Digidesign® Plug-ins

Version 7.3

contents

chapter 1

Introduction

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

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

These plug-ins include:

- Bruno® & Reso® cross-synthesis plug-ins
- D-Fi™ creative sound design plug-ins
- DINR™ intelligent noise reduction
- Impact™
- Maxim™ peak limiter/sound maximizer
- Reverb One™
- ReVibe™
- Smack!™
- SoundReplacer™ drum and sound replacement plug-in
- X-Form high-quality time compression and expansion plug-in

⚠ *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.*

Contents of the Boxed Version of Your Plug-in

If you purchased the boxed version of a plug-in, it contains the following:

- Installation disc
- Digidesign Registration Information Card
- One of the following authorization cards for authorizing plug-ins with an iLok USB Smart Key (not supplied):
  - Activation Card with an Activation Code

  – or –

  - License Card

System Requirements

To use Digidesign plug-ins you need the following:

- An iLok USB Smart Key
- An iLok.com account for managing iLok licenses
- One of the following:
  - A Digidesign-qualified Pro Tools|HD® system or Pro Tools LE system.
  - 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 Digidesign-qualified 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).

Register Your Plug-ins

If you purchase a plug-in online from the DigiStore (www.digidesign.com) using Internet Activation, you are automatically registered.

If you purchase a plug-in boxed version, check the included Digidesign Registration Information Card for information about how to register online.

Registered users receive periodic software update and upgrade notices.

Please refer to the Digidesign website (www.digidesign.com) or the included Digidesign Registration Information Card for information on technical support.

Working with Plug-ins

Besides the information provided in this guide, refer to the *DigiRack Plug-ins Guide* for general information on working with plug-ins, including:

- Inserting Plug-ins on Tracks
- Clip Indicators
- The Plug-in Window
- Adjusting Parameters
- Automating Plug-ins
- Using the Librarian

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 enclosed registration form for details.

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

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

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

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

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

chapter 2

Installation

Installing Plug-ins

Installers for your plug-ins can be downloaded from the DigiStore (www.digidesign.com) or can be found on the plug-in installer disc (included with boxed versions of plug-ins). An installer may also be available on a Pro Tools software bundle installer disc.

Installation steps are essentially the same, regardless of the bundle, package, or system you purchase.

Updating Older Plug-ins

Because the Digidesign Plug-in installers contain the latest versions of the Digidesign plug-ins, use them to update any plug-ins you already own.

⚠ *Be sure to use the most recent versions of Digidesign plug-ins available from the Digidesign website (www.digidesign.com).*

Installation

To install a plug-in:

1 Do one of the following:

- Download the installer for your computer platform from the Digidesign website (www.digidesign.com). After downloading, make sure the installer is uncompressed (.ZIP on Windows or .SIT on Mac).

– or –

- Insert the Installer disc into your computer.

2 Double-click the plug-in installer application.

3 Follow the on-screen instructions to complete the installation.

4 When installation is complete, click Finish (Windows) or Quit (Mac).

When you open Pro Tools, you are prompted to authorize your new plug-in.

Authorizing Plug-ins

Digidesign plug-ins are authorized using the iLok USB Smart Key (iLok), manufactured by PACE Anti-Piracy, Inc.

iLok USB Smart Key

The iLok is similar to a dongle, but unlike a dongle, it is designed to securely authorize multiple software applications from a variety of software developers.

This key can hold over 100 licenses for all of your iLok-enabled software. Once an iLok is authorized for a given piece of software, you can use the iLok to authorize that software on any computer.

⚠ *The iLok USB Smart Key is not supplied with your plug-in or software option. You can use the one included with certain Pro Tools systems (such as Pro Tools|HD-series systems), or purchase one separately.*

If you purchased a plug-in online at the DigiStore (www.digidesign.com), authorize the plug-in by downloading licenses from iLok.com to an iLok.

▤ *See the iLok Usage Guide for details, or visit the iLok website (www.iLok.com).*

If you purchased a boxed version of a plug-in, it comes with either an Activation Code (on the included Activation Card) or an iLok License card:

• To authorize plug-ins using an Activation Code, see "Authorizing Plug-ins Using an Activation Code" on page 6.

• To authorize plug-ins using an iLok License Card, see "Authorizing Plug-ins Using a License Card" on page 7

Authorizing Plug-ins Using an Activation Code

To authorize a plug-in using an Activation Code:

1 If you do not have an existing iLok.com account, visit www.iLok.com and sign up for an iLok.com account.

2 Transfer the license for your plug-in to your iLok.com account by doing the following:

• Visit http://secure.digidesign.com/activation.

• Input your Activation Code (listed on your Activation Card) and your iLok.com User ID. Your iLok.com User ID is the name you create for your iLok.com account.

3 Transfer the licenses from your iLok.com account to your iLok USB Smart Key by doing the following:

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

• Go to www.iLok.com and log in.

• Follow the on-screen instructions for transferring your licences to your iLok.

▤ *For information about iLok technology and licenses, see the electronic PDF of the iLok Usage Guide.*

4 Launch Pro Tools.

5 If you have any installed unauthorized plug-ins or software options, you are prompted to authorize them. Follow the on-screen instructions to complete the authorization process.

Authorizing Plug-ins Using a License Card

License Cards are specific to each plug-in or software option. You will receive the appropriate License Cards for the plug-ins that you purchase. License Cards have a small punch-out plastic chip called a GSM cutout.

The authorization steps in this section must be repeated for purchased plug-in.

📖 *For additional information about iLok technology and authorizations, see the electronic PDF of the iLok Usage Guide.*

To authorize a plug-in using a License Card:

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

2 Launch Pro Tools. You are prompted to authorize any installed unauthorized plug-ins or software options.

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

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

4 Separate the GSM cutout from the larger protective card by pulling it up and out with your thumb. Do not force the cutout down with your finger.

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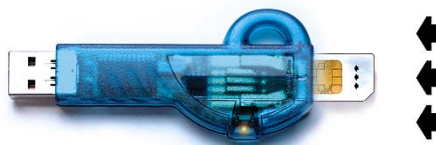
iLok with License Card

6 Follow the on-screen instructions to complete the authorization process for each plug-in.

7 After the authorization has completed, remove the GSM cutout from the iLok. (If you have to remove the iLok from the computer to remove the cutout, be sure to re-insert the iLok in any available USB port on your computer when you are finished.)

Removing Plug-ins

If you need to remove a plug-in from your system, follow the instructions below for your computer platform.

Windows

To remove a plug-in:

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

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

3 Follow the on-screen instructions to remove the plug-in.

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

Mac OS X

To remove a plug-in:

1 Locate and open the Plug-ins folder on your Startup drive (Library/Application Support /Digidesign/Plug-ins).

2 Do one of the following:

- Drag the plug-in to the Trash and empty the Trash.

– or –

- Drag the plug-in to the Plug-ins (Unused) folder.

chapter 3

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 "Adjusting Controls Using a Mouse" on page 9 and "Editing Parameters Using a Computer Keyboard" on page 10.

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

Adjusting Controls 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.

- 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 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 on-screen.

chapter 4

Bruno and Reso

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

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

Bruno and Reso each use a different sound generation method:

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

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

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

Maximum Voices with HD Accel Card

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

Bruno features include:

- Time-slice tone generation with adjustable crossfade
- Polyphony: Up to 62 voices of polyphony (on Pro Tools|HD Accel systems)
- Multi-voice detuning
- Editable ADSR envelope generator
- Portamento
- Velocity-sensitive gain and detuning
- Time-slice switching using envelope triggering or MIDI beat clock
- Voice-stacking
- Side-chain input for control using an external audio source
- Supports sample rates up to 192 kHz
- Online help

Reso features include:

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

DSP Requirements

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

DSP and Voice Polyphony

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

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

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

Inserting Bruno/Reso onto an Audio Track

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

To add Bruno/Reso as a track Insert:

1 Click the Insert selector on the desired track and select Bruno or Reso.

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

3 Play Bruno/Reso with the on-screen keyboard or by MIDI control. See "Playing Bruno/Reso" on page 13.

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

Playing Bruno/Reso

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

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

– or –

◆ Using MIDI

Using the On-Screen Keyboard

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

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

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

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

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

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

To latch keys on the on-screen keyboard:

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

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

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

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

Using MIDI

You can play Bruno/Reso live using a MIDI keyboard controller. You can also use the MIDI keyboard controller to record your performance on an Instrument track or a MIDI track routed to Bruno/Reso for playback.

To configure Bruno/Reso for MIDI input:

1 Insert Bruno/Reso on an audio track.

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

3 Click the track's MIDI Output selector and choose Bruno or Reso.

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

4 Record-enable the Instrument or MIDI track.

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

To play Bruno/Reso with a MIDI controller:

1 Start audio playback.

2 Play your MIDI keyboard while audio plays.

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

Using MIDI Playback

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

To play Bruno/Reso using an Instrument or MIDI track:

1 Insert Bruno or Reso on an audio track.

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

3 Start Pro Tools playback.

Using an External Key Input for Side-Chain Processing

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

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

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

To use a key input for side-chain processing:

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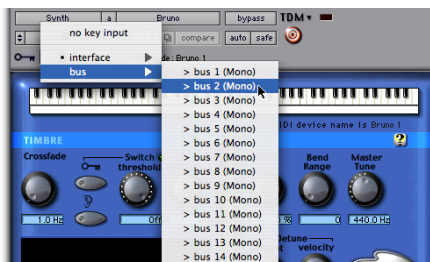
Selecting a Key Input

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

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

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

💡 *Remember to disable Key Listen to resume normal plug-in monitoring.*

5 Adjust other controls to create the desired effect.

Bruno Controls

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

Bruno

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

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

On-Screen Keyboard

The on-screen keyboard

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

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

Timbre Controls

Timbre controls

Crossfade

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

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

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

Switch

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

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

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

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

Key Listen When enabled, Key Listen monitors the source of the key input. It is often useful to do this in order to fine tune Bruno's settings to the key input. See "Using an External Key Input for Side-Chain Processing" on page 14.

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

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

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

Timbrometer
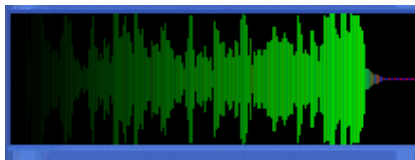
Timbrometer

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

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

Amplitude Controls

Amplitude controls

Gain Amount

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

Gain Velocity

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

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

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

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

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

Mix

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

Spread

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

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

ADSR Envelope Generator

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

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

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

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

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

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

Pitch Controls

Pitch controls

Glide

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

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

Bend Range

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

Master Tune

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

Detune Amount

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

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

Detune Velocity

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

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

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

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

Voice Controls

Voice controls

These controls set Bruno's voice polyphony and allocation.

Mode

Mono (Monophonic)

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

Poly (Polyphonic)

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

Voice Stack

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

Voice Stack

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

- 32 notes in a 1-voice stack
- 16 notes in a 2-voice stack
- 4 notes in a 4-voice stack
- 2 notes in an 8-voice stack
- 1 note in an 12-voice (All) stack

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

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

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

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

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

Online Help

Online help

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

Reso Controls

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

Reso

On-Screen Keyboard

On-screen keyboard

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

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

Timbre Controls

Timbre controls

Resonance Amount

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

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

Resonance Velocity

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

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

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

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

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

Damping Amount

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

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

Damping Velocity

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

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

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

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

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

Harmonics

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

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

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

Toggle

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

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

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

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

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

📖 *See "Using an External Key Input for Side-Chain Processing" on page 14.*

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

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

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

Amplitude Controls

Amplitude controls

Gain Amount

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

Gain Velocity

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

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

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

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

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

Mix

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

Spread

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

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

ADSR Envelope Generator

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

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

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

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

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

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

Pitch Controls

Pitch controls

Glide

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

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

Bend Range

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

Master Tune

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

Detune Amount

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

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

Detune Velocity

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

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

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

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

LPF/Voice Controls

LPF and Voice controls

LPF (Low-Pass Filter)

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

Frequency

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

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

Q

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

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

The range of this control is from 0 to 10.

Follower

The Follower is an envelope follower that lets the filter cutoff frequency dynamically follow the amplitude of the source audio signal.

The range of this control is from a low of –10 to a high of +10. With positive values, the louder the source audio, the higher the cutoff frequency and the wider the filter will open for a brighter sound. With negative values, the louder the source audio, the lower the cutoff frequency and the more the filter will close for a duller sound.

The effectiveness of the Follower depends on the filter's Frequency setting. For example, setting the Follower to +10 and selecting a low Frequency setting will sweep the filter wide on loud passages. However, if the cutoff frequency is at its maximum, setting the Follower to +10 will not sweep the filter at all since it is already completely open.

When used with high Q settings and a relatively low cutoff frequency, the Follower can be used to produce an automatic wah-wah-type effect.

Mono (Monophonic)

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

Poly (Polyphonic)

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

⚠ *Polyphony will be reduced by half at 96 kHz.*

Voice Stack

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

Voice Stack

In a 96 kHz session, Reso on Pro Tools|HD Accel systems can simultaneously play up to:

- 32 notes in a 1-voice stack
- 16 notes in a 2-voice stack
- 4 notes in a 4-voice stack
- 2 notes in an 8-voice stack
- 1 note in an 14-voice (All) stack

In a 44.1 kHz or 48 kHz session on Pro Tools|HD systems not equipped with an HD Accel card, the standard Reso module can simultaneously play up to:

- 28 notes in a 1-voice stack
- 14 notes in a 2-voice stack
- 7 notes in a 4-voice stack
- 3 notes in an 8-voice stack
- 1 note in a 28-voice (All) stack

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

Online Help

Online help

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

chapter 5

D-Fi

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

Lo-Fi

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

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

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

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

Sci-Fi

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

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

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

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

Recti-Fi

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

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

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

Vari-Fi

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

- Speed up from a complete stop to normal speed
- Slow down to a complete stop from normal speed

Vari-Fi is an AudioSuite plug-in only.

Purposely Degrading Audio

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

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

Lo-Fi

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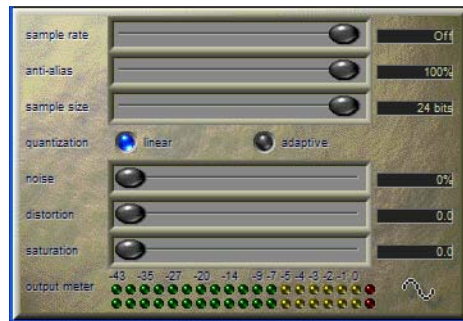
Lo-Fi

Lo-Fi Controls

Sample Rate

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

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

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

Anti-Alias Filter

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

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

Sample Size

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

Quantization

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

Linear Linear quantization abruptly cuts off sample data bits in an effort to fit the audio into the selected bit resolution. This imparts a characteristically raunchy sound to the audio that

becomes more pronounced as the sample size is reduced. At extreme low bit-resolution settings, linear quantization will actually cause abrupt cut-offs in the signal itself, similar to gating. Thus, linear resolution can be used creatively to add random percussive, rhythmic effects to the audio signal when it falls to lower levels, and a grungy quality as the audio reaches mid-levels.

