

Pro Tools DigiRack Plug-Ins Guide

**Version 6.1 for TDM or LE Systems on Windows or Macintosh
Version 5.1.1 to 5.3x for TDM or LE Systems on Macintosh**

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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

(TDM Systems Only)

TDM 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 on TDM-based Pro Tools 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 addi-

tional DSP cards in your computer. On HD-series systems, use HD Core or HD Process cards; on MIX-series systems, use MIX Core, MIX Farm, or DSP Farm cards. This power-on-demand aspect is a significant advantage of TDM-based systems.

▲ *MIX Farm and DSP Farm cards are not supported on HD-series systems.*

RTAS Plug-Ins

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

Because of this dependence on the CPU or *host processing*, the more RTAS plug-ins you use concurrently in a session, the greater the impact 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.

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

AudioSuite Plug-Ins

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

System Requirements

To use DigiRack plug-ins you need:

- ◆ A Digidesign-approved Pro Tools system
 - or –
- ◆ A third-party software application that supports the Digidesign TDM or AudioSuite plug-in standard


Digidesign can only assure compatibility and provide support for hardware and software it has tested and approved. For a list of Digidesign-qualified computers, operating systems, hard drives, and third-party devices, refer to the latest compatibility information on 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 Session	Choose Save Session from the File menu
Control+N	Hold down the Control key and press the <i>N</i> key
Control-click	Hold down the Control key and click the mouse button
Right-click (Windows)	Click with the right mouse button

The following symbols are used to highlight important information:

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

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

 *Shortcuts show you useful keyboard or mouse shortcuts.*

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

chapter 2

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.

▲ *DigiRack plug-ins versions 5.3.x and earlier are not supported in Pro Tools 6.x.*

Allocating Additional Memory to Pro Tools

(Mac OS 9 Only)

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

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

To allocate additional memory to Pro Tools:

- 1 Start Pro Tools so that it can calculate its basic memory allocation.
- 2 Go to the Finder and choose About This Computer from the Apple menu.
- 3 If you have 3 MB or more of memory available (as indicated in the Largest Unused Block portion of this window), go to step 4. If you have less than 3 MB of free memory, stop here: Do not allocate additional memory to Pro Tools unless you install additional RAM in your computer.
- 4 Quit Pro Tools.
- 5 Open the Pro Tools folder, select the Pro Tools application, and choose Get Info from the Finder's File menu.
- 6 Choose Memory from the Show menu.
- 7 Enter the desired amount of memory *above the minimum requirement* in the Preferred Size field. For example, if the Preferred Size field currently says "30410"K and you want to allocate an additional 3 MB of memory (1 megabyte equals 1,024 kilobytes), enter "33482"K into the Preferred Size field.
- 8 Close the Get Info dialog.

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

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 TDM systems.

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

Processing Power Requirements of TDM and RTAS Plug-Ins


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

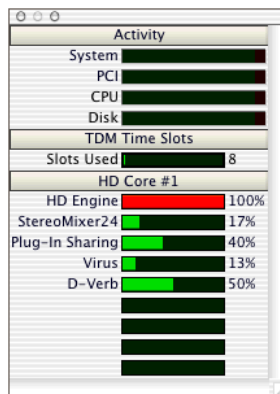
TDM Plug-Ins

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


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

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

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



System Usage window

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

DSP Sharing Between TDM Plug-Ins

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


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 (Windows > Show System Usage) shows how much of your computer’s processing power is currently being used by Pro Tools, and how much is still available.

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


Improving RTAS Plug-In Performance

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

Hardware Buffer Size

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

- ◆ Lower Hardware Buffer Size settings reduce monitoring latency, and are useful when you are recording live input.
- ◆ Higher Hardware Buffer Size settings allow more audio processing and effects, and are useful when mixing and using more RTAS plug-ins.

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

To change the Hardware Buffer Size:

- 1 Choose Setups > Playback Engine.
- 2 From the H/W Buffer Size pop-up menu, select the audio buffer size, in samples.
- 3 Click OK.

CPU Usage Limit

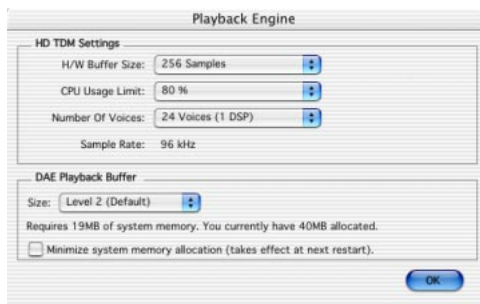
The CPU Usage Limit controls the percentage of CPU resources allocated to Pro Tools host processing tasks.

- ◆ Lower CPU Usage Limit settings limit the effect of Pro Tools processing on other CPU-intensive tasks, such as screen redraws, and are useful when you are experiencing slow system response, or when running other applications at the same time as Pro Tools.
- ◆ Higher CPU Usage Limit settings allocate more processing power to Pro Tools, and are useful for playing back large sessions or using more real-time plug-ins.

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

To change the CPU Usage Limit:

- 1 Choose Setups > Playback Engine.
- 2 From the CPU Usage Limit pop-up menu, select the percentage of CPU processing you want to allocate to Pro Tools.



Playback Engine dialog

- 3 Click OK.

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

Delay in Digital Signal Processing

DSP processing in digital audio systems incurs signal delay of varying amounts. Such delays can vary from as short as 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 with TimeAdjuster

(TDM Systems Only)

You can compensate for TDM plug-in-induced delays by using the TimeAdjuster plug-in. With this plug-in you can 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 53.

Plug-Ins as Inserts

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

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

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

Inserts can be used in two ways:

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

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

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

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

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

Pre-Fader Operation

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

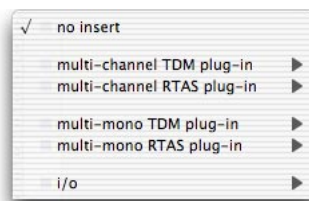
Real-time plug-ins are pre-fader, but post-disk. This means that if you record to disk with a plug-in inserted on the record track, you will hear the effect of the plug-in, but the effect will not be recorded to disk.

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

Mono, Multi-Mono, and Multichannel Plug-Ins

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

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



Plug-in formats

Mono Plug-Ins Used on mono tracks. Some mono plug-ins (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 will analyze and process each channel independently without regard to each other. Controls for all channels are linked by default so that you can adjust them in tandem. You can unlink controls for independent adjustment using the Master Link button. See “Linking and Unlinking Controls on Multi-Mono Plug-ins” on page 14.

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

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

Inserting Plug-Ins on Tracks

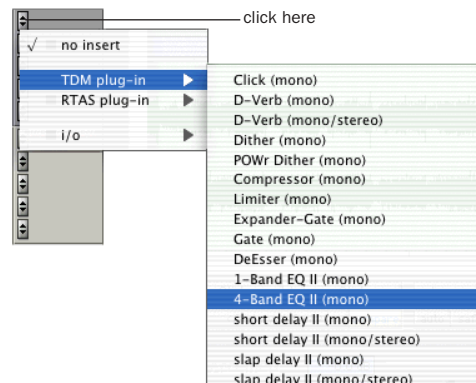
To use a real-time plug-in in a Pro Tools session, insert it on a track. Before doing so, make sure the Inserts View is shown in the Mix window.

To show inserts in the Mix window:

- Choose Display > Mix Window Shows > Inserts View.

To insert a plug-in on a track:

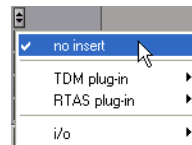
- Click the Insert Selector on the track and select the plug-in that you want to use.



Inserting a plug-in

To remove an insert from a track:

- Click the Insert Selector and select No Insert.



Removing a plug-in

Inserting Plug-Ins During Playback

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

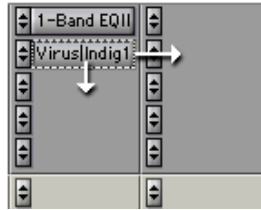
- ◆ Plug-ins cannot be inserted or removed during recording (Pro Tools 6.x only).
- ◆ A plug-in cannot be dragged to a different insert location during playback or recording. Stop playback to do this.
- ◆ Plug-ins that change a track's format (a mono-to-stereo plug-in, for example) cannot be inserted or removed during playback. Stop playback to do this.
- ◆ Plug-ins that contain automation cannot be removed during playback. Stop playback to do this.
- ◆ Playback must be stopped when enabling plug-in controls for automation.
- ◆ Side-chain inputs cannot be created during playback. Stop playback to do this.

Moving and Duplicating Inserts

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

To move an insert:

- Drag the insert to a new insert location.



