SAWStudio Levelizer

PLUG-IN FOR USE WITH
SAWSTUDIO AND SAWSTUDIOLITE

For PC Compatible Systems
Running Windows NT or Windows 2000

USER'S GUIDE

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OVERVIEW

The *Studio Levelizer* Plug-In module is a professional quality dynamics processing and mastering tool that offers enhanced *Gate*, *Compressor* and *Peak Limiter / Normalizer* functions to the SAWStudio user. This module also offers accurate *Gain Reduction Metering* of all combined functions, a *Remove Silence* function, and the ability to scan and display the highest 50 peaks within any range of MultiTrack audio.

Each setting can be adjusted with an incredibly powerful new interface control design that allows value scrolling, stepping and instant selection from a popup display of every possible value.

The *Gate* has adjustments for the *Attack* and *Release* times, the *Floor* level that the gate closes to, and the *Threshold* level. The *Threshold* setting can also be visibly displayed against any SoundFile in the SoundFile View. The *Gate* also has a reverse mode which can turn into a keyed ducking processor, or a special effects self-keyed reverse gate limiter.

The *Compressor* has adjustments for the *Attack* and *Release* times, the *Ratio* of compression, and the *Threshold* level. The *Threshold* setting can also be visibly displayed against any SoundFile in the SoundFile View.

Both the *Gate* and *Compressor* can be *Keyed* from their own track's data or from a mix of one or more other track's data, allowing for audio processing arrangements such as *Ducking* and *Remote Keying* to be easily setup.

The *Key* buffer can also be filtered with the use of powerful *Hi-Cut* and *Lo-Cut EQ* functions that allow the *Studio Levelizer* to be easily used as a *De-Esser* or other special purpose dynamics processor. Both *EQ* filters can cover the entire frequency range and can even be set to overlap in cutoff frequencies, allowing the user complete control over frequency bandwidth.

A special *Key Listen* switch allows you to listen to the actual *Key* data instead of the main audio data stream. This makes it easy to adjust the *EQ* and hear exactly what is triggering the *Gate* and *Compressor* functions.

The *Gain Reduction Meter* is an awesome addition to the module and offers an incredibly accurate means of displaying just how much

adjusting the module is doing. Its accurate tracking of levels, attack and release times, and its timing of the audio transients is unmatched in any software based Windows plug-in and rivals the reaction time of hardware displays.

The *Remove Silence* function can automatically split and region track data based on the *Gate* settings. For this function, the *Attack* time is used as a negative pre-threshold split point. If SAWStudio's *Auto Zero Cross Detection* option is active, the region splits will be perfectly adjusted to *Zero Cross* points making for incredibly silent region transitions. This function can also be used in a *Split / Mark* mode whereby all regions are split and marked for viewing before a simple keystroke finishes the job.

The *Find Peaks* function scans a range of audio data and displays a list of the highest 50 peak values within that range. The values are displayed as HEX data values, percent of clipping values, and decibel values. The actual sample position where the peak occurred is also displayed. Clicking on any of the listed items, automatically jumps the MultiTrack cursor to the exact sample position, allowing instant viewing and playback.

The **Studio Levelizer** module represents one of the most precision dynamics processing and mastering tools available anywhere, and can easily replace many of the high priced hardware based units currently on the market.

Have fun maximizing, finalizing and levelizing your audio projects.

A TOUR OF THE INTERFACE

The *Studio Levelizer* window displays all of the adjustable controls for each of the various dynamics processing tools. Clicking on the blue pyramid logo displays a popup options menu, and clicking on the blue bar in the upper right-hand corner of the window closes the module.

The control interface is a uniquely designed method of changing values with great action and display efficiency. Take a moment to learn and get a feel for adjusting the various controls.

Each control can be adjusted in one of three ways:

- Clicking in the right half of any control display will increment or decrement the setting by one value, depending on whether you click above or below the vertical center of the control display.

