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Introduction to the Soundtrack Pro Plug-ins

Soundtrack Pro includes a comprehensive collection of powerful effect plug-ins.

This manual will introduce you to the individual effects and their parameters. Using plug-ins is much easier if you are familiar with the basic functions of Soundtrack Pro. Information about these can be found in the Soundtrack Pro User Manual.

Soundtrack Pro Effects
The following table outlines the effects included with Soundtrack Pro.

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Delay

Delay effects store the input signal and hold it for a short time before sending it to the effect input or output.

Most delays allow you to feed a percentage of the delayed signal back to the input, creating a repeating echo effect. Each subsequent repeat is a little quieter than the previous one.

The delay time can often be synchronized to the project tempo by matching the grid resolution of the project, usually in note values or milliseconds.

You can use delays for:
- Doubling individual sounds, making them sound like a group of instruments playing the same melody
- Creating echo effects, placing the sound in a large “space”
- Enhancing the stereo position of tracks in a mix

Delay effects are usually used as channel insert or bussed effects. They are rarely used on an overall mix (in an output channel), unless you’re trying to achieve a special effect, such as an otherworldly-sounding mix.

This chapter describes the delay effects included with Logic Studio:
- “Delay Designer” on page 10
- “Stereo Delay” on page 28
- “Tape Delay” on page 29
Delay Designer

Delay Designer is a multi-tap delay. Each tap is an independent delay. Unlike simple delay effects that only offer one or two delays (or taps), Delay Designer offers you up to 26 individual taps. In other words, you can think of Delay Designer as 26 separate delay processors—in one effect unit.

Delay Designer provides control over the following aspects of each tap:

- Level and pan position
- Highpass and lowpass filters
- Pitch transposition (up or down) over one octave

Further effect-wide parameters include synchronization, quantization, and feedback.

As the name implies, Delay Designer offers significant sound design potential. You can use it for everything from a basic echo effect to an audio pattern sequencer. You can create complex, evolving, moving rhythms by synchronizing the placement of taps—coupled with judicious use of pitch transposing and filtering. Alternatively, you can set up numerous taps as “repeats” of other taps, much as you would use the feedback control of a simple delay—but with individual control over each repeat.

You can use Delay Designer on channels with stereo or surround inputs and outputs. See “Working with Delay Designer in Surround” on page 27 for details on using it in surround channels.
The Delay Designer interface consists of five main sections:

- **Tap display**: This blue “view screen” display features a graphic representation of all taps. You can see and edit the parameters of each tap in this area. See “The Tap Display,” next, for a more detailed look.

- **Tap parameter bar**: Offers a numeric overview of the current parameter settings for the selected tap. You can view and edit the parameters of each tap in this area. See “The Tap Parameter Bar” on page 18.

- **Sync section**: You can set all Delay Designer synchronization and quantization parameters in this section. See “Syncing Delay Taps” on page 24 for more information.

- **Tap pads**: You can use these two pads to create taps in Delay Designer. See “Creating and Deleting Taps” on page 14.

- **Master section**: This area contains the global Mix and Feedback parameters. See “The Master Section” on page 26 for details.
The Tap Display
You can see and interact with taps in the Tap display. The display is divided into a number of sections:

- **View buttons:** Determine the parameter or parameters represented in the Tap display.
- **Autozoom:** When engaged, the main display is zoomed out, making all taps visible. You can override Autozoom and manually zoom the Tap display by dragging the Zoom slider.
- **Overview display:** Shows all the taps in the time range.
- **Toggle buttons:** Click to toggle (switch) the parameters of a particular tap. The parameter being toggled is chosen with the View buttons. The label at the left of the toggle bar always indicates the parameter being toggled. See “Using the Toggle Buttons to Edit Tap Parameters” on page 23 for more information.
- **Main display:** Offers a visual representation of each tap as a shaded blue line. Each tap contains a bright bar (or dot for stereo panning) that indicates the value of the parameter. You can directly edit tap parameters with the mouse in the main display area. See “Editing Taps” on page 19 for more details.
- **Identification bar:** Includes an identification letter for each tap, along with handles that allow you to move the selected tap backward or forward in time.
The View Buttons
The View buttons determine which parameter is represented in the main display.

- **Cutoff**: When clicked, the taps in the main display show the highpass and lowpass filter cutoff frequencies.
- **Reso**: When clicked, the main display shows the filter resonance value of each tap.
- **Transp**: Click to show the pitch transposition of each tap in the main display.
- **Pan**: Click to show the pan parameter of each tap in the main display.
  - For stereo channels, each tap contains a dot showing its stereo balance. A line (extending outward from the dot) indicates its stereo spread.
  - For surround channels, each tap contains a line representing its surround angle (see “Working with Delay Designer in Surround” on page 27 for details).
- **Level**: Click to show the relative volume level of each tap in the main display.

The Overview Display
You can use the Overview display to zoom and navigate the main display area.

**To zoom the main display, do one of the following:**
- Drag the bright rectangle in the overview display up or down.
Drag the highlighted bars (that are to the left or right of the bright rectangle) to the left or right.

*Note:* The Autozoom button needs to be turned off for this to work. When you zoom in on a small group of taps, the overview display continues to show all taps. The area shown in the Tap display is indicated by the bright rectangle.

**To move to different sections of the Tap display:**
- Drag the bright rectangle to the left or right.

The zoomed view in the main display updates as you drag.

**Creating and Deleting Taps**
You can create new delay taps in three different ways: by using the identification bar, by using the Tap pads, or by copying existing taps.

**To create taps in the identification bar:**
- Click at the desired position.
To create taps with a Tap pad:

1. Click the upper Start pad.

   *Note:* Whenever you click the Start pad, it automatically erases all existing taps. Given this behavior, once you have created your initial taps, you will want to create subsequent taps by clicking the identification bar.

   The upper pad label changes to Tap, and a red tap recording bar appears in the strip below the View buttons.

2. Click the Tap button to record new taps on the fly.

   New taps are created (at the exact moments in time) of each click, adopting the rhythm of your click pattern.

3. To finish creating taps, click the Last Tap button.

   This adds the final tap, ending tap recording and assigning the last tap as the *feedback tap* (see “The Master Section” on page 26 for an explanation of the feedback tap).

   *Note:* If you do not click the Last Tap button, tap recording automatically stops after 10 seconds, or when the 26th tap is created, whichever comes first.

To copy taps in the identification bar:

- Option-drag a selection of one or more taps to the desired position.

   The delay time of copied taps is set to the drag position.

**Tap Creation Suggestions**

- The fastest way to create multiple taps is to use the Tap pads. If you have a specific rhythm in mind, you might find it easier to tap out your rhythm on dedicated hardware controller buttons, instead of using mouse clicks.

   Whenever you click the Start Tap pad, it automatically erases all existing taps. Given this behavior, once you have created your initial taps, you will want to create subsequent taps by clicking in the identification bar.

   Once a tap has been created, you can freely adjust its position. See “Moving Taps” on page 18 for details.
**Identifying Taps**
Taps are assigned letters based on their order of creation. The first tap to be created is assigned as Tap A, the second tap is assigned as Tap B, and so on. Once assigned, each tap is always identified by the same letter, even as taps are moved in time, and therefore reordered. As an example, if you initially create three taps, they are named Tap A, Tap B, and Tap C. If you then change the delay time of Tap B so that it precedes Tap A, it is still called Tap A.

The identification bar shows the letter of each visible tap. The Tap Delay field of the Tap parameter bar displays the letter of the currently selected tap or the letter of the tap being edited when multiple taps are selected (see “Selecting Taps” on page 17 for details).

**Deleting Taps**
To delete a tap, select it and press the Delete or Backspace key. You can also drag a tap down, below the Tap display.

These methods also work when more than one tap is selected.

Finally, you can right-click or Control-click any tap in the Delay Designer interface and then choose “Delete tap(s)” from the shortcut menu to delete all taps.
Selecting Taps
There is always at least one selected tap. You can easily distinguish selected taps by color—the toggle bar icons and identification bar letters of selected taps are white.

To select a tap, do one of the following:
- Click a tap in the main display.
- Click the desired tap letter in the identification bar.
- Click the downward pointing arrow in the Tap field of the Tap parameter bar, then choose the desired tap letter from the pop-up menu.

You can select the next or previous tap by clicking the arrow buttons to the left of the Tap name.

To select multiple taps, do one of the following:
- Drag across the background of the main display to select multiple taps.
- Shift-click specific taps in the Tap display to select multiple non-adjacent taps.
Moving Taps

You can move a tap backward or forward in time.

**Note:** When you move a tap, you are actually editing its delay time.

To move a tap in time:

- Select the tap in the identification bar, and drag it forward in time (left) or backward in time (right).

**Note:** Editing the Delay Time parameter in the Tap Delay field of the Tap parameter bar also moves a tap in time. See “The Tap Parameter Bar” on page 18 and “Editing Taps” on page 19 for more details on the Tap Delay field and editing taps.

The Tap Parameter Bar

The Tap parameter bar shows the current numeric values for every parameter of the selected tap. You can directly edit these parameters in the Tap parameter bar.

The parameters shown are:

- **Filter On/Off button:** Enables or disables the highpass and lowpass filters for the selected tap.
- **HP – Cutoff – LP:** You can view and set the cutoff frequencies (in Hz) for the highpass and lowpass filters here.
- **Slope:** Determines how steep the highpass and lowpass filters will be. Click the 6 dB button for a gentler filter slope or the 12 dB button for a steeper, more pronounced filtering effect. You cannot set the slope of the highpass and lowpass filters independently.
- **Reso:** Sets the amount of filter resonance for both filters.
- **Tap Delay field:** This displays both the number (top), name, and delay time (bottom) of the selected tap.
- **Pitch On/Off button:** Enables or disables pitch transposition for the selected tap.
- **Transpose:** Use the first field to set the amount of pitch transposition in semitones, and the second field to fine-tune each semitone step in cents (1/100th of a semitone).
• **Flip:** Swaps the left and right side of the stereo or surround image. In other words, clicking this button reverses the tap position from left to right, or vice versa. For example, if a tap is set to 55% left, clicking the flip button will swap it to 55% right.

• **Pan:** The Pan parameter controls the pan position for stereo and the surround angle for surround. The pan parameter displays a percentage between 100% (full left) and –100% (full right), which represents the pan position or balance of the tap. A value of 0% represents the center panorama position. When used in surround, a surround panner replaces the percentage representation. See “Working with Delay Designer in Surround” on page 27 for more information.

• **Spread:** When a stereo instance of Delay Designer is used, this parameter allows you to set the width of the stereo spread for the selected tap.

• **Mute:** Clicking this button mutes or unmutes the selected tap.

• **Level:** Determines the output level for the selected tap.

**Editing Taps**

You can edit taps both graphically, using the main Tap display, and numerically, using the Tap parameter bar. All tap edits are reflected both graphically and numerically.

**Editing Taps in the Tap Parameter Bar**

You can edit every parameter in the Tap parameter bar by clicking or by dragging.

To edit a parameter in the Tap parameter bar:

- Click a button or the up or down arrow to enable, disable, or alter a parameter value.

- Drag a parameter value up or down to change it.

If you have multiple taps selected in the Tap display, the values of all selected taps are increased or decreased. These changes are relative to other taps.

Option-clicking a parameter resets it to the default setting. If multiple taps are selected, Option-clicking a parameter of one tap resets that parameter to its default value for all selected taps.
**Editing Parameters in the Tap Display**

You can graphically edit any tap parameter that is represented as a vertical line in the main Tap display.

**To edit a tap parameter in the Tap display:**

1. Click the View button of the parameter you want to edit.
2. Drag the bright line of the tap you wish to edit up or down (or drag one of the selected taps, if multiple taps are selected).

If you have multiple taps selected, the values of all selected taps are increased or decreased relative to other taps.

You can also set the value of multiple taps by Command-dragging horizontally and vertically across several taps in the Tap display. As you do so, the parameter value changes to match the mouse position as you drag across the taps. Command-dragging across several taps allows you to “draw” in values.
You can also hold down the Command key and click the Tap display before dragging. This results in a line trailing behind the pointer. The values of the taps are aligned along the line when you release the mouse button.

Option-clicking a tap resets the chosen parameter to its default setting. If multiple taps are selected, Option-clicking one tap resets that parameter to its default value for all selected taps.

**Editing the Filter Cutoff Parameters in the Tap Display**

While the steps outlined above apply for most graphically editable parameters, the Cutoff and Pan parameters work in a slightly different fashion.

In the Cutoff view, each tap actually shows two parameters—highpass and lowpass filter cutoff frequency. The filter cutoff values can be adjusted independently by dragging the specific cutoff frequency line (upper line is highpass, lower line is lowpass), or both cutoff frequencies can be adjusted by dragging between them.

When the highpass filter cutoff frequency value is lower than that of the lowpass cutoff frequency, *only one line is shown*. This line represents the frequency band that passes through the filters (in other words, the filters act as a bandpass filter). In this configuration, the two filters operate *serially*, meaning the tap first passes through one filter, then the other.
If the highpass filter’s cutoff frequency value is above that of the lowpass filter cutoff frequency, the filter switches from serial operation to parallel operation, meaning the tap passes through both filters simultaneously. In this case, the space between the two cutoff frequencies represents the frequency band being rejected (in other words, the filters act as a band-reject filter).

**Editing the Pan Parameter in the Tap Display**

The way that the Pan parameter is represented in the Pan view is entirely dependent on the input channel configuration of Delay Designer.

With stereo configurations, the Pan parameter adjusts the stereo balance, not the position of the tap in the stereo field. The Pan parameter takes the form of a stereo balance dot on the tap representing its stereo balance. To adjust the balance, drag the stereo balance dot up or down the tap.

By default, the stereo spread is set to 100%. To adjust this, drag either side of the dot. As you do so, the height of the line (extending vertically upward or downward from the dot) changes. Keep an eye on the spread parameter in the Tap parameter bar to view the spread percentage numerically.

In surround configurations, the bright line represents the surround angle. See “Working with Delay Designer in Surround” on page 27 for more information.
Using the Toggle Buttons to Edit Tap Parameters

Each tap has its own toggle button in the Toggle bar. These buttons offer you a quick way to graphically activate and deactivate parameters. The specific parameter being toggled by the toggle buttons depends on the current View button selection:

- **Cutoff view**: Toggle buttons turn the filter on or off.
- **Reso view**: Toggle buttons switch filter slope between 6 dB and 12 dB.
- **Pitch view**: Toggle buttons switch pitch transposition on or off.
- **Pan view**: Toggle buttons switch between the Flip modes.
- **Level view**: Toggle buttons mute or unmute the tap.

Option-Command-clicking a toggle button switches the mute state, regardless of the current view. When you release the Option and Command keys, the toggle buttons return to their standard functionality (in the active View mode).

**Note**: The first time you edit a filter or pitch transpose parameter, the respective module automatically turns on. This saves you the effort of manually turning on the filter or pitch transposition module before editing. Once you manually turn either of these modules off, however, you will need to manually switch it back on.

Editing Tap Parameters Using the Shortcut Menu

Right-clicking or Control-clicking a tap opens a shortcut menu that contains the following commands:

- **Copy sound parameters**: Copies all parameters—except the delay time—of the selected tap or taps onto the Clipboard.
- **Paste sound parameters**: Pastes the tap parameters stored in the Clipboard into the selected tap or taps. If there are more taps in the Clipboard than are selected in the main Tap display, the extra taps in the Clipboard are ignored.
- **Reset sound parameters to default values**: Resets all parameters of all selected taps—except the delay time—to the default values.
- **Delete tap(s)**: Deletes all taps.
Parameter Editing Suggestions
In general, you’ll find editing in the Tap parameter bar fast and precise when you want
to edit the parameters of one tap at a time. All parameters of the selected tap are
available, with no need to switch display views or estimate values with vertical lines.

If you want to edit the parameters of one tap relative to other taps, use the Tap display.
Also, if you want to edit multiple taps at once, you can use the Tap display to select
multiple taps and then edit them together.

Don’t forget Command-dragging to draw in different values for multiple taps.

Syncing Delay Taps
Delay Designer can either synchronize to the project tempo or operate independently.
When Delay Designer is in synchronized mode (Sync mode), taps snap to a grid of
musically relevant positions based on note durations. You can also set a Swing value
when in Sync mode; this varies the precise timing of the grid, resulting in a more laid
back, less robotic feel for each tap. When not in Sync mode, taps don’t snap to any grid,
and you cannot apply the Swing value.

Activating Sync Mode
Sync mode is turned on or off by clicking the Sync button in the Sync section.

An orange ring is shown around the Sync button when Sync mode is on, and a grid
that matches the chosen Grid parameter value is shown in the identification bar.

Once Sync mode is activated, all taps move toward the closest delay time value on the
grid. When you subsequently create or move taps, they always move in increments
based on the current grid setting or are created at a “snapped” position on the grid.
**Setting the Grid Resolution**

The Grid menu offers several grid resolutions, which correspond to musical note durations. The grid resolution, along with the project tempo, determines the length of each grid increment.

**To set the grid resolution:**

- Click the Grid field, then choose the desired grid resolution from the pop-up menu.

As you change grid resolutions, the increments shown in the identification bar change accordingly. This also determines a step limitation for all taps.

For example, the current project tempo is set to 120 beats per minute, and the Delay Designer Grid parameter is set to 1/16th notes. At this tempo and grid resolution, each grid increment is 125 milliseconds apart. If Tap A is currently set to 380 ms, turning on Sync mode would immediately shift Tap A to 375 ms. If you subsequently moved Tap A forward in time, it would snap to 500 ms, 625 ms, 750 ms, and so on.

At a resolution of 1/8th notes, the steps are 250 milliseconds apart, so Tap A would automatically snap to the nearest division (500 ms) and could be moved to 750 ms, 1000 ms, 1250 ms, and so on.

**Setting the Swing Value**

The Swing value determines how close to the absolute grid position every second grid increment will be. A Swing setting of 50% means that every grid increment has the same value. Settings below 50% result in every second increment being shorter in time. Settings above 50% result in every second grid increment being longer in time.

**To adjust the Swing value:**

- Drag up or down in the Swing field to raise or lower the Swing value.

By subtly varying the grid position of every second increment (values between 45% and 55%), the Swing function creates a less rigid rhythmic feel. This can be a very “humanizing” effect, but you are not limited to using the Swing function in this way.

Extremely high Swing settings are not subtle at all, as they place every second increment directly beside the subsequent increment. You can use this feature to create interesting and intricate double rhythms with some taps, while retaining the grid to lock other taps into a more rigid synchronization with the project tempo.
**Saving Sync Settings**
When you save a Delay Designer setting, the Sync mode status, Grid, and Swing values are all saved. When you save a setting with Sync mode on, the grid position of each tap is also stored. This ensures that a setting loaded into a project with a different tempo (to that of the project that the setting was created in) will retain the relative positions and rhythm of all taps—at the new tempo.

One point to bear in mind, however, is that Delay Designer offers a maximum delay time of 10 seconds. This means that if you load a setting into a project with a slower tempo than the tempo at which it was created, some taps may fall outside the 10 second limit. In such cases, these taps are not played, but are still retained as part of the setting.

**The Master Section**
The Master section incorporates parameters for two global functions: delay feedback and dry/wet mix.

**Using Feedback**
In simple delays, the only way for the delay to repeat is to use feedback. Because Delay Designer offers 26 taps, you can use these to create repeats, rather than requiring discrete feedback controls for each tap.

Delay Designer’s global feedback parameter allows you to send the output of one tap back through the effect input to create a self-sustaining rhythm or pattern.

This tap is called the *feedback tap*. 
To toggle feedback on or off:

- Click the Feedback button.

When the Feedback button is turned on, it is lit. The orange track around the Feedback Level knob indicates the current feedback level.

**Note:** If feedback is turned on and you begin creating taps using the Tap pads, feedback is automatically switched off. When you stop creating taps with the Tap pads by clicking the Last Tap button, feedback is automatically turned back on.

To determine the feedback tap:

- Click the Feedback Tap field, then choose the desired tap from the pop-up menu.

You can vary the output level of the feedback tap back into Delay Designer’s input between 0% (no feedback) and 100% (the feedback tap is fed back at full volume).

To set the feedback level of the feedback tap, do one of the following:

- Drag the Feedback Level knob.
- Drag the Feedback Level field.

**The Mix Sliders**

Use the Mix sliders to adjust the level of the dry input signal and the (post-processing) wet signal.

**Working with Delay Designer in Surround**

Delay Designer is optimally designed for use in surround configurations. With 26 taps, you can fly delay taps all over the surround field for a variety of rhythmic effects.

When using Delay Designer in a surround configuration, the pan percentage on the Tap parameter bar is replaced with a surround panner, allowing you to determine the surround position of each tap.

The movement of the surround position is made easier with these functions:

- Hold down the Command key to lock diversity.
- Hold down the Command and Option keys to lock the angle.
- Option-click the blue dot to reset angle and diversity.

In the Tap display’s Pan view, you can adjust only the angle of the tap between 0 and 360 degrees, not its diversity.
Delay Designer always processes each input channel independently. In surround configurations, Delay Designer processes each surround channel independently, and the surround panner lets you adjust the surround balance of each tap in the surround field.

**Note:** The Delay Designer generates separate automation data for stereo pan and surround pan operations. This means that when using the Delay Designer in surround channels, it will not react to existing stereo pan automation data, and vice versa.

**Stereo Delay**

The Stereo Delay works much like the Tape Delay (see “Tape Delay” on page 29), but allows you to set the Delay, Feedback, and Mix parameters separately for the left and right channels.

The effect also features a Crossfeed knob for each stereo side. It determines the feedback intensity—or the level at which each signal is routed to the opposite stereo side.

You can freely use the Stereo Delay on mono tracks or busses, when you want to create independent delays for the two stereo sides.

**Note:** If you do use the effect on mono channel strips, the track or bus will have two channels from the point of insertion (all Insert slots after the chosen slot will be stereo).

This section only covers the additional features offered by the Stereo Delay. For more information about the parameters shared with the Tape Delay, see “Tape Delay” on page 29.

- **Left Input and Right Input:** Use these to choose the input signal for the two stereo sides. Options include Off, Left, Right, L+R, and L-R.
- **Feedback Phase button:** Use to invert the phase of the corresponding channel’s feedback signal.
- **Crossfeed Left to Right and Crossfeed Right to Left**: Use to transfer the feedback signal of the left channel to the right channel, and vice versa.
- **Crossfeed Phase buttons**: Use to invert the phase of the crossfed feedback signals.

