ArtsAcoustic CL Series Manual
CL-1, CLMS-1, CLMS-1 XL

Thank you for choosing an ArtsAcoustic product. We have been putting a lot of research, heart and passion into this product and we hope that you greatly enjoy it as much as we do.

Contents

Introduction

First let us say thank you for purchasing the ArtsAcoustic CL Series. You are now able to use and enjoy a high end CL Series giving you a stunning amount of dynamics related sound capabilities coupled with most accurate analog authenticity.

What exactly is the ArtsAcoustic CL Series?

Easily spoken it is a bundle of currently three analog modeled, optoelectronic dynamics processors (compressor/limiter) featuring internal and external side chaining. Even if they share the same core algorithm each Plugin offers special functionality. To keep the features useable and to avoid an overflow of functionality we have developed three independent Plugins, that will provide you with features depending on individual tasks.

While the CL-1 is a solid and easy to use stereo Compressor-Limiter, the CLMS-1 is an enhanced version additionally providing simple but cleverly implemented Mid/Side possibilities – this often does miracles to a stereo signal.

The CLMS-1 XL therefore is the "flagship" version featuring full Mid/Side and dual mono (Left/Right) functionality. Two compressors independently working on the mid and the side or left and right signal. That means that on both, the audible to-be-compressed signal, as well as on the control signal, you can select to use the mid, the side, or any mix of both signals separately. You are able to compress two mono signals totally independent, being able to select freely, which signal controls the compressors: mid, side or a mix of both signals. Of course it provides external side chaining or each, the mid and the side compressor.

For an in depth description on Mid/Side techniques see chapter "Mid/Side techniques".

Philosophy of the ArtsAcoustic CL Series

There are already loads of compressors and limiters out in Plugin Land. So why did we decide to provide one more compressor/limiter bundle?

We are sure that this question is easily answered when you use the Plugins the first time. It is the sound. Being long time musicians/engineers ourselves, we have a tendency to absolutely love the sound of certain old analog gear. Some of those rare units are just delivering color, depth and definition to the signal that is still hard to be found in digital gear.

The key factor is the following: when working completely analog millions of very small side effects of the components that are built into a mixing console, a compressor or an EQ are altering the signals phase, the dynamic, adding small amounts of saturation, etc. . So there are an awful load of tiny, individually not noticeable things happening to a signal (except just adding a saturation wherever in the signal path) that adds up to be audible in summary.

This is why we all love the sound of analog gear. In contrary in the digital domain the signal is frozen as is as soon as it passed the converters. Having the signal recorded into your DAW the signal more or less stays unaltered unless you alter the signal yourself. If you do not simulate the analog behavior yourself the signal will stay as clean as you recorded it.

In order to get that analog "warmth" and "depth" added to your digital signal you must add it yourself. Therefore you would have to know exactly what happens with the signal in the analog domain, but unless you are not a deeply involved into the actual creation of analog circuits is most likely not the case.

Here our work begun: working with dynamic processors since more than two decades, we analyzed every promising compressor we could get into our hands to study the signal paths, the build in units, etc. . Based on this comprehensive knowledge the ArtsAcoustic CL Series came to life. It is not a clone of one particular dynamic processor, but some kind of a summary statement based on every analog dynamic processor we found to be worthy.

We bring the most authentic analog sound to these Plugins. We even modeled the function of some circuits of the crucial parts in a dynamic processor, such as resistors, capacitors, etc. . Now we provide the exact same effects to the Plugins core algorithm which is why the outcome is stunningly real, as you will soon discover yourself.

While several compressors in the digital domain claim to deliver this beautiful analog sound to our ears there is still room for improvement. This is our contribution to bring analog sound to the digital domain, hopefully raising the bar again.

System requirements

- This software is a Plugin for Microsoft Windows / Mac OS X (further details below). You will need a host software where you can run it in. It cannot be run as a stand-alone application.
- The Plugin relies on the VST™ 2.4 (Virtual Studio Technology) specification by Steinberg™ (Windows and Mac) or the Audio Units Standard by Apple (Mac Only). It can only be run in a host that supports at least one of these standard architectures.
- Please refer to your hosts manual to assure that your host software supports either the VST-Technology or the Audio Units Standard.

Windows	Macintosh
Required: • Pentium / Athlon 800 MHz • 128 MB RAM • Windows XP, Windows Vista (32/64 bit), Windows 7 (32/64 bit) • Windows MME compatible audio hardware • VST 2.4 (32/64 bit) compatible Host • Display Resolution 1024 x 768 pixels	Required: • Macintosh with G4 or Core Solo Processor • 128 MB RAM • OS X Version 10.5.x – 10.6.x • CoreAudio compatible audio hardware • VST 2.4 (32 bit) or AU (32/64 bit) compatible Host • Display Resolution 1024 x 768 pixels
Recommended: • Pentium / Athlon 1.8 GHz or faster • 512 MB RAM • Display Resolution 1152 x 864 pixels, dual monitor setup • ASIO compatible sound card recommended	Recommended: • Power Mac G4 Dual 1.25 GHz or faster • 512 MB RAM • Display Resolution 1152 x 864 pixels, dual monitor setup

Installation

This section will assist you with installing the ArtsAcoustic CL Series on a Windows system.

Installation - Windows

Double-click the installer to start the installation.

Step 1 – Welcome
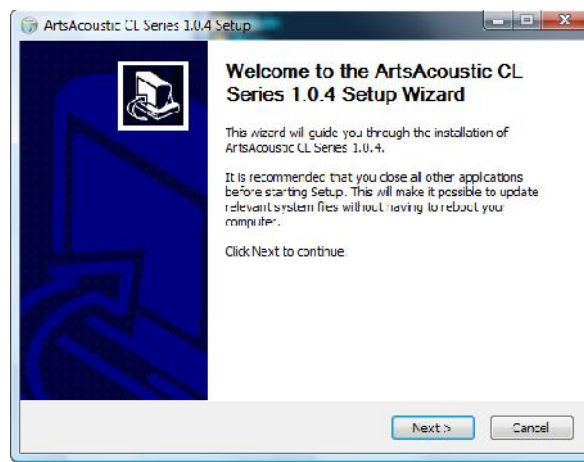
Click Next> to continue.

Step 2 – License Agreement
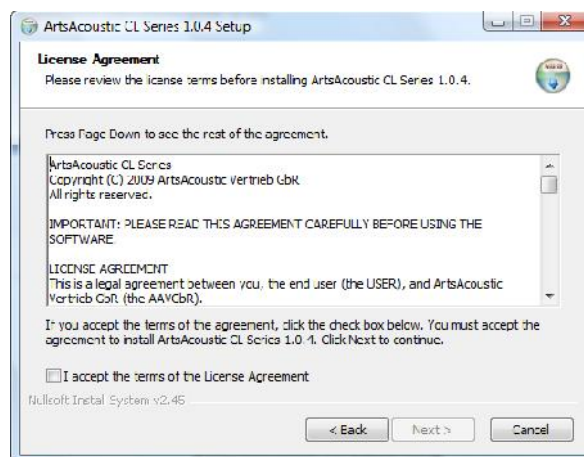
Please read the license agreement and click the checkbox if you agree to the terms.
Click Next> to continue.

