

DigiRack™ Plug-ins

Version 7.4



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chapter 1

Introduction

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

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

A *References to Pro Tools LE™ in this guide are usually interchangeable with Pro Tools M-Powered™ or Pro Tools Academic™, except as noted in the Pro Tools M-Powered and Academic Getting Started Guides.*

Plug-in Formats

There are four plug-in formats used in Pro Tools:

- TDM plug-ins (real-time, DSP-based)
- RTAS® plug-ins (real-time, host-based)
- AudioSuite™ plug-ins (non-real-time, file-based processing)
- Elastic Audio plug-ins (both real-time and non-real time, host-based Time Compression and Expansion)

TDM Plug-ins

(Pro Tools HD Only)

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

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

RTAS Plug-ins

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

Because of this dependence on the CPU or *host processing*, the more RTAS plug-ins you use concurrently in a session, the greater the impact it will have on other aspects of your system's performance, such as maximum track count, number of available voices, the density of edits possible, and latency in automation and recording.

RTAS plug-ins can be used with Pro Tools HD, Pro Tools LE™.

AudioSuite Plug-ins

AudioSuite plug-ins 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 either creates an entirely new audio file, or alters the original source audio file. AudioSuite plug-ins can be used on all Pro Tools systems and Avid software.

Elastic Audio Plug-ins

Elastic Audio plug-ins provide Time Compression and Expansion (TCE) algorithms for applying Real-Time and Rendered, track-based Elastic Audio processing. The selected Elastic Audio plug-in determines how the audio is processed. Each plug-in uses a unique algorithm appropriate for different types of audio (for example, use the Rhythmic plug-in for processing drum loops).

Elastic Audio plug-ins have their own unique selector that is only available in the Edit window. You cannot insert Elastic Audio plug-ins on track Plug-in inserts (TDM, RTAS, and hardware inserts).

Elastic Audio plug-ins can be used on all Pro Tools systems.



For information about working with Elastic Audio and using Elastic Audio plug-ins, see the Pro Tools Reference Guide.

System Requirements

To use DigiRack plug-ins you need any the following:

- A Digidesign-qualified Pro Tools system running Pro Tools
- A Digidesign-qualified Pro Tools system and a third-party software application that supports the Digidesign TDM, RTAS, or AudioSuite plug-in standards
- A qualified Avid® Xpress®, Avid Xpress DV, or Avid DNA™ system (AudioSuite only)
- A qualified Digidesign VENUE system (TDM only)

For complete system requirements visit the Digidesign website (www.digidesign.com).

Compatibility Information

Digidesign can only assure compatibility and provide support for hardware and software it has tested and approved.





For a list of Digidesign-qualified computers, operating systems, hard drives, and third-party devices, visit the Digidesign website (www.digidesign.com).

Conventions Used in This Guide

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

Convention	Action
File > Save	Choose Save from the File menu
Control+N	Hold down the Control key and press the N key
Control-click	Hold down the Control key and click the mouse button
Right-click	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 this guide and other Digidesign guides.*

About www.digidesign.com

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

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

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

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

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

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

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

chapter 2

Installation and Configuration

Installing Plug-ins

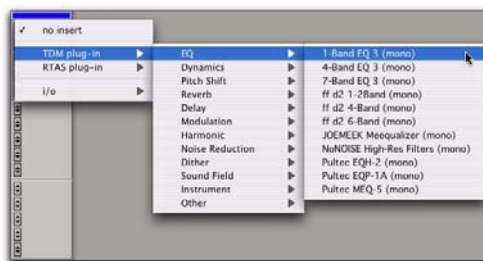
The Digidesign DigiRack plug-ins are installed when you install Pro Tools. For more information about installing Pro Tools, see the *Getting Started Guide* that came with your system.

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

- 1 DigiRack plug-ins (installed with Pro Tools).
- 2 Packs and other bundles included with your Digidesign system.
- 3 Any additional plug-ins. Check with the manufacturer of your plug-ins for compatibility information and support.

Organizing Plug-ins

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



Plug-ins organized by categories



Plug-ins organized by manufacturer

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

“Organize Plug-in Menus By” Option

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

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

Plug-in categories include:

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

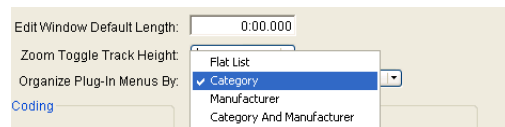
Manufacturer Organizes plug-ins by their manufacturer (such as Digidesign, Eventide, Line 6, McDSP), with individual plug-ins listed in the manufacturer submenus. Plug-ins that do not have a Manufacturer defined will appear in the “Other” manufacturer folder.

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

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

To select how plug-ins are organized:

- 1 Open or create a session.
- 2 Choose Setup > Preferences, and click the Display tab.
- 3 Click the “Organize Plug-in Menus By” pop-up menu, and select one of the view options.



Choosing Category view for plug-in menus

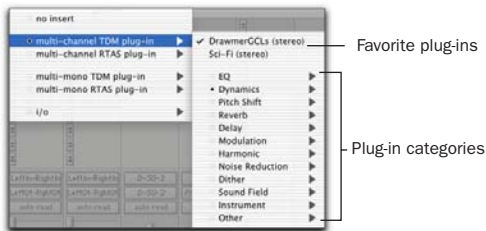
- 4 Click Done.

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

Plug-in Favorites

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

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



Menu display of favorite plug-ins (Category view shown)

To designate a plug-in as a favorite:

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

To change the status of a plug-in favorite:

- 1 Control-click (Windows) or Command-click (Mac) on a plug-in Insert button, and select a plug-in favorite that you no longer want to designate as a favorite.
- 2 When selecting the plug-in favorite to change, you can select the plug-in from the list of favorites at the top of the plug-in menu, or in its plug-in category/manufacturer sub-menu (where the plug-in appears in bold).

Default Plug-ins

You can set a default EQ plug-in and Dynamics plug-in, which places them at the top of the Insert selector pop-up menu on-screen, and first in the list of menu choices when assigning inserts on ICON worksurfaces.



Default plug-in display in Insert selector menu

To set a default plug-in:

- 1 Choose Setup > Preferences and click Mixing.
- 2 Under Setup, choose a plug-in from the Default EQ or Default Dynamics pop-up menu.
- 3 Click OK to close the Preferences window.

chapter 3

Working with Real-Time Plug-ins

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

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

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

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

Processing Power Requirements of TDM and RTAS Plug-ins

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


TDM Plug-ins

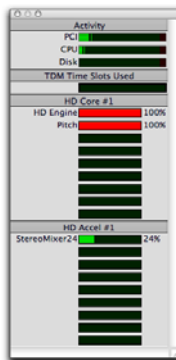
Each real-time TDM plug-in that is inserted in a Pro Tools session uses a portion of your system's total available DSP resources.

Since these DSP resources reside on the cards that make up your particular Pro Tools hardware configuration, the amount of DSP available depends entirely on the number and type of DSP cards in your system.


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

The System Usage window (Window > System Usage) shows how much DSP is available on your system, and gives an approximation of how it is currently being used.

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



System Usage window

 *See Appendix A, “DSP Requirements for TDM Plug-ins” for relative numbers of instances of each DigiRack TDM plug-in that can be powered by a single DSP chip. DSP capacity differs with card type.*

DSP Sharing Between TDM Plug-ins

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


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

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

RTAS Plug-ins

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

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

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


Improving RTAS Plug-in Performance

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

Hardware Buffer Size

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

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

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

To change the Hardware Buffer Size:

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

RTAS Processors

The RTAS Processors setting determines the number of processors in your computer allocated for RTAS plug-in processing.

With computers that have multiple processors, or that feature multi-core processing or hyper-threading, this setting lets you enable multi-processor support for RTAS processes. Used in combination with the CPU Usage Limit setting, the RTAS Processors setting lets you control the way RTAS processing and other Pro Tools tasks are carried out by the system.

- ◆ A higher number of processors reserves more CPU processing capacity for RTAS plug-in processing. This is useful for sessions with large number of RTAS plug-ins.
- ◆ A lower number of processors leaves more CPU processing capacity for automation, screen redraws, and video playback in Pro Tools, or for other application running at the same time as Pro Tools.


To set the number of RTAS Processors:

- 1 Choose Setup > Playback Engine.
- 2 From the RTAS Processors pop-up menu, select the number of available processors you want to allocate for RTAS plug-in processing.
- 3 Click OK.

CPU Usage Limit


The CPU Usage Limit controls the percentage of CPU resources allocated to Pro Tools host processing tasks such as RTAS plug-in performance and screen redraws.

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

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

To change the CPU Usage Limit:

- 1 Choose Setup > Playback Engine.
- 2 From the CPU Usage Limit pop-up menu, select the percentage of CPU processing you want to allocate to Pro Tools. (On dual-processor Mac computers, this setting controls the allocation of a single processor.)
- 3 Click OK.

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

RTAS Error Suppression

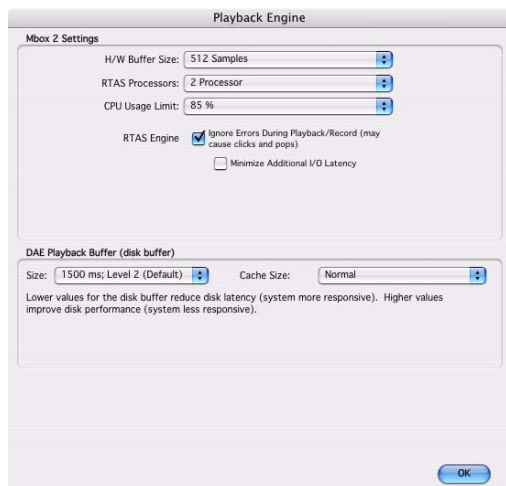
You should only enable RTAS error suppression if you are experiencing frequent RTAS errors that are interrupting your creative workflow. When RTAS error suppression is enabled, you can experience a degradation of audio quality. However, this may be acceptable in order to avoid interrupting playback and recording when working with instrument plug-ins. Be sure to disable RTAS error suppression when you need to ensure the highest possible audio quality, such as for the final mix or audio layback.



Enabling Ignore Errors During Playback/Record only suppresses error reporting. RTAS errors can result in pops and clicks in the audio during playback and recording. To avoid this, try increasing the CPU Usage Limit and the HW Buffer Size, and reduce the number of RTAS plug-ins in use in the session.

To enable RTAS error suppression:

- 1 Choose Setup > Playback Engine.



Playback Engine dialog, Mbox 2 (Mac OS)

- 2 Select Ignore Errors During Playback/Record.

- 3 If available, you can also select Minimize Additional I/O Latency.

- 4 Click OK.

RTAS Error Suppression Options

Ignore Errors During Playback/Record When enabled, Pro Tools continues to play and record even if the RTAS processing requirements exceed the selected CPU Usage Limit. This can result in pops and clicks in the audio during playback and recording, but does not stop the transport.

Minimize Additional I/O Latency When enabled, any additional latency due to suppressing RTAS errors during playback and record is minimized to 128 samples. Suppressing RTAS errors requires at least 128 samples of additional buffering on some systems. If the Minimize Additional I/O Latency option is disabled, the buffer is half the H/W Buffer Size, or at least 128 samples (which ever is greater). If you are on an older, slower computer, you may not want to enable this option since doing so can adversely affect performance.

The Minimize Additional I/O Latency option is only available if the Ignore Errors During Playback/Record option is enabled.

This option is only available on Pro Tools systems that require additional buffering for RTAS error suppression, as follows:

- Windows XP:
 - Mbox[®] 2 Pro
 - Digidesign-qualified M-Audio interfaces with Pro Tools M-Powered or Pro Tools Academic
- Mac OS X:
 - Mbox
 - Mbox 2
 - Mbox 2 Pro
 - Mbox 2 Mini
 - Digi 002[®] and Digi 002 Rack
 - All Pro Tools M-Powered systems
 - All Pro Tools Academic systems

RTAS Plug-ins on Auxiliary Inputs or Master Faders


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

This can affect voice usage and total latency, and limitations on inserting or removing plug-ins during playback. For more information, see “RTAS Plug-ins on Auxiliary Input and Master Fader Tracks” on page 27.

Delay in Signal Processing

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

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


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

Compensating for Delay

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


Delay Compensation

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

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


TimeAdjuster

You can compensate for TDM or RTAS plug-in-induced delays by using the TimeAdjuster plug-in. This plug-in lets you apply a specific number of samples of delay to the signal path of a Pro Tools track. TimeAdjuster provides settings files that apply the correct compensation time in samples for delay introduced by one or more plug-ins. See “TimeAdjuster” on page 89 for more information about the TimeAdjuster plug-in.

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

Manual Delay Compensation


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

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

Low-Latency Recording with Instrument Plug-ins

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

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

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

MIDI and Audio Processing Plug-ins

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

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

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

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

- 1 Start-Control-click (Windows) or Command-Control-click (Mac) the Track Compensation indicator for the Auxiliary Input to bypass delay compensation.
- 2 Enter the total system delay into the User Offset field.

Plug-ins as Inserts

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

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

Inserts can be used in two ways:

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

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

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

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

Pre-Fader Operation

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

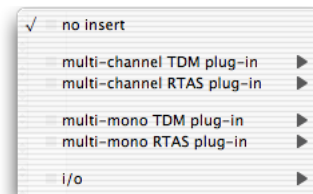
Real-time plug-ins are pre-fader, but post-disk. This means that if you record to disk with a plug-in inserted on the record track, you will hear the effect of the plug-in, but the effect will not be recorded to disk.

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

Mono, Multi-Mono, and Multichannel Plug-ins

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

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



Plug-in formats

Mono Plug-ins Used on mono tracks. Some mono plug-ins (such as Mod Delay) can generate stereo or greater-than-stereo output from a mono channel. Plug-ins that occur on a track after a stereo plug-in are used in stereo as well.

Multi-Mono Plug-ins Used on stereo or greater-than-stereo multichannel tracks when a multichannel version of the plug-in is not available. Multi-mono plug-ins can analyze and process each channel independently without regard to each other. Controls for all channels are linked by default so that you can adjust them in tan-

dem. You can unlink controls for independent adjustment using the Master Link button. See “Linking and Unlinking Controls on Multi-Mono Plug-ins” on page 21.

⚠ *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 or Edit window.

To show inserts in the Mix window:

- Choose View > Mix Window > Inserts.

To show inserts in the Edit window:

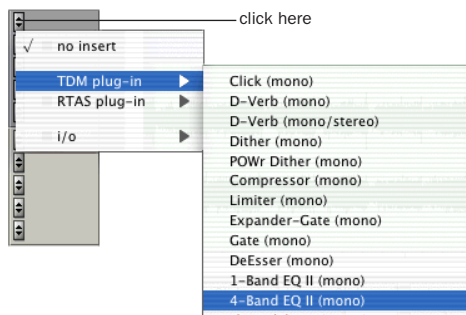
- Choose View > Edit Window > Inserts.

To insert a plug-in on a track:

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



Inserting a plug-in with Plug-ins by Category enabled



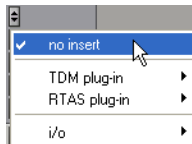
Inserting a plug-in (standard view)

💡 You can also drag and drop plug-in settings files (.tfx) from DigiBase browsers, or Windows Explorer or the Mac Finder, to Inserts to insert the corresponding plug-in with those settings.

Removing an Insert from a Track

To remove an insert from a track:

- Click the Insert selector and select No Insert.



Removing a plug-in

Configuring Plug-ins During Playback

You can change, add, or remove real-time plug-ins and hardware inserts during playback. On Pro Tools|HD systems, adding or removing RTAS plug-ins that change voicing assignments (for example, inserting an RTAS plug-in after a TDM plug-in) temporarily interrupts playback, but without stopping the transport.

Plug-in Changes During Record

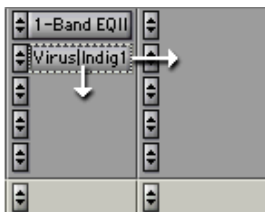
Changing plug-ins during audio recording is disallowed so that you don't inadvertently interrupt audio recording. However, you can make changes while recording MIDI only.

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 replaces it.

To duplicate an insert:

- Alt-drag (Windows) or Option-drag (Mac) 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, do one of the following:

- Control-Start-click (Windows) or Command-Control-click (Mac) 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:

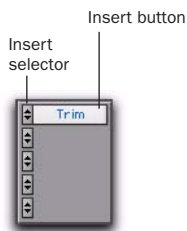
- Control-Start-Alt click (Windows) or Command-Control-Option click (Mac) an Insert button in the position you want to toggle.

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

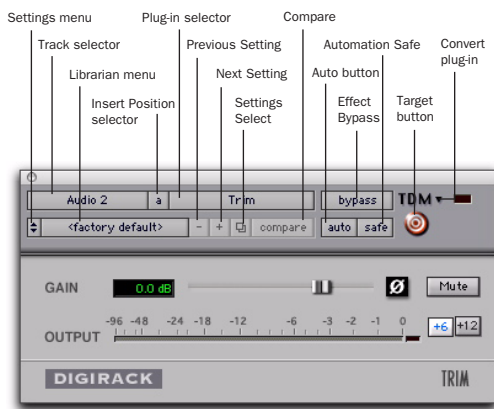
- Control-Start-Alt-Shift-click (Windows) or Command-Control-Option-Shift-click (Mac) an Insert button in the position you want to toggle.

The Plug-in Window

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



Insert button and selector on track



Plug-in window (mono Trim shown)

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

Insert Position Selector Accesses any insert on the current track.

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

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

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

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

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

Librarian Menu Recalls settings files saved in the plug-in's root settings folder or in the current session's Settings folder. See "Using the Librarian" on page 127.

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

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

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

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

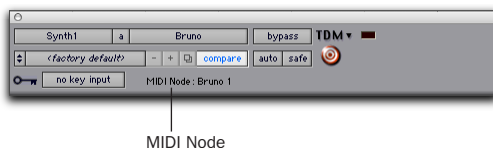
Auto Lets you enable individual plug-in controls for automation recording. See "Automating Plug-ins" on page 24.

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

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

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

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



Midi Node display in a plug-in header (Bruno shown)

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

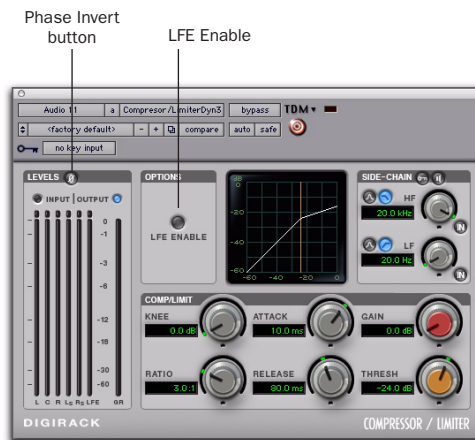


Plug-in window (multi-mono 1-Band EQ shown)

Link Enable Buttons Let you selectively link or unlink the controls of specific channels of a multi-mono plug-in. Each square represents a speaker channel. The Master Link button must be disabled to use the Link Enable buttons. See “Linking and Unlinking Controls on Multi-Mono Plug-ins” on page 21.

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

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



Plug-in window (multichannel Compressor/Limiter shown)

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

Tempo Sync Enables compatible plug-ins to automatically synchronize to the session tempo, for rhythmic delay, echo, and similar effects. See “Tempo Sync” on page 23 for more information.

Opening Plug-in Windows

To open a Plug-in window:

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

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

Opening Multiple Plug-in Windows

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

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

To open an additional Plug-in window:

- Shift-click the Insert button for the additional plug-in.

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

- Alt-click (Windows) or Option-click (Mac) the Channel selector in the Plug-in window of the multi-mono plug-in.

To close all currently open Plug-in windows:

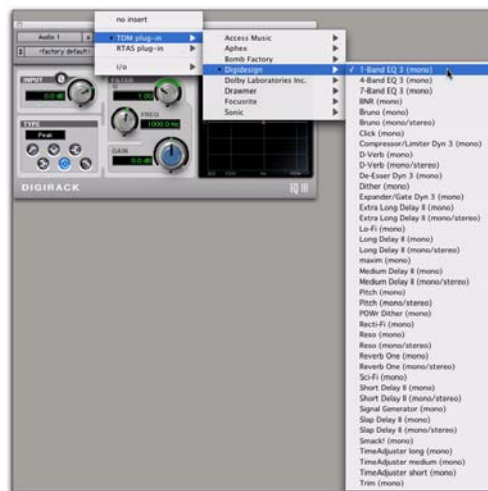
- Alt-click (Windows) or Option-click (Mac) the close box of any currently open Plug-in window.

Plug-in Window Controls

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

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

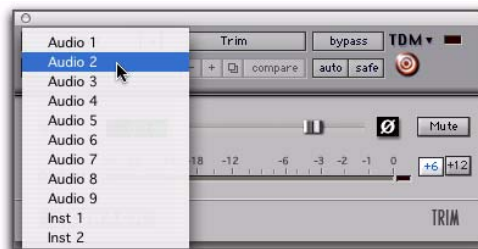
- Click the Insert selector and select a plug-in from the pop-up menu (or Category sub-menu).



Selecting a plug-in from the Plug-in window

To choose a different track:

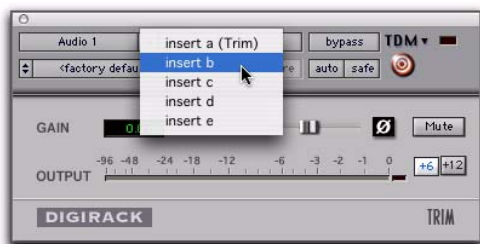
- Click the Track selector and choose a track from the pop-up menu.



Choosing a track from the Plug-in window

To choose a different insert:

- Click the Insert Position selector and choose an insert from the pop-up menu.




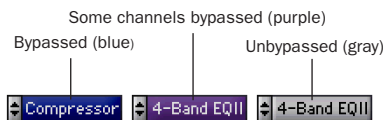
Choosing an Insert from the Plug-in window

Bypassing Plug-ins

To bypass a plug-in, do one of the following:

- Click the Plug-in window's Bypass button.
- OR —
- Control-click (Windows) or Command-click (Mac) 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.



Bypass states indicated by color

Linking and Unlinking Controls on Multi-Mono Plug-ins

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

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



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

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.

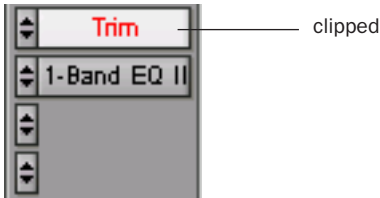
Clip Indicators

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

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



Plug-in Clipping indicator in the plug-in header



Plug-in Clipping indication in the Mix window

To clear plug-in clipping:

- Click the clip LED in the plug-in header

To clear a clip indicator:

- Click the clip LED in the plug-in header

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

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

- Alt-click (Windows) or Option-click (Mac) any meter.
- Press Alt-C (Windows) or Option-C (Mac).
- Choose Track > Clear All Clip Indicators.

