

Vienna
Symphonic
Library

Vienna Suite

Manual

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Introduction

SIMPLIFY – one word is the mantra that accompanies the Vienna Symphonic Library team every day. The search for the best algorithms in audio processing in the digital domain is showing long-awaited and surprising results.

In combination with the taste of the experts, trained from years of experience with the most sensible sounds available in virtual orchestration, the Vienna Symphonic Library Development Team, Martin Saleteg and George Yohng have created a meticulously engineered plug-in bundle to produce musical results in a wide variety of applications.

Intuitive handling of complex audio signals calls for new approaches to their design. As with the Vienna Instruments, visual feedback for the well-known basic audio processes in the background is a great feature for everybody involved in the evolution of sound from the first sketches to the final mix, serving both newcomers and professionals.

Most of the plug-ins include tailor-made presets for Vienna Instruments which will give you a lot of creative input for your soundscapes and provide quite a number of sonic solutions and ideas for both individual instruments as well as complete mixes. Our research was following musical guidelines, and the decision to let you benefit from our findings in the form of intelligible User Interfaces was easily made after the huge success of the Vienna Instruments. Enjoy this new array of great-sounding yet CPU-friendly plug-ins for the everyday audio processing requirements that are easy to operate, yet super-efficient.

Walkthrough Videos



Watch the video

General Plugin Operation

All plug-ins share some operational features. Here's a description of those common controls:

- Ctrl+Click (Windows), Command+Click (macOS) resets the selected control to its default value.
- Pressing Shift while dragging a control increases its resolution for fine-tuning adjustments.
- **Double-clicking** on a control allows you to enter a numerical value using the computer keyboard.
- **Factory Presets** lets you select one of the pre-configured settings for the plug-in, many of which are specifically adapted for the Vienna Instruments.
- You can **Load** and **Save** your custom presets on your computer, and exchange them with other users of the Vienna Suite. Make sure to back up your custom presets regularly – you don't want to lose them...

The default path for the **Preset Folder** is:

- *Windows:*
  - [YourName]/AppData/Roaming/Vienna Suite Presets/[PluginName]
    [YourName] is the name you're logged in with on the computer.
- *macOS:*
  - [Macintosh HD]/Users/YOUR_NAME/Library/Application Support/Vienna Suite
    [Macintosh HD] is the name of the hard disk the operating system is installed on.

> **Important**
>
> Plug-ins are only guaranteed to work at a host sampling rate of 44,100 Hz and higher. If the rate is set to, e.g., 22,050 or 33,000 Hz, plug-ins may perform improperly or in a limited way.

> **Important**
>
> Under some conditions, certain plug-ins will change the output signal's phase, so that using them, e.g., on a 'send' bus may not produce the expected result.

Hybrid Reverb

The Hybrid Reverb combines the natural versatility of convolution impulses (maximum length for Factory Presets: 1 second) for early reflections with the vivid, larger-than-life reverb tail of an algorithmic reverb. You get the best of both worlds, which helps you sculpt the much sought-after contemporary sound that is the ideal of today's mixing standards.

We have provided many starting points from multiple preset variations that let you dig deeper into the secrets of modern reverberation.

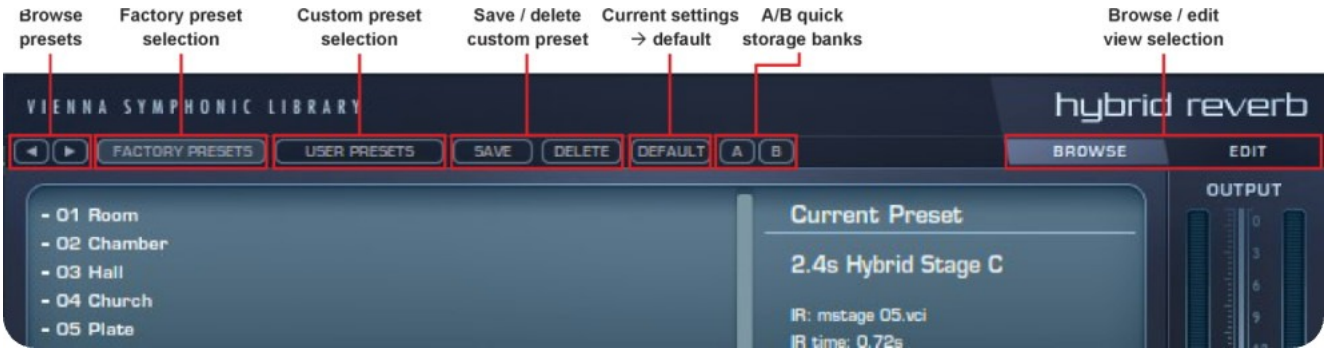
Switch between **Browse View** and **Edit View** with the buttons below the Hybrid Reverb Logo, or by clicking anywhere on the top header bar.

Walkthrough Video



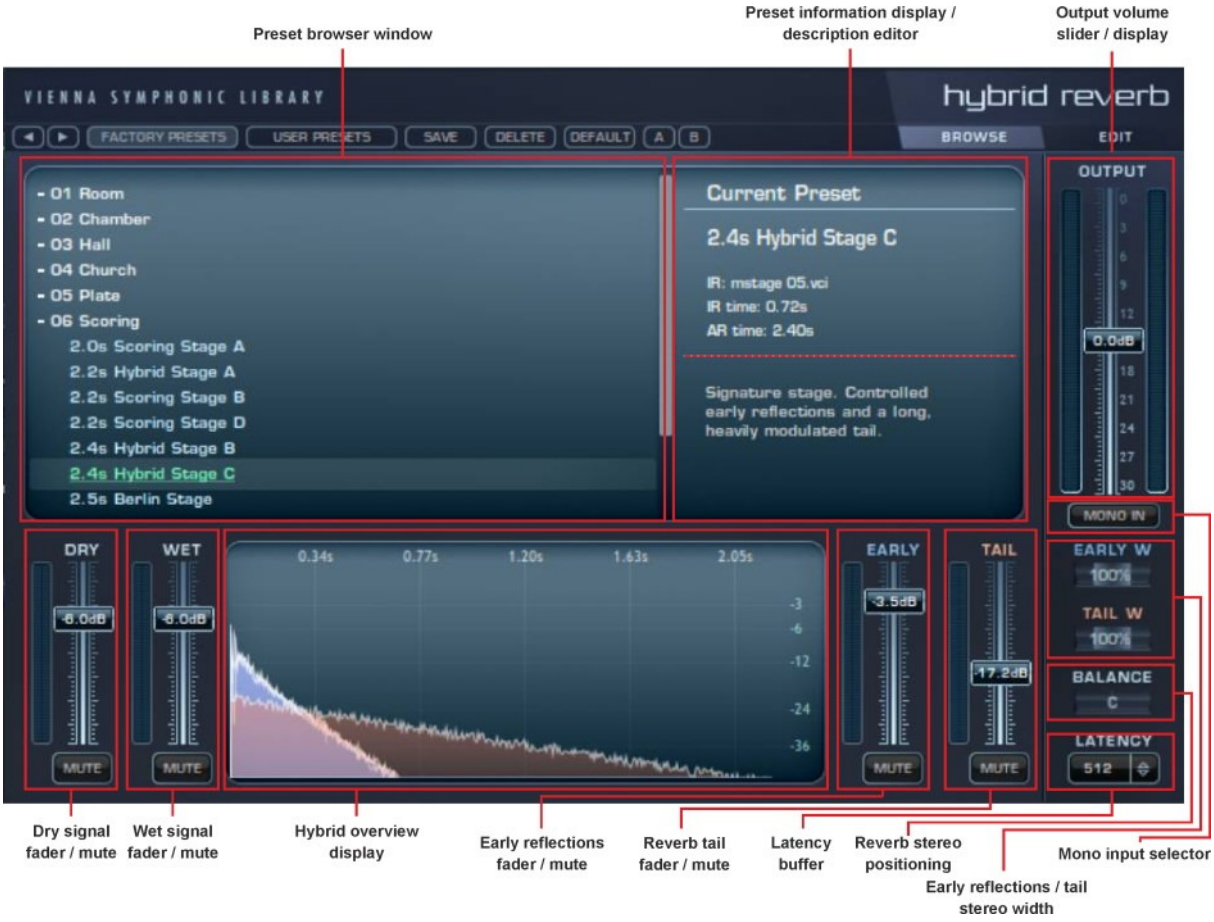
Watch the video

Preset Management



- **Arrows**: Use the arrows above the browser to step through the presets.
- **Factory Presets**: Display all available Factory Presets in the Browser Window below.
- **User Presets**: Manage your customized presets in the Browser Window below.
- **Default**: Saves the current settings as the default (startup) preset.
- **A/B**: Provides quick-store of settings, to compare two different settings comfortably.
- **Browse/Edit**: Switch between Browse View and Edit View with the buttons below the Hybrid Reverb Logo.