Step 3 – Destination Folder
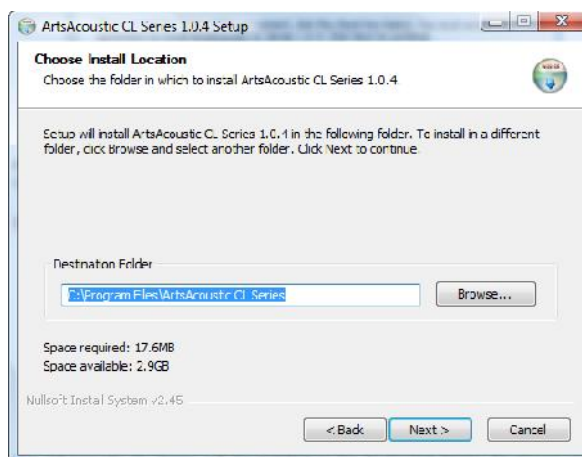
The ArtsAcoustic CL Series will be installed in the Destination Folder. To choose another location, click Browse. Click Next> to continue.

Step 4 – VSTPlugins Folder
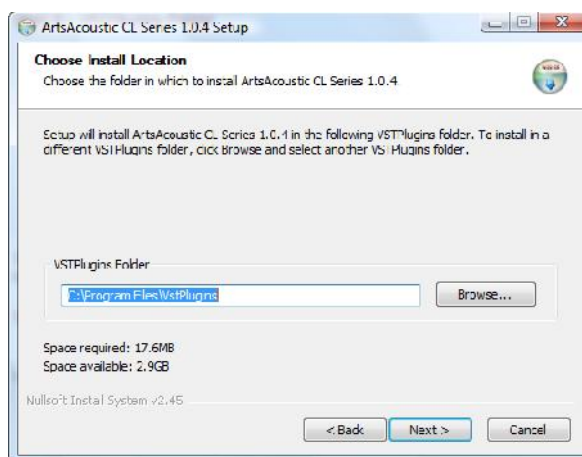
The VST part of the Plugin will be installed in the VSTPlugins Folder. To choose another location, click Browse. Click Next> to continue.

Step 5 – User Preset Folder
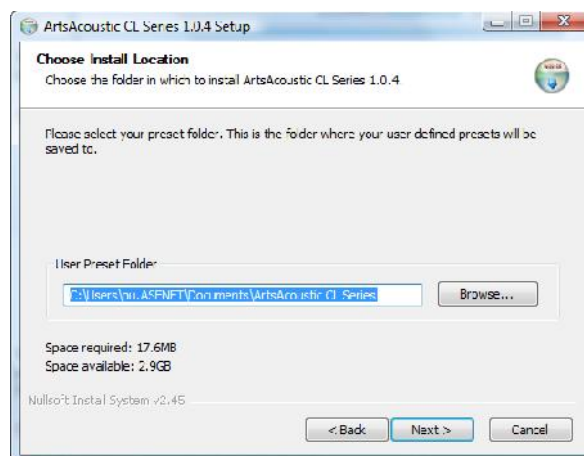
The ArtsAcoustic CL Series User Presets will be saved in the User Preset Folder. To choose another location, click Browse. Click Next> to continue. You can change the User Preset Folder afterwards within the Plugin editor.

Step 6 – Start menu Folder
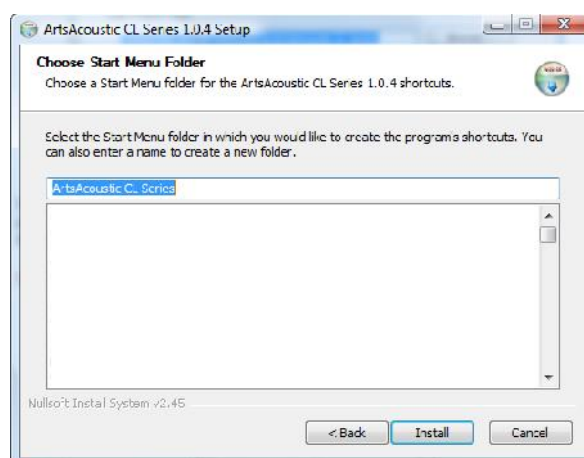
Click Install to continue.

Installation - Mac

Double-click the installer package icon to start the installation.

Step 1 – Install/Uninstall

Choose what you want to do. You can either install the actual version or uninstall a older version.

Step 2 – Introduction
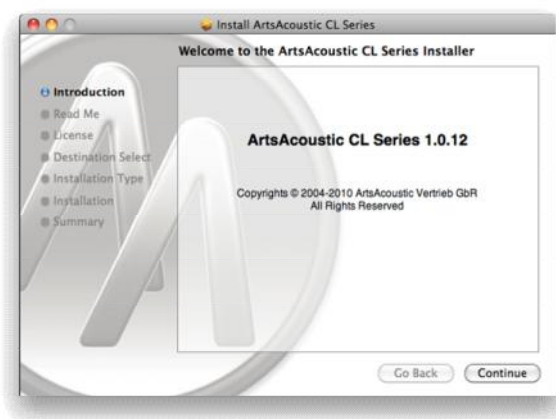
Click 'Continue' to continue.

Step 3 – Read Me
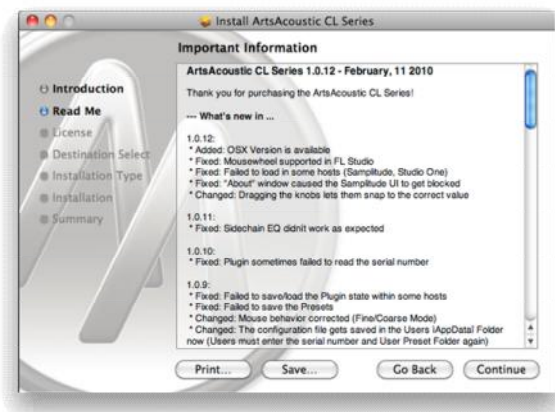
Click 'Continue' to continue.

Step 4 – License
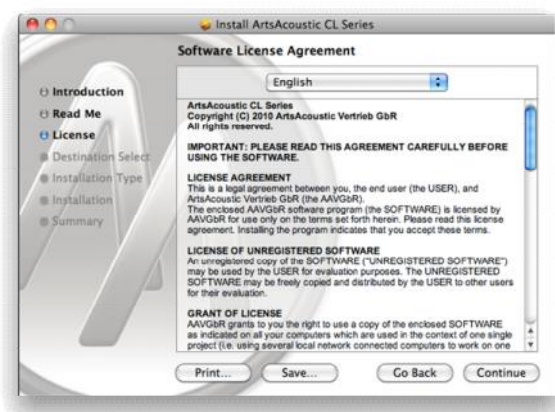
Click 'Continue' to continue.

Step 5 – Select Destination

Click 'Continue' to continue.

Step 6 – Installation Type
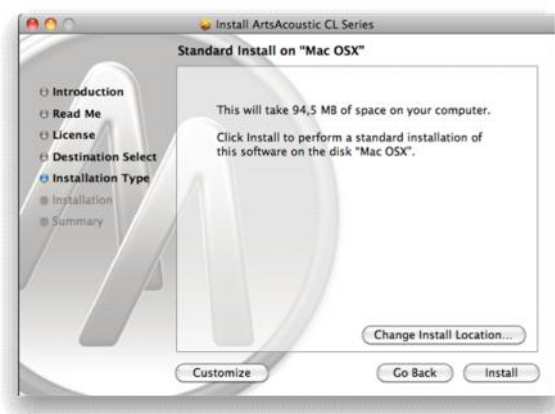
If you want to individually select the components to be installed click 'Customize'. Here you can choose if you want to install the AU and/or VST version of the ArtsAcoustic CL Series.
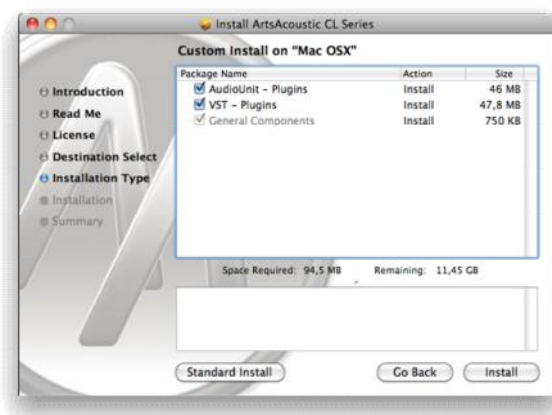
If you are performing a new installation click 'Install'. Otherwise click 'Upgrade' for upgrading an existing installation.