- Clicking and holding in the right half of any control display will change the mouse pointer to a plus/minus shape and place it in the vertical center position of the control display. If you now slide the mouse slightly up or down, the setting will scroll through the values in an increasing or decreasing manner. The farther you push the mouse off the vertical center, the faster the scroll will change the values. You may also tap the right mouse button at any time to snap the control to its default setting.

- Clicking in the upper left half of any control will popup a menu display of every possible setting for that control. You may then select any value with one mouse click. The current value is displayed with a checkmark and the default value is displayed as bold. Clicking outside the menu will cancel any selection change.

The *Studio Levelizer* is a powerful and complex processing tool. It takes practice and a certain feel to master its use. Have some patience when learning to apply its various functions. With practice and applied experience you will learn when and how to introduce its many powers into your audio projects.

THE GATE

The *Gate* is used to help eliminate unwanted background noise during low volume areas of your soundfile. The *Gate* starts with the volume down and raises the volume to full when the audio signal rises above the *Threshold* setting. When the signal level falls below the *Threshold* setting again, the *Gate* turns the volume back down.

By experimenting with different combinations of control settings, you can effectively clean up otherwise noisy soundfiles.

ATTACK

The *Attack* setting adjusts how quickly the volume is ramped up once the signal level rises above the *Threshold* setting. Speeds range from 0 to 2 seconds.

Setting the *Attack* time to zero can introduce a slight percussive effect when the *Gate* opens. This can add some clarity to an otherwise sloppy instrument attack. Adjusting the *Attack* time to 1 ms or higher eliminates this effect.

RELEASE

The *Release* setting adjusts how quickly the volume is ramped down once the signal level falls below the *Threshold* setting again. Speeds range from 20 milliseconds to 4 seconds.

By using a longer *Release* time, the *Gate* will operate smoother as the signal momentarily fluctuates above and below the *Threshold*.

By using a shorter *Release* time, you may effectively control the decay of sustained sounds and actually tighten and clean up the overall blend of a mix.

FLOOR

The *Floor* setting adjusts the lowest volume level that the Gate uses when it is triggered *Off*. Settings range from minus infinity to zero dB.

In some cases, gating a noisy track completely *On* and *Off* can be very noticeable and annoying in a mix as background hiss levels appear and disappear with the opening and closing of the *Gate*. Setting the *Floor* to an intermediate level, allowing a constant small amount of hiss to

remain throughout the mix, can sometimes create a more pleasing result.

THRESHOLD

The *Threshold* setting determines the signal level which triggers the *Gate* into action.

A special link to the SoundFile View provides a unique and powerful feature to assist in adjusting the *Threshold* settings. Double-clicking an *MT Entry* from the current track into the SoundFile View provides the opportunity for a visual display of the current *Threshold* setting referenced against the audio waveform. When adjusting the *Threshold* setting, a pair of horizontal lines for each channel will appear superimposed on the audio waveform in the SoundFile View window.

You may zoom and magnify the display to reference the *Threshold* setting directly against the unwanted noise in the waveform data.

Setting the *Threshold* too low can cause the *Gate* to remain open too much of the time, effectively allowing unwanted noise to leak through.

Setting the *Threshold* too high can cause the *Gate* to shut down prematurely, effectively chopping the audio and removing parts of the desired signal.

RVRS

The *Gate Rvrs* switch causes the gate to operate in reverse. When the signal is greater than the threshold, the gate closes down to the floor setting and when the signal falls below the threshold, the gate opens back up to full. This can be used to create a simple and powerful ducking processor by assigning the key to another track which will then control when this track lowers its level. Set the floor value to the desired ducked level and adjust the attack, release and threshold to control the overall timing. You can also use this mode with self keying to create a new sounding limiting effect. When the signal rises above the threshold on loud transients, the gate will lower the level to the fixed floor level. This sounds different than limiting or compression, because the transient is actually dropped below the level of the rest of the average signal.

GATE ON

The *Gate On* switch turns the *Gate* on and off. When the *Gate* is off, it adds no processing time or effect to the ***Studio Levelizer*** functions.

