MELDAproduction

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GENERAL MELDAPRODUCTION SOFTWARE INFORMATION

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1. Run Update manager from start menu, using "update.cmd" file in the installation directory or using setup.exe in the installation directory. It will locate any necessary updates or packages and install them for you. Requires internet connection. In some cases, such as major updates, this method may not be available.

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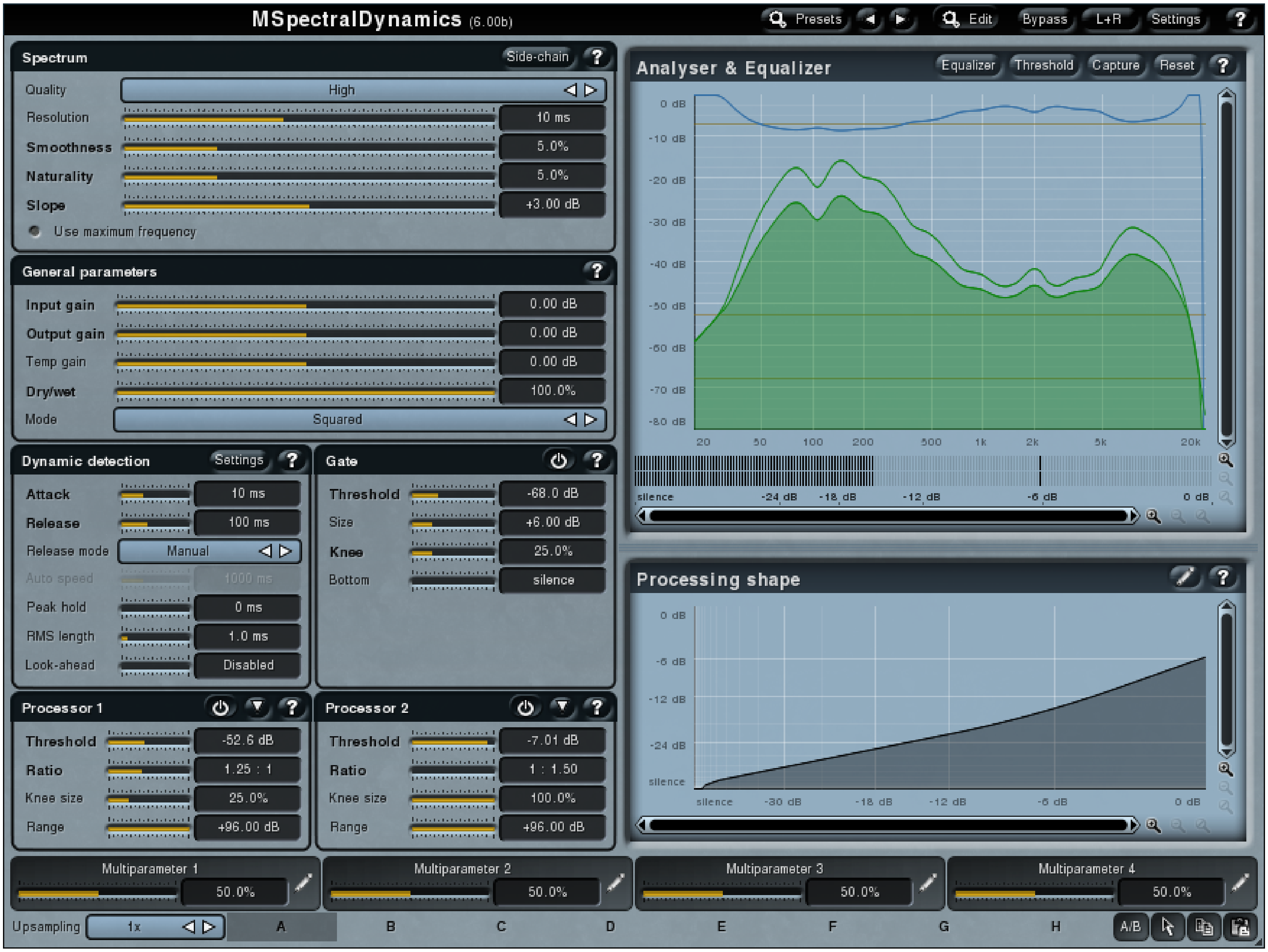
MELDAPRODUCTION MSPECTRALDYNAMICS



OVERVIEW

MSpectralDynamics presents a true mastering revolution. It can do many things, but mostly it serves as a modern hi-tech replacement of multiband compressors and loudness maximizers. However with top-class features such as custom processing shape it can provide very wide range of effects. Additionaly it provides a free-form linear phase equalizer with range from -80dB to 0dB, which you can use to fix particular problems in your recording.