Step 7 – Summary

After performing the installation a 'Summary' will be displayed.

Copy protection

The Plugin has to be activated using a serial number. The serial number has been sent to you with your delivery notification email.

Open the file including your serial number which is attached to the delivery notification email and copy the complete content into your Clipboard.

Load any Plugin of the ArtsAcoustic CL Series into your preferred host. In case the Plugin finds a valid serial number in the Clipboard it will register the ArtsAcoustic CL Series without any further actions on your side.

Otherwise you can start the registration process from within the Plugin by either clicking on the text "Unregistered Copy."
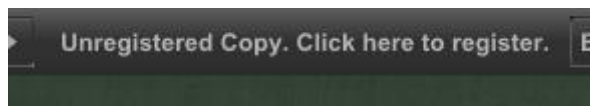
or using the file menu "File|Register…".
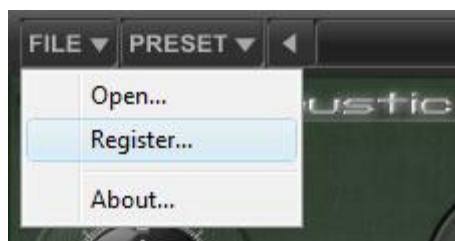
Open the file including your serial number which is attached to the delivery notification email.
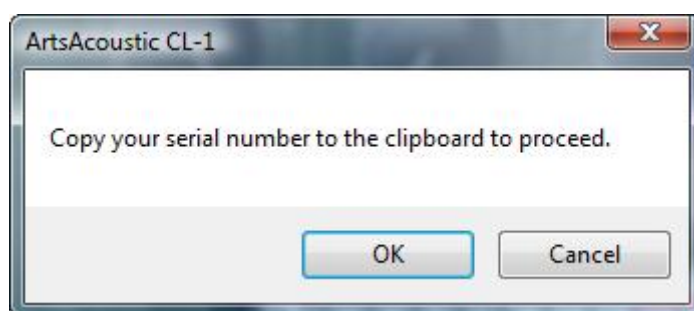
Copy the complete content into your Clipboard and return to the Plugin. Press the button OK and it will register the ArtsAcoustic CL Series.

In case the serial number is not valid the Plugins of the ArtsAcoustic CL Series will not process any sample data.

Uninstall

- On Windows use the system tools to uninstall the ArtsAcoustic CL Series.
- On Mac OS X use the Uninstaller which is located in the Installation *.dmg file.

Basics

Compression/Limiting

This is just a very basic explanation of how a compressor/limiter works. Depending on the compressor/limiter there is usually a lot more going on in the specific circuits.

Compression

The intention of using a compressor is to reduce and/or control the dynamic range of a signal over the time period it lasts. Loud signals are lowered while quiet signals are passing unaltered. To be able to do this you first need to specify the point in level at which the signal should be reduced, i.e. at which the compressor should start to work.

Threshold

This is done by a parameter called Threshold. If the input signal is louder as the Threshold, the compressor starts to work, if it is lower, the compressor doesn`t work, leaving the signal as it is.

Ratio

The Ratio defines the amount of the level reduction, depending on how much the signal is over the Threshold. For example with a Ratio of 4:1, when the input level is 4.0 dB over the Threshold, the signal level at the compressors output will be 1.0 dB over the Threshold, essentially providing 3.0 dB in level reduction.

Attack and Release

In order to control the time at which the signal should be lowered, as well as the time at which it should be recovering to normal level again, the parameters Attack and Release are provided. The Attack determines the time passing, before the compressor works when the signal is over the Threshold. The Release controls the time the signal needs to recover back to normal level, after the signal completely passed the attack stage.

Limiting

Limiting is very similar to compression. Usually there are two main differences. While the Ratio of a compressor is adjustable, the limiter mostly features a fixed infinite ratio. As the name "limiting" already suggests, the input signal should be limited to a maximum of level you specify. Ideally it shouldn`t pass the maximum level you adjust with the Threshold. so a signal that is above the Threshold will be reduced exactly to the Thresholds level. The second difference is, that there is no Attack parameter. Again as the signal should not exceed a certain level you set up with the Threshold the reduction should be instantly so there is no need for an attack time. Therefore the attack time in a limiter is fixed at fastest possible point.

Mid/Side techniques

The Mid/Side technique is very old. It is used by professional engineers since a long time for both, recording and mixing, to gain control over the stereo image of a signal. By using this technique it is possible to control the stereo image of a stereo signal (making only the side signal louder is a common usage), as well as it is possible to independently process the mid and/or the side signal.

It just works on real stereo signals: signals that in some way differ on both channels make up the stereo signal. In other words: even if a signal is recorded on a stereo channel in your host but the recorded signal is mono, this technique does not have any effect, as there wouldn`t be any side information existing to be extracted.

One thing is very important

A stereo signal has only 2 channels. Whatever one realizes in the center, is actually a kind of phantom sound. It is not coming from a channel in the middle, but from both speakers. The reason why one realizes the signal coming from the center is because these parts of the signal are absolutely identical on both channels. As soon as certain frequencies or levels of the signal differ one side compared to the other, one will realize the whole signal as stereo.

How are we able to extract the mid from the side signals?

Fact is that we aren`t able to do so at all. Falsely Mid/Side processing is commonly described as having the possibility to extract the center and the side signals separately. This is of course not possible as we stated earlier. We would at least need 3 channels to do that, the left, the right and the center channel. Since stereo is made up of only 2 channels it isn`t possible.

How is this Mid/Side processing working?

To split a stereo signal into Mid/Side there are two things to do. To get the mid signal, you sum up the left and the right channel (or vice versa). In other words it is simply the mono signal of the stereo signal. To get the side signal you sum up the left and the phase inverted right channel. Again in other words you end up with the difference of both channels (the equal signals are cancelled by the phase inversion). This "exclusive information" of the stereo image is summed up as a mono signal again. Now we have the mono mid signal and the mono side signal, thus we are able to process them independently.

However once we have processed the mid and the side signals as we like, we have to decode the signal back to stereo. To get back the left side of the stereo signal we have to sum up the mid and the side signal and cut it into half. To get back the right side of the stereo signal, we have to sum up the mid and the phase inverted side signal and cut it into half. You now have a normal stereo signal again.

CL Series - Overview

Feature

All Plugins feature the following:

Feature	Description
Plugin category	Effect
Plugin Type	Dynamics
Sample rate	22050 to 196000 Hz
Input: dynamic range	-67.0 dB – 0.0 dB (sweet spot) -69.9 dB – +30.0 dB (max.)
32 bit processing	Supported
64 bit processing	Supported
32 bit precision	Supported
64 bit precision	Supported
Latency	0.73 ms

Tips and Tricks

- The Plugins are not to be used as Send Effects. Due to the nature of the Plugins modeling of analog components the phase of the signal is changed. Better use the included Dry/Wet knob to mix the signals.

CL-1 Optoelectronic Compressor-Limiter

This Plugin is a simulation of a conventional general purpose compressor/limiter circuit. Use this one, if you don`t need any Mid/Side functionality, but a solid compression.

Signal Flow Diagram

Features and Functions

General Functions

File Menu

Further details can be found in chapter "Preset Manager".

Preset Menu

Further details can be found in chapter "Preset Manager".