**Tape Delay**
The Tape Delay simulates the warm sound of vintage tape echo machines, with the convenience of easy delay time synchronization to your project tempo.

The Tape Delay is equipped with a highpass and lowpass filter in the feedback loop, making it easy to create authentic dub echo effects. It also includes an LFO for delay time modulation. The LFO produces a triangular wave, with adjustable speed and modulation intensity. You can use it to produce pleasant or unusual chorus effects, even on long delays.

- **Feedback**: Determines the amount of delayed and filtered signal that is routed back to the input of the Tape Delay.
- **Freeze**: Captures the current delay repeats and sustains them until the Freeze parameter is released.
- **Delay**: Sets the current delay time in milliseconds (this parameter is dimmed when you synchronize the delay time to the project tempo).
- **Tempo**: Sets the current delay time in beats per minute (this parameter is dimmed when you synchronize the delay time to the project tempo).
- **Sync button**: Switch this on to synchronize delay repeats to the project tempo (including tempo changes).
- **Note buttons**: Click to set the grid resolution for the delay time, in note durations.
- **Groove slider**: Determines the proximity of every second delay repeat to the absolute grid position (how close every second delay repeat is).
- **Distortion Level (Extended Parameter)**: Determines the level of the distorted (tape saturation) signal.
- **Low Cut and High Cut**: Frequencies below the Low Cut value and above the High Cut value are filtered out of the source signal.
• **LFO Speed**: Sets the frequency (speed) of the LFO.

• **LFO Depth**: Sets the amount of LFO modulation. A value of 0 turns delay modulation off.

• **Flutter parameters**: Simulates the speed irregularities of the tape transports used in analog tape delay units. Flutter Rate adjusts the speed, and Flutter Intensity determines how pronounced the effect is.

• **Smooth**: Evens out the LFO and flutter effect.

• **Dry and Wet**: These individually control the amount of original and effect signal.

**Setting the Feedback**
When you set the Feedback slider to the lowest possible value, the Tape Delay generates a single echo. If Feedback is turned all the way up, the echoes are repeated ad infinitum.

**Note**: The levels of the original signal and its taps (echo repeats) tend to accumulate, and may cause distortion. This is where the internal tape saturation circuit comes to the rescue—it can be used to ensure that these overdriven signals continue to sound good.

**Setting the Groove Value**
The Groove value determines the proximity of every second delay repeat to the absolute grid position. A Groove setting of 50% means that every delay will have the same delay time. Settings below 50% result in every second delay being played earlier in time. Settings above 50% result in every second delay being played later in time. When you want to create dotted note values, move the Groove slider all the way to the right (to 75%); for triplets, select the 33.33% setting.

**Filtering the Delay Effect**
You can shape the sound of the echoes, using the on-board highpass and lowpass filters. The filters are located in the feedback circuit, meaning that the filtering effect increases in intensity with each delay repeat. If you're after an increasingly “muddy” tone, move the High Cut filter slider toward the left. For “thinner” echoes, move the Low Cut filter slider toward the right.

**Note**: If you're unable to hear the effect, even though you seem to have a suitable configuration, be sure to check out both the Dry/Wet controls and the filter settings. Move the High Cut filter slider to the far right and the Low Cut filter slider to the far left.
Distortion

You can use Distortion effects to recreate the sound of analog or digital distortion, and to radically transform your audio.

Distortion effects simulate the distortion created by vacuum tubes, transistors, or digital circuits. Vacuum tubes were used in audio amplifiers before the development of digital audio technology, and are still used in musical instrument amps today. When overdriven, they produce a type of distortion which many people find musically pleasing, and which has become a familiar part of the sound of rock and pop music. Analog tube distortion adds a distinctive warmth and bite to the signal.

There are also distortion effects that intentionally cause clipping and digital distortion of the signal. These can be used to modify vocal, music, and other tracks to produce an intense, unnatural effect or for creating sound effects.

Distortion effects include parameters for tone, which let you shape the way in which the distortion alters the signal (often as a frequency-based filter), and for gain, which let you control how much the distortion alters the output level of the signal.

**Warning:** When set to high output levels, distortion effects can damage your hearing (and speakers). When adjusting effect settings, it is recommended that you lower the output level of the track, and raise the level gradually when you are finished.

The following sections describe the individual effects included with Soundtrack Pro:

- “Bitcrusher” on page 32
- “Clip Distortion” on page 33
- “Distortion” on page 34
- “Distortion II” on page 34
- “Overdrive” on page 35
- “Phase Distortion” on page 36
Bitcrusher

The Bitcrusher is a low-resolution digital distortion effect. You can use it to emulate the sound of early digital audio, create artificial aliasing by dividing the sample rate, or distort signals until they are unrecognizable.

Bitcrusher Parameters

- **Drive slider and field**: Sets the amount of gain (in decibels) applied to the input signal.
- **Resolution slider and field**: Sets the bit rate (between 1 and 24 bits).
- **Downsampling slider and field**: Sets the amount by which the sample rate is reduced. A value of 1x leaves the signal unchanged, a value of 2x halves the sample rate, and a value of 10x reduces the sample rate to one-tenth of the original signal. (For example, if you set Downsampling to 10x, a 44.1 kHz signal is sampled at just 4.41 kHz.)
- **Mode buttons**: Click one of the buttons to set the distortion mode to Folded, Cut, or Displaced (each of which is described in the following section).
- **Clip Level slider and field**: Sets the point below the normal threshold at which the signal starts clipping.

Using the Bitcrusher

Setting the Resolution parameter to a value lower than the bit rate of the original signal degrades the signal, introducing digital distortion. Lowering the value increases the number of sampling errors, generating more distortion. At extremely low bit rates, the amount of distortion can be greater than the level of the usable signal.

The Mode buttons determine whether signal peaks that exceed the clip level are Folded, Cut, or Displaced (as displayed on the button icons and the resulting waveform in the display). The kind of clipping that occurs in digital systems is usually closest to that of the center mode (Cut). Internal distortion may generate clipping similar to the types generated by the other two modes.

Raising the Drive level tends to increase the amount of clipping at the output of the Bitcrusher as well.
Clip Distortion

Clip Distortion is a nonlinear distortion effect that produces unpredictable spectra. You can use it to simulate warm, overdriven tube sounds, and also to create drastic distortion.

Clip Distortion features an unusual combination of serially connected filters. After being amplified by the Drive value, the signal passes through a highpass filter, and is then subjected to nonlinear distortion, as controlled by the Symmetry parameter. After the distortion, the signal passes through a lowpass filter. The effected signal is mixed with the original signal, after which the mixed signal is sent through another lowpass filter. All three filters have a slope of 6 db/octave.

This unique combination of filters allows for gaps in the frequency spectra that can sound quite good with this sort of nonlinear distortion. The clip circuit graphic visually depicts every parameter except for the High Shelving filter parameters.

Clip Distortion Parameters

- **Drive slider and field**: Sets the amount of gain applied to the input signal. After being amplified by the Drive value, the signal passes through a highpass filter.
- **Tone slider and field**: Sets the cutoff frequency (in Hertz) of the highpass filter.
- **Symmetry slider and field**: Sets the amount of nonlinear (asymmetrical) distortion applied to the signal.
- **Clip Filter slider and field**: Sets the cutoff frequency (in Hertz) of the first lowpass filter through which the signal passes after distortion.
- **Mix slider**: Sets the ratio of the effected (wet) signal to the uneffected (dry) signal following the Clip Filter.
- **Sum LPF circular slider and field**: Sets the cutoff frequency (in Hertz) of the lowpass filter through which the mixed signal passes.
- **High Shelving Frequency knob and field**: Sets the frequency (in Hertz) of the high shelving filter.
- **High Shelving Gain knob and field**: Sets the amount of gain applied to the output signal.
Using Clip Distortion
If you set the High Shelving Frequency to around 12 kHz, you can use it like the treble control on a mixer channel strip or a stereo hi-fi amplifier. Unlike those types of treble controls, however, you can boost or cut the signal by up to ±30 dB using the Gain parameter.

Distortion
This Distortion effect simulates the lo-fi, dirty distortion generated by a bipolar transistor. You can use it to simulate playing a musical instrument through a highly overdriven amplifier, or to create unique distorted sounds.

Distortion Parameters

- **Drive slider and field**: Sets the amount of saturation applied to the signal.
- **Tone slider and field**: Sets the frequency at which the signal is filtered by a high cut filter. Filtering the harmonically rich distorted signal produces a somewhat less grating, softer tone.
- **Output slider and field**: Sets the output volume level. This allows you to compensate for increases in loudness caused by adding distortion.

Distortion II
Distortion II emulates the distortion effect section of a Hammond B3 organ. You can use it on musical instruments to recreate this classic effect, or use it creatively when designing new sounds.
Distortion II Parameters

- **PreGain dial**: Sets the amount of gain applied to the input signal.
- **Drive dial**: Sets the amount of saturation applied to the signal.
- **Tone dial**: Sets the frequency at which the signal is filtered. Filtering the harmonically rich distorted signal produces a somewhat less grating, softer tone.
- **Type pop-up menu**: Choose the type of distortion you want to apply. The choices are: Growl, Bity, and Nasty.
  - **Growl**: Emulates a two-stage tube amplifier, similar to the type found in a Leslie 122 model, often used together with a Hammond B3 organ.
  - **Bity**: Emulates the sound of a bluesy (overdriven) guitar amp.
  - **Nasty**: Produces hard distortion, suitable for creating very aggressive sounds.

Overdrive

The Overdrive effect emulates the distortion produced by a field effect transistor (FET), which is commonly used in solid-state musical instrument amplifiers and hardware effects devices. When saturated, FETs generate a warmer sounding distortion than bipolar transistors.

Overdrive Parameters

- **Drive slider and field**: Sets the amount of saturation of the transistor.
- **Tone slider and field**: Sets the cutoff frequency at which the signal is filtered. Filtering the harmonically rich distorted signal produces a somewhat less grating, softer tone.
- **Output slider and field**: Sets the output volume level. Using the Overdrive plug-in tends to increase the level of the original signal, and you can compensate for this by lowering the Output level.
Phase Distortion
The Phase Distortion effect is based on a modulated delay line, similar to a chorus or flanger effect (for more information about these effects, see Chapter 8, “Modulation,” on page 95). Unlike these effects, however, in the Phase Distortion effect the delay time is not modulated by a low frequency oscillator (LFO), but rather by a lowpass-filtered version of the input signal itself. This means that the signal modulates its own phase position. The input signal only passes the delay line and is not affected by any other process.

Phase Distortion Parameters

- **Monitor button**: Turn on to hear only the input signal, or turn off to hear the mixed signal.
- **Cutoff circular slider and field**: Sets the cutoff frequency of the resonant lowpass filter through which the input signal passes.
- **Resonance circular slider and field**: Sets the resonance of the resonant lowpass filter through which the input signal passes.
- **Mix slider and field**: Adjusts the percentage of the effected signal mixed with the original signal.
- **Max Modulation slider and field**: Sets the maximum delay time.
- **Intensity slider and field**: Sets the amount of modulation applied to the signal.
Using the Phase Distortion

The input signal only passes the delay line and is not affected by any other process. The Mix parameter blends the effected signal with the original signal. The delay time is modulated by a side chain signal—namely, the input signal. The input signal passes through a resonant lowpass filter, with dedicated Cutoff frequency and Resonance controls. You can listen to the filtered side chain (instead of the Mix signal) by turning on the Monitor button. You set the maximum delay time via the Max Modulation parameter. The amount of modulation itself is controlled with Intensity.

Below the other parameters is the Phase Reverse parameter. Normally, a positive input value results in a longer delay time. By turning on the Phase Reverse parameter, positive input values result in a reduction of the delay time on the right channel only. This is only available for stereo instances of the effect.
Dynamics

You can use Dynamics effects to control the perceived loudness of your audio, add focus and punch to tracks and projects, and optimize the sound for playback in different situations.

The *dynamic range* of an audio signal is the range between the softest and loudest parts of the signal (technically, between the lowest and the highest amplitude). Using dynamics effects, you can adjust the dynamic range of individual audio files, tracks, or an overall project to increase the perceived loudness, and highlight the most important sounds while making sure softer sounds are not lost in the mix. Dynamics effects include compressors, limiters, and noise gates.

**Compressors**

A compressor works like an automatic volume control, lowering the volume whenever it rises above a certain level, called the *threshold*. Why would you want to reduce the dynamic level? By cutting the highest parts of the signal (called *peaks*), the compressor lets you raise the overall level of the signal, thereby increasing the perceived volume. This gives the sound more focus by making the louder foreground parts stand out while keeping the softer background parts from becoming inaudible. Compression also tends to make sounds tighter or punchier because transients are emphasized (depending on attack and release settings) and because the maximum volume is reached more swiftly.

In addition, compression can help make a project sound better when played back in different audio environments. For example, the speakers on a television set or in a car sound system typically have a narrower dynamic range than the sound system in a theater. Compressing the overall mix can help make the sound fuller and clearer in lower-fidelity playback situations.

Compressors are typically used on vocal tracks to make the vocals prominent in an overall mix. They can also be used on music and sound effects tracks, but rarely on ambience tracks.
Some compressors, called multiband compressors, can divide the incoming signal into
different frequency bands, and apply different compression settings to each band. This
helps achieve the maximum level without introducing compression artifacts, and is
typically used on an overall project mix.

Expanders
Expanders are similar to compressors, except that they raise, rather than lower, the
signal when it exceeds the threshold. Expanders are used to enliven the audio signal.

Limiters
Limiters (also called peak limiters) work in a similar way as compressors, in that they
reduce the audio signal when it exceeds a set threshold. The difference is that while a
compressor gradually lowers the signal level above the threshold, a limiter quickly
reduces any signal louder than the threshold to the threshold level. The main use of a
limiter is to prevent clipping while preserving the maximum overall signal level.

Noise Gates
Noise gates alter the signal in the opposite way that compressors or limiters do. While a
compressor lowers the level when the signal passes above the threshold, a noise gate
lowers the signal wherever it is below the threshold. Louder sounds pass through
unchanged, but softer sounds, such as ambient noise or the decay of a sustained
instrument, are cut off. Noise gates can be used mainly to eliminate low-level noise or
hum from an audio signal.

The following sections describe the effects included with Soundtrack Pro:

- “Adaptive Limiter” on page 41
- “Compressor” on page 42
- “DeEsser” on page 45
- “Enveloper” on page 46
- “Expander” on page 48
- “Limiter” on page 49
- “Multipressor” on page 50
- “Noise Gate” on page 53
- “Surround Compressor” on page 55
Adaptive Limiter
The Adaptive Limiter is a versatile tool for controlling the perceived loudness of sounds. It works by rounding and smoothing peaks in the signal, producing an effect similar to an analog amplifier being driven hard. Like an amplifier, it can slightly color the sound of the signal. You can use the Adaptive Limiter to achieve maximum gain without clipping (exceeding 0 dBFS).

The Adaptive Limiter is typically used in the final mix, where it may be placed after a compressor (such as the Multipressor) and before a final gain control in order to produce a mix of maximum loudness. Using the Adaptive Limiter can produce a louder-sounding mix than can be achieved by simply normalizing the signal.

Note: Using the Adaptive Limiter adds latency when the Lookahead parameter is active. In most situations, it should be used for mixing and mastering previously recorded tracks, not when recording.

Adaptive Limiter Parameters

- **Input Scale knob**: Scales the input level. Scaling is useful with very high or low input signals, to bring the level into the most effective range for the Gain knob to work effectively. In general, it should never exceed 0 dBFS.
- **Gain knob**: Sets the amount of gain after input scaling.
- **Out Ceiling knob**: Sets the maximum output level, or ceiling, above which the signal will not rise.

Input meters (to the left of the control dials) show the input levels in real time as the file or project plays. The Output meters show the output levels, allowing you to see the results of the Adaptive Limiter. The two Margin fields show the highest level for input and output respectively (since the start of playback). You can reset the Margin fields by clicking them.
Compressor

The Compressor is designed to emulate the sound and response of a professional-level analog (hardware) compressor. It tightens up your audio by reducing sounds that exceed a certain threshold level, smoothing out the dynamics and increasing the overall volume—the perceived loudness. Compression helps bring the key parts of a track or a mix into focus while preventing softer parts from being inaudible. It is probably the most versatile and widely used sound-shaping tool used in mixing, next to EQ.

You can use the Compressor with individual tracks, including vocal, instrumental, and effects tracks, as well as on the overall mix. In most cases, you’ll want to insert the Compressor directly into a channel.

Compressor Parameters

- **Circuit Type slider and field**: Choose the type of circuit emulated by the Compressor. The choices are Platinum, Classic A_R, Classic A_U, VCA, FET, and Opto (optical).
- **Gain Reduction display**: Shows the amount of compression applied as the audio plays.
- **Attack knob and field**: Sets the attack time (the amount of time it takes for the compressor to react when the signal exceeds the threshold).
- **Release knob and field**: Sets the release time (the amount of time it takes for the compressor to stop reducing the signal when the signal falls below the threshold).
- **Auto button**: When selected, the release time dynamically adjusts to the audio material.
- **Compression curve display**: Shows the compression curve created by the Ratio and Knee parameters, with input as the X-axis and output as the Y-axis.
- **Ratio slider and field**: Sets the compression ratio (the ratio by which the signal is reduced when it exceeds the threshold).
- **Knee slider and field**: Adjusts whether the signal is compressed immediately or more gradually at levels close to the threshold.
- **Compression Threshold slider and field**: Sets the threshold for the Compressor (the level above which the signal is reduced).
• **Peak/RMS buttons:** Turn on one or the other to set whether the Compressor analyzes the signal using the Peak or RMS method when using the Platinum Circuit Type.

• **Gain slider and field:** Sets the amount of gain applied to the output signal.

• **Gain pop-up menu:** Choose a value to raise the output level in order to compensate for volume reduction caused by compression. The choices are OFF, 0 dB, and –12 dB.

• **Limiter Threshold slider and field:** Sets the threshold level for the limiter.

• **Limiter button:** Turns the integrated limiter on or off.

**Using the Compressor**

The following sections provide information on using each of the main Compressor parameters.

**Threshold and Ratio**

The most important Compressor parameters are Threshold and Ratio. The Threshold is the level (in decibels) above which the signal is reduced by the amount set as the Ratio. Because the Ratio is a percentage of the overall level, the more the signal exceeds the threshold, the more it is reduced. For example, with the Threshold set at –6 dB and the Ratio set to 4:1, a –2 dB peak in the signal (4 dB louder than the threshold) is reduced by 3 dB so that it is just 1 dB above the threshold, while a +6 dB peak (12 dB above the threshold) is reduced by 9 dB so that it is 3 dB above the threshold. The scale of dynamics is preserved, but the differences between the peaks are evened out.

**Attack and Release**

After Threshold and Ratio, the most important parameters are Attack and Release. You use the Attack and Release parameters to shape the dynamic response of the Compressor. The Attack parameter sets the amount of time after the audio exceeds the threshold before the Compressor starts reducing the signal. For many sounds, including voices and musical instruments, the initial attack is important for defining the sound, and setting the Attack parameter higher ensures that the original attack is not altered. To maximize the level of an overall mix, setting the Attack parameter lower ensures that the Compressor starts reducing the signal right away.

Similarly, the Release parameter controls how quickly the Compressor stops reducing the signal when it falls below the threshold. Setting the Release parameter higher makes the difference in dynamics smoother, while setting it lower can make the difference more abrupt. Adjusting the Attack and Release parameters properly can help avoid pumping, a common side effect of compression.
Knee
The Knee parameter smooths the effect of the Compressor by controlling whether the signal is slightly compressed as it approaches the threshold. Setting the Knee parameter close to 0 (zero) means that levels just below the threshold are not compressed at all (1:1 ratio), while levels at the threshold are compressed by the full Ratio amount (4:1, 10:1, or more). This is what audio engineers call hard knee compression, which can cause the transition to be abrupt as the signal reaches the threshold. Increasing the value of the Knee parameter applies some compression to the signal as it approaches the threshold, creating a smoother transition. This is called soft knee compression. Setting the Knee parameter controls the shape of compression around the threshold, while the Threshold and Ratio parameters control its intensity.

Other Parameters
Because the Compressor works by reducing levels, the overall volume of its output is typically lower than the input signal. You can adjust the output level using the Gain slider.

You can use the Auto Gain parameter to compensate for the reduction in gain produced by compression, referenced to either –12 dB or 0 dB. Auto Gain sets the level of gain (amplification) to a value of $T - (T/R)$, where $T =$ the Threshold and $R =$ the Ratio.

The Gain Reduction Meter displays the amount of compression occurring as the signal plays. It's useful to watch how much your tracks are being compressed, and to make sure they're not being overly compressed.

When using the Platinum Circuit Type, the Compressor can analyze the signal using one of two methods: Peak or RMS (root mean square). While Peak is more technically accurate, RMS provides a better indication of how people perceive the signal's loudness. When using the Compressor primarily as a limiter, select the Peak button. When compressing individual tracks, especially music tracks, select the RMS button.

If you activate Auto Gain and RMS simultaneously, the signal may be saturated. If you hear any distortion, switch Auto Gain off and adjust the Gain slider until the distortion is gone.
DeEsser
The DeEsser is a frequency-specific compressor, designed to compress only a particular
frequency band within a complex audio signal. It is used to eliminate hiss (also called
sibilance) from the signal. The advantage of using the DeEsser instead of an EQ effect to
cut high frequencies is that it compresses the signal dynamically rather than statically.
This prevents the sound from becoming darker when no sibilance is present in the
signal. The DeEsser features extremely fast attack and release times.

When using the DeEsser, you can set the frequency range being compressed (the
Suppressor frequency) independently of the frequency range being analyzed (the
Detector frequency). The two ranges appear separately in the DeEsser window for easy
comparison. The DeEsser performs gain reduction on the Suppressor frequency range
for as long as the threshold for the Detector frequency is exceeded.

The DeEsser does not use a frequency dividing network (a crossover utilizing low and
high pass filters). Rather, it is based on subtracting the isolated frequency band, and so
does not alter the phase curve.

DeEsser Parameters

The Detector parameters are on the left side of the DeEsser window, and the
Suppressor parameters are on the right. The center section includes the Detector and
Suppressor displays and the Smoothing slider.

