

# **Digidesign Legacy Plug-ins**

Version 7.3



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# Introduction

### **Digidesign Legacy Plug-ins**

Digidesign legacy plug-ins are plug-ins that are no longer supported with current Pro Tools systems. They are included only for compatibility with sessions created in older versions of Pro Tools, but have not been qualified with the latest version of Pro Tools. Whenever possible, use only the latest versions of any Digidesign plug-ins.

Digidesign legacy plug-ins covered in this guide include:

- Dyn II
- EQ II
- · Mod Delay I
- SurrroundScope



See the DigiRack Plug-ins Guide and the Digidesign Plug-in Guide for more information on plug-ins.

# **System Requirements**

To use Digidesign Legacy plug-ins you need any of the following:

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

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

#### **Compatibility Information**

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

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

### **Conventions Used in This** Guide

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

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

The following symbols are used to highlight important information:



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



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



Shortcuts show you useful keyboard or mouse shortcuts.



Cross References point to related sections in this guide and other Digidesign guides.

### **About www.digidesign.com**

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

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

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

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

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

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

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

# Installation

# **Installing Legacy Plug-ins**

- When you install Pro Tools, EQ III and Dynamics III plug-ins are installed. The following legacy plug-ins are installed in the Plug-ins (Unused) folder:
  - EO II
  - Dynamics II



For more information about installing Pro Tools, see the Getting Started Guide that came with your system.

- ◆ The Mod Delay I plug-in is not installed with Pro Tools, but is available on the Pro Tools installer disc in the Additional Files\Unsupported Software\Plug-ins folder.
- ◆ When you install Pro Tools, Surround Tools plug-ins are installed. If the SurroundScope plug-in was previously installed, it is moved to the Plug-ins (Unused) folder.

#### To make a legacy plug-in available for use:

- 1 If Pro Tools is currently running, quit Pro Tools.
- **2** Locate the plug-in in the Plug-ins (Unused) folder (on your Startup drive) or in the Unsupported Software\Plug-ins folder (on the Pro Tools Installer disc).
- **3** Drag the plug-in into the Plug-ins folder.



放 Legacy plug-ins should only be used when opening older Pro Tools sessions. You should use the current versions of plug-ins whenever possible.

### **Updating Pro Tools**

If you are updating from an lower version of Pro Tools, the Pro Tools installer checks for any legacy plug-ins currently installed on your system. If the installer detects unsupported plugins in the Pro Tools Plug-ins folder, it notifies you and recommends that you use the current version of the plug-in instead.

Any detected legacy plug-ins are moved from the Plug-ins folder to the Plug-ins (Unused) folder.

### **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.

# **Dyn II Plug-ins**

The Dynamics II plug-in provides a compressor, limiter, expander, gate and de-esser to control the dynamics of an audio selection. It is available in TDM, RTAS, and AudioSuite™ versions.

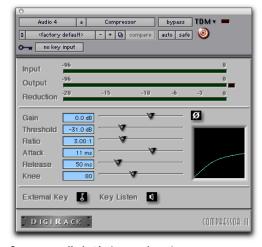


See the DigiRack Plug-ins Guide for more information on DigiRack plug-ins.

### **Dyn II TDM and RTAS Versions**

The Dyn II modules include the Compressor, Limiter, Expander, Gate, and De-Esser.

### Compressor II



Compressor II plug-in (mono shown)

#### **Compressor Controls**

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

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

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

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

Reduction Indicates the amount of gain reduction in dB.

**Threshold** Sets the threshold level. Signals that exceed this level will be compressed. Signals that are below it will be unaffected. A level setting of 0 dB is equivalent to no compression. Unlike scales on analog compressors, metering scales on a digital device reflect a 0 dB value that indicates full scale (fs)—the full-code signal level. There is no headroom above 0 dB.

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

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

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

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

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

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

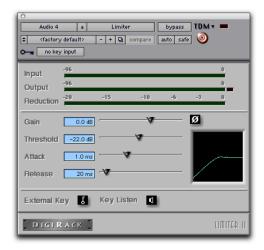
External Key Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger compression.

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



**A** Key Listen is not saved with the other plugin settings.

#### Limiter II



Limiter II plug-in

#### **Limiter Controls**

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

**Gain** Provides overall output gain adjustment.

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

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

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

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

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

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

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

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

External Key Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger limiting.

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



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

#### Gate II



Gate II plug-in

#### **Gate Controls**

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

**Gating** Indicates the amount of reduction in dB.

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

**Attack** Sets the attack time of the Gate.

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

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

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

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

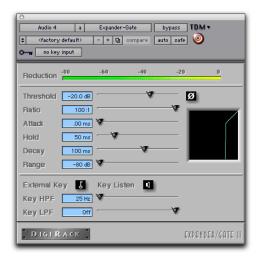
External Key Enables and disables side-chain processing. Side-chain processing uses the varying amplitude of a separate reference track to trigger limiting.

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



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

#### Expander/Gate II



Expander/Gate II plug-in

#### **Expander/Gate Controls**

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

**Reduction Meter** Indicates the amount of signal reduction in dB.

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

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

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

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

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

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

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

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

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

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

External Key Enables and disables side-chain processing. Side-chain processing allows the Expander/Gate to listen to the varying amplitude of a separate reference track and use it to trigger expansion.