Bypass

This button will bypass the complete processing. This can be helpful for A/B controlling the original and the compressed signal.

Compressor

Input

This parameter has two functions. First it levels the signal that is entering the compressor, it is the so called "Input Gain Stage". Second this gain stage is modeled after a tube circuit. The tube adds harmonics from subtle to heavy, when the input signal is over 0.0 dB. Harmonic coloring is often preferred, so simply play around with the Input Gain. However it is crucial to lower the compressors Output Gain accordingly, in order to stay below digital clipping. The controllable range of the Input Gain is -40.0 dB to +40.0 dB. The internal side chain signal is only affected when the Input Gain is below 0.0 db and external side chain signal is not affected by the Input Gain.

> We suggest to always normalize the input signal outside the compressor, but if that is not possible this is the parameter to be used instead.

Tube Indicator (LED)

This led indicates the tube activity. As soon as the tube adds harmonics to the signal, this led will lit up. Signals passing the tube unaltered will not trigger it, so it remains off.

Threshold

The Threshold is the level that must be exceeded by the incoming side chain signal, in order for the compressor to work. The Threshold level could be thought of as the "sensitivity" of an compressor. The controllable range of the Threshold is 0.0 dB to -70.0 dB.

Ratio

The Ratio is the degree to which the compressor is reducing the dynamic range. Ratio therefore indicates the relative gain reduction between the incoming audio signal and the outgoing compressed audio signal. A ratio of 2:1 means that an audio signal of e.g. +2.0 dB gets compressed to an audio signal of just +1.0 dB. The higher the Ratio, the more the signal will be reduced. The controllable range of the Ratio is 1:1 to -5:1. When setting the Ratio to the middle of the controllable range the Ratio is set to "Limit".

> All our compressors feature a technique called "negative ratio". This is a very rare implementation, which actually allows you to reduce the signal below the Threshold. It can be used for drastic gain reductions often heard on drums.

Attack

This adjusts the time that passes before the compressor takes action. The controllable range of the Attack is 0.0 % (~0.1 ms) to 100.0 % (~500.0 ms).

Release

The Release determines the time that the signal needs to recover to its original level. The controllable range of the Release is 0.0 % (~20.0 ms) to 100.0 % (~1000.0 ms).

Dry/Wet

With this parameter you can mix back the original, completely untreated input signal.

> It is often called "New York style compression" to do so. The intention is, to compress the signal in a drastic way but use only a small portion of the wet signal and mix it back to the dry signal. That way one can achieve a very natural sound, featuring both, the captured sound of the compressor itself and the natural sound of the dry signal.

Output

This adjusts the output of the compressors Dry/Wet mix stage. With this parameter you define the level that feeds the limiter. Be careful with this parameter when the limiter is "Off", you need to carefully readjust this parameter in order to not overshoot 0.0 dB. The controllable range of the Output Gain is -40.0 dB to +40.0 dB.

> The Output Gain can be used to either level the compressed audio signal to 0.0 dB (compensate the gain reduction) or if the Input has been driven into the Tube Circuit to set back the compressed and saturated audio signal to a usable range.

Side Chain

This section exclusively affects the internal/external control signal.

External/Internal Signal

Here you specify which signal is used to control the compressor. When set to "Internal" the control signal is grabbed right from the main stereo input, before the "Input Gain Stage". When set to "External" the signal to control the compressor is grabbed from the discrete external stereo input. Whatever signal you feed into these inputs will control the compressor. Please refer to your hosts manual in order to learn how to route the external side chain control signal.

> In case your host does not support more than two input channels the external side chain is also using the internal signal.

EQ On/Off

Turns the equalizer for the control signal (internal/external side chain) on and off.

Highpass/Lowpass Filter

Whatever control signal you use it is routed through these two EQ bands unless you switched off the EQ. It features a 12dB Highpass as well as a 12dB Lowpass.

- The Highpass lets the frequencies above the adjusted value pass through, while cutting the frequencies below it.
- The Lowpass lets the frequencies below the adjusted value pass through, while cutting the frequencies above it.

Whenever you want the compressor to react only on specific frequencies of the control signal, use this EQ to isolate these frequencies. If the knobs are set to "0", the signal passes untreated.

Solo

In order to directly hear how the EQ affects the control signal turn this on. Now you will hear the side chain signal.

Limiter

The limiter sits directly after the compressor output. Therefore the compressor output knob controls the limiter input. The limiter is operating as a "brick wall limiter" and does not allow any signal to pass above -0.1 dB.

On/Off

The switch is used to turn on/off the Limiter. The Attenuation will not be affected by this switch.

Type

The limiter can operate in two modes: Peak and RMS level detection.

Release

The Release determines the time that the signal needs to recover to the compressors output level. The controllable range of the Release is 0.0 % (Clipping), 0.01 % (Peak: ~50.0 ms; RMS: ~1.0 ms) to 100.0 % (~1500.0 ms).
When set to 0.0 % the release is completely disabled and effectively turning the limiter into sample accurate brick wall clipping mode. In this operation mode the limiter Release is extremely fast, but you must take care about the level of the signal you feed into the limiter since it can introduce unpleasant digital distortion.

> Feed the limiter with care, when it is clipping mode.

Attenuation

Attenuation reduces the overall output of the Plugin, it sits right at the end of the complete signal path. The controllable range of the Attenuation is -40.0 dB to 0.0 dB.

This parameter is used for two main tasks:
1. If you compress heavily and drive the limiter hard, you might end up with a lot of average loudness. Most likely this will put you in a situation where you would have to drastically lower the level slider of your hosts channel and then you might not have enough range on that level slider to mix properly. Pull the Attenuation knob down in order to increase the range of your hosts channel level slider.
2. When compressing and limiting, you most likely end up with a slightly louder perceived signal, even if both, the dry and the Plugin output signal are at -0.1 dB in peak level. In order not to get fooled by the loudness increase, you should finally reduce the output to a point that roughly equals the perceived loudness of the dry signal. Use the Bypass switch in the upper right of the user interface to compare both signals frequently. Once you are satisfied with the result, you can turn Attenuation back to 0.0 dB.

Metering

This section visually displays what happens to the signal at various stages. Use this to observe how the input signal is treated by the Plugin.

Input

The input button turns the metering to show all input stages.

Reduction

The reduction button turns the metering to show the individual gain reduction of the compressor and the limiter.

> When the limiter release is set to "brick wall clipping" mode the reduction of the limiter can`t be displayed. You wouldn`t see the "needle" anyway as the brick wall clipping is sample accurate and therefore way too fast to be displayed.

Output

The output button turns the metering to show all output stages.

CLMS-1 Optoelectronic Mid/Side Compressor-Limiter

The CLMS-1 is an enhanced version of the CL-1. Essentially providing you with basic Mid/Side capabilities in the control and the audible signal path.

Signal Flow Diagram

Features and Functions

General Functions

File Menu

Further details can be found in chapter "Preset Manager".

Preset Menu

Further details can be found in chapter "Preset Manager".

Bypass

This button will bypass the complete processing. This can be helpful for A/B controlling the original and the compressed signal.

Compressor

Input

This parameter has two functions. First it levels the signal that is entering the compressor, it is the so called "Input Gain Stage". Second this gain stage is modeled after a tube circuit. The tube adds harmonics from subtle to heavy, when the input signal is over 0.0 dB. Harmonic coloring is often preferred, so simply play around with the Input Gain. However it is crucial to lower the compressors Output Gain accordingly, in order to stay below digital clipping. The controllable range of the Input Gain is -40.0 dB to +40.0 dB. The internal side chain signal is only affected when the Input Gain is below 0.0 db and external side chain signal is not affected by the Input Gain.

> We suggest to always normalize the input signal outside the compressor, but if that is not possible this is the parameter to be used instead.