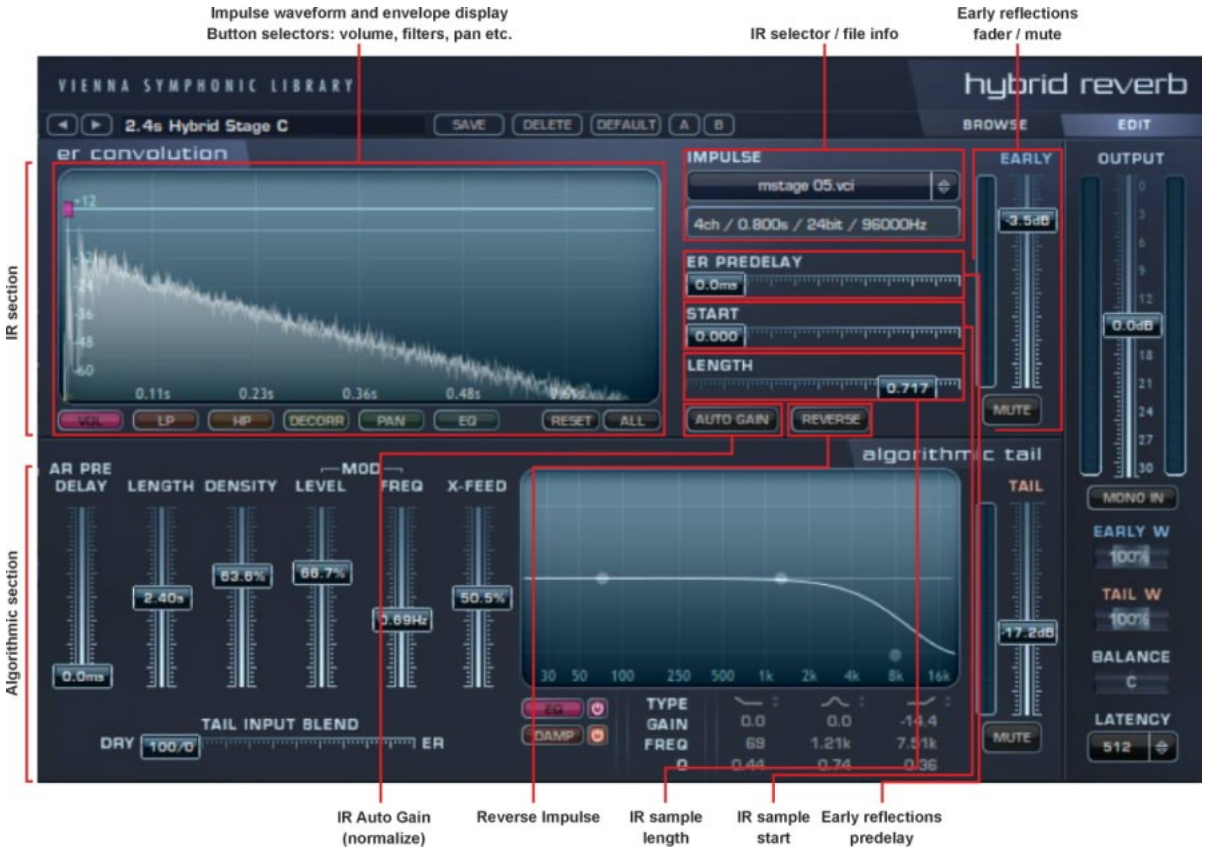
Browse View



- **Preset Browser Window**: Double-click to load a preset.
- **Preset Information Section**: Displays preset name, the loaded impulse response (IR), the impulse response length (IR time) and the Algorithmic Reverb length (AR time) of the last loaded preset.
- **Preset Description Section**: A short description of the loaded preset. Double-click to edit.
- **Output Volume Slider**: Sets the output volume of the Hybrid Reverb.
- **Dry**: The amount of the unprocessed signal. The MUTE button below will silence this signal.
- **Wet**: The amount of the processed signal. The MUTE button below will silence this signal.
- **Hybrid Overview Display**: Shows the Impulse Response of the Early Reflections in combination with the length and shape of the Algorithmic Reverb Tail.
- **Early**: The amount of the Early Reflection signal. The MUTE button below will silence this signal.
- **Tail**: The amount of the Tail signal. The MUTE button below will silence this signal.
- **Mono Input**: Forces inputs to be combined to a mono signal.
- **Early W**: The stereo width of the Early Reflections.
- **Tail W**: The stereo width of the Algorithmic Reverb (Tail).

- **Balance**: Positions the reverb in the stereo field
- **Latency:** By default, the plugin has 512 sample latency, but if desired, latency can be adjusted to a lower setting and even to 0 – with the exception of Apple Logic, which for technical reasons needs non-zero latency. For certain systems, a higher latency setting is preferable at very low latencies of your soundcard, such as 64 or 128 samples. In general, the higher the latency, the lower the CPU usage. It is *NOT* recommended to use latencies lower than the buffer size. This will lead to excessive CPU usage.

Edit View



Early Reflections Convolution Section

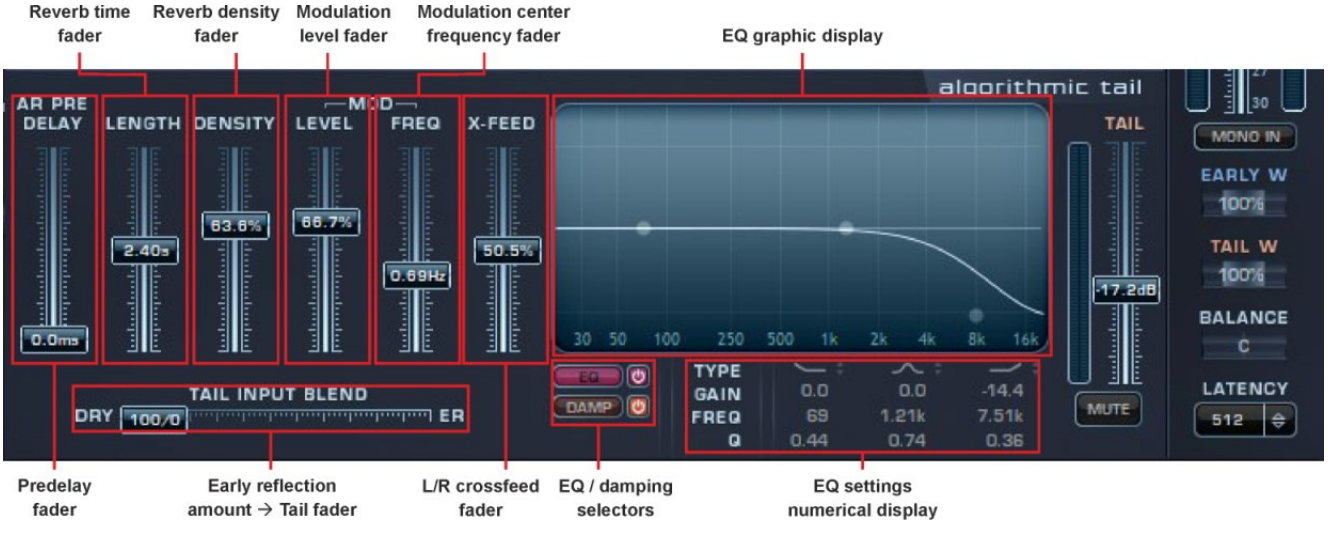
This section lets you manipulate the loaded impulse response that is used exclusively for the early reflections of the processed signal.

- **Impulse Waveform Display:** Displays the waveform of the loaded impulse response and the envelopes of the impulse VOL, LP, HP, DECORR, PAN and EQ. All manipulations are non-destructive and the waveform of the impulse is adapted automatically.
- **Graphic Handles:** Double-click anywhere in the Impulse Waveform Display – a new handle will appear for the selected envelope. This way, you can design your own envelopes and shape the room to your liking.
- **LP:** Adds a low pass(i.e.,high-cut)filter to the impulse response. Values are indicated in Hz.
- **HP:** Adds a high pass(i.e., low-cut)filter to the impulse response. Values are indicated in Hz.

- **Decorr:** Adjusts the amount of decorrelation used on the impulse response. Decorrelation enhances a signal's perceived stereo image, and thus the spatial enveloping and broadness of the resulting reverb.
- **Pan:** Adjusts the impulse response's left/right balance.
- **EQ:** Adjust the sound of your impulse response with this high-quality full-parametric equalizer.
- **Reset:** The chosen envelope displayed in the Impulse Waveform Display is reset.
- **All:** All envelopes in the Impulse Waveform Display are reset.
- **Impulse**: Displays the loaded impulse response. Use the arrows to step through the available impulses. Double-click to open the browser. Information about the current impulse is displayed underneath.
- **ER Predelay:** The amount of time (in milliseconds) between the dry signal and the onset of the impulse response.
- **Start:** Moves the starting point of the impulse response sample (displayed in seconds). One could say that the left border of the IR is moved towards the middle. This is useful for getting rid of any unwanted silence built-in to the IR, or to adjust the length of the early reflection component.
- **Length:** Determines the length of the used impulse response sample (displayed in seconds). One could also say that the right border of the IR is moved towards the middle so that the tail of the IR gets shortened. This can be useful for special effects.
- **Early:** Adjusts the volume of the selected impulse response. All level changes are indicated in dB in relation to the unchanged original volume envelope of the IR. Corresponds to the **Early** slider in Browse View.
- **Auto Gain:** Automatically normalizes the impulse response.
- **Reverse (Impulse):** Reverse your impulse response sample.

Tail Algorithmic Section

This section creates the reverb tail of your processed signal. Control the amount of the early reflection signal that is fed into the algorithmic reverb with the **Tail Input Blend** slider.