Moving a plug-in



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

To duplicate an insert:

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

Making Plug-Ins Inactive

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:

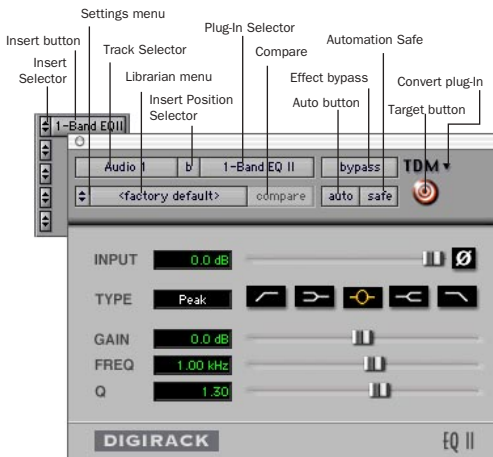
- Control-Start-Alt-Shift-click (Windows) or Command-Control-Option-Shift-click (Macintosh) an Insert button in the position you want to toggle.



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

The Plug-In Window

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



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



Plug-In window (multichannel Compressor shown)

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

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

Librarian Menu Recalls settings files saved in the plug-in's root settings folder or in the current session's Settings folder. See "Using the Librarian" on page 17.

Insert Position Selector Accesses any insert on the current track.

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

Plug-In Selector lets you select any real-time plug-in installed in the Plug-Ins folder.

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

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

Auto lets you enable individual plug-in controls for automation recording. See “Automating Plug-Ins” on page 16.

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

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

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

Phase Invert Button Inverts the phase polarity of the input signal on some plug-ins.

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

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

Link Enable Buttons Let you selectively link the controls of specific channels of a multi-mono plug-in. Each square represents a speaker channel. The Master Link button must be disabled to use the Link Enable buttons. See “Linking and Unlinking Controls on Multi-Mono Plug-ins” on page 14.

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*.

Opening Plug-In Windows

To open a Plug-In window:

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

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

Opening Multiple Plug-In Windows

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

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

To open an additional Plug-In window:

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

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

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

To close all currently open Plug-In windows:

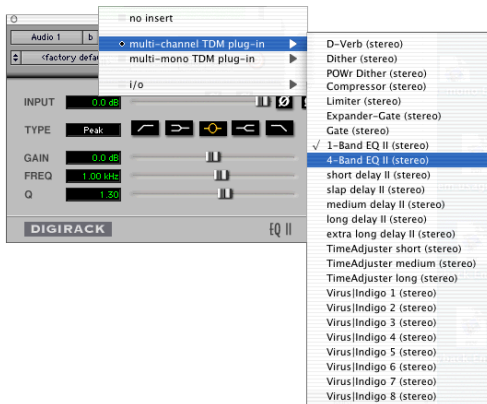
- Alt-click (Windows) or Option-click (Macintosh) the close box of any currently open Plug-In window.

Plug-In Window Controls

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



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

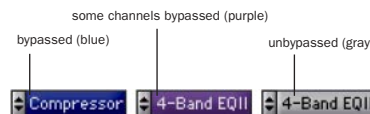
Bypassing Plug-Ins

To bypass a plug-in:

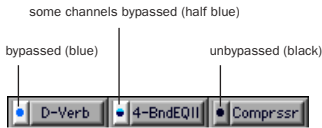
- Click the Plug-In window's Bypass button.
 - or –
- Control-click (Windows) or Command-click (Macintosh) the plug-in's Insert button in the Mix or Edit window.



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 (Pro Tools 6.x) or half blue and half black (Pro Tools 5.x).



Bypass states indicated by color (Pro Tools 6.x)

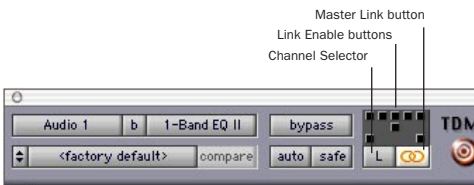


Bypass states indicated by color (Pro Tools 5.3.x)

Linking and Unlinking Controls on Multi-Mono Plug-ins

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

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



Channel Selector and Link controls

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

- Deselect the Master Link button.

To access controls for a specific channel:

- Select the channel from the Channel Selector.

To link the controls of specific channels:

- 1 Deselect the Master Link button if it is not already deselected.
- 2 Click the Link Enable buttons for the channels whose controls you want to link.

Editing Plug-In Controls

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

To adjust a plug-in control:

- 1 Begin audio playback so that you can hear the control changes in real time.
- 2 Adjust the controls of the plug-in for the effect you want.
- 3 Closing the plug-in will save the most recent changes.

Keyboard Shortcuts

- ◆ For finer adjustments, Control-drag (Windows) or Command-drag (Macintosh) the control.
- ◆ To return a control to its default value, Alt-click (Windows) or Option-click (Macintosh) the control.

Keyboard Input for Plug-In Controls

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

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

To adjust controls from a computer keyboard:

- Click in the control text field that you want to edit to activate the field. Type an appropriate 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.

- To increase a value, press the Up Arrow on your keyboard. To decrease a value, press the Down Arrow on your keyboard.
- Press Enter on the numeric keyboard after typing a value to input the value (without leaving the selected control field).
- 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.

Using a Key Input for Side-Chain Processing

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

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

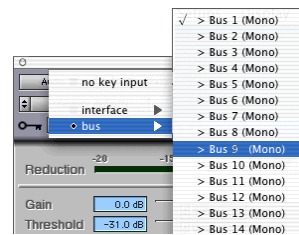
⚠ *RTAS plug-ins do not provide side-chain processing when used on TDM-based systems. If you want to use side-chain processing, use the TDM versions of plug-ins on TDM-based systems.*

Key Input Filters

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

To use a Key Input for side-chain processing:

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




Choosing a Key Input

- 2 Click External Key to activate side-chain processing.
- 3 To hear the audio source you have selected to control side-chain input, click Key Listen.
- 4 To filter the Key Input so that only specific frequencies trigger the plug-in, use the Key HPF and Key LPF controls (if available) to select the desired frequency range.
- 5 Begin playback. The plug-in uses the input or bus that you chose as a Key Input to trigger its effect.
- 6 Adjust the plug-in's Threshold control (if available) to fine-tune Key Input triggering.
- 7 Adjust other controls to achieve the desired effect.

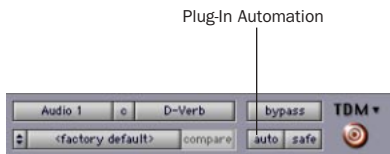
Automating Plug-Ins

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

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.



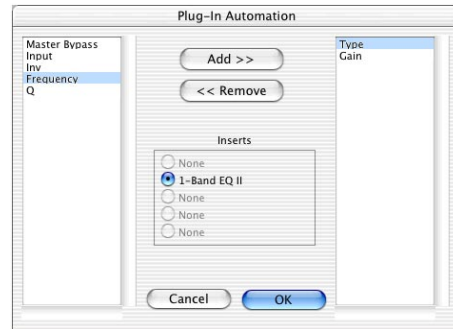
Accessing the Plug-In Automation dialog

– OR –

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


 You can also use this keyboard shortcut to open the Plug-In Automation dialog. 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.

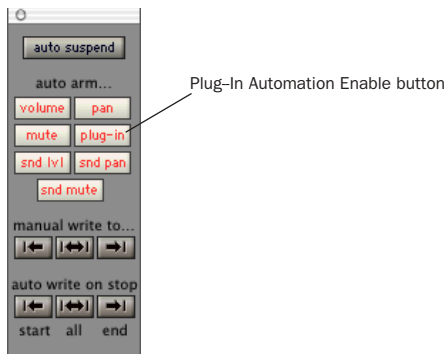
 As an alternative to using the Plug-In Automation dialog, you can enable individual plug-in controls directly from the Plug-In window by Control-Alt-Start-clicking (Windows) or Command-Option-Control-clicking (Macintosh) the control.



Shortcut for enabling plug-in automation

To automate a plug-in:

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



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.



Automation Safe enabled plug-in

To disable plug-in safe mode:

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

For more information on creating and editing automation, refer to the Pro Tools Reference Guide.

Using the Librarian

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

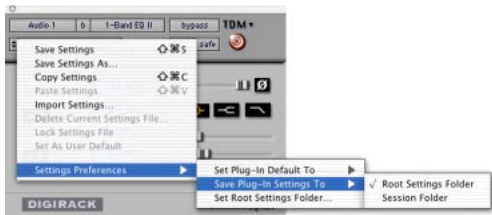


Librarian menu

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

The Settings Menu

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).

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

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

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

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

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

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

Choosing a Destination for Settings

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

To select a destination folder:

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

If you choose Root Settings folder, Pro Tools saves to the default 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 20.*

To save a setting:

- 1 Choose Save Settings from the Settings pop-up menu.
- 2 Type a name and click OK. The setting appears in the Librarian menu.

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

To load a previously saved setting:


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

To import a setting:

- 1 Choose Import Settings from the Settings pop-up menu.
- 2 Locate the settings file you want to import and click Open. Pro Tools loads the setting and copies it to the root destination folder.


To copy a setting:

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

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

To paste a setting:

- 1 Open the destination plug-in.
- 2 Choose Paste Settings from the Settings pop-up menu.

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

To create a custom User Default setting:

- 1 Create and save a setting.
- 2 Choose Set As User Default from the Settings pop-up menu.