Tube Indicator (LED)

This led indicates the tube activity. As soon as the tube adds harmonics to the signal, this led will lit up. Signals passing the tube unaltered will not trigger it, so it remains off.

M/S Mix

With this knob you set up the balance between the mid and the side signal before the audible signal is send into the compressor stage.

Threshold

The Threshold is the level that must be exceeded by the incoming side chain signal, in order for the compressor to work. The Threshold level could be thought of as the "sensitivity" of an compressor. The controllable range of the Threshold is 0.0 dB to -70.0 dB.

Ratio

The Ratio is the degree to which the compressor is reducing the dynamic range. Ratio therefore indicates the relative gain reduction between the incoming audio signal and the outgoing compressed audio signal. A ratio of 2:1 means that an audio signal of e.g. +2.0 dB gets compressed to an audio signal of just +1.0 dB. The higher the Ratio, the more the signal will be reduced. The controllable range of the Ratio is 1:1 to -5:1. When setting the Ratio to the middle of the controllable range the Ratio is set to "Limit".

> All our compressors feature a technique called "negative ratio". This is a very rare implementation, which actually allows you to reduce the signal below the Threshold. It can be used for drastic gain reductions often heard on drums.

Attack

This adjusts the time that passes before the compressor takes action. The controllable range of the Attack is 0.0 % (~0.1 ms) to 100.0 % (~500.0 ms).

Release

The Release determines the time that the signal needs to recover to its original level. The controllable range of the Release is 0.0 % (~20.0 ms) to 100.0 % (~1000.0 ms).

Side Pan

Some signals introduce an uneven stereo balance and therefore one side is louder than the other. When using compression this may even become more audible. Use this slider to adjust the stereo balance. This slider affects exclusively the side signal, the mid signal stays unaltered.

Dry/Wet

With this parameter you can mix back the original, completely untreated input signal.

> It is often called "New York style compression" to do so. The intention is, to compress the signal in a drastic way but use only a small portion of the wet signal and mix it back to the dry signal. That way one can achieve a very natural sound, featuring both, the captured sound of the compressor itself and the natural sound of the dry signal.

Output

This adjusts the output of the compressors Dry/Wet mix stage. With this parameter you define the level that feeds the limiter. Be careful with this parameter when the limiter is "Off", you need to carefully readjust this parameter in order to not overshoot 0.0 dB. The controllable range of the Output Gain is -40.0 dB to +40.0 dB.

> The Output Gain can be used to either level the compressed audio signal to 0.0 dB (compensate the gain reduction) or if the Input has been driven into the Tube Circuit to set back the compressed and saturated audio signal to a usable range.

Side Chain

This section exclusively affects the internal/external control signal.

External/Internal Signal

Here you specify which signal is used to control the compressor. When set to "Internal" the control signal is grabbed right from the main stereo input, before the "Input Gain Stage". When set to "External" the signal to control the compressor is grabbed from the discrete external stereo input. Whatever signal you feed into these inputs will control the compressor. Please refer to your hosts manual in order to learn how to route the external side chain control signal.

> In case your host does not support more than two input channels the external side chain is also using the internal signal.

EQ On/Off

Turns the equalizer for the control signal (internal/external side chain) on and off.

Highpass/Lowpass Filter

Whatever control signal you use it is routed through these two EQ bands unless you switched off the EQ. It features a 12dB Highpass as well as a 12dB Lowpass.

- The Highpass lets the frequencies above the adjusted value pass through, while cutting the frequencies below it.
- The Lowpass lets the frequencies below the adjusted value pass through, while cutting the frequencies above it.

Whenever you want the compressor to react only on specific frequencies of the control signal, use this EQ to isolate these frequencies. If the knobs are set to "0", the signal passes untreated.

Solo

In order to directly hear how the EQ affects the control signal turn this on. Now you will hear the side chain signal.

M/S Mix

This parameter lets you select which signal should be used to control the compressor.

- If the knob is fully left, the mid signal is used to control the compressor.
- If the knob is fully right, only the side signal is used to control the compressor.
- If the knob is anywhere in between, a mix of both the mid and the side signal is used.

> The CLMS-1 solely compresses a stereo signal and not the Mid/Side encoded signal.

Limiter

The limiter sits directly after the compressor output. Therefore the compressor output knob controls the limiter input. The limiter is operating as a "brick wall limiter" and does not allow any signal to pass above -0.1 dB.

On/Off

The switch is used to turn on/off the Limiter. The Attenuation will not be affected by this switch.

Type

The limiter can operate in two modes: Peak and RMS level detection.

Release

The Release determines the time that the signal needs to recover to the compressors output level. The controllable range of the Release is 0.0 % (Clipping), 0.01 % (Peak: ~50.0 ms; RMS: ~1.0 ms) to 100.0 % (~1500.0 ms).

When set to 0.0 % the release is completely disabled and effectively turning the limiter into sample accurate brick wall clipping mode. In this operation mode the limiter Release is extremely fast, but you must take care about the level of the signal you feed into the limiter since it can introduce unpleasant digital distortion.

Feed the limiter with care, when it is clipping mode.

Attenuation

Attenuation reduces the overall output of the Plugin, it sits right at the end of the complete signal path. The controllable range of the Attenuation is -40.0 dB to 0.0 dB.

This parameter is used for two main tasks:

1. If you compress heavily and drive the limiter hard, you might end up with a lot of average loudness. Most likely this will put you in a situation where you would have to drastically lower the level slider of your hosts channel and then you might not have enough range on that level slider to mix properly. Pull the Attenuation knob down in order to increase the range of your hosts channel level slider.

2. When compressing and limiting, you most likely end up with a slightly louder perceived signal, even if both, the dry and the Plugin output signal are at -0.1 dB in peak level. In order not to get fooled by the loudness increase, you should finally reduce the output to a point that roughly equals the perceived loudness of the dry signal. Use the Bypass switch in the upper right of the user interface to compare both signals frequently. Once you are satisfied with the result, you can turn Attenuation back to 0.0 dB.

Metering

This section visually displays what happens to the signal at various stages. Use this to observe how the input signal is treated by the Plugin.

Input

The input button turns the metering to show all input stages.

Reduction

The reduction button turns the metering to show the individual gain reduction of the compressor and the limiter.

When the limiter release is set to "brick wall clipping" mode the reduction of the limiter can`t be displayed. You wouldn`t see the "needle" anyway as the brick wall clipping is sample accurate and therefore way too fast to be displayed.

Output

The output button turns the metering to show all output stages.

CLMS-1 XL Optoelectronic Dual Mid/Side Compressor-Limiter

The CLMS-1 XL is the most versatile unit of the three compressors. It provides you with most flexible control over its mid/side capabilities. Think of two independent, full fledged compressors. one for the mid and one for the side signal.

The input of the CLMS-1 XL is split up into Mid/Side encoded mono signals, which both feed their own compressor stages labeled with "M" and "S". Furthermore the control signal is also Mid/Side encoded and feeding their own mixers. These Mid/Side mixers outputs are each controlling their compressor counterparts. That way you can determine which mix of the Mid/Side control signal controls each compressor individually. Each compressor stage has its own independent side chain/control signal path, which features an EQ for both separately, as well as the ability to use an external side chain signal for each compressor.

The compressors output is then fed into a Mid/Side mixer, which allows you to rebalance the outputs of the compressors. Additionally you can rebalance the side signal separately. The whole Mid/Side mixers output is then decoded back to stereo, and hits the Dry/Wet mixer. Finally the stereo signal is send into the output gain stage and the limiter.