Using the Side-Chain Input

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

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

A typical use for external side-chain processing is to control the dynamics of one audio signal using the dynamics of another signal (the key input). For example, a kick drum track could be used to trigger gating of a bass track to tighten it up, or a rhythm guitar track could be used to gate a keyboard pad.

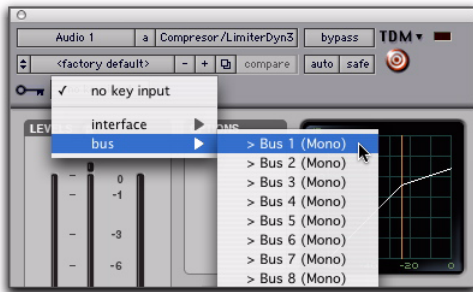
Side-Chain Filters

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

Using a Key Input for External Side-Chain Processing

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

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



Selecting a key input

- 2 Click External Key to activate external side-chain processing.
- 3 Press the Spacebar to begin playback. The plug-in uses the input or bus that you chose as a key input to trigger its effect.
- 4 During playback, do any of the following to fine-tune side-chain triggering:
 - To hear the audio source you have selected to control the side-chain, enable Side-Chain Listen or Key Listen (depending on the plug-in). When finished, disable Side-Chain Listen/Key Listen to resume normal plug-in monitoring.
 - To filter the side-chain so that only specific frequencies trigger the plug-in, use the filtering controls (if available) to select the desired frequency range.
- 5 Adjust the plug-in's Threshold control (if available) and other controls as needed.

Tempo Sync

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

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

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

Compatibility and Settings

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

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

To utilize Tempo Sync:

- 1 Insert a plug-in that supports Tempo Sync, such as the DigiRack Mod Delay II.
- 2 Click the Tempo Sync icon. The tempo shown will change to match the current session tempo.



Tempo Sync icon (Mod Delay II shown)

- 3 To set a rhythmic delay, click a note value (whole, half, quarter, eighth, or sixteenth), then do any of the following to further adjust the rhythm:
 - To enable triplet delay timing, click the 3 button so that it is lit.
 - To set a dotted delay value, click to enable the dotted icon.
 - To apply some swing to the delay value, adjust the Groove slider.

Automating Plug-ins

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



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



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

Preference Setting to Auto Enable All Plug-in Parameters

You can configure Pro Tools to enable all parameters for automation when you first assign a plug-in insert.

To auto-enable all parameters on newly inserted plug-ins:

- 1 Choose Setup > Preferences, and click the Mixing tab.
- 2 In the Automation section, click to select the Plug-in Controls Default to Auto-Enabled setting.
 - When selected, all applicable controls of newly added plug-ins are enabled for automation.
 - When not selected, the controls of newly added plug-in must be manually enabled for automation.

Enabling Plug-in Parameters for Automation

To enable plug-in controls for automation:

- 1 Open the Plug-in window for the plug-in you want to automate.
- 2 Do one of the following:
 - Click the Automation Enable button in the Plug-in window.



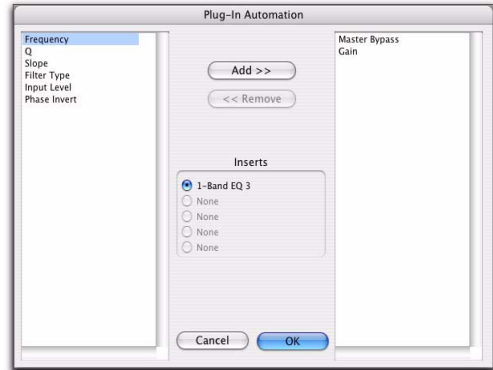
Accessing the Plug-in Automation dialog

– or –

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

💡 You can also Control-Alt-Start-click (Windows) or Command-Option-Control-click (Mac) any plug-in control in the Plug-in window, then choose Open Automation Dialog from the pop-up menu.

- 3 Select 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.

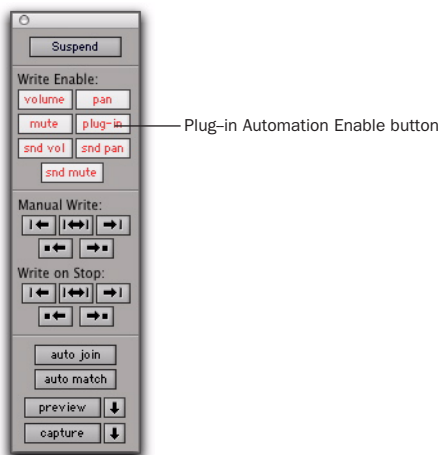
💡 As an alternative to using the Plug-in Automation dialog, you can enable individual plug-in controls directly from the Plug-in window by Control-Alt-Start-clicking (Windows) or Command-Option-Control-clicking (Mac) the control.



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.

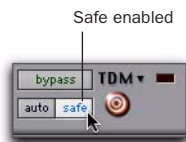
After the initial automation pass, you can write additional automation to the track without completely erasing the previous pass by choosing Auto Touch mode or Auto Latch mode. These modes add new automation only when you actually move the control.

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 enabling a plug-in


To disable plug-in safe mode:

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

Instrument Plug-ins


Instrument plug-ins are unique from other plug-ins in that they generate audio by synthesis or sample playback, and are controlled by MIDI. Typically, plug-in instruments are inserted on Instrument tracks.

You can also insert plug-in instruments on Auxiliary Input tracks and control them with an additional MIDI track. In some cases you may want to insert an instrument plug-in on an instrument track, assign any additional audio outputs in may have to additional Auxiliary Input tracks, and control it from not only the Instrument track on which it is inserted, but also from multiple MIDI tracks (for example, if you are using a multiple output-capable drum machine plug-in).

 *For more information on Instrument tracks, see the Pro Tools Reference Guide.*

Multiple Outputs for Instrument Plug-ins

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

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

Main and Auxiliary Plug-in Outputs

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



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

RTAS Plug-ins on Auxiliary Input and Master Fader Tracks

(Pro Tools HD Only)

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



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



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

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

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

Voice Usage and Total Latency for RTAS Plug-ins

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

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

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

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

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

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

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

Ordering RTAS and TDM Plug-ins on an Audio Track

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

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

TDM Plug-ins Grouped Before RTAS Plug-ins Each initial insert of an RTAS plug-in after a TDM plug-in will cause processing latency, and will use voices as described in “Voice Usage and Total Latency for RTAS Plug-ins” on page 28. RTAS plug-ins will stay active while Record Enable or TrackInput monitoring is enabled for that track.

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

Ordering Recommendations

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

- To ensure that RTAS plug-ins stay active when you record enable a track or use TrackInput monitoring, group all TDM plug-ins before RTAS plug-ins.
 - or –
- To conserve voices and minimize processing latency, group all RTAS plug-ins before TDM plug-ins.

Limitations for Inserting or Removing Plug-ins During Playback

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

- Inserting RTAS and TDM plug-ins on or removing from Auxiliary Input, Master Fader, or Instrument tracks temporarily interrupts playback, but without stopping the transport.
- Inserting an RTAS plug-in after a TDM plug-in on an audio track, or removing an RTAS plug-in from an audio track if it comes after a TDM plug-in, temporarily interrupts playback, but without stopping the transport.
- Inserting a TDM plug-in before an RTAS plug-in on an audio track, or removing a TDM plug-in from an audio track if it comes before an RTAS plug-in, temporarily interrupts playback, but without stopping the transport.

HTDM Plug-in Conversion to RTAS

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

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

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

▲ *When HTDM plug-ins are converted to RTAS, any RTAS plug-ins that precede TDM plug-ins on audio tracks will be bypassed if you enable Record Enable or TrackInput monitoring on that track. See “Ordering RTAS and TDM Plug-ins on an Audio Track” on page 28 for details.*

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.

The AudioSuite Menu

AudioSuite plug-ins are accessed through the AudioSuite menu.

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

 For more information, see “Organizing Plug-ins” on page 5.

AudioSuite Plug-ins

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

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

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

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

Compressor/Limiter III Processes audio with compression, or limiting.

Chorus Processes audio with chorusing.

Flanger Processes audio with flanging.

Expander/Gate III Processes audio with expander-gating.

De-esser III 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.

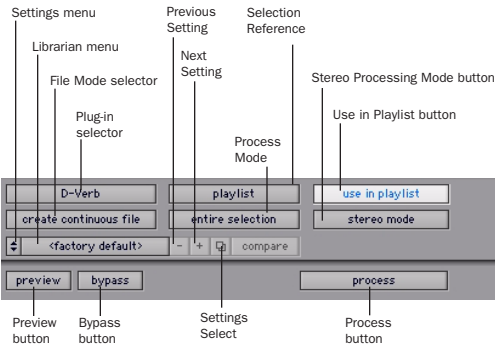
Signal Generator Generates audio tones for equipment calibration purposes.

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

Time Shift Changes an audio file's duration and pitch independently or together (Varispeed).

The AudioSuite Window

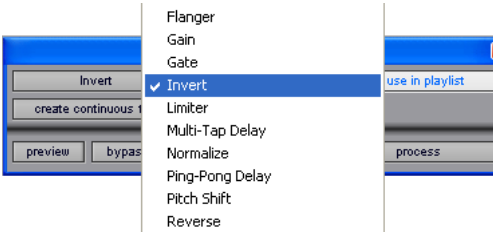
When you choose a plug-in from the AudioSuite menu, the AudioSuite window appears. From this floating window you can access and edit the controls for any non-real-time AudioSuite plug-in.



AudioSuite window

Plug-in Selector

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



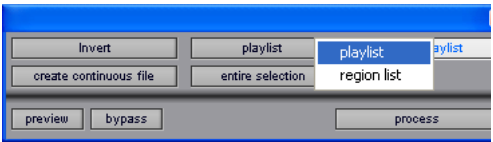
Plug-in selector menu (flat list, non-category view)

Selection Reference

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

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

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



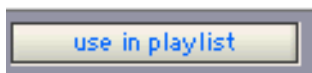
Selection Reference pop-up menu (Invert plug-in shown)

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

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

Use In Playlist Button

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



Use in Playlist button

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

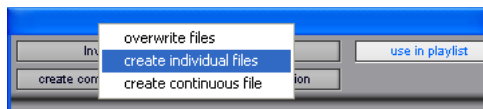
Use In Playlist On, Reference = Region List When Use in Playlist is enabled and the Selection Reference pop-up menu is set to Regions List, all copies of the selected region will be replaced everywhere in the session.

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

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

File Mode Selector

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



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

There are three options:

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

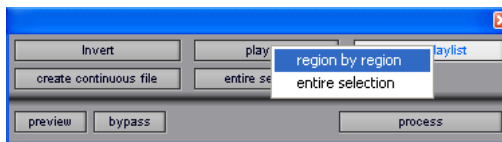
Create Individual Files Processes the selected regions nondestructively, creating a new audio file for each region. The new audio files are added to the session, leaving the original source audio files in the Regions List unchanged. (Whether the processed audio is added to the current playlist or only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

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

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

Process Mode Selector

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



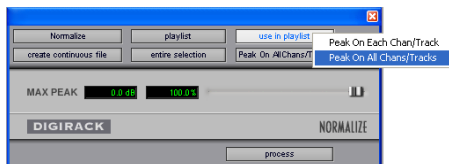
Process Mode selector (Invert plug-in shown)

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

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

Chan/Track Process Mode Selector

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



Chan/Track Process Mode selector pop-up menu (Normalize plug-in shown)


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

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

Preview

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

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

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

Before you use Preview, be aware that:

- ◆ The performance of the Preview function depends on the speed of your CPU. Faster computers preview AudioSuite effects better than slower computers.
- ◆ Regardless of how many tracks and regions are currently selected, the Preview button will audition only the first selected stereo track or the first selected pair of mono tracks.
- ◆ If you are using Region-by-Region processing mode, the Preview function will preview only the first region within a multi-region selection. To hear all selected regions, temporarily select Create Continuous File from the File Mode menu before previewing.
- ◆ The Preview function is affected by the AudioSuite Buffer Size option (on the Processing page of the Pro Tools Preferences dialog). See “AudioSuite Processing Preferences” on page 37.

Bypass

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


Process

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

New files are written to the hard disk specified for that track in the Disk Allocation dialog, or to the same drive as the original file if the region is not currently on a track. See “Auto-File Naming of AudioSuite-Processed Audio” on page 36.

Multichannel Processing

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

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

Undoing AudioSuite Processing

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

⚠ *Undo is not available when a plug-in is configured for destructive editing since the process has already overwritten the source audio file.*

Auto-File Naming of AudioSuite-Processed Audio

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

Auto file-naming follows these rules:

- ◆ New regions are named beginning with the region name, followed by an abbreviation of the current AudioSuite process, followed by standard Pro Tools file and region numbering.
- ◆ If a plug-in's File Mode pop-up is set to Overwrite, the original region's name will not be changed.
- ◆ If a plug-in's File Mode pop-up is set to Create Individual Files, the resulting regions will have an abbreviated version of the plug-in name appended to them.

Other AudioSuite Controls

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

Plug-in Librarian and Settings Menus

The Settings and Librarian pop-up menus that appear in some AudioSuite plug-ins provide a means of saving, loading, copying, pasting and organizing custom plug-in settings files. See “Using the Librarian” on page 127.

Analyze Button

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

Key Input Selector

Selects a track/bus to be used to trigger processing. In order to use this feature, the key input source audio must occur at the same time as the target audio. Side-chain input is monophonic. See “Using the Side-Chain Input” on page 22.

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

Channel Mode Button

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

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

Stereo Mode Processes each contiguous pair of channels within a selected region as a stereo track. When applied to an odd number of channels, the plug-in processes the last odd, unpaired channel as mono, using the left channel settings of the stereo plug-in. If you want the last track to be processed in stereo, you must select an additional track to pair it with—an empty one if necessary.

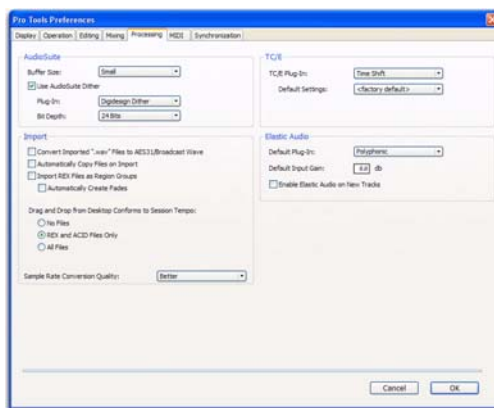
For example, a region comprised of two mono tracks would be processed as a stereo track, with the first mono track processed as the left channel and the second mono track processed as the right channel.

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

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

AudioSuite Processing Preferences

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



Processing Preferences page

AudioSuite

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

To configure the AudioSuite Buffer Size:

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


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

- 4 Click Done.

Use 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 or files that contain a fade in or fade out.

Dither Plug-in Lets you choose the dither plug-in algorithm used during AudioSuite processing.

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 the Pro Tools Reference Guide.*

TC/E

The TC/E section lets you choose the plug-in used for AudioSuite time compression and expansion processing, as well as its default settings.

TC/E Plug-in Selects the Time Compression/Expansion plug-in and default settings used when editing with the TCE Trim 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.

Default Settings Selects the settings used by the selected AudioSuite TC/E plug-in.

 *Refer to the Pro Tools Reference Guide for more information about the Time Compression/Expansion Trim 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.

A *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.

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

Selecting Tracks for Delay or Reverb Processing

Because some AudioSuite effects such as delay and reverb add additional material to the end of the selected audio, it is important that you make a selection that is *longer* than the original source material so that the plug-in can write it to the audio file. If you select only the original material, without leaving additional space at the end, any reverb decay or delay that occurs after the end of the region will be cut off.

To compensate for this, place the region in a track and select the desired audio *plus* an amount of blank space at the end of the region equal to the amount of delay or reverb decay that you have added in the plug-in. The plug-in will then have space at the end of the region in which to write the final delay or decay. If you select more blank space than you need, you can remove it using the Trimmer tool after applying AudioSuite processing.

To process audio with an AudioSuite plug-in:

- 1 Do one of the following:
 - Make an Edit selection of the audio you want to process.
 - or –
 - Select audio regions in the Region List that you want to process.
- 2 Choose an AudioSuite plug-in from the AudioSuite menu.

3 Adjust the plug-in controls for the effect you want. These settings determine how the file is processed and what effect the processing has 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.

4 Click the Preview button to begin audition processing of the selected material.

5 Make further adjustment to the plug-in parameters based if necessary.

6 When you are satisfied with the Preview results, click the Process button.

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

Using AudioSuite Plug-ins in Stereo

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

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



For more information, see “Channel Mode Button” on page 36.

chapter 5

Adjusting Plug-in Controls

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

To adjust a plug-in control:

- 1 Begin audio playback so that you can hear the control changes in real time.
- 2 Adjust the controls of the plug-in for the effect you want. Refer to “Editing Parameters Using a Mouse” on page 41 and “Editing Parameters Using a Computer Keyboard” on page 41.

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



Plug-in controls of similar category (such as EQ or Dynamics plug-ins) can follow track groups. See the Pro Tools Reference Guide for more information.

Editing Parameters Using a Mouse

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

Keyboard Shortcuts

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


Editing Parameters Using a Computer Keyboard

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


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

To change control values with a computer keyboard:

- 1 Click the text box corresponding to the control that you want to adjust.
- 2 Change the value by doing one of the following:
 - To increase a value, press the Up Arrow on your keyboard. To decrease a value, press the Down Arrow on your keyboard.
 - or –
 - Type the desired value.

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

- 3 Do one of the following:
 - Press Enter on the numeric keyboard to input the value and remain in keyboard editing mode.
 - or –
 - Press Enter on the alpha keyboard (Windows) or Return (Mac) 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.*

Editing Parameters Using a Scroll Wheel

Some controls have text boxes that display the current value of the parameter. You can edit the numeric value of a parameter using a scroll wheel.

To change control values using a scroll wheel:

- 1 Click the text box corresponding to the control that you want to adjust.
- 2 To increase a value, scroll up with the scroll wheel. To decrease a value, scroll down with the scroll wheel.

Toggling Switches

To toggle a switch:

- Click the switch.

Groups and Linked Plug-in Controls

(Pro Tools HD Only)

Plug-ins can follow Pro Tools Groups (Mix, Edit, or Mix/Edit) for linked control of multiple inserts. For more information, see the *Pro Tools Reference Guide*.

chapter 6

DigiRack Real-Time TDM and RTAS Plug-ins

EQ III

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

EQ III is available in the following formats:

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

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

EQ III can be operated from the following control surfaces:

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

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

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

EQ III Configurations

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

- 7 Band (“7-Band EQ 3”)
- 2–4 Band (“4-Band EQ 3”)
- 1 Band (“1-Band EQ 3”)

7 Band EQ and 2–4 Band EQ

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

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



7 Band EQ and 2–4 Band EQ window

1 Band EQ

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

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



1 Band EQ window

EQ III Controls

Adjusting EQ III Controls

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

Dragging Plug-in Controls

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



Dragging a plug-in control

Typing Control Values

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



Typing a control value

Inverting Filter Gain

(Peak EQ Bands Only)

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

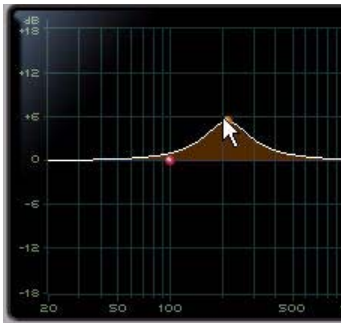
Dragging in the Frequency Graph Display

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

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

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

Q Start-dragging (Windows) or Control-dragging (Mac) a control point up increases the Q setting. Start-dragging (Windows) or Control-dragging (Mac) a control point down decreases the Q setting.



Dragging a control point in the Frequency Graph display

Adjusting Controls with Fine Resolution

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

Resetting Controls to Default Values

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

Using Band-Pass Mode

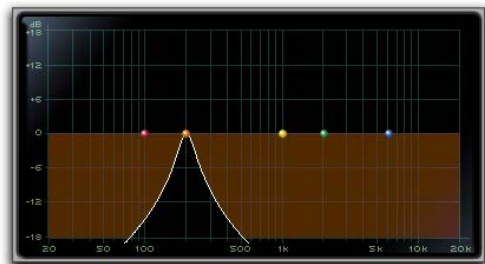
You can temporarily set any EQ III control to Band-Pass monitoring mode. Band-Pass mode cuts monitoring frequencies above and below the Frequency setting, leaving a narrow band of mid-range frequencies. It is especially useful for adjusting limited bandwidth in order to solo and fine-tune each individual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



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

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

- Hold Start+Shift (Windows) or Control+Shift (Mac), and drag any rotary control or control point horizontally or vertically.



EQ III interactive graph displaying Band-Pass mode

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

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

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

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

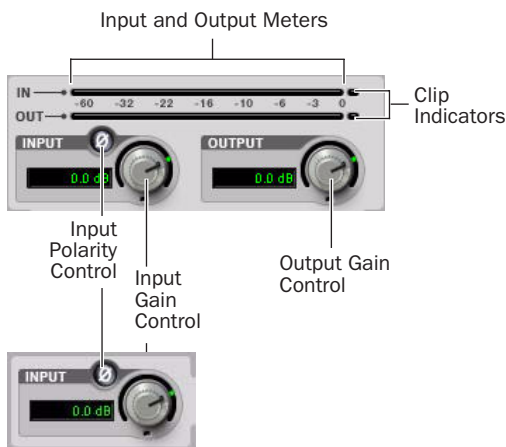
- Release Start+Shift (Windows) or Control+Shift (Mac).

Using a Control Surface

EQ III can be controlled from any supported control surface, including Digidesign's D-Control, D-Command, ProControl, Control24, Digi 002, or Command8. Refer to the guide that came with the control surface for details.

I/O Controls

The following Input and Output controls are found on all EQ III configurations, except where noted otherwise.