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

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

Editing Settings on Unlinked Multi-Mono Plug-ins

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

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

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

Creating Subfolders for Settings

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



Librarian menu with subfolders for settings

To create a settings subfolder:

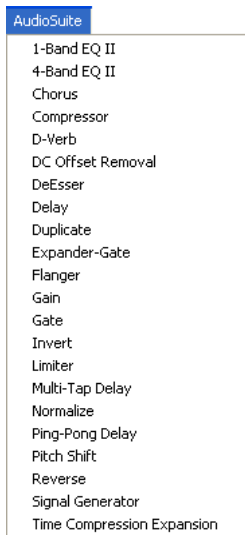
- 1 From the Settings pop-up menu, choose Save Settings.
- 2 Click the New Folder button and type a name for the subfolder.
- 3 Name the setting and click Save. The setting is saved within the subfolder.

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 nondestructively 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.

AudioSuite plug-ins are accessed through the AudioSuite menu.



AudioSuite menu

AudioSuite Plug-Ins

1-Band EQ II Processes audio with high-pass, low-shelf, high-shelf, low-pass or peak EQ filters, in a 1-band module.

4-Band EQ II Processes audio with high-pass, low-shelf, high-shelf, low-pass or peak EQ filters, in a 4-band module.

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 remove 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.

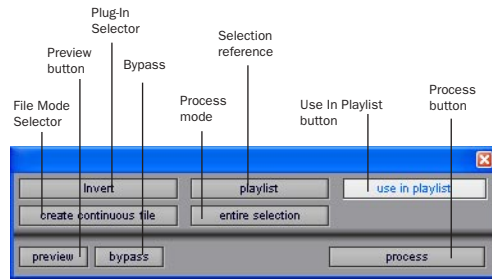
DC Offset Removal Recognizes and removes DC offset from audio files.

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

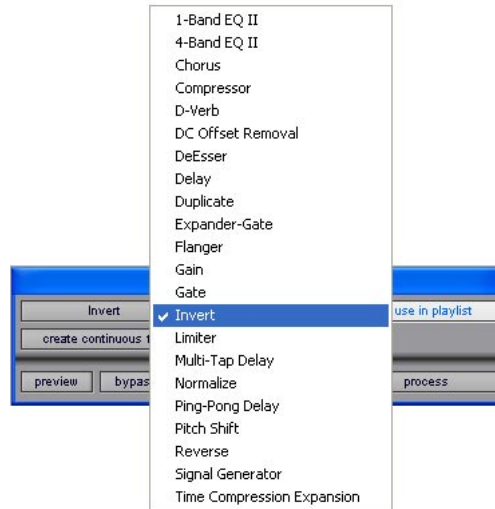
When you choose a plug-in from the AudioSuite menu, the AudioSuite window appears. From this floating window you can access and edit the controls for any non-real-time AudioSuite plug-in.



AudioSuite window

Plug-In Selector

From this menu you can select any AudioSuite plug-in installed in your Plug-Ins folder.

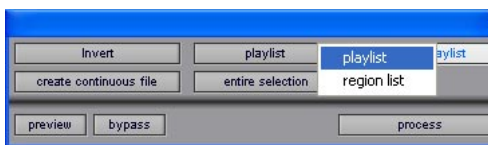


Plug-In Selector menu

Selection Reference

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

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



Selection Reference pop-up menu

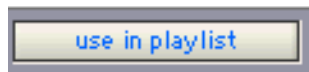
Playlist Applies AudioSuite processing only to regions currently selected in tracks/playlists in the Edit window. Processing will not be applied to regions in the Audio Regions List.

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

⚠ *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.

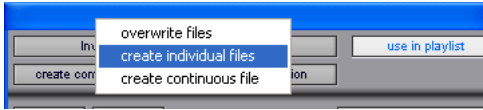
Use In Playlist On, Reference = Region List When Use in Playlist is enabled and the Selection Reference pop-up menu is set to Regions List, all copies of the selected region will be replaced everywhere in the session.

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

From this menu you can select whether or not the AudioSuite plug-in will process the selected audio destructively or non-destructively, and how the original files will be modified.



File Mode pop-up menu

There are three options:

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

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

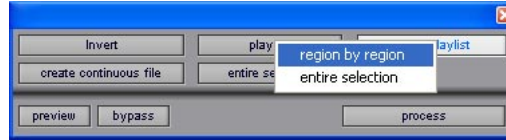
Create Continuous File Processes the selected regions nondestructively and creates a new audio file consisting of the selected regions consolidated into a single, unbroken region. This mode is particularly useful if you are assembling a composite track from multiple takes. This mode is not available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



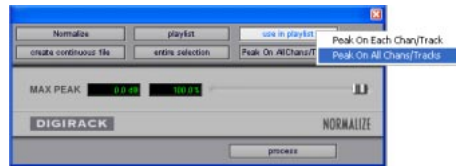
Process Mode Selector

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

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

Chan/Track Process Mode Selector

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



Chan/Track Process Mode Selector pop-up menu


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

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

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 “The AudioSuite Processing Preferences Dialog” on page 27.

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 26.

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 17.

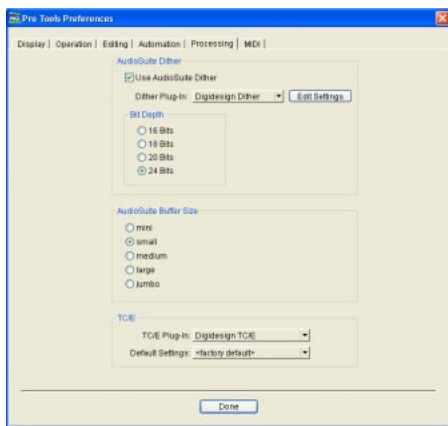
Analyze Button Analyzes a selection without actually processing it. On the DigiRack Gain plug-in, you can use the Analyze feature to determine the maximum peak level on a track at a specific gain value before you process the audio.

Side-chain 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 a Key Input for Side-Chain Processing” on page 15.

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

The AudioSuite Processing Preferences Dialog

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



Processing Preferences page

AudioSuite Dither When selected, dither is applied during certain AudioSuite processes (such as Normalize and Gain), using the dither algorithm selected in the Dither Plug-In pop-up 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 34 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 34.

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


To configure the AudioSuite Buffer Size:

- 1 Choose Setups > Preferences.
- 2 Click Processing.
- 3 Select an AudioSuite Buffer Size.

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

- 4 Click Done.


TC/E 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 regions, partial regions, or selections that are made up of whole and partial regions across one or more tracks.

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


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

Only audio files on locally connected hard drives can be processed with AudioSuite plug-ins. You cannot process audio files on remote hard drives over a network (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 important that you select only those regions, or parts of regions, you actually want to process.

Selecting a region in the Audio Regions List will automatically select it in a track (if it currently resides on one). This is the default behavior for Pro Tools. However, you can change this using the Region Selection Follows Track Selection, or Track Selection Follows Regions List Selection options in the Editing page of the Preferences dialog. 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 (a reverb tail or a delay tap), it is very important that you make a selection that is *longer* than the original source material so that the plug-in can write it into the audio file.

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

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

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

To process audio with an AudioSuite plug-in:

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

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

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

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

- To process the selected region only in the track in which it appears, select **Playlist** from the Selection Reference pop-up. Alternatively, if you want to process the selected region in the Audio Regions list only, choose **Region List** from this pop-up.
- If you want to process and update every occurrence of the selected region throughout your session, enable the **Use In Playlist** button (and also select **Region List** from the Selection Reference pop-up). Alternatively, if you do not want to update every occurrence of the selected region, deselect the **Use In Playlist** button.
- To configure the plug-in for destructive processing, select **Overwrite Files** from the File Mode pop-up menu. This will overwrite and permanently modify the original source audio files.

- Alternatively, to configure the plug-in for nondestructive processing, select **Create Individual Files** from the File Mode pop-up menu. This will create new audio files that have been processed with the AudioSuite Plug-in and leave the original source audio files untouched.
- If you have selected multiple regions for processing and want to create a new file that connects and consolidates all of these regions together, select **Create Continuous File** from the File Mode pop-up menu.

