OhmBoyZ

Reference Manual

Contents

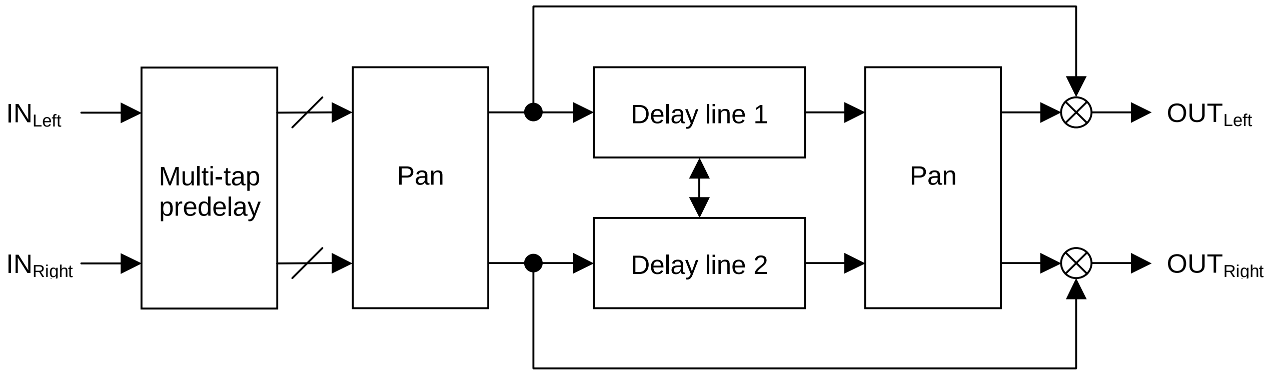
1. Installing the plug-in

Depending on the OhmBoyz version you get, installation procedure may vary. Please refer to the `readme.txt` file included in the archive to know more about installation and specific requirements.

2. Internal structure of the effect

This part is technical but it will help you understand OhmBoyz, and therefore, it will be easier to use it and get results rapidly. First, OhmBoyz can be compared to a stereo delay. But it is , in fact, more complex. There are two different stages :

- 4 predelays that can produce 4 different replicas of the sound shifted in time.
- 2 delay lines, generating an echo for the replicas generated by the predelays.