Detector Section
• **Detector Frequency knob:** Sets the frequency range the DeEsser analyzes.
• **Detector Sensitivity knob:** Sets the degree of responsiveness to the input signal. At
  higher ratios, the Detector reacts more responsively.
• **Monitor pop-up menu:** Choose whether to monitor the filtered Detector signal (**Det.**),
  the filtered Suppressor signal (**Sup.**), or the sound removed from the input signal in
  response to the Sensitivity parameter (**Sens.**). Choose **Off** to hear the DeEsser output.
Suppressor Section
- **Suppressor Frequency knob**: Sets the frequency band that is reduced when the Detector frequency sensitivity threshold is exceeded.
- **Strength knob**: Sets the amount of gain reduction around the Suppressor frequency.

Center Section
- **Detector and Suppressor frequency displays**: The upper display shows the Detector frequency range, and the lower display shows the Suppressor frequency range (in Hz).
- **Smoothing slider**: Sets the reaction speed of the gain reduction start and end phases. Smoothing controls both the attack and release time (as they are used by compressors).

Enveloper
The Enveloper is an unusual effect that lets you shape transients—the attack and release phases of a signal. This gives it a unique capability to shape the signal, and can be used to achieve impressive results different than any other dynamics effect.

Enveloper Parameters
The Gain and Time controls on the left apply to the attack portion of the signal, while the Gain and Time controls on the right apply to the release portion.
- **Threshold slider and field**: Sets the threshold above which the attack and release levels are altered.
- **(Attack) Gain slider and field**: Sets the gain on the attack phase of the signal. When set to the center (0) position, the signal is unaffected.
- **(Attack) Time knob**: Sets the duration from the beginning of the signal considered as the attack.
- **Display area**: Graphically displays the attack and release curves applied to the signal.
- **(Release) Time knob**: Sets the duration of the signal considered as the release.
- **(Release) Gain slider**: Sets the gain on the release phase of the signal. When set to the center (0) position, the signal is unaffected.
- **Out Level slider**: Sets the level of the output signal.
- **Lookahead slider and field**: Adjusts how far the Enveloper looks ahead in the signal.
Using the Enveloper

The most important parameters of the Enveloper are the two Gain sliders, one on each side of the central display area, that govern Attack (left) and Release (right). Raising the gain emphasizes the attack or release phase, respectively, while lowering the gain attenuates the corresponding phase.

For example, boosting the attack gives a drum sound more snap, or amplifies the initial pluck (or pick) sound of a stringed instrument. Cutting the attack causes percussive signals to fade in more softly. You can also mute the attack, making it virtually inaudible. Another handy application for this effect is to mask the poor timing of accompanying instruments.

Emphasizing the release also boosts any reverb applied to the affected track. Conversely, toning down the release phase makes tracks originally drenched in reverb sound drier. This is particularly useful when working with drum loops, but it has many other applications as well. Let your imagination be your guide.

When using the Enveloper, set the Threshold to the minimum value and leave it there. Only when you seriously raise the release phase, which boosts the noise level of the original recording, should you raise the Threshold slider a little. This limits the Enveloper to affecting only the useful part of the signal.

Drastic boosting or cutting of either the release or attack phase may change the overall level of the signal. You can compensate for this by lowering the Out Level slider.

The Time parameters for attack and release (below the display area) enable you to access the time-based intervals that the effect interprets as the attack and release phases. Generally, you’ll find values of around 20 ms (attack) and 1500 ms (release) are fine to start with. Adjust them accordingly for the type of signal that you’re processing.

The lookahead slider allows you to define how far ahead in the signal the Enveloper looks to anticipate future events. Normally, you won’t need to use this feature, except possibly for signals with extremely sensitive transients. If you do raise the Lookahead value, you may need to adjust the attack time accordingly.

In contrast to a compressor or expander, the Enveloper operates independently of the absolute level of the input signal—provided the Threshold slider is set to the lowest possible value.
Expander
The Expander is similar to a compressor except that it increases, rather than reduces, the dynamic range above the Threshold level. You can use the Expander to add liveliness and freshness to your audio, specifically by emphasizing the transients of highly compressed signals.

Expander Parameters

- **Threshold slider and field**: Sets the level above which the Expander expands the signal.
- **Ratio slider and field**: Sets the ratio by which the signal is expanded when it exceeds the threshold.
- **Attack knob and field**: Sets the amount of time it takes for the expander to react when the signal exceeds the threshold.
- **Release knob and field**: Sets the amount of time it takes for the expander to stop expanding the signal when the signal falls below the threshold.
- **Knee knob and field**: Sets whether the signal is slightly expanded at levels just below the threshold.
- **Gain slider and field**: Sets the amount of output gain.
- **Auto Gain button**: When selected, Auto Gain compensates for the increase in gain produced by expansion.
- **Expansion display**: Shows the expansion curve applied to the signal.
- **Peak/RMS buttons**: Turn on one or the other to set whether the Expander uses the Peak or RMS method to analyze the signal.

Because the Expander is a genuine upward expander (as opposed to a downward expander that increases the dynamic range below the Threshold), the Ratio slider features a value range of $1:1$ to $0.5:1$.

When using the Expander with Auto Gain active, the signal will sound softer even when the peak level remains the same; in other words, the expander decreases loudness. If you dramatically change the dynamics of a signal (by setting higher Threshold and Ratio values), you may find that you need to reduce the output level using the Gain slider to avoid distortion. In most cases, turning on Auto Gain will adjust the signal to the correct level.
Limiter
The Limiter functions similarly to a compressor with one important difference: where a compressor proportionally reduces the signal when it exceeds the threshold, a limiter reduces any peak above the threshold to the threshold level, effectively limiting the signal to this level. The Limiter is used primarily as a mastering effect.

Limiter Parameters

- **Gain reduction meter:** Shows the amount of limiting while the signal plays.
- **Gain slider and field:** Sets the amount of gain applied to the input signal.
- **Lookahead slider and field:** Adjusts how far ahead (in milliseconds) the Limiter analyzes the audio signal.
- **Release slider and field:** Sets the amount of time after the signal falls below the threshold before the Limiter stops limiting.
- **Output Level knob and field:** Sets the output level of the signal.
- **Softknee button:** When selected, the signal is limited only when it reaches the threshold. When switched on, the transition to full limiting is nonlinear, producing a softer, less abrupt effect, and reducing distortion artifacts that can be produced by hard limiting.

The Lookahead parameter allows the Limiter to look forward in the audio so that it can react earlier to peak volumes by adjusting the amount of reduction. Using Lookahead causes latency, but this latency has no perceptible effect when you use the Limiter as a mastering effect, on previously recorded material. Set Lookahead to higher values if you want the limiting effect to take place before the maximum level is reached, creating a smoother transition.

Typically, you apply the Limiter as the very last effect in the mastering signal chain. In this case, you use the Limiter to raise the overall volume of the signal, so that it reaches but does not exceed 0 dB.

The Limiter is designed in such a way that if set to 0 dB Gain and 0 dB Output Level, it has no effect (on a normalized signal). If the signal clips (red gain line), the Limiter—using its basic settings—reduces the level before clipping can occur. (However, the Limiter cannot fix audio that was clipped during recording).
Multipressor

The Multipressor (short for multiband compressor) is an extremely versatile tool used in mastering audio. It splits the incoming signal into different frequency bands (from one to four), and allows you to apply compression to each band independently. After compression is applied, the bands are combined into a single output signal.

The advantage of compressing different frequency bands separately is to apply a higher amount of compression to the bands that need it without producing the pumping effect often associated with high amounts of compression. Using the Multipressor, you can apply higher compression ratios to specific frequency bands, and therefore achieve a higher average volume, without causing audible artifacts.

Raising the overall volume level can result in a dramatic increase of the existing noise floor. Each frequency band features downward expansion, which allows you to reduce or suppress this noise. Downward expansion works as a counterpart to compression: while the compressor compresses the dynamic range of the higher volume levels, the downward expander expands the dynamic range of the lower volume levels. With downward expansion, the signal is reduced in level when it falls below the Threshold level. This functions in a similar way as a noise gate, but rather than simply cutting off the sound, it smoothly fades the volume using an adjustable ratio.

Multipressor Parameters

The parameters in the Multipressor window are grouped into three main areas: the upper graphic display section, the lower set of controls for each frequency band, and the output parameters on the right.
Graphic Display Section
- Graphic display: Each frequency band is represented graphically. The amount of gain change from 0 dB is shown graphically by the blue bars. For active bands, the band number appears in the center of its area. You can adjust each frequency band independently in the following ways:
  - Drag the horizontal bar up or down to adjust the gain makeup for that band.
  - Drag the vertical edges left or right band to set the crossover frequencies for that band (which adjusts the band's frequency range).
- Crossover fields: Sets the crossover frequency between adjacent bands.
- Gain Make-up fields: Sets the amount of the gain makeup for each band.

Frequency Band Section
Below the graphic band display are value fields and other parameters for controlling each frequency band:
- Compr Thrsh fields (Compression Threshold): Sets the compression threshold for the selected band. Setting the parameter to 0 dB results in no compression of the band.
- Compr Ratio fields (Compression Ratio): Sets the compression ratio for the selected band. Setting the parameter to 1:1 results in no compression of the band.
- Expnd Thrsh fields (Expansion Threshold): Sets the expansion threshold for the selected band. Setting the parameter to its minimum value (-50 dB) means that only signals that fall below this level are expanded.
- Expnd Ratio fields (Expansion Ratio): Sets the expansion ratio for the selected band.
- Expnd Reduct fields (Expansion Reduction): Sets the amount of downward expansion for the selected band.
- Peak/RMS fields: Enter a smaller value for shorter peak detection, or a larger value for RMS detection (in milliseconds).
- Attack fields: Sets the amount of time (in milliseconds) that the signal exceeds the threshold before compression starts for the selected band.
- Release fields: Sets the time (in milliseconds) required after the signal falls below the threshold before compression stops on the selected band.
- Band on/off buttons: Each band has a button (numbered from 1 to 4). Click the button to turn on the band (the button becomes light blue and the band appears in the graphic display area above). Click the button again to turn off the band.
- Bypass buttons: Turn on to bypass the selected frequency band.
- Solo buttons: Turn on to solo the selected frequency band.
- Level meters: The light blue bar on the left shows the input level, and the dark blue bar on the right shows the output. Drag the upper triangular notch to adjust the Compression Threshold (Compr Thrsh), and drag the lower triangular notch to adjust the Expansion Threshold (Expnd Thrsh). Drag between the two notches to move both.
Output Parameters

- **Auto Gain pop-up menu:** Controls whether the Multipressor references the overall processing of the signal to 0 dB, making the output louder (On), or produces more standard compression, with compressed bands attenuated by the amount the dynamic range is reduced (Off).

- **Lookahead value field:** Adjusts how far the processor looks forward in the audio, in order to react earlier to peak volumes for smoother transitions.

- **Out Gain slider:** Sets the overall gain at the output.

- **Level meter:** Shows the overall output level.

Using the Multipressor

In the graphic display, the blue bars show the gain change, not only the gain reduction as in a standard compressor. The gain change displayed is a composite value of the compression reduction + expander reduction + auto gain compensation + gain make-up.

Compression Parameters

The Compression Threshold and Compression Ratio parameters are the key parameters used to control compression. In most cases, the most useful combinations of these two settings are either a low Compression Threshold with a low Compression Ratio, or a high Compression Threshold with a high Compression Ratio.

Downward Expansion Parameters

The Expansion Threshold, Expansion Ratio, and Expansion Reduction parameters are the key parameters for controlling downward expansion. They determine the strength of expansion applicable to the range that you want to expand.

Peak/RMS, Attack, and Release Parameters

Adjusting the parameter between Peak (0 ms, minimum value) and RMS (Root Meantime Square – 200 ms, maximum value) is dependent on the type of signal you would like to compress. An extremely short Peak detection setting is suitable for compression of short and high peaks of low power, which do not typically occur in music. The RMS detection method measures the power of the audio material over time and thus works much more musically. This is because human hearing is more responsive to the overall power of the signal than to single peaks. As a basic setting for most applications, the centered position is recommended.

Output Parameters

The Out Gain slider sets the overall output level. Set Lookahead to higher values when the Peak/RMS fields are set to higher values (further toward RMS). Setting Auto Gain to On references the overall processing to 0 dB, making the output louder.
Noise Gate
The Noise Gate is commonly used to suppress unwanted noise that is audible when the audio signal is at a low level. You can use it to remove background noise, crosstalk from other signal sources, and low-level hum.

The Noise Gate works by allowing signals above the Threshold level to pass unimpeded while reducing signals below the Threshold level. This allows you to remove lower-level parts of the signal, while allowing the intended parts of the audio to pass.

Noise Gate Parameters

Main Parameters
- **Threshold slider and field:** Sets the level (in decibels) below which the signal is reduced.
- **Reduction slider and field:** Sets the amount by which the signal is reduced.
- **Attack knob and field:** Sets the amount of time it takes to fully open the gate after the signal exceeds the threshold.
- **Hold knob and field:** Sets the amount of time the gate is kept open after the signal falls below the threshold.
- **Release knob and field:** Sets the amount of time it takes to fully close the gate after the signal falls below the threshold.
- **Hysteresis slider and field:** Sets the difference (in decibels) between the threshold values that open and close the gate, to prevent it rapidly opening and closing when the input signal is close to the threshold.
- **Lookahead slider and field:** Sets how far ahead (in milliseconds) the noise gate analyzes the signal.

Sidechain Parameters
- **Monitor button:** Turn on to preview the Sidechain signal, including the effect of the High and Low Cut filters.
- **High Cut slider and field:** Sets the upper cutoff frequency for the sidechain signal.
- **Low Cut slider and field:** Sets the lower cutoff frequency for the sidechain signal.

When no external sidechain is selected, the input signal is used as the sidechain.
Using the Noise Gate

In most situations, setting the Reduction slider to the lowest possible value ensures that sounds below the Threshold are completely suppressed. Setting it to a higher value attenuates low-level sounds but still allows them to pass. You can also set Reduction to a value greater than 0 (zero) to boost the signal by up to 20 dB. This is useful for ducking effects.

The three rotary knobs for Attack, Hold, and Release modify the dynamic response of the Noise Gate. If you want the gate to open extremely quickly, say for percussive signals such as drums, set the Attack knob to a lower value. For other sounds, such as string pads, where the signal fades in more gradually, set Attack to a higher value for a smoother effect. Similarly, when you are working with signals that fade out gradually or that have longer reverb tails, set the Release knob to a higher value so that the signal fades naturally.

The Hold knob determines the minimum amount of time that the gate stays open. This avoids abrupt changes (called chattering) caused when the Noise Gate opens and closes rapidly.

The Hysteresis slider provides another option for avoiding chattering, without needing to define a minimum Hold time. You use it to set the range between the threshold values that open and close the Noise Gate. This is useful when the signal level jitters around the Threshold, fluctuating slightly but rapidly around it. This causes the Noise Gate to switch on and off repeatedly, producing an undesirable chattering effect. Using the Hysteresis slider, you can set the Noise Gate to open at the Threshold level and remain open until the level drops below another, lower, level. As long as the difference between these two values is large enough to contain the fluctuating level of the incoming signal, the Noise Gate can function without creating chatter. This value is always negative. Generally, –6 dB is a good place to start.

In some situations, you may find that the levels of the signal you want to keep and the levels of the noise are close enough to be difficult to separate. For example, if you are recording a drum kit, and using the Noise Gate to isolate the sound of the snare drum, the hi-hat may also open the gate in many cases. To remedy this sort of situation, you can use the Sidechain controls to isolate the desired signal using Hi and Low Cut filters.

To use the Sidechain filters, click the Monitor button to turn on monitoring. This lets you hear how the Hi and Low Cut filters will affect the incoming signal. Now you can drag the High Cut slider to set the frequency above which the signal is filtered out, and drag the Low Cut slider to set the frequency below which the signal is filtered. These filters only allow very high (loud) signal peaks in their range to pass. In our example, you could remove the hi-hat’s signal, which is higher in frequency, using the Hi Cut filter, and allow the snare signal to pass. You can turn monitoring off to set a suitable Threshold level more easily.
Surround Compressor

The Surround Compressor, based on the Compressor, is specially adapted for compressing complete surround mixes. The Surround Compressor is especially useful when inserted in a surround output or on channels and busses that carry multichannel audio.

You can adjust the compression ratio, knee, attack, and release for both the main channels and for the LFE channel. Both the main channels and the LFE channel include an integrated limiter. In addition, you can set the threshold and output level for each channel independently.

You can also link channels by assigning them to one of three groups. When you adjust the threshold or output parameter for any channel assigned to a group, that parameter is adjusted by the same amount for all channels assigned to the group.

Surround Compressor Parameters

The Surround Compressor is divided into three sections: The Link section at the top contains a series of menus where you assign each channel to a group. The Main section includes controls common to all the main channels, and the threshold and output controls for each channel. The LFE section on the lower right includes separate controls for the LFE channel.

- **Circuit Type slider and field:** Choose the type of circuit emulated by the Compressor. The choices are Platinum, Classic A_R, Classic A_U, VCA, FET, and Opto (optical).
- **Detection pop-up menu:** Choose whether the Surround Compressor uses the maximum of all detection signals (Max) or the sum of all detection signals (Sum) to determine if it is above or below the threshold.
**Link Section**
- *Grp. (Group) pop-up menus:* For each channel, choose whether the channel is in group A, B, or C, or in no group (–). Moving the Threshold or Output Level slider for any channel assigned to a group will move the sliders for all channels assigned to that group.
- *Byp (Bypass) buttons:* For each channel, click to bypass that channel.

**Main Section**
- *Ratio slider and field:* Sets the ratio by which the signal is reduced when it exceeds the threshold.
- *Knee knob:* Adjusts whether the signal is compressed immediately or more gradually at levels close to the threshold.
- *Attack knob:* Sets the amount of time it takes to reach full compression after the signal exceeds the threshold.
- *Release knob:* Sets the amount of time it takes to return to zero compression after the signal falls below the threshold.
- *Auto button:* When selected, the release time dynamically adjusts to the audio material.
- *Limiter button:* Turns limiting for the main channels on or off.
- *Threshold knob:* Sets the threshold for the limiter on the main channels.
- *Main Compressor Threshold sliders:* Sets the threshold (the level above which the signal is reduced) for each channel.
- *Main Output Levels controls:* Sets the output level for each channel.

**LFE Section**
- *Ratio slider and field:* Sets the compression ratio for the LFE channel.
- *Knee knob:* Sets the knee for the LFE channel.
- *Attack knob:* Sets the attack time for the LFE channel.
- *Release knob:* Sets the release time for the LFE channel.
- *Auto button:* When selected, the release time automatically adjusts to the audio signal.
- *Limiter button:* Turns limiting for the LFE channel on or off.
- *Threshold knob:* Sets the threshold for the limiter on the LFE channel.
Using the Surround Compressor

Using the Link controls, you can assign each channel independently to one of three groups (Group A, B, or C). When you adjust the Threshold or Output Level slider for any channel in a group, the sliders for all channels in the same group are adjusted by the same amount. Also, clicking the Bypass button for any grouped channel bypasses all channels in the group.

You can temporarily unlink a channel by pressing and holding Command and Option while you move the Threshold or Output Level slider for that channel. When you move either slider without pressing Command-Option, the channels move in step, keeping their relative positions. This allows you to make independent threshold settings while maintaining the sidechain detection link necessary for a stable surround image.
Equalization (EQ) lets you shape the sound of your audio by changing the level of specific frequency bands.

EQ is one of the most commonly used audio effects, both for music projects and in post-production work for video. You can use EQ to shape the sound of an audio file, track, or project by adjusting specific frequencies or frequency ranges. Using EQ, you can create both subtle and extreme changes to the sound of your projects.

EQ effects include a variety of single-band filters and multiband EQs. All EQ effects use filters that allow certain frequencies to pass through unchanged, while raising or lowering the level of other frequencies (also referred to as boosting or cutting frequencies). EQs can be used as “broad brush” effects to boost or cut a large range of frequencies, and some EQs (particularly parametric and multiband EQs) can be used for more precision work.

**Single Band EQs**

The simplest types of EQ effects are single band EQs, which include low and high cut, low and high pass, shelving, and parametric EQ.

- **Low cut EQ** attenuates only frequencies below a specific frequency, called the cutoff frequency, by a fixed number of decibels per octave, called the slope. High cut EQ attenuates only frequencies above its cutoff frequency, by a fixed slope.
- **Low pass EQ** attenuates frequencies above the cutoff frequency, while high pass EQ lowers frequencies below the cutoff. In addition, you can control the slope of the filter (how gradually frequencies beyond the cutoff are attenuated) using the Order parameter.
- **High and low shelving EQ** lets you set the cutoff frequency and also control the gain (the amount of boost or cut), allowing you to change it by a fixed amount rather than a slope.
- **Parametric EQ** boosts or cuts all frequencies close to the center frequency (both above and below the center frequency). You can set the center frequency, and also the bandwidth or Q, which determines how wide a range of frequencies around the center frequency are altered.
Multiband EQs
Multiband EQs give you control over a set of filters that together cover a large part of the frequency spectrum. On multiband EQs, you can set the frequency, bandwidth, and Q of each band independently. Using a multiband EQ (such as the Channel EQ, Fat EQ, or Linear Phase EQ), you can perform extensive tone-shaping on any audio source. Multiband EQs are equally useful for shaping the sound of an individual track or an overall project mix.

The following sections describe the individual effects included with Soundtrack Pro.

- “Channel EQ” on page 60
- “Fat EQ” on page 64
- “Linear Phase EQ” on page 66
- “Match EQ” on page 67
- “Single Band EQs” on page 72
  - “High Cut and Low Cut Filter” on page 72
  - “High Pass and Low Pass Filter” on page 72
  - “High Shelving and Low Shelving EQ” on page 72
  - “Parametric EQ” on page 72

Channel EQ

The Channel EQ is a highly versatile multiband EQ. It gives you eight frequency bands, including low and highpass filters, low and high shelving filters, and four flexible parametric bands. It also features an integrated Fast Fourier Transform (FFT) Analyzer that you can use to view the frequency curve of the audio you want to modify, allowing you to see which parts of the frequency spectrum need to be boosted or cut.

