

Evo Channel

FLUX:: Immersive

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1 Evo Channel

FLUX
sound and picture development

Manual EVO.CHANNEL V1.0

The image displays the EVO.CHANNEL software interface, a digital audio workstation (DAW) plugin. The interface features a frequency response graph at the top, showing a multi-band equalizer with various frequency sliders and gain controls. Below the graph, there are several control sections: a 'DRIVE' section with a gain knob, an 'EQUALIZER' section with four frequency sliders (100 Hz, 331 Hz, 331 Hz, 331 Hz) and their respective gain and Q factor settings, and a 'FILTER HI' section with a high-shelf filter. The interface also includes a 'PHASE' section with a phase shift knob and a 'GAIN' section with a gain knob. The bottom of the interface shows a 'MODE' selector set to 'Clean', a 'Preset name' field, and a 'Copy A' button. A physical product box for the EVO.CHANNEL plugin is shown in the foreground, featuring a portrait of a man and the text 'EVO.CHANNEL' and 'Yves JAGET'.

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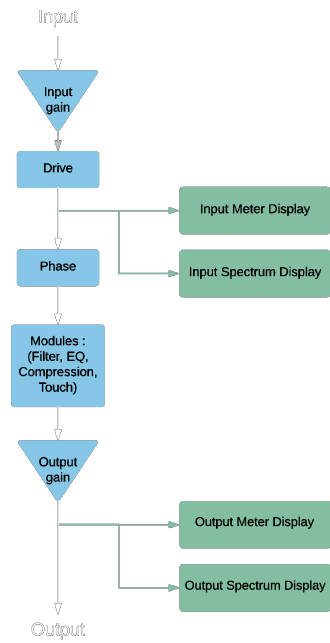
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1.1 Introduction

EVO Channel - The Ultimate Channel Strip Redefined

The philosophy of the analog Channel Strip, the essential element of all analog consoles, is about being efficient, fast, and making things sound great already from the start. EVO Channel is designed to be as efficient, fast and hands-on, adding the advantages of creative digital innovation in terms of workflow and sound, providing everything you need to stay in control of your sound!





EVO Channel architecture :

2 General Settings

2.1 Bypass

Global bypass, when pressed, the signal is routed directly from the inputs to the outputs.

Value Range : Enabled/Disabled

Default Value : Disabled

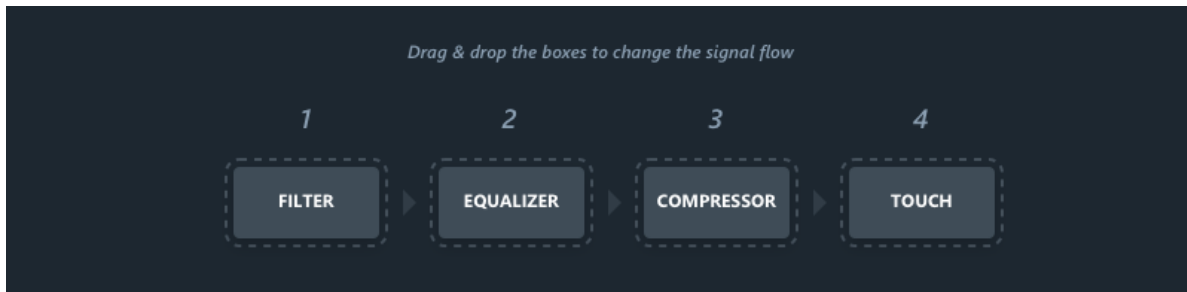
2.2 Skin

The look of the EVO Channel user interface.

Value Range : Light/Dark

Default Value : Light

2.3 Flow



Flow allow you to change the order of the modules in EVO Channel. Simply drag and drop a module to set your perfect signal flow.

3 Module Settings

3.1 Analyser

3.1.1 Analyser Switch

The spectrum analyzer originates from the Flux:: Pure Analyzer and has been optimized for the EVO Channel. It gives you an accurate direct view of what's going on in the different frequency domains of your audio. When switched on, the spectrum analysis is activated and displayed in the equalization view panel. The grey waveform corresponds to your input signal (post gain/drive), the same as the input meter. The black one is the output signal (post output gain), the same as the output meter.

Value Range : Enabled/Disabled

Default Value : Enabled

3.1.2 Analyser Slider

Controls the amount of frequency detail of the curve. Move the slider to the left to get a more smoothed curve, and to the right to get more details.

Value Range : No Value

3.2 Input

3.2.1 Input Gain

The input gain control trims the level of the signal at the input of EVO Channel. The meter shows both RMS signal (VU-Meter, blue) and peak signal (peak meter, green), from -24 to +18 dB range, referenced at -18dB.

Value Range : -24.0 dB / +18.0 dB

Colors : - Blue : RMS Value - Green : Peak Value

Default Value : 0.0 dB

3.2.2 Drive

In EVO Channel a signal Drive is available direct at the input Gain for restoring and maintaining the vitality of the sound.

The drive module has been specially designed to add a soft saturation and warmth to your audio tracks.