I/O controls and meters for 7 Band EQ and 2-4 Band EQ (top) and 1 Band EQ (bottom)

Input Gain Control

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

Output Gain Control

(7 Band EQ and 2-4 Band EQ Only)

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

Input Polarity Control

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

Input and Output Meters

(7 Band EQ and 2-4 Band EQ Only)

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

Green Indicates nominal levels

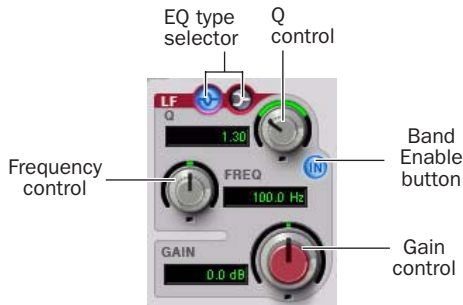
Yellow Indicates pre-clipping levels, starting at -6 dB below full scale

Red Indicates full scale levels (clipping)

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

EQ Band Controls

The individual EQ bands on each EQ III configuration have some combination of the following controls, as noted below.



EQ Band controls (7 Band EQ Low-Mid band shown)

EQ Type Selector

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

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

Band Enable Button

(7 Band EQ and 2–4 Band EQ Only)

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

Band Gain Control

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

Frequency Control

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

Q Control

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

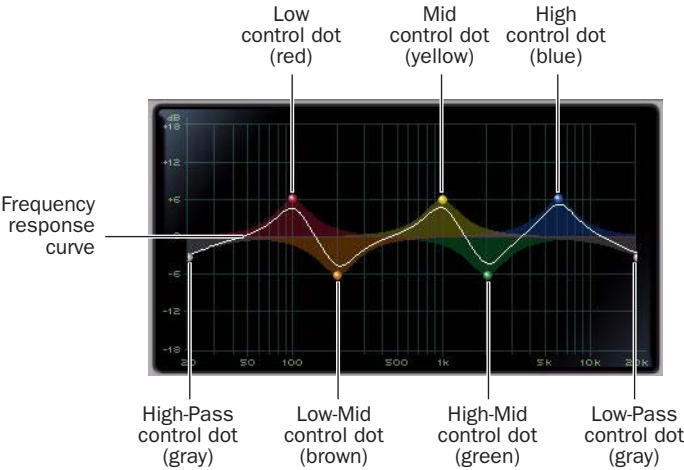
Shelf On Shelf bands, the Q control changes the Q of the shelving filter. Higher Q values represent steeper shelving curves. Lower Q values represent broader shelving curves.

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

Frequency Graph Display

(7 Band EQ and 2-4 Band EQ Only)

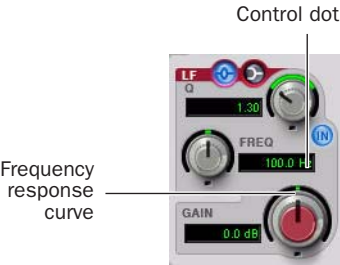
The Frequency Graph display in the 7 Band EQ and the 2-4 Band EQ shows a color-coded control dot that corresponds to the color of the Gain control for each band. The filter shape of each band is similarly color-coded. The white frequency response curve shows the contribution of each of the enabled filters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



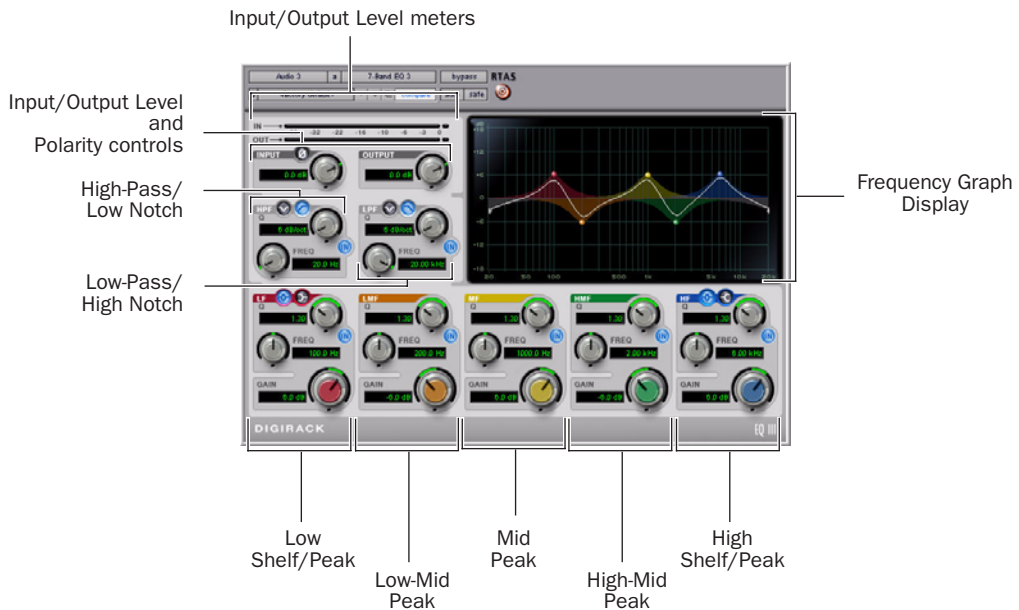
Frequency Graph display for the 7 Band EQ

1 Band EQ

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



7 Band EQ



7 Band EQ and 2-4 Band EQ window

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

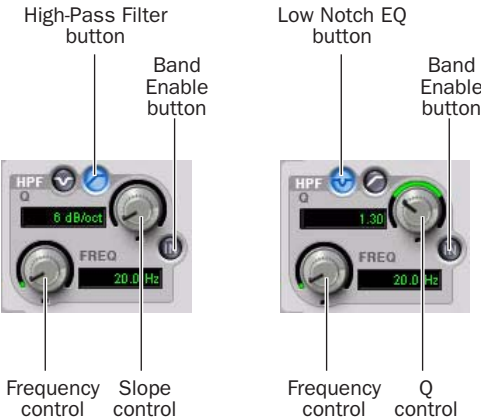
All seven bands are available for simultaneous use. In the factory default setting, the High-Pass/Low Notch and Low-Pass/High Notch bands are out of circuit, the Low Shelf and High Shelf bands are selected and in circuit, and the Low-Mid Peak, Mid Peak, High-Mid Peak bands are in circuit.

High-Pass/Low Notch

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

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

Low-Notch EQ Attenuates a narrow band of frequencies centered around the Frequency setting. The width of the attenuated band is determined by the Q setting.



High-Pass filter (left) and Low Notch EQ (right)

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

High-Pass Filter and Low Notch EQ control values

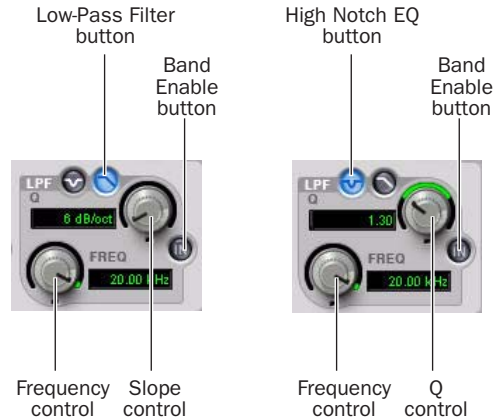
Control	Value
Frequency Range	20 Hz to 8 kHz
Frequency Default	20 Hz
HPF Slope Values	6, 12, 18, or 24 dB/oct
Low Notch Q Range	0.1 to 10.0
Low Notch Q Default	1.0

Low-Pass/High Notch

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

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

High-Notch EQ Attenuates a narrow band of frequencies centered around the Frequency setting. The width of the attenuated band is determined by the Q setting.



Low-Pass filter (left) and High Notch EQ (right)

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

Low-Pass Filter and High Notch EQ control values

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

Low Shelf/Low Peak

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

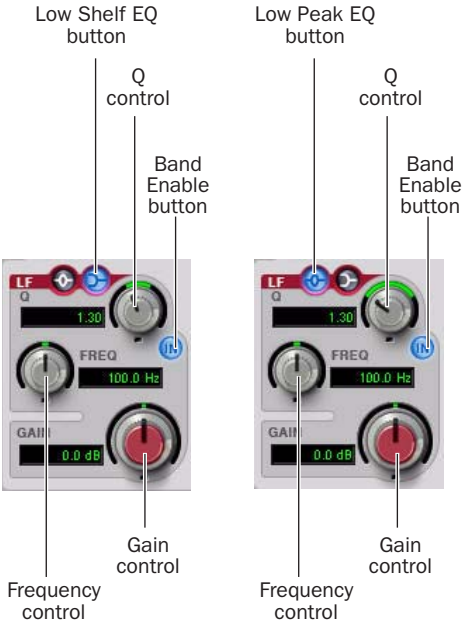
Low-Shelf EQ Boosts or cuts frequencies at and below the Frequency setting. The amount of boost or cut is determined by the Gain setting. The Q setting determines the shape of the shelving curve.

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

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

Low Shelf EQ and Low Peak EQ control values

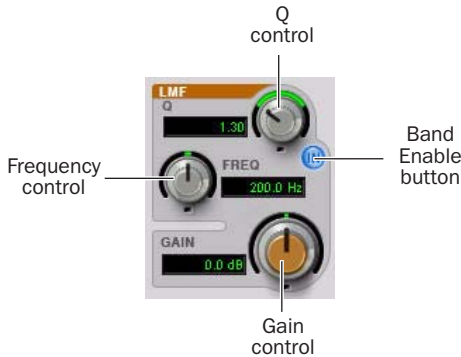
Control	Value
Frequency Range	20 Hz to 500 Hz
Frequency Default	100 Hz
Low Shelf Q Range	0.1 to 2.0
Low Peak Q Range	0.1 to 10.0
Q Default	1.0
Low Shelf Gain Range	-12 dB to +12 dB
Low Peak Gain Range	-18 dB to +18 dB



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

Low-Mid Peak

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



Low-Mid Peak EQ

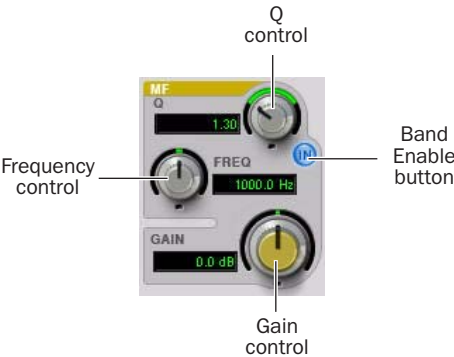
The Low-Mid Gain control and its corresponding graph elements are displayed on-screen in brown.

Low-Mid Peak EQ control values

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

Mid Peak

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



Mid Peak EQ

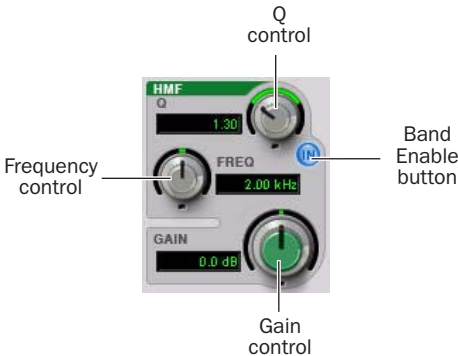
The Mid Gain control and its corresponding graph elements are displayed on-screen in yellow.

Mid Peak EQ control values

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

High-Mid Peak

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



High-Mid Peak EQ

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

High-Mid Peak EQ control values

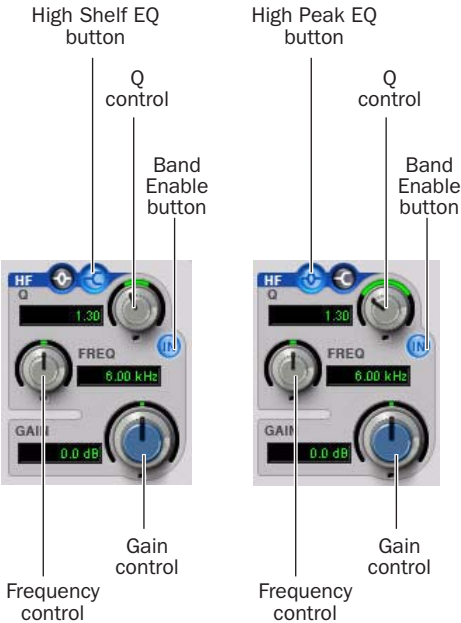
Control	Value
Frequency Range	200 Hz to 18 kHz
Frequency Default	2 kHz
Mid Peak Q Range	0.1 to 10.0
Mid Peak Q Default	1.0
Mid Peak Gain Range	-18 dB to +18 dB

High Shelf/High Peak

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

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

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



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

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

High Shelf EQ and High Peak EQ control values

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

2–4 Band EQ

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

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



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

Changing from 2–4 Band to 7 Band

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

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

Filter Usage with 2–4 Band EQs

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

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

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

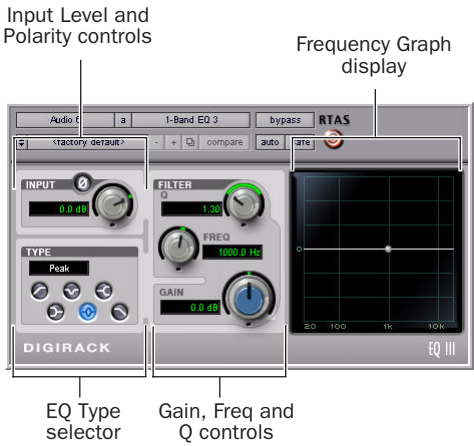
When you switch an existing EQ III plug-in between the 2–4 Band and 7 Band versions, or when you import settings between versions, the change is subject to the following conditions:

Changing from 7 Band to 2–4 Band

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

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

1 Band EQ



1 Band EQ window

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

Band Controls

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

1 Band EQ control values

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

EQ Types

High-Pass Filter

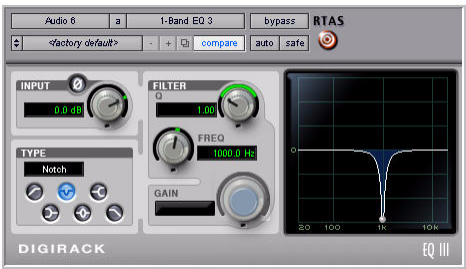
The High-Pass filter attenuates all frequencies below the Frequency setting at the selected rate (6 dB, 12 dB, 18 dB, or 24 dB per octave) while letting all frequencies above pass through. No gain control is available for this filter type.



1 Band EQ set to High-Pass Filter

Notch Filter

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



1 Band EQ set to Notch Filter

High-Shelf EQ

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



1 Band EQ set to High-Shelf EQ

Low-Shelf EQ

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



1 Band EQ set to Low-Shelf EQ

Peak EQ

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



1 Band EQ set to Peak EQ

Low-Pass Filter

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



1 Band EQ set to Low-Pass Filter

Automating EQ III Controls

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

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

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



Enabling automation for a control

To automate multiple controls:

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

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


Automation-enabled buttons




Automation-enabled knobs

Automation-enabled controls

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

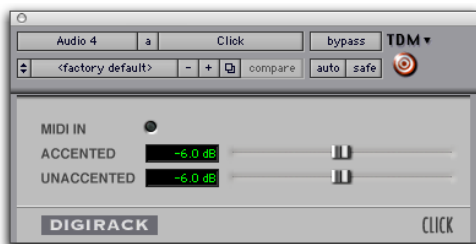
 For complete instructions on automating plug-ins in Pro Tools, see the *Pro Tools Reference Guide*.

EQ II

 For information, see the *Digidesign Legacy Plug-ins Guide*.

Click

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



Click plug-in

Click Controls

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

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

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

Creating a Click Track

To use the Click plug-in:

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

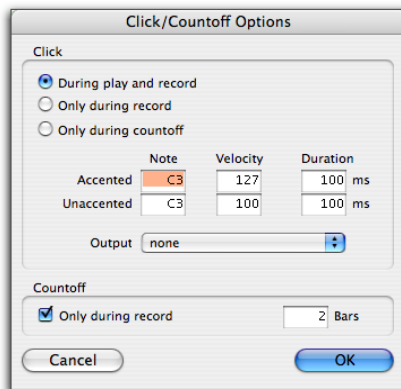
2 Do one of the following:

- Choose Track > Create Click Track. Pro Tools creates a new Instrument track named “Click” with the Click plug-in already inserted. In the Edit window, the click track’s Track Height is set to Mini.
- or –
- Create new a mono Auxiliary Input track and insert the Click plug-in.

3 Select a click sound preset.


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

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



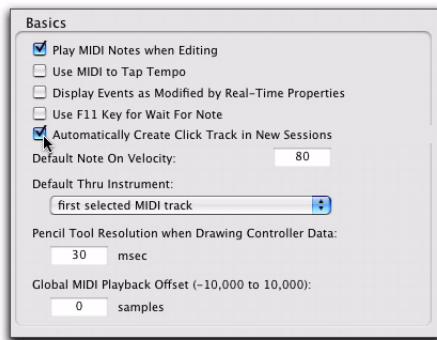
Click Options dialog

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

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

Automatically Create Click Track In New Sessions


To have Pro Tools automatically create a click track when you create a new session, enable the Automatically Create Click Track In New Sessions option in the MIDI Preferences page (Setup > Preferences).

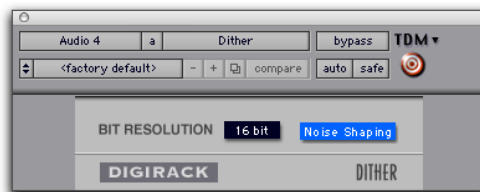


Selecting the Automatically Create Click Track In New Sessions option in the MIDI Preferences page

Dither

The Dither plug-in minimizes quantization artifacts when reducing the bit depth of an audio signal to 16-, 18-, or 20-bit resolution.


 For more advanced dithering, use the DigiRack POW-r Dither plug-in. See “POW-r Dither” on page 60.




Dither plug-in

Whenever you are mixing down or bouncing to disk and your destination bit depth is lower than 24-bit, insert a dither plug-in on a Master Fader that controls the output mix.

Using a dither plug-in on a Master Fader is preferable to an Auxiliary Input because Master Fader inserts are post-fader. As a post-fader insert, the dither plug-in can process changes in Master Fader level.

 For more information on using Dither, see the Pro Tools Reference Guide.

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

 If you are mixing down to an analog destination with any 24-bit capable interface, you do not need to use Dither. This allows maximum output fidelity from the 24-bit digital-to-analog convertors of the interface.

Dither Controls

Bit Resolution

Use this pop-up menu to choose one of three possible resolutions for the Dither processing. Set this control to the maximum bit resolution of your destination.

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

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 as 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.

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

Noise Shaping

Noise shaping can further improve audio performance and reduce perceived noise inherent in dithered audio. Noise shaping uses filtering to shift noise away from frequencies in the middle of the audio spectrum (around 4 kHz), where the human ear is most sensitive.

The Noise Shaping button engages or disengages Noise shaping. Noise shaping is on when the button is highlighted in blue.

POW-r Dither

(Pro Tools HD and Pro Tools LE Only)

The POW-r Dither plug-in is an advanced type of dither that provides optimized bit depth 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 59.



POW-r Dither plug-in

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

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

POW-r Dither Controls

Bit Resolution

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

16-bit Recommended for output to digital 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 can further improve audio performance and reduce perceived noise inherent in dithered audio. Noise shaping uses filtering to shift noise away from frequencies in the middle of the audio spectrum (around 4 kHz), where the human ear is most sensitive.


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

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

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

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

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

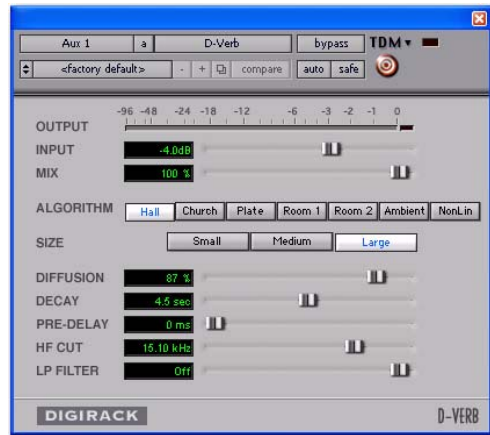
 *For more information on using Dither, see the Pro Tools Reference Guide.*

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 lights remain on. To clear the clip indicator, click it. With longer reverb times there is a greater likelihood of clipping occurring as the feedback element of the reverb builds up and approaches a high output level. See also “Clip Indicators” on page 22.

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 III

Dynamics III provides three modules:

- Compressor/Limiter
- Expander/Gate
- De-Esser

All Dynamics III modules are available in TDM, RTAS, and AudioSuite formats.

Dynamics III supports 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz sample rates. Compressor/Limiter and Expander/Gate modules work with mono, stereo, and greater-than-stereo multichannel formats up to 7.1. The De-Esser module works with mono and stereo formats only.

In addition to standard controls in each module, Dynamics III also provides a graph to track the gain transfer curve in the Compressor/Limiter and Expander/Gate plug-ins, and a frequency graph to display which frequencies trigger the De-Esser and which frequencies will be gain reduced.

Adjusting Dynamics III Controls

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

Editing Parameters Using a Mouse

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



Dragging a plug-in control

Keyboard Shortcuts

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

Typing Control Values

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



Typing a control value

Editing Parameters Using a Computer Keyboard

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

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

To change control values with a computer keyboard:

- 1 Click on the text box corresponding to the control that you want to adjust.
- 2 Change the value by doing one of the following:

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

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

- 3 Do one of the following:

- Press Enter on the numeric keyboard to input the value and remain in keyboard editing mode.
- or –
- Press Enter on the alpha keyboard (Windows) or Return (Mac) to enter the value and leave keyboard editing mode.

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

Enabling Buttons

To enable a button, click it.

Using a Control Surface


Dynamics III can be controlled from any supported control surface, interface, and worksurface. Refer to the appropriate controller product guide for more information.

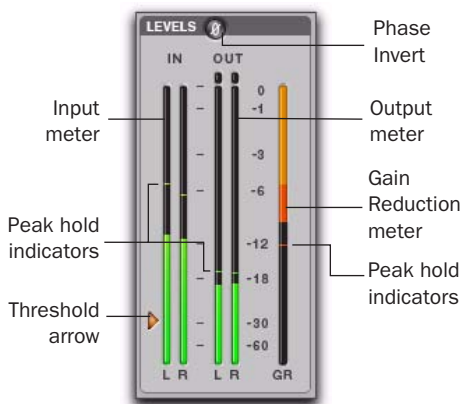
Shared Compressor/Limiter and Expander/Gate Features

The following parts of the user interface are shared between the Compressor/Limiter and Expander/Gate Dynamics III plug-ins.

Levels Section

This indicator let you track input, output, and gain reduction levels, as well as work with phase invert and the threshold setting.

 See “De-Esser Levels Section” on page 74 for more information on De-Esser III Input/Output Level controls.



I/O Meter display (stereo instance shown)

Input and Output Meters


The Input and Output meters show peak signal levels before and after dynamics processing:


Green Indicates nominal levels.

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

Red Indicates full scale levels (clipping).