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

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

Using AudioSuite Plug-Ins in Stereo

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

- ◆ To process a mono track and obtain a stereo result, select the desired track or region plus an empty track or region. Then set the plug-in to **Stereo** mode and select the **Sum Inputs** button (if present) to center the dry signal. When you process the audio, the result will be two tracks or regions that represent the right and left channels of the processed audio. You should then pan these tracks hard right and hard left in your mix.
- ◆ If you set a plug-in to **Stereo** mode, then select an odd number of Pro Tools tracks for processing (as opposed to an even number), the plug-in will process the selected tracks in pairs, in stereo. However, the last odd, unpaired track will be

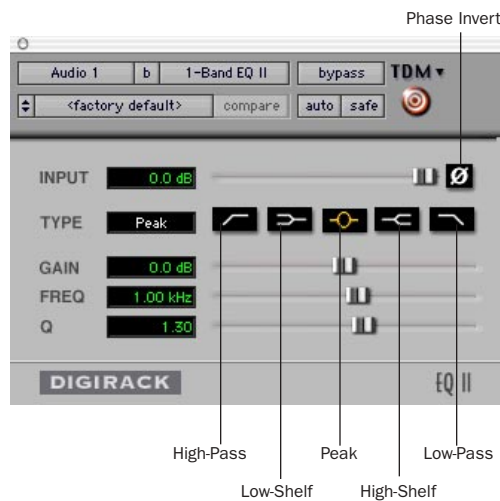
processed as mono, using the left channel settings of the stereo plug-in. If you want the last track to be processed in stereo, you must select an additional track to pair it with—an empty one if necessary.

chapter 5

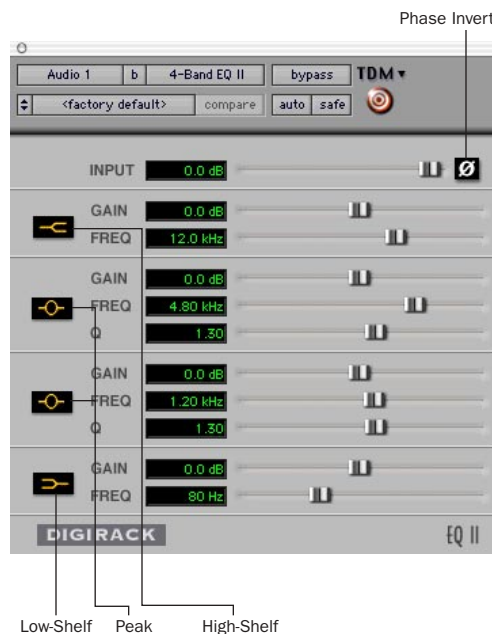
DigiRack Real-Time TDM and RTAS Plug-Ins

EQ II

The EQ II plug-in provides a 1-Band or 4-Band EQ for adjusting the frequency spectrum of an audio selection. You can configure the 1-Band EQ as a high-pass, low-shelf, high-shelf, low-pass, or peak EQ. The 4-Band EQ provides one high-shelf, one low-shelf, and two peak EQs.



1-Band EQ plug-in



4-Band EQ plug-in

Adjusting EQ

A useful way to audition an EQ is to increase or decrease its gain several dB then sweep the frequency up or down until you hear its effect. You can then make more precise adjustments to the settings.

EQ II Controls

Input Controls the input gain of the EQ, letting you prevent clipping. Since Pro Tools inserts are pre-fader, plug-ins used as inserts can cause clipping if their levels are boosted excessively.

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

Type Selects an EQ type (high-pass, low-shelf, peak, high-shelf or low-pass).

Freq Designates the center of the frequency region to be cut or boosted.

Gain Controls the amount that the selected frequencies are cut or boosted.

Q Sets the bandwidth of the Peak filter. Higher values represent narrower bandwidths. Lower numbers represent wider bandwidths. This control is only available on the Peak EQ.

Bypass Bypasses the EQ. The 4-Band EQ has individual Bypass buttons for each band (the black buttons with appropriate EQ curve icons), as well as the standard all-band Bypass button at the top of the Inserts/Sends Editor window.



Bypass EQ band button on the 4-band EQ

High-Pass Filter Attenuates all frequencies below the selected cutoff frequency setting at a rate of 12 dB per octave while letting all others above pass through. For this reason, no gain control is available for this filter. High-Pass filters can be useful for removing low frequency rumble or thinning out the lower end of a sound for special effects such as simulating a telephone effect.

Low-Shelf EQ Produces a lift or a cut at and below the specified frequency.

Peak EQ Boosts or cuts only those frequencies centered around the selected center frequency. The Q button sets the bandwidth of the Peak filter. This determines the width of the filter's overall slope—from a broad bell shape to a narrow notch. Broad curves tend to be most useful for musical applications. Narrow curves are useful for special purpose processing such as hum removal. Higher values represent narrower bandwidths. Lower numbers represent wider bandwidths.

High-Shelf EQ Produces a lift or a cut at and above the specified frequency.

Low-Pass filter Attenuates all frequencies above the selected cutoff frequency setting at a rate of 12 dB per octave while letting all others below pass through. For this reason, no gain control is available for this filter.

Click

(Pro Tools 6.x)

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 accented 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 MIDI > 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 MIDI > Click Options 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

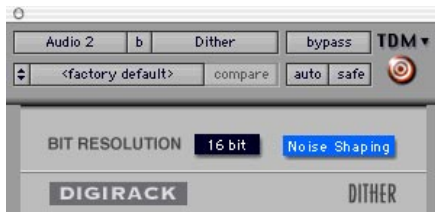
 *Refer to the Pro Tools Reference Guide for more information on configuring Click options.*

- 5 Begin playback. A click is generated according to the tempo and meter of the current session and the settings in the Click 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 DigiRack POW-r Dither plug-in. See “POW-r Dither” on page 35.




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 converters 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. As noted previously, 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 Shape

This button engages or disengages Noise Shaping. Noise shaping is on when the button is highlighted in blue. See “Noise Shaping” on page 34 for more information.

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

POW-r Dither

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 34.



POW-r Dither plug-in

⚠ *The POW-r Dither plug-in does not run on third party software 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.

It is recommended that you 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 during mixdown.*

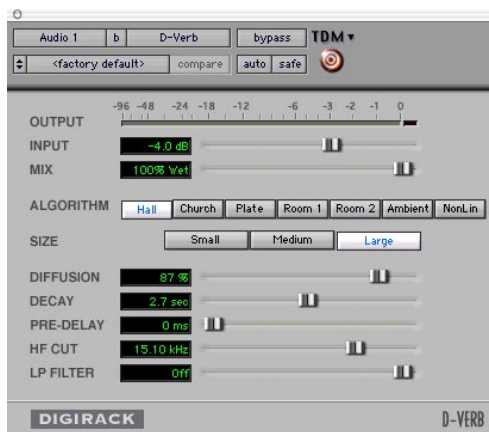
⚠ *On DSP Farm-equipped MIX-series systems, POW-r Dither requires more than one DSP chip when used on LCR and beyond multichannel tracks. On MIX Farm equipped MIX-series systems, POW-r Dither requires more than one DSP chip when used on 7.0-format and beyond multichannel tracks.*

D-Verb

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

A *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 light will 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.

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 rela-

tively 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

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 settings created in version 5.x and later of Pro Tools are not compatible with earlier versions. Saving a session in a pre- 5.x format will cause Compressor settings to be lost.*

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.

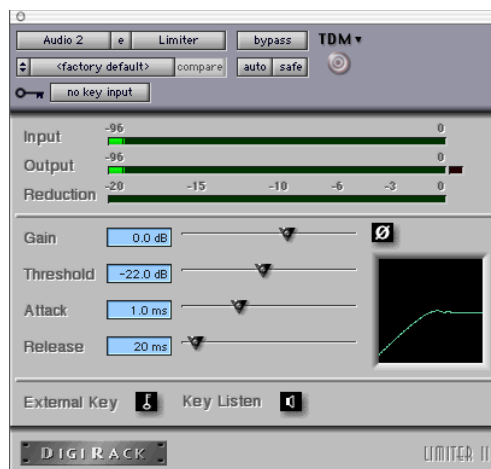
LFE Enable Enables plug-in processing of the LFE (low frequency effects) channel on a multi-channel track formatted for 5.1, 6.1, or 7.1 surround formats on some plug-ins. To disable LFE processing, deselect this button. For more information on the LFE channel, refer to the *Pro Tools Reference Guide*.

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 a Key Input for Side-Chain Processing" on page 15.

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 audits 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 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 limiting. See “Using a Key Input for Side-Chain Processing” on page 15.

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.

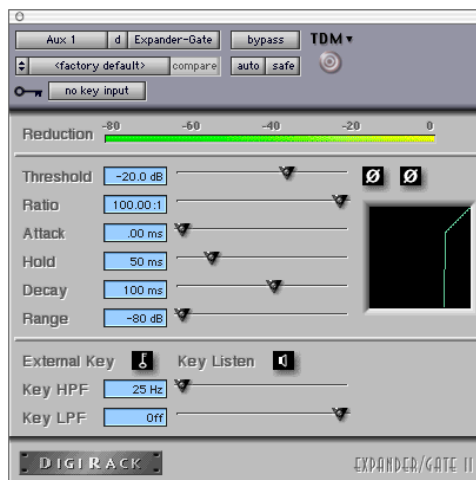
Graph Displays the response curve set by the Gate's Threshold and Range settings. As you adjust these controls, refer to the graph to see how the shape of this curve changes.

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 a Key Input for Side-Chain Processing” on page 15.

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 plugin 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 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*.

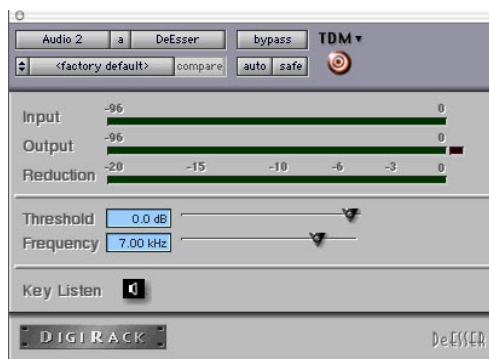
External Key Enables and disables side-chain processing. Side-chain processing allows the Expander/Gate to listen to the varying amplitude of a separate reference track and use it to trigger expansion. See “Using a Key Input for Side-Chain Processing” on page 15.