You can use the Channel EQ in many ways: to shape the sound of individual tracks or audio files, or for tone-shaping on an overall project mix. With its Analyzer and graphic controls, it is very easy to observe the audio signal and make adjustments in real time.
Channel EQ Parameters
On the left side of the Channel EQ window is the Gain control and parameters for the Analyzer, while the central area of the window includes the graphic display and parameters for shaping each EQ band.

- **Master Gain slider and field:** Sets the output level of the signal. After boosting or cutting individual frequency bands, you can use the Master Gain fader to adjust the output level.
- **Analyzer button:** Turns the Analyzer on or off.
- **Pre/Post EQ button:** When Analyzer mode is active, sets whether the Analyzer shows the frequency curve before or after EQ is applied.
- **Resolution pop-up menu:** Choose the sample resolution for the Analyzer from the menu. The choices are: low (1024 points), medium (2048 points), and high (4096 points).

Graphic Display Section
- **Band On/Off buttons:** Located above the graphic display. Click a button to turn the corresponding band on or off. Each button has a icon showing the type of EQ it uses:
  - Band 1 is a highpass filter.
  - Band 2 is a low shelving filter.
  - Bands 3 through 6 are parametric bell filters.
  - Band 7 is a high shelving filter.
  - Band 8 is a lowpass filter.
- **Graphic display:** Shows the current curve of each EQ band. You can adjust the frequency of each band by dragging left or right in the section of the display for that band, and adjust the gain of each band (except bands 1 and 8) by dragging up or down in the band's section. The display reflects your changes immediately.

Parameter Section
Below the graphic display area are controls that you can use to show the settings for each band and adjust each band's settings.
- **Frequency fields:** Adjust the frequency of each band.
- **Gain/Slope fields:** Adjust the amount of gain for each band. For bands 1 and 8, this changes the slope of the filter.
- **Q fields:** Adjust the Q or resonance for each band (the range of frequencies around the center frequency that are affected).

The Q parameter of band 1 and band 8 has no effect when the slope is set to 6 dB/Oct. When the Q parameter of bands 3 through 6 is set to an extremely high value (such as 100), these filters only affect a very narrow frequency band, and can be used as notch filters.
• **Link button:** Activates Gain-Q coupling, which automatically adjusts the Q (bandwidth) when you raise or lower the gain on any EQ band, to preserve the perceived bandwidth of the bell curve.

Setting Gain-Q Couple to **strong** preserves the perceived bandwidth almost entirely, while **light** and **medium** settings allow some change as you raise or lower the gain. The asymmetric settings feature a stronger coupling for negative gain values than for positive values, so the perceived bandwidth is more closely preserved when you cut than when you boost gain.

**Note:** If you play back automation of the Q parameter with a different Gain-Q-Couple setting, the actual Q values will be different than when the automation was recorded.

**Using the Channel EQ**

How you use the Channel EQ depends on your audio and what you intend to do, but a useful workflow for many situations is as follows: with the Channel EQ set to a flat response (no frequencies boosted or cut), turn on the Analyzer and play the audio, observing the graphic display to see which parts of the frequency spectrum have frequent peaks and which stay at a low level. Notice particularly any places where the signal distorts or clips. Then, using the graphic display or the Parameter controls, adjust the frequency bands as desired to obtain the sound you want.

You can attenuate those frequencies that experience clipping to reduce or eliminate the distortion, and raise the quiet areas to make them sound more pronounced. You can adjust the center frequency for bands 2 through 7 to affect a specific frequency (either one you want to emphasize, such as the root note of the music, or one you want to eliminate, such as hum or other noise), and narrow the Q so that only a narrow range of frequencies are affected, or widen it to alter a broad area.

In the graphic display, each EQ band appears as a different color. You can graphically adjust the frequency of a band by dragging horizontally in the area of the band. Drag vertically to adjust the amount of gain for the band (For bands 1 and 8, the slope values can only be changed in the parameter area below the graphic display). Each band has a pivot point, which appears as a small circle on the curve at the location of the band’s frequency; you can adjust the Q or width of the band by dragging the pivot point vertically.

You can also adjust the decibel scale of the graphic display by vertically dragging either the left or right edge of the display (where the dB numbers appear) when the Analyzer is not active. When the Analyzer is active, dragging the left edge adjusts the linear dB scale, and dragging the right edge adjusts the Analyzer dB scale.
To increase the resolution of the EQ curve display in the most interesting area around the zero line, drag the dB scale on the left side of the graphic display upward. Drag downward to decrease the resolution. The overall range is always ±30, but small values are easier to recognize.

When you work with the Channel EQ, you can turn off any bands you are not using to shape the sound. Inactive bands do not use any computer resources.

**Using the Analyzer**

When you turn on the Analyzer, the Channel EQ shows a realtime curve of all frequency components of the signal as the audio plays, superimposed over the EQ curves you set, using a Fast Fourier Transformation (FFT). The Analyzer curve uses the same scale as the EQ curves, allowing you to easily recognize important frequencies in the audio and use the EQ curves to raise or lower them.

As soon as the Analyzer is activated, you can change the Analyzer Top parameter, which alters the scaling of the FFT Analyzer, on the right side of the graphic display. The visible area represents a dynamic range of 60 dB, but by dragging, you can adjust the maximum value between +20 dB and –40 dB. The Analyzer display is always dB-linear.

When choosing a resolution from the menu, keep in mind that the higher the resolution, the more CPU power is required. High resolution is necessary whenever you need reliable results in very low bass frequencies, for example. The bands derived from FFT analysis are divided in accordance with the frequency linear principle, meaning that there are more bands in higher octaves than in lower ones.

**Note:** The FFT Analyzer needs additional CPU resources. In fact, CPU usage increases significantly at higher resolutions. It is recommend that you disable the Analyzer or close the Channel EQ window when you play or record the project, after setting the desired EQ parameters. This will free up CPU resources for other tasks.
Fat EQ

The Fat EQ effect is a versatile multiband EQ with up to five individual frequency bands. You can use Fat EQ for individual tracks or for overall mixes. The Fat EQ includes a graphic display of the EQ curves and a set of parameters for each band.

Fat EQ Parameters
The main area of the Fat EQ window includes a graphic display area and a set of strips with parameters for each frequency band. To the right of the parameter section are the Master Gain slider and field.

Graphic Display Section
- **Band Type buttons**: Located above the graphic display. For bands 1-2 and 4-5, click a button in the button pair to select the type of EQ for the corresponding band.
  - For Band 1, click the highpass or the low shelving button.
  - For Band 2, click the low shelving or the parametric button.
  - Band 3 always acts as a parametric EQ band. (Click the button to turn it on or off.)
  - For Band 4, click the parametric or the high shelving filter.
  - For Band 5, click the high shelving or the lowpass button.
- **Graphic display**: Shows the EQ curve of each frequency band. When you adjust each band's settings using the controls in the Parameter section, the display reflects your changes immediately.
**Parameter Section**
Below the graphic display area are controls that you can use to show the settings for each band and adjust each band’s settings.

- **Frequency fields:** Sets the frequency for each band.
- **Gain knobs:** Sets the amount of gain for each band.
- **Q/Order fields:** Sets the Q or bandwidth for each band (the range of frequencies around the center frequency that are altered). For bands 1 and 5, this changes the slope of the filter.
- **Band On/Off buttons:** Click the numbered button to turn each band on or off. Inactive bands do not use your computer’s resources.

**Master Gain Section**
- **Master Gain slider and field:** Located to the right of the Parameter section. Sets the output level of the signal. After boosting or cutting frequency bands, you can use the Master Gain fader to adjust the output level.

**Using the Fat EQ**
The icons above the graphic display let you switch the type of EQ for each band, except for Band 3, which always operates as a fully parametric bell filter. You can use the controls in the Parameter section to set the frequency, gain, and Q for each band, as well as turn individual bands on or off.

At low Q values, the EQ covers a wider frequency range, while at high Q values, the effect of the EQ band is limited to a very narrow frequency range. Keep in mind that the Q value can significantly influence how audible your changes are: if you’re working with a narrow frequency band, you’ll generally need to cut or boost it more drastically to notice the difference.
Linear Phase EQ

The high-quality Linear Phase EQ effect is similar in appearance to the Channel EQ, sharing the same parameters and eight-band layout. However, the Linear Phase EQ uses a different underlying technology, which preserves the phase of the audio signal 100%—even when you apply the wildest EQ curves to the sharpest signal transients!

The Linear Phase EQ uses more CPU resources than the Channel EQ, and introduces greater amounts of latency. For this reason, it is strongly recommended that you use it for mastering previously recorded audio—and don’t use it when playing software instruments live, for example.

**Linear Phase EQ Parameters**

The parameters of the Linear EQ are identical to the Channel EQ. For information on the Channel EQ parameters, see “Channel EQ Parameters” on page 61.

**Using the Linear Phase EQ**

In operation, the Linear Phase EQ is similar to the Channel EQ. For more information, refer to the section “Channel EQ” on page 60. Because the parameters of the Channel EQ and Linear Phase EQ are almost identical, you may freely copy settings between them.

One difference, however, is that the Linear Phase EQ uses a set amount of your computer’s CPU resources, regardless of how many bands are active.
Match EQ

The Match EQ effect allows you to store the average frequency spectrum of an audio file as a template and apply the template to your project, so that it matches the spectrum of the original file. Using Match EQ you can acoustically match the sound of different songs you plan to include on an album, or impart the particular sound of any source recording to your own projects. Beyond matching the project’s frequency spectrum to the original file’s EQ, you can also manually modify the filter curve before you apply it to your project.

*Note:* Match EQ acoustically matches the frequency curve of two audio signals. However, it does not match any dynamic differences in the two signals.

**Match EQ Parameters**

![Match EQ Parameters](image)

### Left Side

- **Analyzer button:** Turns the Analyzer function on or off.
  
  *Note:* Deactivating the Analyzer frees up processing power for other applications.

- **Position button:** Sets whether the Analyzer looks at the signal before the filter curve is applied (Pre) or after (Post).

- **View pop-up menu:** Choose what information appears on the graphic display. The choices are:
  - **Automatic:** Automatically displays information for the current function, as determined by which of the buttons below the graphic display is selected.
  - **Template:** Displays the frequency curve for the source file learned as the template (shown in red).
  - **Current Material:** Displays the frequency curve for the audio learned as the current material (the track on which the Match EQ is applied, or a loaded plug-in settings file or template—shown in green).
  - **Filter:** Displays the filter curve created by matching the template and the current material (shown in yellow).
• **Format button:** Sets whether the Analyzer displays audio channels via separate curves (L&R for stereo, All Cha for surround) or the summed maximum level (LR Max for stereo, Cha Max for surround).

• **Select buttons:** Click one of the buttons to control whether any changes you make to the filter curve created by matching the template with the current material are applied only to the left channel (L), the right channel (R), or both channels (L+R).

• **Select menu (not pictured):** When using a surround instance of the Match EQ, the Select buttons are replaced by a menu, from which you can choose an individual channel for changes to the filter curve, or choose All.

• **Channel Link slider:** Refines the settings made using the Select buttons or Select menu. When set to 1.0, all channels (L and R for stereo, or all channels for surround) are represented by a common EQ curve. When set to 0.0 (the minimum value), a separate filter curve is displayed for each channel, which can be selected using the Select buttons or Select menu. Settings between 0.0 and 1.0 allow you to blend these values with your changes to the filter curves being transferred to each channel, as determined by the setting.

**Note:** The View, Select, and Channel Link parameters are disabled when using the effect on a mono channel.

**Center Section**

• **Graphic display:** Displays the filter curve created by matching the template to the current material. You can edit the filter curve (see “Editing the Filter Curve”).

• **Template Learn button:** Click to start the process of learning the frequency spectrum of the source file. Click again to stop the learning process.

• **Current Material Learn button:** Click to start the process of learning the frequency spectrum of the project you want to match the source file. Click again to stop the learning process.

• **Current Material Match button:** Matches the frequency spectrum of the current material to that of the template (source) file.

**Right Side**

• **Phase button:** Sets whether processing alters (Minimal) or is prevented from altering the signal phase (Linear). The Linear setting increases latency, while the Minimal setting results in lower latency.

• **Apply slider and field:** Modifies the effect of the filter curve on the signal. Values between 101% and 200% magnify the effect, values between 1% to 99% reduce it, and values from –1% to –100% invert the peaks and troughs in the filter curve. A value of 100% produces no modification of the filter curve.
• **Smoothing slider and field:** Sets the amount of smoothing for the filter curve. A value of 0.0 leaves the filter curve unchanged. At all other settings, the filter curve is smoothed at a constant bandwidth, determined by the set value in semitones. For example, a value of 1.0 means that the smoothing uses a bandwidth of one semitone. A value of 4.0 produces a smoothing bandwidth of four semitones (a major third), a value of 12.0 produces a bandwidth of one octave, and so on.

*Note: Smoothing does not affect any manual changes you make to the filter curve.*

**Using the Match EQ**

Match EQ is a learning equalizer that analyzes or learns the frequency spectrum of an audio signal, such as an audio file, a track input signal, or a template. You can also load a previously saved plug-in settings file, or import the settings of another Match EQ instance by copying and pasting.

Match EQ analyzes the average frequency spectrum of the source file (the template) and of your project (the current material), then matches the two spectra, creating a filter curve. This filter curve adapts the frequency response of the current material to match that of the template. Before applying the filter curve, you can modify it by boosting or cutting any number of frequencies, or by inverting the curve. The Analyzer allows you to visually compare the frequency spectrum of the source file and the resulting curve, making it easier to make manual corrections at specific points within the spectrum.

You can use the Match EQ in different ways, depending on your intent and the audio you're working with. In general, you will want to make your mix sound similar to an existing recording—either your own or that of another artist. Following is a common usage example that you can adapt to your own workflow. In this example, you match the frequency spectrum of a mix to the spectrum of a source audio file.

**To match the EQ of a project mix to the EQ of a source audio file:**

1. In the project you want to match to the source audio file, instantiate a Match EQ (typically on Output 1-2).
2. Drag the source audio file onto the Template Learn button.
3. Return to the start of your mix, click Current Material Learn, then play your mix (the current material) from start to finish.
4. When done, click Current Material Match (this automatically disengages the Current Material Learn button).
To use the matched EQ on a track:

1. Set the track you want to match as a sidechain input to the Match EQ.
2. Click the “Template Learn” button, play the entire source audio track from start to finish, then click the “Template Learn” button again.
3. Return to the start of your mix, click Current Material Learn, then play your mix (the current material) from start to finish.
4. When done, click Current Material Match (this automatically disengages the Current Material Learn button).

Match EQ creates a filter curve based on the differences between the spectrum of the template and the current material. This curve automatically compensates for differences in gain between the template and the current material, with the resulting EQ curve referenced to 0 dB. A yellow filter response curve appears in the graphic display, showing the average spectrum of your mix approximating (mirroring) the average spectrum of your source audio file.

You can also drag an audio file onto the Template Learn or Current Material Learn buttons to use as either the template or the current material. A progress bar appears while the Match EQ is analyzing the file.

Control-clicking (or right-clicking, with an appropriate mouse) either of the Learn buttons opens a shortcut menu containing a variety of options to apply to the spectrum of the template or the current material, including: clearing, copying to the Match EQ clipboard (which you can access from any Match EQ instance in the current project), pasting from the Match EQ clipboard to the current instance, loading from a stored settings file, or generating a frequency spectrum for an audio file you choose in a File dialog.

When you click either of the Learn buttons, the View parameter is set to Automatic, and the graphic display shows the frequency curve for the function matching the selected button. You can review any of the frequency curves when no file is being processed by choosing one of the other View options.

The filter curve is updated automatically each time a new template or current material spectrum is learned or loaded, when the Match button is enabled. You can alternate between the matched (and possibly scaled and/or manually modified) filter curve and a flat response by activating/deactivating the Match button.
Only one of the Learn buttons can be active at a time. For example, if the Learn button in the Template section is active and you press the Learn button in the Current Material section, the analysis of the template file stops, the current status is used as the spectral template, and analysis of the track (Current Material) begins.

**Note:** Each time you match two audio signals—either by loading/learning a new spectrum while Match is activated or by activating Match after a new spectrum has been loaded—any existing changes to the filter curve are discarded, and Apply is set to 100%.

By default, the Apply slider is set to 100% when you learn the frequency curve of an audio signal. In many cases, you may want to lower it slightly to avoid extreme spectral changes to your mix. It is also recommended that you use the Smoothing slider in order to adjust the spectral detail of the generated EQ curve.

**Editing the Filter Curve**

You can graphically edit the matched filter curve in the graphic display by clicking any point on the filter curve. Drag horizontally to shift the peak frequency for this band (over the entire spectrum). Drag vertically to adjust the gain of this band (between –24 to +24 dB). To adjust the Q-Factor, hold down the Shift key and drag vertically. Hold down the Option key while dragging to reset the gain to 0 dB. As you drag, the current values appear in a small box inside the graphic display, allowing you to make precise adjustments graphically.

**Note:** If you manually modify the filter curve, you can restore it to the original (or flat) curve by Option-clicking on the background of the Analyzer display. Option-clicking the background again restores the most recently modified curve.

The Q-factor of the filter is set by the vertical distance between the point where you click and the curve. By clicking on the curve, the maximum Q-value of 10 (for notch-like filters) is used. Clicking above or below the curve decreases the Q-value. The further you click from the curve, the smaller the value (down to the minimum of 0.3).

The colors and modes of the dB scales on the left and right of the display are automatically adapted to the active function. If the Analyzer is active, the left scale displays the average spectrum in the signal, while the right scale serves as a reference for the peak values of the Analyzer. Basically, the Analyzer visualizes a dynamic range of 60 dB. The displayed range can, however, be shifted between the extreme values of +20 dB and –100 dB by dragging on the scale.

If the resulting filter curve is displayed, the left scale—and the right, if the Analyzer is inactive—shows the dB values for the filter curve in an appropriate color. By click-dragging on one of the scales, the overall gain of the filter curve is adjusted in the range from –30 to +30 dB.
Single Band EQs
Following are descriptions of each of the effects found in the Single Band submenu.

High Cut and Low Cut Filter
As their names suggest, the Low Cut Filter attenuates the frequency range below the selected frequency, while the High Cut Filter attenuates the frequency range above the selected frequency. Each has a single parameter letting you set the cutoff frequency.

High Pass and Low Pass Filter
The High Pass Filter affects the frequency range below the set frequency. Higher frequencies pass through the filter. You can use the High Pass Filter to eliminate the bass below a selectable frequency. In contrast, the Low Pass Filter affects the frequency range above the selected frequency. Both filter plug-ins offer the following parameters:
- Frequency field and slider: Sets the cutoff frequency.
- Order field and slider: Sets the filter order.
- Smoothing field and slider: Adjusts the amount of smoothing (in milliseconds).

High Shelving and Low Shelving EQ
The Low Shelving EQ affects only the frequency range below the selected frequency, while the High Shelving EQ affects only the frequency range above the selected frequency. Each has parameters for Gain, which you use to boost or cut the level of the selected frequency band, and Frequency, which you use to set the cutoff frequency.

Parametric EQ
The Parametric EQ is a simple filter with a variable center frequency. It can be used to boost or cut any frequency band in the audio spectrum, either with a wide frequency range, or as a notch filter with a very narrow range. A symmetrical frequency range on either side of the center frequency is boosted or cut. The Parametric EQ offers the following parameters:
- Gain field and slider: Sets the amount of gain.
- Frequency field and slider: Sets the cutoff frequency.
- Q-Factor field and slider: Adjusts the Q (bandwidth).
Frequency Ranges Used With EQ

All sounds can be thought of as falling into one of three basic frequency ranges: bass, midrange, or high (treble). These can each be further divided to include low bass, low and high midrange, and low and high highs. The following table describes some of the sounds that fall into each range:

<table>
<thead>
<tr>
<th>Name</th>
<th>Frequency range</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>High High</td>
<td>8–20 kHz</td>
<td>Includes cymbal sounds and highest harmonics of instruments. Boosting frequencies in this range slightly can add sparkle and presence.</td>
</tr>
<tr>
<td>High</td>
<td>5–8 kHz</td>
<td>This range corresponds roughly to the treble tone control on a stereo. Boosting frequencies in this range can add brightness and shine.</td>
</tr>
<tr>
<td>Low High</td>
<td>2.5–5 kHz</td>
<td>Includes the higher harmonics of voices and musical instruments. This range is important for adding presence. Excessive boosting in this range can sound shrill or harsh.</td>
</tr>
<tr>
<td>High Midrange</td>
<td>1.2–2.5 kHz</td>
<td>Includes the consonants of voices and the high harmonics of musical instruments, especially brass instruments. Excessive boosting in this range can create a pinched, nasal sound.</td>
</tr>
<tr>
<td>Midrange</td>
<td>750 Hz–1.2 kHz</td>
<td>Includes the vowels of voices and the harmonics of musical instruments that create tone color.</td>
</tr>
<tr>
<td>Low Midrange</td>
<td>250–750 Hz</td>
<td>Includes the fundamentals and lower harmonics of voices and musical instruments; careful EQing of each can keep them from competing. Excessive boosting in this range can result in muddy and unclear audio; excessive cutting can produce thin-sounding audio.</td>
</tr>
<tr>
<td>Bass</td>
<td>50–250 Hz</td>
<td>Corresponds roughly to the bass tone control on a stereo. Includes the fundamental frequencies of voices and of musical instruments. Excessive boosting in this range can sound boomy and thick.</td>
</tr>
<tr>
<td>Low Bass</td>
<td>50 Hz and below</td>
<td>Also called sub bass. Very little of the sound of voices or musical instruments falls in this range. Many sound effects used in movies, such as explosions and earthquakes, fall in this range.</td>
</tr>
</tbody>
</table>

Note: The frequencies shown for each range are approximate. Any division of sound into frequency ranges is somewhat arbitrary, and is meant only to give a general indication of each range.
In addition to the filters in EQ effects, you can use filters to change the character of your audio in both familiar and unusual ways.

The Filter submenu contains a variety of filter-based effects that you can use to creatively modify your audio, including autofilters, filter banks, vocoders, wah-wah effects, and a gate that uses frequency rather than the amplitude (volume) as the criteria for which part of the signal is allowed to pass through.

The following sections describe the individual plug-ins included with Soundtrack Pro.