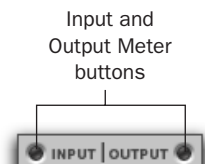
The clip indicators at the top of the Output meters indicate clipping at the input or output stage of the plug-in. Clip indicators can be cleared by clicking the indicator.

 The Input and Output meters display differently depending on the type of track (mono, stereo, or multichannel) on which the plug-in has been inserted.

 When Side-Chain Listen is enabled, the Output meter only displays the levels of the side-chain signal. See “Side-Chain Listen” on page 77.

Toggle Multichannel Input and Output Meters

With multichannel track types LCRS and higher, both Input Level and Output Level meters cannot be shown at the same time. Click either the Input or Output Meter button to display the appropriate level meter. The Input/Output level meters display is toggled to Output by default.



Input and Output meter buttons

Gain Reduction Meter

The Gain Reduction meter indicates the amount the input signal is attenuated (in dB) and shows different colors during dynamics processing:

Light Orange Indicates that gain reduction is within the “knee” and has not reached the full ratio of compression.

Dark Orange Indicates that gain reduction is being applied at the full ratio (for example, 2:1).

Threshold Arrow

The orange Threshold arrow next to the Input meter indicates the current threshold, and can be dragged up or down to adjust the threshold. When a multichannel instance of the plug-in has been configured to show only the Output meter, the Threshold arrow is not displayed.

Phase Invert

The Phase Invert button inverts the phase (polarity) of the input signal, to help compensate for phase anomalies that can occur either in multi-microphone environments or because of mis-wired balanced connections.

LFE Enable

(Pro Tools HD Systems Only)

The LFE Enable button (located in the Options section) is on by default, and enables plug-in processing of the LFE (low frequency effects) channel on a multichannel track formatted for 5.1, 6.1, or 7.1 surround formats.



LFE Enable button (Compressor/Limiter III shown)

To disable LFE processing, deselect this button.

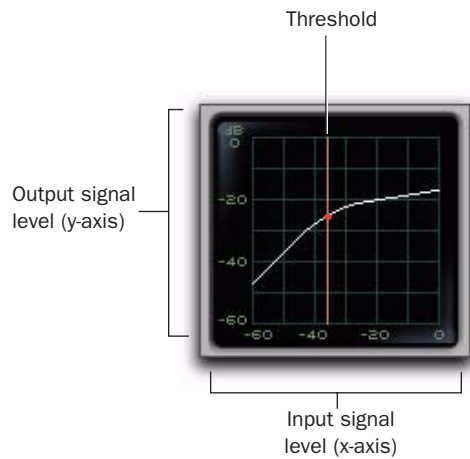


The LFE Enable button is not available if the plug-in is not inserted on an applicable track.

Dynamics Graph Display

The Dynamics Graph display—used with the Compressor/Limiter and Expander/Gate plug-ins—shows a curve that represents the level of the input signal (on the x-axis) and the level of the output signal (on the y-axis). The orange vertical line represents the threshold.

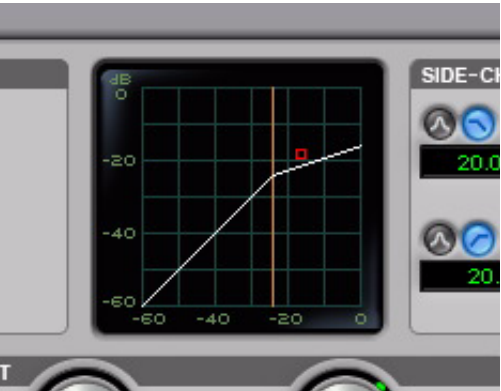
Use this graph as a visual guideline to see how much dynamics processing you are applying.



Dynamics graph display

The Compressor/Limiter and Expander/Gate plug-ins also feature an animated, multi-color cursor in their gain transfer curve displays.

The gain transfer curve of the Compressor/Limiter and Expander/Gate plug-ins shows a moving ball cursor that shows the amount of input gain (x-axis) and gain reduction (y-axis) being applied to the incoming signal.



Gain transfer curve and cursor showing amount of compression

To indicate overshoots (when an incoming signal peak is too fast for the current compression setting) the cursor temporarily leaves the gain transfer curve.

The cursor changes color to indicate the amount of compression applied, as follows:

Dynamics III Plug-in Compression Amount

Cursor Color	Compression Amount
white	no compression
light orange	below full ratio
dark orange	full ratio amount

See “De-Esser Frequency Graph Display” on page 75 for information on using the De-Esser’s graph display.

Side-Chain Section

For information on using the Side-Chain section of the Compressor/Limiter or Expander/Gate, see “Using the Side-Chain Input in Dynamics III” on page 76.

Compressor/Limiter III



Compressor/Limiter III

The Compressor/Limiter plug-in applies either compression or limiting to audio material, depending on the ratio of compression used.

About Compression

Compression reduces the dynamic range of signals that exceed a chosen threshold by a specific amount. The Threshold control sets the level that the signal must exceed to trigger compression. The Attack control sets how quickly the compressor responds to the “front” of an audio signal once it crosses the selected threshold. The Release control sets the amount of time that it takes for the compressor’s gain to return to its original level after the input signal drops below the selected threshold.

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 does not decrease the overall volume too drastically, or eliminate desired attack transients in the program material.

Of course, compression has many creative uses that break these rules.

About Limiting

Limiting prevents signal peaks from ever exceeding a chosen threshold, and is generally 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.

Limiting generally begins with the ratio set at 10:1 and higher. Large ratios effectively limit the dynamic range of the signal to a specific value by setting an absolute ceiling for the dynamic range.

Compressor/Limiter III Controls

This section describes controls for the Compressor/Limiter plug-in.

Input/Output Level Meters

The Input and Output meters show peak signal levels before and after dynamics processing. See “Levels Section” on page 65 for more information.

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.

Compressor/Limiter Graph Display

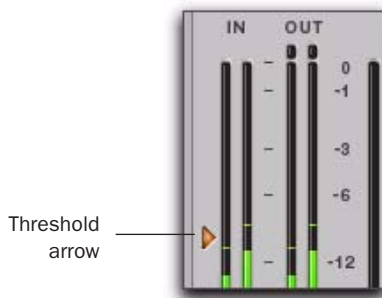
The Dynamics Graph display lets you visually see how much expansion or gating you are applying to your audio material. See “Dynamics Graph Display” on page 66.

Threshold

The Threshold control sets the level that an input signal must exceed to trigger compression or limiting. Signals that exceed this level will be compressed. Signals that are below it will be unaffected.

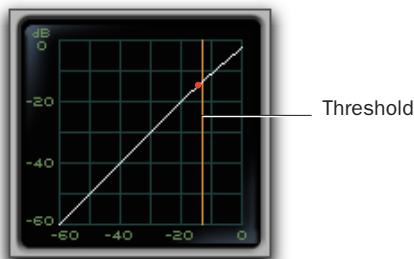
This control has an approximate range of –60 dB to 0 dB, with a setting of 0 dB equivalent to no compression or limiting. The default value for the Threshold control is –24 dB.

An orange arrow on the Input meter indicates the current threshold, and can also be dragged up or down to adjust the threshold setting.



Threshold arrow on input meter

The Dynamics Graph display also shows the threshold as an orange vertical line.



Threshold indicator on Dynamics Graph display

This control ranges from –60 dB (lowest gain) to 0 dB (highest gain).

Ratio

The Ratio control sets the compression ratio, or the amount of compression applied as the input signal exceeds the threshold. For example, a 2:1 compression ratio means that a 2 dB increase of level above the threshold produces a 1 dB increase in output.

This control ranges from 1:1 (no compression) to 100:1 (hard limiting).

Attack

The Attack control sets the attack time, or the rate at which gain is reduced after the input signal crosses the threshold.

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

This control ranges from 10 μ s (fastest attack time) to 300 ms (slowest attack time).

Release

The Release control sets the length of time it takes for the Compressor/Limiter to be fully deactivated after the input signal drops below the threshold.

Release times should be set long enough that if signal levels repeatedly rise above the threshold, the gain reduction “recovers” smoothly. If the release time is too short, the gain can rapidly fluctuate as the compressor repeatedly tries to recover from the gain reduction. If the release time is too long, a loud section of the audio material could cause gain reduction that continues through soft sections of program material without recovering.

This control ranges from 5 ms (fastest release time) to 4 seconds (slowest release time).

Knee

The Knee control sets the rate at which the compressor reaches full compression once the threshold has been exceeded.

As you increase this control, it goes from applying “hard-knee” compression to “soft-knee” compression:

- With hard-knee compression, compression begins when the input signal exceeds the threshold. This can sound abrupt and is ideal for limiting.
- With soft-knee compression, gentle compression begins and increases gradually as the input signal approaches the threshold, and reaches full compression after exceeding the threshold. This creates smoother compression.



Hard knee



Soft knee

Graph examples of hard and soft knee compression

For example, a Knee setting of 10 dB would be the gain range over which the ratio gradually increased to the set ratio amount.

The Gain Reduction meter displays light orange while gain reduction has not exceeded the knee setting, and switches to dark orange when gain reduction reaches the full ratio.

This control ranges from 0 dB (hardest response) to 30 dB (softest response).

Gain

The Gain control lets you boost overall output gain to compensate for heavily compressed or limited signals.

This control ranges from 0 dB (no gain boost) to +40 dB (loudest gain boost), with the default value at 0 dB.

 For more information on the LFE channel, refer to the *Pro Tools Reference Guide*.

Side-Chain Section

The *side-chain* is the split-off signal used by the plug-in's detector to trigger dynamics processing. The Side-Chain section lets you toggle the side-chain between the internal input signal or an external key input, and tailor the equalization of the side-chain signal so that the triggering of dynamics processing becomes frequency-sensitive. See “Using the Side-Chain Input in Dynamics III” on page 76.

Expander/Gate III



Expander/Gate III

The Expander/Gate plug-in applies expansion or gating to audio material, depending on the ratio setting.

About Expansion

Expansion decreases the gain of signals that fall below a chosen threshold. They 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.

About Gating

Gating silences signals that fall below a chosen threshold. To enable gating, simply set the Ratio and Range controls to their maximum values.

Expanders can be thought of as soft noise gates since they provide a gentler way of reducing noisy low-level signals than the typically abrupt cutoff of a gate.

Expander/Gate III Controls

This section describes controls for the Expander/Gate plug-in.

Input/Output Level Meters

The Input and Output meters show peak signal levels before and after dynamics processing. See “Levels Section” on page 65 for more information.

Expander/Gate Dynamics Graph Display

The Dynamics Graph display lets you visually see how much expansion or gating you are applying to your audio material. See “Dynamics Graph Display” on page 66.

Look Ahead Button

Normally, dynamics processing begins when the level of the input signal crosses the threshold. When the Look Ahead button is enabled, dynamics processing begins 2 milliseconds before the level of the input signal crosses the threshold.



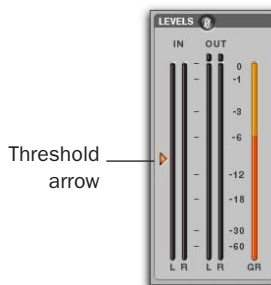
Look Ahead control

The Look Ahead control is useful for avoiding the loss of transients that may have been otherwise cut off or trimmed in a signal.

Threshold

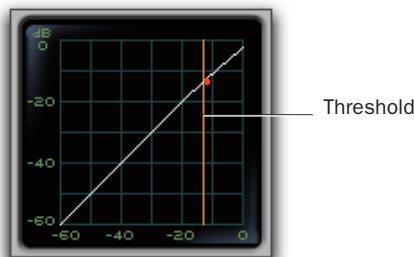
The Threshold control sets the level below which an input signal must fall to trigger expansion or gating. Signals that fall below the threshold will be reduced in gain. Signals that are above it will be unaffected.

An orange arrow on the Input meter indicates the current threshold, and can also be dragged up or down to adjust the threshold setting.



Threshold arrow on Input meter

The Dynamics Graph display also shows the threshold as an orange vertical line.



Threshold indicator on Dynamics Graph display

This control has an approximate range of -60 dB to 0 dB, with a setting of 0 dB equivalent to no compression or limiting. The default value for the Threshold control is -24 dB.

Ratio

The Ratio control 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 cutting off signals that fall below the threshold. As you adjust the ratio control, refer to the built-in graph to see how the shape of the expansion curve changes.

This control ranges from 1:1 (no expansion) to 100:1 (gating).

Attack

The Attack control sets the attack time, or the rate at which gain is reduced after the input signal crosses the threshold. Use this along with the Ratio setting to control how soft the Expander's gain reduction curve is.

This control ranges from 10 μ s (fastest attack time) to 300 ms (slowest attack time).

Hold

The Hold control specifies the duration (in seconds or milliseconds) during which the Expander/Gate will stay in effect after the initial attack occurs. This can be used as a function to keep the Expander/Gate in effect 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 close and open very rapidly.

This control ranges from 5 ms (shortest hold) to 4 seconds (longest hold).

Release

The Release control sets how long it takes for the gate to close after the input signal falls below the threshold level and the hold time has passed.

This control ranges from 5 ms (fastest release time) to 4 seconds (slowest release time).

Range

The Range control sets the depth of the Expander/Gate when closed. 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.

This control ranges from -80 dB (lowest depth) to 0 dB (highest depth).

Side-Chain Section

The *side-chain* is the split-off signal used by the plug-in's detector to trigger dynamics processing. The Side-Chain section lets you toggle the side-chain between the internal input signal or an external key input, and tailor the equalization of the side-chain signal so that the triggering of dynamics processing becomes frequency-sensitive.

See "Using the Side-Chain Input in Dynamics III" on page 76.

De-Esser III



De-Esser III

The De-Esser reduces sibilants 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 De-Esser 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 De-Esser operates.

Using De-essing Effectively

To use de-essing most effectively, insert the De-Esser after compressor or limiter plug-ins.

The Frequency control 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 in an undesired manner.

Similarly, the Range control should be set to a dB level low enough so that de-essing is triggered only by sibilants. If the Range is set too high, 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 Range control so that it is lower on soft sections.

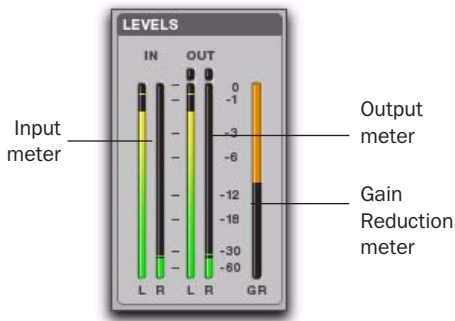
💡 *The De-Esser has no control to directly adjust the threshold level (the level that an input signal must exceed to trigger de-essing). The amount of de-essing will vary with the input signal.*

De-Esser III Controls

This section describes controls for the De-Esser plug-in.

De-Esser Levels Section

These controls let you track input, output, and gain reduction levels.



De-Esser III I/O Meter display

Input and Output Meters

The Input and Output meters show peak signal levels before and after dynamics processing:

Green Indicates nominal levels.

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

Red Indicates full scale levels (clipping).

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

Gain Reduction Meter

The Gain Reduction meter indicates the amount the input signal is attenuated, in dB. This meter shows different colors during de-essing:

Light Orange Indicates that gain reduction is being applied, but has not reached the maximum level set by the Range control.

Dark Orange Indicates that gain reduction has reached the maximum level set by the Range control.

Frequency

The Frequency control sets the frequency band in which the De-Esser operates. When HF Only is disabled, gain is reduced in frequencies within the specified range. When HF Only is enabled, the gain of frequencies above the specified value will be reduced.

This control ranges from 500 Hz (lowest frequency) to 16 kHz (highest frequency).

Range

The Range control defines the maximum amount of gain reduction possible when a signal is detected at the frequency set by the Frequency control.

This control ranges from -40 dB (maximum de-essing) to 0 dB (no de-essing).

HF Only

When the HF Only button is enabled, gain reduction is applied only to the active frequency band set by the Frequency control. When the HF Only button is disabled, the De-Esser applies gain reduction to the entire signal.

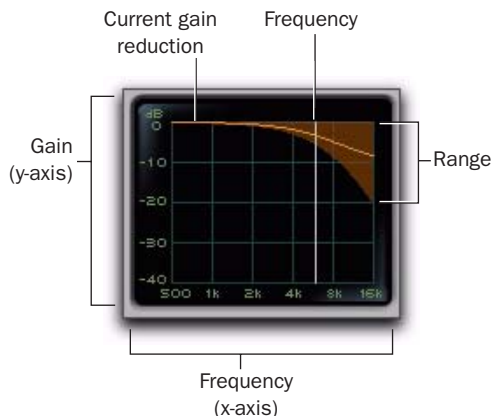
Listen

When enabled, the Listen button lets you monitor the sibilant peaks used by the De-Esser as a side-chain to trigger compression. This is useful for listening only to the sibilance for fine-tuning De-Esser controls. To monitor the whole output signal without this filtering, deselect the Listen button.

De-Esser Frequency Graph Display

The De-Esser Frequency Graph display shows a curve that represents the level of gain reduction (on the y-axis) for the range of the output signal's frequency (on the x-axis). The white line represents the current Frequency setting, and the animated orange line represents the level of gain reduction being applied to the signal.

Use this graph as a visual guideline to see how much dynamics processing you are applying at different points in the frequency spectrum.




De-Esser graph display

Using the Side-Chain Input in Dynamics III

(Compressor/Limiter and Expander/Gate Only)

Dynamics processors typically use the detected amplitude of their input signal to trigger gain reduction. This split-off signal is known as the *side-chain*. The Compressor/Limiter and Expander/Gate plug-ins feature external key capabilities and filters for the side-chain.

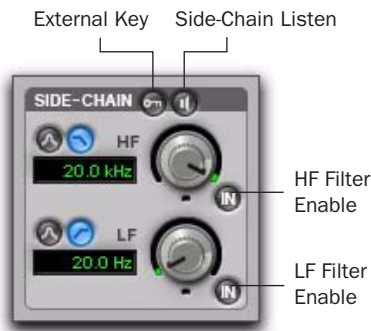
With external key side-chain processing, you trigger dynamics processing using an external signal (such as a separate reference track or audio source) instead of the input signal. This external source is known as the *key input*.

 In Pro Tools TDM 6.9.x and lower, RTAS plug-ins do not provide external key side-chain processing. If you want to use side-chain processing, use the TDM versions of plug-ins on Pro Tools|HD systems.

With side-chain filters, you can make dynamics processing more or less sensitive to certain frequencies. For example, you might configure the side-chain so that certain lower frequencies on a drum track trigger dynamics processing.

Side-Chain Section

The Side-Chain section lets you toggle the side-chain between the internal input signal or an external key input, listen to the side-chain, and tailor the equalization of the side-chain signal so that the triggering of dynamics processing becomes frequency-sensitive.



Compressor/Limiter and Expander/Gate Side-Chain

External Key

The External Key toggles external side-chain processing on or off. When this button is highlighted, the plug-in uses the amplitude of a separate reference track or external audio source to trigger dynamics processing. When this button is dark gray, the External Key is disabled and the plug-in uses the amplitude of the input signal to trigger dynamics processing.

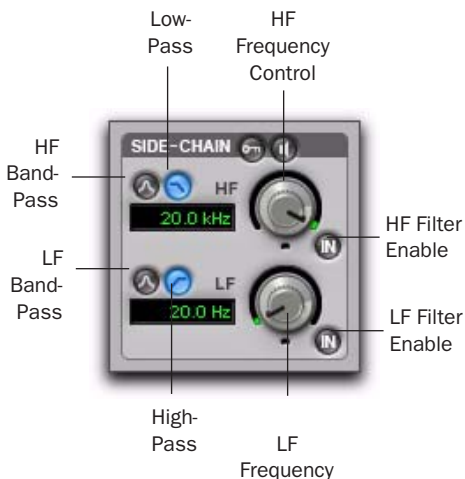
Side-Chain Listen

When enabled, this control lets you listen to the internal or external side-chain input by itself, as well as monitor its levels with the Output meter. This is especially useful for fine-tuning the plug-in's filter settings or external key input.

⚠ *Side-Chain Listen is not saved with other plug-in settings.*

HF and LF Filter Enable Buttons

The HF Filter Enable and LF Filter Enable buttons toggle the corresponding filter in or out of the side-chain. When this button is highlighted, the filter is applied to the side-chain signal. When this button is dark gray, the filter is bypassed and available for activation.



HF and LF Filter Side-Chain

High-Frequency (HF) Filter Type

The HF filter section lets you filter higher frequencies out of the side-chain signal so that only certain bands of high frequencies or lower frequencies pass through to trigger dynamics processing. The HF side-chain filter is switchable between Band-Pass and Low-Pass filters.

Band-Pass Filter Makes triggering of dynamics processing more sensitive to frequencies within the narrow band centered around the Frequency setting, and rolling off at a slope of 12 dB per octave.

Low-Pass Filter Makes triggering of dynamics processing more sensitive to frequencies below the Frequency setting rolling off at a slope of 12 dB per octave.

HF Frequency Control

The Frequency control sets the frequency position for the Band-Pass or Low-Pass filter, and ranges from 80 Hz to 20 kHz.

Low-Frequency (LF) Filter Type

The LF filter section lets you filter lower frequencies out of the side-chain signal so that only certain bands of low frequencies or higher frequencies are allowed to pass through to trigger dynamics processing. The LF side-chain is switchable between Band-Pass and High-Pass filters.

Band-Pass Filter Makes triggering of dynamics processing more sensitive to frequencies within the narrow band centered around the Frequency setting, and rolling off at a slope of 12 dB per octave.

High-Pass Filter Makes triggering of dynamics processing more sensitive to frequencies above the Frequency setting rolling off at a slope of 12 dB per octave.

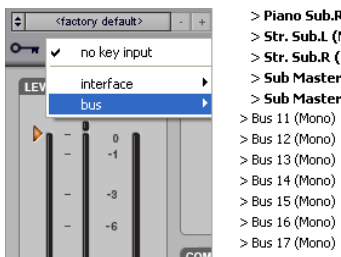
LF Frequency Control

The Frequency control sets the frequency position for the Band-Pass or High-Pass filter, and ranges from 25 Hz to 4 kHz.

Using an External Key Input for Side-Chain Processing

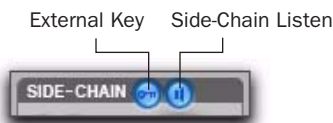
To use a filtered or unfiltered external key input to trigger dynamics processing:

- 1 Click the Key Input selector and select the input or bus carrying the audio from the reference track or external audio source.