Multiband compressors became a very important tool required to balance the spectrum and maximize loudness. However multiband compressors are very clumsy, their effects are often too unnatural and it is incredibly simple to completely destroy the audio material.

MSpectralDynamics on the other hand does not have any bands and works with the entire spectrum instead. It approximates energy located in each frequency and surroundings and applies the dynamics to it separately. This algorithm is extremely complicated but provides state-of-the-art sound quality and features you could not even dream of before!

You can perform spectral compression to balance the spectral energy and increase loudness with minimal amount of artifacts. You can apply spectral expansion to excite frequencies and increase sound clarity. And this is just a beginning...

INTRODUCTION

Processor 1 (panel on the left in the middle) is enabled by default and performs a little compression. On the Analyser view you can see the threshold of the processor displayed using an orange line. The power spectrum of your recording is displayed using green line and area. You can drag the orange line, the threshold. When you move the threshold below the topmost hill of the spectrum, the green area splits from the green line. Green line defines input signal, green area defines output signal. This way you can see the processor in action. Moreover a blue line appears on the top of the analyser view and shows the gain reduction for each frequency.

Let's say you want to maximize loudness and balance frequencies in the recording. In this case you want to make the output spectrum (green area) as horizontal as possible (except the total low-end and high-end which usually needs to be cut-off). Such spectrum makes the recording represent all frequencies with similar power which makes the brain consider it louder, hence "nicer".

The lower the threshold is, the more the spectrum is balanced, but also more artefacts it can cause. Other parameters related to the processor one we are using you can find in the Processor 1 panel. Besides threshold, ratio is also very important. It generally defines how much the power will be reduced (or conversely) above the threshold. Your goal is to find optimal balance between the threshold and ratio which increases loudness as much as possible and causes minimal artefacts.

Spectrum panel is also extremely important and defines the way the spectrum is understood by the plug-in and you. You should keep High quality whenever your CPU can handle it. Another very important parameter is Smoothness which generally defines how much each frequency affects its neighbours. You can expect more natural results for higher values, however it also maximizes the loudness less, because even weak frequencies look like their power is higher due to their neighbours.

Presets button

Presets button displays a window where you can load and manage available presets.

◁ button

the button loads previous preset.

▷ button

the button loads next preset.

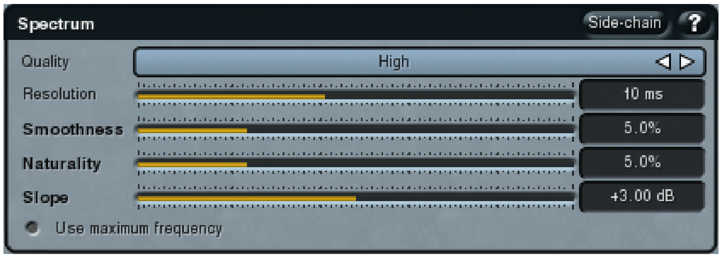
Channel mode button

Channel mode button shows current processing channel mode, such as L+R (which is default) and means processing left and right channels separately, or M+S which makes the plugin process mono and stereo channels separately.

Settings button

Settings button shows menu with additional settings and functions.

SPECTRUM PANEL



Spectrum panel contains properties that affect how the spectral information are detected and processed. All of these parameters affect the processing too, which is not true for most MeldaProduction plug-ins containing spectral analyser.

Quality

Quality defines analyser processing quality, therefore quality of the output signal. It can have significant impact on the performance. Note that different quality settings also produce different sound character, so you should not find optimal settings in low quality mode and then render the project in the high quality mode, since the sound would be simply different.

Resolution

Resolution defines how accurately the analyser can control attack and other parameters. Lower value causes the processing to be more

accurate but require much more CPU power. This parameter does not have significant impact on the quality of resulting signal, however it controls how precisely the requested time domain parameters are changed, mostly Attack. Optimally the value should be lower than attack.
Range: 1.0 ms to 50 ms, default 10 ms

Smoothness

Smoothness makes the analyser smooth out the curve, so it contains less bumping. It approximates energy in each frequency and the resulting graph may be easier to understand. Higher smoothness value typically produces more natural results, since it removes unwanted peaks.
Range: 0.00% to 20.0%, default 5.0%

Naturality

Naturality makes the processor smooth out the processing curve, so it avoids disruption of frequency correlations, that may cause some artefacts. Higher value produces more natural results, however requires more CPU power and lowers the processing precision.
Range: 0.00% to 20.0%, default 5.0%