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

Noise Generator

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

Distortion/Saturation

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

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

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

Output Meter

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

Sci-Fi

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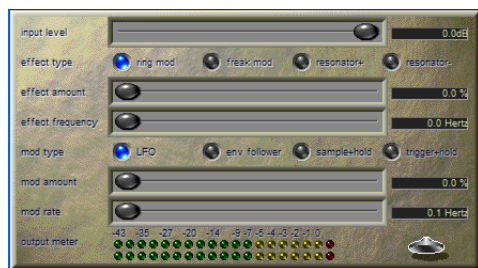
Sci-Fi

Sci-Fi Controls

Input Level

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

Effect Type

Sci-Fi provides four different types of effects:

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

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

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

Effect Amount

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

Effect Frequency

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

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

To use keyboard note entry:

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

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

Sci-Fi Keyboard Note Entry

Modulation Type

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

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

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

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

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

Mod Amount and Mod Rate

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

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

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

Output Meter

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

Recti-Fi

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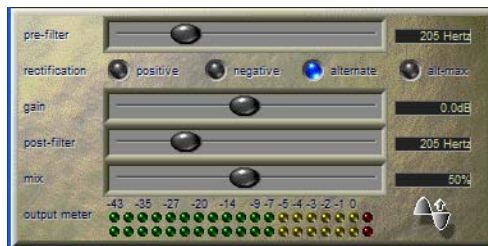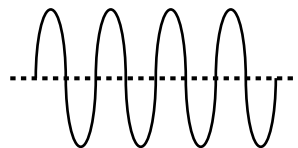

Recti-Fi

Recti-Fi Controls

Pre-Filter

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

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

Normal waveform

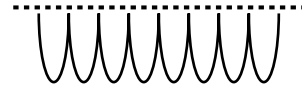## Rectification

Positive Rectification

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

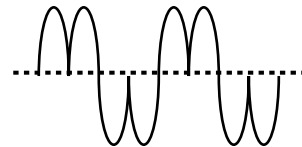*Positive rectification*

Negative Rectification

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

*Negative rectification*

Alternating Rectification

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

Alternating rectification

Alt-Max Rectification

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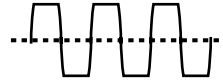
Alt-Max rectification

Gain

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

Post-Filter

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

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

Mix

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

Output Meter

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

Vari-Fi

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

Vari-Fi

Vari-Fi Controls

Speed Up

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

Slow Down

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

D-Fi Demo Session

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

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

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

Sci-Fi Examples

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

Choosing a Sci-Fi setting

Hi-Hat Loop

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

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

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

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

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

6 Deselect the Bypass button to hear the effect.

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

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

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

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

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

Drum Kit Loop

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

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

3 Choose "Ring Mod Trig & Hold Kit."

4 Deselect Bypass to hear the effect.

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

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

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

8 Experiment with the other settings.

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

Wah Guitar Loop

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

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

3 Choose "Freq Mod Env. Follower Wah."

4 Deselect Bypass to hear the effect.

5 Press the Spacebar to audition this loop.

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

Lo-Fi Examples

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

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

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

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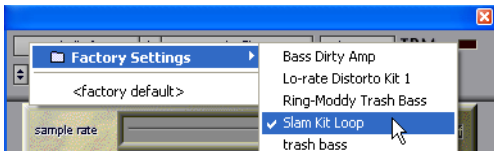
Choosing a Lo-Fi setting

Slam Kit Loop

1 Open the Lo-Fi Plug-in window.

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

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

4 Press the Spacebar to audition the loop.

5 Deselect Bypass to hear the effect.

6 Try each Lo-Fi setting with this loop.

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

Drum Kit Loop

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

2 Choose "Lo-rate Distorto Kit."

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

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

Bass Only

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

2 Choose "Bass Dirty Amp."

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

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

4 Choose "Trash Bass."

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

5 Choose "Ring Moddy Bass."

This setting demonstrates extreme Lo-Fi processing.

Recti-Fi Examples

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

Before you begin:

1 Open the demo session.

2 Choose Windows > Show Memory Locations.

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

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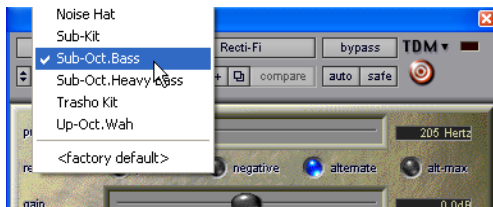
Choosing a Recti-Fi setting

Sub Octave Bass

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

2 Choose "Sub Octave Bass."

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

Sub-Oct. Heavy Bass

■ Choose "Sub-Oct. Heavy Bass."

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

Drum Kit Loop

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

2 Choose "Sub Kit."

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

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

Hat Loop

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

2 Choose "Noise Hat."

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

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

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

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

Wah Guitar

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

2 Choose "Up Octave Wah."

3 Play the audio.

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

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

Slam Kit Loop

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

2 Choose "Trasho Kit."

3 Play the audio.

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

DINR

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

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

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

Broadband Noise Reduction

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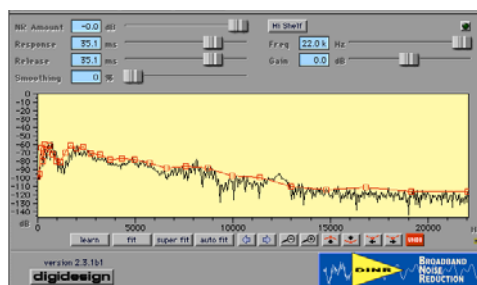
BNR

How Broadband Noise Reduction Works

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

Creating a Noise Signature

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

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

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

The Contour Line

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

Psychoacoustic Effects of Noise Reduction

One of the psychoacoustic effects associated with broadband noise reduction is that listeners often perceive the loss of noise as a loss of high frequencies. This occurs because the noise in the higher frequency ranges fools the ear into thinking the original signal has a great deal of energy in that range. Consequently, when the noise is removed it feels as if there has been a loss of high-frequency signal. DINR's High-Shelf EQ is useful for compensating for this effect. See "High-Shelf EQ" on page 45.

Limitations of Noise Reduction

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

◆ The amount of *noise* removed from the signal

◆ The amount of *signal* removed from the signal

◆ The number of *artifacts* added to the signal

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

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

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

Broadband Noise Reduction Controls

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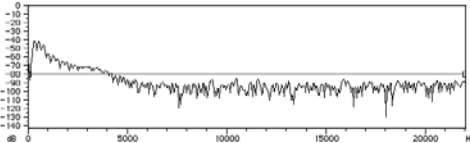
Spectral Graph

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

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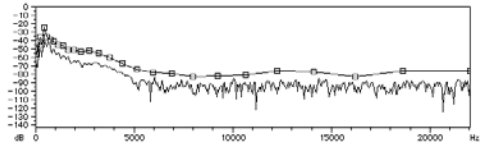
Contour Line

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

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

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

to follow the noise signature as closely as possible will also help maximize noise reduction and minimize signal loss. See "Editing the Contour Line" on page 49.

NR Amount, Response, Release, and Smoothing

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

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

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

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

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

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

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

High-Shelf EQ

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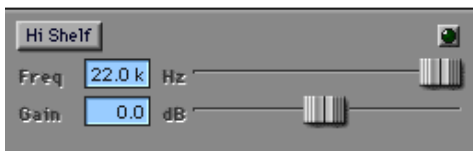
High-Shelf EQ

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

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

Learn

Learn

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

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

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

Fit

Fit

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

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

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

Super Fit

Super Fit

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

Auto Fit

Auto Fit

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

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

Scroll Left/Right

Scroll Left/Right

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

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

Zoom Out/In

Zoom Out/In

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

Move Breakpoints Up/Down/Left/Right

Move Breakpoints Up/Down/Left/Right

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

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

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

Undo

UNDO

Undo

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

Using Broadband Noise Reduction

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

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

To use Broadband Noise Reduction:

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

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

3 Do one of the following:

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

– or –

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

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

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

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

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

8 To further increase noise reduction, edit the Contour Line. The quickest way to do this is to move the entire Contour Line upwards. In the Spectral Graph, Control-drag (Windows) or Command-drag (Mac) to select the entire waveform range. Then click the Move Breakpoint Up button. The higher you move the Contour Line above the noise signature, the more noise is removed. See "Editing the Contour Line" on page 49.

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

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

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

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

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

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

Some editing of the newly generated Contour Line will probably be necessary to yield optimum results, since it is not based entirely on noise from your audio file. See "Editing the Contour Line" on page 49.

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

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

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

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

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

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

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

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

8 Since the Contour Line is not based entirely on noise from your audio file, you may also want to edit its envelope in order to fine-tune the noise reduction. See "Editing the Contour Line" on page 49.
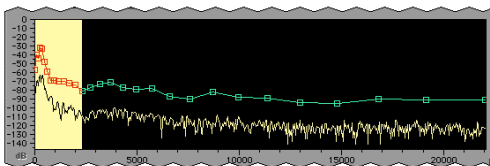
Noise components on the Spectral Graph

Editing the Contour Line

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

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

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

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

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

2 Select Options > Loop Playback.

3 Begin playback.

To edit the Contour Line:

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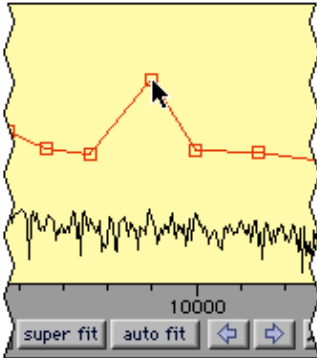
Dragging a breakpoint

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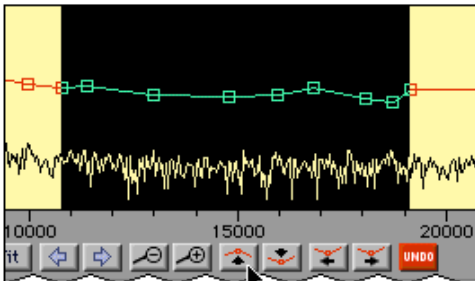
Moving selected breakpoints

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

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

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

Using BNR AudioSuite

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

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

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

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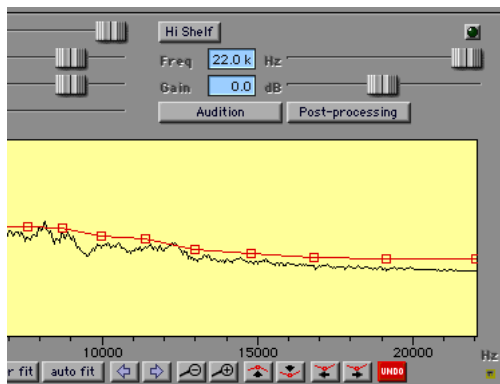
BNR AudioSuite

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

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

2 From the Pro Tools AudioSuite menu, choose BNR.

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

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

5 Click Preview to begin playback of the selected material.

6 Adjust BNR controls and fine-tune the noise reduction using the techniques explained above (See "Using Broadband Noise Reduction" on page 47.)

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

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

To begin AudioSuite processing:

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

◆ Decide where the selected region should be processed:

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

  – or –

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

◆ Decide if you want to update every occurrence of the selection region:

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

  – or –

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

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

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

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

3 Click Process.

chapter 7

Impact

Impact is a high-quality compressor plug-in that provides critical control over the dynamic range of audio signals. Impact is a real-time TDM plug-in with the look and sound of a mixing console's stereo-bus compressor.

Impact provides support for 192 kHz, 176.4 kHz, 96 kHz, 88.2 kHz, 48 kHz, and 44.1 kHz sessions.

Impact provides support for mono, stereo, and all Pro Tools-supported multichannel audio formats.

Using the Impact Compressor

Compressors reduce the dynamic range of audio signals that exceed a user-selectable threshold by a specific amount. This is accomplished by reducing output levels as input levels increase above the threshold.

The amount of output level reduction that Impact applies as input levels increase is referred to as the *compression ratio*. This parameter is adjustable in discrete increments. If you set the compression ratio to 2 (a ratio of 2:1), for each 2 dB that the signal exceeds the threshold, the output will increase only by 1 dB. With a setting of 4 (a ratio of 4:1), an 8 dB increase in input will produce only a 2 dB increase in output.

Impact plug-in

Side-Chain Processing

Compressors generally use the detected amplitude of their input signal as a control source. However, you can also use other signals, such as a separate reference track or an external audio signal as a control source. This is known as *side-chain* processing.

Side-chain processing lets you control Impact compression using an independent audio signal (typically, another Pro Tools track). In this way you can compress the audio of one track using the dynamics of a different audio track.

The reference track or external audio source used for triggering side-chain processing is referred to as the *Key Input*.

📄 *See "Using a Key Input for External Side-Chain Processing" on page 57 for instructions on setting up and using a key input.*

Impact Parameters

Ratio

Ratio sets the compression ratio. If the ratio is set to 2:1 for example, it will compress changes in signals above the threshold by one half. This control provides four fixed compression ratios, 2:1, 4:1, 10:1, and 20:1. Selecting 2:1 applies very light compression; selecting 20:1 applies heavy compression, bordering on limiting.

Ratio

Attack

Attack sets the compressor attack time. 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. The range of this control is from 0.1 ms to 30.0 ms.

Attack

Threshold

Threshold sets the decibel level that a signal must exceed for Impact to begin applying compression. Signals that exceed the Threshold will be compressed by the amount of gain reduction set with the Ratio control. Signals that are below the Threshold will be unaffected. The range of the Threshold control is from –70 dB to –0 dB. A setting of –0 dB is equivalent to no compression.

Threshold

Release

Release controls the length of time it takes for the compressor to be fully deactivated after the input signal drops below the threshold level. In general, this setting should be longer than the attack time and long enough that if signal levels repeatedly rise above the threshold, they cause

gain reduction only once. If the release time is too long, a loud segment of audio material could cause gain reduction to persist through a low-volume segment (if one follows). Setting this control to its maximum value, Auto, selects a release time that is program dependent, based on the audio being processed. The range of this control is from 20 milliseconds to 2.5 seconds.

Release

Make-Up

Make-Up adjusts the overall output gain. Because large amounts of compression can restrict dynamic range, the Make-Up control is useful for compensating for heavily compressed signals and making up the resulting difference in level. When Impact is used on stereo or multi-channel tracks, the Make-Up control determines master output levels for all channels. The range of this control is from 0 dB of attenuation to +40 dB of gain.

Make-Up

⚠ *Applying large amounts of Make-Up gain will boost the level of any noise or hiss present in audio material, making it more audible.*

External On/Off

External On/Off enables and disables side-chain processing. With side-chain processing you can trigger compression from a separate reference track or external audio source. The source used for triggering side-chain processing is referred to as the *Key Input*.

📄 *See "Using a Key Input for External Side-Chain Processing" on page 57 for instructions on setting up and using a key input.*

External On/Off

Listen On/Off

Key Listen On/Off enables and disables auditioning of the Key Input (the reference track or external audio source used for triggering side-chain processing). This is useful for fine-tuning Impact's compression settings to the Key Input.

Listen On/Off

Gain Reduction Meter

The Gain Reduction meter is an analog-style meter that indicates the amount of gain reduction in dB. The range of this meter is from 0 dB to 40 dB. The gain reduction meter displays the amount of gain reduction linearly from 0–20 db, and non-linearly from 20–40 dB.

Gain Reduction meter

Input/Output Meters

The Input/Output meters indicate input and output signal levels in dB. When Impact is used in mono or stereo, both input and output meters are displayed. When Impact is used in a multichannel format, only output meters are displayed by default. You can toggle the meter display to show only input meters by clicking the blue-green rectangle at the lower right of the meter display.
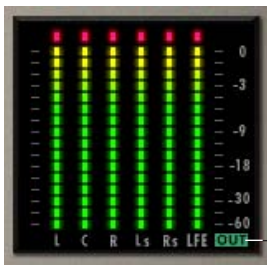
click here to toggle between input and output meters

Output meters (5.1 surround format shown)

A red clip indicator appears at the top of each meter. Clicking a clip indicator clears it. Alt-clicking (Windows) or Option-clicking (Mac) clears the clip indicators on all channels.

clip indicator
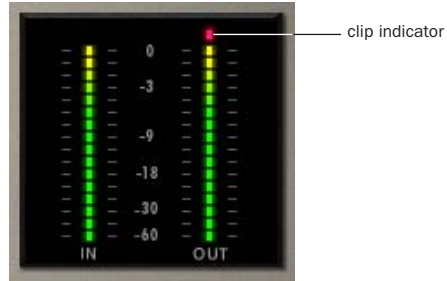
Input/Output meters (mono shown)
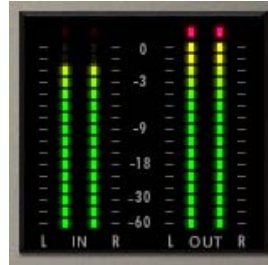
Input/Output meters (stereo shown)

Using a Key Input for External Side-Chain Processing

Impact provides side-chain processing capabilities. Side-chain processing lets you control Impact compression using an independent audio signal (typically, another Pro Tools track). In this way you can compress the audio of one track using the dynamics of a different audio track.