- **AR Predelay:** The amount of time (in milliseconds) between the dry signal and the onset of the algorithmic reverb.
- **Length:** The length of the reverb tail.
- **Density:** Adjusts **** the complexity and richness of the reverb tail. Please note that this setting also affects CPU usage.
- **Mod Level:** Controls the level of modulation in the tail. Higher values might create a sense of pitch shift in steady instruments. Helps to create a more vivid sounding decay and frequency spread. The modulation engine is turned off when setting the Mod Level to 0%, and CPU usage will be somewhat lower.
- **Mod Frequency:** Controls the frequency of modulation in the algorithm. Modulation frequencies will be spread out randomly around the set value.
- **X-Feed:** Controls the crossfeed level from left to right inside the reverb algorithm. 0% means that an input signal only on the left channel will yield an output signal only on the left channel. 100% will produce an even distribution in the stereo field, resulting in a more consistent sound.
- **Tail Input Blend:** Adjusts the amount of the Early Reflection Signal that is fed into the algorithmic reverb tail.
- **Filter Response Display:** Shows the response curve and frequency analysis for the current settings. The (horizontal) frequency range is from 20 Hz to 20 KHz, and the (vertical) amplitude scale displays a range of 36 dB (-18 dB to +18 dB).
- **Filter handle:** Drag this handle to graphically adjust the filter's boost/cut and/or frequency. When adjusting the filter handle, Shift locks the frequency; Alt / Option locks the gain. Double-clicking a handle resets the respective band's gain to the default.
- **Composite curve:** Shows the overall response curve of the current settings in the Filter Response Display.
- **3 band EQ:** Enable the 3 bands of the algorithmic reverb tail EQ independently.
- **3 Band Damping EQ:** For the damping filter, the Q value is limited between 0.1 and 0.5 to keep the filters from going resonant.– As the characteristics of any room will have an essential influence on the damping of each frequency range over time, this parameter is the decisive factor in successfully recreating these properties. The CPU usage of the reverb is affected by how many damping filters you enable.
- **Type:** Choose between Peak, Low Shelf, High Shelf, Low Cut and High Cut filter types. The damping EQ offers three fixed (Low Shelf, High Shelf, LowPass) types.
- **Gain:** Adjusts the peak gain of the corresponding filter in a range from -18 dB to +18 dB.
  Damping EQ: -3 dB to 0 dB.
- **Frequency:** Adjust the centre frequency of the given frequency band.
- **Q setting:** Adjusts the filter Q of the corresponding filter band from 0.1 to 10. The smallest setting of 0.1 will create a very wide filter, whereas a setting of 10 will give you the smallest possible filter width.
  Damping EQ: 0.1 to 0.5. Use the mouse wheel (or press Ctrl/Cmd + click and drag) to adjust the Q setting for the currently selected filter. The given filter handle has to be highlighted.

Convolution Reverb

The primary goal of a convolution reverb is to sample real spaces, in order to simulate the acoustics of the sampled space. A straightforward and simple example of capturing an impulse response would be to set up a pair of microphones in a concert hall and to place the microphone in the centre of the auditorium. Next, produce a very brief pulse of sound, and record everything that the microphone picks up, which includes both the original sound and the response of the room to it.

This recording of an impulse in a room or another location is used as a fingerprint of the room. As a result, the processed audio will sound as if it were played in the same location.

Sample Development is an art – and this is also true for sampling rooms. We are proud to have great partners on board who have contributed great-sounding spaces in addition to the Vienna Impulse Responses – and there will be many more available for download!

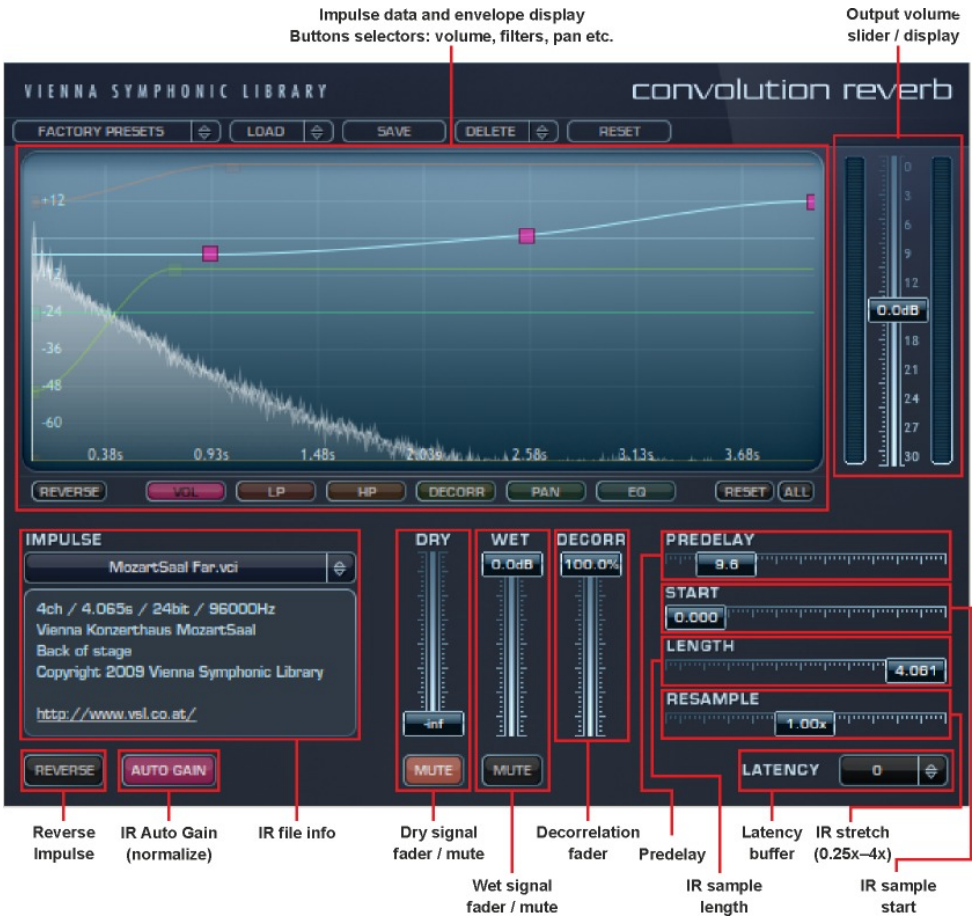
With the Vienna Suite Convolution Reverb, you can shape the sound of your own spaces with the many options explained below, or you can make use of the versatile presets available!

Walkthrough Video



Watch the video

Controls



- **Impulse Waveform Display:** Displays the waveform of the loaded impulse response and the envelopes of the impulse VOL, LP, HP, DECORR, PAN and EQ. All manipulations are non-destructive and the waveform of the impulse is adapted automatically.
- **Graphic Handles:** Double-click anywhere in the Impulse Waveform Display – a new handle will appear for the selected envelope. This way, you can design your own envelopes and shape the room to your liking.
- **Reverse (Envelopes):** You can reverse all envelopes applied to your impulse response here.
- **Vol:** Adjusts the Volume of the selected impulse response.
- **LP:** Adds a low pass filter to the impulse response.
- **HP:** Adds a high pass filter to the impulse response.
- **Decorr:** Adjusts the amount of decorrelation used on the impulse response. Only effective if the general DECORR Slider (underneath) is up.
- **Pan:** Position your impulse response.
- **EQ:** Tune the sound of your impulse response with this high-quality Equalizer.
- **Reset:** The chosen envelope displayed in the Impulse Waveform Display is reset.
- **All:** All envelopes in the Impulse Waveform Display are reset.

- **Output Volume Slider:** Determines the response's output volume.
- **Impulse:** Load impulses of your choice or choose from many presets.
- **Impulse Data Window:** Shows the properties of the loaded impulse response: channels / length / bit / Hz
- **Reverse (Impulse):** Reverse your impulse response sample.
- **Auto Gain:** Automatically normalizes the impulse response.
- **Dry:** The amount of the unprocessed signal. The MUTE Button below will silence this signal.
- **Wet:** The amount of the processed signal. The MUTE Button below will silence this signal.
- **Decorr:** Advanced frequency-domain decorrelation algorithm. Using the decorrelation slider and the decorrelation graph, you can bring life to dull, narrow impulses and create a lush, rich reverb tail.
- **Predelay:** The amount of time (in milliseconds) between the dry signal and the onset of the reverb.
- **Start:** Moves the starting point of the impulse response sample (displayed in seconds).
- **Length:** Determines the length of the used impulse response sample (displayed in seconds).
- **Resample:** Stretches the impulse response sample (0.25x – 4x).
- **Latency:** By default, the plugin has 512 sample latency, but if desired, latency can be adjusted to a lower setting and even to 0 – with the exception of Apple Logic, which for technical reasons needs a non-zero latency. For certain systems, a higher latency setting is preferable at very low latencies of your soundcard, such as 64 or 128 samples. At a buffer size of 256 or 512 samples, the best performance can be achieved with zero latency.

Equalizer

The Vienna Suite Equalizer operates like a standard EQ filter, but it is powered by highly sophisticated 64-bit processing algorithms. The filter response display provides comprehensive control and visual feedback of the EQ curve being applied. Five fully parametric bands of EQ plus additional high and low pass filters are completed with an integrated 120-band spectrum analyzer.

We have included **Factory Presets** for many instruments contained in the wide array of Vienna Instruments Collections, making it very comfortable to start your mixing session.

These Factory Presets are subdivided into so-called "Resonance Menu" Presets and "Character" Presets. In the "Resonance Menus", up to 5 frequencies were localized for the respective instrument/ensemble which could be problematic in the mix. Simply activate the desired EQ filter band's preset parameters by clicking on it; now you can determine whether the disturbing resonances of this instrument have been filtered out. Also, try combinations of several bands, and reduce or increase the attenuation as needed. The idea is to get a better placement of the original, pure sound of the Vienna Instruments in a complex mix without larger adjustments.

"Character" Presets work out an instrument's or ensemble's manifold sound possibilities. Here, the pure

sound of a Vienna Instrument is deliberately altered in different directions. These Presets multiply the sounds of your Vienna Instruments Collection – a true fountain of inspiration!