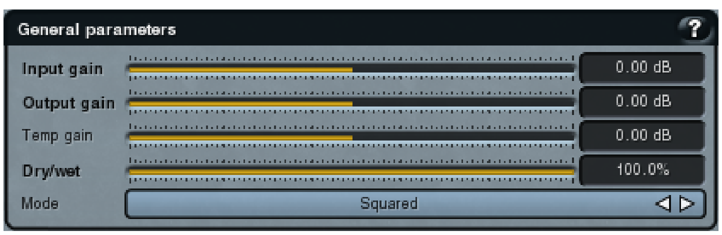
Slope

Slope makes the analyser increase magnitude of higher frequencies, since they are typically lower in energy. 3dB per octave is a default value, which makes the graph horizontal if it contains equal energy in each octave. It helps you understand the spectral properties of your audio signal and can be also used to change the global output signal character like this: Lower value typically creates a descent, which makes higher frequencies not exceed the threshold so only low-end is processed. Conversely higher value lets higher frequencies exceed a threshold and get processed more than the low-end.
Range: 0.00 dB to +6.00 dB, default +3.00 dB

Use maximum frequency

Use maximum frequency makes the processor use maximum magnitude of all the frequencies and apply them to the whole spectrum. It is useful for example to compress spectrum where the problematic frequency is changing.

GENERAL PARAMETERS PANEL



General parameters panel contains parameters related to dynamic processing.

Input gain

Input gain defines gain applied on the incoming signal. If you set ratio to 1:1 and custom shape is disabled, then the plug-in works simply as a fast gain processor.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Output gain

Output gain defines gain applied on the output signal. If you set ratio to 1:1 and custom shape is disabled, then the plug-in works simply as a fast gain processor.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

Temp gain

Temp gain defines temporary gain applied on the input signal and then reversed on the output. You can do the same effect by setting Input gain to a value g and Output gain to value -g. This plug-in moreover tries to approximate the gain reduction. The accurate approximation is not possible, however when you set the parameters so that the level is touching the threshold with temporary gain at 0dB, then any change to the temporary gain should change the amount of compression but keep the output level stable. Therefore the temporary gain in fact controls amount of compression.
Range: -24.00 dB to +24.00 dB, default 0.00 dB

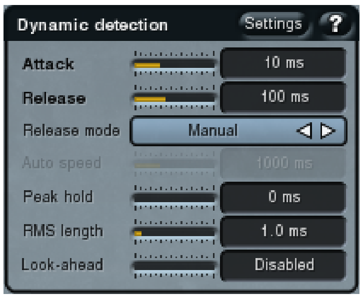
Dry/wet

Dry/wet defines ratio between dry and wet signals. 100% means fully processed, 0% means no processing at all.
Range: 0.00% to 100.0%, default 100.0%

Mode

Mode affects the processing shape. The plug-in features special non-linear transfer shapes which affect the way the signal is processed. Logarithmic produces classic dynamic processing where a signal exceeding the threshold by 10dB at a compression ratio of 2 : 1 produces 5dB attenuation in output level. In this same scenario, Squared mode produces a slightly greater output attenuation of 6.4dB and Linear mode produces a still greater value of 7.5dB. Thus, Squared and Linear modes produce progressively more compression / expansion. There is no compromise in sound quality between the different modes. Comparing the three modes, Linear mode requires the least amount of CPU power, and Logarithmic the most.

DYNAMIC DETECTION PANEL



Dynamic detection panel contains parameters defining how the plug-in determines level of the source signal.

Settings button

Settings button shows additional dynamics detector settings.

Attack

Attack defines how quickly can the processor increase the measured input level.
When using as a compressor, attack controls how quickly can the plugin start compressing. As a result, too small attack time will compress even beginning transient thus removing punch from a snare drum for example. Too high attack on the other hand will avoid even reaching the threshold, so the compressor may not do anything.
When used as a limiter, attack becomes a very sensitive control defining how much is the signal limited and how much saturated/clipped. If the attack is very low, the limiter catches most peaks itself and reduces them. That provides lower distortion, but can cause pumping. On the other hand higher attack (even above 1ms) may let most peaks through the limiter to the clipper or saturator, which causes more distortion but less pumping.
Range: 0 ms to 1000 ms, default 10 ms

Release

Release defines how quickly can the processor decrease the measured input level.
When used as compressor, release time defines how quickly can the compressor stop working. If the release is high, then after every peak, which drives the compressor above the threshold, it stays above it and compresses the audio a lot. As a result this attenuates sustain. Very low values usually cause distortions as the measured level is jumping quickly giving very nonuniform changes in level.
When used as limiter, the release time should be low enough, say about 10ms. With higher values the limiter may start pumping, because every peak makes it start compressing the audio, but the release time leaves the measured level above the threshold for too long. In an extreme case with very low attack and very high threshold the measured level may stay virtually constant causing the limiter to perform just a simple gain.
Range: 1.0 ms to 5000 ms, default 100 ms

Release mode

Release mode defines how the plug-in performs when decreasing level. In manual mode this is based only on the release time, which is suitable for most cases, when the signal has constant character. Automatic release modes can adapt to signals with unstable characteristics.