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.*

⚠ *Expander/Gate settings created in version 5.x and later of Pro Tools are not compatible with earlier versions. Saving a session in a pre- 5.x format will cause settings to be lost.*

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 and Mod Delay II

The DigiRack Mod Delay and Mod Delay II plugins provide time delay-based effects such as echo, slap echo, doubling, chorusing, and flanging. Mod Delay II improves on Mod Delay by supporting tempo controls and sample rates up to 192 kHz.

⚠ *It is recommended that you only use Mod Delay II for all new sessions. Use Mod Delay only for compatibility with existing sessions. If you do not need to use Mod Delay, move it from the Plug-Ins folder into the Plug-Ins (Unused) folder. Mod Delay settings can be imported into Mod Delay II.*

Mod Delay

(Pro Tools 5.3 and Earlier Only)

There are four different Mod Delays, each of which is capable of a different maximum delay time:

- The Short Delay provides 1024 samples of delay (23.2 milliseconds at 44.1 kHz or 21.3 ms at 48 kHz).
- The Slap Delay provides 7186 samples of delay (162 ms at 44.1 kHz or 149 ms at 48 kHz).
- The Medium Delay provides 16384 samples of delay (371 ms at 44.1 kHz or 341 ms at 48 kHz).

- The Long Delay provides 162474 samples of delay (3.68 seconds at 44.1 kHz or 3.38 seconds at 48 kHz).

⚠ *The Long Delay TDM plug-in is only available on systems equipped with HD-series or MIX-series systems.*



Mod Delay plug-in (Pro Tools 5.3)

⚠ *Mod Delay is not supported in sessions at sample rates greater than 48 kHz.*

Multiple Mod Delay plug-ins can be cascaded on inserts one into another to achieve even longer delay times. For each delay, set the Feedback to 0% and the Mix to 100% to get a single long delay (the sum of all the delay times). Adjust the various controls of each of the delays independently to create complex delay patterns.

The Mod Delays provide a Mono In/Stereo Out option. Choosing a delay in the Mono In/Stereo Out format “stereoizes” the output of a mono track. A track created in this way will then have stereo pan faders for controlling each channel of the stereo signal. As explained previously, any inserts that occur on a track after a mono-to-stereo insert will automatically become stereo as well.

Mod Delay 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.

⚠ *The Mix control is fixed at 100% in the TDM version of the mono-to-stereo and stereo Mod Delay Long Delay on Pro Tools|HD-series systems.*

LPF (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 attenuated.

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.

Automation

Mod Delay cannot have automation data copied and pasted to the adjacent left or right audio channel. This occurs even when the automation data is to or from the same control. For example, Feedback Left automation data cannot be copied to Feedback Right.

To apply automation data to both the left and right channels of the Mod Delay stereo outputs, use multi-mono plug-ins in place of the stereo or mono-to-stereo Mod Delay. Multi-mono plug-in automation is linked, providing the same automation data to all linked channels.

Mod Delay II

There are five different Mod Delay II plug-ins, each of which is capable of a different maximum delay time:

- 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 plug-in (Long Delay shown)

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.

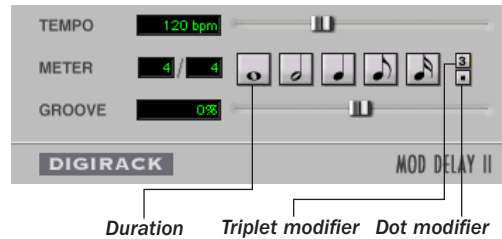
Tempo Selects the desired tempo in beats per minute (bpm). This setting is independent of Pro Tools’ tempo. When a specific Duration is selected (see “Duration” below), moving this control will affect the Delay setting. Likewise, the range of both controls will be limited to the maximum available delay with the currently selected Duration. To enter very short or long delays it may be necessary to deselect all Duration buttons.

 Use *Edit > Identify Beat or 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, eighth 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 eighth 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 value 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 the 96 I/O or 192 I/O and other elements of your studio. Refer to the *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 signal generator produces a tone as soon as it is inserted on a track. To mute the Signal Generator, use the Bypass button.


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.

 The pink noise generator in the Signal Generator plug-in is calibrated at RMS level, rather than peak level.

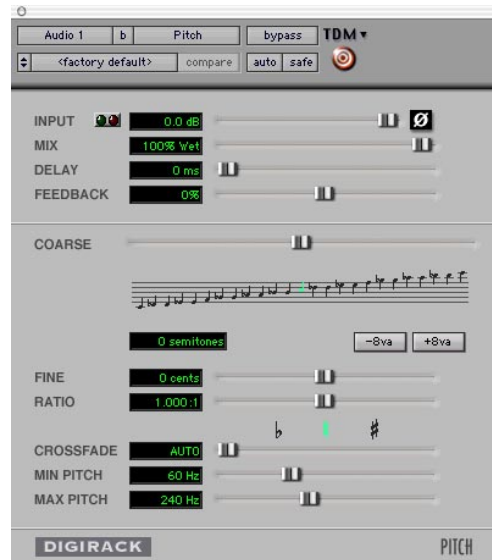
Signal These buttons select the waveform. Choices are sine, square, sawtooth, triangle, white noise, and pink noise.

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

Pitch

(TDM Systems 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 cross-fades 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.

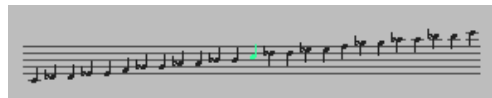
Feedback Controls the amount and type of feedback (positive or negative) applied from the output of the delay portion of the Pitch plug-in back into its input. It also controls the number of repetitions of the delayed signal. You can use it to produce effects that spiral up or down in pitch, with each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

(TDM Systems Only)

The TimeAdjuster plug-in provides three functions:

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



TimeAdjuster plug-in

The TimeAdjuster plug-in is the most efficient way to compensate for DSP 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 Delay Compensation” on page 54).

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 Delay Compensation

DSP 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 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 TDM plug-ins display their delay values in the Mix window's 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 In the Mix window, 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

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, 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 in TDM systems, refer to the Pro Tools Reference Guide.*

Trim

The Trim plug-in can be used to attenuate an audio signal from $-\infty$ (Infinity) dB to +6 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 $-\infty$ dB to +6 dB of gain adjustment.

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

Mute Mutes the signal output.

chapter 6

DigiRack AudioSuite Plug-Ins

EQ II

The EQ II plug-in provides a 1-Band or 4-Band EQ for adjusting the frequency spectrum of audio material. You can configure the 1-Band EQ as a high-pass, low-shelf, high-shelf, low-pass, or peak EQ. The 4-Band EQ provides one high-shelf, one low-shelf, and two peak EQs.




1-Band EQ II AudioSuite plug-in

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



4-Band EQ II AudioSuite plug-in

 For information on the individual controls for these plug-ins, see “EQ II” on page 31.

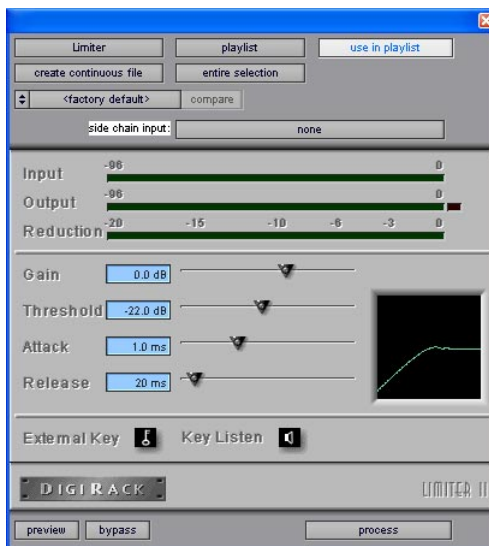
Dynamics II

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

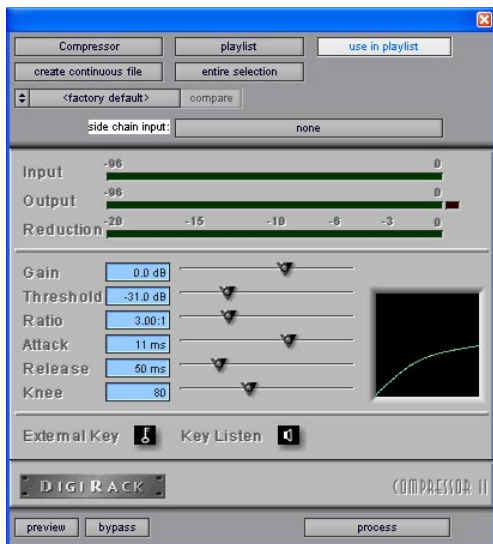
- Compressor II
- Limiter II
- Gate II
- Expander/Gate II
- DeEsser