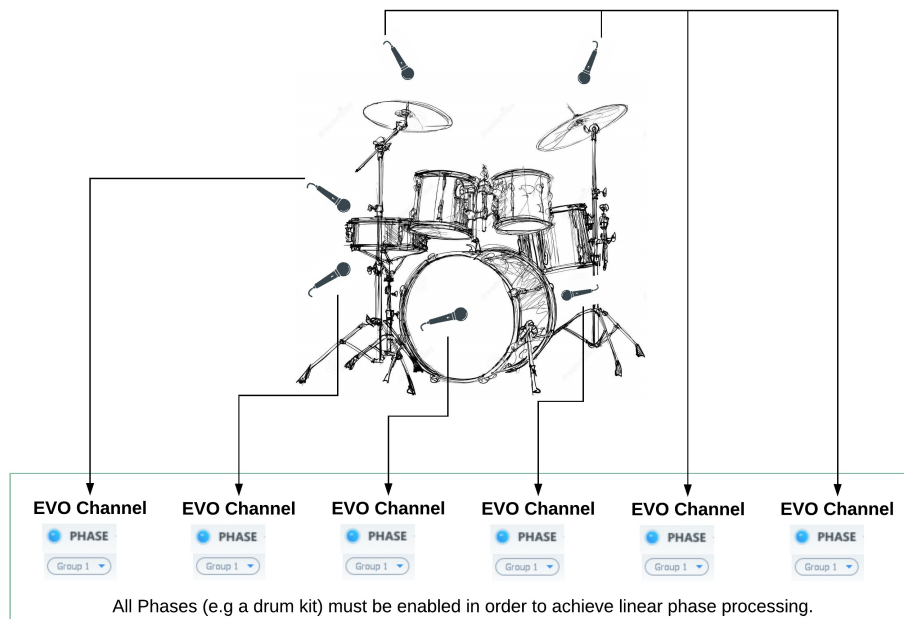
Value Range : 0% / 100%

Default Value : 0%

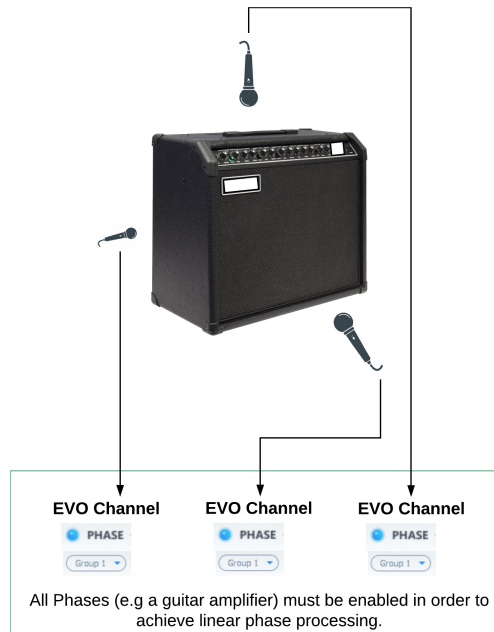
3.3 Phase

One of the most crucial parameters to take into account; Phase, or “The art of using multiple microphones on one and the same sound source”, a classic example is when setting up the sound for a drumkit.

Controlling the Phase of microphones is universal and relevant both for studio and live use. In EVO Channel an Arbitrary and Linear Phase Rotation throughout the spectrum is provided, to accomplish the same natural thing as when physically moving a microphone, of course without adding any latency.



The phase must be enabled for all the drum tracks. Then set the same group number for all these tracks. It allows to enable and disable the phase for all the drums in one click (simply switch on/off for the phase in one of the tracks).



3.3.1 Phase Switch

The Phase is the result of Flux:: proprietary research, this unique algorithm allows linear phase correction with zero latency.

Activate the phase module on all of the tracks that were recorded at the same time and in the same room (multi- microphones tracking or live recording in exemple), and then use the slider on the different tracks until you get full control over the phase and everything sounds correct.

When enabled, the phase correction is turned on.

Value Range : Enabled/Disabled

Default Value : Disabled

3.3.2 Phase Group

Add an EVO Channel instance to a Phase Group. You can add an instance of EVO Channel to a group, all the members of the same group can then have their Phase enabled or disabled at one and the same time. To do so, add the instance to a group by selecting the group number for the desired group. Now when you then turn the phase switch on or off in any of the instances in the group, this will affect all instances that are members of the same group.

Value Range : None/Group number

Number of groups : 8 groups

Default Value : Disabled

3.3.3 Phase Correction

Set the shift value used to correct the phase.

Value Range : -180° / 180°

Default Value : 0°

3.3.4 Polarity Invert

When enabled, polarity inversion is applied to the signal.

Value Range : Enabled / Disabled

Default Value : Disabled

3.4 Equalization module

One of the corner stones in a Channel Strip is a well-built, efficient and complete Equalizer.

The EVO Channel EQ is a straightforward comprehensive 4-band equalizer with additional 6-24 dB Hi/Lo filters.

The graphical interactive EQ curve is layered on top of the built in Spectrum Analyzer for hands-on editing with direct connection to the visual response.

Built with our proprietary State-Space technology, like in Epure, the EQ section has been carefully tuned to preserve the optimal signal to noise ratio independent of the parameter settings preventing the signal from deteriorating when the gain is reduced. It's an efficient sharp-edged surgical precision tool for the most demanding equalizing and filtering tasks conceivable.

Equalization module is divided in two parts : - A filtering part composed of a low cut, a low shelf, a high cut and a high shelf filter. - A parametric equalization stage made of four parametric EQs.



3.4.1 Low-Cut Switch

Enables the low cut filter.

Value Range : Enabled / Disabled

Default Value : Enabled

3.4.2 Low-Cut Cutoff Frequency

Cutoff frequency of the low cut filter.

Value Range : 20Hz to sampling rate / 2

Default Value : 20Hz

3.4.3 Low-Cut Slope

Slope of the low cut filter.

Value Range : 6dB/oct - 24dB/oct

Default Value : 6dB/oct

3.4.4 Low Shelf Switch

Enables the low shelf.

Value Range : Enabled / Disabled

Default Value : Enabled

3.4.5 Low Shelf Cutoff

Cutoff frequency of the low shelf.

Value Range : 20Hz / 22050Hz

Default Value : 100Hz

3.4.6 Low Shelf Gain

Target gain for frequencies below the cutoff frequency.

Value Range : -24.0dB / +24.0dB

Default Value : 0.0dB

3.4.7 High-Cut Switch

Enables the high cut filter.

Value Range : Enabled / Disabled

Default Value : Enabled

3.4.8 High-Cut Cutoff Frequency

Cutoff frequency of the high cut filter.

Value Range : 20Hz to sampling rate / 2

Default Value : Sampling rate / 2

3.4.9 High-Cut Slope

Slope of the high cut filter.

Value Range : 6dB/oct - 24dB/oct

Default Value : 6dB/oct

3.4.10 High Shelf Switch

Enables the high shelf.

Value Range : Enabled / Disabled

Default Value : Enabled

3.4.11 High Shelf Cutoff

Cutoff frequency of the high shelf.

Value Range : 20Hz to sampling rate / 2

Default Value : 5000Hz

3.4.12 High Shelf Gain

Target gain for frequencies above the cutoff frequency.

Value Range : -24.0dB / +24.0dB

Default Value : 0.0dB

3.4.13 Parametric Equalization Switches

Enables the corresponding parametric equalizer.

Value Range : Enabled / Disabled

Default Value : Enabled

3.4.14 Parametric Equalization Gain

Gain stage of parametric equalization.

Value Range : -24.0dB / +24.0dB

Default Value : 0.0dB

3.4.15 Parametric Equalization Center Frequency

Center frequency of parametric equalization.

Value Range : 20Hz to sampling rate / 2

Default Value : Default value depends of the equalizer used.

- Low parametric equalizer : 200Hz
- Mid-Low parametric equalizer : 500Hz
- Mid-High parametric equalizer : 1300Hz
- High parametric equalizer : 2500Hz

3.4.16 Parametric Equalization Q

Q value of parametric equalization. Defines the width of the EQ band.

Value Range : 1.0 Q / 100.0 Q

Default Value : 5.0 Q

3.4.17 Equalization Output Gain

A stage of gain at the output of the equalization module.