Selecting a Key Input

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



Side-Chain section

- 3 To listen to the signal that will be used to control side-chain input, click Side-Chain Listen to enable it (highlighted).
- 4 To filter the key input so that only specific frequencies trigger the plug-in, use the HF and LF controls to select the desired frequency range.
- 5 Begin playback. The plug-in uses the input or bus that you chose as an external key input to trigger its effect.
- 6 Adjust the plug-in's Threshold control to fine-tune external key input triggering.
- 7 Adjust other controls to achieve the desired effect.

Using a Filtered Input Signal for Side-Chain Processing

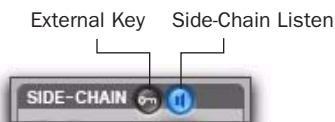
To use the filtered input signal to trigger dynamics processing:

- 1 Ensure the Key Input selector is set to No Key Input.



Key Input selector


- 2 Ensure that the External Key button is disabled (dark gray).



Side-Chain section

- 3 To listen to the signal that will be used to control side-chain input, click Side-Chain Listen to enable it (highlighted).
- 4 To filter the side-chain input so that only specific frequencies within the input signal trigger the plug-in, use the HF and LF controls to select the desired frequency range.
- 5 Begin playback. The plug-in uses the filtered input signal to trigger dynamics processing.
- 6 To fine-tune side-chain triggering, adjust the plug-in controls.

Dynamics II

 For information, see the Digidesign Legacy Plug-ins Guide.

Mod Delay II



Mod Delay II plug-in (Long Delay shown)

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

- The Short Delay provides 43 ms of delay at all sample rates.
- The Slap Delay provides 171 ms of delay at all sample rates.

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

- The Medium Delay provides 341 ms of delay at all sample rates.
- The Long Delay provides 683 ms of delay at all sample rates.
- The Extra Long Delay provides 2.73 seconds of delay at all sample rates.

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

Mod Delay II Controls

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

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

LPF (Low-Pass Filter) Controls the cutoff frequency of the Low-Pass Filter. Use the LPF setting to attenuate the high frequency content of the feedback signal. The lower the setting, the more high frequencies are attenuated. The maximum value for LPF is Off. This lets the signal pass through without limiting the bandwidth of the plug-in.

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

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

Rate Controls the rate of modulation of the delayed signal.

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

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

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

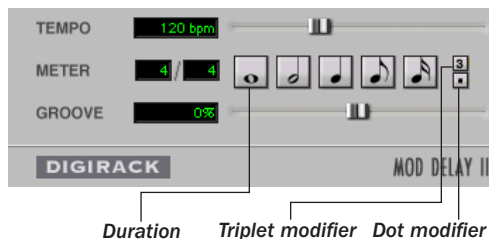
For more information, see “Tempo Sync” on page 23.

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

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

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

Duration Specifies a desired delay from a musical perspective. Enter the desired delay by selecting appropriate note value (whole note, half note, quarter note, 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 range. If you encounter this behavior, switch to a version of Mod Delay II that has a longer delay time (for example, switch from Medium Delay to Long Delay).*

Multichannel Mod Delay II


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



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


Signal Generator

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

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



Signal Generator plug-in


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

Signal Generator Controls

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

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

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

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

Peak Generates signal at the maximum possible level without clipping.

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

SignalTools

DigiRack SignalTools metering plug-ins provide two metering modules: SurroundScope and PhaseScope.

The SignalTools plug-ins support TDM and RTAS plug-in formats at all sample rates.

SurroundScope

(Pro Tools HD Only)

SurroundScope is a plug-in that provides surround metering for multichannel track types from 3 channels (LCR) to 8 channels (7.1 surround). Stereo and mono tracks are not supported.

💡 *This version of SurroundScope is compatible with sessions that used the previous versions of SurroundScope.*



Surround Display

SurroundScope detects the multi-channel format of the track and displays each channel in the signal in a circle around the Surround Display.



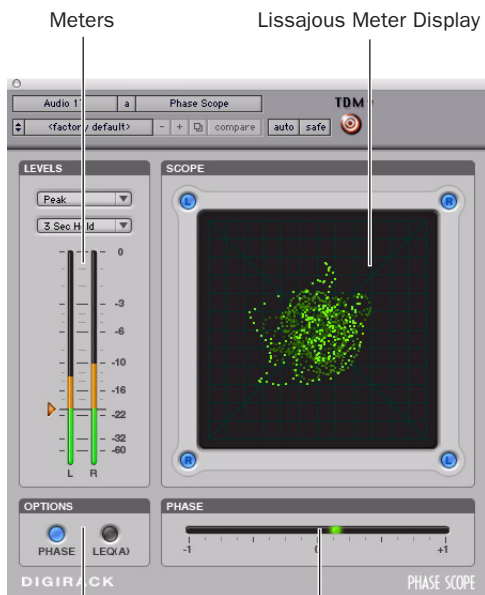
SurroundScope Surround Display (5.0 shown)

The Surround Display generates a composite image that indicates relative signal strength in the displayed channels.

- ◆ A circle in the center of the display indicates a surround signal that is panned equally to all channels.
- ◆ An irregular shape that is closer to one side of the display indicates that the channels on that side have a stronger signal.
- ◆ A teardrop shape that points toward a single channel indicates that the signal is panned to that channel.

PhaseScope

PhaseScope is a multichannel metering plug-in that provides signal level and phase information for stereo tracks only. (Mono and LCR or greater multichannel tracks are not supported.)



Display Options

Phase/Leq(A) Display

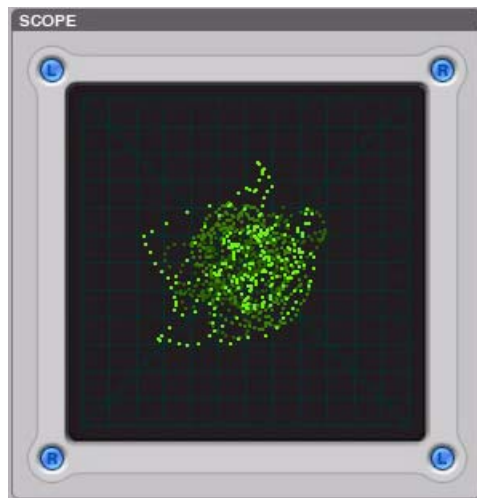
PhaseScope

Lissajous Meter Display

The PhaseScope Lissajous Meter displays the relationship between the amplitude and phase of a stereo signal, enabling you to monitor stereo imaging graphically.



A “Lissajous curve” (also known as a Lissajous figure or Bowditch curve) is a type of graph that is able to describe complex harmonic motion. To learn more, search the Web or your local library for information on its origins and its two principal developers, Jules Antoine Lissajous, and Nathaniel Bowditch.



PhaseScope Lissajous Meter Display

The Lissajous Meter display is divided into four quadrants, with left and right channels arranged diagonally. When audio is panned predominantly to a particular speaker channel, a diagonal line appears, indicating the channel.

The Lissajous Meter displays in-phase material as a vertical line and out-of-phase material as a horizontal line.

SignalTools Display Options

Both SignalTools plug-ins offer two display options: Phase Meter Display and Leq(A) Meter Display.

To choose a display option:

- Click the corresponding button in the Options section of the plug-in window.



SignalTools display options

Phase Meter Display

The Phase Meter indicates the phase coherency of two channels of a multi-channel signal.



SignalTools Phase Meter

The Phase Meter is green when the channels are positively out of phase (values from 0 to +1) and red when the channels are negatively out of phase (values from 0 to -1).

At the center or zero position, the signal is a perfect stereo image. At the +1 position, the signal is a perfect mono image. At the -1 position, the signal is 100% out of phase.

SurroundScope With SurroundScope you can select the two channels to compare by clicking the channel buttons around the Surround Display. Selected channels are indicated in blue.



Selecting SurroundScope channels for phase metering

PhaseScope With PhaseScope, the left and right channels are always compared.

Leq(A) Meter Display

The Leq(A) Meter display lets you view the *true weighted average* of the power level sent to any channel or combination of channels (except the LFE channel) in a multichannel track.

The Leq(A) Meter display shows a floating average for the level over the interval chosen in the Window menu. For example, with a setting of 2 seconds, the display shows the average value for the most recent 2 seconds of audio playback.



SignalTools Leq(A) meter and controls

Selecting Channels for Leq(A) Metering

SurroundScope With SurroundScope, you can select any combination of channels for Leq(A) metering by clicking the channel buttons around the Surround Display. Selected channels are indicated in green.



Selecting SurroundScope channels for Leq(A) metering

PhaseScope With PhaseScope, you can select either or both channel for Leq(A) metering by clicking the channel buttons in the corners of the Lissajous display. Selected channels are indicated in green.



Selecting PhaseScope channels for Leq(A) metering

Leq(A) Metering Controls

Window The Leq(A) window menu lets you choose the length of time the signal is measured before an average value is calculated. Settings range from 1 second to 2 minutes.

When the Leq(A) meter is in INF (infinite) mode it is constantly averaging the signal without a floating averaging window.

Reset The Reset button lets you manually reset the start time of the Leq(A) measurement window.

Auto Reset When enabled, causes the start time of the Leq(A) measurement window to be automatically reset whenever playback starts in Pro Tools.

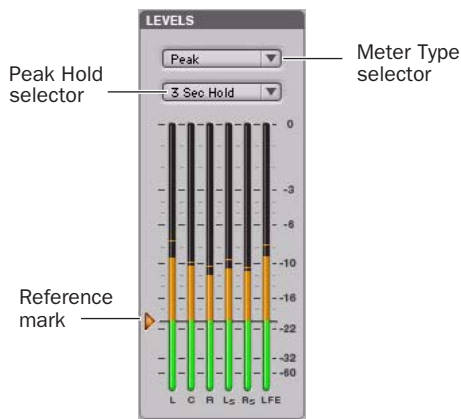
Hold on Stop When enabled, causes the Leq(A) measurement window timer to pause when playback stops, and resume when playback begins again.



In any of the Loop Transport modes, the measurement start time is automatically reset each time playback goes back to the beginning of the loop.

SignalTools Level Meters

The SignalTools plug-ins let you choose the type of metering to use. Each meter type has a different metering scale and response.




SignalTools level meters

Meter Types

Peak (Default meter type) Uses the metering scale in DigiRack EQ III and Dynamics III plug-ins.

RMS (Root Mean Square) was used in previous versions of the Digidesign SurroundScope plug-in and uses the same “true” RMS metering scale.

 *The “true” RMS meter scale is not the same as the AES 17 RMS scale. For a sine wave with a peak value of -20 dBFS, the “true” RMS meter will show a value of -23 dBFS. (The same sine wave will show a value of -20 dBFS on an AES 17 RMS meter.,*

Peak + RMS Uses a multi-color display to show both types of metering. Peak metering is shown in conventional green color, while RMS metering is shown in blue.


VU (Volume Unit) Uses AES standards for signal level indication.

BBC Uses IEC-IIa standards for signal level indication. This style of metering suppresses short duration peaks that would not affect broadcast program material. Reference calibration (4 dB) is -18 dBFS.

Nordic Uses IEC Type I standards for signal level indication and provides greater resolution for readings between -10 dBu and $+4$ dBu. Reference calibration (0 dB) is -18 dBFS.

DIN Uses IEC Type I standards for signal level indication and provides greater resolution for readings between -10 dBu and $+5$ dBu. Reference calibration (-9 dB) is -18 dBFS.

VENUE Provides Peak metering behavior with a meter scale calibrated specifically for Digidesign VENUE systems. Reference calibration (0 dB) is -20 dBFS.

 *Meter values are always displayed on control surfaces in dBFS values, regardless of the Meter Type setting.*

Meter Peak Hold Options

Each of the SignalTools plug-ins lets you choose the style of peak hold when peaks are shown in the plug-in meters.


3 Second Displays peak levels for 3 seconds

Infinite Displays peak levels until meters are cleared

No Peak Hold Does not display peak levels

Reference Mark Options

Each SignalTools meter type lets you adjust the level of the reference mark on the side of the meter display. The mark is set by default to the reference level for the corresponding meter type.

 *SignalTools meters also change color to show different ranges of level. The relative range of color automatically adjusts to follow the current Reference Mark setting in all meter types (except Peak+RMS).*

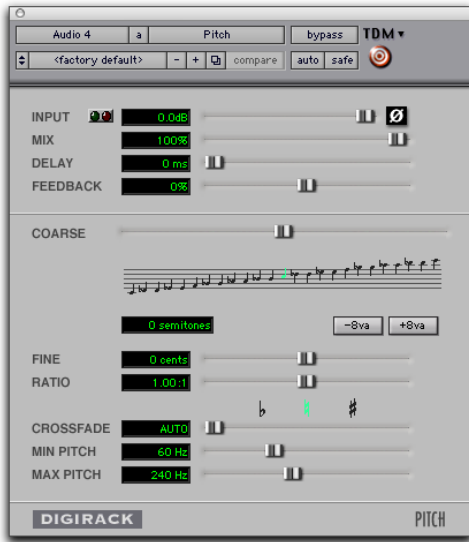
To change the reference level of a SignalTools meter:

- Drag the reference mark to a different location on the meter scale.

Pitch

(Pro Tools HD Only)

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



Pitch plug-in

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

Pitch transposition with the Pitch plug-in involves a much more complex technique: digitally adding or subtracting portions of the audio waveform itself, while using de-glitching 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 (Mac) on the staff will set the coarse pitch change value to zero.



Relative pitch entry

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

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

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

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

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

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

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

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

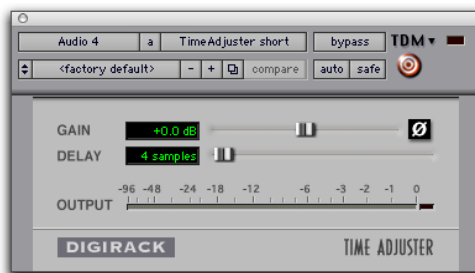
The range of this slider is from 15 Hz to 1000 Hz. The default setting is 60 Hz. Adjustment is tied to the current setting of the Maximum Pitch control so that the minimum range is never less than one octave, and the maximum range never more than five octaves.

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

TimeAdjuster

The TimeAdjuster plug-in provides three functions:

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



TimeAdjuster plug-in



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

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

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

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

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

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

TimeAdjuster Controls

Phase Invert Inverts the phase (polarity) of the input signal. While most Digidesign plug-ins supply a phase invert button of their own, some third-party plug-ins may not. Phase inversion is also useful for performing delay compensation by tuning unknown delay factors by ear (see “Using TimeAdjuster for Manual Delay Compensation” on page 90).

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


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

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


Using TimeAdjuster for Manual Delay Compensation

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

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

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

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

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

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

Phase Inversion

If you are working with phase-coherent track pairs, or tracks recorded with multiple microphones, you can invert the phase to negate the delay. If you don’t hear any audio when you invert a signal’s phase, you have precisely adjusted and compensated for the delay. This is because when you monitor duplicate signals and invert

the polarity (phase) of one of them, the signals will be of opposite polarity and cancel each other out. This technique is convenient for finding the exact delay setting for any plug-in.

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

- 1 Place duplicate audio regions on two different audio tracks and pan them to the center (mono).
- 2 Apply the plug-in whose delay you want to calculate to the first track, and a Time Adjuster plug-in to the second track.
- 3 With TimeAdjuster, invert the phase.
- 4 Control-drag (Windows) or Command-drag (Mac) to fine-tune delay in one sample increments, or use the up/down arrow keys to change the delay one sample at a time until the audio signal disappears.
- 5 Change the polarity back to normal.
- 6 Save the TimeAdjuster setting for later use.

Comb-Filter Effect Cancellation

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

Viewing Channel Delay

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

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

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




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

- 2 Apply the TimeAdjuster plug-in to the track whose delay you want to increase, and Control-click (Windows) or Command-click (Mac) its Track Level indicator until the channel delay value is displayed for that track.
- 3 Change the delay time in TimeAdjuster by moving the Delay slider or entering a value in the Delay field, until the channel delay value matches that of the first track.
- 4 Test the delay values by duplicating an audio track and reversing its phase while compensating for delay.

When to Compensate for Delays

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

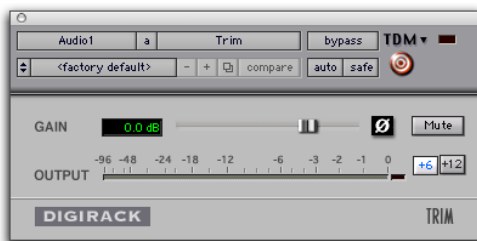
However, this may not always be necessary. You may only really need to compensate for delays between tracks where phase coherency must be maintained (as with instruments recorded with multiple microphones or stereo pairs). If you are working with mono signals, and the accumulated delays are small (just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

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

Trim

The Trim plug-in can be used to attenuate an audio signal from $-\infty$ (Infinity) dB to +6 dB or $-\infty$ (Infinity) dB to +12 dB. For example, using a multi-mono Trim plug-in on a multi-channel track provides simple, DSP-efficient muting control over the individual channels of the track.

This capability is useful, since Track Mute buttons mute *all* channels of a multi-channel track and do not allow muting of individual channels within the track.



Trim plug-in

💡 *Alt-click (Windows) or Option-click (Mac) 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 or +12 dB of gain adjustment, depending whether the Gain toggle is set to +6 or +12.

+6/+12 Gain toggle Switches the maximum level of attenuation between $-\infty$ dB to +6 dB and $-\infty$ dB to +12 dB.

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

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

Mute Mutes the signal output.

📖 *See also “Linking and Unlinking Controls on Multi-Mono Plug-ins” on page 21.*

chapter 7

DigiRack AudioSuite Plug-ins

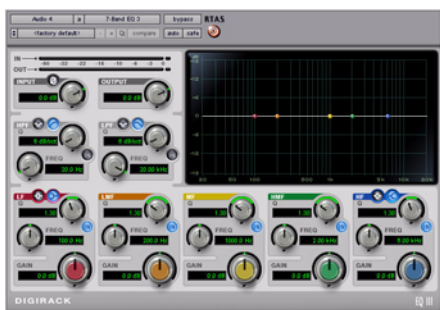
EQ III

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

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

7 Band EQ and 2–4 Band EQ

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



7 Band EQ and 2–4 Band EQ window


1 Band EQ

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




1 Band EQ window

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

 For information on the individual controls for these plug-ins, see “EQ III” on page 43

EQ II


 For information, see the Digidesign Legacy Plug-ins Guide.

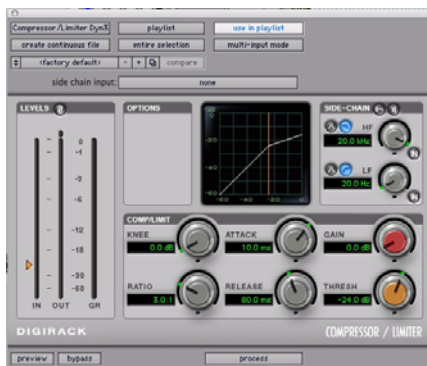
Dynamics III

Dynamics III provides three modules:

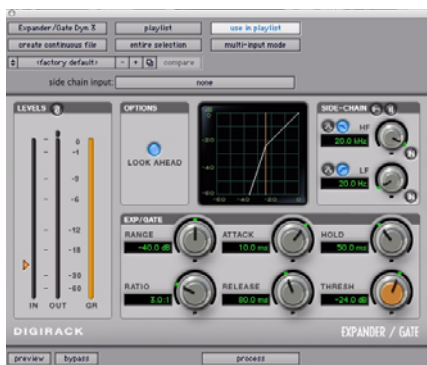
- Compressor/Limiter
- Expander/Gate
- De-Esser

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

 For information on individual controls in these plug-ins, see “Dynamics III” on page 63.



Compressor/Limiter 3 plug-in




Expander/Gate 3 plug-in



De-Esser 3 plug-in

Dynamics II

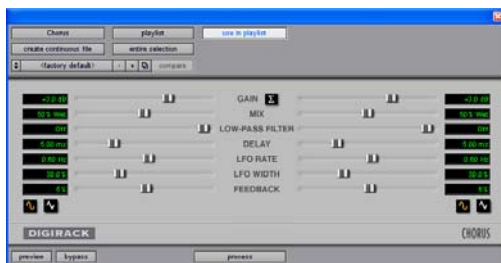
 For information, see the Digidesign Legacy Plug-ins Guide.

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 combining 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.

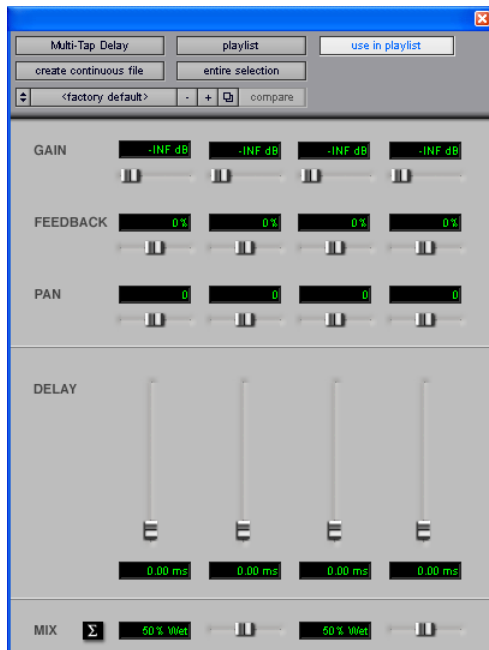
LFO Waveform Selects a sine wave or triangle wave for the LFO. This affects the character of the modulation. The sine wave has a gentler ramp and peak than the triangle wave.

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

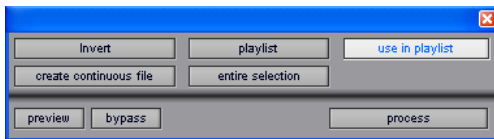
Because Ping-Pong Delay adds additional material (a delay tap) to the end of selected audio, make a selection that is longer than the original source material to allow the delay tap to be written into the end of the audio file.

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.

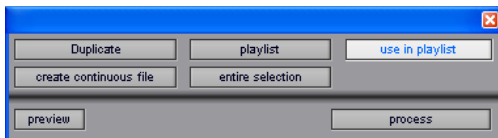
This process is useful for altering the phase or *polarity* relationship of tracks. The Invert plug-in is useful during mixing for modifying frequency response between source tracks recorded with multiple microphones. You can also use it to correct audio recorded out of phase with an incorrectly wired 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 79.*

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

 *For information on the individual controls for the AudioSuite Short Delay, Slap Delay, Medium Delay, and Long Delay plug-ins, see “Mod Delay II” on page 79.*

Selecting Audio Delay Processing

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

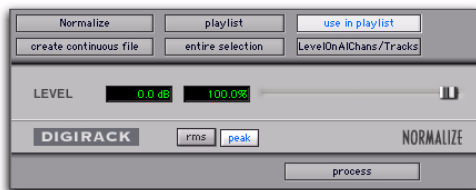
Selecting only the original material, without leaving additional space at the end, will cause delay taps that occur after the end of the region to be cut off.