- “AutoFilter” on page 76
- “Spectral Gate” on page 80
- “Soundtrack Pro Autofilter” on page 82
AutoFilter

The AutoFilter is a versatile filter effect with several unique features. You can use it to create classic, analog-style synthesizer effects, or as a tool for creative sound design. The filter cutoff can be dynamically modulated using both a synthesizer-style ADSR envelope and a low frequency oscillator (LFO). In addition, you can choose between different filter types and slopes; control the amount of resonance; add distortion for more aggressive sounds; and mix the original, dry signal with the processed signal.

AutoFilter Parameters

The main areas of the AutoFilter window include the Envelope, LFO, Filter, and Distortion sections. The overall Threshold control is in the upper-left corner, and the Output controls are on the right side of the window.

Threshold Slider

The Threshold slider sets the cutoff frequency that applies to both the envelope and LFO. When the input signal level exceeds the Threshold level, the envelope and LFO are retriggered. The Threshold parameter always applies to the envelope. It applies to the LFO only if the Retrigger button is selected.

Envelope Section

- *Attack knob and field:* Sets the attack time for the envelope.
- *Decay knob and field:* Sets the decay time for the envelope.
- *Sustain knob and field:* Sets the sustain time for the envelope.
- *Release knob and field:* Sets the release time for the envelope.
- *Dynamic knob and field:* Sets the amount by which the input signal modulates the peak value of the envelope.
- *Cutoff Mod. slider and field:* Sets the intensity of the control signal’s effect on the cutoff frequency.
LFO Section
- **Coarse and Fine Rate knobs and field:** Use together to set the frequency of the LFO. Drag the Coarse slider to set the LFO frequency in Hertz, then drag the Fine slider to fine tune the frequency in 1/000s of a Hertz.
- **Beat Sync button:** When selected, the LFO is synchronized to the sequencer’s tempo.
- **Phase knob:** Lets you shift the phase relationship between the LFO and the sequencer when Beat Sync is active.
- **Decay/Delay knob and field:** Sets the amount of time the LFO takes to go from 0 to its maximum value.
- **Rate Mod. knob and field:** Sets the rate of modulation for the LFO frequency, independent of the input signal level. When the input signal exceeds the Threshold, the modulation width of the LFO increases from 0 to the Rate Mod. value.
- **Stereo Phase knob and field:** For stereo instances of the AutoFilter, sets the phase relationships of the LFO modulations on the two stereo channels.
- **Cutoff Mod. slider and field:** Sets the intensity of the control signal’s effect on the cutoff frequency.
- **Retrigger button:** When selected, the waveform starts at 0° as soon as the Threshold is exceeded.
- **Waveform buttons:** Click one of the buttons to set the shape of the LFO waveform.
- **Pulse Width slider and field:** Lets you shape the curve of the selected waveform.

Filter Section
- **Cutoff Freq. knob:** Sets the cutoff frequency for the lowpass filter.
- **Resonance knob:** Sets the width of the frequency band around the cutoff frequency that is emphasized.
- **Fatness slider and field:** Adjusts the amount of fatness (low-frequency boost). When you set Fatness to its maximum value, adjusting Resonance has no effect on frequencies below the cutoff frequency.
- **State Variable Filter buttons:** Click one of the buttons to set whether the filter is a highpass (HP), bandpass (BP), or lowpass (LP) filter.
- **4-Pole Lowpass Filter buttons:** Click one of the buttons to set the slope of the lowpass filter to 6, 12, 18, or 24 dB per octave.

Distortion Section
- **Input knob:** Sets the amount of distortion applied before the filter section.
- **Output knob:** Sets the amount of distortion applied after the filter section.

Output Section
- **Dry Signal slider and field:** Sets the amount of the original (dry) signal added to the filtered signal.
- **Main Out slider and field:** Sets the final output volume of the AutoFilter.
Using the AutoFilter
The following section provides additional information on using the parameters in the AutoFilter window.

Filter Parameters
The most important parameters are located on the right side of the AutoFilter window. The Filter Cutoff knob determines the point where the filter kicks in. Higher frequencies are attenuated, while lower frequencies are allowed to pass through.

The Resonance knob controls how much frequencies around the cutoff frequency are emphasized. When you turn Resonance up sufficiently, the filter itself begins oscillating at the cutoff frequency. Self-oscillation begins before you max out the Resonance parameter, just like the filters on a Minimoog synthesizer. Increasing Resonance causes the lowpass filter to cut the bottom end, making the signal sound thinner. You can compensate for this thinness using the Fatness slider.

Both the envelope and LFO parameters are used to dynamically modulate the cutoff frequency. The Threshold parameter at the upper-left corner of the AutoFilter window applies to both sections, and analyzes the level of the input signal. If the input signal level exceeds the Threshold level, the envelope and LFO are retriggered.

Envelope Parameters
When the input signal exceeds the Threshold level, the control signal is triggered at the minimum value. Over the period of time determined by the Attack parameter, the signal reaches its maximum level. It then decreases for the period of time defined by the Decay value, and then stays at a constant level for the duration of the Sustain value. Once the signal level drops below the Threshold value, it decreases to its minimum value over the time period determined by the Release parameter. If the input signal falls below the Threshold level before the control signal has reached the Sustain level, the Release phase is triggered. You can modulate the peak value of the Envelope section using the level of the input signal by adjusting the Dynamic parameter. The Cutoff Mod. slider determines the intensity of the control signal’s effect on the cutoff frequency.
LFO Parameters
You set the waveform of the LFO by clicking one of the Waveform buttons. The choices are: descending sawtooth (saw down), ascending sawtooth (saw up), triangle, pulse wave, or random (random values, Sample & Hold). Once you select a waveform, you can shape the curve with the Pulsewidth slider. Use the Coarse and Fine Frequency knobs to set the LFO frequency. The Rate Mod. (Rate Modulation) knob controls modulation of the LFO frequency independent of the input signal level. If the input signal exceeds the Threshold level, the modulation width of the LFO increases from 0 to the Rate Mod. value. You can also define the amount of time this process takes by entering the desired value with the Decay/Delay knob. If the Retrigger button is turned on, the waveform starts at 0° whenever the Threshold is exceeded. For stereo instances of the AutoFilter, you can control the phase relationships of the LFO modulations on the two stereo sides with the Stereo Phase knob.

Turning on Beat Sync synchronizes the LFO to the sequencer’s tempo. The speed values include bar values, triplet values, and more. These are determined by the Rate knob next to the Beat Sync button. Use Sync Phase to shift the phase relationship between the LFO and the sequencer.

Distortion Parameters
The Distortion Input and Output parameters let you individually control pre-input and post-output distortion. Although the two distortion modules work in an identical way, their respective positions in the signal chain—before and after the filter, respectively—result in remarkably different sounds.

Output Parameters
The Dry Signal parameter sets the level ratio of the non-effected (dry) signal mixed with the processed signal. The Main Out parameter can lower the output volume by as much as 50 dB, allowing you to compensate for higher levels caused by adding distortion or other processing.
Spectral Gate
The Spectral Gate separates the signal above and below the Threshold level into two independent frequency ranges that you can modulate separately. It can produce some unusual and rich filtering effects.

Spectral Gate Parameters

- **Threshold slider and field**: Sets the threshold level at which the frequency band defined by the Center Freq. and Bandwidth parameters is divided into upper and lower frequency ranges.
- **Speed slider and field**: Sets the modulation frequency for the defined frequency band.
- **CF (Center Frequency) Modulation slider and field**: Sets the intensity of center frequency modulation.
- **BW (Band Width) Modulation slider and field**: Sets the amount of bandwidth modulation.
- **Graphic display**: Shows the frequency band defined by the Center Freq. and Bandwidth parameters.
- **Center Freq. (Frequency) knob and field**: Sets the center frequency of the frequency band to be processed by the Spectral Gate.
- **Bandwidth knob and field**: Sets the bandwidth of the frequency band to be processed by the Spectral Gate.
- **Low Level slider and field**: Blends the frequencies of the original signal below the selected frequency band with the processed signal.
- **Super Energy knob and field**: Controls the level of the frequency range above the threshold.
- **Sub Energy and field**: Controls the level of the frequency range below the threshold.
- **High Level slider and field**: Blends the frequencies of the original signal above the selected frequency band with the processed signal.
- **Gain slider and field**: Adjusts the amount of gain for the final output signal.
Using the Spectral Gate
Using the Center Freq. and Bandwidth parameters, set the frequency band you want to process using the Spectral Gate. The graphic display visually indicates the band defined by these two parameters.

Once the frequency band is defined, use the Threshold parameter to set the level above and below which the frequency band is divided into upper and lower ranges. Use the Super Energy knob to control the level of the frequencies above the Threshold, and use the Sub Energy knob to control the level of the frequencies below the Threshold.

You can also mix the frequencies from the original signal outside the frequency band defined by the Center Freq. and Bandwidth with the processed signal. Use the Low Level slider to blend the bass frequencies below the defined frequency band with the processed signal, and use the High Level slider to blend in frequencies above the defined frequency band.

You can modulate the defined frequency band using the Speed, CF Modulation, and BW Modulation parameters. Speed determines the modulation frequency, CF (Center Frequency) Modulation defines the intensity of the center frequency modulation, and BW (Band Width) Modulation controls the bandwidth modulation.

After making your adjustments, you can use the Gain slider to adjust the final output level of the processed signal.

One way to get better acquainted with the operation of the Spectral Gate would be to start with a drum loop. Set the Center Freq. to its minimum (20 Hz) and the Bandwidth to its maximum (20000 Hz) value (so that the entire frequency range is processed). Turn up the Super Energy and Sub Energy knobs, one at a time, then try different Threshold settings. This should give you a good sense of how different Threshold levels affect the sound of Super Energy and Sub Energy. When you come across a sound that you like or consider useful, narrow the Bandwidth drastically, gradually increase the Center Freq., and then use the Low Level and High Level sliders to mix in some treble and bass from the original signal. At lower Speed settings, turn up the CF Mod. or BW Mod. knobs.
Soundtrack Pro Autofilter
The Soundtrack Pro Autofilter is a simple lowpass resonance filter that offers the following parameters:

- **Cutoff Frequency slider and field:** Sets the cutoff frequency for the lowpass filter.
- **Resonance slider and field:** Adjusts the amount of emphasis in the frequency band around the cutoff frequency.
You can use the Soundtrack Pro Imaging plug-ins to extend the stereo base of a recording, and to alter perceived signal positions.

These effects enable you to make certain sounds, or the overall mix, seem wider and more spacious. You can also alter the phase of individual sounds within a mix, to enhance or suppress particular transients.

The following sections describe the Imaging plug-ins included with Soundtrack Pro:
- “Direction Mixer” on page 83
- “Stereo Spread” on page 86

Direction Mixer
You can use the Direction Mixer plug-in to decode middle and side (MS) audio recordings (see “What is MS?” on page 85), or to spread the stereo base of a left/right recording, and determine its pan position.

- **Input buttons:** Use the LR or MS buttons to determine whether the input signal is a standard left/right signal, or if you’re dealing with an MS encoded (middle and side) signal.
- **Spread slider and field:** Determines the spread of the stereo base.
- **Direction knob and field:** Determines the direction from which the middle of the recorded stereo signal will emanate from within the mix, or in less complicated terms, its pan position.
Using the Direction Mixer

The Direction Mixer is a simple plug-in to use, as it only offers two parameters: Spread and Direction. Each alters the incoming signal differently when either the LR or MS Input buttons are active.

Using the Spread Parameter on LR Input Signals

At a neutral value of 1, the left side of the signal is positioned precisely on the left, and the right side precisely on the right. As you decrease the Spread value, the two sides move toward the center of the stereo image. A value of 0 produces a mono signal (both sides of the input signal are routed to the two outputs at the same level—a true middle signal). At values greater than 1, the stereo base is extended out to an imaginary point beyond the spatial limits of the speakers.

Note: If simply using the Direction Mixer to spread the stereo base, monaural compatibility decreases with Spread values above 1. After a stereo signal has been processed at an extreme Spread setting of 2, the signal will be canceled out completely if played back in mono.

Using the Spread Parameter on MS Input Signals:

When you alter MS levels with the Spread parameter (above a value of 1), the level of the side signal becomes higher than that of the middle signal. At a value of 2, you will only hear the side signal (on the left, you’ll hear L-R and on the right, R-L).

Setting the Direction Parameter

When Direction is set to a value of 0, the middle of the stereo recording will be dead center within the mix. If you use positive values, the midpoint of the stereo recording is moved toward the left. Negative values move the midpoint to the right. Here’s how this works:

- At 90˚, the midpoint of the stereo recording is panned hard left.
- At –90˚, the midpoint of the stereo recording is panned hard right.
- Higher values move the midpoint back toward the center of the stereo mix, but this also has the effect of swapping the stereo sides of the recording. For example, at values of 180˚ or –180˚, the midpoint of the recording is dead center in the mix, but the left and right sides of the recording are swapped.
What is MS?
Relegated to obscurity for a good long while, MS stereo (middle-side as opposed to left-right) has recently enjoyed a renaissance of sorts.

Making a Middle Side Recording
Two microphones are positioned as closely together as possible (usually on a stand or hung from the studio ceiling). One is a cardioid (or omnidirectional) microphone that directly faces the sound source that you want to record—in a straight alignment. The other is a bidirectional microphone, with its axes pointing to the left and right (of the sound source) at 90° angles.

• The cardioid microphone records the middle signal to the left side of a stereo track.
• The bidirectional microphone records the side signal to the right side of a stereo track.

MS recordings made in this way can be decoded by the Direction Mixer.

Why Make MS Recordings?
The advantage that MS recordings have over XY recordings (two cardioid microphones that are directed to a point halfway to the left and right of the sound source) is that the stereo middle is actually located on the on-axis (main recording direction) of the cardioid microphone. This means that slight fluctuations in frequency response that occur off the on-axis—as is the case with every microphone—are less troublesome.

In principle, MS and LR signals are equivalent, and can be converted at any time. When “–” signifies a phase inversion, then the following applies:

\[
M = L + R \\
S = L - R
\]

In addition, L can also be derived from the sum of—and R, from the difference between—M and S.

Here’s some interesting trivia for you: Radio (FM) broadcasts feature M and S stereo. The MS signal is actually converted to a signal suitable for the left and right speakers by the receiver.
Stereo Spread

The Stereo Spread effect is typically used for mastering. There are several ways to extend the stereo base (or perception of space), including use of reverbs and other effects and altering the signal's phase. They can all sound great, but can also weaken the overall sound of your mix by ruining transient responses, for example.

The Stereo Spread plug-in extends the stereo base by distributing a selectable number of frequency bands from the middle frequency range to the left and right channels. This is done alternately—middle frequencies to the left channel, middle frequencies to the right channel, and so on. This greatly increases the perception of stereo width without making the sound totally unnatural, especially when used on mono recordings.

Stereo Spread Parameters

- **Lower Int. slider and field:** Sets the amount of stereo base extension for the lower frequency bands.
- **Upper Int. slider and field:** Sets the amount of stereo base extension for the upper frequency bands.

A point to note when you are setting the Lower and Upper Int. sliders is that the stereo effect is most apparent in the middle and higher frequencies, and that distributing low frequencies between the left and right speakers significantly reduces the energy from both speakers. For this reason, you should use a lower Lower Int. setting, and avoid setting the Lower Freq. below 300 Hz.

- **Graphic display:** Shows the number of bands the signal is divided into, and the intensity of the Stereo Spread effect in the upper and lower frequency bands. The upper section represents the left channel, and the lower section represents the right channel. The frequency scale displays frequencies in ascending order, from left to right.
- **Upper and Lower Freq. slider and fields:** Use these to determine the upper and lower limits of the highest frequency, and lowest band, to be distributed in the stereo image.
- **Order knob:** Sets the number of frequency bands that the signal is divided into. A value of 8n is usually sufficient for most tasks, but you can use up to 12 bands.
You can use the Metering plug-ins of Soundtrack Pro to analyze audio in a variety of ways.

Each Metering plug-in allows you to view different characteristics of an audio signal. For example: The BPM Counter displays the pitch of a note, the Correlation Meter displays the phase relationship, and the Level Meter displays the level of an audio recording.

This chapter describes the Metering plug-ins included with Soundtrack Pro:

- “Correlation Meter” on page 87
- “MultiMeter” on page 88
- “Surround MultiMeter” on page 91
- “Tuner” on page 92

Correlation Meter

The Correlation Meter displays the phase relationship of a stereo signal.

- A correlation of +1 (plus one, the far right position) means that the left and right channels correlate 100% (they are completely in-phase).
- A correlation of 0 (zero, the center position) indicates the widest permissible left/right divergence, often audible as an extremely wide stereo effect.
- Correlation values lower than zero indicate that out-of-phase material is present, which can lead to phase cancelations if the stereo signal is combined into a monaural signal.
MultiMeter

The MultiMeter provides a collection of professional gauge and analysis tools in a single window. It includes:

- An Analyzer to view the level of each 1/3-octave frequency band.
- A Goniometer for judging the phase coherency in the stereo sound field.
- A Correlation Meter to spot mono phase compatibility.
- An integrated Level Meter to view the signal level for each channel.

There is also a surround version of the MultiMeter, with parameters for each channel and a slightly different layout. For more information on the Surround MultiMeter, see “Surround MultiMeter” on page 91.

MultiMeter Parameters

You can view either the Analyzer or Goniometer in the main display area. You switch the view and set other MultiMeter parameters using the controls on the left side of the window. The Phase Correlation meter is always visible at the bottom of the window, and the Level Meters are visible on the right.

Analyzer Section

- **Analyzer button:** When selected, displays the Analyzer in the center of the window.
- **Left, Right, LRmax, and Mono buttons:** Sets which channels are displayed in the Analyzer. Selecting Left or Right display only those channels, respectively, selecting LRMax displays the maximum level of the stereo inputs, and selecting Mono displays the spectrum of the mono sum of both (stereo) inputs.
- **Mode buttons:** Click one of the buttons to set whether levels are displayed using Peak, Slow RMS, or Fast RMS characteristics.
Goniometer Section

- Goniometer button: When selected, displays the Goniometer in the center of the window.
- Auto Gain field (display only): Sets the amount by which the display compensates for low input levels. You can set Auto Gain levels in 10% increments, or set it to off.
- Decay field: Sets the amount of time it takes for the Goniometer trace to fade to black.

Peak Section

- Hold button: When selected, activates peak hold for all of the metering tools in the MultiMeter. Peak hold has the following effects:
  - In the Analyzer, a small yellow segment above each 1/3 octave level bar labels the most recent peak level.
  - In the Goniometer, all illuminated pixels of the display are held in place during peak hold.
  - In the Correlation Meter, a growing horizontal area around the white correlation indicator shows any phase correlation deviations—in both directions. A vertical red line to the left of the correlation indicator permanently shows the maximum negative phase deviation value. You can reset this line by clicking on it during playback.
  - In the Level Meter, a small yellow segment above each stereo level bar labels the most recent peak level.
- Hold Time pop-up menu: When peak hold is active, sets the hold time for all metering tools to either 2 s, 4 s, 5 s, 6 s, or infinite.
- Reset button: Click to reset the peak hold segments of all metering tools.

Graphic Display
Displays either the Analyzer or Goniometer in the center of the window. The Correlation Meter always appears in the bottom of the window, and the Level Meters always appear on the right side of the window.

Using the MultiMeter
The following section provides information on using the different meters in the MultiMeter.

Analyzer
The Analyzer displays the frequency spectrum of the input signal in 31 independent frequency bands. Each frequency band represents one-third of an octave.

You can display only part of the input signal using one of the channel buttons. Select either the Left or Right button to view these channels independently, select LRmax to see the maximum band levels of either channel, or select Mono to view the levels of the stereo signal as a summed mono entry.
You can alter the scale of values displayed in the Analyzer in several ways. Use the View parameters, which let you set the maximum level displayed and the overall dynamic range, by vertically dragging the dB scale on the left edge of the Analyzer. Adjusting the scale is useful when analyzing highly compressed material, so that you can identify smaller level differences more easily by moving and/or reducing the display range. You can also change the range (the minimum and maximum values displayed) by vertically dragging in the Analyzer display area, and adjust the maximum value shown by dragging vertically in the display area.

The two RMS modes with Slow and Fast response settings show the effective signal average, and provide a representative overview of the perceived volume levels. The Peak mode shows level peaks accurately.

**Goniometer**

A Goniometer helps you to judge the coherence of the stereo image and determine phase differences between the left and right channels. Phase problems are easily spotted as trace cancelations along the center line (M—mid/mono).

The idea of the Goniometer was born with the advent of early two-channel oscilloscopes. To use such devices as Goniometers, users would connect the left and the right stereo channels to the X and Y inputs, while rotating the display by 45 degrees to produce a useful visualization of the signal’s stereo phase. The signal trace slowly fades to black, imitating the retro glow of the tubes found in older Goniometers, while also enhancing the readability of the display.

In the MultiMeter’s Goniometer, the Auto Gain parameter lets you obtain a higher readout on low-level passages, by allowing the display to automatically compensate for low input levels. You can set the amount of compensation in 10% increments. Remember that, for the MultiMeter, Auto Gain is a display parameter only, and increases display levels in order to enhance readability. It does not change the actual audio levels.

**Correlation Meter**

The Correlation Meter provides an additional phase measurement tool that gauges the phase relationship of a stereo signal. The Correlation Meter’s scale ranges from –1 to +1, and different values provide the following indications:

- A +1 correlation value indicates that the left and right channels correlate 100%. In other words, the left and right signals are in phase and are the same shape.
- Correlation values in the blue zone (between +1 and the middle position) indicate that the stereo signal is mono compatible.
- The middle position indicates the highest permissible amount of left/right divergence, which is often audible as an extremely wide stereo effect.
- When the correlation meter moves into the red area to the left of the center position, out-of-phase material is present. This will lead to phase cancelations if the stereo signal is combined into a mono signal.
Level Meter (Peak/RMS Meter)
The Level Meter displays the current signal level on a logarithmic decibel scale. The signal level for each channel is represented by a blue bar. RMS and Peak levels are shown simultaneously, with RMS levels appearing as dark blue bars, and Peak levels appearing as light blue bars. When the level exceeds 0 dB, the portion of the bar above the 0 dB point becomes red.

The current peak values are displayed numerically (in dB increments) above the Level Meter. The values are reset by clicking in the display.

Surround MultiMeter
The surround version of the MultiMeter includes some additional parameters for use in analyzing multichannel surround files.