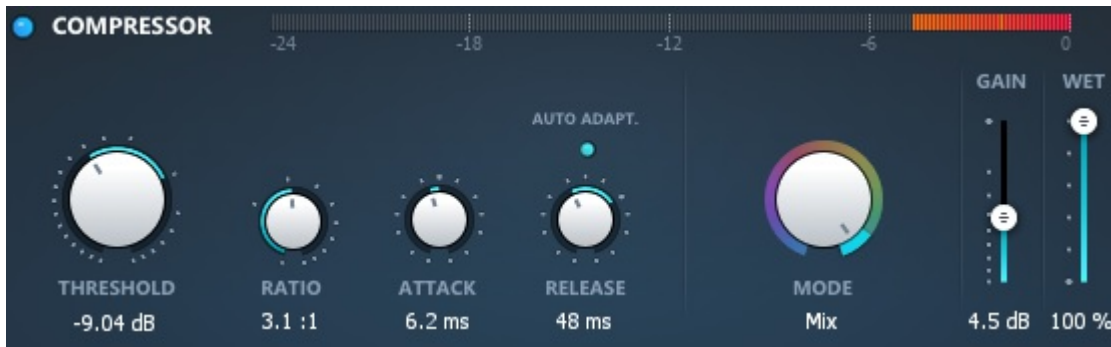
Value Range : -12.0dB / +12.0dB

Default Value : 0.0dB

3.5 Compressor

In addition to controlling the signal dynamics, the compressor is often used for shaping the attitude of a sound. To use a compressor in a creative and artistic fashion it's important that it's easy to use and has the ability to create an interesting sound.

The EVO Channel's compressor module is based on the Pure Compressor's dynamics engine, and the same range of compression types are available in EVO Channel through the different modes available (each mode corresponds to a fine tuning of Pure Compressor). As some modes use a LID compression (Level Independent Detection) in parallel, a gain reduction may be processed even if the audio level is below the threshold.



3.5.1 Mode

The compressor module gives you up to 9 modes of compression.

Available modes :

- Start
- Kick/Snare
- Overhead
- Drum Bus
- Bass
- Acoustic
- Piano
- Vocal
- Mix

Default Value : Start

3.5.2 Threshold

Threshold value of the compressor.

Value Range : -42.0dB / +18.0dB

Default Value : Depends on the Mode.

3.5.3 Ratio

Compression ratio parameter.

Value Range : 1.0:1 / 10.0:

Default Value : Depends on the Mode.

3.5.4 Attack

Attack value of the compressor.

Value Range : 0.1ms / 1000.0ms

Default Value : Depends on the Mode.

3.5.5 Release

Release value of the compressor.

Value Range : 1ms / 10000ms

Default Value : Depends on the Mode.

3.5.6 Auto Adapt.

When enabled, the compressor adapts its release time to the input signal depending on the audio signal, but won't exceed the release time value.

Value Range : Enabled / Disabled

Default Value : Enabled

3.5.7 Gain Reduction Display

Displays the gain reduction performed by the compressor.

Value Range : 0dB / -24dB

3.5.8 Compressor Output Gain

Gain stage at the output of the compressor module.

Value Range : 0.0dB / 24.0dB

Default Value : 0.0dB

3.5.9 Wet

Wet parameter defines how much of the compressed signal is mixed with the original signal, for parallel compression.

Value Range : 0% / 100%

Default Value : 100%

3.6 Touch

Different types of material require different tools, whether it's a vocal cut, drums, guitar, piano or something else, it requires the accurate specific treatment for the material.

The Touch module in EVO Channel is a polymorphic section offering a variety of different processors designed to adapt to the requirements of the material, with seven different processing modes including a DeEsser, Expander and a Transient and Sustain Designer.

With the adjustable signal flow in EVO Channel the Touch module can be inserted where it's best doing its job, like having the DeEsser or Expander first in the flow, or the Transient designer after the compressor.



3.6.1 Mode

Mode parameter defines the processor used by the touch module. Seven different modes are provided.

Available modes :

- Transient Boost
- Transient Kill
- Sustain Boost
- Sustain Kill
- DeEsser 1
- DeEsser 2
- Expander

Default Value : DeEsser 2

3.6.2 Amount

The amount of signal processed by the Touch module.

Value Range : 0% / 100%

Default Value : 0%

3.6.3 Release/Range

There is a release parameter for the following modes :

- Transient Boost
- Transient Kill
- Sustain Boost

- Sustain Kill
- Expander

Value Range : 1ms / 1000ms

Default Value : 20ms

The two deesser modes offers a range parameter, to limit the maximum gain reduction.

Value Range : 0dB / 24dB

Default Value : 12dB

3.6.4 Frequency Range

The Touch module works on a defined frequency range.

Value Range :

Minimal bound : 20Hz

Maximal bound : Sampling Rate / 2

Default Value :

Inferior bound : 3.9kHz

Superior bound : up to Sampling Rate / 2

Range width : Depends on the sampling rate.

3.6.5 Listen

When enabled, you can monitor the actual signal setup with the frequency range.

Value Range : Enabled / Disabled

Default Value : Disabled

3.7 Output

3.7.1 Output Gain

The output gain control trims the level of the signal at the output of EVO Channel. The meter shows both RMS signal (VU-Meter, blue) and peak signal (peak meter, green), from -24 to +18 dB range, referenced at -18dB.

Value Range : -24.0 dB / +18.0 dB

Colors : - Blue : RMS Value - Green : Peak Value

Default Value : 0.0 dB

4 Geek Settings

These settings are available by clicking on the “Yves Jaget” icon.

4.1 Compressor

4.1.1 Mode

Default Value: ClF Slow

8 different detection modes are available:

- Solera: The Attack setting also controls the integration time for RMS detection.
- Solera Feed Backward: The Attack setting also controls the integration time for RMS detection which is done on the output of the processor. Note also that the Solera Feed Backward prevents to use the external side chain because it's the processed signal which feed the side chain.
- Classic Fast: The integration time for RMS detection is 10 ms with no direct relation with the Attack setting.
- Classic Medium: The integration time for RMS detection is 40 ms with no direct relation with the Attack setting.
- Classic Slow: The integration time for RMS detection is 80 ms with no direct relation with the Attack setting.
- Classic Feed Backward Fast: The integration time is 10 ms for RMS detection which is done on the output of the processor. Note also that the Feed Backward mode prevents to use the external side chain because it's the processed signal which feed the side chain.
- Classic Feed Backward Medium: The integration time is 40 ms for RMS detection which is done on the output of the processor. Note also that the Feed Backward mode prevents to use the external side chain because it's the processed signal which feed the side chain.
- Classic Feed Backward Slow: The integration time is 80 ms for RMS detection which is done on the output of the processor. Note also that the Feed Backward mode prevents to use the external side chain because it's the processed signal which feed the side chain.

Note

These Feed Backward modes have been inspired by vintage hardware architectures. they create a sort of auto regulation of the processing which produces a naturally beefy sound. The output volume also control the feedback level.