Walkthrough Video



Watch the video

Controls



- **Filter Response Display:** Shows the response curve and frequency analysis for the current settings. The (horizontal) frequency range is from 20 Hz to 20 KHz, and the (vertical) amplitude scale is in dB and is adjustable between 6 and 24 dB.
- **dB Scale:** Lets you specify the vertical scale (dB) of the filter response display in 6-dB steps (6, 12, 18, 24 dB). Of course, this doesn't change the sound itself.
- **Volume Slider:** lets you adjust the volume of the processed signal.
- **EQ Filter Bands:** the five bands of the EQ that can be independently enabled and programmed. The numbers and colours of the EQ Filters correspond to the numbers and colours of the filter handles in the filter Response Display. Click on the coloured filter band buttons to enable/disable the corresponding filter.
- **Low Cut and High Cut:** Fixed at Butterworth Q (0.71), both High and Low Cut offer 4 filter slopes for different slope (fall off) characteristics (12, 24, 36, 48 dB)
- **Gain:** Adjusts the peak gain of the corresponding filter in a range from –24 dB to +24 dB.
- **Filter handle:** Drag this handle to graphically adjust the filter's boost/cut and/or frequency. When adjusting the filter handle, `Shift` locks the frequency; `Alt` / `Option` locks the gain. Double-clicking a handle resets the respective band's gain to the default.
- **Q setting:** Adjusts the filter width of the corresponding filter band from 0.1 to 10. The smallest setting of 0.1 will create a shelving EQ, whereas a setting of 10 will give you the smallest possible filter width. Use the mouse wheel (or press `Ctrl/Cmd + click and drag`) to adjust the Q setting for the currently selected filter. The given filter handle has to be highlighted.
- **Composite curve:** Shows the overall response curve of the current settings in the Filter Response Display.
- **Individual Filter Curve:** Each individual filter's response curve is displayed in the filter's color when selected.
- **Analyzer:** Enable/disable a 120-band spectrum analyzer. If you're going for even less CPU consumption, the analyzer should be switched off.
- **Analyzer Display Mode:** Choose between Input and Output signal display (pre/post EQ).

Compressor

The Vienna Suite Compressor lowers the level of the input when it exceeds the threshold. The amount of attenuation is determined by the Attenuation / Ratio and input level. When the input level goes above the threshold, attenuation is added gradually to reduce distortion. Usually, the incoming signal is identical to the input to the gain reduction stage. The compressor's attack time determines how fast the compression is applied – this way you can sculpt the attack of your instruments, making them softer or more aggressive. The release time determines the time it takes the compressor to decrease the applied gain reduction. Depending on the length of the release time you can minimize the "sonic body" of an instrument in comparison to the attack sound, or you can apply extremely short release times to emphasize it.

The Vienna Suite Compressor can shape the controlling signal (aka "Sidechain") to allow for even more musical compression modes.

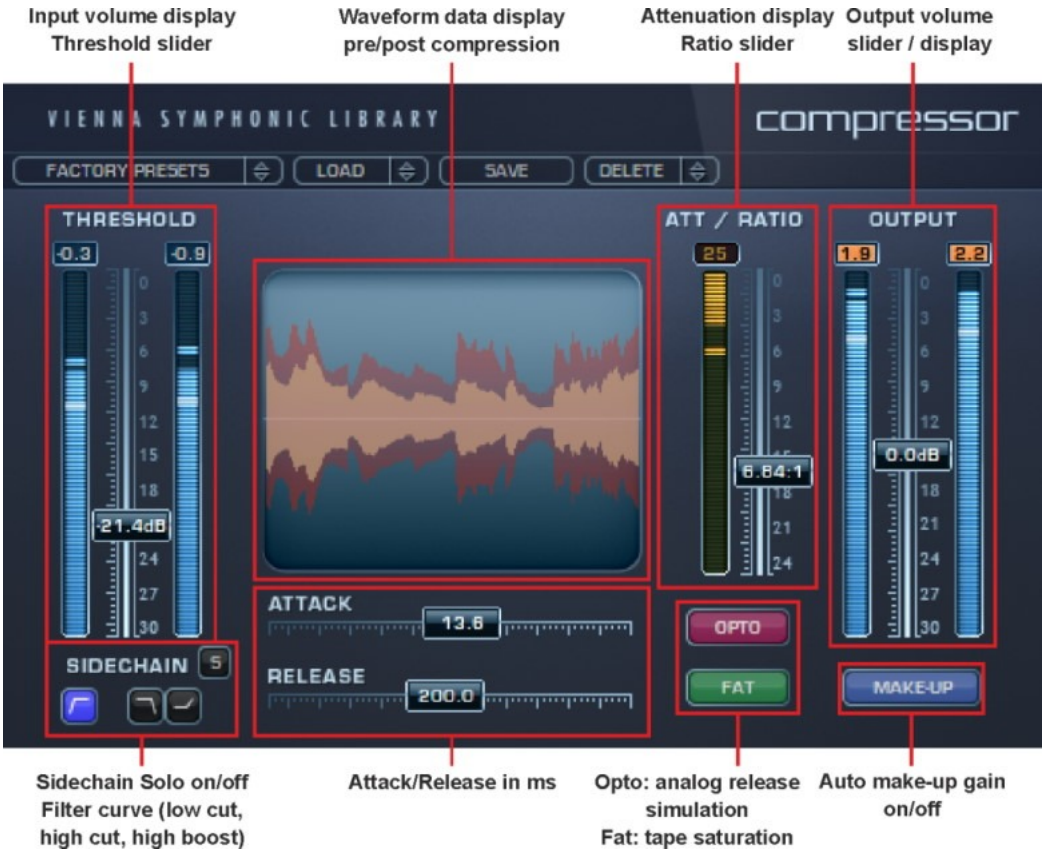
Additionally, you can use the Compressor as a Sidechain plug-in within your sequencer.

Walkthrough Video



Watch the video

Controls



- **Threshold:** is the level above which the signal is reduced. A low threshold means that a larger portion of the signal will be processed.
- **Sidechain:** Choose between Low Cut (120 Hz), High Cut (4 kHz) (both at 12 dB/oct) and High Boost (+6 dB at 4 kHz). The Sidechain works exclusively with the internal sidechain filter. Use low cut when the signal starts to "pump" due to high energy in the lower frequencies. Use high boost when you want to get rid of excessive high-frequency content, e.g. Voices with strong "sss" and "shhhh". Use the high cut when the signal's attacks begin to sound lifeless and flat, especially when using fast attack times. The settings in the Sidechain only affect the Compressor's controls, NOT the signal itself!
- **Sidechain Solo:** Solos the sidechain signal.
- **Waveform Data Display:** Shows the input and output signals in real-time.
- **Attack:** Sets the time the compressor needs to increase the gain reduction to reach the level determined by the ratio. Values: 0.1 – 500 ms.
- **Release:** The 'release phase' is the period when the compressor decreases gain reduction to the level determined by threshold and ratio (or to zero, once the level has fallen below the threshold). Values: 10–5000 ms.
- **Attenuation:** Displays the current amount of compression applied to the signal.
- **Ratio:** The ratio determines the input/output ratio for signals above the threshold. For example, a 3:1 ratio means that a signal overshooting the threshold by 3 dB will leave the compressor 1 dB above the threshold.

- **Opto:** Activates the optical release simulation, resulting in a more "breathing" sound. Compared to the same settings without Opto Mode, a little less compression will be applied to the signal.
- **Fat:** Activates a process which makes the signal sound "thicker", similar to tape compression. When FAT mode is on, the sound is hard-clipped at 0 dB. With a certain configuration (when the signal may exceed 0 dB), this may cause distortion in the attack segment. In this case, please switch off the make-up gain or turn off FAT mode.
- **Make-up:** Automatically compensates for the gain loss in the compression based on the current threshold, ratio and attack settings.
- **Output:** Determines the signal's output volume.

Limiter

The Vienna Suite Limiter maximizes the apparent volume within the available dynamic range by reducing the distance between average program levels and their associated peaks. Too much limiting will make your signal sound unnatural. Just the right amount of maximization will make your signal seem loud without ever exceeding the available dynamic range. Processing at 64-bit floating point resolution, which provides the highest fidelity with no added noise, the comfort of a real-time input/output waveform display combined with an auto-mode for transparent maximization makes it very intuitive to operate.

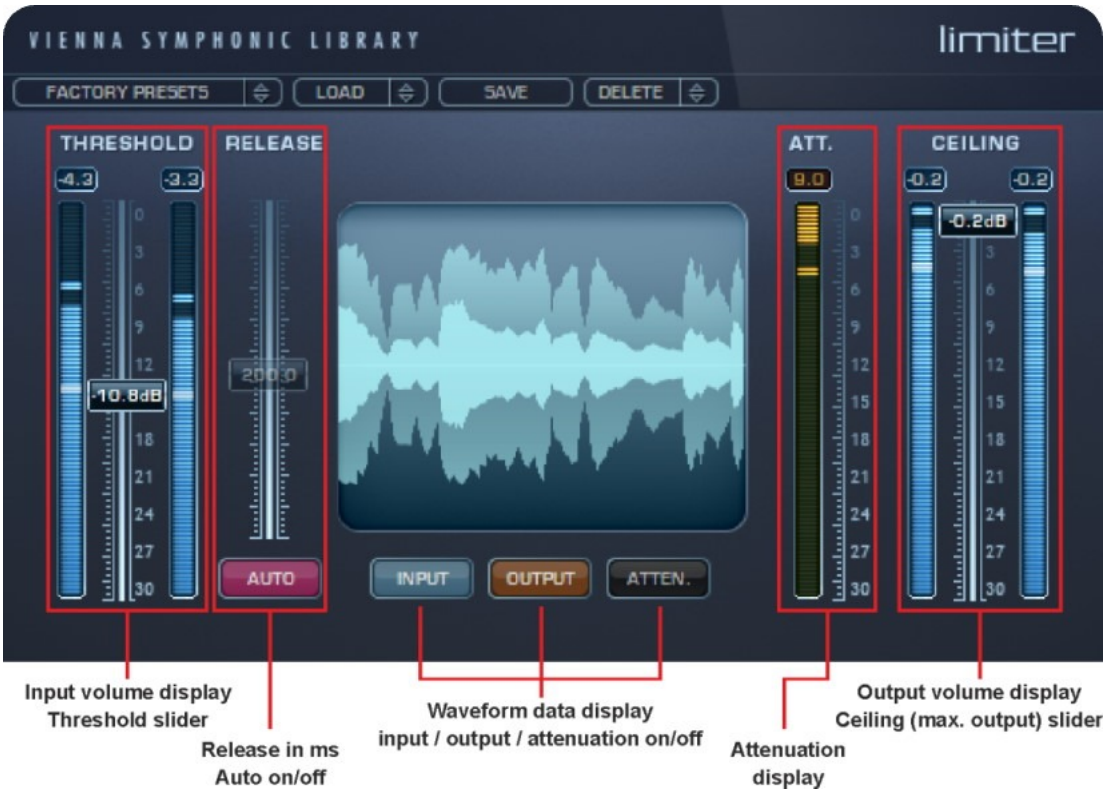
> **Important**
>
> In spite of the Vienna Suite Limiter's excellent quality, exaggerated limiting will result in unnaturally sounding signals.

