Pure DCompressor

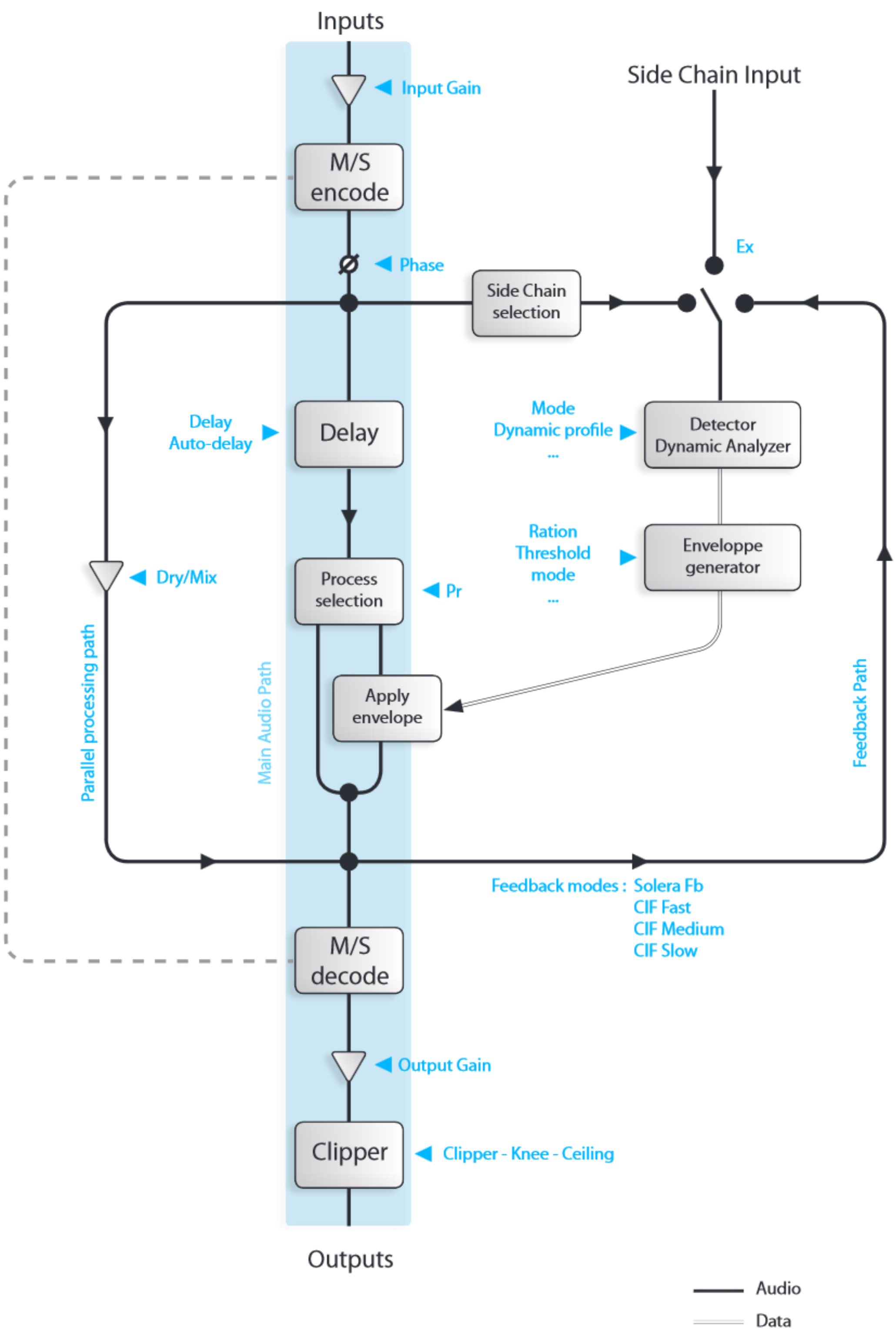
FLUX:: Immersive

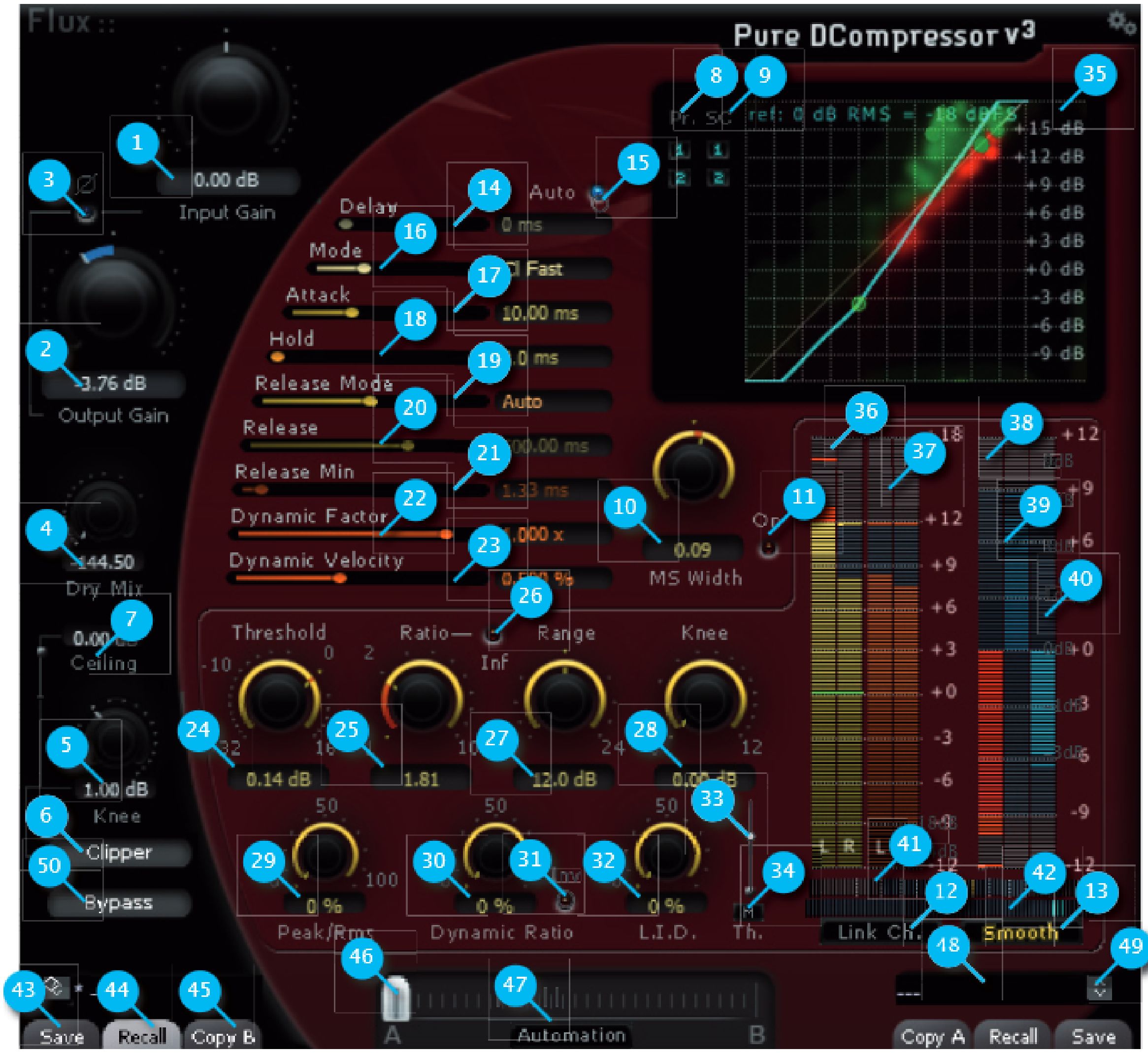
2/6/23

Table of contents

1 Pure DCompressor

Product Page | Shop Page


Inputs
Input Gain
Side Chain Input
M/S
encode
Ex
Phase
Side Chain
selection
Delay
Auto-delay
Delay
Mode
Dynamic profile
...
Detector
Dynamic Analyzer
Ration
Threshold
mode
...
Enveloppe
generator
Dry/Mix
Process
selection
Pr
Apply
envelope
Parallel processing path
Main Audio Path
Feedback Path
Feedback modes : Solera Fb
CIF Fast
CIF Medium
CIF Slow
M/S
decode
Output Gain
Clipper
Clipper - Knee - Ceiling
Outputs
Audio
Data