4.1.2 Hold

Unit: ms

Value Range: 0 ms / 500 ms.

Default Value: 0 ms

This parameter is the only one in the time related settings, that is independent per dynamic processor. The compressor and the expander may have different hold time.

Note

Used in the Expander section, this setting allows very precise gating of drum tracks. It can also be used for creative purpose on the other dynamic sections.

4.1.3 Release Minimum

Unit: ms

Value Range: 0.67ms / 5000.00

Step: 0.01

Default Value: 1.30 ms

Sets the minimum release value when in Advanced Mode.

4.1.4 Dynamic Factor

Unit: x

Value Range: 0 / 3.0

Step: variable.

Default Value: 1

Amplify or dim the extracted real time dynamic informations.

4.1.5 Dynamic Velocity

Unit: %

Value Range: 10 / 1000

Step: 1

Default Value: 50 %

Sets the speed of variation on the dynamic informations.

4.1.6 SC Lo Cut

Unit: Hz

Value Range: 20 to 24000 Default Value: 20

Filters out the low-end from the detection circuit.

4.1.7 SC Hi Cut

Unit: Hz

Value Range: 20 to 24000 Default Value: 20000

Filters out the high-end from the detection circuit.

4.1.8 Knee (28)

Unit: dB

Value Range: 0 to +24

Default Value: 0

Sets the smoothness of the transmission curve for the specific dynamic processing section. The curve is smoothed around the threshold value of the dB amount set with the knee value.

4.1.9 Peak Detection Amount (29)

Unit: %

Value Range: 0 / 100

Step: 1

Default Value: 0 %

Instant peak value can be added to the RMS signal feeding the detector section, making the dynamic processing more sensitive to audio transients.

4.1.10 Dynamic Ratio (30)

Unit: %

Value Range: 0 / 100

Step: 1

Default Value: 0 %

This setting relaxes the ratio applied to the processor section when the detected signal dynamic raises.

This setting literally opens the sound, increases the dynamic impression and keeps some crest by adjusting in real time the ratio of every dynamic processing section regarding both their current settings about ratio and the signal content (mainly dynamic range). To start understanding this setting and easily hear it, take a full mixed drum kit or a complete mix with punchy drums, set the compression threshold, ratio to get something near pumping or an aggressive compression.

Then increase the output gain to compensate the gain lost and then toggle between 0 and 100% of Dynamic Ratio. At 100 % you should hear more air in the sound, more transient and less compression impression; especially in terms of attack.

4.1.11 L.I.D.. (Level Independent Detector) (32)

Unit: %

Value Range: 0 / 100

Step: 1

Default Value: 0 %

Allows process the audio signal independently of the sound level but regarding the signal dynamic range. It is mixed with the standard compression scheme.

Note

Take a piece of full mixed music, set the ratio to 3-4 and the compression will start working. Now set the threshold of the compressor to the maximum value, the compressor will stop working because the sound level will never reach the threshold. Then increase the L.I.D.. and you will see (and hear) the compression working again!!! Now decrease or increase the input gain (in Solera or before, as you want) and you will see that the compression will continue to work equally; it's totally, completely independent of the sound level and depends only on Ratio, Knee and sound content. How can this be used? When you have too much dynamic in the sound, going for e.g. from -3, -6 dB Vu (or less) to +12 dB; If you want to compress the low levels you will hear the sound TMpumping when the sound reaches the High levels and the only thing to do with standard compressor

will be to increase the threshold to rescue some airiness in the sound. But when doing that the compressor will not work any more on the low levels and you will hear some sound differences (in term density, live space, grain etc^o) especially when the compressor starts working. With Solera L.I.D., adjust the threshold and ratio on the High levels to what you think OK, then increase the L.I.D.. (from 20 to 50 %) and listen now the low levels and especially the transition between Low and High levels. You can also start increasing the ratio to increase the effect. You'll then notice that the compression will always be active but can still take care of High, loud levels (unless you set 100% L.I.D..) and make the compression very smooth and no more pumping^o In addition with the Dynamic Ratio function, you'll be able to set a constant and very natural envelop that allows to increase low levels, low frequency and to keep important transients.

4.1.12 L.I.D.. Threshold (33)

Sets the gain range of the L.I.D. parameter.

- Up: Increasing of the L.I.D. action
- Down: Decreasing of the L.I.D. action

The current L.I.D. Threshold value is reflected by two blue lines on the Dynamic Activity display.

For Compressor and DCompressor sections, the L.I.D. action is effective only when the orange Dynamic Activity (18) exceeds the area between the two blue lines. For Expander and DExpander sections, the L.I.D. action is effective only when the orange Dynamic Activity (18) stays inside the area between the two blue lines.

4.1.13 L.I.D.. Maximum (34)

When engaged, the threshold for the processing is determined by the maximum values from RMS/peak detection OR from the signal dynamic detection. The L.I.D. Threshold is still active, but the L.I.D. mix button is disabled.

Note

This feature allows the whole process to be more reactive to the signal content. It worth to be tried on drum tracks.

4.2 Touch

4.2.1 Slope

The Slope sets the number of frequency-dependent components controlling the slope of the frequency response. One component, referred to as 1 (First-Order), gives a 6 dB/Octave slope, adding a second component, referred to as 2 (Second-Order), gives a 12 dB/Octave slope etc. The higher the Frequency Order is, the steeper the slope of the frequency response becomes.

Value Range: 6 / 12 / 18 / 24 - dB Per Octave

Default Value: 6 dB/o

4.2.2 Release

Adjusts the release time for the transient processing envelope.

Value Range: 0.00 ms / 500.00 ms

Default Value: 0.00 ms

4.2.3 Bandwidth

Value Range: Static BW/Dynamic BW/Flat Sum

Default Value: Static BW

Static bandwidth provides a constant Q factor, no matter of the gain factor of the filter. It's computed by having a constant bandwidth at +- 3 dB from 0 dB.

Dynamic bandwidth provides a dynamic Q factor, dependent on the gain factor of the filter. It's computed by having a constant bandwidth at +- 3 dB from peak levels of the filter.