Walkthrough Video



Watch the video

Controls



- **Threshold:** Sets the point at which the limiter starts to affect the input signal. Lowering the threshold

setting has the effect of raising the output gain of the signal. The average level of the signal is increased while the limiter assures that the signal will not exceed the ceiling level.

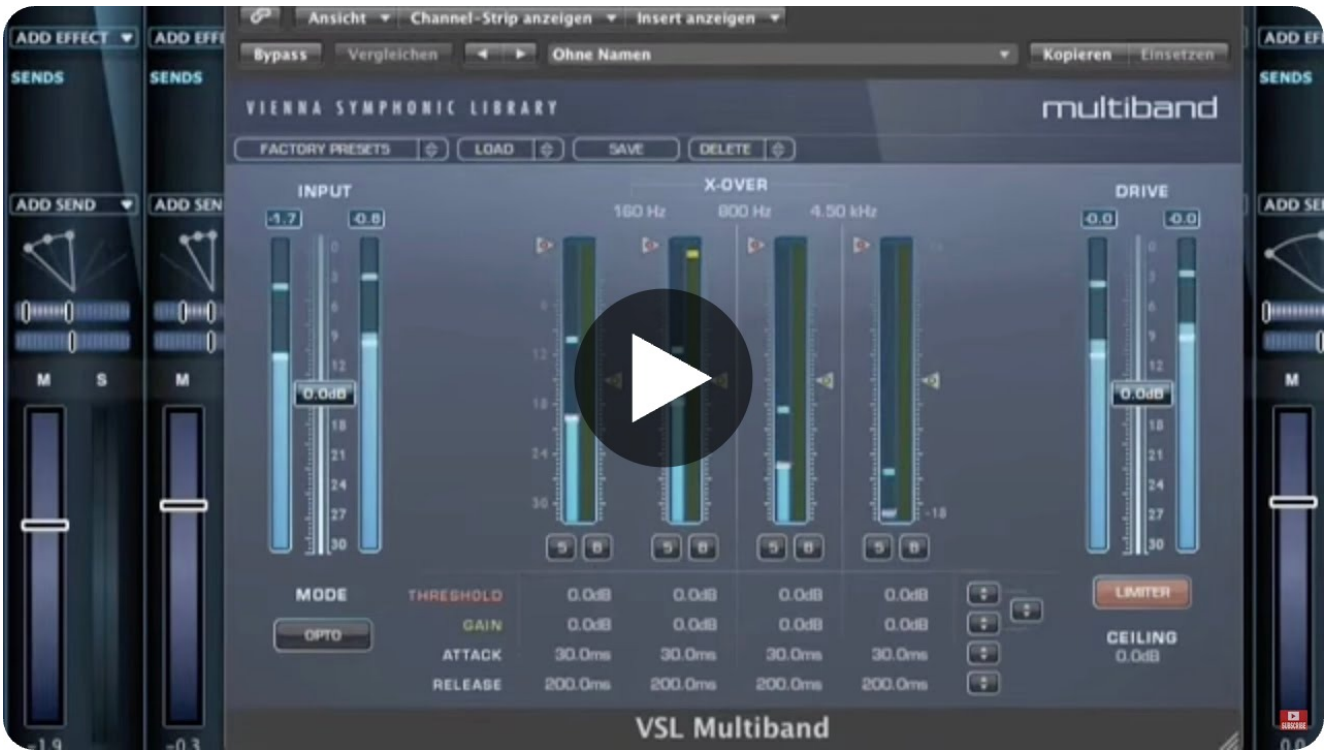
- **Release:** Controls how long it takes for the limiter to recover from attenuating the signal after it has dropped below the threshold. Values: 10–5000 ms.
- **Auto:** Disables the manual release slider and lets the limiter control the release time depending on the input signal, resulting in the most transparent limiting possible.
- **Waveform Data Display:** Shows the input and output signals as well as the attenuation graph, which can be individually switched on and off with the buttons below. If you need to reduce CPU load a bit, you can switch off the entire waveform display.
- **Attenuation:** Displays the current amount of limitation applied to the signal.
- **Ceiling:** Sets the maximum output of the limiter.

Clicking on the numerical peak displays above the meters resets the peak values.

Master Equalizer

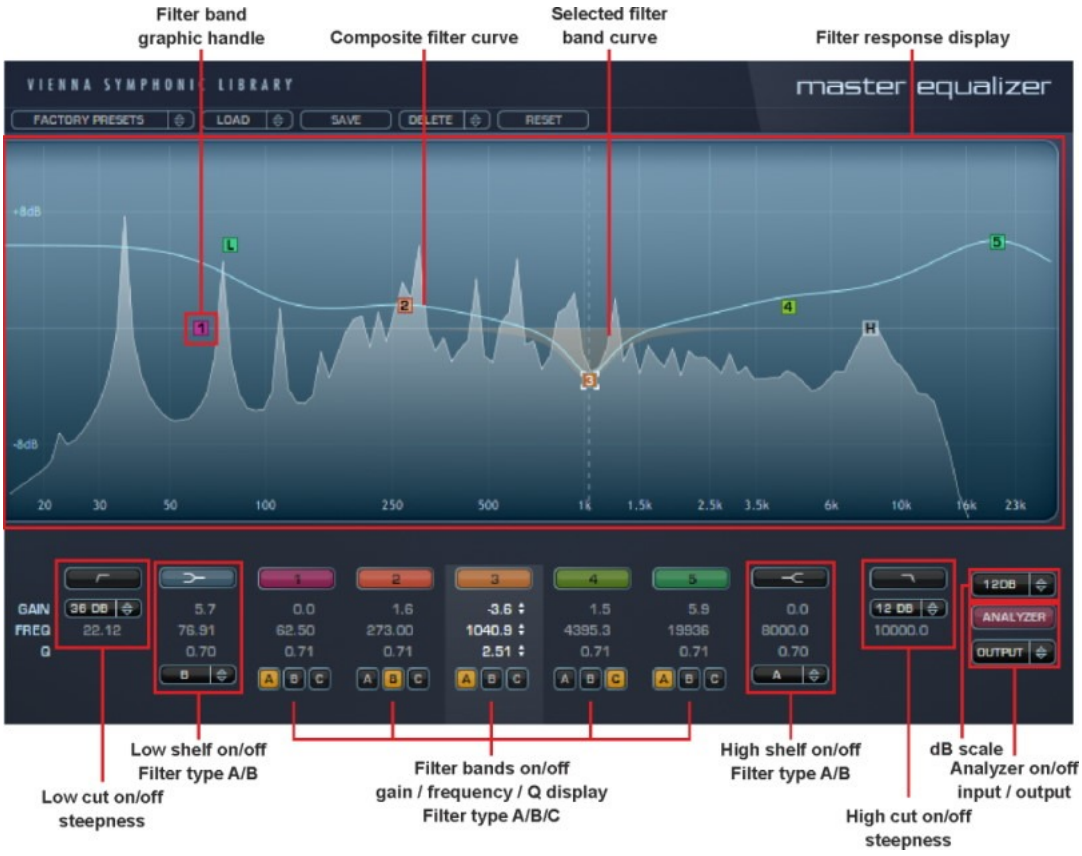
A precision tool for advanced application, the Master Equalizer is the deluxe version of the Vienna Suite Equalizer. There are 3 filter types to choose from on 5 frequency bands and an additional high shelf filter – as well as Low Cut and High Cut filters. The use of 4x oversampling makes internal calculations even more precise. The filter response display is bigger, allowing a more detailed resolution of the frequency spectrum. This Master Equalizer will need more CPU power than the standard Vienna Suite Equalizer and is the first choice for delicate sonic tasks.