Pure DCompressor is the de-compressor section of the Solera. A De-compressor must be used to automatically increase the peak dynamic. Pure DCompressor allows you to restore the original dynamic of a sound. It's very useful for heavily compressed signal. Pure DCompressor adds some naturalness.

The Threshold value is expressed in dB – The plug-in compares it to the RMS (Root Mean Square or effective power) of the input signal. This value is displayed as a green rectangle on the input level meter. The level variations above the threshold are affected by the Ratio value. For a 1:1 de-compressor ratio, the processed signal isn't affected by the processing: A 1 dB variation above the threshold at the input is reflected by a 1 dB variation at the output. Let's apply now a 3:1 ratio. If the input signal rises above 1 dB the threshold value, the output signal rises from 3 dB: Here is the de-compressor action. The input signal gain is increased by a 3:1 ratio above the threshold point.

The Knee sets how progressive is the start up of the compressor action – In other words, It smoothes the transition point between no processing and full processing. If the Knee value expressed in dB is increased, the progressiveness of the action will be spread below and above the threshold point. The Range value sets the processing maximum action. No gain variation can exceed this value.

2 General Settings

Unit: dB
Value Range: -48 / +48
Step: 0.01
Default Value: 0 dB

Sets the gain applied to the dynamic processing input.

2.1 Output Gain (2)

Unit: dB
Value Range: -48 / +48
Step: 0.01
Default Value: 0 dB

Sets the global gain applied to the dynamic processing output.

2.2 Invert Phase (3)

Default Value: Off

When this button is engaged, the phase of the processed signal is inverted.

This 180° phase shift is also applied on the Solo of the detector Equalizer.

2.3 Dry Mix (4)

Default Value: -144 dB

This slider controls the amount of the original signal that can be added to the processed audio. This feature is dedicated to mastering works requiring both heavy processing and subtle control. The mix is done before the output gain.

2.4 Clipper Knee (5)

Unit: dB
Value Range: 0 / +3
Step: 0.01
Default Value: 1 dB

Sets the smoothness of the transmission curve.

2.5 Enable Clipper (6)

The Clipper is the very last stage of the processing chain.

2.6 Clipper Celling (7)

The Clipper is the very last stage of the processing chain.

2.7 Bypass (50)

It's a global bypass.

2.8 Channel Processing Selector (8)

When operating on a multi-channel bus, all channels are processed by default, but it can be useful to remove some channels from the processing for some reasons. This selector allows to keep the unchecked channels untouched.

This feature may be used if different settings are required. Several instances of a plug-in can be used in series, each one processing a particular channel with its own settings.

2.9 Channel Side Chain Routing (9)

Side chain is only available for mono and stereo.

When operating on a multi-channel bus, all channels are feeding the side chain by default, but it can be useful to prevent some channels feeding the side chain for some reasons. The external side chain button is the one that appear with the channel count + 1 label (for mono, label is “2”, for stereo label is “3”).

Here are the different behaviors for external side chain depending of the host applications:

- Audio Unit : The side chain button is displayed when the side chain is connected by the host.
- AAX : The side chain button is only active if a side chain buss is selected from the plug-in handler.
- VST : No side chain feature is available.

2.10 MS Width Control (10)

Unit: dB
Value Range: -6 / +6
Step: 0.01
Default Value: 0

Sets the stereo width of the processed signal. A -6 dB value deceases the stereo width. A +6 dB value increases the wideness of the stereo mix but can produce phase issue.

2.11 MS Mode On/Off (11)

Default Value: Off

Enables one MS encoding matrix at the input and one MS decoding matrix at the output of the dynamic processing in order to control the stereo width of the mix. When engaged, the side chain is fed by a MS encoded signal that is reflected in the display section. M channel corresponds to the normal left channel. And the S channel corresponds to the normal right channel This feature is only available when two channels (no more, no less) are processed.