A typical use for side-chain processing is to control the dynamics of one audio signal using the dynamics of another signal (referred to as the Key Input). For example, you could use a lead vocal track to trigger compression of a background vocal track so that their dynamics match.

To use a Key Input signal for side-chain processing:

1 Click the Send button and select a bus path for the audio track or Auxiliary Input you want to use as the side-chain signal.

2 From Impact's Key Input menu, select the input or bus path carrying the audio you want to use as the side-chain signal to trigger Impact compression. The Key Input source must be monophonic.
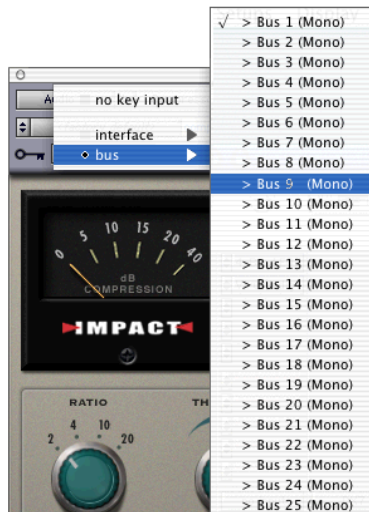
Choosing a Key Input

3 To activate external side-chain processing, click Ext.

4 Begin playback. Impact uses the input or bus that you selected as a Key Input to trigger its effect.

5 If you want to hear the audio source you have selected as the side-chain input, click Listen. (To stop listening to the side-chain input, click Listen again.)

☼ *Remember to disable Listen to resume normal plug-in monitoring.*

6 Adjust Impact's Threshold parameter to fine-tune Key Input triggering.

7 Adjust other parameters to achieve the desired effect.

chapter 8

Maxim

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

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

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

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

Maxim features include:

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

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

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

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

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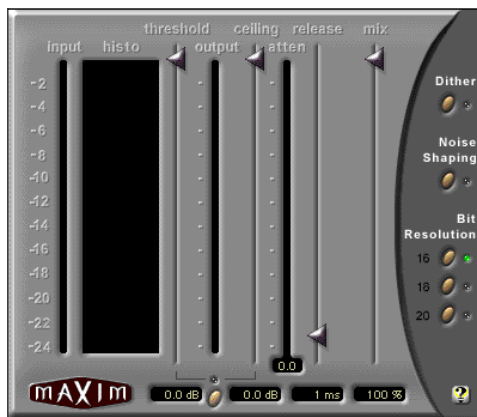
Maxim

About Peak Limiting

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

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

Limiting has two main uses in the audio production cycle:

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

• Adjusting the dynamic range of individual instruments for creative purposes

Limiting a Mixdown

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

Limiting Individual Instruments

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

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

How Maxim Differs From Conventional Limiters

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

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

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

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

Maxim Controls and Meters

Maxim features the following controls and indicators:

Input Level Meter

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

Histogram

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

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

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

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

dB level of waveform peaks

density of waveform peaks at each level
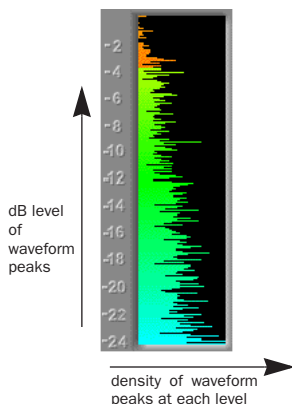
Histogram

Threshold Slider

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

Output Meter

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

Ceiling Slider

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

Attenuation Meter

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

Release Slider

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

Mix Slider

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

Link Button

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

Link button

Dither Button

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

Applying dither helps reduce quantization noise that can occur when you are mixing from a 24-bit TDM environment to a 16-bit destination, such as CD-R or DAT. If you are using Maxim on

a Master Fader during mixdown, Maxim's built-in dither function saves you the trouble and DSP resources of having to use a separate Dither plug-in.

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

Noise Shaping

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

Bit Resolution Button

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

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

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

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

Using Maxim

Following are suggestions for using Maxim most effectively.

To use Maxim:

1 Insert Maxim on the desired track.

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

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

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

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

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

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

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

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

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

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

Maxim and Mastering

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

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

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

Reverb One

Reverb One is a world-class reverb processing TDM plug-in. It provides a level of sonic quality and reverb-shaping control previously found only on the most advanced hardware reverberation units.

A set of unique, easy-to-use audio shaping tools lets you customize reverb character and ambience to create natural-sounding halls, vintage plates, or virtually any type of reverberant space you can imagine.

Reverb One features include:
- Editable Reverb EQ graph
- Editable Reverb Color graph
- Reverb Contour graph
- Dynamic control of reverb decay
- Chorusing
- Early reflection presets
- Extensive library of reverb presets
- Supports 44.1 khz, 48 kHz, 88.2 kHz, and 96 kHz processing.

⚠ *For sessions with a sample rate greater than 96 kHz, Reverb One will downsample and upsample accordingly.*

A Reverb Overview

Digital reverberation processing can simulate the complex natural reflections and echoes that occur after a sound has been produced, imparting a sense of space and depth—the signature of an acoustic environment. When you use a reverberation plug-in such as Reverb One, you are artificially creating a sound space with a specific acoustic character.

This character can be melded with audio material, with the end result being an adjustable mix of the original dry source and the reverberant wet signal. Reverberation can take relatively lifeless mono source material and create a stereo acoustic environment that gives the source a perceived weight and depth in a mix.

Creating Unique Sounds

In addition, digital signal processing can be used creatively to produce reverberation characteristics that do not exist in nature. There are no rules that need to be followed to produce interesting treatments. Experimentation can often produce striking new sounds.

Acoustic Environments

When you hear live sound in an acoustic environment, you generally hear much more than just the direct sound from the source. In fact, sound in an anechoic chamber, devoid of an acoustic space's character, can sound harsh and unnatural.

Each real-world acoustical environment, from a closet to a cathedral, has its own unique acoustical character or sonic signature. When the reflections and reverberation produced by a space combine with the source sound, we say that the space is *excited* by the source. Depending on the acoustic environment, this could produce the warm sonic characteristics we associate with reverberation, or it could produce echoes or other unusual sonic characteristics.

Reverb Character

The character of a reverberation depends on a number of things. These include proximity to the sound source, the shape of the space, the absorptivity of the construction material, and the position of the listener.

Reflected Sound

In a typical concert hall, sound reaches the listener shortly after it is produced. The original direct sound is followed by reflections from the ceiling or walls. Reflections that arrive within 50 to 80 milliseconds of the direct sound are called *early reflections*. Subsequent reflections are called *late reverberation*. Early reflections provide a sense of depth and strengthen the perception of loudness and clarity. The delay time between the arrival of the direct sound and the beginning of early reflections is called the *pre-delay*.

The loudness of later reflections combined with a large pre-delay can contribute to the perception of largeness of an acoustical space. Early reflections are followed by reverberation and repetitive reflections and attenuation of the original sound reflected from walls, ceilings, floors, and other objects. This sound provides a sense of depth or size.

Reverb One provides control over these reverberation elements so that extremely natural-sounding reverb effects can be created and applied in the Pro Tools mix environment.

Reverb One Controls

Reverb One has a variety of controls for producing a wide range of reverb effects. Controls can be adjusted by dragging their sliders or typing values directly in their text boxes.
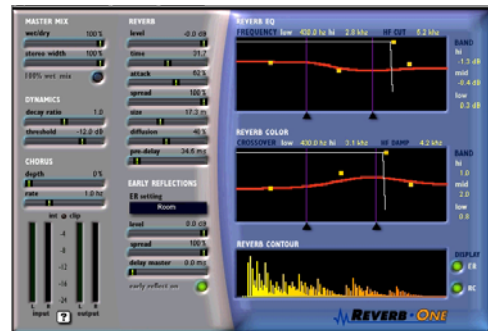
Reverb One

Editing Graph Values

In addition to the standard slider controls, the Reverb EQ and Reverb Color graph settings can be adjusted by dragging elements of the graph display.

To cut or boost a particular band:

■ Drag a yellow breakpoint up or down.

To adjust frequency or crossover:

■ Drag a triangular slider right or left.

To adjust high-frequency cut or damp:

■ Drag the yellow dot right or left.
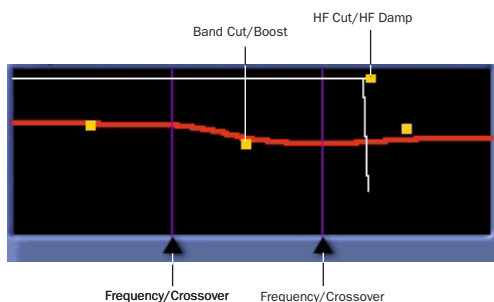
Adjusting graph controls

Master Mix Controls

The Master Mix section has controls for adjusting the relative levels of the source signal and the reverb effect, and also the width of the reverb effect in the stereo field.

Master Mix section

Wet/Dry Adjusts the mix between the dry, unprocessed signal and the reverb effect.

Stereo Width Controls the width of the reverb in the stereo field. A setting of 0% produces a mono reverb. A setting of 100% produces maximum spread in the stereo field.

100% Wet Toggles the Wet/Dry control between 100% wet and the current setting.

Dynamics Controls

The Dynamics section has controls for adjusting Reverb One's response to changes in input signal level.

Dynamics can be used to modify a reverb's decay character, making it sound more natural, or conversely, more unnatural, depending on the desired effect.

Typically, dynamics are used to give a reverb a shorter decay time when the input signal is above the threshold, and a longer decay time when the input level drops below the threshold.

This produces a longer, more lush reverb tail and greater ambience between pauses in the source audio, and a shorter, clearer reverb tail in sections without pauses.

For example, on a vocal track, use Dynamics to make the reverb effect tight, clear, and intelligible during busy sections of the vocal (where the signal is above the Threshold setting), and then "bloom" or lengthen at the end of a phrase (where the signal falls below the threshold).

Similarly, Dynamics can be used on drum tracks to mimic classic gated reverb effects by causing the decay time to cut off quickly when the input level is below the threshold.

To hear examples of decay dynamics, load one of the Dynamics presets with the Librarian.
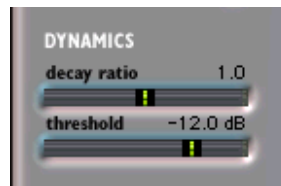
Dynamics section

Decay Ratio Controls the ratio by which reverb time is increased when a signal is above or below the Threshold level. Dynamics behavior differs when the Decay Ratio is set above or below 1. A ratio setting of greater than 1 increases reverb time when the signal is *above* the threshold. A ratio setting of less than 1 increases a reverb's time when the signal is *below* the threshold.

For example, if Decay Ratio is set to 4, the reverb time is increased by a factor of 4 when the signal is above the threshold level. If the ratio is 0.25, reverb time is increased by a factor of 4 when the signal is below the Threshold level.

Threshold Sets the input level above or below which reverb decay time will be modified.

Chorus Controls

The Chorus section has controls for setting the depth and rate of chorusing applied to a reverb tail. Chorusing thickens and animates sounds by adding a delayed, pitch-modulated copy of an audio signal to itself.

Chorusing produces a more ethereal or spacey reverb character. It is often used for creative effect rather than to simulate a realistic acoustic environment.

💡 *To hear examples of reverb tail chorusing, load one of the Chorus presets with the Librarian.*
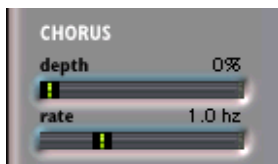
Chorus section

Depth Controls the amplitude of the sine wave generated by the LFO (low frequency oscillator) and the intensity of the chorusing. The higher the setting, the more intense the modulation.

Rate Controls pitch modulation frequency. The higher the setting, the more rapid the chorusing. Setting the Rate above 20 Hz can cause frequency modulation to occur. This will add sideband harmonics and change the reverb's tone color, producing some very interesting special effects.

Reverb Controls

The Reverb section has controls for the various reverb tail elements, including level, time, attack, spread, size, diffusion, and pre-delay. These determine the overall character of the reverb.

Reverb section

Level Controls the output level of the reverb tail. When set to 0%, the reverb effect consists entirely of the early reflections (if enabled).

Time Controls the rate at which the reverberation decays after the original direct signal stops. The value of the Time setting is affected by the Size setting. You should adjust the reverb Size

setting before adjusting the Time setting. If you set Time to its maximum value, infinite reverberation is produced. The HF Damping and Reverb Color controls also affect reverb Time.

Attack Attack determines the contour of the reverberation envelope. At low Attack settings, reverberation builds explosively, and decays quickly. As Attack value is increased, reverberation builds up more slowly and sustains for the length of time determined by the Spread setting.

When Attack is set to 50%, the reverberation envelope emulates a large concert hall (provided the Spread and Size controls are set high enough).

Spread Controls the rate at which reverberation builds up. Spread works in conjunctions with the Attack control to determine the initial contour and overall ambience of the reverberation envelope.

Low Spread settings result in a rapid onset of reverberation at the beginning of the envelope. Higher settings lengthen both the attack and buildup stages of the initial reverb contour.

Size Determines the rate of diffusion buildup and acts as a master control for Time and Spread within the reverberant space.

Size values are given in meters and can be used to approximate the size of the acoustic space you want to simulate. When considering size, keep in mind that the size of a reverberant space in meters is roughly equal to its longest dimension.

Diffusion Controls the degree to which initial echo density increases over time. High Diffusion settings result in high initial buildup of echo density. Low Diffusion settings cause low initial buildup.

After the initial echo buildup, Diffusion continues to change by interacting with the Size control and affecting the overall reverb density. Use high Diffusion settings to enhance percussion. Use low or moderate settings for clearer, more natural-sounding vocals and mixes.

Pre-Delay Determines the amount of time that elapses between the original audio event and the onset of reverberation. Under natural conditions, the amount of Pre-delay depends on the size and construction of the acoustic space, and the relative position of the sound source and the listener. Pre-delay attempts to duplicate this phenomenon and is used to create a sense of distance and volume within an acoustic space. Long Pre-Delay settings place the reverberant field behind rather than on top of the original audio signal.

☼ *For an interesting musical effect, set the Pre-Delay time to a beat interval such as 1/8, 1/16, or 1/32 notes.*

Early Reflection Controls

The Early Reflections section has controls for the various early reflection elements, including ER setting, level, spread, and delay.

Calculating Early Reflections

A particular reflection within a reverberant field is usually categorized as an early reflection. Early reflections are usually calculated by measuring the reflection paths from source to listener. Early reflections typically reach the listener within 80 milliseconds of the initial audio event, depending on the proximity of reflecting surfaces.

Simulating Early Reflections

Different physical environments have different early reflection signatures that our ears and brain use to pinpoint location information. These reflections influence our perception of the size of a space and where an audio source sits within it. Changing early reflection characteristics changes the perceived location of the reflecting surfaces surrounding the audio source.

This is commonly accomplished in digital reverberation simulations by using multiple delay taps at different levels that occur in different positions in the stereo spectrum (through panning). Long reverberation generally occurs after early reflections dissipate.