Normalize

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

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

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



Normalize plug-in

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

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

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

Normalizing Multiple Regions Across Tracks

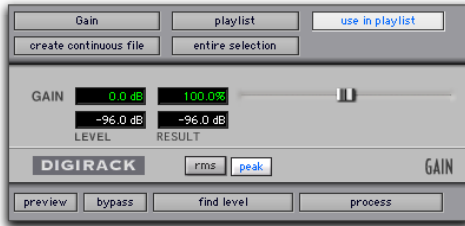
When multiple regions are selected across multiple tracks, the Normalize plug-in can search for peaks in two different modes:

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

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

Gain

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



Gain plug-in

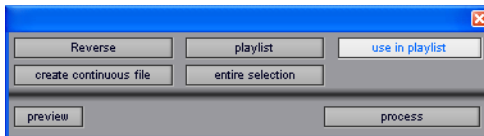
Gain Specifies the desired gain level. Set this value by manually adjusting the Gain slider, or by entering a numeric decibel value, or by entering a percentage.

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

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

Reverse

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



Reverse plug-in

Signal Generator

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



Signal Generator plug-in

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

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

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

- 1 Make a selection in the Tracks window.
- 2 Choose AudioSuite > Signal Generator.
- 3 Enter values for the Frequency, Level, and Signal controls.
- 4 Click Process in the Signal Generator plug-in.

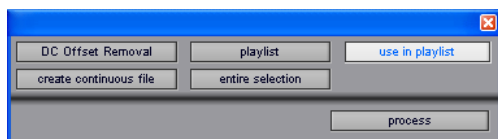
💡 *Select the Create Continuous File option for greater flexibility in making audio selections for use with the Signal Generator plug-in.*

📖 *For information on the individual controls for the Signal Generator plug-in, see “Signal Generator” on page 81.*

DC Offset Removal

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

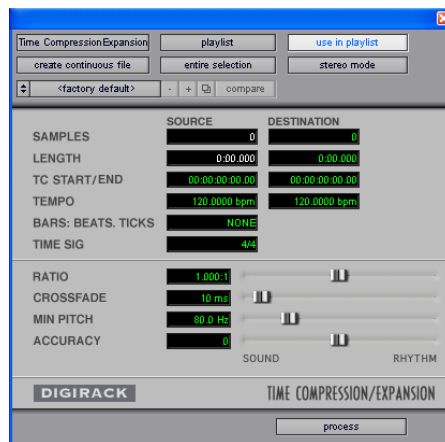
To check for DC offset, find a silent section in the audio material. If DC offset is present, a near-vertical fade-in with a constant or steady-state offset from zero will appear in the waveform. Use the DC Offset Removal plug-in to remove it.



DC Offset Removal plug-in

Time Compression/Expansion

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



Time Compression/Expansion plug-in

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

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

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

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

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

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

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

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


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

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

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

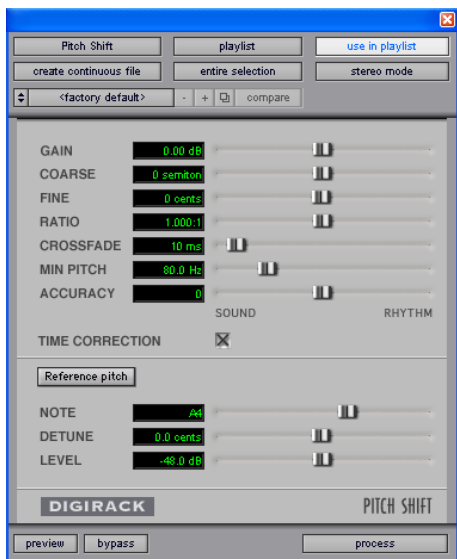
Accuracy Prioritizes the processing resources allocated to audio quality (Sound) or timing (Rhythm). Moving the slider towards "Sound" generally results in better sonic quality and fewer audio artifacts. Moving the slider towards "Rhythm" puts the emphasis on keeping the tempo consistent.

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

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

Pitch Shift

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



Pitch Shift plug-in

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

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

Time Correction Disabling this option has the effect of “permanently varispeeding” your audio file. The file’s duration will be compressed or extended according to the settings of the Coarse and Fine pitch controls. When Time Correction

is enabled, fidelity can be affected. For example, time expansion as a result of Time Correction when lowering pitch can cause the audio to sound granulated.

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

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

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

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

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

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

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

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

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

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

Gain Adjusts the volume of the reference tone in dB.

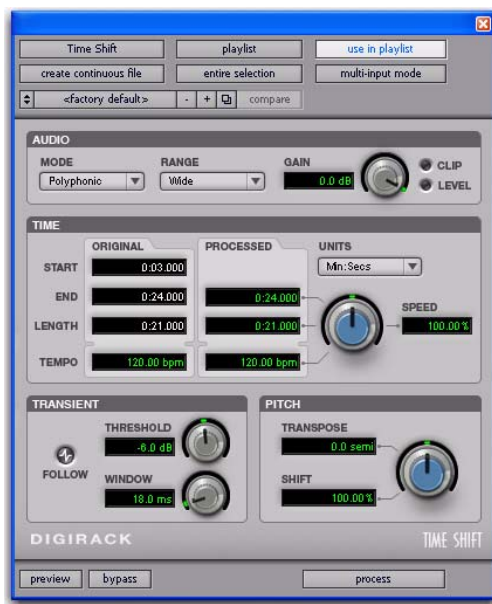
To use the Reference Pitch:

- 1 Select the audio material you want to use as a pitch reference. Click the preview button to begin playback of the selected audio.
- 2 Click the Reference Pitch button to activate the reference sine wave tone.
- 3 Adjust the Note and Detune settings to match the reference tone to the pitch of the audio playback. Adjust the Gain setting to change the relative volume of the reference tone. It may also be helpful to toggle the Reference Pitch on and off to compare pitch.
- 4 Select the audio material to be pitch shifted.
- 5 Adjust the Coarse and Fine Pitch Shift controls to match the pitch of the audio playback to the reference pitch.
- 6 Click Process to apply pitch shift to the selection.

Time Shift

Time Shift provides high quality time compression and expansion (TCE) algorithms and formant correct pitch-shifting. Time Shift is ideal for music production, sound design, and post production applications. Use it to manipulate audio loops for tempo matching or to transpose vocal tracks using formant correct pitch shifting. You can also use it in audio post production for pull up and pull down conversions as well as for adjusting audio to specific time or SMPTE durations for synchronization purposes.

Time Shift Displays and Controls



Time Shift plug-in

The interface for Time Shift is organized in four parts: Audio, Time, Formant or Transient, and Pitch.

Audio Use the controls in the Audio section to select the most appropriate time compression and expansion algorithm (mode) for the type of material you want to process, and to attenuate the gain of the processed audio to avoid clipping.

Time Use the controls in the Time section to specify the amount of time compression or expansion you want to apply.

Formant or Transient Use the controls in the Formant or Transient section to adjust either the amount of formant shift or the transient detection parameters depending upon which mode you have selected in the Audio section. The Formant section is only available when Monophonic is selected as the Audio Mode. The Transient section is available with slightly different controls depending on whether Polyphonic or Rhythmic is selected as the Audio Mode.

Pitch Use the controls in the Pitch section to apply pitch shifting. Pitch shifting can also be formant correct if you select the Monophonic audio setting.

Audio

The Audio section of Time Shift provides controls for specifying the type of audio you want to process and gain attenuation of the processed signal to avoid clipping.



Time Shift plug-in, Audio section

Mode

The Audio Mode pop-up menu determines the type of TCE and pitch shift algorithm for processing audio: Monophonic, Polyphonic, Rhythmic, or Varispeed.

Monophonic Use for processing monophonic sounds (such as a vocal melody).

Polyphonic Use for processing complex sounds (such as a multipart musical selection).

Rhythmic Use for processing percussive sounds (such as a mix or drum loop).



Rhythmic mode uses transient analysis for time shifting. If you select audio with no apparent transients, or set the Transient Threshold control to a setting above any detected transients, Time Shift assumes a “virtual-transient” every three seconds to be able to process the file. Consequently, the file should be 20 bpm or higher (one beat every three seconds) to achieve desirable results. For material that has no apparent transients, use Monophonic or Polyphonic mode.

Varispeed Use to link time and pitch change for tape-like pitch and speed change effects, and post production workflows.

Range

The Audio Range pop-up menu determines the frequency range for analysis: Low, Mid, High, or Wide. For low-range material, such as a bass guitar, select Low. For mid-range material, such as male vocals, select Mid. The high range control is for material with a high fundamental frequency, such as female vocals. For more complex material that covers a broad frequency spectrum, select Wide.

In Polyphonic mode, Wide is the default setting and is usually best for all material when using the Polyphonic audio type.

In Monophonic mode, Mid is the default setting and is usually matches the range of most monophonic material.

The range pop-up menu is unavailable in Rhythmic mode and Varispeed mode.

Gain

The Audio Gain control attenuates the input level to avoid clipping. Adjust the Gain control from 0.0 dB to -6.0 dB to avoid clipping in the processed signal.

Clip Indicator

The Clip indicator indicates clipping in the processed signal. When using time compression or pitch shifting above the original pitch, it is possible for clipping to occur. The Clip indicator lights when the processed signal is clipping. If the processed signal clips, undo the AudioSuite process and attenuate the input gain using the Gain control. Then, re-process the selection.

Level Indicator

The Level indicator displays the level of the output signal using a plasma LED, which uses the full range of plasma level metering colors.

Time

The Time section of Time Shift provides controls for specifying the amount of time compression or expansion as well as the timebase used for calculating TCE. Adjust the Time control to change the target duration for the processed audio.



Time Shift plug-in, Time section

Original Displays the Start and End times, and Length of the edit selection. Times are displayed in units of the timebase selected in the Units pop-up menu.

Processed Displays the target End time and Length of the processed signal. Times are displayed in units of the timebase selected in the Units pop-up menu. You can click the Processed End and Length fields to type the desired values. These values update automatically when adjusting the Time control.

Tempo Displays the Original Tempo and Processed Tempo in beats per minute (bpm). You can click the Original Tempo and Processed Tempo fields to type the desired values. The Processed Tempo value updates automatically when adjusting the Time control.

Unit Select the desired timebase for the Original and Processed time fields: Bars|Beats, Min:Sec, Time Code, Feet+Frames, or Samples.

⚠ *Time Shift does not receive Bars|Beat and Feet+Frame information from Pro Tools 7.0 or 7.1. Consequently, Bars|Beats and Feet+Frames are displayed as "N/A."*

Speed Displays the target time compression or expansion as a percentage of the original. Adjust the Time control or click the Speed field and type the desired value. Time can be changed from 25.00% to 400.00% of the original speed (or 4 to 1/4 times the original duration). The default setting is 100.00%, or no change. 25.00% results in 4 times the original duration and 400.00% results in 1/4 of the original duration.

The Speed field only displays up to 2 decimal places, but lets you type in as many decimal places as you want (up to the IEEE standard). While the display rounds to 2 decimal places, the actual time shift is applied based on the

number you typed. This is especially useful for typing post production pull up and pull down factors (see “Post Production Pull Up and Pull Down Tasks” on page 110).

Formant

The Formant section of Time Shift lets you shift the formant shape of the selected audio independently of the fundamental frequency. This is useful for achieving formant correct pitch shifting. It can also be used as an effect. For example, you can formant shift a male vocal up by five semitones and it will take on the characteristics of a female voice.

The Formant section is only available when Monophonic is selected as the Audio Type. The Formant section provides a single control for transposing the formants of the selected audio by -24.00 semitones (-2 octaves) to $+24.00$ semitones ($+2$ octaves), with fine resolution in cents. Adjust the Formant Shift control or click the Shift field and type the desired value.



Time Shift plug-in, Formant section



Audio with a fundamental pitch has an overtone series, or set of higher harmonics. The strength of these higher harmonics creates a formant shape, which is apparent if viewed using a spectrum analyzer. The overtone series, or harmonics, have the same spacing related to the pitch and have the same general shape regardless of what the fundamental pitch is. It is this formant shape that gives the audio its overall characteristic sound or timbre. When pitch shifting audio, the formant shape is shifted with the rest of the material, which can result in an unnatural sound. Keeping this shape constant is critical to formant correct pitch shifting and achieving a natural sounding result.

Transient

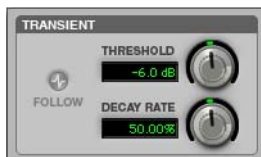
The Transient section is only available when Polyphonic or Rhythmic is selected as the Audio Type, and provides slightly different controls for each.

When Polyphonic is selected as the Audio Type, the Transient section provides controls for setting the transient detection threshold and for adjusting the analysis window length for processing audio.



Time Shift plug-in, Transient section with Polyphonic selected as the Audio Type

When Rhythmic is selected as the Audio Type, the Transient section provides controls for setting the transient detection threshold, and for adjusting the decay rate of the transients in the processed audio when time stretching.



Time Shift plug-in, Transient section with Rhythmic selected as the Audio Type

Follow Enables an envelope follower that simulates the original acoustics of the audio being stretched. Click the Follow button to enable or disable envelope following. Follow is only available when Polyphonic is selected as the Audio Type.

Threshold Sets the transient detection threshold from 0.0 dB to -40.0 dB. Disable transient detection by setting the Threshold control to Off (turn the knob all the way to the right). Part of Time Shift's processing relies upon separating "transient" parts of the selection from "non-transient" parts. Transient material tends to change its content quickly in time, as opposed to parts of the sound which are more sustained. Adjust the Threshold control or click the Threshold field and type the desired value.

The default value for Threshold is -6.0 dB. For highly percussive material, lower the threshold for better transient detection, especially with the Rhythmic audio setting. For less percussive material, and for shifting with the Polyphonic audio setting, a higher setting can yield better results. Experiment with this control, especially when shifting drums and percussive tracks, to achieve the best results.

Window Sets the analysis window length for processing audio. You can set the Window from 6.0 milliseconds to 185.0 milliseconds. Adjust the Window control or click the Window field and type the desired value. The Window control is only available when Polyphonic is selected as the Audio Type.

The default for Window size is 18.0 milliseconds and works well for many applications, but you may want to try different Window settings to get the best results. Try larger window sizes for low frequency sounds or sounds that do not have many transients. Try smaller window sizes for drums and percussion. 37.0 milliseconds tends to work well for polyphonic instruments such as piano or guitar. A setting as large as 71.0 milliseconds works well for bass guitar. Settings in the 12 millisecond range work well on drums or percussion.

Decay Rate Determines how much of the decay from a transient is heard in the processed audio when time stretching. When time stretching using the Rhythmic setting, the resulting gaps between the transients are filled in with audio, and Decay Rate determines how much of this audio is heard by applying a fade out rate. Decay Rate is only available when Rhythmic is selected as the Audio Type. Adjust the Decay Rate up to 100% to hear the audio that is filling the gaps created by the time stretching with only a slight fade, or adjust down to 1.0% to completely fade out between the original transients.

Pitch

The Pitch section provides controls for pitch shifting the selected audio. Use the Pitch control to transpose the pitch from –24.00 semitones (–2 octaves) to +24.00 semitones (+2 octaves), with fine resolution in cents.



Time Shift plug-in, Pitch section

Transpose Displays the transposition amount in semitones. You can transpose pitch from –24.00 semitones (–2 octaves) to +24.00 semitones (+2 octaves), with fine resolution in cents. Adjust the Pitch control or click the Transpose field and type the desired value.

Shift Displays the pitch shift amount as a percentage. You can pitch shift from 25.00% (–2 octaves) to +400.00% (+2 octaves). Adjust the Pitch control or click the Shift field and type the desired value. The default value is 100% (no pitch shift).

Post Production Pull Up and Pull Down Tasks

Table 8 on page 110 provides information on TCE settings for common post production tasks. Type the corresponding TCE% (represented to 10 decimal places in Table 8) in the Time Shift field for the corresponding post production task and the process the selected audio.



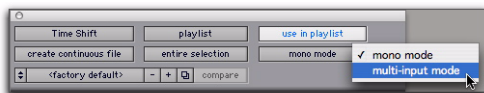
Use the corresponding Time Shift Plug-in Setting for the desired post production task.

Table 8. Most commonly used TCE settings in Post workflows

Post workflow plug-in setting	TCE% (to 10 decimal places)	Frames
Pal to Film –4%.tfx	96.0%	25 to 24/30
PAL to NTSC –4.1%.tfx	95.9040959041%	25 to 23.976/29.97
Film to PAL +4.1667%.tfx	+104.1666666667%	24/30 to 25
Film to NTSC –0.1%.tfx	99.9000999001%	24/30 to 23.976/29.97
NTSC to Pal +4.2667%.tfx	+104.2708333333%	23.976/29.97 to 25
NTSC to Film +0.1%.tfx	+100.10%	23.976/29.97 to 24/30

AudioSuite Input Modes

Time Shift supports the Pro Tools AudioSuite Input Mode selector for use on mono or multi-input processing.



AudioSuite Input mode selector

Mono Mode Processes each audio region as a mono file with no phase coherency maintained with any other simultaneously selected regions.

Multi-Input Mode Processes up to 48 input channels and maintains phase coherency within those selected channels.

AudioSuite Preview

Time Shift supports Pro Tools AudioSuite Preview and Bypass. For more information on using AudioSuite Preview and Bypass, see the DigiRack Plug-ins Guide.

⚠ *AudioSuite Preview and Bypass are unavailable with Time Shift in Pro Tools 7.0 and 7.1.*


AudioSuite TCE Plug-in Preference

The Time Shift plug-in's high quality time compression and expansion algorithms that can be used with the Pro Tools TCE Trim tool.



TCE Plug-in option in Processing Preferences page

⚠ *Time Shift is not available with the TCE Trim tool in Pro Tools 7.0 and 7.1.*

 *Refer to the Pro Tools Reference Guide for more information about the TCE Trim tool.*

To select Time Shift for use with the TCE Trim tool:

- 1 Choose Setup > Preferences.
- 2 Click the Processing tab.
- 3 From the TC/E Plug-in pop-up menu, select Time Shift.
- 4 Select the desired preset setting from the Default Settings pop-up menu.
- 5 Click OK.

Processing Audio

Time Shift lets you change the time and pitch of selected audio independently or concurrently.



Normalizing a selection before using Time Shift may produce better results.

To change the time of a selected audio region:

- 1 Select AudioSuite > Pitch Shift > Time Shift.
- 2 Select the Audio Mode appropriate to the type of material you are processing (Monophonic, Polyphonic, or Rhythmic).
- 3 In Monophonic or Polyphonic mode, select the appropriate Range for the selected material (Low, Mid, High, or Wide).
- 4 If compressing the duration of the selection, attenuate the Gain control as necessary.
- 5 If using Monophonic mode, adjust the Formant Shift control as desired.
- 6 If using Polyphonic or Rhythmic mode, adjust the Transient controls as desired.
- 7 Make sure Pitch Shift is set to 100% (unless you also want to change the pitch of the selection).
- 8 Adjust the Time Shift control to the desired amount of time change. Time change is measured in terms of the target duration using the selected timebase or as a percentage of the original.
- 9 Click Process.

To change the pitch of a selected audio region:

- 1 Select AudioSuite > Pitch Shift > Time Shift.
- 2 Select the Audio Mode appropriate to the type of material you are processing (Monophonic, Polyphonic, or Rhythmic).
- 3 In Monophonic or Polyphonic mode, select the appropriate Range for the selected material (Low, Mid, High, or Wide).
- 4 If transposing the pitch of the selection up, attenuate the Gain control as necessary.
- 5 If using Monophonic mode, adjust the Formant Shift control as desired.
- 6 If using Polyphonic or Rhythmic mode, adjust the Transient controls as desired.
- 7 Make sure Time Shift is set to 0% (unless you also want to change the duration of the section).
- 8 Adjust the Pitch Shift control to the desired amount of pitch change. Pitch change is measured in semitones (and cents) or as a percentage of the original.
- 9 Click Process.

To change the time and pitch of a selected audio region:

- 1 Select AudioSuite > Pitch Shift > Time Shift.
- 2 Select Varispeed from the Audio Mode pop-up menu.
- 3 Adjust either the Time Shift or Pitch Shift control to match the desired amount of time and pitch change in terms of a percentage of the original.
- 4 Click Process.



Using the Monophonic, Polyphonic, or Rhythmic modes, you can adjust both the Time Shift and Pitch Shift controls independently before processing.

chapter 8

Elastic Audio Plug-ins

Elastic Audio Plug-in Parameters

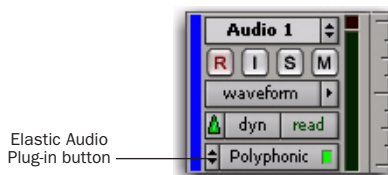
Pro Tools provides several TCE algorithms for real-time and rendered track-based Elastic Audio processing. The selected Elastic Audio plug-in determines how the audio is processed. Each plug-in uses different algorithms for Elastic Audio processing. Different algorithms are better suited to different types of audio. Select the plug-in whose algorithm is appropriate to the material on the track you want to process.

Elastic Audio plug-ins have their own unique selector that is only available in the Edit window. You cannot insert Elastic Audio plug-ins on track Plug-in Inserts (TDM, RTAS, and hardware inserts).

Elastic Audio plug-ins provide few to no adjustable parameters. The adjustable controls, if any, for Elastic Audio plug-ins are available in the Elastic Audio Plug-in window. Elastic Audio plug-in parameters cannot be automated.

To open an Elastic Audio Plug-in window:

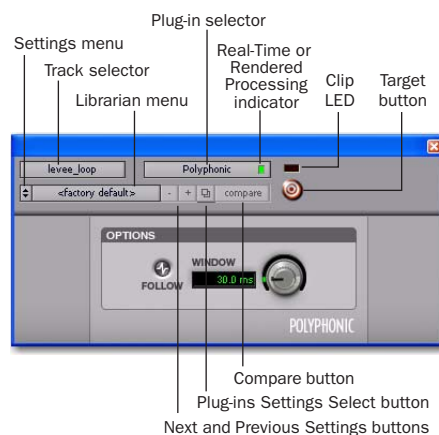
- Click the Elastic Audio Plug-in button.



Elastic Audio plug-in button

Elastic Audio Plug-in Window Controls

The Elastic Audio Plug-in window provides a set of controls in the window header that are common to all Elastic Audio plug-ins.



Elastic Audio Plug-in controls

Track Selector Accesses any other audio track in the session.

Plug-in Selector Lets you select any Elastic Audio plug-in or disable Elastic Audio.