2.12 Link (12)

Default: Enabled

By default the maximum value issued from all channels feeding the side chain is retained as source for processing. This manner, the space information is kept for the processed multi-channel signals.

When disabled, every channel uses its own value for individual processing.

This configuration may be used in conjunction with the MS width section which encode the signal in MS before processing, and decode at the output. This manner, the M signal can be processed while keeping the S channel untouched.

2.13 Smooth (13)

Default: Enabled

When engaged, the side chain is set to preserve bass frequencies from excessive processing.

3 Time Related Settings

3.1 Delay (14)

Unit: ms
Value Range: 0 to 50.0 ms
Default Value: 0 ms

A delay reflecting the attack time can be introduced into the signal path in order to produce a zero attack time for the dynamic processing. Shifting the delay value from the attack time allows to control transients. A delay value inferior to the attack value lets peaks untouched by the processing.

> Note that the different delay values of every band are automatically compensated. Solera can't be used to produce delay based special effects.

!> Warning: Morphing between presets with different delay values produces sound artefacts. Of course this delay introduces latency in the processing.

3.2 Auto Delay (15)

Default Value: Off

When enabled, the delay value is linked to the attack value. Be aware that the latency introduced by this function is now equal to your attack time devide by 2.

3.3 Mode (16)

Default Value: Solera

8 different detection modes are available:

- Solera: The Attack setting also controls the integration time for RMS detection. When "Auto" is engaged for the delay value, the produced attack time is zero.

- Solera Feed Backward: The Attack setting also controls the integration time for RMS detection which is done on the output of the processor. This mode disables the Delay feature. 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. But when "Auto" is engaged for the delay value, the produced attack time is zero.
- Classic Medium: The integration time for RMS detection is 40 ms with no direct relation with the Attack setting. But when "Auto" is engaged for the delay value, the produced attack time is zero.
- Classic Slow: The integration time for RMS detection is 80 ms with no direct relation with the Attack setting. But when "Auto" is engaged for the delay value, the produced attack time is zero.
- Classic Feed Backward Fast: The integration time is 10 ms for RMS detection which is done on the output of the processor. This mode disables the Delay feature. 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.

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.

3.4 Attack (17)

Unit: ms
Value Range: 0 ms to 100 ms
Default Value: 0.0 ms

Sets the attack time of the processing envelop. It also controls the manner the RMS value is computed from the incoming signal.

!> Warning : The Attack setting also controls the integration time for RMS detection.

3.5 Hold (18)

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.

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.

3.6 Release Mode (19)

Default Value: Auto

Three release modes are available for the envelop of the dynamic processing.

- Manual corresponds to the value you have set.
- Auto enables our specific algorithm to generate a signal dependent value to avoid typical pumping effects.
- Advanced gives access to two different values for release and to the control of the velocity of the variations between the maximum and the minimum release values.

3.7 Release (20)

Unit: ms
Value Range: 0.67 ms / 10000.00 ms
Default Value: 500.00 ms

Sets the manual release value and the maximum release value when in Advanced Mode.

3.8 Release Minimum (21)

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.

3.9 Dynamic Factor (22)

Unit: x
Value Range: 0 / 3.0
Step: variable.
Default Value: 1

Amplify or dim the extracted real time dynamic informations.

3.10 Dynamic Velocity (23)

Unit: %
Value Range: 10 / 1000
Step: 1
Default Value: 50 %

Sets the speed of variation on the dynamic informations.

4 Dynamic Section Settings

4.1 Threshold (24)

Unit: dB
Value Range: -32 to +16
Default Value: 0

Sets the threshold of the specific dynamic processing section. This dB scale refers to an RMS value.

The threshold effective value is modified by the L.I.D., the L.I.D. Threshold and, the L.I.D. Maximum settings.