Surround MultiMeter Parameters

Analyzer Section
These parameters are similar to those in the stereo MultiMeter, but include buttons for each surround channel. You can select a single channel or a combination of channels. When analyzing a combination of channels, you can set the Analyzer to display either the maximum or sum of the selected channels using the Sum and Max buttons.

Goniometer Section
These parameters are similar to those in the stereo MultiMeter, but include buttons for each channel pair (L–R, Lm–Rm, Ls–Rs) or for all channel pairs (both). When using the Surround MultiMeter in configurations with exactly two channel pairs (quad, 5.1, and 6.1 configurations), the Goniometer can optionally display both pairs by selecting Both. One pair (for L–R) appears in the upper center of the graphic display, and one (for Ls-Rs) appears in the lower center.
Balance/Correlation Section
In the Surround MultiMeter, the Correlation Meter is not visible when the Analyzer or Goniometer are active, but is displayed separately when you select the Balance/Correlation button. The highlighted area indicates the overall balance of the surround signal.

The Balance/Correlation Meter combines two meters into one compact, easy-to-read display. It is based on the balance meter, showing the sound placement, while also incorporating a correlation display that shows strongly correlated signals as a sharp marker, less strongly correlated signals as a blurred area, and negative correlations as red.

Peak (Level Meter) Section
These parameters are the same as those in the stereo MultiMeter.

Tuner
You can tune both acoustic and electric music instruments connected to your system using the Tuner. Tuning your instruments ensures that your recordings will be in tune with any software instruments, existing samples, or existing recordings in your projects.

Tuner Parameters

- Graphic tuning display: As you play, the pitch of the note appears in the semicircular area, centered around the Keynote. If the highlight bar moves to the left of center, the note is flat; if the highlight bar moves to the right of center, the note is sharp. The numbers around the edge of the display show the variance, in cents, from the target pitch.

- Keynote/Octave display: The upper Keynote area shows the target pitch of the note you play (the closest pitch in tune). The lower Octave area indicates which octave the note belongs to. This matches the MIDI octave scale, with the C above middle C displayed as C4 and middle C displayed as C3.
• *Tuning Adjustment knob and field:* Sets the pitch of the note used as the basis for tuning. By default, the Tuner is set to concert pitch $A = 440$ Hz. Drag the knob left to lower the pitch corresponding to $A$, or drag the knob right to raise the pitch corresponding to $A$. The current value is displayed in the field.

**Using the Tuner**

Using the Tuner is simple. With your instrument (or microphone capturing the sound of an acoustic instrument) connected to the channel with the Tuner, play a single note and watch the display. If the note is flat of the Keynote, the segments left of center light, showing how far (in cents) the note is off pitch. If the note is sharp, the segments right of center light. Adjust the tuning of your instrument until the center segment lights (red).

On the tuning display, the range is marked in single semitone steps $\pm 6$ cents close to the center, and then in larger increments to a maximum of $\pm 50$ cents.
Modulation

Modulation effects are used to add motion and depth to your sound.

Modulation effects include chorus, flanging, and phasing among others, which make sounds richer or more animated. This is often achieved through the use of an LFO, which is controlled with parameters such as speed or frequency, and depth (also called width, amount, or intensity). You can also control the ratio of the affected (wet) signal and the original (dry) signal. Some modulation effects include feedback parameters, which add part of the effect’s output back into the effect input.

Soundtrack Pro includes the following modulation effects:

- “Chorus” on page 96
- “Ensemble” on page 96
- “Flanger” on page 97
- “Modulation Delay” on page 98
- “Phaser” on page 99
- “RingShifter” on page 101
- “Scanner Vibrato” on page 106
- “Tremolo” on page 107
Chorus
The Chorus effect delays the original signal. The delay time is modulated with an LFO. The delayed, modulated signal is mixed with the original, dry signal.

You can use the Chorus effect to enrich the sound and create the impression that it’s being played by multiple instruments or voices, in unison. The slight delay time variations generated by the LFO simulate the subtle pitch and timing differences heard when several people perform together. Using chorus also adds fullness or richness to the signal, and can add movement to low or sustained sounds.

- **Intensity slider and field**: Defines the modulation amount.
- **Rate knob and field**: Defines the frequency, and therefore the speed, of the LFO.
- **Mix slider and field**: Determines the balance of dry and wet signals.

Ensemble
The Ensemble combines up to eight chorus effects. Two standard LFOs and one random LFO (which generates random modulations) enable you to create complex modulations. The Ensemble’s graphic visually represents the processed signals.

- **Voices slider and field**: Determines how many individual chorus instances are used and therefore how many voices (or signals) are generated, in addition to the original signal.
- **Rate knobs and fields**: Use the respective knob to control the frequency of each LFO.
- **Intensity sliders and fields**: Use these to set the amount of modulation for each LFO.
• **Phase knob and field**: Controls the phase relationship between the individual voice modulations. The value that you choose here is dependent on the number of voices, which is why it is shown as a percentage value rather than degrees. The value 100 (or −100) is equal to the greatest possible distance between the modulation phases of all voices.

• **Spread slider and field**: Used to distribute the voices across the stereo or surround field. When you set a value of 200%, the stereo or surround base is expanded artificially. Please note that monaural compatibility may suffer if you choose to do this.

• **Mix slider and field**: Determines the balance between dry and wet signals.

• **Effect Volume knob and field**: Use this to independently determine the level of the effects signal. This is a useful tool that compensates for changes in volume caused by changes to the Voices parameter.

Flanger
The Flanger effect works in much the same way as the Chorus effect, but uses a significantly shorter delay time. In addition, the effect signal can be fed back into the input of the delay line.

Flanging is typically used to create changes sometimes described as adding a “spacey” or underwater effect.

- **Rate knob and field**: Defines the frequency, and therefore the speed, of the LFO.
- **Intensity slider and field**: Determines the modulation amount.
- **Feedback slider and field**: Determines the amount of the effect signal that is routed back into the input. Negative values invert the phase of the routed signal.
- **Mix slider and field**: Determines the balance between dry and wet signals.
Modulation Delay

The Modulation Delay effect is based on the same principles as the Flanger and Chorus effects, but you can set the delay time, thereby allowing both chorus and flanging effects to be generated. It can also be used—without modulation—to create resonator or doubling effects. The modulation section consists of two LFOs with variable frequencies.

- **Feedback slider and field**: Determines the amount of the effect signal that is routed back to the input. If you’re going for radical flanging effects, enter a high Feedback value. If simple doubling is what you’re after, you won’t want any feedback at all.
- **Flanger-Chorus knob and field**: Sets the basic delay time. Set to the far left position to create flanger effects, to the center for chorus effects, and to the far right to hear clearly discernible delays.
- **Intensity slider and field**: Sets the modulation amount.
- **De-Warble button**: Switch on to ensure that the pitch of the modulated signal remains constant.
- **Constant Mod. (Constant Modulation) button**: Switch on to ensure that the modulation width sounds constant, regardless of the modulation rate. Note that when switched on, higher modulation frequencies will reduce the modulation width.
- **LFO Mix slider and fields**: Determines the balance between the two LFOs.
- **LFO 1 and LFO 2 Rate knobs and fields**: Use the left knob to set the modulation rate for the left stereo channel, and the right knob to set the modulation rate of the right stereo channel. The right LFO Rate knob is only available in stereo and surround instances, and can only be set separately if the Left Right Link button is not enabled. In Surround instances, the center channel is assigned the middle value of the LFO left and LFO right Rate knobs. The other channels are assigned values between the left and right LFO rates.
- **LFO Left Right Link button (only available in stereo and surround instances)**: Switch on to tie the modulation rates of the left and right stereo channels to each other.
• **LFO Phase knob and field (only available in stereo and surround instances)**: Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or –180° is equal to the greatest possible distance between the modulation phases of the channels. The LFO Phase parameter is available only if the LFO Left Right Link button is active.

• **Distribution menu (only available in surround instances)**: Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose “new random” in the Distribution menu.

• **Vol.Mod. (Volume Modulation) slider and field**: Use this to determine how much of an impact the LFO modulation has on the amplitude of the effect signal.

• **Output Mix slider and field**: Determines the balance between dry and wet signals.

**Phaser**
The Phaser effect combines the original signal with a copy that is slightly out of phase with the original. This means that the amplitude of the two signals reach their highest and lowest points at slightly different times. The timing differences between the two signals is modulated by two independent LFOs.

In addition, the Phaser includes a filter circuit and a built-in envelope follower that tracks any volume changes in the input signal, thereby generating a dynamic control signal.

- **Filter button**: Click to activate the filter section, which processes the feedback signal of the Pitch Shifter.
- **LP and HP knobs and fields**: Use these to set the cutoff frequency of the filter section’s highpass and lowpass filters.
- **Feedback slider and field**: Determines the amount of the effect signal that is routed back into the input of the effect.
- **Ceiling and Floor slider and fields**: Use the individual slider handles to determine the frequency range that will be affected by the LFO modulations.
- **Order slider and field**: Allows you to choose between different phaser algorithms. The more orders a phaser has, the heavier the effect.

- **Env Follow slider and field (Sweep section)**: Determines how much the frequency range (as set with the Ceiling and Floor controls) is modulated by the level of the input signal.

- **LFO 1 and LFO 2 Rate knobs and fields**: Use to set the speed for each LFO independently.

- **LFO Mix slider and fields**: Determines the balance between the two LFOs.

- **Env Follow slider and field (LFO section)**: Use this to set how much the speed of LFO 1 is modulated by the level of the input signal.

- **Phase knob and field (only available in stereo and surround instances)**: Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or –180° is equal to the greatest possible distance between the modulation phases of the channels.

- **Distribution menu (only available in surround instances)**: Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose "new random" in the Distribution menu.

- **Output Mix slider and field**: Determines the balance of dry and wet signals. Negative values result in a phase inverted mix of the effect and direct (dry) signal.

- **Warmth button**: Click to switch on an additional distortion circuit, which allows the creation of warm overdrive effects.

### Setting the Phaser Orders

The more orders a phaser has, the heavier the effect. The 4, 6, 8, 10, and 12 settings put five different phaser algorithms at your fingertips, all of which replicate the analog circuits that they are modeled on, each designed for a specific application. You are free to select odd numbered settings (5, 7, 9, 11), which, strictly speaking, don't generate actual phasing. The more subtle comb filtering effects produced by odd numbered settings can, however, come in handy on occasion.
RingShifter
The RingShifter effect combines a ring modulator with a frequency shifter effect. Both effects were popular during the 1970s, and are currently experiencing something of a renaissance.

- The ring modulator modulates the amplitude of the input signal using either the internal oscillator or a side chain signal. The frequency spectrum of the resulting effect signal equals the sum and difference of the frequency content in the two original signals. Its sound is often described as metallic or clangorous. The ring modulator was used extensively on jazz rock and fusion records in the early 70s.

- The frequency shifter moves the frequency content of the input signal by a fixed amount and, in doing so, alters the frequency relationship of the original harmonics. The resulting sounds range between sweet and spacious phasing effects to strange robotic timbres. Frequency shifting should not be confused with pitch shifting. Pitch shifting transposes the original signal, leaving its harmonic frequency relationship intact.

The RingShifter effect consists of the following parameter groups:

- **Mode buttons**: Determine whether the RingShifter operates as a frequency shifter or ring modulator.

- **Oscillator parameters**: Use these to configure the internal sine wave oscillator, which modulates the amplitude of the input signal—in both frequency shifter modes and the ring modulator OSC mode.

- **Envelope follower and LFO parameters**: The oscillator frequency and output signal can be modulated with an envelope follower and LFO.

- **Delay parameters**: Use these to delay the effect signal.

- **Output parameters**: The output section of the RingShifter includes a feedback loop and controls to set the stereo width and amount of the dry and wet signals.
Modes
The four mode buttons determine whether the RingShifter operates as a frequency shifter or as a ring modulator.

- **Single (Frequency Shifter) button**: The frequency shifter generates a single, shifted effect signal. The oscillator Frequency control determines whether the signal is shifted up (positive value) or down (negative value).

- **Dual (Frequency Shifter) button**: The frequency shifting process produces one shifted effect signal for each stereo channel—one is shifted up, the other is shifted down. The oscillator Frequency control determines the shift direction in the left versus the right channel.

- **OSC (Ring Modulator) button**: The ring modulator uses the internal sine wave oscillator to modulate the input signal.

- **Side Chain (Ring Modulator) button**: The ring modulator modulates the amplitude of the input signal with the audio signal assigned via the side chain input. The sine wave oscillator is switched off, and the Frequency controls are not accessible when Side Chain mode is active.

The Oscillator
In both frequency shifter modes, and the ring modulator OSC mode, the internal sine wave oscillator is used to modulate the amplitude of the input signal. The Frequency control sets the frequency of the sine wave oscillator.

- In the frequency shifter modes, this parameter controls the amount of frequency shifting (up and/or down) applied to the input signal.

- In the ring modulator OSC mode, this parameter controls the frequency content (timbre) of the resulting effect. This timbre can range from subtle tremolo effects to clangorous metallic sounds.
• **Frequency control:** Sets the frequency of the sine oscillator.

• **Lin(ear) and Exp(onential) buttons:** Use these buttons to switch the scaling of the Frequency control:
  - The exponential scaling offers extremely small increments around the 0 point, which is useful for programming slow moving phasing and tremolo effects.
  - In the Lin(ear) mode, the resolution of the scale is even across the entire control range.

• **Env Follower slider and field:** Use to determine how much the oscillator is modulated by the level of the input signal.

• **LFO slider and field:** Use to determine the amount of oscillator modulation by the LFO.

**Delay**
The effect signal is routed through a delay, following the oscillator.

- **Time knob and field:** Sets the delay time.
- **Sync button:** Turn this on to synchronize the delay to your project tempo in musical note values.
- **Level knob and field:** Sets the level of the delay added to the ring modulated or frequency shifted signal. A Level value of 0 passes the effect signal directly to the output (bypass).
Output

- **Feedback knob and field**: Sets the amount of the signal that is routed back to the effect input.
- **Stereo Width knob and field**: Determines the breadth of the effect signal in the stereo field. Stereo Width affects only the effect signal of the RingShifter, not the dry input signal.
- **Dry/Wet knob and field**: Set the mix ratio of the dry input signal and the wet effect signal.
- **Env Follower slider and field**: Use to determine how much the Dry/Wet parameter is modulated by the level of the input signal.
- **LFO slider and field**: Sets the modulation depth of the Dry/Wet parameter by the LFO.

**Setting the Feedback**

Feedback gives the RingShifter sound an additional edge and is useful for a variety of special effects. It produces a rich phasing sound when used in combination with a slow oscillator sweep. Comb filtering effects are created by using high Feedback settings with a short delay time (< 10 ms). Using longer delay times, in conjunction with Feedback, creates spiralling, continuously rising and falling frequency shift effects.
Modulation Sources
The oscillator Frequency and Dry/Wet parameters can be modulated via the internal envelope follower and LFO. The oscillator frequency even allows modulation through the 0 Hz point, thus changing the oscillation direction.

Envelope Follower
The envelope follower analyzes the amplitude (volume) of the input signal and uses this to create a continuously changing control signal—a dynamic volume envelope of the input signal. This control signal can be used for modulation purposes.

- **Power button:** Turns the envelope follower on or off.
- **Sensitivity slider and field:** Determines how responsive the envelope follower is to the input signal. At lower settings, the envelope follower will only react to the most dominant signal peaks. At higher settings, the envelope follower will track the signal more closely, but may react less dynamically.
- **Attack slider and field:** Sets the response time of the envelope follower.
- **Decay slider and field:** Controls the time it takes the envelope follower to return from a higher to a lower value.

LFO
The LFO is the second modulation source. The LFO produces continuous, cycled control signals.

- **Power button:** Turns the LFO on or off.
- **Symmetry and Smooth sliders and fields:** These controls shape the LFO waveform. The LFO waveform display provides visual feedback.
- **Rate knob and field:** Sets the cycle speed of the LFO.
- **Sync button:** Turn this on to synchronize the LFO cycles (LFO rate) with the project tempo, using musical note values.
Scanner Vibrato

The Scanner Vibrato effect simulates the scanner vibrato section of a Hammond organ. You can choose between three different vibrato and chorus types. The stereo version of the effect features two additional parameters: Stereo Phase and Rate Right. These allow you to set the modulation speed independently for the left and right channels.

- Vibrato knob: Sets the desired vibrato or chorus types. The C0 setting disables both the vibrato and chorus effect.
- Chorus Int knob: Sets the intensity of a chosen chorus effect type. If a vibrato effect type is chosen, this parameter has no effect.
- Stereo Phase knob: If set to a value between 0 and 360 degrees, Stereo Phase determines the phase relationship between left and right channel modulations, thus enabling synchronized stereo effects. If “free” is chosen, you can set the modulation speed of the left and right channel independently.
- Rate Left knob: Sets the modulation speed of the left channel when Stereo Phase is set to free. If Stereo Phase is set to a value between 0 and 360 degrees, Rate Left sets the modulation speed for both the left and right channels. Rate Right has no function when in this mode.
- Rate Right knob: Sets the modulation speed of the right channel when Stereo Phase is set to free.
**Tremolo**

The Tremolo effect modulates the amplitude of a signal, resulting in periodic volume changes. You’ll recognize this effect from vintage guitar combo amps (where it is sometimes incorrectly referred to as *vibrato*). The graphic display shows all parameters, except Rate.

- **Depth slider and field**: Determines the modulation amount.
- **Rate knob and field**: Defines the frequency, and therefore the speed, of the LFO.
- **Symmetry and Smoothing knobs and fields**: Use these to set the shape of the modulation.
- **Phase knobs and fields (only available in stereo and surround instances)**: Controls the phase relationship between the individual channel modulations. At 0°, the extreme values of the modulation are achieved simultaneously for all channels. 180° or −180° is equal to the greatest possible distance between the modulation phases of the channels.
- **Distribution menu (only available in surround instances)**: Defines how the phase offsets between the individual channels are distributed in the surround field. You can choose from Circular, Random, Front <> Rear, and Left <> Right distribution. When you load a setting that uses the Random option, the saved phase offset value is recalled. If you want to randomize the phase setting again, choose “new random” in the Distribution menu.

**Symmetry and Smoothing**

If Symmetry is set to 50% and Smoothing to 0%, the modulation has a rectangular shape. This means that the timing of the full volume signal is equal to that of the low volume signal, and that switching between both states occurs abruptly. You can define the loud/quiet time ratio with Symmetry and make it fade gently in or out with Smoothing.
You can use the Pitch effects of Soundtrack Pro to transpose the pitch of audio tracks.

These effects can also be used for creating unison or slightly thickened parts, or even the creation of harmony voices.

Soundtrack Pro includes the following Pitch effects:

- “Pitch Shifter II” on page 109
- “Vocal Transformer” on page 110

**Pitch Shifter II**

The Pitch Shifter II provides a simple way to combine a pitch-shifted version of the signal with the original signal.

**Pitch Shifter II Parameters**

- *Semi Tones slider and field*: Sets the pitch shift value in semitones.
- *Cents slider and field*: Controls detuning of the pitch shift value in cents (100ths of a semitone).
• **Drums, Speech, and Vocals buttons:** Select one of the three presets to optimize Pitch Shifter II operation for common types of audio material:
  - Drums leaves the groove of the original track intact.
  - Vocals retains the intonation of the original with no change. Thus, Vocals is well-suited for any signals that are inherently harmonic or melodious, such as string pads.
  - Speech provides a compromise between the two by attempting to retain both the rhythmic and harmonic aspects of the signal. This is suitable for complex signals such as spoken-word recordings, rap music, and other hybrid signals such as rhythm guitar.

• **Mix slider and field:** Sets the amount of the processed signal mixed with the original signal.

**Using Pitch Shifter II**
Set the amount of transposition (pitch shift) with the Semi Tones parameter, then set the amount of detuning with the Cents parameter. Use one of the three presets (Drums, Vocals, or Speech) depending on the material you are working with. For other types of material, you can try each of the presets (starting with Speech), compare the results, and use the one that best suits your material. While you are auditioning and comparing different settings, it’s often a good idea to temporarily set the Mix parameter to 100% to hear the maximum effect of the processing. Keep in mind that Pitch Shifter II artifacts are much harder to hear with Mix set to a smaller percentage.

In the Pitch Shifter II Controls view, you can create your own presets using the Delay and Crossfade parameters. These parameters are effective only when you select the Manual option in the Timing menu. You can also select the Auto option here—the Pitch Shifter will then automatically create presets by analyzing the incoming signal. The Stereo Link parameter allows you to invert the stereo channel’s signals, with processing for the right channel occurring on the left and vice versa.

**Vocal Transformer**
The Vocal Transformer allows you to manipulate vocal tracks in many different ways. You can use it to transpose the pitch of a vocal line, to augment or diminish the range of the melody, or even to reduce it to a single note—to mirror the pitches of a melody. No matter how you change the pitches of the melody, formants remain the same. You can shift the formants independently, which means that you can turn a vocal track into a Mickey Mouse voice, while maintaining the original pitch.

The Vocal Transformer is well suited to extreme vocal effects. The best results are achieved with monophonic signals, including monophonic instrument tracks. The plug-in is not designed for polyphonic voices (a choir on a single track, for example) or other chordal tracks.
Vocal Transformer Parameters

- **Pitch knob and field**: Determines the amount of transposition applied to the input signal.
- **Formant knob and field**: Shifts the formants of the input signal.
- **Robotize button**: Click to switch the Vocal Transformer to Robotize mode. Robotize mode is used for augmenting, diminishing, or mirroring the melody.
- **Tracking slider and buttons (available only in Robotize mode)**: Control how the melody is changed in Robotize mode.
- **Pitch Base slider and field (available only in Robotize mode)**: Use to transpose the note that the Tracking parameter is following.
- **Mix slider and field**: Defines the level ratio between the original (dry) and effect signals.

### Setting the Pitch and Formant Parameters

The Pitch parameter transposes the pitch of the signal up to two octaves upward or downward. Adjustments are made in semitone steps. Incoming pitches are indicated by a vertical line below the Pitch Base field.

Transpositions of a fifth upward (Pitch = +7), a fourth downward (Pitch = –5), or by an octave (Pitch = ±12) are the most useful, harmonically.

As you alter the Pitch parameter, you might notice that the formants don’t change.