Reverb One provides a variety of early reflections models. These let you quickly choose a basic acoustic environment, then tailor other reverb characteristics to meet your precise needs.

Early Reflections section

ER Settings Selects an early reflection preset. These range from realistic rooms to unusual reflective effects. The last five presets (Plate, Build, Spread, Slapback and Echo) feature a nonlinear response.

Early reflection presets include:

- Room: Simulates the center of a small room without many reflections.
- Club: Simulates a small, clear, natural-sounding club ambience.
- Stage: Simulates a stage in a medium-sized hall.
- Theater: Simulates a bright, medium- sized hall.
- Garage: Simulates an underground parking garage.
- Studio: Simulates a large, live, empty room.
- Hall: Places the sound in the middle of a hall with reflective, hard, bright walls.
- Soft: Simulates the space and ambience of a large concert hall.
- Church: Simulates a medium-sized space with natural, clear-sounding reflections.
- Cathedral: Simulates a large space with long, smooth reflections.
- Arena: Simulates a big, natural-sounding empty space.
- Plate: Simulates a hard, bright reflection. Use the Spread control to adjust plate size.
- Build: A nonlinear series of reflections
- Spread: Simulates a wide indoor space with highly reflective walls.
- Slapback: Simulates a large space with a long-delayed reflection.
- Echo: Simulates a large space with hard, unnatural echoes. Good for dense reverb.

Level Controls the output level of the early reflections. Turning the Early Reflections Level slider completely off produces a reverb made entirely of reverb tail.

Spread Globally adjusts the delay characteristics of the early reflections, moving them closer together or farther apart. Use Spread to vary the size and character of an early reflection preset. Setting the Plate preset to a Spread value of 50%, for example, will change the reverb from a large, smooth plate to a small, tight plate.

Delay Master Determines the amount of time that elapses between the original audio event and the onset of early reflections.

Early Reflect On Toggles early reflections on or off. When early reflections are off, the reverb consists entirely of reverb tail.

Reverb Graphs

The reverb graphs display information about the tonal spectrum and envelope contour of the reverb. The Reverb EQ and Reverb Color graphs provide graphic editing tools for shaping the harmonic spectrum of the reverb.

Reverb EQ

You can use this 3-band equalizer to shape the tonal spectrum of the reverb. The EQ is post-reverb and affects both the reverb tail and the early reflections.
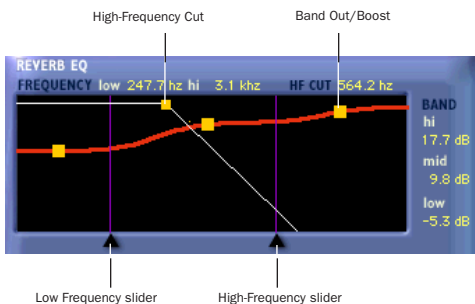
Reverb EQ graph

Frequency Sliders Sets the frequency boundaries between the low, mid, and high band ranges of the EQ.

The low frequency slider (60.0 Hz–22.5 kHz) sets the frequency boundary between low and mid cut/boost points in the EQ.

The high-frequency slider (64.0 Hz–24.0 kHz) sets the frequency boundary between the mid and high cut/boost points in the EQ.

Band Breakpoints Control cut and boost values for the low, mid, and high frequencies of the EQ. To cut a frequency band, drag a breakpoint downward. To boost, drag upward. The adjustable range is from –24.0 dB to 12.0 dB.

HF Cut Breakpoint Sets the frequency above which a 6 dB/octave low pass filter attenuates the processed signal. It removes both early reflections and reverb tails, affecting the overall high-frequency content of the reverb. Use the HF Cut control to roll off high frequencies and create more natural-sounding reverberation. The adjustable range is from 120.0 Hz to 24.0 kHz.

Reverb Color

You can use the Reverb Color graph to shape the tonal spectrum of the reverb by controlling the decay times of the different frequency bands. Low and high crossover points define the cut and boost points of three frequency ranges.

For best results, set crossover points at least two octaves higher than the frequency you want to boost or cut. For example, to boost a signal at 100 Hz, set the crossover to 400 Hz.

Set the crossover to 500 Hz to boost low frequencies most effectively. Set it to 1.5 kHz to cut low frequencies most effectively.
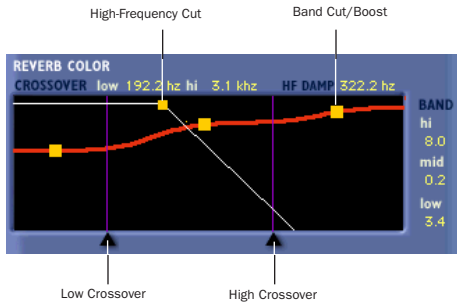
Reverb Color graph

Crossover Sliders Sets the frequency boundaries between the low, mid, and high frequency ranges of the reverberation filter.

The low-frequency slider sets the crossover frequency between low and mid frequencies in the reverberation filter. The adjustable range is from 60.0 Hz to 22.5 kHz.

The high-frequency slider sets the crossover frequency between mid and high frequencies in the reverberation filter. The adjustable range is from 64.0 Hz to 24.0 kHz.

Band Breakpoints Controls cut and boost ratios for the decay times of the low, mid, and high-frequency bands of the reverberation filter. To cut a frequency band, drag a breakpoint downward. To boost, drag it upward. The adjustable range is from 1:8 to 8:1.

HF Damp Breakpoint Sets the frequency above which sounds decay at a progressively faster rate. This determines the decay characteristic of the high-frequency components of the reverb.

HF Damp works in conjunction with HF Cut to shape the overall high -frequency contour of the reverb. HF Damp filters the entire reverb with the exception of the early reflections. At low set-

tings, high frequencies decay more quickly than low frequencies, simulating the effect of air absorption in a hall. The adjustable range is from 120.0 Hz to 24.0 kHz.

Reverb Contour

The Reverb Contour graph displays the envelope of the reverb, as determined by the early reflections and reverb tail.

Reverb Contour graph

ER and RC Buttons Toggles the display mode. Selecting ER (early reflections) displays early reflections data in the graph. Selecting RC (reverb contour) displays the initial reverberation envelope in the graph. Early Reflections and Reverb Contour can be displayed simultaneously.

Other Controls

In addition to its reverb-shaping controls, Reverb One also features online help and level metering.

Online Help

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

Online help button

Input Level Meters

Input meters indicate the input levels of the dry audio source signal. Output meters indicate the output levels of the processed signal.

An internal clipping LED will light if the reverb is overloaded. This can occur even when the input levels are relatively low if there is excessive feedback in the delay portion of the reverb. To clear the Clip LED, click it.
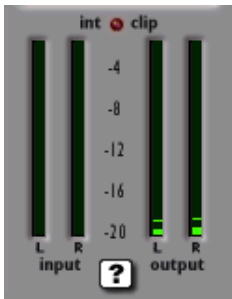
Reverb One meters

ReVibe

ReVibe provides studio-quality reverb and acoustic environment modeling for mono, stereo, and greater-than-stereo multichannel audio formats. Revibe offers extensive control over reverb characteristics, and a diverse array of room reflection and coloration presets.

ReVibe makes it possible to model extremely realistic acoustic spaces and place audio elements within them in a Pro Tools mix.

⚠ *ReVibe requires one or more HD Accel cards.*

ReVibe plug-in

Reverberation Concepts

Digital reverberation processing can simulate the complex natural reflections and echoes that occur after a sound has been produced, imparting a sense of space and depth—the signature of an acoustic environment. When you use a reverberation plug-in such as ReVibe, you are artificially creating a sound space with a specific acoustic character.

This character can be melded with audio material, with the end result being an adjustable mix of the original dry source and the reverberant wet signal. You can use reverberation to enhance relatively lifeless mono source material with a stereo acoustic environment that gives the source audio a perceived weight and depth in a mix.

Acoustic Environments

When you hear live sound in an acoustic environment, you generally hear much more than just the direct sound from the source. In fact, sound in an anechoic chamber, devoid of an acoustic space's character, can sound harsh and unnatural.

Each real-world acoustical environment, from a closet to a cathedral, has its own unique acoustical character or sonic signature. When the reflections and reverberation produced by a space combine with the source sound, the space is said to be *excited* by the source. Depending on the acoustic environment, this could produce the warm sonic characteristics associated with reverberation, or it could produce echoes or other unusual sonic characteristics.

Reverb Character

Reverb character depends on many factors including the shape of the space, the reflectivity of the construction material, the proximity of reflective elements to the sound source, and the position of the listener.

Reflected Sound

In a typical concert hall, sound reaches the listener shortly after it is produced. The original direct sound is followed by reflections from the ceiling or walls. These discrete reflections, which usually arrive within 100 milliseconds of the direct sound, are called *early reflections*. The subsequent, and more diffuse reflections, are called the *reverb tail*. The delay time between the arrival of the direct sound and the beginning of the reflected sounds is called the *pre-delay*.

The loudness and panning of early reflections combined with the length of the pre-delay can contribute to the perception of size of an acoustical space.

ReVibe also uses *Room Coloration* to accurately model acoustic spaces and effects. Room Coloration is a complex filter process, similar to EQ, that models the frequency shape of each room or effect.

ReVibe provides control over these reverb elements so that extremely natural-sounding reverb effects can be created and applied in the Pro Tools mix environment.

ReVibe can also be used to produce reverb characteristics that do not exist in nature. There are no rules that you need to follow to produce interesting treatments. Experimentation can often produce striking results.

Using ReVibe

ReVibe supports 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz sessions. ReVibe works with mono and stereo formats, and LCR, LCRS, quad, 5.0, and 5.1 greater-than-stereo multichannel formats.

In general, when working with stereo and greater-than-stereo tracks, use the multichannel version of ReVibe.

Revibe supports the following combinations of track types and plug-in insert formats:

Table 11. Supported multichannel formats for ReVibe

Track	Plug–in Insert Format						
	Mono	**Stereo**	**L C R**	**L C R S**	**Quad**	**5.0**	**5.1**
Mono	●	●	●	●	●	●	●
Stereo		●	●	●	●	●	●
L C R			●	●	●	●	●
L C R S				●			
Quad					●		
5.0						●	
5.1							●

Adjusting ReVibe Parameters

Editing Slider Controls with a Mouse

You can adjust slider controls with a mouse by dragging horizontally. Parameter values increase as you drag to the right, and decrease as you drag to the left.

Some sliders (such as the Diffusion slider) are *bipolar*, meaning that their zero position is in the center of the slider's range. Dragging to the right of center creates a positive parameter value; dragging to the left of center generates a negative parameter value.

Editing Graph Parameters with a Mouse

You can adjust parameters on the Decay Color & EQ graph with a mouse by dragging the appropriate dot on the graph.

To cut or boost a particular EQ band:

■ Drag a control dot up or down.

Cutting or boosting an EQ frequency band

To adjust EQ frequency crossover:

■ Drag the control dot right or left.

Setting the EQ crossover frequency

To adjust high frequency rear cut:

■ Drag the control dot right or left.

Setting the rear cut frequency

Editing Parameters with a Computer Keyboard

Each control has a corresponding parameter text field that displays the current value of the parameter. You can edit the numeric value of a parameter with your computer keyboard.

To change control values with a computer keyboard:

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

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.

For parameters with values in kilohertz, typing "k" after a number value will multiply the value by 1000. For example, type "8k" to enter a value of 8000.

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 from a selected parameter to the next parameter, press the Tab key. To move backward, press Shift+Tab.

Enabling Switches

To enable a switch, click on the switch (the round LED indicator next to each switch name). Switch LEDs illuminate when enabled.

Early Reflection switch LED

ReVibe Controls

Master Mix Section

The Master Mix section has controls for adjusting the relative levels of the source signal and the reverb effect.

Master Mix controls

Wet/Dry Control

Wet/Dry adjusts the mix between the dry, unprocessed signal and the reverb effect. If you insert the ReVibe plug-in directly onto an audio track, settings from 30% to 60% are a good starting point for experimenting with this parameter. The range of this control is from 0% to 100%.

You can also achieve a 100% wet mix by clicking the 100% Wet Mix button.

Stereo Width Control

Stereo Width controls the stereo field spread of the front reverb channels. A setting of 0% produces a mono reverb, but leaves the panning of the original source signal unaffected. A setting of 100% produces a hard panned stereo image.
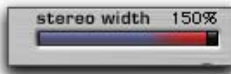
Stereo Width control

Settings above 100% use phase inversion to create an even wider stereo effect. The Stereo Width slider displays red above the 100% mark to remind you that a phase effect is being used to widen the stereo field.

The range of this control is from 0% to 150%. The default setting is 100%.

⚠ *The Stereo Width control does not affect the reverberation effect coming through the rear channels. If you want to produce a strictly mono reverb, be sure to set the Rear Reverb parameter (Levels section) to –INF dB.*

100% Wet Mix Button

This button toggles the Wet/Dry control between 100% wet and the current setting. A 100% wet mix contains only the reverb effect with none of the direct signal. This setting can be useful when using pre-fader sends to achieve send/return bussing. The wet/dry balance in the mix can be controlled using the track faders for the dry signal, and the Auxiliary input fader for the effect return.

Chorus Section

The Chorus section has controls for adjusting the depth and rate of chorusing applied to the reverb tail. Chorusing thickens and animates sounds and produces a more ethereal reverb character. It is often used for creative effects rather than to simulate a realistic acoustic environment.
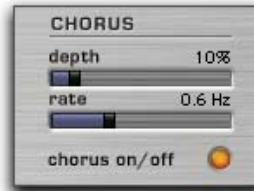
Chorus controls

Depth Control

Depth controls the amplitude of the sine wave generated by the LFO (low frequency oscillator) and the intensity of the chorusing. The higher the setting, the more intense the modulation. The range of this control is from 0% to 100%.

Rate Control

Rate controls the frequency of the LFO. The higher the setting, the more rapid the chorusing. The range of this control is from 0.1 Hz to 30.0 Hz.

Setting the Rate above 20 Hz can cause frequency modulation to occur. This will add sideband harmonics and change the reverb's tone color, producing interesting effects. Typical settings are between 0.2 Hz and 1.0 Hz.

Chorus On/Off Button

This button toggles the chorus effect on or off.

Chorus on/off button

Early Reflection Section

Different physical environments have different early reflection signatures that our ears and brain use to pinpoint location information in physical space. These reflections influence our perception of the size of a space and where an audio source sits within it.

Changing early reflection characteristics changes the perceived location of the reflecting surfaces surrounding the audio source. In general, the reverb tail continues after early reflections dissipate.

ReVibe room presets use multiple delay taps at different levels, different times, and in different positions in the multichannel environment (through 360° panning) to create extremely realistic sounding environments.

The Early Reflect section has controls for adjusting the various early reflection elements, including level, spread, and pre-delay.
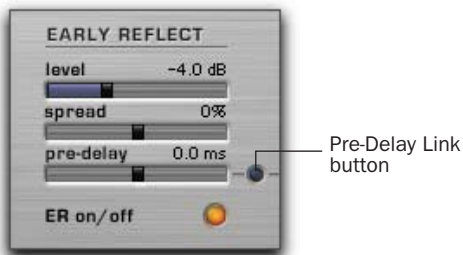
Early Reflect section

Level Control

Level controls the output level of the early reflections. Setting the Level slider to –INF (minus infinity) eliminates the early reflections from the reverb effect. The range of this control is from –INF to 6.0 dB.

Spread Control

Spread globally adjusts the delay characteristics of the early reflections, moving the individual delay taps closer together or farther apart. Use Spread to vary the size and character of an early reflection preset. The range of this control is from –100% to 100%.

At 0%, the early reflections are set to their optimum value for the room preset. Typical spread values range between –25% and 25%.