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

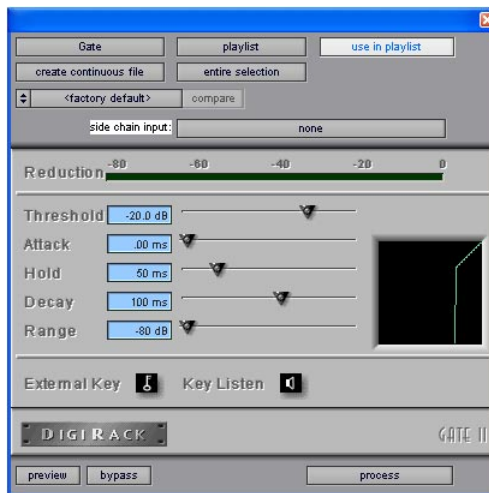
☞ *For information on the individual controls for these plug-ins, see “Dynamics II” on page 39.*



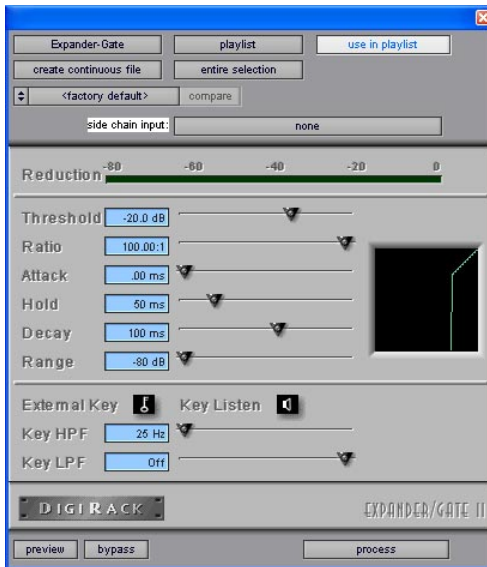
Limiter II AudioSuite plug-in



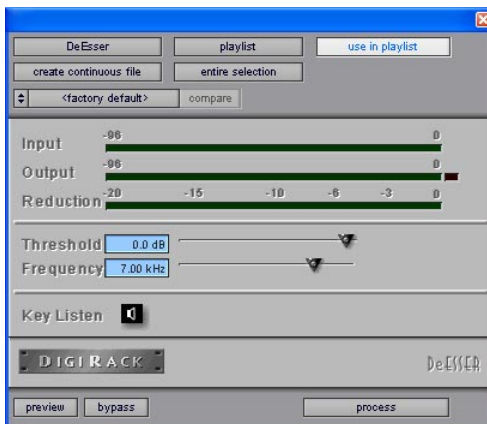
Compressor II AudioSuite plug-in



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.

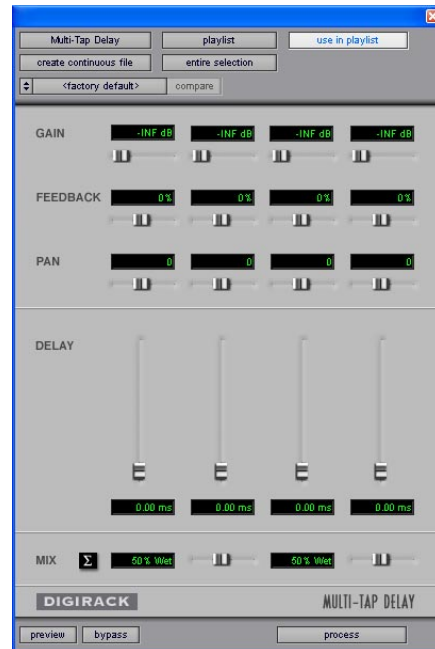
Feedback Controls the amount of feedback applied from the output of the delayed signal back into its input. Negative settings provide a more intense effect.

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.

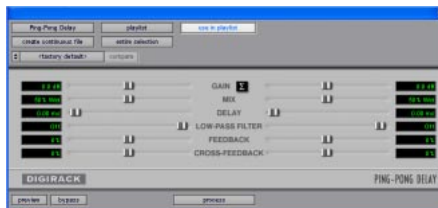
Select only the original material, without leaving additional space at the end, will produce a delay that occurs after the end of the region will be cut off.

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

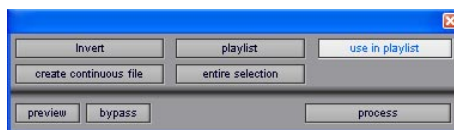
Because Ping-Pong Delay adds additional material (a delay tap) to the end of selected audio, make a selection that is longer than the original source material to allow the delay tap to be written into the end of the audio file.

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.

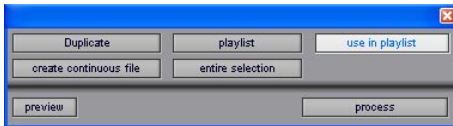
This process is useful for altering the phase or *polarity* relationship of tracks. The Invert plug-in is useful during mixing for modifying frequency response between source tracks recorded with multiple microphones. You can also use it to correct audio recorded out of phase with an incorrectly wired 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 48.

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” on page 46.

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.

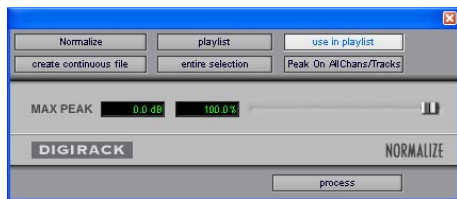
Selecting only the original material, without leaving additional space at the end, will cause delay taps that occur after the end of the region to be cut off.

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.

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.



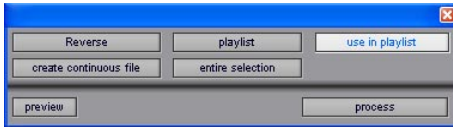
Gain plug-in

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

Reverse

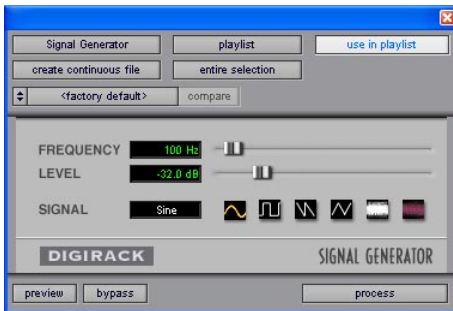
The Reverse plug-in replaces the audio with a reversed version of the selection.



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.

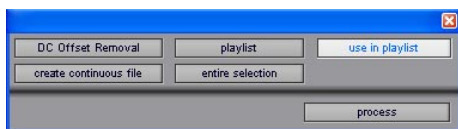
 Select the Create Continuous File option for greater flexibility in making audio selections for use with the Signal Generator plug-in.

 For information on the individual controls for the Signal Generator plug-in, see “” on page 49.

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.

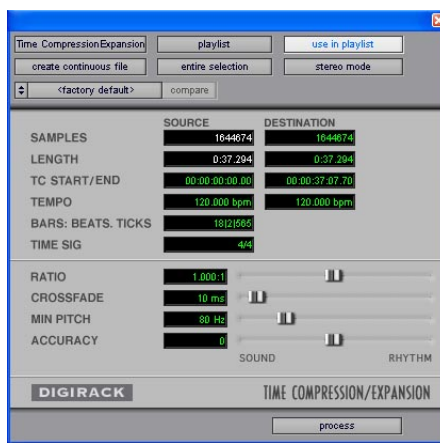
To check for DC offset, find a silent section in the audio material. If DC offset is present, a near-vertical fade-in with a constant or steady-state offset from zero will appear in the waveform. Use the DC Offset Removal plug-in to remove it.



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.

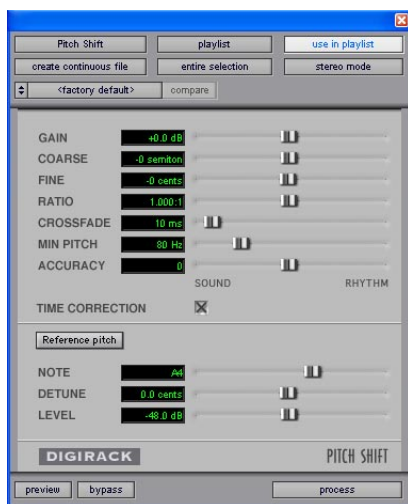
Accuracy Prioritizes the processing resources allocated to audio quality (Sound) or timing (Rhythm). Moving the slider towards "Sound" generally results in better sonic quality and fewer audio artifacts. Moving the slider towards "Rhythm" puts the emphasis on keeping the tempo consistent.

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 bass guitar or audio material with a low frequency range. Use higher values when processing higher frequency audio material.

Accuracy Prioritizes 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. This feature has the following controls:

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 6.1 Only)

Pro Tools supports ReWire version 2.0 technology developed by Propellerheads Software. ReWire is available as a DigiRack RTAS plug-in for Pro Tools TDM and LE systems on Windows XP and Mac OS X.