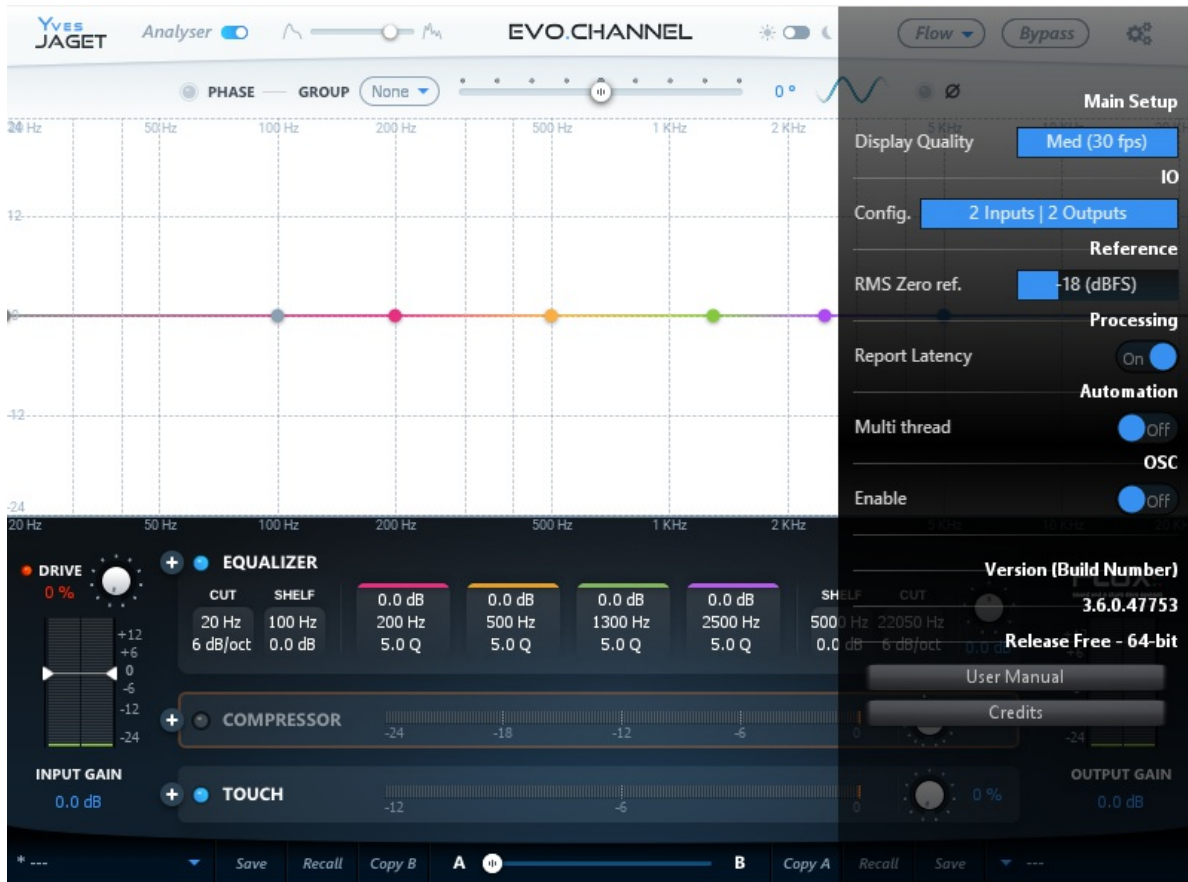
Flat Sum is a more traditional way of computing the Q factor. The bandwidth is constant for the gain factor divided by two.

4.2.4 Zero Crossing Threshold

The Zero Crossing refers to the amount of time the signal crosses the point of an amplitude of zero. It gives an indication on transient detection, as they tend to produce bursts of zero crossing.

5 Plugin Settings

Clicking the cogwheel symbol opens a window with a range of general settings and a direct access button to the user manual.



5.1 Main Setup

5.1.1 UI Refresh Rate

Max refresh rate of the plug-in's UI.

5.2 I/O

5.2.1 Input / Output

I/O Config and Layout is not always available, though it is always displayed, it can only be edited in some configurations and formats.

5.2.2 Config

Current I/O configuration, is only available in certain VST hosts; typically hosts with limited capabilities for handling multichannel configurations.

5.2.3 Layout

Available I/O routings based on current I/O configuration. Layout is available for editing if more than two input channels are available. If the Layout is changed from the default value, an asterisk * is displayed next to the Layout information in the Input section.

5.3 Processing

5.3.1 Report Latency

Enables/Disables the latency reporting to the host.

5.4 Automation

5.4.1 Multithread

Enables/Disables Multithread Automation.

5.5 OSC

OSC is available in EVO Channel.

5.5.1 Enable

Enables/Disables OSC control and mapping of the plug-in's parameters.

5.6 Version Information

Plug-in version and build-number information.

5.7 User Manual / Credits

Quick link to the User Manual. Plug-in creation credits.

6 Shortcuts

Shortcuts have been added to further enhance the user interaction and improve the workflow.

Where	Shortcut	Description
EQ Display	Mouse Click + Alt	Reset to default value
EQ Display	Mouse Move + Ctrl	Q change only
EQ Display	Mouse Move + Shift	Gain change only
EQ Display	Mouse Move + Ctrl + Shift	Frequency change only
EQ Display	Wheel + Ctrl	Q change only
EQ Display	Wheel + Shift	Gain change only
Compressor Mode	Mouse Click + Alt + Shift	Reset all compressor parameters to the default value of the Mode

7 Preset Management

EVO Channel, as well as all other Flux:: plug-ins, provides two preset slots referred to as slot A and slot B, which provide access to two sets of parameter settings simultaneously. In addition to just recall the settings for each of the slots individually and alternate between their settings, a morphing slider is provided offering the possibility to morph between the slots and their corresponding settings. When clicking on one of the preset slots, the built in preset manager appears.



7.1 Preset Sections

EVO Channel provides two preset sections referred to as section A and section B, offering simultaneous access to two full sets of parameter settings. Clicking the A section (bottom left) or the B section (bottom right), or clicking the arrow in the Current Selected Preset display, opens a new window accessing the built-in preset manager.

7.2 Save

Save replaces the selected preset by a new one under the same name featuring the current settings. If you want to keep an existing preset without your new modifications, just select an empty place into the preset list, enter a new name for this modified preset featuring the current settings and press Save. Recall

Once a preset is selected from the preset list it must be explicitly loaded into section A or the section B by using the recall button. A preset is effective only after it has been recalled.