Signal Flow Diagram
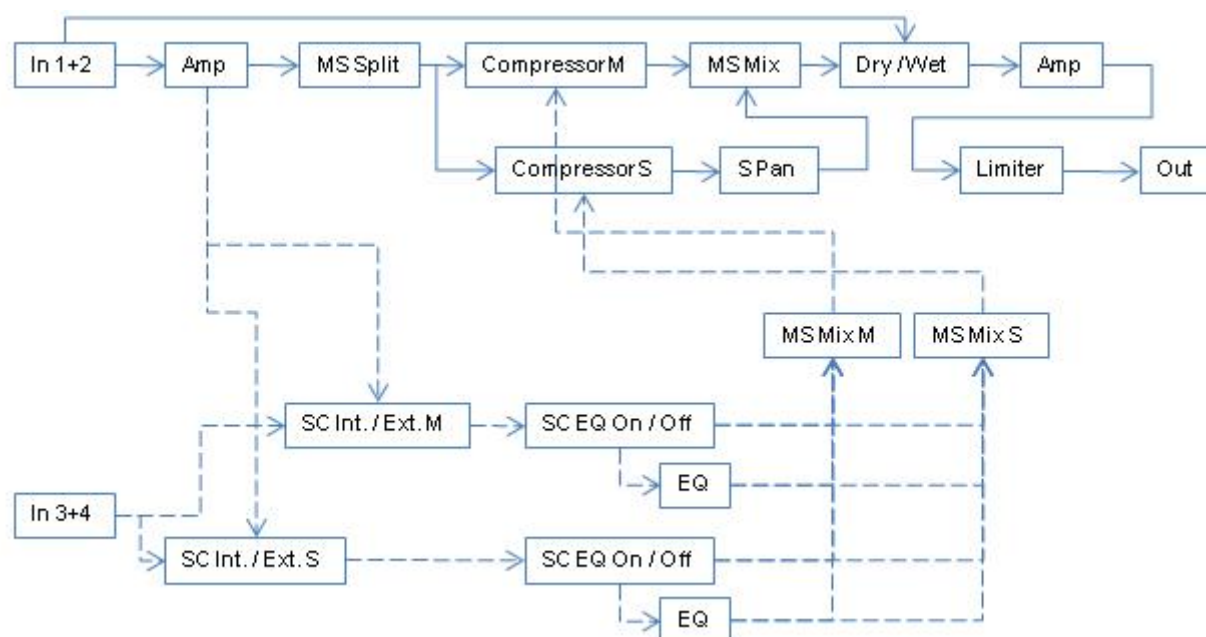
Features and Functions

General Functions

File Menu

Further details can be found in chapter "Preset Manager".

Preset Menu

Further details can be found in chapter "Preset Manager".

Bypass

This button will bypass the complete processing. This can be helpful for A/B controlling the original and the compressed signal.

Compressor

M/S - L/R

The Plugin can be used in two different modes. One is the Mid/Side compression and the other is a dual mono mode (Left/Right). When switched to L/R all switches, knobs and sliders labeled with an "M" are corresponding to the left signal where as all labeled with an "S" correspond to the right channel.

> The "S Pan" parameter has no functionality in L/R mode.

Input

This parameter has two functions. First it levels the signal that is entering the compressor, it is the so called "Input Gain Stage". Second this gain stage is modeled after a tube circuit. The tube adds harmonics from subtle to heavy, when the input signal is over 0.0 dB. Harmonic coloring is often preferred, so simply play around with the Input Gain. However it is crucial to lower the compressors Output Gain accordingly, in order to stay below digital clipping. The controllable range of the Input Gain is -40.0 dB to +40.0 dB. The internal side chain signal is only affected when the Input Gain is below 0.0 db and external side chain signal is not affected by the Input Gain.

> We suggest to always normalize the input signal outside the compressor, but if that is not possible this is the parameter to be used instead.

Tube Indicator (LED)

This led indicates the tube activity. As soon as the tube adds harmonics to the signal, this led will lit up. Signals passing the tube unaltered will not trigger it, so it remains off.

Threshold M

The Threshold is the level that must be exceeded by the incoming side chain signal, in order for the compressor to work. The Threshold level could be thought of as the "sensitivity" of an compressor. The controllable range of the Threshold is 0.0 dB to -70.0 dB. This parameter is exclusively for the mid compressor, labeled with the "M".

Ratio M

The Ratio is the degree to which the compressor is reducing the dynamic range. Ratio therefore indicates the relative gain reduction between the incoming audio signal and the outgoing compressed audio signal. A ratio of 2:1 means that an audio signal of e.g. +2.0 dB gets compressed to an audio signal of just +1.0 dB. The higher the Ratio, the more the signal will be reduced. The controllable range of the Ratio is 1:1 to -5:1. When setting the Ratio to the middle of the controllable range the Ratio is set to "Limit". This parameter is exclusively for the mid compressor, labeled with the "M".

> All our compressors feature a technique called "negative ratio". This is a very rare implementation, which actually allows you to reduce the signal below the Threshold. It can be used for drastic gain reductions often heard on drums.

Attack M

This adjusts the time that passes before the compressor takes action. The controllable range of the Attack is 0.0 % (~0.1 ms) to 100.0 % (~500.0 ms). This parameter is exclusively for the mid compressor, labeled with the "M".

Release M

The Release determines the time that the signal needs to recover to its original level. The controllable range of the Release is 0.0 % (~20.0 ms) to 100.0 % (~1000.0 ms). This parameter is exclusively for the mid compressor, labeled with the "M".

Threshold S

This parameter is exclusively for the side compressor, labeled with the "S". A description can be found at the corresponding mid compressor parameter.

Ratio S

This parameter is exclusively for the side compressor, labeled with the "S". A description can be found at the corresponding mid compressor parameter.

Attack S

This parameter is exclusively for the side compressor, labeled with the "S". A description can be found at the corresponding mid compressor parameter.

Release S

This parameter is exclusively for the side compressor, labeled with the "S". A description can be found at the corresponding mid compressor parameter.

Link

With this switch both compressors parameters can be linked. The parameters Threshold, Ratio, Attack and Release can be adjusted in parallel. The parameters will not change their values instantly but as soon as you turn a knob the corresponding parameter will also be adjusted.

Side Pan

Some signals introduce an uneven stereo balance and therefore one side is louder than the other. When using compression this may even become more audible. Use this slider to adjust the stereo balance. This slider affects exclusively the side signal, the mid signal stays unaltered.

M/S Mix

The mid and the side compressors outputs are send into a mid/side mixer. With this knob you set up the balance between the outputs of the mid and the side compressor.

Dry/Wet

With this parameter you can mix back the original, completely untreated input signal.

It is often called "New York style compression" to do so. The intention is, to compress the signal in a drastic way but use only a small portion of the wet signal and mix it back to the dry signal. That way one can achieve a very natural sound, featuring both, the captured sound of the compressor itself and the natural sound of the dry signal.

Output

This adjusts the output of the compressors Dry/Wet mix stage. With this parameter you define the level that feeds the limiter. Be careful with this parameter when the limiter is "Off", you need to carefully readjust this parameter in order to not overshoot 0.0 dB. The controllable range of the Output Gain is -40.0 dB to +40.0 dB.

> The Output Gain can be used to either level the compressed audio signal to 0.0 dB (compensate the gain reduction) or if the Input has been driven into the Tube Circuit to set back the compressed and saturated audio signal to a usable range.

Side Chain M/S

This section exclusively affects the internal/external control signal.

External/Internal Signal

Here you specify which signal is used to control the compressor. When set to "Internal" the control signal is grabbed right from the main stereo input, before the "Input Gain Stage". When set to "External" the signal to control the compressor is grabbed from the discrete external stereo input. Whatever signal you feed into these inputs will control the compressor. Please refer to your hosts manual in order to learn how to route the external side chain control signal. This can be set for each side of the side chain path individually.