Since it may be hard to understand how each mode works, this is kind of an simplified explanation. In Automatic and Automatic fast modes, the longer you stay above threshold, the higher the release time will be and thus longer it will take to get below the threshold. In Linear 1 and Linear 2 modes, higher level makes longer release, so it tends to compress more once it finds higher levels. In Opto mode, higher level causes shorter release, so it tends to perform less compression.

For example, let's say you are compressing a full drum set. The sound of the hi-hat is very sharp and short, so it is appropriate to have short release times. Bass drum, however, often tends to have long decay, so it may be useful to have longer release times.

Automatic and Automatic fast modes are increasing release time when the signal is above threshold and vice versa. The speed depends on Auto speed parameter. Automatic fast mode gains full speed immediately after crossing the threshold, automatic mode varies the speed according to the current signal level.

Linear 1 and Linear 2 modes set release time based on current signal level only - below threshold it equals attack time, above threshold the release time is raising from the attack time up to specified release time parameter. Linear 1 mode usually provides higher release times.

Opto is oposite to Linear modes. It sets release time based on current signal level only - below threshold it equals release time, above threshold the release time is decreasing.

Auto speed

Auto speed defines how quickly the automatic release works. Specifically how the release time raises/lowers per second. It is relevant only in automatic release modes.

For example if you set 5000ms, the release time will be able to increase by 1000ms in 5000ms, when incoming signal exceeds the lowest threshold.
Range: 1.0 ms to 10000 ms, default 1000 ms

Peak hold

Peak hold defines the time that signal level holds its maximum. It can be used to improve gating.
Range: 0 ms to 1000 ms, default 0 ms

RMS length

RMS length defines time to reach approximately 36% of original volume intensity when receiving silence. Therefore increasing this value provides slower response. Conversely, setting this value to minimal value makes it a peak compressor.
Range: Peak to 100 ms, default 1.0 ms

Look-ahead

Look-ahead makes the processor use signal that has not actually arrived for dynamic calculation. This way it can respond even faster to dynamic changes and may help processing transients. This feature is useful for mastering, however it naturally induces latency, therefore it is not convenient for mixing and recording.
Range: Disabled to 1000 ms, default Disabled

GATE PANEL

Gate panel contains parameters for the noise-gate.

Threshold

Threshold defines maximal signal level, when the effect is applied.
Range: -80.0 dB to 0.00 dB, default -68.0 dB

Size

Size defines size of the interval between the gate threshold and point when the output signal level reaches zero.
Range: 0.00 dB to +24.00 dB, default +6.00 dB

Knee

Knee defines size of the smoothing knee.
Range: 0.00% to 100.0%, default 25.0%

Bottom

Bottom defines volume reached when the gate is fully closed, hence the threshold minus size. In most cases you leave this silence.
Range: silence to -80.0 dB, default silence

PROCESSOR 1 PANEL

Processor 1 panel contains parameters of the primary processor, which can behave like a compressor or expander.

▽ button

the button switches the processor into a downward. expander. In this mode the processor reacts to levels below the threshold, instead of above the threshold in normal mode. Despite downward compression can be done too, this mode is particularly useful for expansion, since upward expansion is somewhat dangerous as it can significantly amplify the audio, way above 0dB.

Threshold

Threshold determines minimal signal level, when the effect is applied.
Range: -80.0 dB to 0.00 dB, default -40.0 dB

Ratio

Ratio defines compression ratio of input signal above threshold.
Range: 1 : 1.50 to Infinity, default 1.50 : 1

Knee size

Knee size defines size of the knee.
Range: 0.00% to 100.0%, default 25.0%

Range

Range defines size of the interval above the threshold after which the original signal ratio is restored.
Range: +1.00 dB to +96.00 dB, default +96.00 dB

PROCESSOR 2 PANEL

Processor 2 panel contains parameters of the secondary processor, which can behave like a compressor or expander.

▽ button

the button switches the processor into a downward. expander. In this mode the processor reacts to levels below the threshold, instead of above the threshold in normal mode. Despite downward compression can be done too, this mode is particularly useful for expansion, since upward expansion is somewhat dangerous as it can significantly amplify the audio, way above 0dB.