THE COMPRESSOR

The *Compressor* is used to help squeeze high volume areas of your soundfile to a lower volume. The Compressor starts with the volume at full and lowers the volume down when the audio signal rises above the *Threshold* setting.

ATTACK

The *Attack* setting adjusts how quickly the *Compressor* springs into action, and begins the volume squeeze once the signal level rises above the *Threshold* setting. Speeds range from 0 to 2 seconds.

Using a short *Attack* setting can effectively compress the attack transient and the sustain of a musical instrument, whereas a longer *Attack* setting lets the attack transient through at full volume compressing only the sustain portion of the signal. Experimenting with the *Attack* setting allows you to drastically alter the dynamic quality of different types of audio signals.

Generally, an *Attack* time between 10 and 100 ms is a good starting point for most types of audio signals.

RELEASE

The *Release* setting adjusts how quickly the *Compressor* releases its effect on the signal, and begins ramping the volume back up once the signal level falls below the *Threshold* setting again. Speeds range from 20 milliseconds to 4 seconds.

Setting the *Release* time too short can introduce a fluttering type of distortion to sustained areas of music as the *Compressor* continuously adjusts the volume up and down on signals that are riding just at the *Threshold* level. Using a slightly longer *Release* time will smooth out the overall audio results in these situations.

Generally, a Release time between 200 and 400 ms is a good starting point for most types of audio signals.

THRESHOLD

The *Threshold* setting determines the signal level which triggers the *Compressor* into action. The actual *Threshold* setting you choose will

depend on whether you are attempting to create a noticeable change in the sound or keep the effect as transparent as possible. The lower the *Threshold*, and therefore the deeper into the bulk of the audio, the more audibly noticeable the compressing effect will be.

A special link to the SoundFile View provides a unique and powerful feature to assist in adjusting the *Threshold* setting. Double-clicking an *MT Entry* from the current track into the SoundFile View provides the opportunity for a visual display of the current *Threshold* setting referenced against the audio waveform. When adjusting the *Threshold* setting, a pair of horizontal lines for each channel will appear superimposed on the audio waveform in the SoundFile View window.

You may zoom and magnify the display to reference the *Threshold* setting directly against the unwanted noise in the waveform data.

Setting a low *Threshold* value can cause the *Compressor* to effectively squash and/or thump the sound.

Setting a higher *Threshold* value can allow the *Compressor* to work more transparently, effectively holding the audio within volume limits with less audio artifacts.

RATIO

The *Ratio* setting determines how much the *Compressor* squeezes the audio (thereby lowering the volume) once it kicks into action. The higher the *Ratio,* the lower the volume is dropped.

The interaction of the *Compressor* controls allow you to alter the dynamics of your audio file drastically. By adjusting the *Threshold* down into the bulk of the audio material and using a large *Ratio* setting, you can effectively squash sharp transient sounds and balance out the overall volume level of the entire signal, or create unnatural audio effects by adjusting the settings to extremes.

By using a small *Ratio* setting, the effects of the compression will be more subtle, effectively smoothing out only the rough edges of the signal.

COMP ON

The *Comp On* switch turns the *Compressor* on or off. When the *Compressor* is off, it adds no processing time or effect to the **Studio Levelizer** functions.

THE KEY BUFFER

The *Key Buffer* is the actual source of audio data that triggers the *Gate* and *Compressor*. The audio data found in this buffer is used to determine whether the signal is above or below the *Threshold* level and thereby set the *Gate* and/or *Compressor* in motion.

There are three choices for the *Key*:

- **Self** - This setting causes the *Key* information to be taken from the same audio data that is being processed. A copy of the current audio buffer is passed through the *EQ* filters and then used as a trigger source for the processing.

- **A and B** - This setting causes the *Key* information to be taken from one of the two internal virtual buffers that can contain audio data sent and mixed from other tracks in the *MultiTrack*. The virtual buffer is passed through the *EQ* filters and then used as a trigger source for the processing.