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



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

#### DeEsser



DeEsser plug-in



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

#### **DeEsser Controls**

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

**Output Meter** Indicates the output level of the DeEsser.

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

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

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

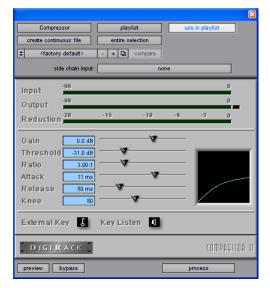
**Key Listen** Monitors the sibilant peaks used by the DeEsser as a Key Input to trigger compression. This is useful for listening only to the sibilance and fine tuning settings to remove them.

# **Dynamics II AudioSuite**

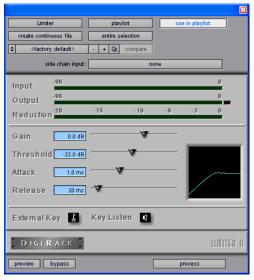
The features of these plug-ins are non-real-time, but otherwise identical to their real-time counterparts.



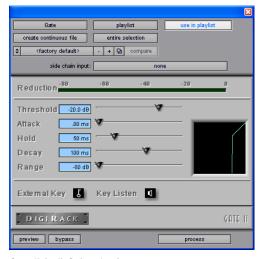
For information on the individual controls for these plug-ins, see "Dyn II TDM and RTAS Versions" on page 5.



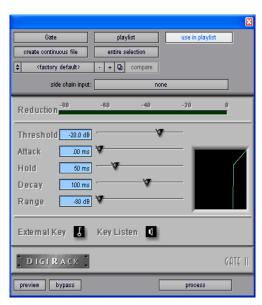
Compressor II AudioSuite plug-in



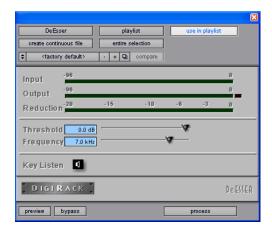
Limiter II AudioSuite plug-in



Gate II AudioSuite plug-in



Expander/Gate II AudioSuite plug-in



DeEsser AudioSuite plug-in

# **EQ II Plug-ins**

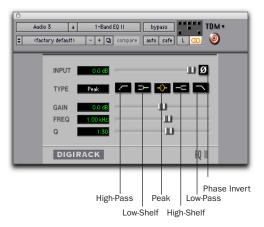
The EQ II plug-in provides a 1-Band or 4-Band EQ for adjusting the frequency spectrum of an audio selection. It is available in TDM, RTAS, and AudioSuite versions.



See the DigiRack Plug-ins Guide for more information on DigiRack plug-ins.

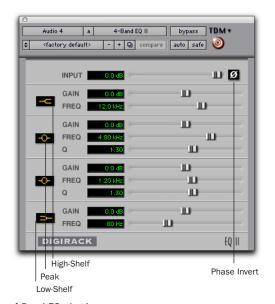
### **EQ II TDM and RTAS Versions**

You can configure the 1-Band EQ as a high-pass, low-shelf, high-shelf, low-pass, or peak EQ.



1-Band EQ plug-in

The 4-Band EQ provides one high-shelf, one low-shelf, and two peak EQs.



4-Band EQ plug-in

# **Adjusting EQ**

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

#### **EQ II Controls**

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

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



Phase Invert

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

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

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

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

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



Bypass EQ band button on the 4-band EQ

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

available for this filter. High-Pass filters can be useful for removing low frequency rumble or thinning out the lower end of a sound for special effects such as simulating a telephone effect.

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

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

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

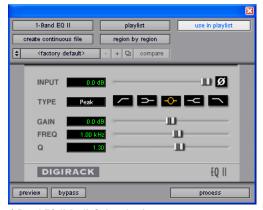
**Low-Pass filter** Attenuates all frequencies above the selected cutoff frequency setting at a rate of 12 dB per octave while letting all others below pass through. For this reason, no gain control is available for this filter.

# **EQ II AudioSuite Versions**

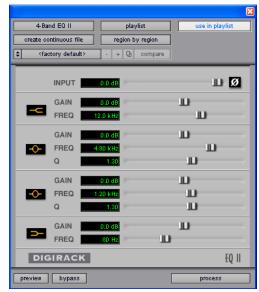
The features of these plug-ins are non-real-time, but otherwise identical to their real-time counterparts.



For information on the individual controls for these plug-ins, see "EQ II Controls" on page 14.



1-Band EQ II AudioSuite plug-in



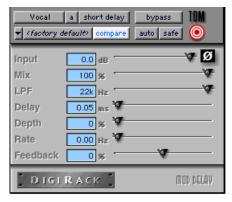
4-Band EQ II AudioSuite plug-in

# **Mod Delay I Plug-ins**

### **Mod Delay**

(Pro Tools 5.3 and Lower Only)

The DigiRack Mod Delay plug-ins provide time delay-based effects. Effects obtained through the use of Mod Delay include echo, slap echo, doubling, chorusing, and flanging. It is available in TDM, RTAS, and AudioSuite versions.