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 TDM or Pro Tools LE software version 6.1 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 on TDM Systems

On TDM systems, the ReWire RTAS plug-in must be inserted on an audio 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.

 *The standard Hardware Buffer size of 512 samples is strongly recommended for using ReWire in sessions with sample rates above 48 kHz on Pro Tools|HD systems.*

ReWire and Track Count on Pro Tools LE Systems

On Pro Tools LE-based systems, 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/compto).

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 File > New Track and specify one audio track (TDM systems) or one Auxiliary Input track (LE systems), then click Create.

3 Choose File > New Track and specify one MIDI track, then click Create.

4 In the Mix window, click the Insert Selector on the audio or Auxiliary Input track and assign the ReWire RTAS client plug-in to the track insert.

▲ *On TDM systems, the ReWire RTAS plug-in can only be inserted on an audio track. On LE systems, the ReWire plug-in can be inserted on an audio track or an Auxiliary Input.*

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

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

6 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.*

7 In the Mix window, click the MIDI track's Device/Channel Selector and select the ReWire client application. Some ReWire clients (such as Reason) may list multiple devices. If so, choose the device that you want.

8 Select MIDI > Thru and record enable the MIDI track. Play some notes on your MIDI controller to trigger the client application.

9 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 7 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.*

chapter 8

Using DirectConnect (Mac OS 9 Only)

DirectConnect is a Macintosh-only Pro Tools plug-in that allows up to 32 separate audio channel outputs from compatible host-based applications, such as software-based synthesizers or samplers, to be independently routed, recorded, processed and mixed within the Pro Tools environment.

⚠ *DirectConnect is only supported on Mac OS 9. It is not supported by Pro Tools 6.x on Mac OS X. For information on using host-based applications such as software synthesizers or samplers with Pro Tools 6.x on Windows XP or Mac OS X, see Chapter 7, “Using Digi Re-Wire (Pro Tools 6.1 Only).”*

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

- Assign them to busses for plug-in processing
- Automate volume, pan, and plug-in controls
- Bounce To Disk
- Take advantage of the audio outputs of your Digidesign audio interfaces

📖 *If you have downloaded DirectConnect from the Digidesign Web site, see “Installing DirectConnect” on page 77 for installation instructions.*

DirectConnect Requirements

To use DirectConnect, you will need:

- A Pro Tools HD-series or MIX-series system, or Pro Tools LE software running on a Digidesign-approved Pro Tools configuration
- Digidesign Soft SampleCell or Third-party client software compatible with DirectConnect, such as:
 - *Studio9000* from Koblo
 - *Retro AS-1*, *Unity DS-1*, *Voodoo*, or *Phrazer* from Bitheadz
 - *Absynth*, *B-4*, *Battery*, *Dynamo*, *FM7*, or *Reaktor* from Native Instruments
 - *Spark XL* from TC|Works
 - *Alkali* from Audio Genetics


⚠ *Third-party client software must support and be configured for the same sample rate as the session to use with DirectConnect. For example, third-party client software that does not support sample rates beyond 48 kHz cannot be used in 96 kHz Pro Tools sessions.*

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

DirectConnect and Track Count on TDM Systems

Each channel of audio transmitted through DirectConnect uses the same amount of resources as a single audio track in Pro Tools. Consequently, you can only use a total combination of disk tracks and DirectConnect audio streams that does not exceed the maximum number of possible voices for your system. For example, if you are playing 40 disk tracks in a 96 kHz/24-bit session on a Pro Tools|HD 1 core system, that will leave 8 channels of audio (8 mono, or 4 stereo) that can be used with DirectConnect.

Using DirectConnect 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 total CPU usage in the Pro Tools System Usage window, making sure to not overtax your system.

 *The standard sample buffer size of 512 samples is strongly recommended for using DirectConnect in sessions with sample rates above 48 kHz on Pro Tools|HD systems.*

The number of available audio streams at different sample rates are:

- 32 streams at 44.1/48 kHz
- 16 streams at 88.2/96 kHz
- 6 streams at 176.4/192 kHz

If you attempt to use more channels than are available with DirectConnect, you may encounter a DAE error -6042 (PCI bus too busy).

DirectConnect and Track Count on Pro Tools LE Systems


On Pro Tools LE-based systems, performance is determined by several factors, including host CPU speed, available memory, and buffer settings. Digidesign cannot guarantee 32 simultaneous audio channel outputs with DirectConnect on all computer configurations.

For the latest information on recommended CPUs and system configurations, check the Digidesign Web site.

DirectConnect and DSP Resources (TDM Systems Only)

Similar to some plug-ins, DirectConnect requires a specific type of DSP chip (SRAM) to function; there are nine of these particular DSP chips on an HD Core card.

When DirectConnect is in use, it takes up one of these DSPs and makes it unavailable for any other plug-ins.


 *You can use the Show DSP Usage window (available from the Pro Tools Windows menu) to see how much DSP is available on your system and how it is currently being used. For details, see the Pro Tools Reference Guide.*


Sample Rate and Latency

On Pro Tools HD-series systems, DirectConnect supports sample rates up to 192 kHz. Since the buffer size setting for DirectConnect is fixed, latency will decrease as the session sample rate is increased. This may cause synchronization problems with MIDI tracks if MIDI track offsets have been configured for a different sample rate.

DirectConnect and Other DAE Applications

DirectConnect will work with DAE-compatible applications other than Pro Tools, including sequencers from other third-party software companies.

 *You must have completed a full install of Pro Tools before you can use DirectConnect with third-party sequencer software.*

 *Contact your sequencer manufacturer for information on compatibility with DirectConnect, and DAE and DigiSystem INIT 5.3 or later.*


Installing DirectConnect

To use DirectConnect, you must install the *plug-in description* file for your software synth or sampler. This file must reside in the Plug-Ins folder inside the DAE Folder. Some DirectConnect client applications will do this for you during installation, such as Soft SampleCell.

DirectConnect also requires the following files, that are installed automatically with the Pro Tools software:

DirectConnect This plug-in file resides in the Plug-Ins folder inside the DAE Folder (that is in the System Folder).

Digidesign StreamManager This file resides in the Extensions folder inside the System Folder.

 *Contact the manufacturer to obtain the plug-in description file for your DirectConnect-compatible software if it is not installed when you install the DirectConnect client application.*

Using DirectConnect

Several DirectConnect client applications have OMS drivers, such as Soft SampleCell. Once you have installed a DirectConnect client application, be sure to create a new OMS Setup that includes the DirectConnect client application's OMS driver, or update your existing OMS Setup to include the OMS driver of the DirectConnect client application.

To update your OMS Current Studio Setup:

- 1 Launch the OMS Setup application.
- 2 Verify that your Current Studio Setup is open.
- 3 Choose Studio > MIDI Cards & Interfaces and follow the on-screen instructions.
- 4 Your DirectConnect client application's OMS driver should appear in your Studio Setup. If not, you will need to use the OMS IAC driver (see "Configuring the IAC Driver" on page 77).
- 5 Choose File > Save.
- 6 Quit OMS Setup.

DirectConnect, OMS, and IAC

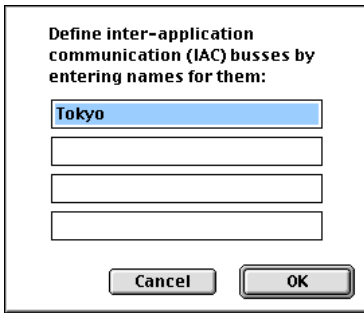
DirectConnect client applications that do not have an OMS driver, such as Studio9000 by Koblo, will require the OMS IAC Driver. Pro Tools will be able to send MIDI data to the software synth or sampler using the OMS IAC bus.

Configuring the IAC Driver

To control a software synth that does not have its own OMS driver in Pro Tools, you will need to first configure the OMS IAC driver.

To configure the OMS IAC driver:

- 1 If it is not already installed, install the OMS IAC driver. To do this, perform a Custom Install from the OMS Installer.
- 2 Once the IAC driver is installed and you have restarted your computer, you will need to update your OMS Studio setup. In the OMS Setup application, choose Studio > MIDI Cards and Interfaces.
- 3 Once the IAC driver appears in your Studio Setup, double-click its icon and rename one of the busses as desired.



Renaming an IAC bus

For example, if you are connecting to the Tokyo software from Koblo, rename the bus “Tokyo.”

- 4 Click OK.
- 5 Choose File > Save and then close OMS Setup.

Configuring the Software Synth

Next you’ll need to route the software synthesizer’s audio output to DirectConnect, and assign its MIDI control to the IAC bus. Following is an example for configuring the Tokyo software for Koblo synths.

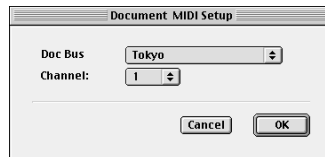
To configure the Tokyo application for DirectConnect:

- 1 Launch the Tokyo application and select the desired synth from the Synths menu.
- 2 Choose File > Select Drivers. Set the Audio Driver to DirectConnect and the MIDI Driver to OMS.
- 3 Click OK.