- **Channels 1-72** - These settings allow you to key directly from any track on the *MultiTrack* with no concern about track order.

THE EQ HI CUT FILTER

This setting adjusts the *Hi Frequency Cut-Off* value that is applied to the *Key Buffer*. This filter is not applied directly to the audio signal path, and is used only as a filter for the *Keying* information.

This can be used to selectively control how the processing triggers. For example, setting this value to a frequency near 100 cycles, can cause the *Compressor* to only squeeze kick drums and bass guitar signals, leaving the rest of the audio untouched.

By adjusting the *Hi Cut* and *Lo Cut* frequencies at the same time, you can control the actual bandwidth of frequencies that are allowed to pass.

THE EQ LO CUT FILTER

This setting adjusts the *Lo Frequency Cut-Off* value that is applied to the *Key Buffer*. This filter is not applied directly to the audio signal path, and is used only as a filter for the *Keying* information.

This can be used to selectively control how the processing triggers. For example, setting this value to a frequency near 7000 cycles, can allow the *Compressor* to act as a *De-Esser*, effectively squeezing loud '*S*' signals, leaving the rest of the audio untouched.

By adjusting the *Hi Cut* and *Lo Cut* frequencies at the same time, you can control the actual bandwidth of frequencies that are allowed to pass.

THE SEND CONTROL

The *Send* control can be used to send a track's audio data to one of the internal virtual buffers that can be used by another ***Studio Levelizer*** on another track as *Keying* information.

The data is actually mixed to the destination buffer, allowing for multiple tracks to combine signals into the same virtual buffer.

THE KEY LISTEN SWITCH

This switch allows you to hear the Key Buffer audio data instead of the actual track audio data. This is very useful for monitoring the Key data while adjusting EQ filters, for example.

THE PEAK LIMITER

The *Peak Limiter* is used to limit large transient spikes to a specified volume. The **Studio Levelizer** can perform this process absolutely transparently, effectively removing large transient spikes without audibly affecting the overall sound of the waveform! Most hardware peak limiters can only chop the transient at the specified volume level, introducing many noticeable audible side effects. Using a unique design concept, the **Studio Levelizer** actually scales the volume level of each half cycle (positive and negative) of the waveform independently, from zero cross to zero cross, resulting in an instantaneous blending of the transient into the rest of the signal, with no noticeable thumping or pumping side effects.

The *Peak Limiter* does not utilize the *Key Buffer* in its processing.

PEAK LIMIT

The *Peak Limit* setting adjusts the ceiling threshold, which triggers the *Peak Limiter* into action, thus creating the maximum audio level that any transient in the file can attain. If set to *Off*, the *Peak Limiter* adds no processing time or effect to the **Studio Levelizer** functions.

You may also Right-click in the *Peak Limit* control to turn the control *On* and *Off* without losing its value setting.

A special link to the SoundFile View provides a unique and powerful feature to assist in adjusting the *Peak Limit* setting. Double-clicking an *MT Entry* from the current track into the SoundFile View provides the opportunity for a visual display of the current *Peak Limit* setting referenced against the audio waveform. When adjusting the *Peak Limit* setting, a pair of horizontal lines for each channel will appear superimposed on the audio waveform in the SoundFile View window.

You may zoom and magnify the display to reference the *Threshold* setting directly against the unwanted noise in the waveform data.

When adjusting this setting, attempt only to clip the peaks that stick out above the bulk of the audio material. If you place the setting down into the bulk of the audio you can introduce varying amounts of audible distortion.

If you desire to limit down into the bulk of the audio, apply the *Compressor* at the same time with an *Attack* time of around 10 - 100

ms and a *Release* time of around 200 - 300 ms. The slow *Attack* time allows the fast transients to slip through without being squashed, and the added *Release* time smoothly compresses and holds the sustained signals below the *Peak Limit* setting, thereby allowing much lower limiting thresholds without the audio distortion artifacts of too low of a *Peak Limit* setting.

Note that turning the *Peak Limiter* off also turns the *Normalizer* off. The *Normalizer* depends on a valid *Peak Limiter* setting in order to operate.