Threshold

Threshold determines minimal signal level, when the effect is applied.
Range: -80.0 dB to 0.00 dB, default -52.0 dB

Ratio

Ratio defines compression ratio of input signal above threshold.
Range: 1 : 1.50 to Infinity, default 1.50 : 1

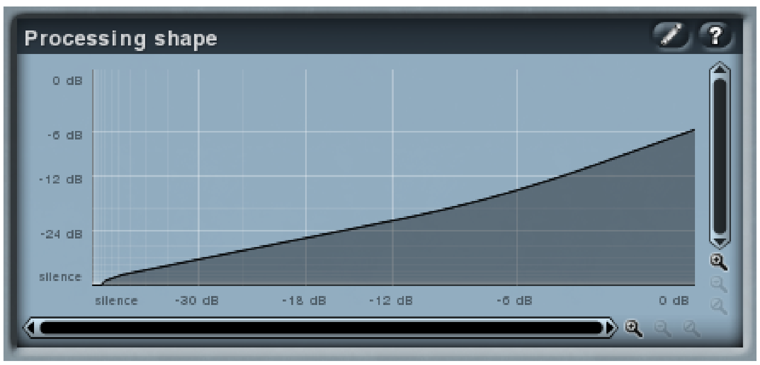
Knee size

Knee size defines size of the knee.
Range: 0.00% to 100.0%, default 25.0%

Range

Range defines size of the interval above the threshold after which the original signal ratio is restored.
Range: +1.00 dB to +96.00 dB, default +96.00 dB

LEVEL SHAPE GRAPH

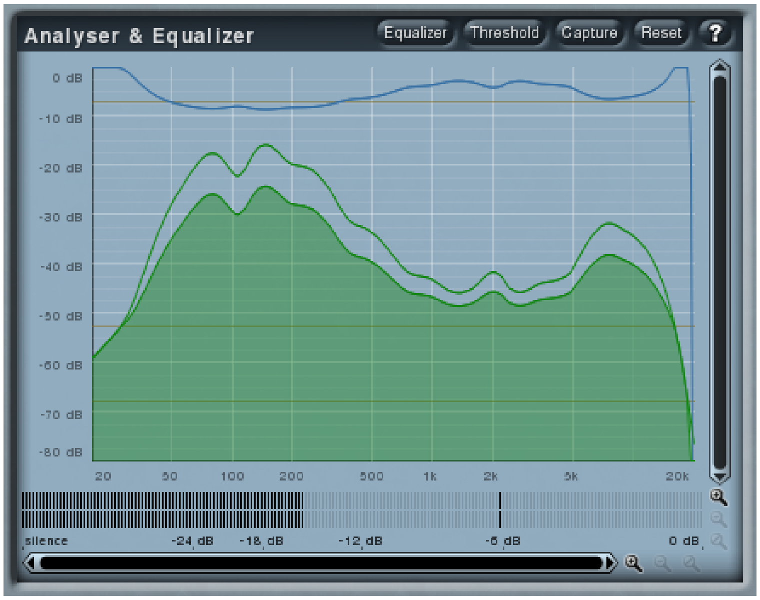


Level shape graph defines the dynamic processing envelope. X axis contains input signal level, Y axis defines output level.
Note that this display is not logarithmical. This can lead to a confusion, because for example moving expander's threshold changes the graph slope despite the ratio stays the same. This is however necessary, because logarithmical display can never contain silence, as it is minus infinity decibels, and silence point is essential for gates for example. The display is therefore a compromise between usability and accuracy.

button

the button enables or disables custom level shape. When enabled, it inherits the automatic settings and you can draw any processing shape you want.

ANALYSER & EQUALIZER PANEL



Analyser & Equalizer panel contains dynamic analysis of the signal and a free-form equalizer applied before the signal is processed through the dynamic processor. The equalizer has no CPU requirements, so do not be afraid to use it to correct the source material and remove unwanted frequencies etc.
On top you can see 2 buttons - Equalizer and Threshold. The first you already know. The latter is attached to the Processor 1 threshold, however it modifies threshold of any processor or custom shape point. This way you can define amount of processing for each frequency.
Green area contains signal power of the output signal after processing. Dark green line describes signal power of the incoming signal before processing. Blue line on top contains gain reduction for each frequency. Horizontal orange lines indicate thresholds of enabled processors unless custom shaping mode is enabled.

Equalizer button

Equalizer button shows or hides the free-form equalizer.

Threshold button

Threshold button shows or hides the free-form threshold editor.

Capture button

Capture button sets the threshold graph to current analysis. This can be used to provide a very transparent spectral compression for example. Hold Ctrl to avoid computing long-term average and use instant analysis. Conversely hold Shift to perform a long-term analysis with especially long averaging time.

Reset button

Reset button restores the original free-form equalizer or threshold settings - settings of the current graph. If you hold Ctrl both are reset.