Formants are characteristic emphases of certain frequency ranges. They are static and do not change with pitch. Formants are responsible for the specific timbre of a given human voice.

The Pitch parameter is expressly used to change the pitch of a voice, not its character. If you set negative Pitch values for a female soprano voice, you can turn it into an alto voice, without changing the specific character of the singer’s voice.

The Formant parameter shifts the formants, while maintaining—or independently altering—the pitch. If you set this parameter to positive values, the singer sounds like Mickey Mouse. By altering the parameter downward, you can achieve vocals reminiscent of Darth Vader.
**Tip:** If you set Pitch to 0 semitones, Mix to 50%, and Formant to +1 (with Robotize switched off), you can effectively place a singer (with these different vocal characteristics) next to the original singer. Both will sing with the same voice—in a choir of two. This choir effect is quite effective, and is easily controlled with the Mix parameter.

**Using Robotize Mode**

If you switch Robotize on, the Vocal Transformer can augment or diminish the melody. You can control the intensity of this distortion with the Tracking parameter.

The four –1, 0, 1, and 2 buttons set the Tracking slider to values of –100%, 0%, 100%, and 200%, respectively. These buttons are convenience controls that make it easier to set the Tracking parameter to the most useful settings.

- At a value of 100% (switch 1), the range of the melody is maintained. Higher values augment, and lower values diminish, the melody.
- At a setting of 200% (switch 2), the intervals are doubled.
- A setting of 0% (switch 0) delivers interesting results, with every syllable of the vocal track being sung at the same pitch. Low values turn sung lines into spoken language.
- At a setting of –100% (switch –1), all intervals are mirrored.

The Pitch Base parameter is used to transpose the note that the Tracking parameter is following, for example: the note that is spoken, if Tracking is set to 0%.
You can use Reverb effects to simulate the sound of acoustic environments such as rooms, concert halls, caverns, or the sound of infinite space.

Sounds bounce off the surfaces of any space, or off objects within a space, repeatedly, gradually dying out until they are inaudible. The bouncing soundwaves result in a reflection pattern, more commonly known as a reverberation (or reverb).

The early part of a reverb consists of a number of discrete reflections that you can clearly discern before the diffuse reverb tail builds up. These early reflections are essential to how you perceive the space of a room. All information about the size and shape of a room that the human ear can discern is contained in these early reflections.
Plates, Digital Reverb Effects, and Convolution Reverb
The first form of reverb used in music production was actually a special room with hard surfaces (called an echo chamber). It was used to add echoes to the signal. Mechanical devices, including plates and springs, were used to add reverberation to the output of musical instruments and microphones.

Digital recording introduced digital reverb effects, which consist of thousands of delays of varying lengths and intensities. The time between the original signal and the arrival of the early reflections can be adjusted by a parameter commonly known as predelay. The average number of reflections in a given period of time is determined by the density parameter. The regularity or irregularity of the density is controlled with the diffusion parameter.

Ever-increasing computing power has made it possible to sample the reverb characteristics of real spaces, using convolution reverbs. These room characteristic sample recordings are known as impulse responses.

Convolution reverbs work by convolving (combining) an audio signal with the impulse response recording of a room’s reverb character.

This chapter describes the reverb effects included in Soundtrack Pro:
- PlatinumVerb
- Space Designer: Space Designer is a convolution reverb and is described separately in Chapter 11, “Convolution Reverb: Space Designer,” on page 119.
- “Soundtrack Pro Reverb” on page 117

PlatinumVerb
The PlatinumVerb effect allows you to edit both the early reflections and diffuse reverb tail separately, making it easier to precisely emulate real rooms. Its dual-band Reverb section splits the incoming signal into two bands, each of which is processed (and can be edited) separately.
The interface can be divided into four parameter groups:

- **Early Reflections parameters:** Emulates the original signal's first reflections as they bounce off the walls, ceiling, and floor of a natural room.
- **Reverb parameters:** Controls the diffuse reverberations.
- **Balance ER/Reverb parameter:** Controls the balance between the Early Reflections and Reverb section. When you set the slider to either of its extreme positions, the unused section is deactivated.
- **Output section:** Determines the balance between the effected (wet) and direct (dry) signals.

### Early Reflection Parameters

- **Predelay:** Determines the amount of time between the start of the original signal and the arrival of the early reflections.
- **Room Shape:** Defines the geometric form of the room. The numeric value (3 to 7) represents the number of corners in the room. The graphic display visually represents this setting.
- **Room Size:** Determines the dimensions of the room. The numeric value indicates the length of its walls—the distance between two corners.
- **Stereo Base (only available in stereo instances):** Defines the distance between the two virtual microphones that you are using in the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. More realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head.

### Reverb Parameters

- **Initial Delay:** Sets the time between the original signal and the diffuse reverb tail.
- **Spread:** Controls the stereo image of the reverb. At 0%, the effect generates a monaural reverb. At 200%, the stereo base is artificially expanded.
- **Crossover:** Defines the frequency at which the input signal is split into two frequency bands, for separate processing.
- **Low Ratio:** Determines the reverb time of the bass band in relation to the reverb time of the high band. It is expressed as a percentage, ranging from 0 to 200%.
- **Low Freq Level:** Sets the level of the bass reverb. At 0 dB, the volume of the two bands is equal.
- **High Cut:** Frequencies above the set value are filtered from the reverb signal.
- **Density:** Controls the density of the diffuse reverb tail.
- **Diffusion:** Sets the diffusion of the reverb tail.
- **Reverb Time:** Determines the reverb time of the high band.
Output Parameters

- **Dry**: Controls the amount of the original signal.
- **Wet**: Controls the amount of the effect signal.

Setting Predelay and Initial Delay

In practice, too short a Predelay tends to make it difficult to pinpoint the position of the signal. It can also color the sound of the original signal. On the other hand, too long a Predelay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, which leaves an audible gap.

The optimum Predelay setting depends on the type (or envelope) of signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually. A good practice is to use the longest Predelay possible before you start to hear undesirable side effects, such as an audible echo.

If you’re going for a natural-sounding, harmonic reverb, the transition between the early reflections and the reverb tail should be as smooth and seamless as possible. Set the Initial Delay so that it is as long as possible, without a noticeable gap between the early reflections and the reverb tail.

Setting Density and Diffusion

Ordinarily, you want the signal to be as dense as possible. However, use of a lower Density value means the effect eats up less computing power. Beyond this, in rare instances, a high Density value can color the sound, which you can fix by simply reducing the Density knob value. Conversely, if you select a Density value that is too low, the reverb tail will sound grainy.

High Diffusion values represent a regular density, with few alterations in level, times, and panorama position. Low Diffusion values result in the reflection density becoming irregular and grainy. The stereo spectrum changes, too.

Setting the Reverb Time

Reverb Time is commonly considered as the amount of time it takes for the level of a reverb signal to drop by 60 dB. This is why the reverb time is often indicated as RT60. Most natural rooms have a reverb time somewhere in the range of one to three seconds, a value that absorbent surfaces and furniture reduces. Large empty halls or churches have reverb times of up to eight seconds, and some cavernous or cathedral-like venues even beyond that.

Setting the High Cut

Uneven or absorbent surfaces (wallpaper, wood paneling, carpets, and so on) tend to reflect lower frequencies better than higher frequencies. The High Cut filter replicates this effect. If you set the High Cut filter so that it is wide open, the reverb will sound as if it is reflecting off stone or glass.
Setting the Reverb Time and Level of the Low Frequency Band

You can use the Low Ratio control to offset the reverb time of the low frequency band. At 100%, the reverb times for the two bands are identical. At lower values, the reverb time of the frequencies below the crossover frequency is shorter. At values greater than 100%, the reverb time for low frequencies is longer.

Both of these phenomena occur in nature. In most mixes, a shorter reverb time for bass frequencies is preferable. For example, if you’re using the PlatinumVerb on a kick and snare drum loop, a short reverb time for the kick drum allows you to set a substantially higher wet signal.

The Low Freq Level slider allows you to boost or attenuate the level of the low frequency band. In the vast majority of mixes, your best bet is to set a lower level for the low frequency reverb signal. This enables you to turn up the level of the bass instrument—making it sound punchier. This also helps to counter bottom-end masking effects.

Soundtrack Pro Reverb

The Soundtrack Pro Reverb provides a simple reverb effect that requires only modest CPU resources. You can use it on both musical and non-musical audio material.

- **Mix (dry/wet mix %) slider and field**: Sets the ratio of the original (dry) signal to the effected (wet) signal at the output.
- **Decay (decay %) slider and field**: Sets the percentage of the processed signal fed back into the effect.
Convolution Reverb: Space Designer

Space Designer is a convolution reverb effect. You can use it to create exceptionally realistic reverberations.

Space Designer generates reverb by convolving, or combining, an audio signal with an impulse response (IR) reverb sample. For example, imagine that you apply the Space Designer to a vocal track. If you load an IR recorded from an actual opera house into Space Designer, it will convolve the IR of the opera house with your vocal track and put the singer inside the opera house.

An impulse response is a recording of a room's reverb character; to be more precise: a recording of all the reflections in a given room, following an initial signal spike. The actual impulse response file is a standard audio file—it is not the file type that is unique, it is how the file is used.

Space Designer can operate as mono, stereo, true stereo (meaning each channel is processed discretely), or surround effect. Space Designer not only loads existing impulse responses, but offers sound sculpting features such as envelopes, filters, EQ, and stereo/surround balance controls that provide you with unprecedented control over dynamics, timbre, and length via a comprehensive set of parameters. In addition, Space Designer includes an on-board impulse response synthesis facility.

You can use Space Designer to create both a highly realistic reverb if you use an IR recorded from a real space, or a completely unique effect if you use a synthesized IR that doesn't represent any real space. Convolution can be used to put your audio signal inside any space, including a speaker cabinet, plastic toy, and so on, if you make an IR from it. And with Space Designer's extensive audio processing features, you can exactly tailor its space to your material.
Space Designer consists of the following parameter groups:

- **Impulse response parameters:** Use these parameters to load, save, or manipulate impulse response files. The IR file you choose determines what Space Designer will use to convolve with your audio signal. These parameters will be the initial parameters you use to load your IR file, as well as the last if you want to save your synthesized IR. See “Impulse Response Parameters” on page 121.

- **Global parameters:** After your IR is loaded, you will use these general parameters to adjust how Space Designer operates on the signal and IR. Global parameters include input and output parameters, delay and volume compensation, predelay, and so on. These parameters affect the overall processing of Space Designer, as opposed to the more specific parameter groups that affect one particular aspect of Space Designer’s processing. See “Global Parameters” on page 125.

- **Envelope and EQ display:** Use the button bar at the top to switch the display between envelopes and the EQ. You can edit the selected parameters both graphically and numerically in the display itself.

- **Volume envelope parameters:** Use the volume envelope to dynamically animate the volume of your reverb over the duration of the impulse response. See “Volume Envelope Parameters” on page 133.

- **Filter parameters:** You can further modify the timbre of the Space Designer reverb using these resonant filter parameters. You can choose from a number of filter modes, adjust the resonance of the filter, as well as adjust the filter envelope dynamically over time, as you can with the volume envelope. See “Filter Parameters” on page 134.
Synthesized impulse response parameters: If you do enough processing of the original IR, you may want to synthesize a new IR from your edited parameters. Use these parameters to adjust the density envelope and other synthesized IR parameters. See “Synthesizer Impulse Response Parameters” on page 136.

EQ: For final sound sculpting, Space Designer includes a built-in four-band EQ: two shelving filters, and two parametric filters. Use these parameters to fine tune the sound of your reverb to your taste. See “EQ Parameters” on page 138.

Important: In order to convolve audio in real time, Space Designer must first calculate any parameter adjustments to the impulse response. This requires a moment or two, following parameter edits, and is indicated by a blue progress bar. During this time, you can continue to adjust the parameter. Once calculation starts, the blue bar is replaced by a red bar, advising you that calculation is taking place.

Impulse Response Parameters
Space Designer can use either impulse response files or its own synthesized impulse responses. The circular area to the left of the Envelope and EQ display contains the impulse response parameters. Here you can determine the Impulse Response mode (IR Sample mode or Synthesized IR mode), load or create impulse responses, and set the sample rate.

- IR Sample button: Click to switch Space Designer to the IR Sample mode. The loaded impulse response sample is used to generate reverberation.
- IR Sample Arrow button: Click to load an impulse response.
- Sample Rate parameter: Determines the sample rate of the loaded impulse response.
- Preserve Length option: Activate to preserve the length of the impulse response when changing the sample rate.
• **Length parameter:** Adjusts the length of the impulse response.

• **Synthesized IR button:** Click to switch Space Designer to Synthesized IR mode. In this mode, Space Designer generates a new synthesized impulse response from the values of the Length, envelope, filter, EQ, and Spread parameters. You may freely switch between a loaded impulse response sample and a synthesized impulse response without losing the settings of the other. See “Synthesizer Impulse Response Parameters” for more information on working in Synthesized IR mode.

**Working in IR Sample Mode**
When you first click the IR Sample button, a file selector box opens, allowing you to select the desired impulse response file from a folder on your hard disk or CD. If you have already loaded an impulse response file, this button switches back from Synthesized IR to IR Sample mode.

**Loading IRs**
To change the currently loaded impulse response, click the downward pointing arrow to the right of the button. This accesses the following menu functions:

• **Load IR:** Loads an impulse response sample without changing the envelopes.

• **Load IR & Init:** Loads an impulse response sample and initializes the envelopes.

• **Show in Finder:** Opens a Finder window that displays the file location.

The name of the loaded impulse response file and its length are displayed in the main display's Envelope view.

All impulse responses that ship with Soundtrack Pro are installed in the /Library/Audio/Impulse Responses/Apple folder. The default name for deconvolution files consists of the source file name, appended with an ".sdir" file extension.

**Impulse Response Formats**
Any mono, stereo, AIFF, SDII, or WAV file can be used. In addition, because Space Designer supports surround formats up to 7.1 surround, discreet audio files and B-format audio files that comprise a single surround IR can also be used.
Setting the Sample Rate
The Sample Rate slider is used to determine the sample rate of an impulse response. You can choose between the following settings:

- **Orig**: Space Designer uses the current project sample rate. When loading an impulse response, Space Designer automatically converts the sample rate of the impulse response to match the current project sample rate—should it be necessary. For example, this allows you to load a 44.1 kHz impulse response into a project running at 96 kHz, and vice versa.

- **/2, /4, /8**: These settings are half-divisions of the preceding value—one-half, one-quarter, one-eighth. For example:
  - If the project sample rate is 96 kHz, the options will be 48 kHz, 24 kHz, and 12 kHz.
  - If 44.1 kHz is the project sample rate, the options will be 22.05 kHz, 11.025 kHz, and 5512 Hz.

Changing the sample rate increases or reduces the frequency response and length of the impulse response and, to a degree, the overall sound quality of the reverb. That said, don’t worry too much if the maximum bandwidth of the reverb tail is reduced to 11.025 kHz when you select a sample rate of 22.05 kHz (half of 44.1 kHz). Natural room surfaces (concrete and tiles excluded) barely reflect such high frequencies.

By selecting half the sample rate, the impulse response becomes twice as long. The highest frequency that can be reverberated will be halved. This results in a behavior that is much like doubling every dimension of a virtual room (multiplying a room’s volume by eight).

Another benefit of reducing the sample rate is that processing requires significantly less CPU power, making half sample rate settings the ideal solution for large, open spaces.
Activating the Preserve Length button preserves the length of the impulse response when the sample rate is changed. Manipulating these two parameters as you see fit can lead to interesting results.

The lower sample rates can also be used for interesting tempo, pitch, and retro-digital sounding effects.

If running Space Designer in a project that uses a higher sample rate than the impulse response, you may also want to reduce the impulse response sample rate. Make sure the Preserve Length function is enabled. This cuts CPU power consumption, without compromising reverb quality. There is no loss in reverb quality, because the impulse response will not benefit from the higher project sample rate.

Similar adjustments can be made while running in Synthesized IR mode. Most typical reverb sounds don’t feature an excessive amount of high frequency content. If you were running at 96 kHz, you would need to make use of some deep lowpass filtering to obtain the mellow frequency response characteristics of many reverb sounds. As a different approach, you are better served to first reduce the high frequencies by 1/2 or even 1/4 using the Sample Rate slider, and then apply the lowpass filter. This conserves a considerable amount of CPU power.

**Setting the Length of the Impulse Response**

You can use the Length parameter to set the length of the impulse response (sampled or synthesized).

All envelopes are automatically calculated as a percentage of the overall length, which means that if this parameter is altered, your envelope curves will stretch or shrink to fit, thus saving you time and effort.

When using an impulse response file, the Length parameter value cannot exceed the length of the actual impulse response sample. Longer impulse responses (sampled or synthesized) place a higher strain on the CPU.
Global Parameters

Space Designer’s Envelope and EQ display contains most of Space Designer’s interface elements that change to reflect the current parameter group you are adjusting. The global parameters, spread throughout the interface around and below the Envelope and EQ display, remain constant.

The upper (raised) section of Space Designer contains the following global parameters:

- **Input slider**: Determines how Space Designer processes a stereo or surround input signal. See “Input Slider” on page 126 for more information.
- **Latency Compensation button**: Switches Space Designer’s internal latency compensation feature on or off. See “Latency Compensation” on page 127 for more information.
- **Definition area**: Lets you configure Space Designer to switch to a less defined IR set in order to emulate reverb diffusion and save computer processing power. See “Definition” on page 128 for more information.
- **Rev Vol Compensation**: Engages Space Designer’s internal IR volume matching (see “Rev Vol Compensation” on page 128 for more information).
- **Output sliders**: Adjust output levels (see “Output Parameters” for more information).
The lower (flat) section of Space Designer contains the following global parameters:

- **Filter parameters**: Activate, adjust the resonance, and select the mode of Space Designer’s resonant filter. See “Filter Parameters” for more information.

- **Pre-Dly knob**: Sets the reverb’s pre-delay time, or time between the original signal and the first reflections from the reverb. See “Pre-delay” on page 130 for more information.

- **IR Start knob**: Shifts the point at which the impulse response will play back. See “IR Start” on page 130 for more information.

- **Spread knob**: For synthesized impulse responses, this adjusts the perceived stereo width (for stereo instances of Space Designer) or surround width (for surround instances) to widen the perceived stereo or surround field and enhance the stereo or surround effect. See “Spread Parameters” on page 137 for more information.

- **Xover knob**: Sets the crossover frequency below which synthesized IRs will be processed by the Spread knob. See “Spread Parameters” on page 137 for more information.

### Input Slider

The Input slider functions as either a stereo processing slider for stereo instances of Space Designer, or as an LFE to Reverb slider in surround mode. The Input slider does not appear in mono or mono-to-stereo instances of Space Designer.

**Stereo Mode**

For stereo instances of Space Designer, the Input slider determines how a stereo signal is processed:

- **Stereo setting (top of slider)**: The signal is processed on both channels, retaining the stereo balance of the original signal.

- **Mono setting (middle of slider)**: The signal is processed in mono.

- **XStereo setting (bottom of slider)**: The signal is inverted, with processing for the right channel occurring on the left, and vice versa.

- **In-between positions**: A mixture of stereo to mono crossfeed signals is produced.

**Tip**: The three base positions of the Input slider are clickable key parameter positions—if you click them, the slider will immediately jump to the clicked position.
**Surround Mode**
For surround instances of Space Designer, the Input slider determines how much of the LFE signal is mixed with the surround channels feeding the reverb.

At its lowest setting, the slider acts as an LFE bypass, with all the LFE signal passed through the reverb unprocessed.

**Latency Compensation**
The complex calculations made by Space Designer take time. This time results in a processing latency, or delay, between the direct (input) signal, and the processed (output) signal. The Latency Compensation button determines how Space Designer delays the direct signal in relation to the processed signal.

Space Designer’s processing latency is 128 samples at 44.1 kHz, and doubles at each lower sample rate division. For example, if you set Space Designer’s Sample Rate slider to “/2,” the processing latency increases to 256 samples. The processing latency does not increase in surround mode or with higher sample rates than 44.1 kHz.

When activated, this parameter delays the direct signal (in the Output section) to match the processing delay of the effect signal. This is not related to latency compensation in the host application—this compensation occurs within Space Designer and applies only to Space Designer.
**Rev Vol Compensation**

Rev Vol Compensation (Reverb Volume Compensation) attempts to match the perceived (not actual) volume differences of impulse response files.

It is switched on by default and should generally be left in this mode, although you may find that it isn’t successful with all types of impulse responses. In such situations, switch it off and adjust the input and output levels accordingly.

**Definition**

The Definition parameter appears in the definition area at the bottom of the Envelope and EQ display.

Calculating every precise detail of an impulse response derived reverb uses significant CPU power. The Definition parameter emulates the diffusion of natural reverb patterns while at the same time reducing Space Designer’s CPU consumption.

Natural reverbs contain most of their spacial information in the first few milliseconds. Toward the end of the reverb, its reflection pattern diffuses more and more, containing less spacial information. In order to emulate this phenomenon—as well as conserve CPU power—you can configure Space Designer to only use the full IR resolution at the onset of the reverb, and to use a reduced IR resolution toward the end of the reverb.

The Definition parameter acts as the crossover point at which this switch to the reduced IR resolution occurs. The parameter is displayed in both milliseconds (indicating when the crossover occurs) and a percentage (100% is equal to the length of the full resolution IR).

**Note:** The Definition slider appears only when you have loaded CPU intensive impulse response formats, such as true stereo.
Output Parameters
The output parameters let you adjust the mix between the direct (dry) and processed signals. Which parameters are available depends on Space Designer’s input configuration.

Mono and Stereo Configurations
If you insert Space Designer as mono, mono-to-stereo, or stereo effect, Space Designer offers two output sliders: one for the direct signal, and one for the reverb signal.

- **Dry slider**: Sets the level of the noneffected (dry) signal. Set this to a value of 0 (mute) if Space Designer is inserted in a bus channel, or when using modeling impulse responses such as speaker simulations.
- **Rev(erb) slider**: Adjusts the output level of the effected (wet) signal.

Surround Configuration
In surround configurations, Space Designer offers four output sliders that together comprise a small surround output mixer.