7.3 Copy A / Copy B

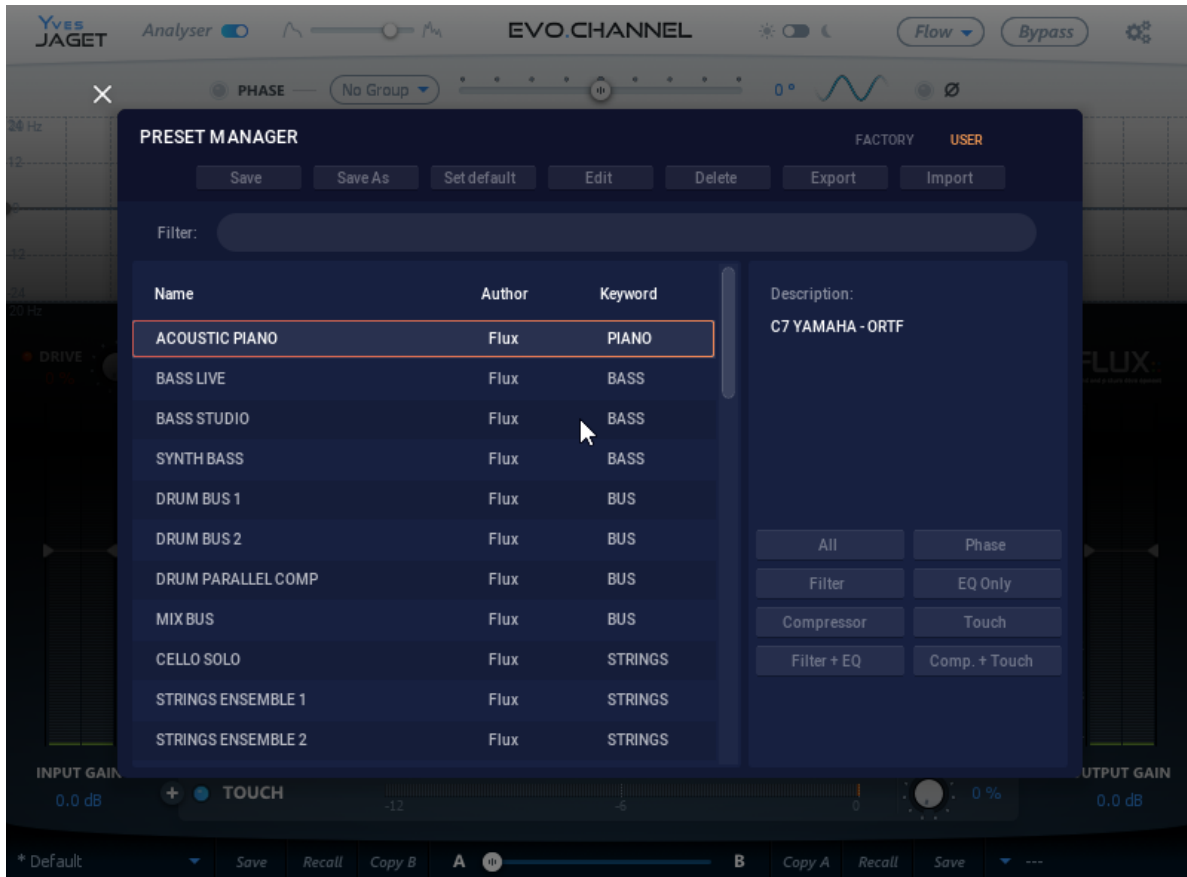
The current parameters of a section are copied to the other one. The section A or B is re-initialized with the current values and the morphing slider is parked at 100% of the corresponding section.

7.4 Morphing Slider

Morphs the parameter values of both parameter sections, it has no unity or specific value display; it provides morphing of the current values from both of the parameter sections (A & B). A double-click on one side of the slider area toggles between the two parameter sections. The actual result of the morphed parameter settings can be saved as a new preset.

8 Preset Manager

The preset manager contains two preset banks, the Factory bank contains factory presets, this bank is not available for saving of presets but any of the presets can be loaded into a preset slot and then saved into, the User bank, where all user presets are saved.