Walkthrough Video



[Watch the video]

Controls



- **Filter Response Display:** Shows the response curve and frequency analysis for the current settings.
- **dB Scale:** Lets you specify the vertical scale (in dB) of the filter response display in 6-dB steps (6, 12, 18, 24 dB). The sound will not be affected by this setting.
- **EQ Filter Bands:** The EQ's five bands which can be independently enabled and programmed. The numbers and colours of the EQ Filters correspond with the numbers and colours of the filter handles in the Filter Response Display. Click on the coloured filter band buttons to enable/disable the corresponding filter.
- **Filter Types for the peaking filters:**
  - *Type A:* the most versatile filter type, will give excellent results over the entire frequency range.
  - *Type B:* this filter can be more detailed in the mid and low-frequency ranges.
  - *Type C*: common standard digital filter type.
- **Low Shelf / High Shelf Filter:** Choose between two shelving types. *Mode A* behaves like a "normal" shelving filter and allows for Q adjustment. *Mode B* is a special filter which yields a better phase response and thus creates a more "musical" result, disregarding the Q settings.
- **Low Cut and High Cut:** With an adjustable Q between 0.40 and 2.50, both High and Low Cut offer 4 filter slopes for different slope (fall-off) characteristics (12, 24, 36, 48 dB)
- **Gain:** Adjusts the peak gain of the corresponding filter in a range from –24 dB to +24 dB.
- **Filter handle:** Drag this handle to graphically adjust the filter's boost/cut and/or frequency. When adjusting the filter handle, `Shift` locks the frequency; `Alt` / `Option` locks the gain.
- **Q setting:** Adjusts the filter width of the corresponding filter band. Use the mouse wheel (or press `Ctrl/Cmd + click and drag`) to adjust the Q setting for the currently selected filter. The given filter handle has to be highlighted.
- **Composite curve:** Shows the overall response curve of the current settings in the Filter Response Display.
- **Selected Filter Band Curve:** Each individual filter's response curve is displayed in the filter's color when selected.
- **Analyzer:** Enable/disable a 120-band spectrum analyzer. If you're going for less CPU consumption, the analyzer should be switched off.
- **Analyzer Display Mode:** Choose between Input and Output signal display (pre/post EQ).

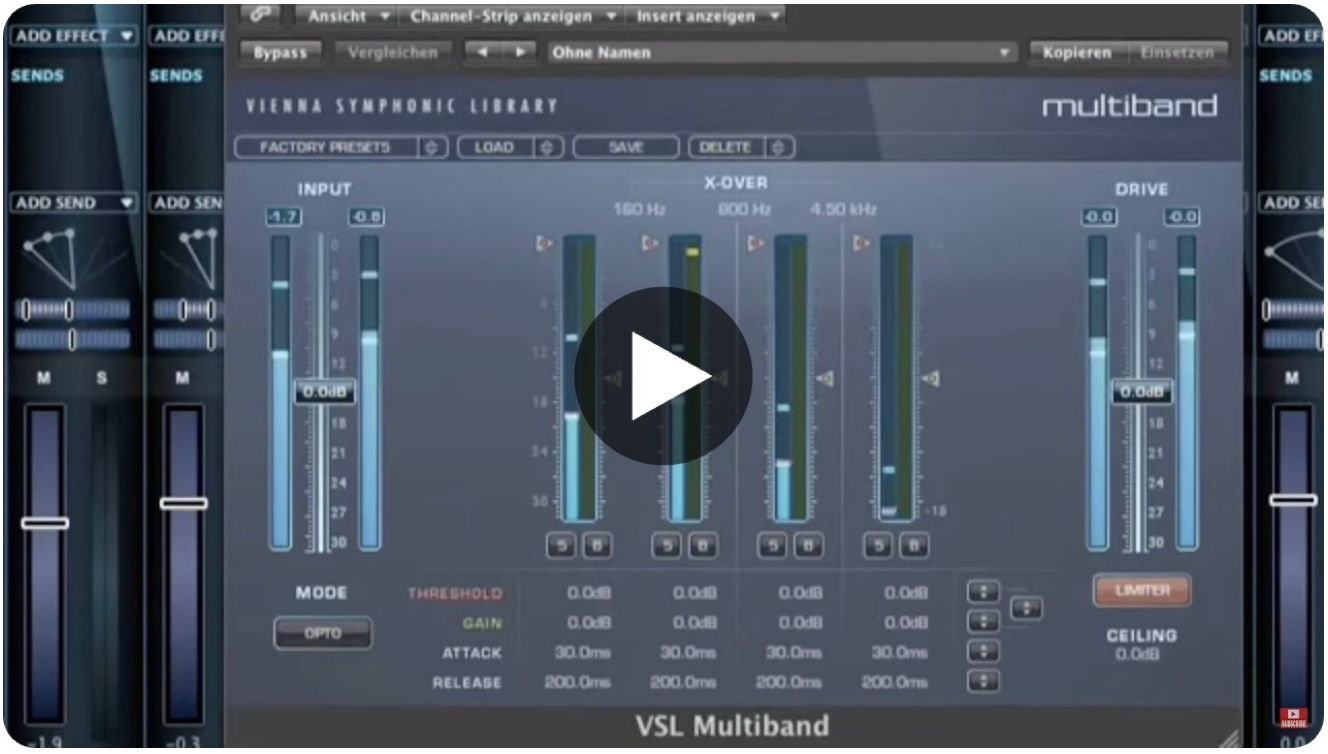
Multiband Limiter

The Vienna Suite Multiband Limiter offers precise level control of 4 specific frequency bands of your audio signal. It can be used in the mastering process to control bass, mid-range or treble frequencies independently of each other. A wide-band limiter (like the Vienna Suite Limiter) would respond to the loudest parts of the audio signal regardless of their frequency. Another common use of frequency-specific limiting/compression is de-essing. This allows you to control the sibilants of an audio signal without affecting lower frequencies.

A specially designed crossover network guarantees a transparent signal flow and contributes to the unique character of the Multiband Limiter. Please note that due to these crossover filters the Vienna Suite Multiband Limiter can't be bit-transparent, even without any gain reduction applied.

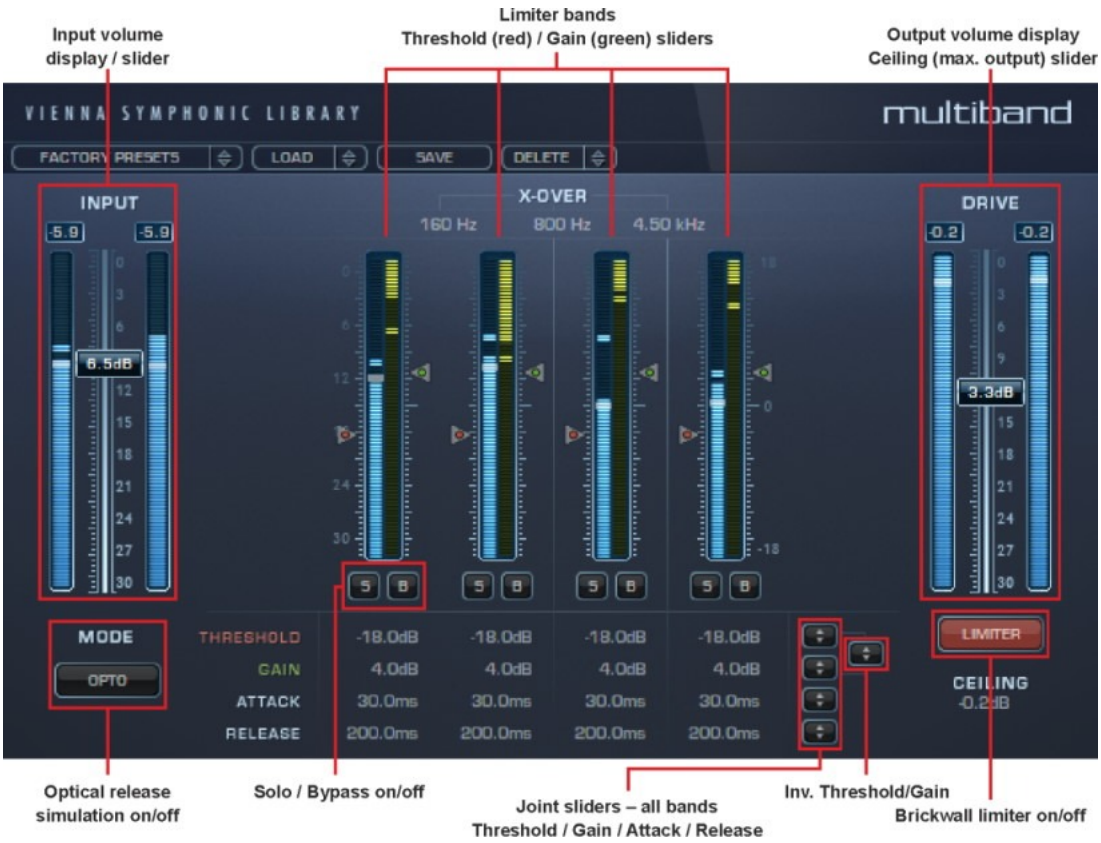
The Multiband Limiter will allow you to get a maximum of perceived loudness on an individual track or the entire mix while maintaining sonic clarity. The integrated output brickwall limiter will take care that your output signal will not exceed the specified ceiling value.

Walkthrough Video



Watch the video

Controls



- **Input Gain:** Boosts or attenuates the incoming signal (max. +18 dB).
- **Opto Mode:** activates the optical release simulation, resulting in a more "breathing" sound. Compared to the same settings without Opto Mode, a little less compression will be applied to the signal.
- **X-Over:** 3 adjustable crossover points separate the frequency spectrum into 4 bands for separate limiting.
- **Threshold:** Determines the level above which the limiter will have an effect. Threshold can be set with the red triangular slider to the left of the individual band meters by clicking and dragging the mouse over the value box, or by double-clicking the value box and entering the value with the keyboard (hit enter/ return to confirm).
- **Gain:** Boosts or attenuates the given frequency band. Gain can be set with the green triangular slider to the right of the individual band meters by clicking and dragging the mouse over the value box, or by double-clicking the value box and entering the value with the keyboard (hit enter/return to confirm).
- **Inverse Threshold/Gain Box:** Clicking this box and dragging the mouse downwards will lower all thresholds while simultaneously increasing the gain for all bands, and vice versa. This control provides the easiest and fastest way to use the Multiband Limiter.
- **Attack:** Determines the time it takes for the limiter to react after the signal has exceeded the threshold.
- **Release:** Determines the time it takes for the limiter to return to zero after the input signal has dropped below the threshold level.