Real-Time or Rendered Processing

Indicator Lights when Elastic Audio processing is real-time and dims when Elastic Audio processing is rendered (see “Real-Time and Rendered Elastic Audio Processing” on page 120).

Clip LED Lights red to indicate clipping in the Elastic Audio processing. Clipping can occur when time compressing audio. If clipping occurs, you can lower the Input Gain in the Elastic Properties window (see “Input Gain” on page 124), or attenuate the Default Input Gain in the Processing Preferences page (see “Elastic Audio Preferences” on page 125).

You can also undo any Elastic Audio processing and apply an appropriate amount of AudioSuite Gain reduction, and then try the Elastic Audio processing again. The Clip LED only functions for Real-Time Elastic Audio processing; it does not function for Rendered Elastic Audio processing.



For Rendered Elastic Audio processing, clipping is indicated by the appearance of the Elastic Audio Processing Clip indicator on the region.

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

Librarian Menu Recalls settings files saved in the plug-in’s root settings folder or in the current session’s Settings folder.

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



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

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

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

Target Button When multiple Elastic Audio Plug-in windows are open, clicking this button selects that plug-in as the target for any computer keyboard commands. Deselecting the Target button lets you open another targeted Elastic Audio Plug-in window.

Polyphonic

The Polyphonic plug-in is the default plug-in for previewing and importing. It is a general, all-purpose algorithm that is effective with a wide range of material. For complex loops and multi-instrument mixes, use the Polyphonic plug-in.



Elastic Audio plug-in: Polyphonic


Follow

Selecting the Follow option enables an envelope follower that simulates the original acoustics of the audio being stretched. The Follow button highlights when the Follow option is selected.

Window

The Polyphonic plug-in provides a single control for adjusting the analysis window length for TCE processing. Experiment with adjusting the Window size for different types of material until you achieve the best results. The following table provides some recommended Window sizes for different types of material.

Type of material	Recommended Window size
General purpose	30–40 ms
Percussive	20 ms or lower
Pads and other legato material	60 ms or higher

 *If you frequently work with distinct types of material, you may want to find the best Window settings for each type of material and save them as Plug-in Settings for quick and easy recall.*

Rhythmic

The Rhythmic plug-in is best suited to material with clear attack transients, such as drums.



Elastic Audio plug-in: Rhythmic

Decay Rate

The Decay Rate control determines how much of the decay from a transient is heard in the processed audio when time stretching. When the Rhythmic plug-in is selected, any gaps between transients resulting from time stretching are filled in with audio. The Decay Rate determines how much of this audio is heard by applying a fade out rate. Adjust the Decay Rate up to 100% to hear the audio that is filling the gaps created by the time stretching with only a slight fade, or adjust down to 1.0% to completely fade out between the original transients.

Monophonic

The Monophonic plug-in is best suited to monophonic material where you want to keep the formant relationships intact, such as with vocals. The Monophonic plug-in is also well suited to monophonic instrumental lines, such as a bass track. The Monophonic plug-in provides no plug-in specific parameters.



Elastic Audio plug-in: Monophonic

⚠ The Monophonic plug-in analyzes pitch as well as peak transients in order to provide higher quality TCE to pitched material. Consequently, analysis for the Monophonic algorithm takes longer than for the Polyphonic, Rhythmic, and Varispeed algorithms. Also, when switching from another Elastic Audio plug-in to Monophonic, Pro Tools reanalyzes any audio on the track to collect the extra analysis data.

Varispeed

Use the Varispeed plug-in to link time and pitch change for tape-like pitch and speed change effects, and post production workflows. The Varispeed plug-in provides no plug-in specific controls.



Elastic Audio plug-in: Varispeed

💡 Try using Varispeed with Quantize on drum loops for some interesting effects.

X-Form

(Rendered Only)

The X-Form Elastic Audio plug-in is a modified version of Digidesign's standalone X-Form AudioSuite processing plug-in. The X-Form Elastic Audio plug-in is for Rendered Elastic Audio processing only and cannot process in real-time or as an AudioSuite plug-in. It provides the highest quality time compression algorithms for music production, sound design, and audio loop applications.



Elastic Audio plug-in: X-Form

Quality

Select either Maximum or Low (Faster) from the Quality pop-up menu. Maximum is the slowest processing algorithm, but provides the highest quality results. Low (Faster) produces relatively good results and is much faster than the Maximum setting.

Formant

For audio material with clear formants, enable Formant to preserve the formant shape of audio when applying TCE processing.

💡 *Audio with a fundamental pitch has an overtone series, or set of higher harmonics. The strength of these higher harmonics creates a formant shape, which is apparent if viewed using a spectrum analyzer. The overtone series, or harmonics, have the same spacing related to the pitch and have the same general shape regardless of what the fundamental pitch is. It is this formant shape that gives the audio its overall characteristic sound or timbre. When pitch shifting audio, the formant shape is shifted with the rest of the material, which can result in an unnatural sound. Keeping this shape constant is critical to formant correct pitch shifting and achieving a natural sounding result.*

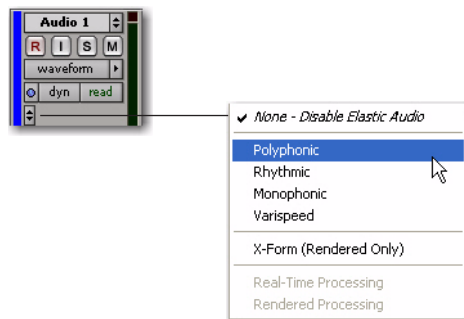
Elastic Audio Tracks

Any audio track in your Pro Tools session can be Elastic Audio enabled. Elastic Audio tracks can be either sample- or tick-based. However, only tick-based Elastic Audio tracks automatically apply TCE based on tempo changes.

📖 *For complete information on working with Elastic Audio, see the Pro Tools Reference Guide.*

To enable Elastic Audio on a track:

- 1 Create a new audio track or use an existing audio track.
- 2 From the track's Elastic Audio Plug-in selector, select the Elastic Audio plug-in appropriate to your track material.




Elastic Audio Plug-in Selector pop-up menu

Any audio regions on the track temporarily go offline while they are analyzed and come back online when the analysis is finished. The waveform for audio regions that are offline for Elastic Audio analysis appears grayed out.

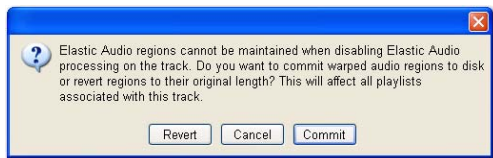
When recording to a real-time Elastic Audio-enabled track, once recording is stopped, the newly recorded audio temporarily goes offline while it is analyzed and comes back online as soon as the analysis is finished.

3 From the track's Elastic Audio Plug-in pop-up menu, select Real-Time Processing or Rendered Processing for Real-Time or Rendered Elastic Audio processing.

 *Elastic Audio processing can be Real-Time or Rendered. Real-Time Elastic Audio processing is immediate, but it is more demanding of system resources. Rendered Elastic Audio processing is non-real-time, but is less demanding of system resources. If you are working on a system with limited resources, use Rendered Elastic Audio processing. For more information, see “Real-Time and Rendered Elastic Audio Processing” on page 120.*

To disable Elastic Audio on a track:

- 1** From the track's Elastic Audio Plug-in selector, select *None – Disable Elastic Audio*.
- 2** If the track had any Elastic Audio-based processing, you are prompted to Cancel, Revert, or Commit.



Commit Elastic Audio dialog

Cancel Leaves Elastic Audio enabled on the track.

Revert Removes any Elastic Audio processing, reverts regions on the track to their original duration, and disables Elastic Audio on the track.

Commit Renders and commits any Elastic Audio processing on the track, writes new files to disk, and disables Elastic Audio on the track.

Committed Regions

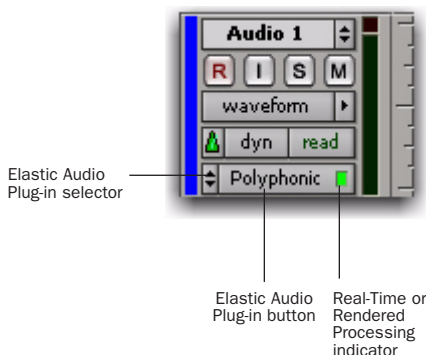
Regions that are committed, either by disabling Elastic Audio on a track or by moving a region to a track without Elastic Audio enabled, are written to disk as new audio files. These new audio files include the audio in the region, plus any fades, and also an additional 5 seconds of audio before and after the region, if available.

Alternate Playlists

Disabling Elastic Audio on a track affects all playlists on the track. All playlists on the track are committed or reverted depending on which option you choose.

Elastic Audio Track Controls

Audio tracks provide a new set of controls for Elastic Audio.



Elastic Audio Track controls

Elastic Audio Plug-in Selector Selects the Elastic Audio plug-in for Elastic Audio processing. You can also use the Elastic Audio Plug-in selector to disable Elastic Audio processing, and to select either Real-Time or Rendered processing (see “Real-Time and Rendered Elastic Audio Processing” on page 120).

Elastic Audio Plug-in Button Displays the name of the selected Elastic Audio plug-in. Click the Elastic Audio Plug-in button to open the Elastic Audio Plug-in window (see “Elastic Audio Plug-in Parameters” on page 113).

Real-Time or Rendered Processing

Indicator Lights when Elastic Audio processing is Real-Time and dims when Elastic Audio processing is Rendered (see “Real-Time and Rendered Elastic Audio Processing” on page 120).

Elastic Audio Analysis

When recording, pasting, moving, or importing un-analyzed audio to an Elastic Audio-enabled track, or when enabling Elastic Audio on an existing audio track, Pro Tools automatically analyzes the audio for transient events. In Waveform view, the waveform initially appears grayed out because regions go offline during Elastic Audio analysis. Once the analysis is complete, the audio comes back online. Elastic Audio analysis is file-based, which means that even if you are only working with a small region of a large file, the entire audio file is analyzed.

Elastic Audio analysis detects *transient events* in the audio file. These transient events are indicated by Event markers. Event markers are displayed in both the Warp and Analysis track views. Elastic Audio analysis also calculates the native tempo of the analyzed audio file and its duration in bars and beats.

Elastic Audio analysis data (detected events, tempo, and duration in Bars:Beats) is stored with the file. In DigiBase browsers, analyzed audio files are indicated by a check mark to the left of the file name, and these files display their duration in Bars|Beats, their timebase as ticks, and their native tempo in BPM.

Tempo Detection


Elastic Audio analysis does its best to detect a regular tempo for all analyzed audio. Any audio containing regular periodic rhythm can be successfully analyzed for tempo and duration in bars and beats. Analyzed files in which a tempo was detected are treated as tick-based files. Tick-based files can be conformed to the session tempo for preview and import.

Analyzed files in which no tempo was detected are treated as sample-based files. If there is only a single transient in the file (such as with a single snare hit), no tempo will be detected. Also, longer files that contain tempo changes or rubato, or that do not contain regular periodic rhythmic patterns will probably not have a detected tempo and will be treated as sample-based files.

Event Confidence

Transient events are detected with a certain degree of confidence. The level of confidence is based on the relative clarity of transients.


For example, a drum loop is likely to have clear, sharp transients. These will be detected with a high degree of confidence. However, a legato violin melody may not have clear, sharp transients, so transients will be detected with a lower degree of confidence.


 *Note that peak amplitude is not the most important measure for event confidence. The clarity of transients is measured in part by the spectral transition from one moment to the next. This tends to favor higher frequency content in terms of event confidence. For example, changing the Event Sensitivity in the Elastic Properties window for a selected drum loop results in the clearer transients of the higher frequency hi-hat hits having more event confidence than the less well defined transients of the lower frequency kick drum hits even though kick drum hits have a higher peak amplitude.*

Pro Tools is very aggressive with its transient detection in order to detect every possible audio event. Consequently, it is possible to have erroneously detected events. These events will be analyzed with a low degree of confidence and can be filtered out by lowering the Event Sensitivity in the Elastic Properties window. Filtering out any false transients can result in higher quality Elastic Audio processing.

Conversely, for material with few clear transients, Elastic Audio analysis may not accurately detect audio events, and those detected may be detected with a low level of confidence. You can relocate or add Event markers where necessary in Analysis view.


For example, a slow synth pad may not have clear transients and detected Event markers may be incorrectly located or may have not been detected at all. In Analysis view, you can relocate, remove, and add Event markers as necessary (such as where the chord changes). Accurately locating Event markers yields better results when applying Elastic Audio processing.

 *When processing non-rhythmic audio (such as legato strings) with the Polyphonic plugin, reducing the number of Event markers often yields better sounding results. You can either manually remove Event markers in Analysis view or lower the Event Sensitivity setting in the Elastic Properties window for the region you want to process (see “Event Sensitivity” on page 124).*

 *For information on editing in Analysis view, see the Pro Tools Reference Guide.*

Real-Time and Rendered Elastic Audio Processing

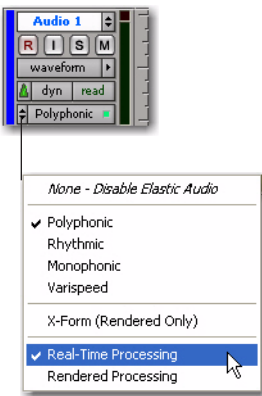
Real-Time Elastic Audio processing means that any changes to Elastic Audio processing—whether it be a tempo change or a manual TCE warp—take effect immediately. However, Real-Time Elastic Audio processing can be very demanding on your system’s resources. Consequently, you may want to use Rendered Elastic Audio processing in some cases. With Rendered Elastic Audio processing, when you make a change to Elastic Audio processing, any affected audio regions temporarily go offline, a new “rendered” audio file is generated and the affected audio regions come back online.

 *In Rendered mode, any change you make is rendered using the original source audio to avoid generational loss from multiple subsequent edits.*

To set an Elastic Audio track to Real-Time or Rendered:

- 1 Click the track's Elastic Audio Plug-in selector.
- 2 From the pop-up menu, select the desired Elastic Audio plug-in.
- 3 From the pop-up menu, select one of the following:
 - Real-Time Processing
 - Rendered Processing

💡 *Select Real-Time processing for tick-based Elastic Audio processing (such as tempo changes). Rendered audio processing can be useful for sample-based Elastic Audio processing or if you need to save system resources.*



Elastic Audio Plug-in Selector pop-up menu

⌘ *To toggle between Real-Time and Rendered Elastic Audio processing, Control-Start-click (Windows) or Command-Control-click (Mac) the Elastic Audio Plug-in button.*

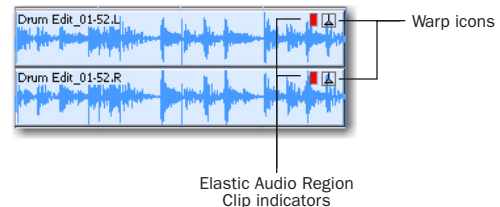
About Rendered Files

Pro Tools creates temporary files for Rendered Elastic Audio processing. These files are kept in a new, auto-created Rendered Files folder in the session folder. Once you commit Rendered Elastic Audio processing to a track (by disabling Elastic Audio on a track and clicking Commit in the Commit Elastic Audio dialog), a new file is written to disk in the Audio Files folder, and the temporary rendered file from which it was created is deleted from the Rendered Files folder.

Region Clip Indicator

(Rendered Processing Only)

Since Rendered Elastic Audio processing is not real-time, it does not report clipping due to Elastic Audio processing in the Elastic Audio plug-in window. Instead, when clipping occurs due to Rendered Elastic Audio processing, the region displays an Elastic Audio Processing Clip indicator just to the left of the Warp icon. To avoid clipping on playback, lower the Input Gain for the region in the Elastic Properties window (see “Elastic Properties Window” on page 122).



Elastic Audio Region Clip Indicator

Elastic Properties

Each region on an Elastic Audio-enabled track has specific Elastic Audio properties (such as tempo, meter, TCE factor, and Input Gain).

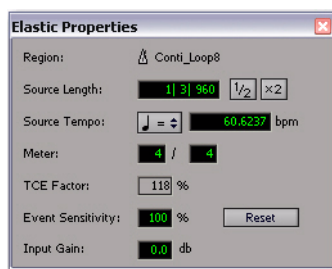
The Elastic Properties window lets you view and change the properties for selected regions. For example, you can reduce the Input Gain for Elastic Audio processing to avoid clipping that can occur during processing.

From the Elastic Properties window, you can also filter Event markers for Elastic Audio regions on tick-based tracks. Filtering Event markers can result in better quality Elastic Audio processing of material with ambiguous transients.

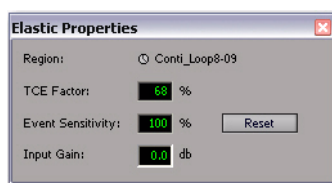
To view the Elastic Properties for a region:

- 1 Select a region on an Elastic Audio-enabled track. The track can be either sample- or tick-based.
- 2 Do one of the following:
 - Choose Region > Elastic Properties.
 - or –
 - Right-click the region and choose Elastic Properties from the pop-up menu.

If the track is sample-based, a reduced set of Elastic Audio Properties are displayed.



Elastic Properties, tick-based



Elastic Properties, sample-based

Elastic Properties Window

The Elastic Properties window displays information about Elastic Audio processing and analysis for one or more regions. You can also use the Elastic Properties window to adjust Elastic Audio processing and analysis data.

Region

The Region property displays the name of the region and indicates whether or not the region is on a tick or sample-based track. If more than one region is selected, the number of selected regions is displayed. If regions on both tick and sample-based tracks are selected, only the reduced set of properties are displayed.

Source Length (Tick-Based Tracks Only)

The Source Length property displays the length of the region based on the source file in bars and beats. If the length of the source file was analyzed incorrectly, you can change it.

To change the Source Length of a region:

- 1 Open the Elastic Properties window for the region.
- 2 Do one of the following:
 - Type the correct bar, beat, and tick values in the Source Length field.
 - If the duration was incorrectly analyzed as being half the duration of the source file, click the x2 button.
 - If the duration was incorrectly analyzed as being twice the duration of the source file, click the 1/2 button.



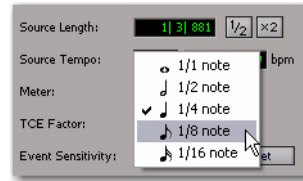
Changing the duration of a region on an Elastic Audio-enabled track using the TCE Trimmer tool updates the Source Length in the Elastic Properties window.

Source Tempo (Tick-Based Tracks Only)

The Source Tempo property displays the average tempo of the source file, which is calculated based on the detected transient events and the length of the file. If the tempo of the source file was analyzed incorrectly, you can change it.

To change Source Tempo of a region:

- 1 Open the Elastic Properties window for the region.
- 2 Select the correct note value for the beat from the Tempo Resolution pop-up menu. For example, if the source file is in 5/8, select 1/8 note.



Elastic Properties, selecting the Tempo Resolution

- 3 Do one of the following:
 - Type the tempo in the Source Tempo field.
 - If the tempo was incorrectly analyzed as being half the tempo of the source file, click the x2 button.
 - If the tempo was incorrectly analyzed as being twice the tempo of the source file, click the 1/2 button.

Meter (Tick-Based Only)

Elastic Audio analysis does not calculate meter and assumes all files are in 4/4. If you work with meters other than 4/4, use the Meter property to correctly identify the source meter.

To change Meter of a region:

- 1 Open the Elastic Properties window for the region.
- 2 Click in the Meter nominator field and type the correct number of beats per measure. For example, if the region is in 3/4, type 3.
- 3 Click the Meter denominator field and type the correct note value. For example, if the meter is in 6/8, type 8.

TCE Factor

The TCE Factor property displays the percentage of change applied to the source file. On tick-based tracks, this property is display only and cannot be edited. On sample-based tracks, you can change the TCE Factor to apply TCE to the region. The percentage of the TCE factor is always calculated in relation to the source file.

Event Sensitivity

The Event Sensitivity property lets you filter Event markers based on the analysis confidence level. The confidence level for any detected transient event is based, in part, on the clarity of the transient. For example, if the file is a drum loop, loud accented hits will be analyzed with a higher degree of confidence than a soft, unaccented hit.

The Event Sensitivity acts like a threshold for showing only the transient events that were detected with a high degree of confidence. In Warp or Analysis track views you will see the number of Event markers decrease or increase as you lower or raise the Event Sensitivity.

Lowering the event sensitivity can help reduce the number of erroneously detected transients. In turn, this can result in better sounding Elastic Audio processing. Pro Tools preserves the detected transients when applying TCE in order to avoid flanging and granulation of the transients. Consequently, false transients are also preserved and the resulting sound quality can be less than desirable. If you are working with audio material that does not have clearly defined

transients, you may want to lower the Event Sensitivity in the Elastic Properties, or you may want to even edit the Event markers in Analysis view.



For REX files, Pro Tools converts the slices as Events with 100% confidence. Consequently, setting the Event Sensitivity from 99% to 1% does not filter out any events based on slices. However, if you set the Event Sensitivity to 0%, all markers are filtered out regardless.

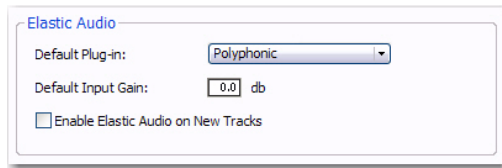
Input Gain

It is possible to encounter clipping when time compressing audio. Real-Time Elastic Audio plug-ins provide a Clip indicator to let you know when clipping has occurred. Rendered Elastic Audio regions display a Clip indicator on the region when clipping has occurred (see “Region Clip Indicator” on page 121).

If you encounter clipping when processing Elastic Audio, use the Input Gain property to attenuate the gain of the audio signal before it is processed.

Elastic Audio Preferences

Pro Tools provides several preferences for Elastic Audio that determine which Elastic Audio plug-in is used for preview and import, and whether or not new tracks are created with Elastic Audio enabled using the selected default plug-in.



Processing Preferences, Elastic Audio

Default Plug-in Lets you select any Real-Time Elastic Audio plug-in as the default plug-in for previewing and importing Elastic Audio. The selected Default Elastic Audio plug-in is also used when new tracks are created with Elastic Audio enabled. The Elastic Audio Plug-in selector in DigiBase browsers inherits the selected Default Plug-in. Likewise, changing the selected plug-in in DigiBase browsers updates the selected Default Plug-in in the Processing preferences.

Default Input Gain Lets you attenuate the signal input to Elastic Audio plug-ins by 0 to –6 dB for preview and import. If you experience clipping due to Elastic Audio processing during preview or after import, you may want to set the Default Input Gain to slightly attenuate the audio signal input for Elastic Audio processing. This preference also applies to any audio imported to an Elastic Audio-enabled track.