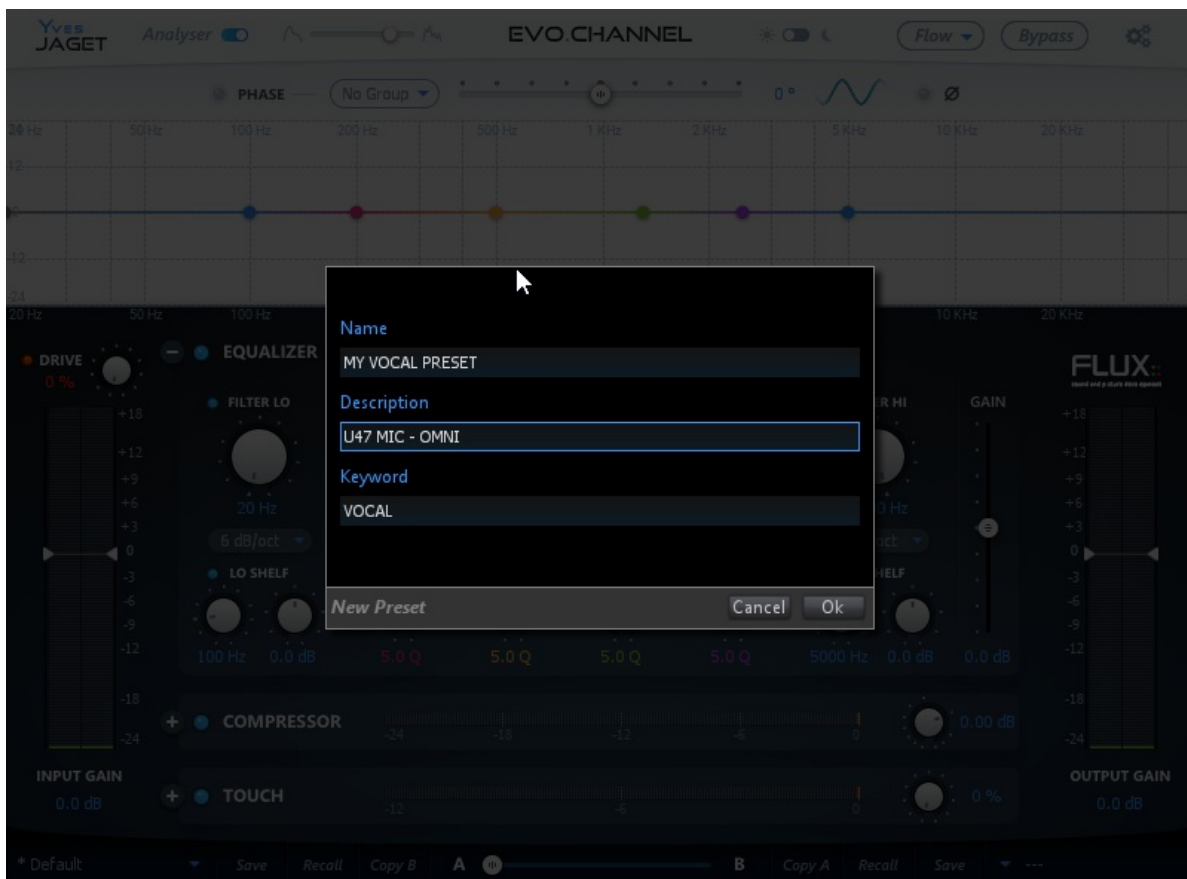


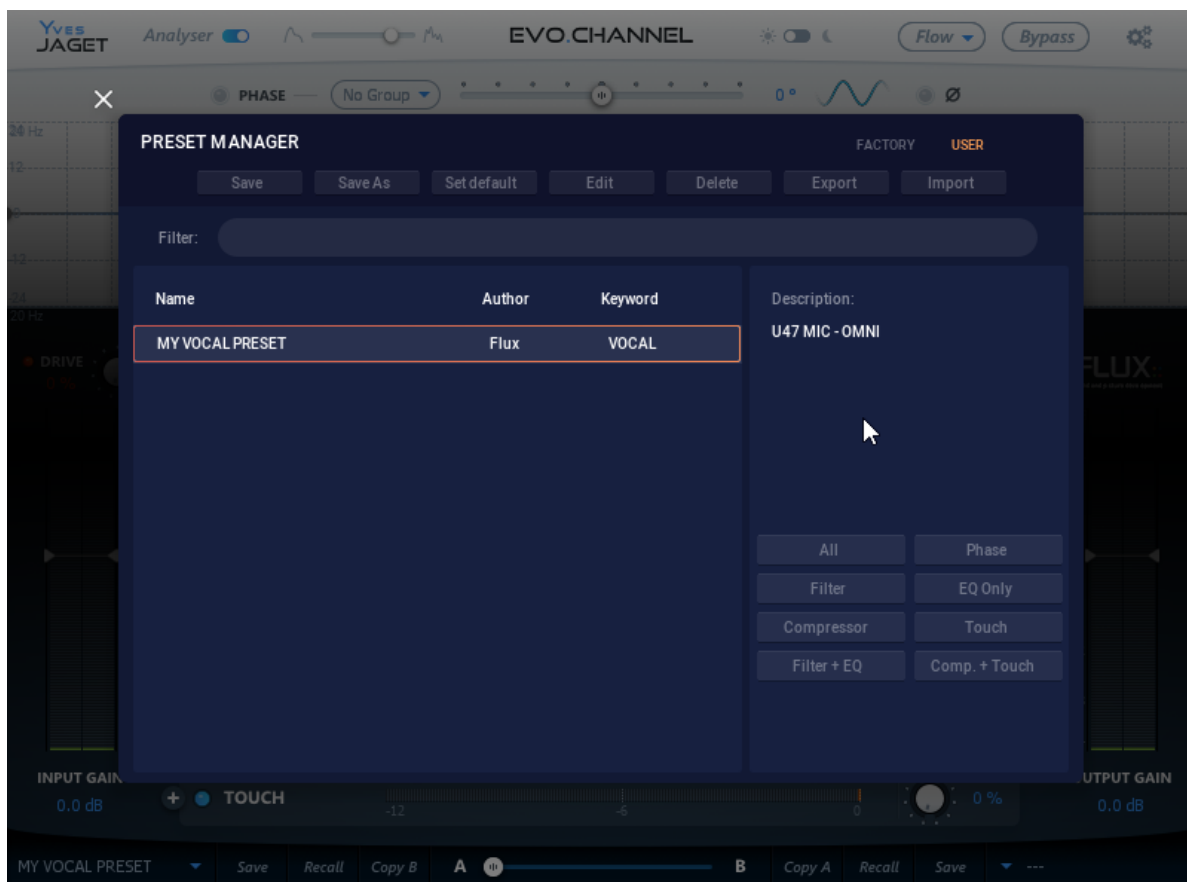
In the preset manager, any preset can be loaded into a preset slot by double clicking on the name of the desired preset in the actual preset list, the preset will then be loaded into the preset slot corresponding to the position of the morphing slider.

- Additional controls in the preset manager
- Recall A loads the selected preset into the corresponding slot.
- Recall B loads the selected preset into the corresponding slot.

- Update, saves the current settings into the selected preset. (Available in User Bank only)
- New, saves the current settings into a new preset. (Available in User Bank only)
- Duplicate creates a copy of the selected preset and saves it to the list.
- Edit allows for changes to the preset meta properties. (Available in User Bank only)
- Delete, removes the selected preset. (Available in User Bank only)
- Export, creates a file reflecting the content of the current preset bank.
- Import, allows for import of a preset bank file by adding the imported banks content to the content in the current preset bank.

When saving or editing a preset, an option to protect the preset is presented. The preset protection, if engaged, only allows the original preset author to uncheck and edit the preset. This means that you can protect your presets in a multi-user configuration. Protected presets can only be modified using the session used for their creation. If used in another user session they can only be imported or deleted.





9 Specifications

9.1 Availability

EVO Channel is available in:

AU / VST / VST3 / AAX Native/ *AAX AudioSuite* / Waves WPAPI

** AAX Native & AAX AudioSuite in Pro Tools 11 and later*

9.2 Processing

EVO Channel provides :

- Up to 16 channels Input/Output in AAX/AU/VST/VST3.
- Up to 8 channels in WPAPI for Waves Soundgrid.
- 64-bits internal floating point processing.
- Sampling rate up to 384 kHz.

9.3 Hardware Requirements

A graphic card fully supporting OpenGL 2.0 is required.

- macOS : OpenGL 2.0 required – Mac Pro 1.1 & Mac Pro 2.1 are not supported.
- Windows : If your computer has an ATi or NVidia graphics card, please assure the latest graphic drivers from the ATi or NVidia website are installed.

9.4 Software License Requirements

In order to use the software an iLok.com user account is required (the iLok USB Smart Key is not required).

9.5 Compatibility

All major native formats are supported

9.5.1 Windows – 10, in 64 bits only.

- VST (2.4)
- VST3 (3.1)
- AAX Native*
- AAX AudioSuite*
- Waves WPAPI

9.5.2 macOS (Intel and ARM)

All versions from Sierra (10.12) to latest. (Compatible with previous versions but not supported)

- VST (2.4)
- VST3 (3.1)
- AU
- AAX Native*
- AAX AudioSuite*
- Waves WPAPI

* *AAX Native & AAX AudioSuite in Pro Tools 11 and later*

A Release Notes

A.1 Build 23.1.0.50251 - All plugins

A.1.1 New features

- New plugins Evo Compressor, Evo Touch and Evo EQ.
- VST3 support
- ARM support for AAX, AU and VST3
- Plugins are now resizable
- Elixir now supports 32 channels
- Alchemist, BitterSweet, Epure, Pure Compressor, Pure DCompressor, Pure Expander, Pure DExpander, PureLimiter, Solera, Syrah now support 16 channels