> In case your host does not support more than two input channels the external side chain is also using the internal signal.

M/S Mix M

This knob controls the balance between the mid and the side control signal, on which the mid compressor reacts.

M/S Mix S

This knob controls the balance between the mid and the side control signal, on which the side compressor reacts.

EQ On/Off

Turns the equalizer for the control signal (internal/external side chain) on and off.
There is a EQ On/Off for each side of the side chain path.

Highpass/Lowpass Filter

Whatever control signal you use it is routed through these two EQ bands unless you switched off the EQ. It features a 12dB Highpass as well as a 12dB Lowpass.
- The Highpass lets the frequencies above the adjusted value pass through, while cutting the frequencies below it.
- The Lowpass lets the frequencies below the adjusted value pass through, while cutting the frequencies above it.

Whenever you want the compressor to react only on specific frequencies of the control signal, use this EQ to isolate these frequencies. If the knobs are set to "0", the signal passes untreated. There is a EQ stage for each side of the side chain path.

Solo

In order to directly hear how the EQ affects the control signal turn this on. Now you will hear the side chain signal. There is a solo button for each side of the side chain path.

Depending on the Mid/Side mix value you specified with the corresponding slider, you will hear the exact mix you set up with this slider.

Limiter

The limiter sits directly after the compressor output. Therefore the compressor output knob controls the limiter input. The limiter is operating as a "brick wall limiter" and does not allow any signal to pass above -0.1 dB.

On/Off

The switch is used to turn on/off the Limiter. The Attenuation will not be affected by this switch.

Type

The limiter can operate in two modes: Peak and RMS level detection.

Release

The Release determines the time that the signal needs to recover to the compressors output level. The controllable range of the Release is 0.0 % (Clipping), 0.01 % (Peak: ~50.0 ms; RMS: ~1.0 ms) to 100.0 % (~1500.0 ms).
When set to 0.0 % the release is completely disabled and effectively turning the limiter into sample accurate brick wall clipping mode. In this operation mode the limiter Release is extremely fast, but you must take care about the level of the signal you feed into the limiter since it can introduce unpleasant digital distortion.

Feed the limiter with care, when it is clipping mode.

Attenuation

Attenuation reduces the overall output of the Plugin, it sits right at the end of the complete signal path. The controllable range of the Attenuation is -40.0 dB to 0.0 dB.

This parameter is used for two main tasks:
1. If you compress heavily and drive the limiter hard, you might end up with a lot of average loudness. Most likely this will put you in a situation where you would have to drastically lower the level slider of your hosts channel and then you might not have enough range on that level slider to mix properly. Pull the Attenuation knob down in order to increase the range of your hosts channel level slider.
2. When compressing and limiting, you most likely end up with a slightly louder perceived signal, even if both, the dry and the Plugin output signal are at -0.1 dB in peak level. In order not to get fooled by the loudness increase, you should finally reduce the output to a point that roughly equals the perceived loudness of the dry signal. Use the Bypass switch in the upper right of the user interface to compare both signals frequently. Once you are satisfied with the result, you can turn Attenuation back to 0.0 dB.

Metering

This section visually displays what happens to the signal at various stages. Use this to observe how the input signal is treated by the Plugin.

Input

The input button turns the metering to show all input stages.

Reduction

The reduction button turns the metering to show the individual gain reduction of the compressor and the limiter.

> When the limiter release is set to "brick wall clipping" mode the reduction of the limiter can`t be displayed. You wouldn`t see the "needle" anyway as the brick wall clipping is sample accurate and therefore way too fast to be displayed.

Output

The output button turns the metering to show all output stages.

Usage

All Plugins can be operated the same way. We have implemented a very simply but efficient set of commands which are mainly available using your mouse.

Knobs, Sliders

Command	Description
Left mouse button	Coarse parameter entry using vertical mouse movement. • You will be able to enter the values using a coarse increment, e.g. 0.0 dB, 1.0 dB, 2.0 dB • The mouse pointer will be hidden as soon as you click on the control
Ctrl + Left mouse button Shift + Left mouse button	Fine parameter entry using vertical mouse movement. • You will be able to enter the values using the finest available increment, e.g. 0.0 dB, 0.1 dB, 0.2 dB • The mouse pointer will be hidden as soon as you click on the control Note: On OS X you cannot use the "Ctrl" modifier.
Right mouse button	An edit box will be displayed to enter the parameter value directly. The entered value will be used as soon as you press the ENTER key.
Double Click	Reset parameter to the default value.
Mouse wheel	Fine parameter entry. • You will be able to enter the values using the finest available increment, e.g. 0.0 dB, 0.1 dB, 0.2 dB
Ctrl + Mouse wheel Shift + Mouse Wheel	Coarse parameter entry. • You will be able to enter the values using a coarse increment, e.g. 0.0 dB, 1.0 dB, 2.0 dB Note: On OS X you cannot use the "Ctrl" modifier.

Buttons, Switches

Command	Description
Left mouse button	A single click on the control will trigger the command.

Parameter Automation

All parameters can be automated using your hosts capabilities.

Preset Manager

The ArtsAcoustic Plugins features a built in Preset Manager. The Preset Manager can be used to open, save, delete, manage and share the presets.

FILE ▼ PRESET ▼ ◀ default - CL1 ▶

> For best cross platform and host to host compatibility we do not recommend to use the Preset Manager provided by the host software. The Preset Manager of the ArtsAcoustic Plugins uses its own format which is compatible on all platforms, hosts and future versions of the Plugins.

Presets

The presets are split up into two different main categories
• Factory Presets
• User Presets

Each of the Plugins of the ArtsAcoustic CL Series has its own preset file format and can manage separate folders for the User Presets. It is also possible to use the same folder and therefore the same sub folder structure to manage the User Presets.

Factory Presets

The Factory Presets can be accessed using the "Preset" menu. Factory Presets cannot be changed or deleted using the Preset Manager. All Factory Presets are saved in a special factory Preset Folder. A list of all available Factory Presets can be found in chapter "Appendix I – Factory Presets".

User Presets

The User Presets can be accessed using the "Preset" menu. All User Presets are saved in a customizable folder and can be fully managed using the Preset Manager.

Autoload Preset

The Plugin will try to load a preset called "autoload" from within the root of the User Preset Folder on startup. If it cannot find this preset the Plugin will be initialized with its default settings.

File Menu

The "File" menu is used to load and save presets beside the Preset Manager. This way it is possible to share settings with other users or save presets in other folders than the defined User Preset Folder.
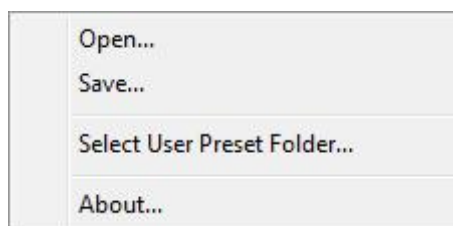
Open…

A preset can be opened using this menu. A file system browser will be opened to select the wanted preset.
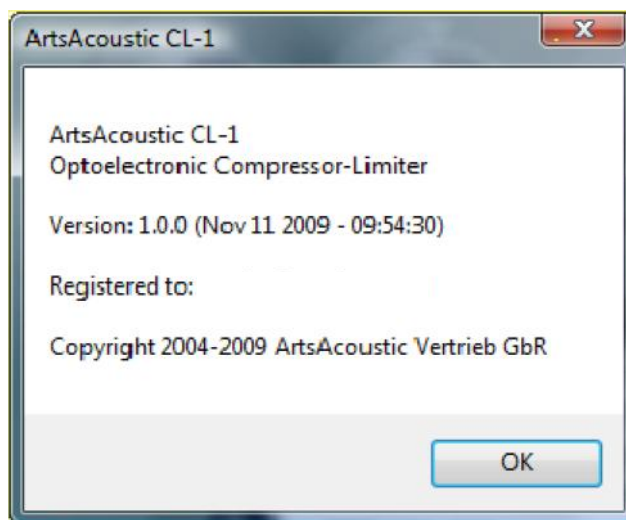
Save…

A preset can be saved using this menu. A file system browser will be opened to select the wanted preset name and location. The preset must not be saved within the User Preset Folder.