These sliders have the following functions:
- **C(enter)**: Adjusts the center reverb level.
- **Bal(ance)**: Sets the balance between the L-C-R front and the Ls-Rs rear speakers. In 7.1 ITU surround, the balance pivots around the Lm-Rm speakers, taking the surround angles into account. With 7.1 SDDS surround, the Lc-Rc speakers are considered front speakers.
- **Rev(erb)**: Adjusts the output level of the effected (wet) signal.
- **Dry**: Sets the level of the noneffected signal. Set this to a value of 0 (mute) when using Space Designer on an aux channel.
**Predelay**

Predelay is the amount of time that elapses between the original signal and the initial early reflections from the reverb generated by Space Designer.

For a room of any given size and shape, predelay determines the distance between the listener and the walls, ceiling, and floor. Of course, Space Designer allows you to adjust this parameter separately from, and over a greater range than what is considered natural for predelay. In practice, too short a predelay tends to make it difficult to pinpoint the position of the signal. It can also color the sound of the original signal.

On the other hand, too long a predelay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, leaving an audible gap between the signals. The ideal predelay setting to create a realistic space depends on the properties of, or more accurately the envelope of, the original signal. Percussive signals generally require shorter predelays than signals where the attack fades in gradually. A good rule of thumb is to use the longest predelay possible before undesirable side effects, such as an audible echo, begin materializing.

Obviously, these guidelines are designed to help you design realistic sounding spaces. If you want to create unnatural soundscapes using Space Designer, experiment with the Predelay parameter to create otherworldly reverbs and echoes.

**IR Start**

The IR Start parameter enables you to shift the playback point into the impulse response, which will effectively cut off the beginning of the impulse response.

For example, the IR Start parameter can be used to eliminate any peaks at the beginning of the impulse response sample. It also offers a number of creative options, such as its use when combined with the Reverse function (see “Button Bar” on page 131).

*Note:* The IR Start parameter is not available in Synthesized IR mode. In the Synthesized IR mode this parameter is not required as, by design, the Length parameter provides identical functionality.
Envelope and EQ Display

Space Designer’s Envelope and EQ display features two components: the button bar at the top and the main display (including its parameter bar). The display itself shows either the envelope being edited or the EQ curve, depending on which button you engage.

**Button Bar**

The Envelope and EQ display’s button bar includes buttons to switch the main display between envelopes and the EQ, as well as some function buttons.

- **Reset button**: Click to reset the currently displayed envelope or EQ to its default values.
- **All button**: Click to reset all envelopes and the EQ to default values.
- **Volume Env button**: Click this button to bring the volume envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See “Volume Envelope Parameters” on page 133 for information on the volume envelope.
- **Filter Env button**: Click this button to bring the filter envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See “Filter Parameters” on page 134 for information on the filter envelope.
- **Density Env button**: Click this button to bring the density envelope to the front of the main display. The other envelope curves are shown as transparencies in the background. See “Synthesizer Impulse Response Parameters” on page 136 for information on the density envelope.
- **EQ button:** Click this button to switch the main display to Space Designer’s four-band parametric EQ. See “EQ Parameters” on page 138 for more information on Space Designer’s EQ.

- **Reverse button:** Click to reverse the impulse response together with its envelopes. When you reverse the impulse response, you are effectively using the tail rather than the front end of the sample. As such, you may need to use lower or even negative Predelay values when reversing.

### Additional View Items in Envelope View
When displaying envelopes, the main display offers a few other buttons and an overview that is not part of the EQ view:

- **Impulse response overview:** Indicates which portion of the impulse response file is currently visible, helping you to orientate yourself when zoom is active.

- **Zoom to Fit button:** Switch on to display the entire impulse response waveform. The display will automatically follow any envelope length changes.

- **A and D buttons:** Click to limit the Zoom to Fit function to the attack and decay portions of the (currently selected) envelope. The A and D buttons are available only to the volume and filter envelopes.

### Setting Envelope Parameters
Space Designer allows you to edit the volume and filter envelopes of all IRs, and the density envelope of synthesized IRs. All three envelopes can be adjusted both graphically and numerically (in the parameter bar).

While some parameters are envelope-specific, all envelopes consist of the Attack Time and Decay Time parameters: The combined total of the Attack Time and Decay Time parameters is equal to the total length of the (synthesized or sampled) impulse response (determined by the Length parameter, see “Setting the Length of the Impulse Response” on page 124), unless the Decay time is reduced.
• You can change the curve shape by dragging the envelope curve directly. Use the small nodes attached to a line for finer adjustments to envelope curves. They are tied to the envelope curve itself, so you can view them as envelope handles. Moving the nodes vertically or horizontally will change the shape of the envelope curve.

• The large nodes are value indicators of the parameters that appear in the horizontal parameter bar below—Init Level, Attack Time, Decay Time, and so on. If you edit any numerical value, the corresponding node will move in the main display. Try this with each numerical parameter to establish which node is which. When you move the pointer over one of these nodes, you’ll see a pair of arrows. The arrows simply indicate the possible directions that the node can be moved.

Volume Envelope Parameters
The volume envelope lets you set the reverb’s initial level and adjust how the volume will change over time. You can edit all of the volume envelope parameters numerically, and many of them can also be edited graphically using the techniques discussed in “Setting Envelope Parameters” on page 132.

The volume envelope includes the following parameters:
• ***Init Level***: Sets the initial volume level of the impulse response attack. It is expressed as a percentage of the full-scale volume of the impulse response file. The attack phase is generally the loudest point of the impulse response. Set Init Level to 100% to ensure maximum volume for the early reflections.
• ***Attack Time***: Determines the length of time before the decay phase of the volume envelope begins.
• ***Decay Time***: Sets the length of the decay phase.
• **Volume decay mode buttons:** Click to choose the volume decay curve.
  • *Exp:* The output of the volume envelope is shaped by an exponential algorithm in order to generate the most natural sounding reverb tail.
  • *Lin:* The volume decay will be more linear (and less natural sounding).
  • *End Level:* Sets the end volume level. It is expressed as a percentage of the overall volume envelope. If you set this parameter to 0%, the reverb tail cuts off abruptly, which is great for gated reverb effects.

**Filter Parameters**
Space Designer’s filter provides control over the timbre of the reverb. Its controls are distributed between two parts of the Space Designer interface: The main filter parameters are found in Space Designer’s lower left corner, and the filter envelope appears on the Envelope and EQ display when its Filter button is engaged. You can select from several filter types, but you also have envelope control over the filter cutoff, independent from the volume envelope. Changes to the filter settings result in a recalculation of the impulse response, rather than a straight change to the sound as it plays through the reverb.

**Main Filter Parameters**

• **Filter On/Off button:** Switches the filter section on and off.
  • **Filter Mode knob:** Selects between the four filter modes.
  • **Reso(nance) knob:** Adjusting this parameter emphasizes frequencies above, around, or below the cutoff frequency. As you increase the resonance value, the sound will lose bass and become thinner. How much effect the resonance value has on the sound also depends on the selected filter mode, with steeper filter modes resulting in more pronounced resonance.
Setting the Filter Mode
The Filter Mode knob switches between four modes. Click the desired LP (lowpass) 6 dB and 12 dB, BP (bandpass) or HP (highpass) value.

- **6 dB (LP):** Bright, good general-purpose filter mode. It can be used to retain the top end of most material, while still providing some filtering.
- **12 dB (LP):** Useful where you want a warmer sound, without drastic filter effects. It is handy for smoothing out bright reverbs.
- **BP:** 6 dB per octave design. Reduces the amount of signal that surrounds the midtones of the input material, leaving the frequencies around the cutoff frequency intact.
- **HP:** 12 dB per octave/two-pole design. This filter reduces the level of frequencies that fall below the cutoff frequency.

Filter Envelope Parameters
The Filter Envelope lets you control the filter’s cutoff frequency over time. All of the filter envelope’s parameters can be adjusted either numerically in the parameter area or graphically in the main display using the techniques discussed in “Setting Envelope Parameters” on page 132.

The parameters for the Filter Envelope are as follows:

- **Init Level:** Sets the initial cutoff frequency of the Filter Envelope.
- **Attack Time:** Determines the time required to reach the Break Level (see below) value.
- **Break Level:** Sets the maximum filter cutoff frequency the envelope reaches. It also acts as the break point between the attack and decay phases of the overall filter envelope. In other words, when this level has been reached after the attack phase, the decay phase will begin. You can create interesting filter sweeps by setting the Break Level to a value lower than that of the Init Level.
- **Decay Time:** Determines the time required (after the Break Level point) to reach the End Level value.
- **End Level:** Sets the filter end cutoff frequency.
Synthesizer Impulse Response Parameters

In Synthesizer IR mode, Space Designer generates a synthesized impulse response determined by the values of the Length, envelopes, filter, EQ, and spread parameters. To switch to Synthesizer IR mode, enable the Synthesized IR button in the impulse response parameters section.

Clicking the activated Synthesized IR button will randomly generate new impulse responses with slightly different reflection patterns. The current impulse response state will always be saved with a setting, allowing for an accurate reproduction of the reverberation sound when next loaded.

Density Envelope

The density envelope allows you to control the density of the synthesized impulse response over time. You can adjust the density envelope numerically in the parameter bar, and you can edit the Init Level, Ramp Time, and End Level parameters using the techniques described in “Setting Envelope Parameters” on page 132.

Please note that the Density Envelope is only available when in the Synthesized IR mode.
The density envelope offers the following parameters:

- **Init Level**: Sets the initial density (the average number of reflections in a given period of time) of the reverb. Lowering the density levels will result in audible reflections patterns and discreet echoes.
- **Ramp Time**: Adjusts the length of time elapsed between the Initial and End Density levels.
- **End Level**: Sets the density of the reverb tail. If you select an End Level value that is too low, the reverb tail will sound grainy. You may also find that the stereo spectrum is affected by lower values.
- **Reflection Shape**: Determines the steepness (shape) of the early reflection clusters as they bounce off the walls, ceiling, and furnishings of the virtual space. Small values result in clusters with a sharp contour, and large values result in an exponential slope and a smoother sound. This is handy when recreating rooms constructed of different materials. Reflection Shape, in conjunction with suitable settings for the envelopes, density, and early reflection will assist you in creating rooms of almost any shape and material.

**Spread Parameters**
The Spread and Xover knobs adjust the perceived stereo or surround width of a synthesized impulse response. While the Spread and Xover knobs are placed among the global parameters, they only function in Synthesized IR mode.

![Spread and Crossover knobs](image)

**Note**: As these parameters adjust stereo or surround processing, they have no impact when using the Space Designer as a mono plug-in.
Spread extends the stereo or surround base to frequencies that fall below the frequency determined by the Xover (crossover) parameter.

At a Spread value of 0, no stereo or surround information is added (although the inherent stereo or surround information of a signal and reverb will be retained). At a value of 100, the left and right channel divergence is at its maximum.

The Xover parameter is set in Hertz. Any synthesized impulse response that falls below this threshold value will be processed by adjustments over 0 for the Spread parameter.

The effect enhances the perceived width of the signal, without losing the directional information of the input signal normally found in the higher frequency range. Low frequencies are spread to the sides, reducing the amount of low frequency content in the center and allowing the reverb to nicely wrap around the mix.

**EQ Parameters**

Space Designer features a four-band EQ comprised of two parametric mid-bands plus two shelving filters (one low shelving filter and one high shelving filter).
The EQ has the following parameters:

- **EQ On/Off button**: Click to switch the entire EQ section on or off.
- **Individual EQ buttons (1 through 4)**: Click to turn individual EQ bands on or off.
- **Frequency**: Sets the frequency for the selected EQ band.
- **Gain**: Adjusts the gain cut or boost for the selected EQ band.
- **Q**: Sets the Q-factor for the two parametric bands. The Q can be adjusted from 0.1 (very narrow) to 10 (very wide).

You can edit the EQ parameters numerically in the parameter bar or graphically on the main display. Move the mouse horizontally over the display. When your mouse cursor is in the access area of a band, its individual curve and parameter area will be highlighted and a pivot point appears.

- Drag any band to the right or left to adjust its frequency.
- Drag any band up to increase the gain, or down to lower the gain.
- Position the pointer directly on the (illuminated) pivot point of a parametric band and drag up to raise the Q or down to lower the Q.

**Automating Space Designer**

Space Designer cannot be fully automated as per most other Soundtrack Pro plug-ins. This is because Space Designer needs to reload the impulse response (and recalculate the convolution) before audio can be routed through it.

You can, however, record, edit, and play back any movement of the following Space Designer parameters:

- Stereo Crossfeed
- Direct Output
- Reverb Output
Specialized

Soundtrack Pro includes a bundle of specialized plug-ins designed to address tasks often encountered during audio production.

Consider using these specialized effects if you want to do any of the following:

- Eliminate or reduce noise below a threshold level (see “Denoiser” on page 141).
- Add life to digital recordings by adding additional high frequency components (see “Exciter” on page 143).
- Add an artificial bass signals, derived from the incoming signal (see “SubBass” on page 144).

Denoiser
The Denoiser eliminates or reduces any noise below a threshold volume level.

Denoiser Parameters

- Threshold slider and field: Sets the volume level (threshold) below which the DeNoiser reduces the signal.
- Reduce slider and field: Sets the amount of noise reduction applied to sounds below the threshold. When reducing noise, remember that each 6 dB reduction is equivalent to halving the volume level (and each 6 dB increase equals a doubling of the volume level).
For example, if the noise floor of your recording is very high (more than –68 dB), reducing it to a level of –83 to –78 dB should be sufficient, provided this does not introduce any audible side effects. This effectively reduces the noise by more than 10 dB, to less than half of the original (noise) volume.

- **Noise Type slider and field**: Set to a value appropriate to the type of noise you want to reduce.
  - A value of 0 equals white noise (equal frequency distribution).
  - Positive values change the noise type to pink noise (harmonic noise; greater bass response).
  - Negative values change the noise type to blue noise (hiss—tape noise).
- **Smoothing Frequency knob**: Adjusts how smoothing is applied to neighboring frequencies. If the Denoiser recognizes that only noise is present on a certain frequency band, the higher you set the Frequency Smoothing parameter, the more it changes the neighboring frequency bands to avoid glass noise.
- **Smoothing Time knob**: Sets the time required by the Denoiser to reach (or release) maximum reduction. This is the simplest form of smoothing.
- **Smoothing Transition knob**: Adjusts how smoothing is applied to neighboring volume levels. If the Denoiser recognizes that only noise is present in a certain volume range, the higher you set the Transition Smoothing parameter, the more it also changes similar level values to avoid glass noise.
- **Graphic display**: Shows how the lowest volume levels of your audio material (which should be mostly or entirely noise) are reduced. Changes to parameters are instantly reflected here.

**Using the Denoiser**

Locate a section of the audio where only noise is audible, and set the Threshold value so that only signals at, or below, this level are filtered out. Then start playback and set the Reduce value as you listen to the audio, so that as much noise as possible is reduced, but as little of the desired signal is reduced.

The Denoiser uses Fast Fourier Transform (FFT) analysis to recognize frequency bands of lower volume and less complex harmonic structure, and then reduces them to the desired dB level. In principle, this method is completely discrete, as neighboring frequencies are also affected.

If you use the Denoiser too aggressively, however, the algorithm will produce artifacts, such as glass noise, which are obviously artificial and therefore less desirable than the existing noise in most cases. If using the Denoiser produces these artifacts, you can use the three Smoothing knobs to reduce or eliminate them.
**Exciter**
The Exciter generates high-frequency components that are not part of the original signal, using a nonlinear distortion process that resembles overdrive and distortion effects. Unlike those effects, however, the Exciter passes the input signal through a highpass filter before feeding it into the harmonics (distortion) generator. This results in the artificial harmonics added to the signal having frequencies at least one octave above the threshold of the highpass filter. The distorted signal is then mixed with the original, dry signal.

You can use the Exciter to add life to digital recordings. It is especially well suited to audio tracks with a weak treble frequency range. The Exciter is also useful for enhancing guitar tracks.

**Exciter Parameters**

- **Frequency slider and field:** Sets the cutoff frequency (in Hertz) of the highpass filter. The input signal passes through this filter before (harmonic) distortion is introduced.
- **Frequency display:** The graphic displays the frequency range that is used as the source signal for the process.
- **Input button:** When selected, the original (pre-effect) signal is mixed with the effected signal. If you disable Input, only the effected signal is heard.
- **Harmonics knob and field:** Sets the amount of the effected signal that is mixed with the original signal (expressed as a percentage). If the Input button is turned off, this has no effect on the signal. In most cases, higher Frequency and Harmonics values are preferable, because human ears cannot easily distinguish between the artificial and original high frequencies.
- **Color 1 and Color 2 buttons:** Click Color 1 to generate a less dense harmonic distortion spectrum. Click Color 2 for a more intense distortion. Color 2 also introduces more (unwanted) intermodulation distortions.
**SubBass**

The SubBass plug-in generates frequencies below those of the original signal—in other words, an artificial bass. The simplest use for the SubBass is as an octave divider, similar to Octaver effect pedals for electric bass guitars. Where such pedals can only process a monophonic input sound source of clearly defined pitch, SubBass can be used with complex summed signals as well. SubBass creates two bass signals, derived from two separate portions of the incoming signal. These are defined with the High and Low parameters.

**Warning:** Using the SubBass can produce extremely loud output signals! Choose moderate monitoring levels, and only use loudspeakers that are actually capable of reproducing the very low frequencies produced. Never try to force a loudspeaker to output these frequency bands with an EQ.

**SubBass Parameters**

- **High Ratio knob:** Adjusts the ratio between the generated signal and the original upper band signal.
- **High Center knob:** Sets the center frequency of the upper band.
- **High Bandwidth knob:** Sets the bandwidth of the upper band.
- **Graphic display:** Shows the selected upper and lower frequency bands.
- **Mix slider and field:** Adjusts the mix ratio between the upper and lower frequency bands.
- **Low Ratio knob:** Adjusts the ratio between the generated signal and the original lower band signal.
- **Low Center knob:** Sets the center frequency of the lower band.
- **Low Bandwidth knob:** Sets the bandwidth of the lower band.
- **Dry slider and field:** Sets the amount of dry (non-effected) signal.
- **Wet slider and field:** Sets the amount of wet (effected) signal.
Using the SubBass
Unlike a pitch shifter, the waveform of the signal generated by the SubBass is not based on the waveform of the input signal, but is sinusoidal (it uses a sine wave). Given that pure sine waves rarely sit well in complex arrangements, you can control the amount of (and balance between) the generated and original signals using the Dry and Wet sliders.

You define the two frequency bands (which the SubBass uses to generate tones) with the High and Low parameters. High Center and Low Center define the center frequency of each band, and High Bandwidth and Low Bandwidth define the bandwidth of each band.

The High Ratio and Low Ratio knobs define the amount that the generated signal is transposed for each band. This is expressed as a ratio of the original signal. As an example; Ratio = 2 transposes the signal down one octave.

Important: Within each frequency band, the filtered signal should have a reasonably stable pitch in order to be analyzed correctly.

In general, narrow bandwidths produce the best results, because they avoid unwanted intermodulations. Set High Center a fifth higher than Low Center, which means a factor of 1.5 for the center frequency. Derive the sub-bass to be synthesized from the existing bass portion of the signal, and transpose by one octave in both bands (Ratio = 2). Do not overdrive the process or you will introduce distortion. If you hear frequency gaps, move one or both Center frequency knobs, or widen the Bandwidth (of one or both frequency ranges) a little.

Tip: Be prudent when using the SubBass, and compare the extreme low frequency content of your mixes with other productions. It is very easy to go overboard with it.
The Utility plug-ins are handy tools that can help you with routine tasks and situations that you may encounter when producing music.

This includes the following tasks:

- Adjusting the level or phase of input signals (see “Gain” on page 147 and “Multichannel Gain” on page 148)
- Generating a static frequency or sine sweep (see “Test Oscillator” on page 149)

**Gain**

Gain lets you amplify (or reduce) the signal by a specific decibel amount. It is very useful when you are working with automated tracks during post-processing and want to quickly adjust levels, for example, when you have inserted another effect that doesn’t have its own gain control or when you want to change the level of a track for a remix version.

**Gain Parameters**

- *Gain slider and field*: Sets the amount of gain.
- *Phase Invert Left and Right buttons*: When selected, inverts the phase of the left and right channels, respectively.
- *Balance knob*: Adjusts the balance of the incoming signal between the left and right channels.
• **Swap L/R (Left/Right) button:** When selected, swaps the left and right output channels. The swapping occurs after the Balance in the signal path.

• **Mono button:** When selected, outputs the summed mono signal on both the left and right channels.

**Using Phase Inversion**

Inverting phase lets you combat time alignment problems, particularly those caused by recording with multiple microphones at the same time. When you invert the phase of a signal heard in isolation, it sounds identical to the original. When the signal is heard in conjunction with other signals, however, phase inversion has an audible effect. As an example, if you place microphones above and below a snare drum, you should invert the phase of the bottom microphone signal, so that it is in-phase with the top microphone signal.

**Multichannel Gain**

The Multichannel Gain lets you control the gain (and phase) of each channel of a surround track or bus independently.

• **Master slider:** Sets the master gain for the combined channel output.

• **Channel gain sliders:** Each slider sets the gain for its channel.

• **Phase Invert buttons:** When selected, the phase of the selected channel is inverted.

• **Mute buttons:** When selected, the channel is muted from the overall output.
Test Oscillator
The Test Oscillator generates a static frequency or a sine sweep. The latter is a user-defined frequency spectrum tone sweep.

Test Oscillator Parameters

- **Waveform buttons**: Select the type of waveform to be used for test tone generation.
  - The Square Wave and Needle Pulse waveforms are available as either aliased or anti-aliased versions. The latter is used in conjunction with the Anti Aliased button.
  - Needle Pulse is a single needle impulse waveform.
  - If the Sine Sweep button is active, the fixed oscillator settings in the Waveform section above are disabled.
- **Frequency**: Determines the frequency of the oscillator (default is 1 kHz).
- **Level**: Determines the overall output level of the Test Oscillator.
- **Sine Sweep button**: Activate to generate a user-defined frequency spectrum sine wave sweep.
  - **Time field**: Determines the duration of the sweep.
  - **Start Freq and End Freq fields**: Define the oscillator frequency at the beginning and end of the sine sweep.
  - **Trigger button**: Click to trigger the sine sweep. The behavior of the Trigger button can be switched via the menu below:
    - **Single**: Clicking the Trigger button triggers the sweep once.
    - **Continuous**: Clicking the Trigger button triggers the sweep indefinitely.