Multiparameter button

Multiparameter button displays settings of particular multiparameter, which can control multiple other parameters at once. the button shows smart learn menu. You can also use right mouse button. In smart learn mode the modulator/multiparameter does not work and rather records your actions. You can use every automatable parameter and use it normally. When you change a parameter, the plugin adds it to the modulator/multiparameter and also notices the interval of values you set.
For example, to record a frequency slider and make a modulator control it from 1Hz to 10Hz, just enable the smart learn mode, click the slider the move it from 1Hz to 10Hz. Then disable the learning mode by clicking on the button.

 button

the button shows smart learn menu. You can also use right mouse button. In smart learn mode the modulator/multiparameter does not work and rather records your actions. You can use every automatable parameter and use it normally. When you change a parameter, the plugin adds it to the modulator/multiparameter and also notices the interval of values you set.
For example, to record a frequency slider and make a modulator control it from 1Hz to 10Hz, just enable the smart learn mode, click the slider the move it from 1Hz to 10Hz. Then disable the learning mode by clicking on the button.

Upsampling

Upsampling can potentially improve sound quality by performing processing at a higher sample rate, which can avoid aliasing. However upsampling has a huge impact on the CPU requirements. Also since upsampling is essentially filtering, it can add some artifacts on its own, and for some algorithms processing at higher sampling rates can lower the audio quality, so you should use it only if you need it. Note that high-quality upsampling mode induces latency, which usually cannot be reported to the host. As an alternative you can simply work at higher sampling rates. Upsampling is usually useless when processing in 96 kHz or higher. We recommend recording and processing in 96 kHz, which is absolutely sufficient without upsampling in most cases.

Presets selector

Presets selector defines current preset. The plugin can handle multiple presets at once. When you change any parameter, only current preset is modified. All presets are stored in the project. This way you can easily check changes and find the best settings for your case. Preset selection is not automatable.

A/B button

A/B button switches between this and previous preset. You can do the same thing by clicking on particular preset, but this makes it easier letting you close your eyes and listen, hence avoiding prejudice.

Morph button

Morph button let's morph between ABCD settings. Note that if you have selected e.g. A setting, you will actually change it, so it is suitable to select for example E settings and then use morphing. Also note that there are parameters which cannot be morphed.

 button

the button copies current settings to clipboard. Other presets and upsampling settings are not copied.

 button

the button pastes settings from clipboard into current preset.

MULTIPARAMETER EDITOR

Name

Name specifies name of the parameter, which is shown on the multiparameter button. The name is also used for active presets - if the name is specified, then the multiparameter serves as a parameter for the active preset. However if the first character is *, then the parameter is hidden. This is useful if you need some internal multiparameters, which you don't want to show.

Group

Group can be used to put some multiparameters into the same group, which results in placing them in the same panel on the active preset editor.

Info

Info may contain additional information about the multiparameter.

Mode

Mode controls the behaviour of the multiparameter. Normal mode makes the multiparameter work like any other slider. Switch mode hides the slider and shows a button instead. The button has 2 states. By pushing the button the multiparameter, the multiparameter value goes from 0% to 100% over a specified time interval. By unpushing the button the value goes back to 0%. You could do the same thing having the multiparameter in normal mode and moving the slider from left to right and then back, but this performs that manually and maintains the time interval. Trigger mode is similar to switch mode, but the button has only a single state and when you push it, the value automatically goes from 0% to 100% and then back without any need to push the button again.

Value mode

Value mode controls units displayed on the multiparameter. Percents mode lets the plugin display percents from 0% to 100%. By first parameter mode uses the current value of the first parameter controlled by the multiparameter. For example, if you want to control a plugin gain, but also in addition to the changed gain control other parameters, you may still want to call the multiparameter gain and the units should be decibels as usual, not percents which do not make much sense for that parameter.

Switch time

Switch time defines time needed to switch from the minimum value to the maximum one, or conversely. It is used only in switch and trigger modes.

Set current value as defaut

Set current value as defaut stores current value as default one for the multiparameter.

Parameter

Parameter defines target parameter being controlled. The set contains all automatable parameters.

Range mode

Range mode defines from which range are the values taken.
Up and down mode makes the values go above and below selected Value, which is considered the center. The interval is compressed if necessary. For example, when value is 10% and range 100%, possible outputs are going from 0% to 20%, thus maximal interval around 10%.
Full range mode is similar, except the interval is never compressed, so the selected value may not be the center anymore. For example, when value is 10% and range 50%, possible outputs are going from 0% to 50%. But if value is 50%, then the interval is from 25% to 75%.
Up/down only mode goes from the selected value up/down only. For example, when value is 10% and range 50%, possible outputs are going from 10% to 60% in up only mode.
Interval mode is the most simple and simply goes between specified value and maximal value.

Value

Value defines center value of the modulation.

Maximal value

Maximal value defines limit when interval mode is being used.

Depth

Depth defines modulation range, size of the interval from which the values are used. Higher depth causes higher modulation and more audible effect.

Invert

Invert checkbox inverts the modulator shape, so minimum becomes maximum etc.

Use first parameter's range

Use first parameter's range makes the parameter use the same range as the first parameter in the list. This is often useful if want to control the range somehow and apply the range to multiple parameters.