Select User Preset Folder…

It is possible to set the User Preset Folder using this menu. The User Preset Folder must be setup for each Plugin of the ArtsAcoustic CL Series separately. As long as the User Preset Folder has not been selected most functionality of the Preset Manager is not available.
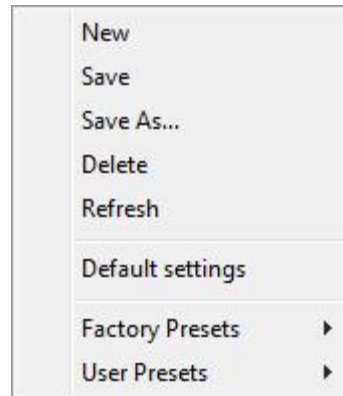
About…

The about window is used to display information about the active Plugin.

Preset Menu

The "Preset" menu is used to manage the presets within the Preset Manager. All functions within this menu will only operate on the special folders used by the Plugin.

New
Save
Save As...
Delete
Refresh

Default settings

Factory Presets ▶
User Presets ▶

New

A new preset can be created using this menu.

Save

The currently opened preset can be saved using this menu. As long as no preset has been opened using the Preset Manager an error message will be displayed.

Save As…

A preset can be saved using this menu. A file system browser will be opened to select the wanted preset name and location. The preset must be saved within the User Preset Folder or its sub folders.

> Attention: This function can be used to overwrite existing presets. Any overwritten preset cannot be recreated from within the Plugin.

> Use the menu "File|Save…" to save a preset into different folders than the User Preset Folder.

Delete

The currently opened preset can be deleted using this menu. As long as no preset has been opened using the Preset Manager an error message will be displayed.
Before the preset gets deleted an additional requester asks for permission to proceed.

> Attention: The preset will be physically removed from the file system and cannot be accessed anymore.

Refresh

The preset folder structures will be reloaded. This can be useful if presets in the User Preset Folder have been changed manually.

Default settings

The Plugin will be reset to its default settings.

Factory Presets >

From within this menu all Factory Presets are available. A Factory Preset cannot be deleted or overwritten using the Preset Manager functionality. When changes have been made to a Factory Preset it is possible to save it to the User Preset Folder using "Save As…". A list of all available Factory Presets can be found in chapter "Appendix I – Factory Presets".

User Presets >

From within this menu all User Presets are available.

Preset Browser

The Preset Browser can be used to browse through all Factory Presets and User Presets.

It is possible to browse through the presets using the buttons "Left Arrow" and "Right Arrow". It will cycle through all presets backwards ("Left") until it reaches the first Factory Preset or forward ("Right") until it reaches the last User Preset.

The name of the currently active preset is displayed in the middle between the buttons "Left Arrow" and "Right Arrow".

Final Words

If you managed to read the manual up to here you should be well equipped to use the ArtsAcoustic CL Series on a professional basis. Secondly we will have reached our goal of writing a comprehensive, coherent, informative and perhaps also entertaining guidebook. We hope that both has become the truth and wish you a lot of fun while working with the Plugins of the ArtsAcoustic CL Series.

Your ArtsAcoustic Team

Support

Use the online forum provided on our website:

http://www.artsacoustic.com

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Appendix I - Factory Presets

Following you can find a list of all available Factory Presets.

Category	Name	Comment
CL-1		
	all-in pumper	
	concrete upfront	
	NY style compression	
	subtle generic compression	
CLMS-1		
	attack me	
	extreme negative ratio	
	smack me	
	transient taming	
CLMS-1 XL		
	dance drum enhancement	
	drum room stereo enhancement	
	gentle master	
	subtle stereo enhancement	

Appendix II – Version History

Following is a list of all released versions.

Date	Version	Comment
2009-12-23	1.0.4	• Initial release
2009-12-25	1.0.5	• Fixed: Didn't work in some hosts (Ableton Live, Samplitude)
2009-12-26	1.0.6	• Fixed: Didn't work in some hosts (Wavelab, Soundforge)
2009-12-30	1.0.7	• Fixed: Didn't save the Plugin state within some hosts • Fixed: Installer copied the Factory Presets into wrong folder. • Changed: Manual updated
2010-01-01	1.0.8	• Fixed: Didn't load the Plugin state within some hosts
2010-01-10	1.0.9	• Fixed: Failed to save/load the Plugin state within some hosts • Fixed: Failed to save the Presets • Changed: Mouse behavior corrected (Fine/Coarse Mode) • Changed: The configuration file gets saved in the Users "AppData" Folder now (Users must enter the serial number and User Preset Folder again) • Changed: Preset Manager runs in its own thread now.
2010-01-18	1.0.10	• Fixed: Plugin sometimes failed to read the serial number
2010-01-19	1.0.11	• Fixed: Sidechain EQ didn't work as expected
2010-04-21	1.0.12	• Added: Mac OS X support • Added: Limiter - RMS level detection • Added: CL1 can be used as Mono Plugin • Fixed: Mouse wheel supported in FL Studio • Fixed: Failed to load in some hosts (Samplitude, Studio One) • Fixed: About Window caused the Samplitude UI to get blocked • Fixed: Some serial numbers have not been saved properly • Changed: The Plugins introduce a latency of 0.73 ms. Plugin Latency Compensation by the host is supported. • Changed: Dragging the knobs lets them snap to the correct value
2010-10-21	1.0.13	• Added: Windows 64bit version • Added: OSX Carbon UI - implemented, but not supporting sub menus yet. • Changed: negative Input Gain now also affects the internal side chain level. • Changed: the RMS Limiter is the default • Fixed: "Midi out" caused an error in some hosts (AudioMulch, Bidule, Kore2) • Fixed: some repaint issues • Fixed: OSX mouse wheel behavior is now more like under Windows • Fixed: lookahead buffer was not properly setup in the CLMS1 and CLMS1-XL

Date	Version	Comment
		• Fixed: Mousewheel support added (Nuendo) • Fixed: some more possible heap corruptions • Fixed: disabled some dbl clicks on switch style controls
2011-06-26	1.0.19	• Added: OSX 64bit version • Fixed: CLS old preset formats (version 1) were not loaded properly • Fixed: demo versions not working properly in some hosts (Energy XT 1/2) • Fixed: OSX Carbon UI - did not save presets properly in some hosts • Fixed: AU version not working in some hosts (Studio One, Maschine, Kore) • Fixed: OSX Carbon UI - did cause trouble with the preset manager (it always loaded the previous preset) • Fixed: OSX Cocoa UI - did cause crashes in Logic. • Fixed: DEMO versions did not work properly • Fixed: removed some unnecessary repaints • Fixed: Windows - hiding the mouse cursor uses another more compatible method • Fixed: OSX - open/save dialog does not shutdown Logic anymore • Fixed: AU demo version do share the same name as the full version • Fixed: CL1 the menu "Preset|New" did not work • Fixed: sometimes the first control was not updated properly and remains unuseable (bound to max value) • Fixed: when switching a preset while playing or changing a parameter the audio engine stopped (OSX/Logic) • Fixed: Mono and StereoToMono versions didn't work properly in some hosts (OSX/Logic) • Fixed: the param smoother did not always work properly therefore the smoothing has been enhanced