THE NORMALIZER

The *Normalizer* automatically calculates and determines the exact amount of volume adjustment (up or down) needed for the soundfile to attain a certain percentage of clipping.

The *Normalizer* does not utilize the *Key Buffer* in its processing.

You may also Right-click in the *Normalizer* control to turn the control *On* and *Off* without losing its value setting.

A special link to the SoundFile View provides a unique and powerful feature to assist in adjusting the *Normalizer* setting. Double-clicking an *MT Entry* from the current track into the SoundFile View provides the opportunity for a visual display of the current *Normalizer* setting referenced against the audio waveform. When adjusting the *Normalizer* setting, a pair of horizontal lines for each channel will appear superimposed on the audio waveform in the SoundFile View window.

You may zoom and magnify the display to reference the *Threshold* setting directly against the unwanted noise in the waveform data.

The *Normalizer* depends on a valid *Peak Limiter* setting in order to operate. If you desire to *Normalize* the volume without limiting any audio peaks, set the *Peak Limit* setting to the tip of the highest visible peak using the special SoundFile link view of the *Peak Limit* adjustment.

Combining the use of the *Peak Limiter*, and possibly the *Compressor*, and the *Normalizer* is an incredibly effective way to maximize the overall volume level of your final mixdowns. For example, in order to master a CD at the maximum level, apply the *Peak Limiter* to limit transient peaks to a level just above the bulk of the audio material, and add the *Normalize* function at a setting of around 98%. This should maximize the final output level with no audible side effects.

THE OUT LEVEL

The *Out Level* control can be used to adjust the overall volume of the **Studio Levelizer** output. You may use this to increase the overall average sound level of your waveform by compressing or limiting the peaks, and at the same time raising the overall gain of the entire output.

The *Out Level* processing is applied before the *Normalizer* and will not override the final *Normalizer* output.

THE GAIN REDUCTION METER

The *Gain Reduction Meter* displays the amount of gain reduction occurring at any moment, which encompasses the combined effect of all **Studio Levelizer** functions that are active. The meter is highly accurate and responsive, making it an invaluable aid in adjusting your settings.

It will track slight adjustments made to *Attack* and *Release* settings, and also *Threshold* and *Ratio* depth, giving you excellent visual indication of how the **Studio Levelizer** is affecting the audio stream.

THE TRACK NUMBER DISPLAY

The *Track Number Display* shows the reference track for the currently displayed settings. Opening a new **Studio Levelizer** on another track will change settings and track reference numbers on the same open window.

The **Studio Levelizer** window may remain open even after closing the Effects Patch Builder window, and the *Track Number Display* becomes the reference to which track settings are currently being viewed.

You may click in the *Track Number Display* to display a list of all tracks that have a **Studio Levelizer** assigned, and select another track to view.

There is also an *Auto Tracking* option at the top of the list which allows the display to change automatically with the current *Hot Track* selection in the MultiTrack view. This option may be saved with the module preferences.

Click anywhere in the **Studio Levelizer** window to close the popup list box without changing selections.

THE BYPASS SWITCH

The *Bypass* switch bypasses the ***Studio Levelizer*** without affecting other modules patched into this track.

THE OPTIONS ICON

Clicking on the blue pyramid *Options* icon displays a small popup menu containing the following options:

HELP CONTENTS

This option displays this Help File.

SAVE PREFERENCES

This option allows you to save preferences for the *Studio Levelizer* module which includes the load and save file paths and the current *Studio Levelizer* window position.

UNDO EDL (KEEP STUDIO LEVELIZER OPEN)

This option activates the *Undo EDL* function of the SAWStudio program. Unlike the normal *Undo* function in SAWStudio, which normally closes all open FX modules, this option keeps the active *Studio Levelizer* onscreen.

This makes it easy to undo operations such as the *Remove Silence* function, in order to adjust a setting and try again.