4.2 Ratio (25)

Unit: dB
Value Range: 1 to 10
Step: 0.01
Default Value: 1

Sets the ratio of the specific dynamic processing section.
The ratio effective value is modified by the Dynamic Ratio amount.

4.3 Infinite (26)

Sets the ratio to its maximum.

4.4 Range (27)

Unit: dB
Value Range: 0 to 48
Default Value: 24

Sets the maximum allowed gain variation for a specific dynamic processing section.

4.5 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.

4.6 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.

4.7 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.

4.8 Dynamic Ratio Inverter (31)

When engaged, the behavior of the Dynamic Ratio is inverted. The ratio value is increased depending of the detected signal dynamic.

4.9 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.

> *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 "pumping" 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...) 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... 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.

4.10 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.

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

5 Display

5.1 Resulting Transfer Curve (35)

Auto Scale depending of the threshold value(s)

5.2 Input Level Meter (36)

Vu-meter not peak-meter, referenced to -16 dB Fs by default, with auto scale depending of the threshold values. When the MS Width section is engaged, the M (Middle) level is displayed on the left meter. S (Side) is displayed on the right meter.

The green index reflects the threshold value.

5.3 Output Level Meter (37)

Vu-meter not peak-meter, referenced to -16 dB Fs by default, with auto scale depending of the threshold values. When the MS Width section is engaged, the M (Middle) level is displayed on the left meter. S (Side) is displayed on the right meter.

5.4 Resultant Envelop (38)

Vu-meter not peak-meter, referenced to -16 dB Fs by default. The scale is +/- 12 dB.
This is the compression, decompression, expander and de-expander summing envelop.

5.5 Dynamic difference between in and out (39)

Vu-meter not peak-meter, referenced to -16 dB Fs by default. The scale is +/- 12 dB.

5.6 Level difference between in and out (40)

Vu-meter not peak-meter, referenced to -16 dB Fs by default. The scale is +/- 12 dB.
This is the compression, decompression, expander and de-expander summing envelop which also takes account of the input and output gains of the band.

5.7 Dynamic Activity Display (41)

No scale
The current L.I.D. Threshold value is reflected by two blue lines on the Dynamic Activity display.
The L.I.D. action is effective only when the orange Dynamic Activity (41) exceeds the area between the two blue lines.

5.8 Instant Release Value (42)

Auto Scale depending of the release value(s)

6 Preset Management

6.1 Save (43)

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.

6.2 Recall (44)

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.

6.3 Copy A / Copy B (45)

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.

6.4 Morphing Slider (46)

This horizontal slider has no unity nor specific value display. It allows to morph current settings between two loaded presets. Double-click on one side of the slider area toggles between full A and full B settings.

The results of an in-between setting can be saved as a new preset.

A global preset including the two loaded presets and the morphing slider position can also be saved from preset management window.

6.5 Automation Control of the Morphing Slider (47)

Default Value: Off

When this button is disabled, all the plug-in parameters values are recorded when writing automation. The morphing slider is ignored.

When reading automation, if this button is disabled, all the plug-in parameters are controlled by the host automation except the morphing slider.

When this button is engaged, all parameters are recorded when writing automation uncluding the morphing slider.

When this button is engaged, ONLY the morphing slider value is applied when reading automation.

The Automation button must be engaged if the morphing slider has to be mapped on a control surface.

6.6 Loaded Preset Display (48)

6.7 Preset Manager Access (49)

7 Specifications

7.1 Processing Specifications - Pure DCompressor

- Up to 16 channels Input/Output.
- 64-bits internal floating point processing.
- Sampling rate up to 384 kHz DXD (Pyramix and Ovation MassCore/Native).
- Sampling rate up to 192 kHz for Native (AU/VST/ST3/AAX/AAX AudioSuite).

7.2 Processing Specifications - Pure DCompressor Session

- Mono/Stereo Input/Output.
- 64-bits internal floating point processing.
- Sampling rate up to 96 kHz.

7.3 Licence Requirements

In order to use Pure DCompressor or Pure DCompressor Studio Session, one of the following is required: - An iLok.com user account (the iLok USB Smart Key is not required). - A Flux:: USB Dongle (Available in our online store).