💡 *Setting Spread to 100% produces very widely spaced early reflections that may sound unnatural. At –100% the early reflections have no spread at all, and are heard as a single reflection.*

Pre-Delay Control

The Pre-Delay control in the Early Reflect section determines the amount of time that elapses between the onset of the dry signal and the first early reflection delay tap. Some Room Types, such as those that produce slapback effects, have additional built-in pre-delay. The range of this control is from –300.0 ms to 300.0 ms.

Negative Pre-Delay times imply that some early reflection delay taps should sound before the original dry signal. Since this is not possible, any of the delay taps that would sound before the dry signal are not used and do not sound.

When Pre-Delay Link is enabled, negative early reflection Pre-Delay times can be used to make the early reflections start before the reverb tail, if desired.

Pre-Delay Link Button

The Pre-Delay Link button toggles linking of the Early Reflection Pre-Delay control and the Reverb Pre-Delay control. When linked, the Early Reflection Pre-Delay is offset by the Reverb Pre-Delay amount, so that the total delay for the early reflections is the sum of the Early Reflection Pre-Delay and the Reverb Pre-Delay.

ER On/Off Button

This button toggles early reflections on or off. When early reflections are off, the reverb effect consists entirely of reverb tail.
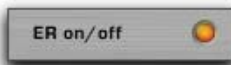
ER On/Off button

Levels Section

The Levels section has controls for adjusting source input and ReVibe output levels. ReVibe provides individual output level controls for front, center, rear reverb, and rear early reflections.
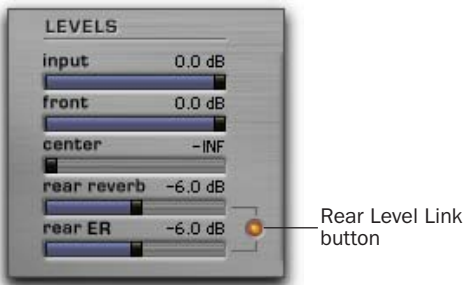
Levels controls

In stereo and greater-than-stereo formats where there is no center channel or where there are no rear channels, the center and rear level controls can be used to augment the reverb sound. Reverb and early reflections that would be heard either from the center channel or from the rear channels can be mixed into the front left and right channels.

Input Control

Input adjusts the level of the source input to prevent internal clipping. The range of this control is from –24.0 dB to 0.0 dB. Lowering the Input control does not change the levels shown on the input side of the Input/Output meter, which shows the level of the signal before the Input control.

Front Control

Front controls the output level of the front left and right outputs. Front is also the main level control for stereo. The range of this control is from –INF (minus infinity) to 0.0 dB.

Center Control

Center controls the output level of the center channel outputs of multichannel formats that have a center channel (such as LCR or 5.1).

When ReVibe is used in a multichannel format that has no center channel (such as stereo or quad), the Center level control adjusts a phantom center channel signal that is center-panned to the front left and right outputs.

The range of this control is from –INF (minus infinity) to 0.0 dB.

Rear Reverb Control

Rear Reverb controls the output level of the rear outputs of multichannel formats that have rear channels (such as quad or 5.1).

When ReVibe is used in a multichannel format that has no rear channels (such as a stereo or LCR) the Rear level control instead adjusts rear channel signals hard-panned to the front left and right outputs.

The range of this control is from –INF (minus infinity) to 0.0 dB.

Rear ER Control

Rear ER controls the output level of early reflections in the rear outputs. The range of this control is from –INF (minus infinity) to 0.0 dB.

The Rear ER control has no effect when the early reflections are turned off with the ER On/Off button.

Rear Level Link Button

The Rear Level Link button toggles linking of the Rear Reverb and Rear ER controls on or off. The Rear Reverb and the Rear ER controls are linked by default. When linked, the Rear ER and Rear Reverb controls move in tandem when either is adjusted. When unlinked, the Rear ER and the Rear Reverb controls can be adjusted independently.
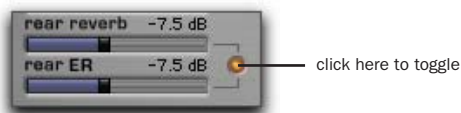
Rear Level Link button

Room Type Section

The controls in the Room Type section let you select a Room Type, which models early reflection characteristics for specific types of rooms or effects devices. Each Room Type also incorporates a complex room coloration EQ, which models the general frequency response of various rooms and effects devices.

Choosing a new Room Type changes the early reflections and room coloration EQ only. All of the other ReVibe parameters and setting remain unchanged. To create a preset that includes all parameters, use the Settings Librarian.

For more information on saving and importing plug-in settings using the Setting Librarian, see the Digidesign Plug-ins Guide.
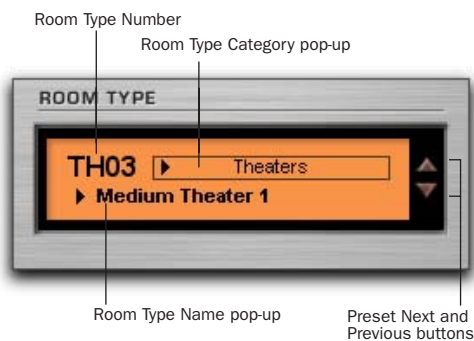
Room Type display and controls

The Room Type display shows the Room Type Category, Room Type Name, Room Type Number and the Next and Previous browse buttons.

Room Type Category Menu

Clicking on the Room Type Category pop-up menu lets you select one of the 14 Room Type categories, and selects the first Room Type preset in that category.

Room Type Name Menu

Click the Room Type Name pop-up menu to select from a list of all available Room Type presets.

📖 *See "ReVibe Room Types" on page 90 for a list of room presets.*

Room Type Number Field

The Room Type Number field displays the Room type number for the current Room Type.

Next and Previous Buttons

Click the Next or Previous buttons to choose the next or previous Room Type.

Room Coloration Section

The Room Coloration controls work in conjunction with the selected Room Type. Coloration takes the characteristic resonant frequencies or EQ traits of the room and allows you to apply this spectral shape to the reverb.

In addition to letting you adjust the overall sound of the room, the high-frequency and low-frequency components are split to allow you to emphasize or de-emphasize the low and high frequency response of the room.
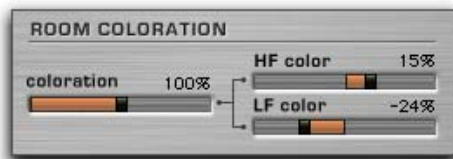
Room Coloration controls

Coloration Control

Coloration adjusts how much of the EQ characteristics of the selected Room Type are applied to the original signal. The range of this control is from 0% to 200%. A setting of 100% provides the optimum coloration for the room type. Settings above 100% will tend to produce extreme and unnatural coloration.

HF Color Control

HF Color adds or subtracts additional high frequency coloration, or relative brightness, to the acoustic model of the room. The range of this control is from –50.0% to 50.0%.

LF Color Control

LF Color adds or subtracts additional low frequency coloration, or relative darkness, to the acoustic model of the room. The range of this control is from –50.0% to 50.0%.

Reverb Section

The Reverb section has controls for the various reverb tail elements, including type, level, time, size, spread, attack time, attack shape, rear shape, diffusion, and pre-delay. These determine the overall character of the reverb tail.
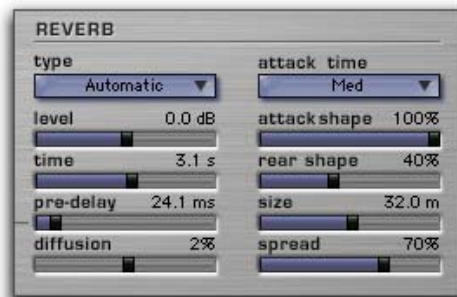
Reverb Controls

Type Menu

Type is a pop-up menu that sets the type of reverb tail. There are nine basic reverb types, plus the *Automatic* type. Selecting the Automatic reverb type will select the type of reverb tail that is stored with the currently selected room type. The reverb types are:

- *Automatic* selects the reverb tail type stored with the room type.
- *Natural* is an average reverb tail type with no extreme characteristics.
- *Smooth* is optimized for large rooms.
- *Fast Attack* can be useful for plate reverbs.
- *Dense* is similar to smooth, and can also be good for a plate reverb.
- *Tight* is good for small to medium rooms.
- *Sparse 1* produces sparse early reflections with a high diffusion buildup.
- *Sparse 2* can be useful for a spring reverb.
- *Wide* is a generic large reverb.
- *Small* is optimized for small rooms.

Level Control

Level controls the output level of the reverb tail. When set to –INF (minus infinity) no reverb tail is heard, and the reverb effect consists entirely of the early reflections (if enabled). The range of this control is from –INF to 6.0 dB.

Time Control

Time controls how long the reverberation continues after the original source signal stops. The range of this control is from 100.0 ms to Inf (infinity). Setting Time to its maximum value will produce infinite reverberation.

Pre-Delay Control

The Pre-Delay control in the Reverb section sets the amount of time that elapses between signal input and the onset of the reverb tail.

Under natural conditions, the amount of predelay 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. Extremely long pre-delay settings produce effects that are unnatural but sonically interesting.

The range of this control is from 0.0 ms to 300.0 ms.

Diffusion Control

Diffusion controls the rate that the sound density of the reverb tail increases over time. The control ranges between –50% and 50%. At 0%, diffusion is set to an optimal preset value. Positive Diffusion settings create a longer initial buildup of echo density. At negative settings, the buildup of echo density is slower than at the optimal preset value.

Attack Time Control

Attack Time adjusts the length of time between the start of the reverb tail and its peak level. Settings are Short, Medium, or Long.

Attack Shape Control

Attack Shape determines the contour of the attack portion of the reverberation envelope. At 0%, there is no buildup contour, and the reverb tail begins at its peak level. At a high Attack Shape setting the reverb tail begins at a relatively low initial level and ramps up to the peak reverb level. The range of this control is from 0% to 100%.

Rear Shape Control

Rear Shape adjusts the envelope of the reverb in the rear channels to control the length of the attack time. This gives more reverb presence and a longer reverb bloom in the rear channels. The range of this control is from 0% to 100%.

Size Control

The Size control adjusts the apparent size of the reverberant space from small to large. Set the Size control to approximate the size of the acoustic space you want to simulate. Size values are given in meters. The range of this control is from 2.0 m to 60.0 m (though relative size will change based on the current Room Type).

Larger settings of the Size parameter increase both the Time and Spread parameters.

☼ *When specifying reverb size, keep in mind that the size of a reverberant space in meters is approximately equal to its longest dimension. In general, halls range from 25 m to 50 m; large to medium rooms range from 15 m to 30 m; and small rooms range from 5 m to 20 m. Similarly, a Room Size setting of 20m corresponds roughly to a 4x8 plate.*

Spread Control

Spread controls the rate at which reverberation builds up. Spread works in conjunction with the Attack Shape control to determine the initial contour and overall ambience of the reverberation envelope.

At low Spread settings there is a rapid onset of reverb at the beginning of the reverberation envelope. Higher settings lengthen both the attack and buildup of the initial reverb contour. The range of this control is from 0% to 100%.

Decay Color & EQ Section

The Decay Color and EQ section provides an editable graphic display of reverb decay color parameters and EQ parameters. Click the EQ button to toggle the display to show EQ parameters. Click the Color button to toggle the display to show Color parameters. To edit a parameter on the graph, drag the appropriate dot.
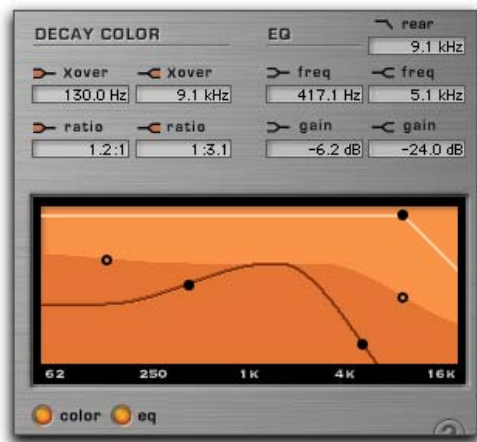
Decay Color & EQ display

Each control point (dot) on the graph has corresponding parameter text fields above the display that show the current parameter values. You can edit the numeric value of a parameter with your computer keyboard. (See "Editing Parameters with a Computer Keyboard" on page 78.)

Decay Color Section

You can use the controls in the Decay Color section to shape the tonal spectrum of the reverb by adjusting the decay times of the low and high frequency ranges. Low and high crossover points define the cut and boost points of three frequency ranges.

For best results, set crossover points at least one octave higher than the frequency you want to boost or cut. To boost a signal at 200 Hz, for example, set the crossover to 400 Hz.

Low Frequency Crossover Control

Low Frequency Crossover sets the crossover frequency at which transitions from low frequencies to mid frequencies take place in the reverberation filter. The range of this control is from 50.0 Hz to 1.5 kHz.

Low Frequency Crossover control

Low Frequency Ratio Control

Low Frequency Ratio sets cut or boost ratios for the decay times of the low and mid frequency bands of the reverberation filter. The range of this control is between 1:16.0 and 4.0:1.

Low Frequency Ratio control

High Frequency Crossover Control

High Frequency Crossover sets the crossover frequency at which transitions from mid frequencies to high frequencies take place in the reverberation filter. The range of this control is from 1.5 kHz to 20.0 kHz.

High Frequency Crossover control

High Frequency Ratio Control

High Frequency Ratio sets cut or boost ratios for the decay times of the mid and high frequency bands of the reverberation filter. The range of this control is between 1:16.0 and 4.0:1.

High Frequency Ratio control

Decay EQ Section

Low Frequency Control

Low Frequency sets the frequency boundary between low and mid cut or boost points in the reverb EQ. The range of this control is from 50.0 Hz to 1.5 kHz.

Low Frequency control

Low Gain Control

Low Gain sets cut and boost values for the low and mid frequencies of the reverb decay EQ. The range of this control is from –24.0 dB to 12.0 dB.
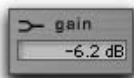
Low Gain control

High Frequency Control

High Frequency sets the frequency boundary between mid and high cut or boost points in the reverb EQ. The range of this control is from 1.5 kHz to 20.0 kHz.
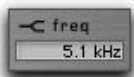
High Frequency control

High Gain Control

High Gain sets cut and boost values for the mid and high frequencies of the reverb decay EQ. The range of this control is from –24.0 dB to 12.0 dB.
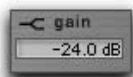
High Gain control

High Frequency Rear Cut Control

High Frequency Rear Cut rolls off additional high frequencies in the rear channels of the early reflections and reverb tail. The application of this filter is distinct from the application of Decay Color and Decay EQ. The range of this control is from 250.0 Hz to 20.0 kHz.
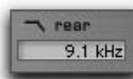
High Frequency Rear Cut control

Online Help Button

Click the name of any control and information about that control will appear. Clicking the Online Help button provides additional details on using this feature.

Online Help

Contour Display

The Contour display shows the current reverb shape and early reflections as a two-dimensional graph. Both front and rear reverb tail shapes and early reflections can be viewed at the same time. Buttons below the display allow you to select the type of data being displayed.
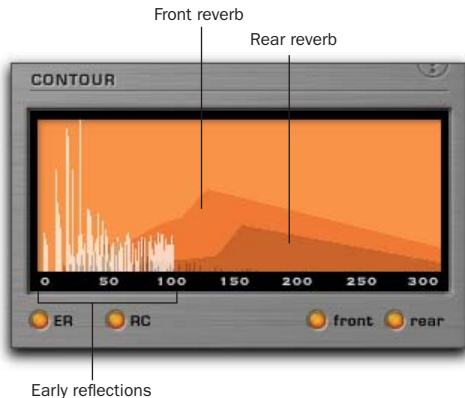
Contour display

ER Button

The ER (early reflections) button toggles display of early reflections on or off within the Contour display. When the ER button is illuminated, early reflections data is displayed. When the ER button is not illuminated, early reflections data is not displayed. Both early reflections and reverb contour data can be displayed simultaneously.