If you commit any regions on a track (by disabling Elastic Audio on the track) that were imported with Input Gain, their region-based Input Gain is reset to 0.

The Elastic Properties window inherits the Default Input Gain preference. To apply further region-based Input Gain for Elastic Audio processing, select the region and adjust the Input Gain setting in the Elastic Properties window (see “Elastic Properties Window” on page 122).

Enable Elastic Audio on New Tracks When selected, new tracks are created with Elastic Audio enabled. The selected Default Plug-in is used.



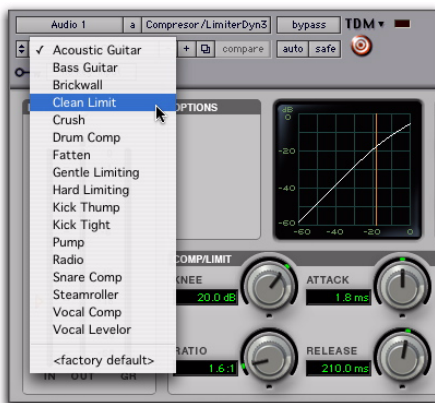
If the Enable Elastic Audio on New Tracks option is selected, you may want to also select the New Tracks Default to Tick Time-base option in the Editing Preferences page.

chapter 9

Plug-in Settings

Using the Librarian

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



Librarian menu

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



You can also use the Plug-in Settings dialog to select settings or audition them. See “Plug-in Settings Dialog” on page 130.

The Settings Menu

Use the Settings menu commands to save, copy, paste, and manage plug-in settings files.



Settings menu

Settings menu commands include:

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

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

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

Paste Settings Pastes settings copied with the Copy Settings command.

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

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

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

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

Settings Preferences

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

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


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

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

The default location for the Root Plug-in Settings folder is as follows:

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

Mac Library/Application Support/
Digidesign/Plug-in Settings

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

Choosing a Destination for Settings

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

To select a destination folder:

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


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

To select a different Root Settings folder:

- 1 From the Settings pop-up menu, choose Settings Preferences > Set Root Settings Folder.
- 2 Select the folder you want to use as your Root folder and click Select.


Managing Settings

Use the Settings pop-up menu to manage settings.

 *Unlinked multi-mono plug-ins have specific rules for settings. See “Editing Settings on Unlinked Multi-Mono Plug-ins” on page 130.*

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 (Mac) 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 (Mac) to copy plug-in settings.*

To paste a setting:

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

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

To create a custom User Default setting:

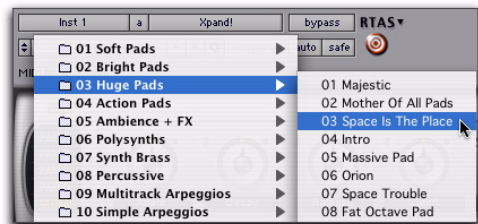
- 1 Create and save a setting.
- 2 Choose Set As User Default from the Settings pop-up menu.

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

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

Creating Subfolders for Settings

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



Librarian menu with subfolders for settings

To create a settings subfolder:

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

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

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

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

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

Editing Settings on Unlinked Multi-Mono Plug-ins

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

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

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

Plug-in Settings Dialog

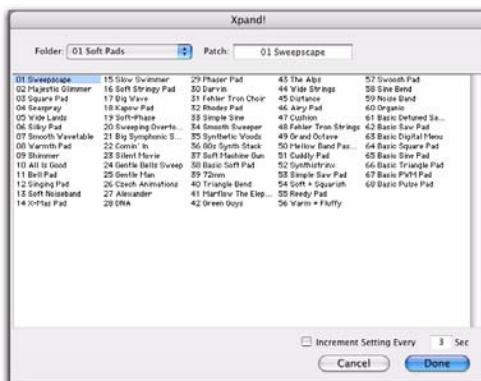
The Plug-in Settings dialog lists the settings files for the current plug-in type. From this list, you can select a new setting, or audition a series of settings. Auditioning plug-in settings is especially useful for selecting patch names for virtual MIDI instrument plug-ins or trying out different effects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

The Plug-in Settings dialog includes the following:

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

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

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



Plug-in Settings dialog

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

- 1 Click the Settings Select button.



Plug-in header settings



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

- 2 Do one of the following:
 - From the dialog list, select the desired plug-in setting.
 - or –
 - Use the Arrow keys to scroll through the available settings.
- 3 Click Done.

Auditioning Plug-in Settings

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

To audition plug-in settings:

- 1 Click the Settings Select button.
- 2 From the Settings dialog, click a setting—the starting point from which you will scroll through the settings files.
- 3 Enter a value for the number of seconds that will elapse between each program change, and press Enter.
- 4 Select the option for Increment Patch.

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

- 5 When you are finished auditioning settings, do one of the following:
 - Click Done to stop incrementing patches, and to close the Plug-in Settings dialog.
 - or –
 - Select the Increment Patch option to stop incrementing patches and keep the Plug-in Settings dialog open.

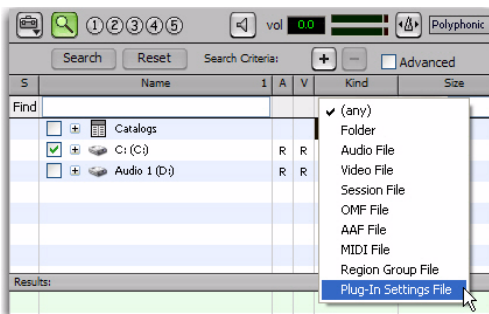
DigiBase and Plug-in Settings Files

DigiBase browsers recognize plug-ins settings files (.txf) as a known Type and lets you drag and drop them to Pro Tools plug-in windows and Inserts. You can also search for plug-in settings by type.

Searching for Plug-in Settings Files

To search for plug-in settings files:

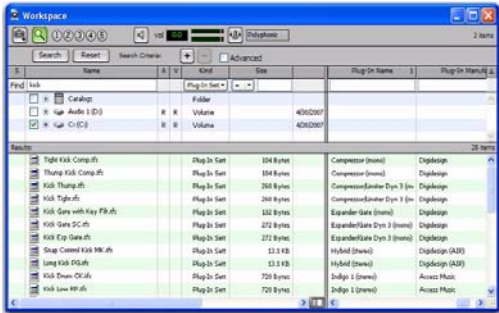
- 1 In the Workspace browser, click the Search icon.
- 2 Select Plug-in Settings File from the Type pop-up menu.



Searching for plug-in settings files

- 3 To further narrow your search, type all or part of the name of the plug-in settings file you want.
- 4 Click Search.

5 To sort your search by Plug-in Name or Manufacturer, click the corresponding column title.



Plug-in settings files found in the Workspace browser and sorted by Plug-in Name

💡 With DigiBase Pro (Pro Tools HD and LE with DV Toolkit 2 only), you can create catalogs of specific plug-in settings that you work with frequently to streamline your workflow. For more information on DigiBase Pro and Catalogs, see the Pro Tools Reference Guide.

Dragging and Dropping Plug-in Settings Files

You can insert a plug-in with a specific plug-in settings file, as well as change plug-in settings of inserted plug-ins by dragging and dropping plug-in settings files from a DigiBase browser, Windows Explorer, or the Mac Finder.

To insert a plug-in by drag and drop:

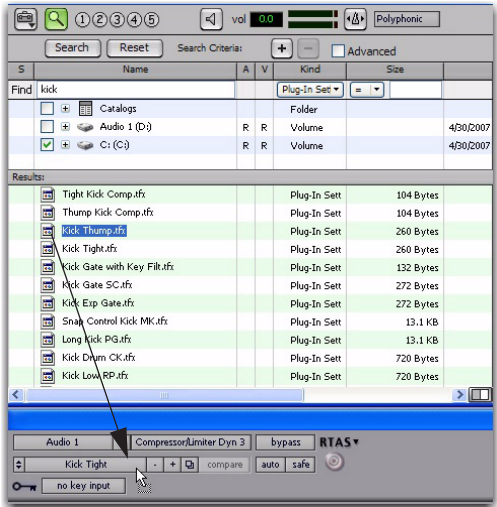
- Drag and drop a plug-in settings file from a DigiBase browser, Windows Explorer, or the Mac Finder to an Insert in the Mix (or Edit) window.



Dragging and dropping a plug-in settings file to an Insert

To change plug-in settings by drag and drop:

- Drag and drop a plug-in settings file from a DigiBase browser, Windows Explorer, or the Mac Finder to the plug-in window. The plug-in settings file must be for the same plug-in name. For example, you cannot replace a 1-Band EQ 3 with a plug-in setting for 4-Band EQ 3.



Dragging and dropping a plug-in settings file to a plug-in

chapter 10

Using ReWire with Pro Tools

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

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



ReWire RTAS plug-in

Using ReWire, Pro Tools can send and receive MIDI to and from a ReWire client application, such as a software synthesizer, and receive audio back from the ReWire client. Pro Tools applies MIDI time stamping to all incoming MIDI. For a diagram of audio and MIDI signal flow with ReWire, see Figure 1 on page 136.

⚠ Pro Tools does not support sending audio to 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 automatically launches that application (if the client application supports this feature). Any corresponding MIDI nodes for that application are available in any Instrument track's MIDI Output selector (Instrument view) and any MIDI track's Output selector.

⚠ Not all ReWire client applications support automatic launch from a ReWire-mixer application. For these applications, launch the ReWire client application separately, and then select it as a plug-in insert in Pro Tools.

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.

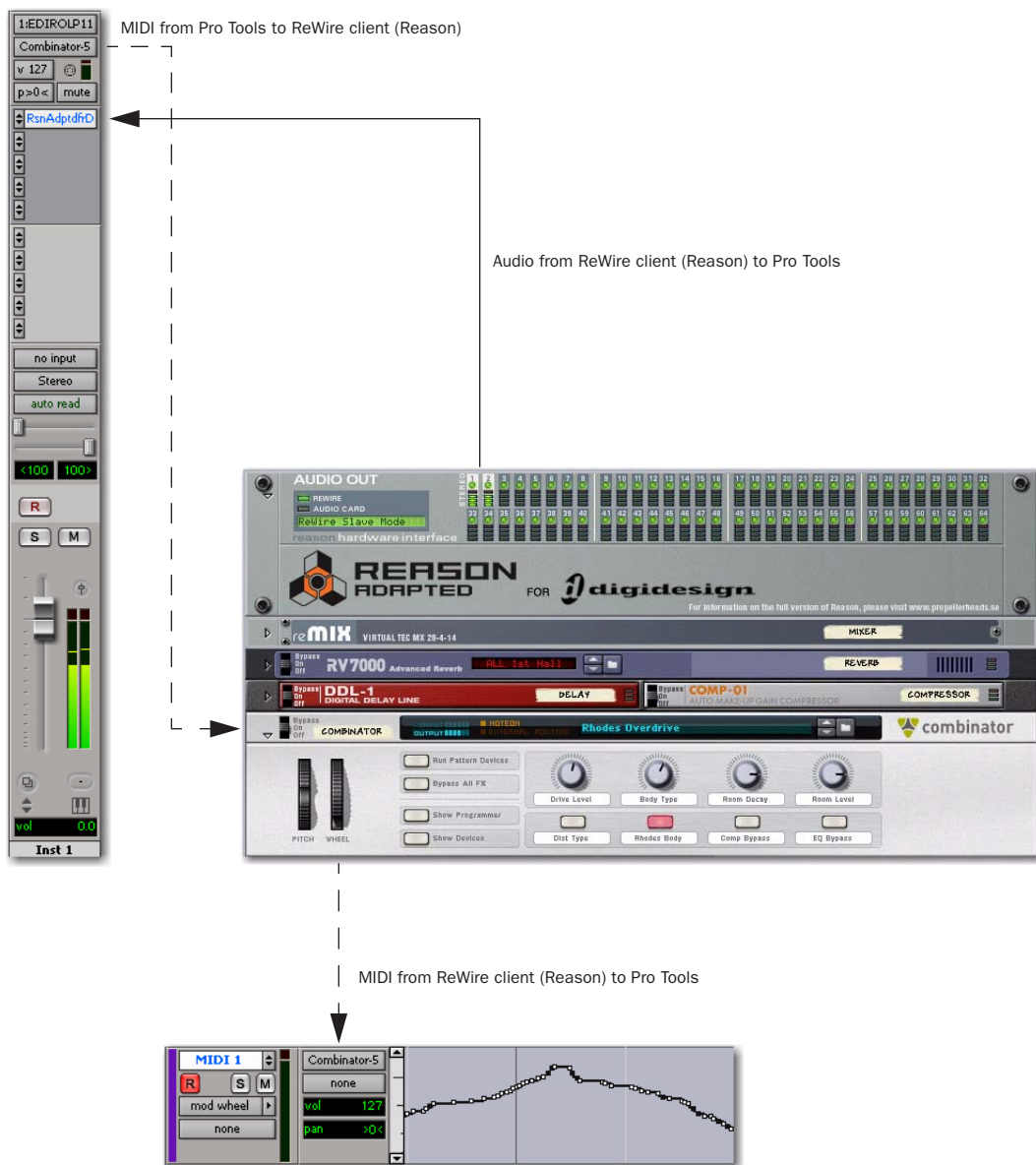


Figure 1. Audio and MIDI signal flow between Pro Tools and a ReWire client application (Reason shown)

ReWire Requirements

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

- A Digidesign-qualified Pro Tools system
- ReWire-compatible client software (such as *Reason* from Propellerheads Software)

⚠ *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 visit the Digidesign website (www.digidesign.com).

ReWire and Track Count with Pro Tools HD

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

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

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

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

ReWire and Track Count on Pro Tools LE Systems

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

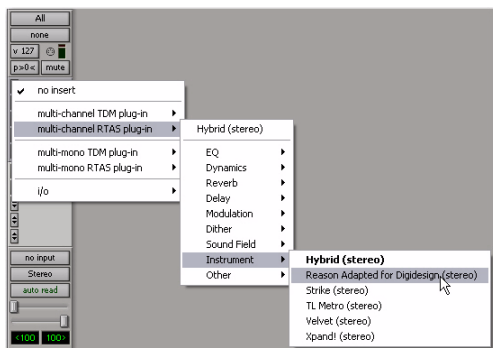
For the latest information on recommended CPUs and system configurations, visit the Digidesign website (www.digidesign.com).

Using ReWire

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


To use a ReWire client application with Pro Tools:

- 1 Make sure that the ReWire client application is installed properly and that you have restarted your computer.
- 2 In Pro Tools, choose Track > New and specify one Instrument track (or audio or Auxiliary Input track), and click Create.
- 3 In the Mix window, click the Insert selector on the track and assign the ReWire RTAS client plug-in to the track insert.



Inserting a ReWire client on a track (Reason shown)

The ReWire client application launches automatically in the background (if the client applications supports auto-launch).

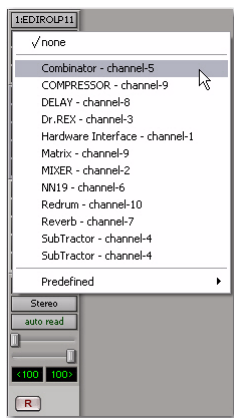
 *If the client application does not support auto-launch, launch it manually. Some ReWire client applications may need to be launched and configured before launching Pro Tools (such as Cycling 74's MAX/MSP). Others may need to be launched after Pro Tools is launched (such as Ableton Live). For more information, consult the manufacturer's documentation for your ReWire client application.*

- 4 Configure the ReWire client application to play the sounds you want.
- 5 In Pro Tools, set the output of the client application in the ReWire plug-in window. This is the audio output of the ReWire client to Pro Tools.



Selecting the audio output from a ReWire client

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



Selecting the ReWire client device to receive MIDI from Pro Tools

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

The selected ReWire device responds to MIDI sent from Pro Tools and plays back audio through the assigned Pro Tools track (Instrument, Auxiliary Input, or audio track).

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 11 for instructions.*

MIDI Automation with ReWire

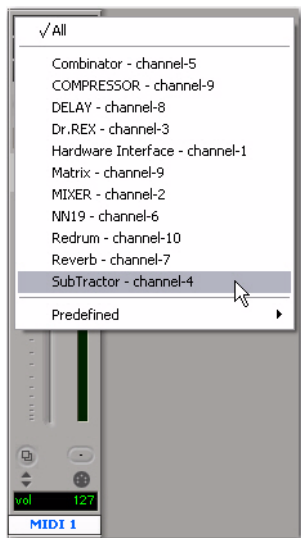
You can use Pro Tools MIDI tracks to record MIDI continuous controller (CC) data from a ReWire client application, and then play back MIDI from Pro Tools to send the recorded MIDI CC data back to the ReWire client application. In this way, you can adjust parameters in the ReWire client application (using the mouse or an external MIDI controller) and record those changes in Pro Tools.

Recording MIDI Continuous Controller Data Over ReWire

The first step in automating a ReWire client application's parameters is to record the CC data to a MIDI track in Pro Tools.

To record MIDI from a ReWire client application in Pro Tools:

- 1 In Pro Tools, create a new MIDI track.
- 2 From the track's MIDI Input selector, select the ReWire device that you want to record.



Selecting the ReWire client device to record MIDI CC data in Pro Tools

⚠ You must select the ReWire device from which you want to record MIDI controller data. Leaving the track's MIDI Input set to All does not record any MIDI data over ReWire.

- 3 Record enable the MIDI track.
- 4 Start recording in Pro Tools.
- 5 Switch to the ReWire client application.

6 Adjust the parameter for which you want to record MIDI CC data. Parameter changes are recorded to the Pro Tools MIDI track as CC data.



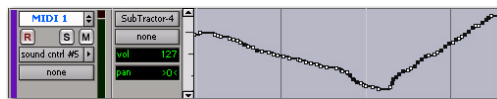
Adjusting a parameter in a ReWire client application (Reason's SubTractor shown)

💡 If your external MIDI controller is correctly mapped to the corresponding ReWire client application's parameters, and it is correctly routed through Pro Tools, use your MIDI controller to adjust the parameter you want to record.

7 When you are done adjusting the parameter, return to Pro Tools and stop recording.

8 Record disable the MIDI track.

9 From the MIDI Track View selector in the Edit window, select the view for the CC data you just recorded.



MIDI CC data recorded from a ReWire client application

Playing Back MIDI Continuous Controller Data Over ReWire


Once you have recorded MIDI CC data from the ReWire client application to a MIDI track, configure the MIDI track to play the ReWire client application. You can also edit the MIDI CC data in Pro Tools until you achieve the best results.

To play back MIDI CC data over ReWire:

- 1 From the MIDI track's MIDI Output selector, select the ReWire client application device you want to control (the same device from which you recorded the MIDI CC data).
- 2 Start playback in Pro Tools.
- 3 Switch to the ReWire client application. Notice that the corresponding parameter changes according the MIDI CC data from Pro Tools.

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 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 Timeline, 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 Timeline, once playback is started, any changes made to loop or playback markers within the ReWire client application will deselect the Pro Tools Timeline selection and remove the loop.*

Automating ReWire Input Switching

ReWire supports automation for switching inputs during playback.

To automate switching inputs during playback:

- 1 Set the track's automation to write.
- 2 Do one of the following:
 - Change the input link pop-up menu manually.
 - or –
 - Draw the automation in the Edit window.



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

appendix a

DSP Requirements for TDM Plug-ins

The number of TDM plug-ins you can use at one time depends on how much DSP power is available in your system. Since the TDM hardware on Pro Tools cards provide dedicated DSP for plug-ins, plug-in performance is not limited by CPU processing power.

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

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

There are a total of nine DSP chips on a Pro Tools|HD card (HD Core, HD Process, and HD Accel). HD Core and HD Process cards provide identical chip sets. HD Accel cards provide newer, more powerful DSP chips (making the HD Accel card ideal for DSP-intensive plug-ins, and for high sample rate sessions).

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

DSP Requirements

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

Mono and Stereo DigiRack Plug-ins on HD Cards

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

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

Plug-in	HD Accel		HD Core and HD Process	
	mono/stereo		mono/stereo	
EQ III (1-band)	66	33	28	14
EQ III (2–4 band)	36	18	15	7
EQ III (7-band)	23	11	10	5
Click	18	n/a	3	n/a
Dither	46	43	34	18
D-Verb	6	6	4	4
Compressor/Limiter III	18	11	7	5
Expander/Gate III	16	11	7	5
De-Esser III	12	8	5	3
Mod Delay II (short)	74	36	43	21
Mod Delay II (slap)	73	36	32	16
Mod Delay II (medium)	36	18	16	8
Mod Delay II (long)	17	7	8	4
Mod Delay II (extra long)	4	2	2	1
Pitch	1	1	1	1
POW-r Dither	22	10	9	4
Signal Generator	48	n/a	25	n/a
PhaseScope (SignalTools)	n/a	13	n/a	5
Time Adjuster (short)	146	72	117	58
Time Adjuster (medium)	32	16	114	56
Time Adjuster (long)	72	36	32	16
Trim	145	n/a	119	n/a

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

Multi-Mono and Multi-Channel DigiRack Plug-ins

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

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

Plug-in	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP Chips per HD Card
DigiRack Pitch	partial	partial	partial	partial	partial	partial	9
D-Verb	partial	partial	partial	partial	partial	partial	9
Compressor/Limiter III	3	3	2	2	2	2	9
Expander/Gate III	3	3	2	2	2	2	9
De-Esser III							
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

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

Plug-in	LCR	Quad & LCRS	5.0	5.1 & 6.0	6.1 & 7.0	7.1	DSP Chips per HD Card
Time Adjuster (medium)	30	22	18	15	13	11	9
Time Adjuster (long)	10	8	6	5	4	4	9
Trim*	39	29	23	19	16	14	9
Signal Generator*	8	6	5	4	3	3	9
SurroundScope (SignalTools)	4	3	2	2	1	1	9

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

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

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

Monitoring DSP Usage

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



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

appendix b

DSP Delays Incurred by TDM Plug-ins

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

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

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



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

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

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

Plug-in	Samples of Delay on HD Cards
EQ III (1-band)	4
EQ III (2–4 band)	4
EQ III (7-band)	4
Compressor II	4
De-esser	4
DigiRack Pitch	3
D-Verb	3
Compressor/Limiter III	4
Expander/Gate III	92
De-Esser III	4
Limiter II	4
Gate II	4
Expander/Gate II	4
Mod Delay I/II (short)	4
Mod Delay I/II (slap)	4

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

Plug-in	Samples of Delay on HD Cards
Mod Delay I/II (medium)	4
Mod Delay I/II (long)	4
Mod Delay II (extra long)	4
Dither	4
POW-r Dither	4
Time Adjuster	4
Signal Generator	4
SignalTools SurroundScope	4
SignalTools PhaseScope	4

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