Drivers dialog in Tokyo application

- 4 Choose Options > MIDI Setup. Set the Doc Bus to Tokyo and the Channel to 1.
- 5 Click OK.




MIDI Setup dialog in Tokyo application

The Koblo Tokyo software synth is now ready for use with Pro Tools and DirectConnect.

Configuring Pro Tools

Configure Pro Tools to connect to your software synth through DirectConnect and OMS.

 *Some third-party products provide both mono and stereo outputs, while others use mono or stereo outputs exclusively. If you cannot instance a DirectConnect plug-in on a mono track, it is likely that it can only be used on stereo tracks. Conversely, if you cannot instance a DirectConnect plug-in on a stereo track, it is likely that it can only be used on a mono track.*

To configure Pro Tools for DirectConnect:

- 1 Launch Pro Tools and open a new or previously saved session.
- 2 Choose File > New Track and specify one Auxiliary Input (stereo) track, then click Create.
- 3 Choose File > New Track and specify one MIDI Track, then click Create.
- 4 In the Mix window, click on the new MIDI track's Device/Channel Selector and select the DirectConnect client application (for example, Soft SampleCell) on channel 1.
- 5 In the Mix window, assign the DirectConnect plug-in to the Auxiliary Input track. Some DirectConnect client applications will launch automatically in the background (such as Soft SampleCell). If your DirectConnect client application does not support this auto-launch feature, open it manually.
- 6 Configure your DirectConnect client application to play the sounds you want. For example, if you are using Soft SampleCell, load the samples, instrument, or bank you want to use.

- 7 In the DirectConnect plug-in window in Pro Tools, set the output to SSC Stereo Out 1-2.



DirectConnect plug-in window (Pro Tools 5.3)

- 8 Select MIDI > Thru and record-enable the MIDI track. Play some notes on your MIDI controller to trigger Soft SampleCell.
- 9 Assign additional plug-ins as desired for the Auxiliary Input track.

Automating DirectConnect


DirectConnect 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.*

Recalling DirectConnect States

Provided your client software supports it, DirectConnect lets you recall the patch settings for your software synth or sampler. Once you save and re-open your Pro Tools session, DirectConnect will automatically recall the settings previously used. Pro Tools will even launch, if necessary, your client software when opening a session that uses DirectConnect.

▲ *If your client software is not running and you open a DirectConnect session, you may need to wait while Pro Tools locates and launches your client software.*

appendix a

DSP Requirements for TDM Plug-Ins

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

There are a total of 9 DSP chips on a Pro Tools|HD card. Though there are a total of 6 DSP chips on a Pro Tools|24 MIX card, certain plug-ins, such as the Mod Delays, require a specific type of DSP chip to function. The tables indicate the number of compatible chips per card for each plug-in.

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.

▲ *The tables that follow show theoretical maximum performance when no other plug-ins are sharing available DSP resources. You will typically use more than one type of plug-in simultaneously.*

Table 2. Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD card at different sample rates (mono and stereo).

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz	
	Mono	Stereo	Mono	Stereo	Mono	Stereo
EQ II (1-band)	119	59	54	27	20	10
EQ II (4-band)	36	18	16	8	6	3
Compressor II	26	17	11	8	4	3
Click	3	n/a	3	n/a	1	n/a
De-esser	20	10*	9	4*	3	1*
DigiRack Pitch	1	1	1	1	1	n/a
Limiter II	26	17	11	8	4	3
Gate II	59	41	27	18	11	7
Expander/Gate II	23	20	10	9	4	3
Mod Delay I (short)	43	21	n/a	n/a	n/a	n/a
Mod Delay I (slap)	32	16	n/a	n/a	n/a	n/a
Mod Delay I (medium)	16	8	n/a	n/a	n/a	n/a
Mod Delay I (long)	8	4	n/a	n/a	n/a	n/a
Mod Delay II (short)	43	21	19	9	7	3
Mod Delay II (slap)	32	16	16	8	7	3
Mod Delay II (medium)	16	8	8	4	4	2
Mod Delay II (long)	8	4	4	2	2	1
Mod Delay II (extra long)	2	1	1	n/a	n/a	n/a
Dither	37	18	17	8	7	3
POW-r Dither	9	4	4	2	1	n/a
Time Adjuster (short)	91	45	41	20	15	7

Table 2. (Continued) Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD card at different sample rates (mono and stereo).

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz	
Plug-In	Mono	Stereo	Mono	Stereo	Mono	Stereo
Time Adjuster (medium)	91	45	41	20	15	7
Time Adjuster (long)	32	16	32	16	15	7
Trim	113	59*	54	27*	20	10*
Signal Generator	27	12*	11	5*	4	2*

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

Table 3. Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD 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
EQ II (1-band)	39	29	28	19	17	14	9
EQ II (4-band)	12	9	7	6	5	4	9
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
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
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

Table 3. (Continued) Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools HD 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
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.*

*** Instances are roughly half at 96 kHz and a quarter at 192 kHz.*

Table 4. Maximum instances of real-time TDM plug-ins per DSP chip for a Pro Tools |24 MIX card

Plug-In	Mono	Stereo	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP chips per Mix card
EQ II (1-band)	63	31	21	15	12	10	9	7	6
EQ II (4-band)	28	14	9	7	5	4	4	3	6
Compressor II	20	13	8	7	6	5	4	4	6
De-esser	16	8	5	4	3	2	2	2	6
DigiRack Pitch	1	1	partial	partial	partial	partial	partial	partial	3
Limiter II	20	13	8	7	6	5	4	4	6
Gate II	31	21	10	7	6	5	4	3	6
Expander/ Gate II	19	15	11	10	9	8	7	7	6
Mod Delay (short)	33	16	3	2	2	1	1	1	5
Mod Delay (slap)	16	8	3	2	2	1	1	1	5
Mod Delay (medium)	8	4	3	2	2	1	1	1	5
Mod Delay (long)	4	2	1	1	–	–	–	–	5
Mod Delay (extra long)	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a
Dither	28	14	9	7	6	4	4	3	6
POW-r Dither	8	4	2	2	1	1	1	1	6
Time Adjuster	63	31	21	15	12	10	9	7	3
Trim	63	31	21	15	12	10	9	7	6
Signal Generator	32	16	10	8	6	5	4	4	6

Table 5. Maximum instances of real-time TDM plug-ins per DSP chip for a DSP Farm card

Plug-In	Mono	Stereo	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP chips per DSP Farm card
EQ II (1-band)	25	12	8	6	5	4	3	3	4
EQ II (4-band)	10	4	3	2	1	1	1	1	4
Compressor II	8	5	3	2	2	2	1	1	4
De-esser	6	2	1	1	–	–	–	–	4
DigiRack Pitch	1	1	partial	partial	partial	partial	partial	partial	1
Limiter II	8	5	3	2	2	2	1	1	4
Gate II	16	10	5	4	3	2	2	2	4
Expander/ Gate II	8	6	3	3	2	2	2	2	4
Mod Delay (short)	10	5	3	3	2	2	2	2	4
Mod Delay (slap)	5	2	1	1	1	–	–	–	4
Mod Delay (medium)	2	1	–	–	–	–	–	–	4
Mod Delay (long)	–	–	–	–	–	–	–	–	–
Mod Delay (extra long)	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a	n/a
Dither	12	6	4	3	2	2	1	1	4
POW-r Dither	3	1	1	0	0	0	0	0	4
Time Adjuster	23	11	9	7	5	5	4	3	4
Trim	29	15	10	7	6	5	4	3	4
Signal Generator	12	6	4	3	2	2	1	1	4

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

appendix b

DSP Delays Incurred by TDM Plug-Ins

This table shows the delays inherent in each type of DigiRack TDM plug-in. Delays matter only if you use a real-time TDM plug-in on one channel of a stereo or multichannel signal but not the others. This can cause the channels to be out of phase.

Use the TimeAdjuster plug-in to compensate for these delays. See “TimeAdjuster” on page 53.

Table 3. Samples of delay incurred by each real-time DigiRack TDM plug-in on HD, MIX, and DSP Farm cards

Plug-In	Samples of delay on HD cards	Samples of delay on MIX cards	Samples of delay on DSP Farm cards
EQ II (1-band)	4	4	4
EQ II (4-band)	4	4	4
Compressor II	4	4	4
De-esser	4	4	4
DigiRack Pitch	3	3	3
Limiter II	4	4	4
Gate II	4	4	4
Expander/Gate II	4	4	4
Mod Delay I/II (short)	4	4	4
Mod Delay I/II (slap)	4	4	4
Mod Delay I/II (medium)	4	4	4
Mod Delay I/II (long)	4	4	n/a

Table 3. (Continued) Samples of delay incurred by each real-time DigiRack TDM plug-in on HD, MIX, and DSP Farm

Plug-In	Samples of delay on HD cards	Samples of delay on MIX cards	Samples of delay on DSP Farm cards
Mod Delay II (extra long)	4	4	n/a
Dither	4	4	4
POW-r Dither	4	4	4
Time Adjuster	4	4	4
Signal Generator	4	4	4

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