RC Button

The RC (reverb contour) button toggles display of the reverb contours for both the front and rear channels on or off within the Contour display. When the RC button is illuminated, the reverberation envelopes are displayed. When the RC button is not illuminated, the reverberation envelopes are not displayed. Both early reflections and reverb contour data can be displayed simultaneously.

Front Button

The Front button toggles display of the front channel reverb contour and the front channel early reflections on or off within the Contour display. When the Front button is illuminated, the initial reverberation envelope and early reflections for the front channels are displayed. When the Front button is not illuminated, they are not displayed.

Rear Button

The Rear button toggles display of the rear channel reverb contour and the rear channel early reflections on or off within the Contour display. When the Rear button is illuminated, the initial reverberation envelope and early reflections for the rear channels are displayed. When the Rear button is not illuminated, they are not displayed.

Input/Output Meter

The Input/Output meter indicates the input signal and the ReVibe output. The range of this meter is from 0 dB to –60 dB. The number of input/output meters that operate simultaneously ranges from a single meter for mono input and output, up to five input/output meters for 5.0 and 5.1 multichannel processing. The meters that operate depend on the channel format of the track on which the plug-in is inserted.
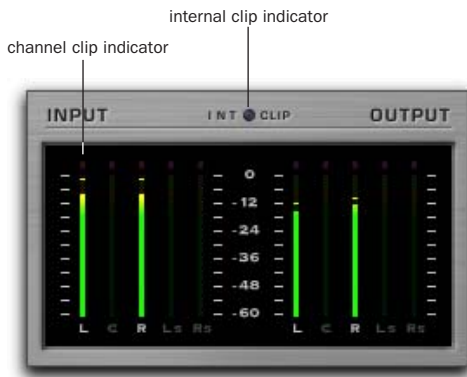
Input/Output Meter

Clip indicators

A red channel clip indicator appears at the top of each meter, and an internal clip meter appears above the meter display itself. The clip indicator lights when the signal level exceeds 0 dB, and stays lit until the user clears it. Clicking a meter's clip indicator will clear that meter.

It is possible to clip internally even when input levels are relatively low. This can occur because a digital reverb is essentially a series of filters and delays. Feedback within the signal paths can cause buildup of the reverb signal, which can cause the level to increase and overload (similar to a delay line with a high level of feedback).

ReVibe Room Types

Revibe comes with over 200 built-in Room Type presets in 14 Room Type categories. These Room Type presets contain complex early reflections and room coloration characteristics that define the sound of the space. The Room Type categories and their presets are as follows:

Studios

Large Natural Studio 1
Large Natural Studio 2
Large Live Room 1
Large Live Room 2
Large Dense Studio 1
Large Dense Studio 2
Medium Natural Studio 1
Medium Natural Studio 2
Medium Natural Studio 3
Medium Natural Studio 4
Medium Live Room 1
Medium Live Room 2
Medium Dense Studio 1
Medium Dense Studio 2
Small Natural Studio 1
Small Natural Studio 2
Small Natural Studio 3
Small Natural Studio 4
Small Natural Studio 5
Small Dense Studio 1
Small Dense Studio 2
Vocal Booth 1
Vocal Booth 2
Vocal Booth 3
Vocal Booth 4

Rooms

Large Bright Room 1
Large Bright Room 2
Large Neutral Room 1
Large Neutral Room 2
Large Dark Room 1
Large Dark Room 2
Large Boomy Room
Medium Bright Room 1
Medium Bright Room 2
Medium Bright Room 3
Medium Neutral Room 1
Medium Neutral Room 2
Medium Neutral Room 3
Medium Dark Room 1
Medium Dark Room 2
Medium Dark Room 3
Small Bright Room 1
Small Bright Room 2
Small Bright Room 3
Small Neutral Room 1
Small Neutral Room 2
Small Neutral Room 3
Small Dark Room 1
Small Dark Room 2
Small Boomy Room

Halls

Large Natural Hall 1
Large Natural Hall 2
Large Natural Hall 3
Large Natural Hall 4
Large Natural Hall 5
Large Natural Hall 6
Large Dense Hall
Large Sparse Hall
Medium Natural Hall 1
Medium Natural Hall 2
Medium Natural Hall 3
Medium Natural Hall 4
Medium Dense Hall
Small Natural Hall 1
Small Natural Hall 2

Theaters

Large Theater 1
Large Theater 2
Medium Theater 1
Medium Theater 2
Small Theater 1
Small Theater 2

Churches

Large Natural Church 1
Large Natural Church 2
Large Dense Church
Large Slap Church
Medium Natural Church 1
Medium Natural Church 2
Medium Dense Church
Small Natural Church 1
Small Natural Church 2

Cathedrals

Natural Cathedral 1
Natural Cathedral 2
Natural Cathedral 3
Dense Cathedral 1
Dense Cathedral 2
Slap Cathedral

Plates

Large Natural Plate
Large Bright Plate
Large Synthetic Plate
Medium Natural Plate
Medium Bright Plate
Small Natural Plate
Small Bright Plate

Springs

Guitar Amp Spring 1
Guitar Amp Spring 2
Guitar Amp Spring 3
Guitar Amp Spring 4
Guitar Amp Spring 5
Guitar Amp Spring 6
Studio Spring 1
Studio Spring 2
Studio Spring 3
Studio Spring 4
Dense Spring 1
Dense Spring 2
Resonant Spring
Funky Spring 1
Funky Spring 2
Funky Spring 3
Funky Spring 4

Chambers

Large Chamber 1
Large Chamber 2
Large Chamber 3
Large Chamber 4
Large Chamber 5
Large Chamber 6
Medium Chamber 1
Medium Chamber 2
Medium Chamber 3
Medium Chamber 4
Medium Chamber 5
Small Chamber 1
Small Chamber 2
Small Chamber 3
Small Chamber 4

Ambience

Large Ambience 1
Large Ambience 2
Large Ambience 3
Large Ambience 4
Medium Ambience 1
Medium Ambience 2
Medium Ambience 3
Medium Ambience 4
Medium Ambience 5
Small Ambience 1
Small Ambience 2
Small Ambience 3
Very Small Ambience

Film and Post

Medium Kitchen
Small Kitchen
Bathroom 1
Bathroom 2
Bathroom 3
Bathroom 4
Bathroom 5
Shower Stall
Hallway
Closet
Classroom 1
Classroom 2
Large Concrete Room
Medium Concrete Room
Locker Room
Muffled Room
Very Small Room 1
Very Small Room 2
Very Small Room 3
Car 1
Car 2
Car 3
Car 4
Car 5
Phone Booth
Metal Garbage Can
Drain Pipe
Tin Can

Large Spaces

Parking Garage 1
Parking Garage 2
Parking Garage 3
Warehouse 1
Warehouse 2
Stairwell 1
Stairwell 2
Stairwell 3
Stairwell 4
Stairwell 5
Gymnasium
Auditorium
Indoor Arena
Stadium 1
Stadium 2
Tunnel

Vintage Digital

Large Hall Digital
Medium Hall Digital
Large Room Digital
Medium Room Digital
Small Room Digital

Effects

Mono Slapback 1
Mono Slapback 2
Mono Slapback 3
Wide Slapback 1
Wide Slapback 2
Wide Slapback 3
Multi Slapback 1
Multi Slapback 2
Multi Slapback 3
Multi Slapback 4
Spread Slapback 1
Spread Slapback 2
Mono Echo 1
Mono Echo 2
Mono Echo 3
Wide Echo 1
Wide Echo 2
Multi Echo 1
Multi Echo 2
Prism
Prism Reverse
Inverse Long
Inverse Medium
Inverse Short
Stereo Enhance 1
Stereo Enhance 2
Stereo Enhance 3

Smack!

The Smack! compressor/limiter plug-in has the following features:

- Three modes of compression:
  - Norm mode emulates FET compressors, which can have faster attack and release times than electro-optical compressors. This mode lets you fine-tune compression precisely by adjusting the attack, release, and ratio controls.
  - Warm mode is based on Norm mode, but has release characteristics more like those of electro-optical limiters.
  - Opto mode emulates classic electro-optical limiters, which tend to have gentler attack and release characteristics than FET compressors. The attack, release and ratio controls are not adjustable in this mode.

- "Key Input" side-chain processing, which lets you trigger compression using the dynamics of another signal.

- Side-Chain EQ filter, which lets you tailor the compression to be frequency-sensitive.

- High-pass filter, which lets you remove "thumps" or "pops" from your audio.

- Distortion control, which lets you add different types of subtle harmonic distortion to the output signal.

☐ *Smack! has no control to directly adjust the threshold level (the level that an input signal must exceed to trigger compression). The amount of compression will vary with the input signal, which is adjustable by the Input control.*

Smack! Plug-in (RTAS version shown)

Using the Smack! Compressor/Limiter

Smack! supports 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz sample rates. It works with mono, stereo, and greater-than-stereo multichannel formats up to 7.1.

⚠ *Sample rates of 176.4 and 192 kHz with the TDM version of Smack! require an HD Accel card, and only work with mono, stereo, and greater-than-stereo multichannel formats up to 7.0. These higher sample rates are not supported by HD Core and HD Process cards*

In general, when working with stereo and greater-than-stereo tracks, use the multichannel version of Smack!.

⚠ *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.*

The TDM version of Smack! introduces 5 samples of delay. The RTAS version of Smack! introduces 1 sample of delay. For more information, see Appendix B, "DSP Delays Incurred by TDM Plug-ins."

Smack! Parameters

Compression Mode Buttons

Smack! has three modes of compression: Norm (Normal), Opto, and Warm. Use the corresponding button to select a mode.

Norm, Warm, and Opto mode buttons

Norm Mode Button

Enable the Norm button to emulate FET compressors, which can have significantly faster attack and release times than opto-electrical-based compressors. It can be used for a wide range of program material and, with extreme settings, can be used for sound effects such as "pumping."

In Norm mode, you can precisely adjust the Ratio, Attack, and Release controls to fine-tune the compression characteristics.

💡 *Some sustained low-frequency tones can cause waveform distortion in Norm mode. The release characteristics of Warm mode (which is based on Norm mode) can be used to remedy this distortion by reducing waveform modulation.*

Warm Mode Button

Enable the Warm button for compression that is based on Norm mode, but which has program-dependent release characteristics. These characteristics, often described as "transparent" or "smooth," can be less noticeable to the listener and can reduce waveform distortion caused by some sustained low-frequency tones.

As with Norm mode, Warm mode can be used for a wide range of program material including vocals or low-frequency instruments such as tom-toms or bass guitar. Extreme settings can be used to produce "pumping" effects. Like Norm mode, Warm mode lets you precisely adjust the Ratio, Attack, and Release controls to fine-tune the compression characteristics.

Opto Mode Button

Enable the Opto button to emulate opto-electro compressors. Opto mode produces "soft knee" compression with gentle attack and release characteristics, and is ideal for compressing thin vocals, bass guitars, kick drums, and snare drums. In Opto mode, only the Input and Output controls are available for adjusting the amount of compression. The Attack, Release, and Ratio controls are greyed out and cannot be manually adjusted.

Input Control

In all Smack! compression modes, Input adjusts the level of input gain to the compressor. For more compression, increase the amount of input gain. For less compression, reduce the amount of input gain.

Setting the Input and Output controls to 5 is equal to unity gain at a compression ratio of 1:1.

Input

Attack Control

In Norm and Warm modes, Attack controls the rate at which gain is reduced after the input signal crosses the threshold.

This control is greyed out in Opto mode.

Set this control to 0 for the fastest attack time, or to 10 for the slowest attack time. Depending on the program material and the parameters used, this represents an approximate range of 100 μs to 80 milliseconds.

Attack

Ratio Control

In the Norm and Warm modes, Ratio controls 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 is greyed out in Opto mode.

Smack! has no control to directly adjust the threshold level (the level that an input signal must exceed to trigger compression). The amount of compression will vary with the input signal, which is adjustable by the Input control.

As you increase the Ratio control, Smack! goes from applying "soft-knee" compression to "hard-knee" compression, as follows:

- With soft-knee compression, gentle compression begins and increases gradually as the input signal approaches the threshold. This creates smoother compression.

- In hard-knee compression, compression begins when the input signal exceeds the threshold. This can sound abrupt, and is ideal for limiting or de-essing.

Smack! compression ratios range from subtle compression to hard limiting. At ratios of 10:1 and higher, Smack! functions as a limiter. Selecting the Smack! setting lowers the threshold slightly and applies hard limiting, which keeps the output level constant regardless of the input level. (This setting can also be used for extreme compression effects.)

Ratio

Release Control

In Norm and Warm modes, Release controls the length of time it takes for the compressor to be fully deactivated after the input signal drops below the threshold level. If the release time is too short, distortion can occur on low-frequency signals.

This control is greyed out in Opto mode.

Set this control to 0 for the fastest release time, or to 10 for the slowest release time. Depending on the program material and the parameters used, this represents an approximate range of 15 ms to 1 second for Norm mode (or the primary release of Warm mode).

Release

Output Control

In all Smack! compression modes, Output adjusts the overall output gain, which lets you compensate for heavily compressed signals by making up the resulting difference in gain.

When you apply Smack! to stereo or multi-channel tracks, the Output control determines master output levels for all channels.

Set this control to 0 for no output gain (silence), or to 10 for the loudest output gain. This represents an approximate range of +40 dB.

Setting the Input and Output controls to 5 is equal to unity gain at a compression ratio of 1:1.

Output

Side-Chain EQ Filter

The side-chain is the signal path that a compressor uses to determine the amount of gain reduction it applies to the signal being compressed. This signal path is derived from the *input signal* or *Key Input*, depending on the user's selection.

When enabled, the Side-Chain EQ filter lets the user tailor the equalization of the side-chain signal so that the compression becomes frequency-sensitive.

📄 *See "Using the Side-Chain Input in Smack!" on page 101 for more information on using the Side-Chain EQ on a Key Input.*

The Side-Chain EQ filter has the following settings:

High-Pass Makes the compressor's detector less sensitive to low frequencies in the input signal or Key Input by rolling off at a rate of 6 dB per octave. For example, you might use this setting on a mix to prevent a bass guitar or bass drum from causing too much gain reduction.

Band-Emphasis Makes the compressor's detector more sensitive to mid-to high frequencies in the input signal or Key Input by boosting those frequencies in the side-chain signal. For example, you might use this setting to reduce sibilance in vocal tracks.

Combined Enables the High-Pass and peak settings simultaneously to make the compressor's detector more sensitive to high frequencies and less sensitive to low frequencies.

Off Disables the Side-Chain EQ control.
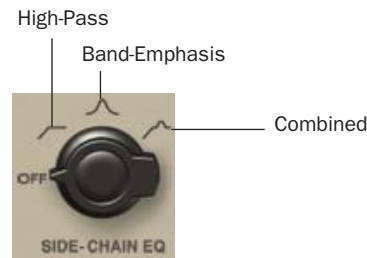
Side-Chain EQ

Distortion Control

When enabled, Distortion adds subtle second-order and third-order harmonic distortion to the output signal.

- Odd harmonics produce waveforms that are more square-shaped and are often described as "harsh" sounding.
- Even harmonics produce waveforms with more rounded edges and are often described as "smooth" sounding.

The amount of distortion that Smack! applies to the input signal depends on both the level of the input signal and the amount of compression being applied.

Odd Applies mostly odd (and some even) harmonics to the distortion.

Even Applies mostly even (and some odd) harmonics to the distortion.

O+E Applies an equal blend of odd and even harmonic distortion.

💡 *The Output control has no effect on the level of distortion applied to the signal.*

Distortion

HPF Toggle Switch

When enabled, the HPF (high-pass filter) toggle switch gently rolls off audio frequencies lower than 60 Hz in the output signal at a rate of 6 dB per octave.