- **Band Solo:** Mutes the non-soloed frequency bands. Very useful for hearing what the limiter does within a specific frequency band.
- **Band Bypass:** Disables limitation for the selected frequency band.
- **Drive:** Sets the input level of the output brickwall limiter.
- **Limiter (Brickwall Limiter):** Maximizes the apparent volume of the output signal within the available dynamic range by reducing the distance between average program levels and their associated peaks. This additional wide-band limiter will not exceed the ceiling level assigned (brickwall limiting). It gradually turns red the more limiting is applied.
- **Ceiling:** Sets the maximum output of the Limiter. You can adjust the value by clicking and dragging your mouse or double-clicking the value box to enter the value with your keyboard (hit enter/return to confirm).

> **Important**
>
> Filtering the signal through 4 separated frequency bands means that even without dynamic adjustments, the output signal will not be bit-identical with the input signal!

Exciter

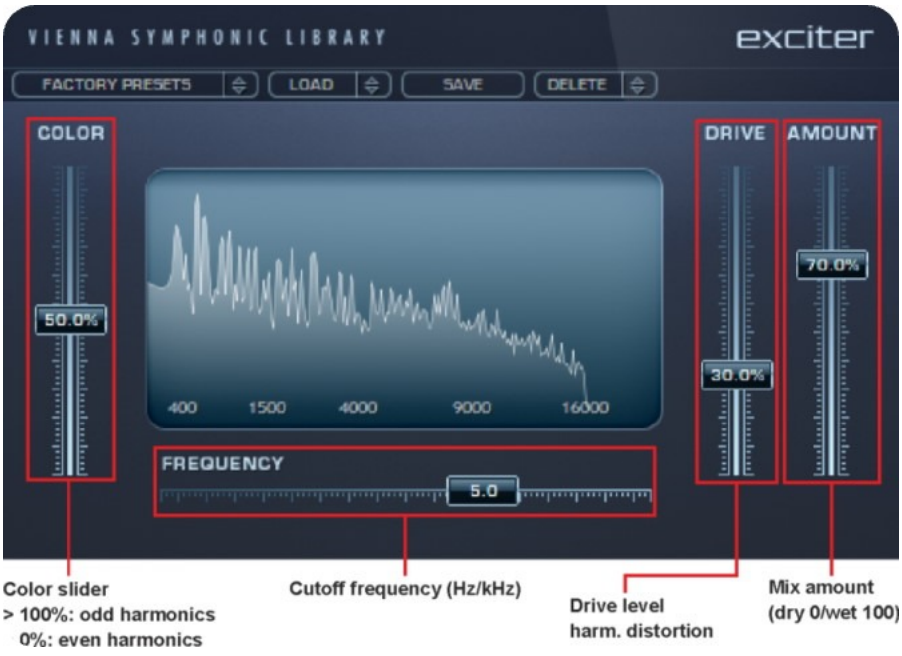
If you are looking for that "glowing" Hollywood sound, the Exciter will create higher-order harmonics from the fundamental frequency signals present in the recording which do just that. Processed with 4 x oversampling and unique distortion algorithms to completely eliminate aliasing, the result can be monitored with a high-frequency FFT spectrum analyzer.

Walkthrough Video



Watch the video

Controls



- **Color Slider:** Changes the musical coloration of the signal between even and odd harmonics. Setting the slide to 100% will only add odd harmonics, whereas 0% will only add even harmonics to your signal.
- **Frequency:** Adjust the cutoff frequency for the Exciter to take effect. The signal part below the chosen cutoff frequency will not be altered.

- **Drive:** Sets the distortion level of the harmonics added by the Exciter.
- **Amount:** Adjusts the mix between the unprocessed signal and the generated harmonics/overtones. This acts like a "send" channel: 0% equals the original signal, and 100% adds the maximum of the processed signal.

PowerPan

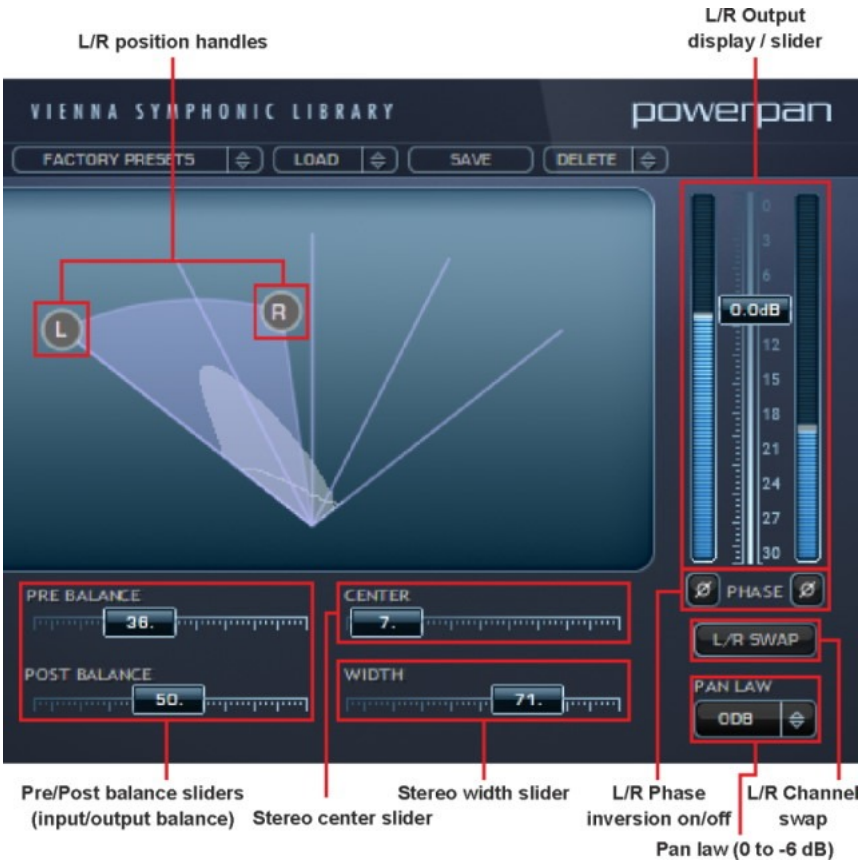
A very useful tool for stage positioning of instruments, featuring real-time audio angular analysis. By collapsing the stereo image of an audio signal, you will not lose any of its information (which happens with "ordinary" balance faders that simply lower the volume of one side of the signal). And there are extras, too: independent Phase Inversion for the left and right channels, and Channel Swap.

Walkthrough Video



Watch the video

Controls



- **Display:** This shows the signal's acoustical position in the stereo field – the two handles L and R let you adjust stereo width and position.
- **Pre Balance:** Adjusts the left or right channel's volume at the plug-in's input.
- **Post Balance:** Adjusts the left or right channel's volume at the plug-in's output.
- **Center:** Positions the audio signal in the stereo field.
- **Width:** Determines the width of the stereo signal. 100% means the full stereo width of the source.
- **Volume:** Boosts or attenuates the output signal.
- **Phase:** Individually inverts the phase of the left and right channels.
- **L/R Swap:** Exchanges the left and right positions of the stereo signal; also displayed in the pan view.
- **Pan Law:** Determines how the signal level is affected by panning. Without compensation, the combined loudness of the left and the right signal will be higher if a channel is panned to the centre than if it is panned to one side. The Pan Law allows for attenuation of the signal when it is panned to the centre. – The optimum value depends a lot on the input source.
  Adjustable from 0 to 6 dB (0 dB actually turns off constant power panning).

Analyzer

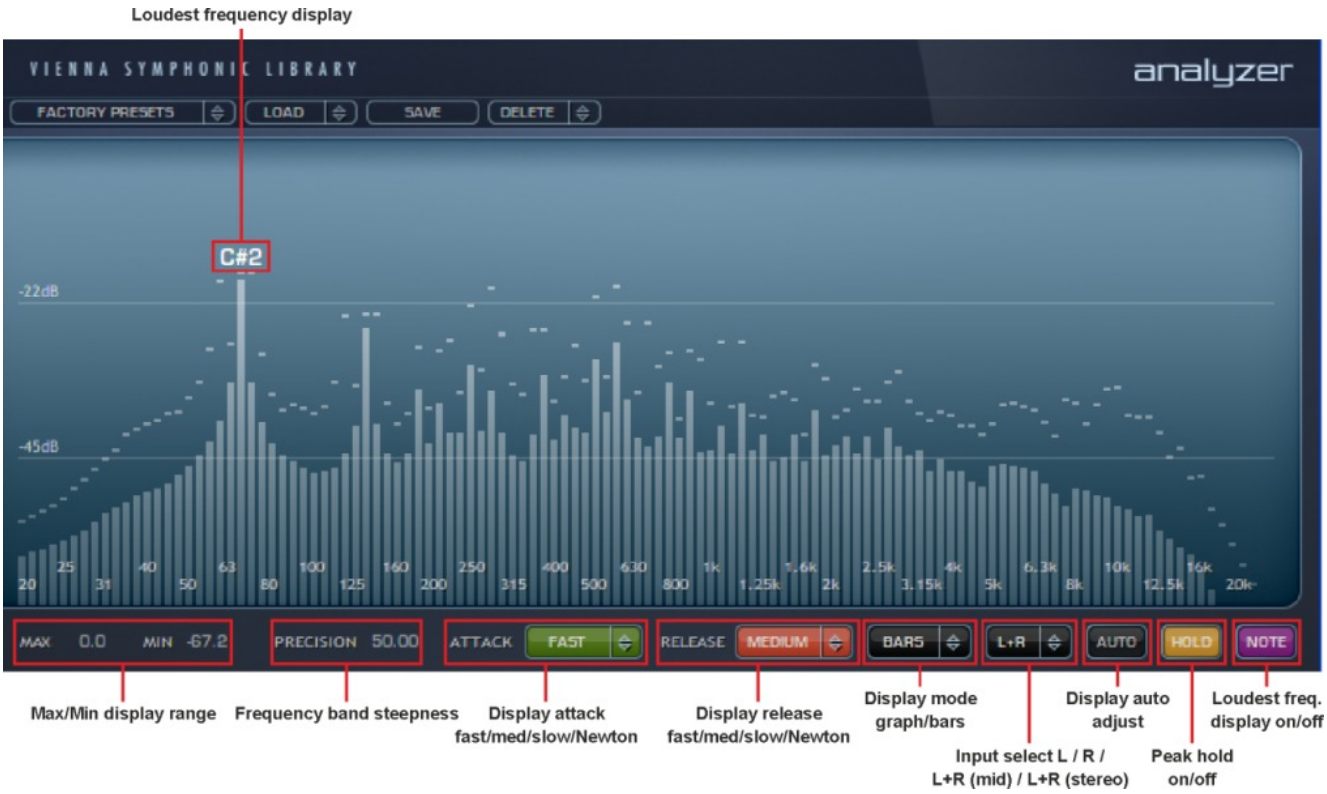
Always convenient: This analog-modeled 120-band spectrum analyzer comes with a fast and accurate display, an auto mode for automatic range management, and a note display that locates the loudest sounding frequency in your input signal.