Mod Delay plug-in (Pro Tools 5.3)

There are four different Mod Delays, capable of different maximum delay times:

- The Short Delay provides 1024 samples of delay (23.2 milliseconds at 44.1 kHz or 21.3 ms at 48 kHz).
- The Slap Delay provides 7186 samples of delay (162 ms at 44.1 kHz or 149 ms at 48 kHz).
- The Medium Delay provides 16384 samples of delay (371 ms at 44.1 kHz kHz or 341 ms at 48 kHz).
- The Long Delay provides 162474 samples of delay (3.68 seconds at 44.1 kHz or 3.38 seconds at 48 kHz).
- ▲ The Long Delay TDM plug-in is available only on HD-series or MIX-series systems.

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

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

The Mod Delays provide a Mono In/Stereo Out option. Choosing a delay in the Mono In/Stereo Out format "stereoizes" the output of a mono track. A track created in this way will then have

stereo pan faders for controlling each channel of the stereo signal. As explained previously, any inserts that occur on a track after a mono-to-stereo insert will automatically become stereo as well.

#### **Mod Delay Controls**

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

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



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

LPF (Low-Pass Filter) Controls the cutoff frequency of the low-pass filter. Use this to attenuate the high frequency content of the feedback signal. The lower the setting, the more high frequencies are attenuated.

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

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

Rate Controls the rate of modulation of the delayed signal.

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

#### **Automation**

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

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

# SurroundScope

SurroundScope assists with multi-channel mixing by visually representing the phase, individual input levels, and position of multi-channel signals in the surround field.

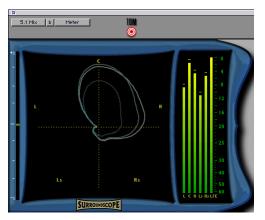
SurroundScope supports all industry-standard, multi-channel formats, including LCR, Quad, LCRS, and 5.0, 5.1, 6.0, 6.1, 7.0, and 7.1.

### **SurroundScope Features**

- 360° Surround Display indicates signal position within the surround field.
- Lissajous meter ensures perfect stereo images by indicating amplitude and phase.
- · High-resolution meters accurately display individual input channel levels as low as -60 dB.
- Stereo phase meter indicates signal phase.
- SurroundScope operates in both TDM and RTAS plug-in formats.

### **SurroundScope Parameters**

Each of SurroundScope's interactive meters displays a different type of information about a multi-channel audio signal. Following is an explanation of these meters and their behavior.

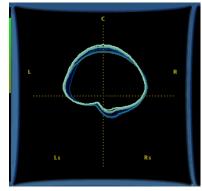


SurroundScope

### **Surround Display**

The Surround Display shows signal position within the surround field. When SurroundScope is inserted on a track, it detects the multi-channel format of the track and displays each speaker channel in the signal in a circle around the Surround Display's center.

Each speaker channel is indicated by letter and is equidistant from the center of the 360° surround field.



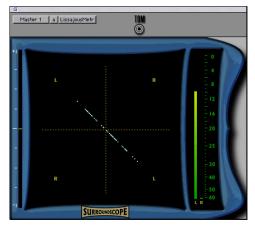
Surround Display

During audio playback, the Surround Display generates a circular metering image whose shape and position change according to the amplitude energy emanating from the various speaker channels in the signal. To understand the precise location of an audio signal within the sound field, observe how the circular sound pattern moves:

- ◆ A perfect circle in the center of the Surround Display denotes a perfectly centered surround signal—one that is panned equally to all channels.
- A signal that balloons toward the outer edges of the display indicates which channels emanate the strongest signal.
- ◆ A signal that is completely panned to one speaker channel appears teardrop-shaped and points toward the channel emanating the signal.

### **Lissajous Meter**

When SurroundScope is inserted on a stereo track, the standard Surround Display is replaced by the Lissajous Meter.



Lissajous Meter with audio panned predominantly left

The Lissajous Meter shows the relationship between the amplitude and phase of a stereo signal, enabling you to monitor and ensure perfect stereo imaging.

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

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

#### **Phase Meter**

The Phase Meter indicates the phase coherency of the left and right channels (exclusively) of a multi-channel signal. This enables you to see at a glance if a signal is in or out of phase.



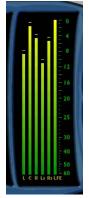
Phase Meter

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

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

#### **Input Level Meters**

The Input Level Meters provide decibel level readings for each channel of a multi-channel signal, including readings for an LFE channel if present. Metering range is from 0 to –60 dB.



Input Level Meters



The input level meters always appear in standard Film layout (L, C, R, Ls, Rs, LFE) for 5.1 format mixes.

### **Using SurroundScope**

Since it has no user-adjustable parameters, SurroundScope is extremely easy to use. Once inserted on a multi-channel track or Master Fader, you can use it to accurately monitor phase, position, and input level metering of multi-channel signals.



For more information about multi-channel signals and multi-channel mixing, refer to the Pro Tools Reference Guide.



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