SHAPE GRAPH

Shape graph lets you tweak the shape of the curve used to control selected parameter. X axis shows the original values, Y axis defines the results. Note that this takes some CPU, therefore you have to enable it using the the button in the caption.

Presets button

Presets button displays a window where you can load and manage available presets.

◁ button

the button loads previous preset.

▷ button

the button loads next preset.

MIDI CONTROLLERS EDITOR

Presets button

Presets button displays a window where you can load and manage available presets.

◁ button

the button loads previous preset.

▷ button

the button loads next preset.

CONTROLLERS PANEL

Controllers panel contains settings of MIDI controllers.

Enable

Enable enables or disables the controller.

Parameter

Parameter defines target parameter being controlled. The set contains all automatable parameters.

Learn

Learn enables or disables MIDI learn.

Channel

Channel defines controller MIDI channel.

Controller

Controller defines source controller.

Value

Value defines center value of the modulation.

MaxValue

MaxValue defines maximum value in case interval mode is used.

Depth

Depth defines modulation range, size of the interval from which the values are used. Higher depth causes higher modulation and more audible effect.

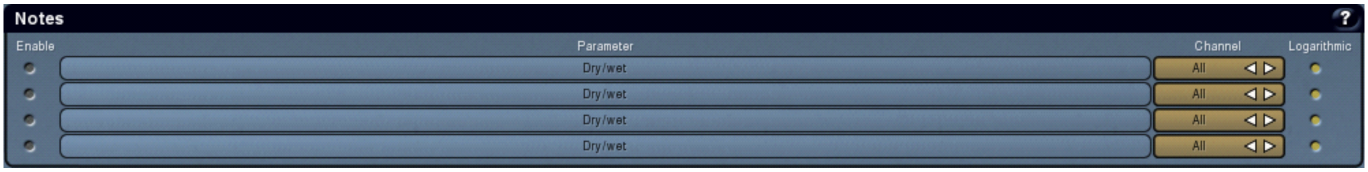
Range mode

Range mode defines from which range are the values taken.
Up and down mode makes the values go above and below selected Value, which is considered the center. The interval is compressed if necessary. For example, when value is 10% and range 100%, possible outputs are going from 0% to 20%, thus maximal interval around 10%.
Full range mode is similar, except the interval is never compressed, so the selected value may not be the center anymore. For example, when value is 10% and range 50%, possible outputs are going from 0% to 50%. But if value is 50%, then the interval is from 25% to 75%.
Up/down only mode goes from the selected value up/down only. For example, when value is 10% and range 50%, possible outputs are going from 10% to 60% in up only mode.
Interval mode is the most simple and simply goes between specified value and maximal value.

Invert

Invert checkbox inverts the modulator shape, so minimum becomes maximum etc.

NOTES PANEL



Notes panel contains settings of MIDI note controllers, thus if you want to control parameters using MIDI keys.

Enable

Enable enables or disables the controller.

Parameter

Parameter defines target parameter being controlled. The set contains all automatable parameters.

Channel

Channel defines controller MIDI channel.

Logarithmic

Logarithmic if logarithmic scale is used which is common for oscillator frequencies, however may not be useful for general parameters.

CONTROL SPECIFICATION

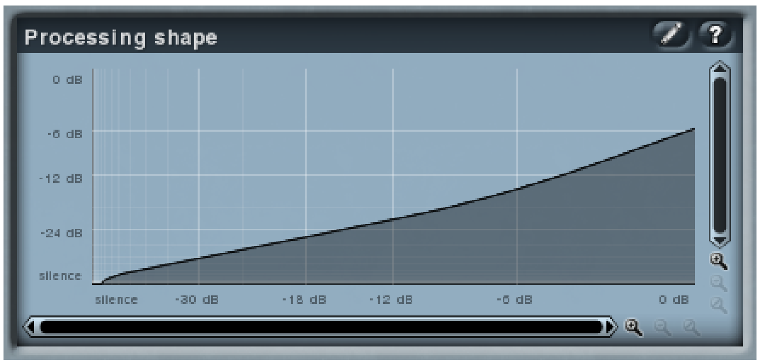
Here we will discuss the general properties of all application controls. As a most important rule you should note, that you can always use any question mark button or F1 key with mouse cursor at a specified control to get detailed information about what it does and how to use it. If the F1 key does not work, it is possible that some other application is using it, so please try holding Ctrl, Alt, Shift or any combination.

ZOOMER

Zoomer provides a simple way to zoom and move in an enlargeable view.

- Plus (+) button zooms-in.
- Minus (-) button zooms-out.
- Slash (/) button zooms to default ratio, which typically means full zoom-out.

GRAPH EDITOR



Graph editor will show and edit one or more graphs.