LOAD PRESETS

This option loads a previously saved preset file containing all adjustable settings for a specific *Studio Levelizer* setup. Selecting this option opens the file dialog box. Select the folder and file name of the preset file you wish to load and all settings will be set accordingly. *Studio Levelizer* preset files use the **.lvl** extension.

SAVE PRESETS

This option saves a preset file containing all adjustable settings for the current *Studio Levelizer* setup. Selecting this option opens the file dialog box. Select the folder and enter a file name for the preset file you wish to save and all settings will be saved to the preset file. *Studio Levelizer* preset files use the **.lvl** extension.

PEAK LIMIT / NORMALIZE DISPLAY AS DB

This option displays *Peak Limit* and *Normalize* settings in decibels, instead of percentages.

REMOVE SILENCE (SPLIT/MARK)

This option operates on the current ***Studio Levelizer*** track, using the current *Gate* settings to actually *Split* and *Mark Regions* at each *Threshold* crossing point. The splits are created according to the following rules:

- Audio sections below the *Threshold* are split and marked for removal. The split point is set at the end of the *Release* cycle.

- Audio sections above the *Threshold* are split and unmarked for keeping. The split point is set in front of the actual *Threshold* crossing point by the amount equal to the *Attack* time.

- If SAWStudio's Auto Zero Cross Detection option is active, each split point is adjusted to the nearest zero cross, creating extremely clean and quiet *Region* boundaries.

Activating this option causes a pre-scan starting from the MultiTrack cursor position to the end of the track, or from the beginning to the end of a marked area. At the end of the pre-scan, the *Regions* are created, leaving you in *Select Mode* to view the results. If all looks well, a simple press of the *Delete* key will remove the silence areas.

If you decide to change your mind, use the *Undo EDL* option on the ***Studio Levelizer***, and you will be instantly returned to your previous condition, ready to adjust the settings and try again.

REMOVE SILENCE (AUTOMATIC)

This option operates on the current ***Studio Levelizer*** track, using the current *Gate* settings to actually *Split* and *Remove Regions* at each *Threshold* crossing point. The splits are created according to the following rules:

- Audio sections below the *Threshold* are split and removed. The split point is set at the end of the *Release* cycle.

- Audio sections above the *Threshold* are split and kept. The split point is set in front of the actual *Threshold* crossing point by the amount equal to the *Attack* time.

- If SAWStudio's Auto Zero Cross Detection option is active, each split point is adjusted to the nearest zero cross, creating extremely clean and quiet *Region* boundaries.

Activating this option causes a pre-scan starting from the MultiTrack cursor position to the end of the track, or from the beginning to the end of a marked area. At the end of the pre-scan, the *Regions* are created, removing the silence areas.

If you decide to change your mind, use the *Undo EDL* option on the **Studio Levelizer**, and you will be instantly returned to your previous condition, ready to adjust the settings and try again.

FIND PEAKS

This option scans the current **Studio Levelizer** track, and finds the highest 50 peaks.

Activating this option causes a pre-scan starting from the MultiTrack cursor position to the end of the track, or from the beginning to the end of a marked area. At the end of the pre-scan, a listbox opens displaying the highest 50 peaks. The peaks are displayed as follows:

- The absolute HEX value of the peak.

- A [CLP] indicator for clipped peaks that go above 100%.

- The percent of clipping the value represents.

- The dB representation for the peak value, relative to 0 dB at clipping.

- The actual sample position where the peak occurs.

24-bit audio data will display values above 100% for clipped peaks. You can easily use these values to determine how much you would need to drop the volume to keep the peaks from clipping.

16-bit audio data on the output track red F-Res FX patch set to 16-bit resolution, usually does not display values above clipping. This data will either wrap-around to become a negative peak or will be flattened at the maximum positive or negative setting by the MultiTrack Soft Clipping option.

THE LEVELIZER

RE-DISPLAY LAST PEAKS LIST

If you close the *Find Peaks* results listbox, this option re-displays the last list, without the need for a new scan.

ABOUT

This option displays the *About* information window including current version information.