Walkthrough Video



Watch the video

Controls



- **Max/Min:** Sets the range of the display in dB.
- **Precision:** Sets the steepness of the analyzed filter bands. Increasing precision will yield a more accurate readout of frequencies.
- **Attack:** Determines the time it takes the Frequency Display to react to the audio signal. Choose between fast, medium, slow and Newton (which emulates the gravity on this planet).
- **Release:** Determines the time it takes the Frequency Display to drop to zero. Choose between fast, medium, slow and Newton (which – again – emulates the gravity on Earth).
- **Display Mode:** Select how the frequency spectrum should be displayed; choose between Bars and Graph.
- **Input Selection:** Choose from left, right, left+right (m, the mono middle, without stereo parts) and left-right (s, the combined stereo signals, without mono middle parts) audio signal.
- **Auto:** Will automatically adapt the graphic display in order to increase the accuracy of the analysis for low-level incoming signals.
- **Hold:** Keeps the highest peaks of all 120 frequency bands displayed. Double-clicking the display will reset the Hold function.

Goniometer

The Vienna Suite Goniometer is a great tool to display the phase differences between two audio channels (resulting from the similarity of the signals). You get a reading of the stereo width and the amount of out-of-phase signal parts.

The additional correlation indicator offers a simple solution to check the mono compatibility of a stereo signal. It also lets you identify unwanted phase shifts or polarity inversions.

We have optimized the Vienna Suite Goniometer to work at the lowest possible CPU demand, which allows you to use it as a constant controlling device for your mixes. Oversampling allows for a precise signal interpretation and the GLOW feature enables you to control the signal without constant observation of the display.



- **Display window:** Shows the Goniometer interpretation of the chosen signals. The display is independent of the absolute level of the signal, the level of the signal is automatically adjusted in a big dynamic range (-50 dB to 0 dB).
- **Analog**: Activates the interpolated display of the signal in the Goniometer (similar to analog goniometers with a cathode ray tube).
- **Detail**: Activates the display of a second, brighter goniometer signal. This feature lets you observe the actual signal while using higher GLOW settings.
- **Clip**: Activates the "Clipping" of signals, starting at 0 dB FS amplitude. This lets you judge whether and how a signal is clipping. Clearly, clipping stereo signals will show a rhombus shape in the goniometer display. If Clip is deactivated, the amplitude of the signal will be adjusted dynamically to an area below 0 dB FS.
- **Goniometer Glow (GON. GLOW):** Lets you adjust the time for the glowing effect in the goniometer display. At "0", older data will fade very quickly. At "100", older data will fade out slower and the goniometer display will show a longer time bracket of the input signal. Clicking on the Goniometer display will reset the Goniometer Glow.
- **Correlation:** Displays the correlation of the input signal in a range from -1 to +1. The determined values mean the following:

- **+1**: Identical signals in both channels.
- **1 to 0**: The input signals are (more or less) mono-compatible.
- **0**: The input signals are totally uncorrelated (or a channel is missing).
- **0 to -1**: The input signals are not mono-compatible.
- **-1**: The input signals are identical, but with polarity inversion on one channel (no mono compatibility).

- **Correlation Glow (COR. GLOW):** Lets you adjust the time for the glowing of the correlation value in the goniometer display. At "0", only the current value is visible. At "100", you can see the progression of the correlation degree over a longer period of time. Clicking on the Goniometer display will reset the Correlation Glow.
- **Input Channel Selection:** Select the input signals that will be analyzed. In a stereo track, only "L-R" will be available, a surround track will let you choose a signal pair from multiple input signals.
- **Level:** Peak-Metering for both selected input signals.

Support and the VSL Forum

We hope that this manual was a helpful companion for getting to know our plug-ins.

However, if you have any further questions don't hesitate to visit our Forum or send an e-mail to our Support Team.

Also, please check out our Video Tutorials which will provide you with further details about Vienna Instruments and our software products.

And now – enjoy the Vienna Suite!
- Your Vienna Symphonic Library Team

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Vienna Suite - Version History

Windows 10 and newer
macOS 10.14 and newer

v2.0.59 - October 20th, 2022

Hotfix

- Win: Plugins asking for a Vienna Imperial license

Improved

- VST: Scanning speeds

v2.0.56 - October 19th, 2022

Changed

- Switch from eLicenser to iLok protection
- Attention: All newer versions are iLok only
- Attention macOS: Mojave (10.14) is the new minimum system requirement

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v1.5.3329c - October 28, 2019

Added

- macOS: Catalina compatibility

v1.5.3329 - July 13, 2018

Fixed

- Crash when using Hybrid Reverb and having audio buffers set to sizes which are not a power of two

Removed

- RTAS plug-ins

v1.5.3285 - December 20, 2017

Fixed

- VST plug-ins were not installed (Bug introduced in v1.5.3282)

v1.5.3282 - December 20, 2017

Added

- New Synchron Strings presets to Hybrid Reverb, Master EQ and PowerPan

Fixed

- Compressor side chain high shelf filter not working
- Potential audio issues due to resampling impulse responses in Convolution/Hybrid Reverb - Fixed: Master EQ Analyzer only showing output values in certain hosts

Attention macOS: Changed minimum requirement to macOS 10.10

v1.5.2049 - April 2, 2015

Fixed

- Analyzer sometimes not responding to audio input
- Latency setting sometimes not recalled in Hybrid Reverb and Convolution Reverb
- Win: Cubase 8 process not closing when Vienna Suite plug-in present in current project

Added

- format selection to Software Installer

v1.5.1929 - October 21, 2014

Fixed

- Win: Runtime dll's not recognized and installed properly

v1.5.1927 - October 20, 2014

Fixed

- Improperly reported plug-in version information
- Mac OS X: Bundle version mismatch on OS X

v1.5.1922 - October 16, 2014

Fixed

- Hybrid Reverb popping dialog when eLicenser key with an appropriate license is not connected
- Improper bypass status with VST3 plug-ins in Cubase/Nuendo
- VST3 plug-ins hanging Cubase 7.5 on exit
- Goniometer signal bypass problem in certain hosts

Added

- Mac OS X: Yosemite compatibility

v1.5.1887 - March 5, 2014

Added

- New presets: Dimension Violas/Basses, Trumpet in Bb muted and more (60 new presets)

Fixed

- Hybrid Reverb: GUI bug in drawing of impulse names
- Convolution Reverb: Don't process HP/LP filters if settings are at min/max
- Convolution Reverb: up/down buttons were not working
- Convolution Reverb: Bug in HP filter processing
- Mac OS X: Rare "installation not allowed, minimum OS version required" problem

v1.5.1880 - October 24, 2013

Added

- mono monitoring button to Goniometer

Fixed

- Hybrid Reverb clicking on some low-frequency sounds

v1.5.1877 - September 2, 2013

Improved

- time needed to reset Convolution and Hybrid Reverb plug-ins

Fixed

- Win: Only install C++ Redistributable if necessary
- Win: Analyzer AAX-64 not showing in ProTools 11
- Win: Won't install on 32-bit Windows versions.

v1.5.1873 - July 29, 2013

Fixed

- Win: Presets couldn't be loaded (bug introduced in 1.5.1872)

v1.5.1872 - July 23, 2013

Fixed

- Win: Analyzer AAX-64 not showing in ProTools 11
- Win: Only install C++ Redistributable if necessary

v1.5.1862 - June 26, 2013

Added

- RTAS-AAX migration

v1.5.1860 - June 22, 2013

Added

- Full support for Pro Tools 11 (please note: AAX version now requires ProTools v10.3.5 or later).
- AAX known issues: RTAS plugins will not automatically be converted to AAX-64 plugins.

Fixed

- Hybrid Reverb pops when switching between certain presets
- Hybrid Reverb crashing with buffers larger than 4096
- Hybrid reverb causing crackling with denormal inputs

v1.5.1833 - November 22, 2012

Fixed

- Goniometer: Changed glow parameter behavior to allow even shorter glowing (former 0% is now around 35%, behavior at default values did not change)
- Analyzer AUTO button causing limited output with VST3 plug-in

v1.5.1819 - October 15, 2012

Added

- 32- and 64-bit AAX plug-in versions
- External sidechain for Compressor
- Dimension Strings presets

Changed

- Mac OS X: Changed system requirements to minimum OS X 10.6 and removed PowerPC support

v1.4.1758 - August 15, 2012

- For these and older changes downlaod the changelog in the Manuals Download Section