This is especially useful for removing "thumps" or "pops" from vocals, bass, or kick-drums.

HPF Toggle Switch

VU Meter

The VU meter displays the amount of input level, output level, or gain reduction from compression, depending on the current Meter Mode button setting. It is calibrated to a reference level of –14 dBFS = 0 VU.
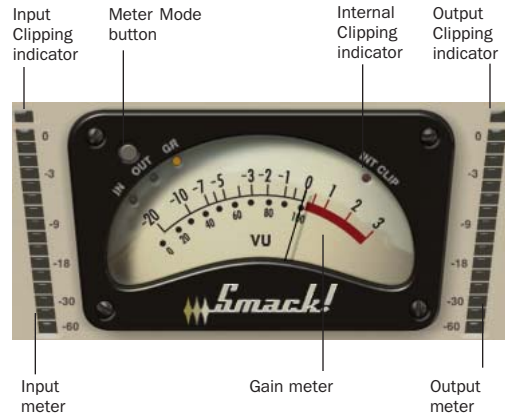
VU Meter

Meter Mode Button and Clip Indicators

The Meter Mode button toggles between displaying three display modes, as follows:

In Displays the input signal level, referenced to –14dBFS = 0 VU.

Out Displays the output signal gain, referenced to –14dBFS = 0 VU.

GR Displays the amount of gain reduction applied by the compressor.

Input and Output Meters

The Input and Output meters indicate input and output signal levels in dBFS (dB relative to full scale or maximum output).

The Internal Clipping indicator (labelled "INT CLIP") turns red when the signal exceeds the available headroom. Clicking the Internal Clipping indicator clears it. Alt-clicking (Windows) or Option-clicking (Mac) clears the clip indicators on all channels.

Using the Side-Chain Input in Smack!

Compressors typically use the detected amplitude of their input signal to cause gain reduction. This split-off signal is called the *side-chain*. However, an external signal (referred to as the *Key Input*) can be used to trigger compression.

A typical use for external side-chain processing is to control the dynamics of one audio signal using the dynamics of another signal. For example, you could use a lead vocal track to duck the level of a background vocal track so that the background vocals do not interfere with the lead vocals.

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

💡 *The Side-Chain EQ filter lets you tailor the equalization of the side-chain signal so that the compression becomes frequency-sensitive. See "Side-Chain EQ Filter" on page 99 for more information.*

To use an external Key Input to trigger compression:

1 Insert Smack! on a track you want to compress using external side-chain processing.

2 On the audio track or Auxiliary Input that you want to specify as the Key Input (the signal that will be used to trigger compression), click the Send button and select the bus path to the track that will use side-chain processing.
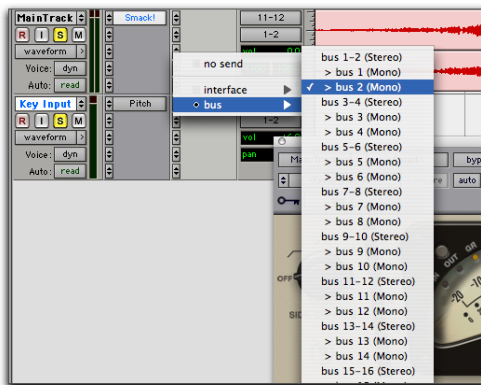
Selecting a Key Input from the Send button

💡 *The Key Input must be monophonic.*

3 In the track that you are compressing, click the instance of Smack! in the Inserts pop-up menu.

4 In the Smack! plug-in window, click the Key Input menu, and select the input or bus path that you have designated as the Key Input.
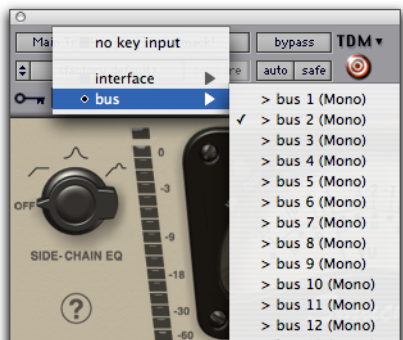
Selecting the input/bus path in Smack!

5 Begin playback. Smack! uses the input or bus that you selected as a Key Input to trigger its effect.

6 To fine-tune the amount of compression, adjust the send level from the Key Input track.

⚠ *When you are using a Key Input to trigger compression, the Input control has no effect on the amount of compression.*

7 To tailor the side-chain signal so that the detector is frequency-sensitive, use the Side-Chain EQ filter (see "Side-Chain EQ Filter" on page 99 for more information).

8 Adjust other parameters to achieve the desired effect.

chapter 12

SoundReplacer

SoundReplacer is an AudioSuite plug-in designed to replace audio elements such as drums, percussion, and sound effects in Pro Tools tracks with alternate sounds. SoundReplacer can quickly and intelligently match the timing and dynamics of original performance material, making it ideal for both music and audio post production.

SoundReplacer features:

- Sound replacement with phase-accurate peak alignment
- Intelligent tracking of source audio dynamics for matching the feel of the original performance
- Three separate amplitude zones per audio event for triggering different replacement samples according to performance dynamics
- Zoomable waveform display for precision threshold/amplitude zone adjustment
- Crossfading or hard-switching of replacement audio in different amplitude zones for optimum realism and flexibility
- Online help

Audio Replacement Techniques

Replacing audio elements during the course of a recording session is a fairly common scenario. In music production it is often done in order to replace or augment an element that lacks punch. In film or video post-production it is typically done to improve or vary a specific sound cue or effect.

In the past, engineers and producers had to rely on sampling audio delay lines or MIDI triggered audio samplers—methods that had distinct disadvantages. Delay lines, for example, support only a single replacement sample, and while they can track the amplitude of the source events, the replacement sample itself remains the same at different amplitude levels.

The result is static and unnatural. In addition to these drawbacks, sample triggers are notoriously difficult to set up for accurate timing.

Similarly, with MIDI triggered samplers, MIDI timing and event triggering are inconsistent, resulting in problems with phase and frequency response when the original audio is mixed with the triggered replacement sounds.

The SoundReplacer Solution

SoundReplacer solves these timing problems by matching the original timing and dynamics of the source audio while providing three separate amplitude zones per audio event. This lets you trigger different replacement samples according to performance dynamics.

Each replacement sample is assigned its own adjustable amplitude zone. Variations in amplitude within the performance determine which sample is triggered at a specific time. For example, you could assign a soft snare hit to a low trigger threshold, a standard snare to a medium trigger threshold, and a rim shot snare to trigger only at the highest trigger threshold.

Replacement samples that are triggered in rapid succession or in close proximity to each other will overlap naturally—avoiding the abrupt sound truncation that occurs on many samplers.

In addition to its usefulness in music projects, SoundReplacer is also an extremely powerful tool for sound design and post production. Morphing gun shots, changing door slams, or adding a Doppler effect can now be accomplished in seconds rather than minutes—with sample-level precision.

Replacement audio events can be written to a new audio track, or mixed and re-written to the source audio track. Sample thresholds can be amplitude-switched between the replacement samples, or amplitude crossfaded for seamless transitions.

SoundReplacer Controls

SoundReplacer

Waveform Display
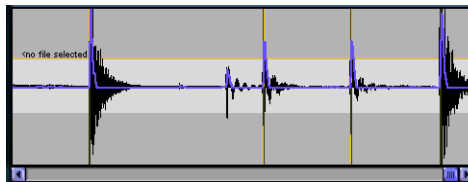
Waveform display with trigger markers shown

The waveform display shows the audio that you have selected for replacement. When you select audio on the source track, then open Sound-Replacer, the audio waveform will automatically be displayed here.

Once the audio selection is displayed, you can load the desired replacement samples and adjust their trigger thresholds while viewing the waveform peaks. Trigger markers then appear in the waveform, indicating the points at which the samples will be triggered.

The color of each marker indicates which threshold/replacement sample will be triggered. The blue Trigger Envelope shows the waveform slope that determines the trigger points. The Zoomer lets you increase or decrease waveform magnification here to help accurately set trigger thresholds.

If you change the audio selection on the source track, click Update to update the waveform display. If Auto Update is selected, SoundReplacer automatically updates the waveform display each time you make a new selection or begin playback.

If you frequently change selections or start and stop playback, turn off Auto Update to prevent too-frequent redraws.

Trigger Threshold

Threshold controls

The color-coded Trigger Threshold sliders set a total of three amplitude zones (one for each replacement audio file) for triggering replacement samples:

• The yellow slider represents amplitude zone 1, the lowest-level trigger.

• The red slider represents amplitude zone 2, the middle-level trigger.

• The blue slider represents amplitude zone 3, the highest-level trigger.

With a replacement sample loaded, drag the Threshold slider to the desired amplitude level. Color-coded trigger markers will appear in the Waveform at points where the source audio signal exceeds the threshold set for that amplitude zone. The replacement sample will be triggered at these points.

The color of the Trigger markers correspond to the matching Threshold slider. This lets you see at a glance which replacement samples will be triggered and where they will be triggered.

If you zoom the waveform display below a specific Trigger Threshold slider's amplitude zone, the slider will be temporarily unavailable. To access the slider again, zoom back out to an appropriate magnification level.

Load/Unload Sound

Load/Unload Sound

Clicking the Load/Unload Sound icons loads or unloads replacement samples for each of the three trigger threshold amplitude zones. Clicking the Floppy Disk icon loads a new sample (or replaces the current sample). Clicking the Trash Can icon unloads the current sample.

⚠ *SoundReplacer does not perform a sample rate conversion before loading replacement samples if they are at a different sample rate from the session. Replacement samples should be at the same sample rate as the session, otherwise they will playback at the wrong speed and pitch.*

To audition a replacement sample before loading it into SoundReplacer, use the Import Audio command in Pro Tools. Once you have located and previewed the desired audio file, you can then load it into SoundReplacer using the Load/Unload Sound icons.

⚠ *SoundReplacer does not load regions that are part of larger audio files. To use a region as a replacement sample, you must first save it as an individual audio file.*

The Zoomer

Zoomer

The Zoomer increases or decreases magnification of the waveform data currently visible in the center of the waveform display so that you can more accurately set sample trigger thresholds.

- To zoom in on amplitude, click the Up Arrow.

- To zoom out on amplitude, click the Down Arrow.

- To zoom in on time, click the Right Arrow.

- To zoom out on time, click the Left Arrow.

⚠ *If you zoom the waveform display below a specific Threshold slider's amplitude zone, the slider will be temporarily unavailable. To access the slider again, zoom back out to an appropriate magnification level.*

Crossfade

When Crossfade is selected, SoundReplacer crossfades between replacement audio files in different amplitude zones. This helps smooth the transition between them.

When Crossfade is deselected, SoundReplacer hard switches between replacement audio files in different amplitude zones.

Crossfading is particularly useful for adding a sense of realism to drum replacement. Crossfading between a straight snare hit and a rim shot, for example, results in a much more "live" feel than simply hard switching between the two samples.

Peak Align

When Peak Align is on, SoundReplacer aligns the peak of the replacement file with the peak of the source file in a way that best maintains phase coherency. When Peak Align is off, SoundReplacer aligns the beginning of the replacement file with the trigger threshold point.

Depending on the characteristics of your source and replacement audio files, using Peak Align can significantly affect the timing of audio events in the replacement file. It is essential that you choose the option most appropriate to the material that you are replacing.

📖 *For more information on using Peak Align, see "Getting Optimum Results with SoundReplacer" on page 108.*

Update

When you click Update, the waveform display is redrawn, based on the audio currently selected on the source track. Each time you make a new selection on a source track, you must click Update for SoundReplacer to draw the waveform of the selection.

Auto Update

When Auto Update is selected, SoundReplacer automatically updates the waveform display each time you make a new selection on a source track. If you frequently change selections or start and stop playback, you may want to deselect Auto Update to prevent frequent redraws.

Mix

Mix adjusts the mix of the replacement audio file with the original source file. Higher percentage values weight the mix toward the replacement audio. Lower percentage values weight the mix toward the original source audio.

The Mix button toggles the Mix control on and off. When Mix is toggled off, the balance is instantly set to 100% replacement audio.

Setting Mix to 50% and clicking Preview lets you audition source audio and replacement audio together to check the accuracy of replacement triggering timing.

Dynamics

Dynamics controls how closely the audio events in the replacement file track the dynamics of the source file:

◆ Setting the ratio to 1.00 matches the dynamics of the source file.

◆ Increasing the ratio above 1.00 expands the dynamic range so that softer hits are softer, and louder hits are louder. This is useful if the source material lacks variation in its dynamic range.

◆ Decreasing the ratio below 1.00 compresses the dynamic range so that there is less variation between loud and soft hits. This is useful if the dynamics of the source material are too extreme.

The Dynamics button provides a quick means of toggling on and off the Dynamics control. When Dynamics is toggled off, SoundReplacer will not track changes in the source audio file's dynamics. Audio events in the resulting replacement audio file will uniformly be at the amplitude of the replacement samples themselves, with no variation in dynamics.

Online Help

Online help

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

Using SoundReplacer

Following are basic guidelines for using SoundReplacer effectively. Also see "Getting Optimum Results with SoundReplacer" on page 108.

To use SoundReplacer:

1 On the source track, select the audio you want to replace. Only selected audio will be replaced.

2 Choose SoundReplacer from the AudioSuite menu.

3 Click the Load Sound icon (the icon beneath the yellow slider) to import the replacement sound for amplitude zone 1.

4 Locate the desired audio file and click Open.

5 Adjust the amplitude zone slider.

6 Repeat steps 3–5 to load replacement sounds into amplitude zones 2 and 3.

If you use only a single replacement sample, you should still set all three amplitude zones for optimum results. This will ensure accurate triggering. For details, See "Mapping The Same Sample Into Multiple Amplitude Zones" on page 109.

7 To align the amplitude peak in the replacement file(s) to threshold trigger markers in the source audio, enable Peak Align.

8 Click Preview to audition the replacement audio.

9 Adjust the Threshold sliders to fine tune audio replacement triggering.

10 Adjust the Dynamics slider to fine tune how SoundReplacer tracks and matches changes in the source audio's dynamics.

11 Adjust the Mix slider to get the desired balance between replacement audio and source audio.

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

◆ Decide where the selected region should be processed:

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

  – or –

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

◆ Decide if you want to update every occurrence of the selection region:

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

  – or –

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

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

Because SoundReplacer does not allow destructive processing, the AudioSuite Overwrite Files option is not available.

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

14 Click the Process button.

Getting Optimum Results with SoundReplacer

Getting optimum results with SoundReplacer generally means making sure that the audio events in the replacement audio file have accurate timing in relation to the source audio. The techniques given here help ensure this.

Using Peak Align

Proper use of the Peak Align feature can significantly improve the results of sound replacement. Since turning Peak Align on or off controls how SoundReplacer aligns the replacement audio with the source audio, it will significantly affect the timing of audio events in the replacement file.

In general:

◆ Turn on Peak Align if you are replacing drum or percussion sounds whose peak level occurs at the initial attack.

◆ Turn off Peak Align if you are replacing sounds whose peak level occurs somewhere *after* the initial attack. Peak Align should also be turned off if the sounds you are replacing are *not* drum or percussion sounds.

To illustrate why Peak Align makes a difference, look at the following illustrations:
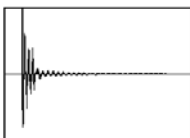
Figure 1. A fast-peaking kick drum
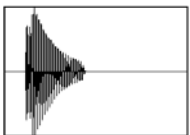
Figure 2. A slower-peaking kick drum

Figure 1 shows a kick drum whose peak level occurs at its initial attack.

Figure 2 shows a kick drum whose peak level occurs *after* its initial attack.