A.1.2 Bugs fixes

- All plugins - Preset Manager - Update user preset do not work
- All plugins - Preset manager - Crash or freeze when saving a preset
- All plugins - UI may be black on Intel UHD 630 graphical cards
- All plugins - AU/VST3 - Preset manager - Default preset is not applied to parameters at plugin instantiation
- All plugins - AAX - Crash with OSC when changing fx slot in Pro Tools
- All plugins - AU - Logic Pro - Automation of boolean/integer parameters broken
- All plugins - AU - Plugins crash in Da Vinci Resolve
- All plugins - DaVinci Resolve - VST - UI is truncated
- All plugins - Streamlabs - Plugins do not work
- All plugins - Licensing issue in DaVinci Resolve and GarageBand
- Alchemist - The range parameter works only for the 1st band
- BitterSweet - Not possible to tweak the Output gain after unlinking it

- BitterSweet - Output gain not reloaded properly when the link is disabled
- BSPro - some modes are not accessible due to GUI issue
- Epure - macOS - Bad graphic scale initialization at 2&4FS
- Evo Channel - Meter reference is not saved
- Syrah - Crash when selecting preset “Static fast compression”
- TRAX Tr - When the link is activated, the Formant slider does not have the expected audio effect
- TRAX Tr - ProTools - Issue in AudioStudio when the modulation is enabled
- VerbSession/VerbSession Studio Session and BSPro StudioSession - Pyramix - VST crash when instantiated
- Verb/Verb Studio Session - Crash when reloading session having 2 instances

A.1.3 Known issues

- All plugins - VST - GUI issue in Izotope Ozone and RX
- All plugins - AAX - Preset manager - Default preset is not applied to parameters at plugin instantiation
- Elixir - Latency not properly compensated after changing stage parameters value in VST and AudioUnit
- TRAX tr - Learn function returning wrong values
- VerbV3 - HOA 3rd order not working properly

A.2 Build 21.12.0.50123 - All plugins except TRAX and StudioSession

Bug fixes

- All plugins AudioUnit - GUI issue with Hdpi displays on macOS Monterey
- All plugins VST - Plugin scan freeze in Wavelab 11 on Mac M1 machines
- All plugins VST - Crash in Adobe Audition on macOS
- All plugins VST macOS - Fix crashes with Ableton live
- Elixir - Automation is not read for toggle parameters.
- Elixir - Crash when clicking on the settings button on Session version

- Elixir - Several fixes on the UI
- Elixir - Windows AAX - Refresh issue with two instances in ProTools
- HEar - Bypass is working in AAX
- HEar AAX - Crash when doing offline bounce on macOS
- HEar AAX - Crash when editing the matrix on macOS
- HEar AAX - Stereo - Change on Matrix are not applied until we change the preset
- HEar AudioUnit - Ableton crashes when inserting a second instance

A.3 Build 21.11.0.50107 (HEar, IRCAM Verb)

NOTE: CURRENTLY NOT COMPLIANT WITH ABLETON LIVE MACOS

Improvement

- HEar - 5.1.4 & 5.0.4 now available

Bug fixes

- HEar - Fix meters refresh issue
- HEar - No verb on some presets
- HEar - Protools crashes when doing offline bounce on macOS

A.4 FLUX:: Immersive - Plugins (including IRCAM Tools) 21.09

This release includes updates for all FLUX::Immersive plugin processing products with the exception of EVO Channel, Epure, IRCAM Trax, Studio Session.

NOTE: CURRENTLY NOT COMPLIANT WITH ABLETON LIVE MACOS

Major optimizations

- Apple computers Big Sur (new M1 chips) AU validation
- Important updates to the Ircam Verb + Session
- Overall better handling of multichannel track setups such for Atmos. (Ircam Hear, Verb and more)
- Automatic detection of track format / channel order for DAWs when possible.

A.4.1 Build 21.9.0.50083

Bug fixes

- Apple computers Big Sur (new M1 chips) AU validation failing
- Empty GUI when close/reopen plugin - Windows 10 - UHD630 graphics
- AudioUnit in Reaper - do not process audio when offline bounce
- Default preset not loaded correctly on instantiation of Verb + Verb Session
- Evo.Channel on Retina - Input and Output Sliders badly scaled
- Incompatible AudioUnit issue in Apple Final Cut Pro
- Plugins: Recall Preset Flags (e.g. “All but setup”) recall always everything
- Preset Manager - UI issue with small plugins when a preset has been created
- Ircam Verb Session reload in VST with audio interruption
- VST Plugins Session not correctly reloaded if it integrate an IO configuration change
- Verb session - Dry/wet not applied in offline render
- Verb v3 Atmos crash on AAX
- Verb: AU validation failed on Apple M1
- Verb: LFE not disabled by default on ProTools
- Verb: Recall Preset may be not correct with double click inside the preset manager
- Verb: disabled channel is not re-injected according to dry/wet parameter (100 % wet means muted)
- Verb: init issue with Nuendo
- AAX - Some plugins - Crash on Mac / No GUI on Windows
- Overall reliability / stability fixes.
- Plugin size not correct
- Potential plugins crash when opening UI

A.5 FLUX:: Immersive - Plugins (including IRCAM Tools) 20.12

This major release includes updates for all **FLUX::Immersive** products with the exception of IRCAM Spat V3 legacy product. Please refer to Spat V3 - Spat Revolution crossgrade options.

Major optimizations

- HiDPI / Retina support + display enhancements and fixes
- Page Table unification for **Avid Control**, S1, S3, S4, S6 and S6L.
- OSC Control for plugins.
- **IRCAM Verb** support for Dolby Atmos, Multichannel support up to 16 channels
- **IRCAM Hear** - Multichannel stability improvement, Now up to 10 channels. (Dolby Atmos 7.1.2)
- **IRCAM Tools** - Audio I/O Matrix and Multichannel enhancement

- Most plugins support of 8 channel.
- 16 channel support for **Bittersweet Pro, Evo In and Evo Channel**

A.5.1 Build 20.12.0.49880

Bug fixes

Core:

- BPro - Latency report issue (AAX)
- IRCAM TRAX Tr - Latency report issue
- IRCAM Verb - Wrong initialization value for Reverb density
- IRCAM Verb -Dry signal still goes out in disabled channels when wet is 100%
- All Pure Dynamics PI + Alchemist - Wrong Thresholds initialization values
- AAX “monolithic” are broken like Hear, TRAX etc...
- Almost all AAX plugins don’t reload parameters from 47856 version session.
- Pure Limiter - Diff feature bypassed the input gain.
- Pure Limiter - Inverted sidechain filters.
- Any plugin except Evo Channel - Research Presets resets when click on a preset.
- Evo channel - Wrong values when reloading touch section.

UI:

- Current preset name disappear on re-opening GUI or session

A.6 Known Issues

- Wavelab “Sample rate not supported” when a plugin is inserted on a clip, track or output section.
- TRAX Tr - Learn frequencies display wrong values (AAX only).
- Hear - Internal config labels change when modify LFE input config from routing matrix.
- When using OSC on a plugin in Pro Tools, a crash will occur if you change/move FX insert slots