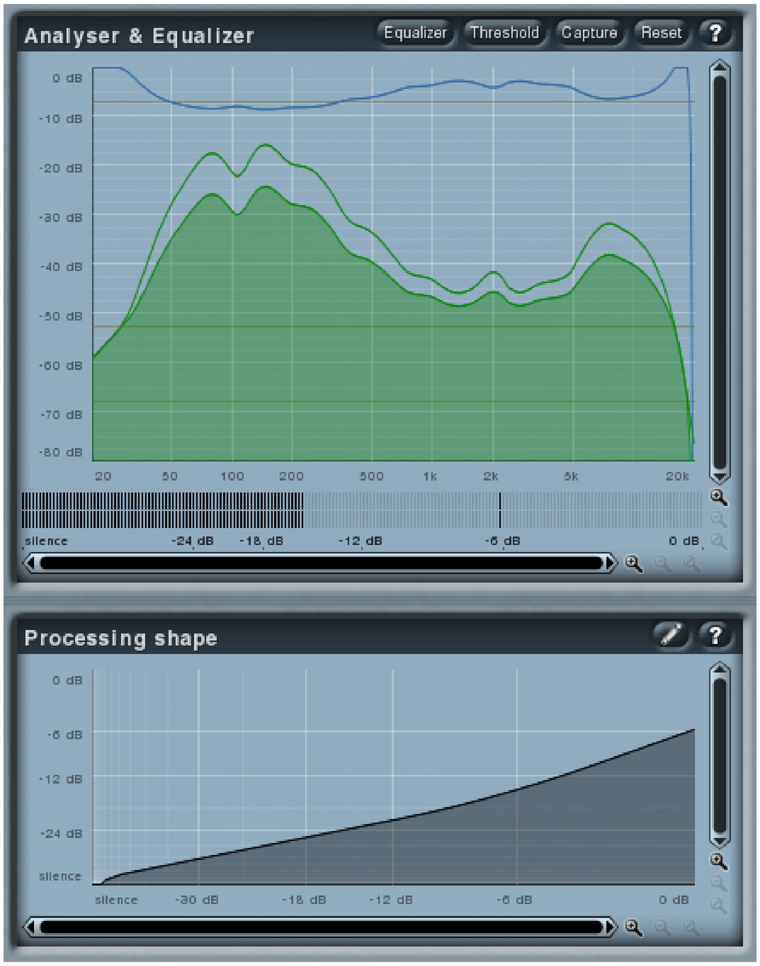
- Zoomers below and on the right control zoom and position of the view.
- Mouse wheel zooms in or out. Alternatively you can zoom in using Alt + right button double click and out using Alt + left button double click. You can also use keyboard numbers 0 to 9 to quickly set zoom level.
- Drag a rectangle using the left mouse button while holding Alt zooms into the selected rectangle if possible.
- Drag using the left mouse button while holding Alt and Ctrl to scroll the view. This is not possible when zoomed all the way out.
- Left mouse button can be used to select points. If there is a *point*, you can move it (or the entire selection) by dragging it. If there is a *curvature circle*, you can setup tension by dragging it. If there is a *line*, you can drag both edge points of it. If there is a *smoothing controller*, you can drag its size. Hold Shift to drag more accurately. Hold Ctrl to create a new point and remove any points above or below.
- Left mouse button double click can be used to create a new point. If there is a *point*, it will be removed instead. If there is a *curvature circle*, zero tension will be set. If there is a *smoothing controller*, zero size will be set.
- Right mouse button shows a context menu relevant to object under the cursor or the entire selection. Hold Ctrl to create or remove any points above or below.
- Middle mouse button drag creates a new point and removes any points above or below. It is equal to holding Ctrl and dragging using left mouse button.
- Mouse wheel over a point modifies its smoothing controller. If the point is selected, the entire selection is modified.
- Ctrl+A selects all points. Delete deletes all selected points.

SWITCHER

Switcher is an alternative to tracker or knob controls, but it has only a limited set of values.

- Left mouse button shows a menu with list of all possible values. This function might be unavailable in certain cases when the number of possible values is too high.
- Up and down arrow keys, buttons in the control and mouse-wheel increase or decrease the value.

SPLITTER



Splitter is typically used to subdivide an area between multiple editors.

- Drag a separator to resize a row or column.
- Drag a crossing of two separators to resize all affected rows and columns.

TRACKER

Tracker (also known an slider) is an alternative to common knob control. However the tracker is typically quite small, easy to use and capable of quite high precision and in most cases provides immediate text or similar representation of value you are editing.

- Click/drag using left mouse button to change the value.
- Right mouse button selects default value.
- Mouse wheel, arrow keys and vertical drag using middle mouse button or using left mouse button while holding Ctrl modifies the value more accurately.
- Home key configures minimal possible value, conversely end key setups a maximal one.
- Shift + left mouse button lets you edit the value as text.