If you turn on Peak Align and attempt to replace the fast-peaking kick with the slow-peaking kick (or vice-versa), SoundReplacer will align their peaks—which occur at different points in the sound. The audible result would be that the replacement audio file (slow-peaking kick) would trigger too early.

Mapping The Same Sample Into Multiple Amplitude Zones

If you are performing drum replacement and intend to use just a single replacement sample, mapping it into multiple amplitude zones will ensure more accurate triggering. Here is why:

Imagine that you are replacing a kick drum part. If you look at the waveform of a kick drum, you will often see a "pre-hit" portion of the sound that occurs as soon as the ball of the kick pedal hits the drum. This is rapidly followed by the denser attack portion of the sound, where most of sound's weight is.
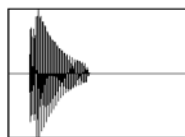
A kick drum with a pre-hit preceding a denser attack

With a sound like this, using a single amplitude threshold presents a problem because typically, in pop music, kick drum parts consist of loud accent hits and softer off-beat hits that are often 6 dB or more lower in level.

If you use a single amplitude threshold to trigger the replacement sample, you have to set the threshold low enough to trigger at the soft hits. The problem occurs at the loud hits: The threshold is now set so low that the pre-hit portion of the loud hits can exceed the threshold—triggering the replacement sample too early. This results in a replacement track with faulty timing.
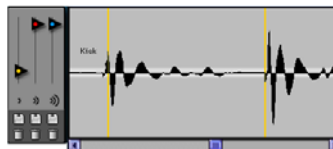
A single low threshold causes the second, louder kick to trigger too early, as evidenced by the trigger marker at the very start of the waveform.

The best way to avoid this problem is to set multiple threshold zones for the same sample using a higher threshold for the louder hit. Soft hits will trigger threshold 1 and louder hits will trigger threshold 2.
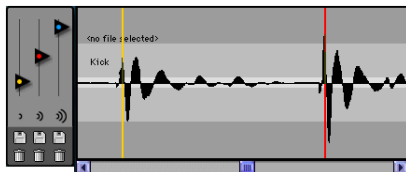
Using a second, higher threshold for the louder kick will make it trigger properly, as shown by the now properly-aligned trigger marker.

To set the precise threshold for louder hits, you may need to zoom in carefully to examine the waveform for trigger points (indicated by color-coded trigger markers) and then Command-drag the Threshold slider for more precise adjustment.

If there is a great deal of variation in the dynamics of the source audio, you may need to use all three Trigger Thresholds/Amplitude Zones for optimum results.

☞ *If only one replacement sample is loaded into SoundReplacer and it is loaded into Trigger threshold/amplitude zone 1 (yellow), SoundReplacer will let you use the red and blue Trigger Threshold sliders to set Amplitude Zones 2 and 3—without having to load the same sample again.*

Using the Audio Files Folder for Frequently Used Replacement Files

If you often use the same settings and replacement sounds in different sessions, Sound-Replacer provides a convenient way to keep the replacement audio files and settings linked together.

When you choose a setting from the Librarian menu, SoundReplacer looks for the replacement audio files associated with the setting. Sound-Replacer first looks in the audio file's original hard disk location (at the time you saved the setting).

If it is not there, SoundReplacer looks in a folder named Audio Files within SoundReplacer's Root Plug-in Settings folder (Plug-in Settings/SoundReplacer/Audio Files).

If SoundReplacer finds the replacement audio file there, the Settings file will load with the associated audio.

By always putting replacement audio files in this special folder, you can freely exchange Sound-Replacer settings—and the audio files associated with them—with other users.

⚠ *Do not create subfolders within Sound-Replacer's Audio Files folder. Files located within subfolders are not recognized.*

X-Form Plug-in Parameters

X-Form

The X-Form AudioSuite plug-in provides the highest quality time compression and expansion algorithms for music production, sound design, and audio loop applications. Use it to manipulate audio loops for tempo matching or to change vocal tracks for formant correct pitch shifting. The X-Form plug-in is useful in audio post-production for adjusting audio to specific time or SMPTE durations for synchronization purposes. X-Form is also ideal for post-production pull up and pull down conversions.

Normalizing a selection before using X-Form may produce better results.

X-Form Displays and Controls

X-Form plug-in

The interface for X-Form is organized in four parts: Audio, Time, Transient, and Pitch.

Audio Use the controls in the Audio section to select the most appropriate time compression and expansion algorithm 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.

Transient Use the controls in the Transient section to adjust the transient detection parameters for high quality time compression or expansion.

Pitch Use the controls in the Pitch section to apply pitch shifting. Pitch shifting can be formant correct with either the Polyphonic or Monophonic algorithm.

Audio

The Audio section of X-Form provides controls for specifying the type of audio you want to process and gain attenuation of the processed signal to avoid clipping.

X-Form plug-in, Audio section

Type

The Audio Type determines the type of TCE and pitch shift algorithm for processing audio: Polyphonic, Monophonic, or Poly (Faster).

Polyphonic Use for processing complex sounds (such as a multipart musical selection).

⚠ *When previewing Polyphonic, Poly (Faster) is used for faster previewing. However, when you process the audio selection, the high-quality Polyphonic setting is used.*

Monophonic Use for processing monophonic sounds (such as a vocal melody).

Poly (Faster) Use for faster previewing and processing, but with slightly reduced audio quality.

Gain

The 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 shifts 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 X-Form 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.

X-Form 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.

⚠️ *X-Form 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."*

Shift Displays the target time compression or expansion as a percentage of the original. Adjust the Time control or click the Shift field and type the desired value. Time can be shifted by as much as 12.50% to 800.00% of the original speed (or 8 times to 1/8 of the original duration) depending on which Range button is enabled (2x, 4x, or 8x). The default setting is 100%, or no time shift.

The Shift 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 post-production pull up and pull down factors (see "Post Production Pull Up and Pull Down Tasks" on page 115).

2x, 4x, and 8x Range Buttons

The 2x, 4x, and 8x Range buttons set the possible range for the Time Shift, Pitch Shift, and Formant Shift controls.

2x Lets you apply Time Shift, Pitch Shift, and Formant Shift from 50.00% to 200.00% (where 50.00% is 2 times the original duration and 200.00% is 1/2 of the original duration).

4x Lets you apply Time Shift, Pitch Shift, and Formant Shift from 25.00% to 400.00% (where 25.00% is 4 times the original duration and 400.00% is 1/4 of the original duration).

8x Lets you apply Time Shift, Pitch Shift, and Formant Shift from 12.50% to 800.00% (where 12.50% is 8 times the original duration and 800.00% is 1/8 of the original duration).

⚠️ *When changing to a smaller Range setting (such as switching from 8x to 2x), the Time Shift and Pitch Shift settings are constrained to the limits of the new, smaller range. For example, with 8x enabled and Time Shift set to 500%, switching to 2x changes the Time Shift value to 200%.*

Transient

The Transient section provides controls for setting the sensitivity for transient detection and for adjusting the analysis window size.

X-Form plug-in, Transient section

Sensitivity Controls how X-Form determines and interprets transients from the original audio. Part of X-Form's processing relies upon separating "transient" parts of the sample 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. Sensitivity is only available when Polyphonic is selected as the Audio Type.

For highly percussive material, lower the Sensitivity for better transient detection, especially with the Rhythmic audio setting. For less percussive material, 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 size. You can adjust the Window from 10.0 milliseconds to 100.0 milliseconds. Adjust the Window control or click the Window field and type the desired value. Window is only available when Monophonic is selected as the Audio Type.

Try larger window sizes for low frequency sounds or sounds that do not have many transients. Try smaller window sizes for tuned drums and percussion. However, the default of 25 milliseconds should work well for most material.

Pitch

The Pitch section provides controls for pitch shifting the selected audio. Use the Pitch control to transpose the pitch from as much as –36.00 semitones (–3 octaves) to +36.00 semitones (+3 octaves), with fine resolution in cents, depending on which Range button is enabled (2x, 4x, or 8x). X-Form also lets you transpose the formant shape independently of the fundamental frequency.

X-Form plug-in, Pitch section

Transpose Displays the transposition amount in semitones. You can transpose pitch by as much as –36.00 semitones (–3 octaves) to +24.00 semitones (+3 octaves), with fine resolution in cents, depending on which Range button is enabled. Adjust the Pitch control or click the Transpose field and type the desired value. The default value is 0.00 semitones, or no pitch shift.

Shift Displays the pitch shift amount as a percentage. You can shift pitch by as much as 12.50% (–3 octaves) to 800.00% (+3 octaves) depending on which Range button is enabled (2x, 4x, or 8x). Adjust the Pitch control or click the Shift field and type the desired value. The default value is 100%, or no pitch shift.

Formant

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 regard-

less 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.

The Pitch section of X-Form lets you pitch shift the formants 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.

To enable or disable formant shifting:

■ Click the In button. The In button lights when formant shifting is enabled.

The Formant field displays the amount of formant pitch shifting from –36.00 semitones (–3 octaves) to +36.00 semitones (+3 octaves), with fine resolution in cents. Adjust the Formant control or click the Formant field and type the desired value. The default value is 0.00 semitones, or no formant shift.

Post Production Pull Up and Pull Down Tasks

Table 14 on page 115 provides information on TCE settings for common post-production tasks. Type the corresponding TCE% (represented to 10 decimal places in Table 14) in the Time Shift field for the corresponding post-production task and the process the selected audio.

🔅 *Use the corresponding X-Form Plug-in Setting for the desired post-production task.*

Table 14. 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

X-Form 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

X-Form 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 X-Form in Pro Tools 7.0 and 7.1.*

AudioSuite TCE Plug-in Preference

The X-Form plug-in's high quality time compression and expansion algorithms that can be used with the Pro Tools TCE Trim tool.
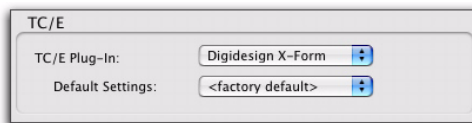
TCE Plug-in option in Processing Preferences page

⚠ *X-Form is not available with the TCE Trim tool in Pro Tools 7.1.x and lower.*

⚠ *When using X-Form for the TCE Trim tool, the default 2x Range is used for an edit range of twice to half the duration of the original audio. If you select a Default Setting that uses either the 4x or 8x Range, the Time Shift and Pitch Shift setting are constrained to the 2x Range limit of 50% to 200%.*

📖 *Refer to the Pro Tools Reference Guide for more information about the TCE Trim tool.*

To select X-Form 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 Digidesign X-Form.

4 Select the desired preset setting from the Default Settings pop-up menu.

5 Click OK.

Processing Audio

X-Form lets you change the time and pitch of selected audio independently or concurrently.

☼ *You can adjust both the Time Shift and Pitch Shift controls independently before processing.*

To change the time of a selected audio region:

1 Select AudioSuite > Pitch Shift > X-Form.

2 Select the Audio Type appropriate to the type of material you are processing (Monophonic or Polyphonic).

3 If compressing the duration of the selection, attenuate the Gain control as necessary.

4 Adjust the Transient controls as desired.

5 Enable the desired Range button (2x, 4x, or 8x) to set the possible range for time change.

6 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 speed.

7 Click Process.

To change the pitch of a selected audio region:

1 Select AudioSuite > Pitch Shift > X-Form.

2 Select the Audio Type appropriate to the type of material you are processing (Monophonic or Polyphonic).

3 If transposing the pitch of the selection up, attenuate the Gain control as necessary.

4 Adjust the Transient controls as desired.

5 Enable the desired Range button (2x, 4x, or 8x) to set the possible range for pitch change.

6 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 pitch.

7 If desired, enable Formant and adjust the Formant control.

8 Click Process.

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 number of instances of each plug-in that can be powered by a single DSP chip on Pro Tools|HD cards. DSP usage differs according to card type.

⚠ *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.

Using Multi-Mono Plug-ins on Greater-Than-Stereo Tracks

Plug-ins used in multi-mono format on greater-than-stereo tracks require one mono instance per channel of the multi-channel audio format. For example, a multi-mono plug-in used on a 5.1 format track, requires six mono instances since there are six audio channels in the 5.1 format.

DSP Requirements

Digidesign TDM plug-ins have the following DSP requirements:

HD Accel Card

Mono and Stereo

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

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz		DSP Chips per HD Accel Card
Plug-in	Mono	Stereo	Mono	Stereo	Mono	Stereo	
D-Fi (Lo-Fi)	16	8	8	4	3	n/a	9
D-Fi (Recti-Fi)	28	14	13	6	6	3	9
D-Fi (Sci-Fi)	18	9	7	4	4	n/a	9
DINR (BNR)	1	1	1	1	n/a	n/a	9
Maxim	20	10	9	4	4	n/a	9
Bruno/Reso	1	1	1	1	n/a	n/a	6
Reverb One	2	2	1	1	1	1	9
Impact	25	19	12	9	5	4	7
ReVibe	2	2	1	1	n/a	n/a	4
Smack!	7	5	3	3	2	1	9

Multichannel

Table 3. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Accel card at 48 kHz

Plug-in	Quad & LCRS	5.1 & 6.0	DSP Chips per HD Accel Card
D-Fi (Lo-Fi)	4	n/a	9
D-Fi (Recti-Fi)	7	n/a	9
D-Fi (Sci-Fi)	4	n/a	9
DINR (BNR)	partial	n/a	9
Maxim	2	1	9
Bruno/Reso	partial	partial	6
Reverb One	partial	partial	9
Impact	14	11	7
ReVibe	2	2	4
Smack!	4	3	9

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

HD Core and HD Process

Mono and Stereo

Table 4. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Core or HD Process card, at different sample rates (mono and stereo).

Sample Rate:	44.1/48 kHz		88.2/96 kHz		174.6/192 kHz		DSP Chips per HD Core or HD Process Card
Plug-in	**Mono**	**Stereo**	**Mono**	**Stereo**	**Mono**	**Stereo**	
D-Fi (Lo-Fi)	7	3	3	1	1	n/a	9
D-Fi (Recti-Fi)	12	6	5	2	2	1	9
D-Fi (Sci-Fi)	7	3	3	1	1	n/a	9
DINR (BNR)	1	1	1	1	n/a	n/a	9
Maxim	8	4	4	2	1	n/a	9
Bruno/Reso	1	1	1	1	n/a	n/a	6
Reverb One	1	1	1	1	1	1	9
Smack!	3	2	1	1	n/a	n/a	9

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

For maximum instance counts for plug-ins that are compatible with the most powerful chips on HD Accel, see "HD Accel Card" on page 120.

Multichannel

Table 5. Maximum instances of real-time TDM plug-ins per DSP chip for an HD Core or HD Process card at 48 kHz

Plug-in	Quad & LCRS	5.1 & 6.0	DSP Chips per HD Core or HD Process Card
D-Fi (Lo-Fi)	1	n/a	9
D-Fi (Recti-Fi)	3	n/a	9
D-Fi (Sci-Fi)	1	n/a	9
DINR (BNR)	partial	partial	9
Maxim	2	1	9
Bruno/Reso	partial	partial	6
Reverb One	partial	partial	9
Smack!	1	1	9

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

Monitoring DSP Usage

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

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

appendix b

DSP Delays Incurred by TDM Plug-ins

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

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

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

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

The following figure shows the delays inherent in each type of Digidesign TDM plug-in:

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

Plug-in	Samples of Delay on Pro Tools|HD Cards
Bruno	3
Lo-Fi	4
Recti-Fi	4
Sci-Fi	4
DINR (BNR)	1538/3074/not supported*
Maxim (TDM and RTAS)	1028/2052/4100*
Reso	3
Impact**	4
ReVibe**	2
Smack! (TDM and RTAS)	5/1

* BNR and Maxim have different delays at different sample rates: 48 kHz/96 kHz/192 kHz.

**Requires an HD Accel card.

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