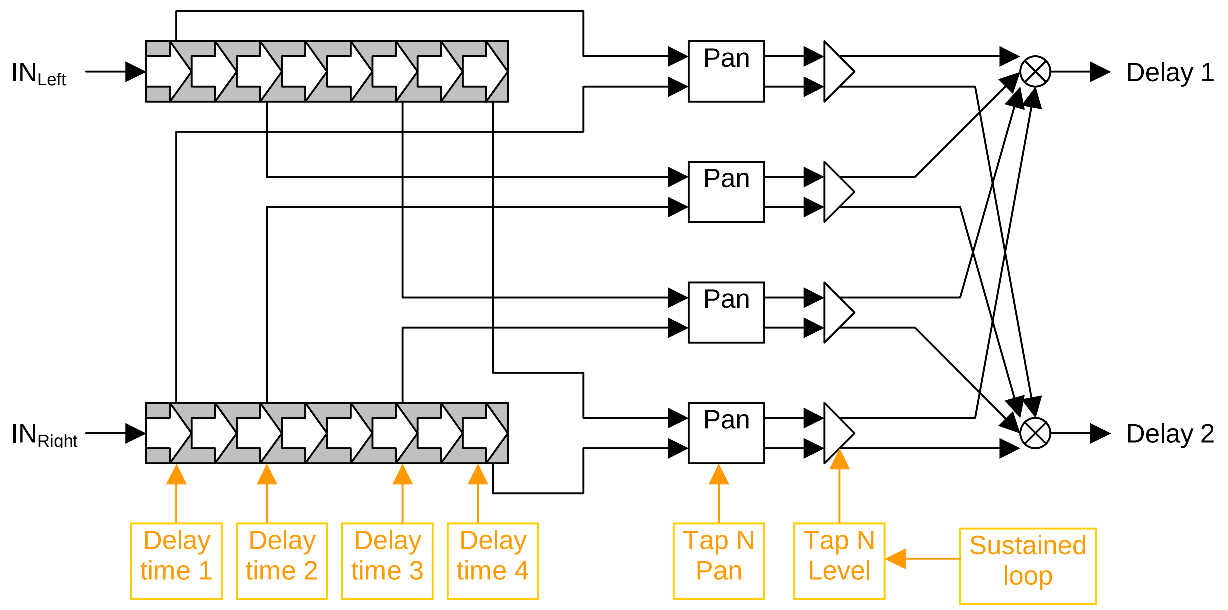


Various effect parameters are connected to LFOs (Low Frequency Oscillator). They enable a parameter to oscillate around its central value. It is therefore possible to get an alive effect that will change all along the time of the song.

2.1 Predelays

Four replicas of the sound are generated by that layer. Each delay of the replicas can be adjusted, as well as its volume and its stereo balance.

When there is only one delay line, the pan of the taps has no effect. If you only want one tap, the 3 other taps volume has to be set on 0.



2.2 Delay lines

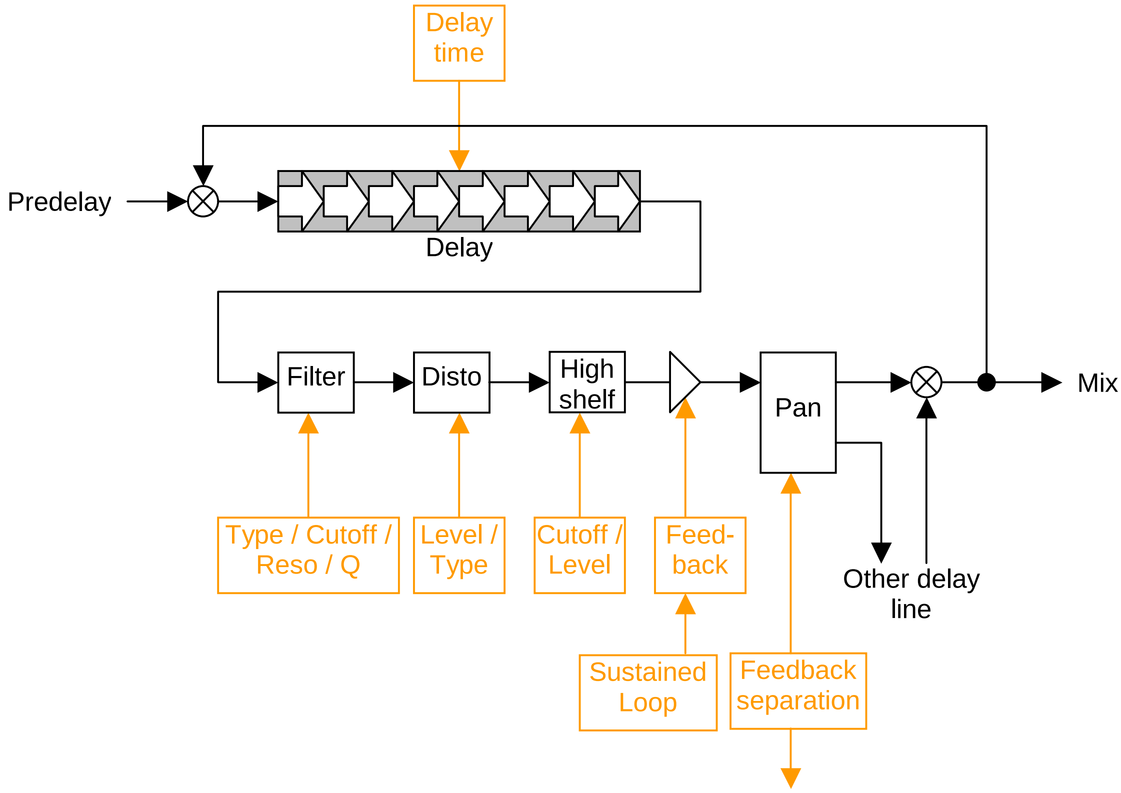
It is the most important part of the sound effect. The predelay stereo output is sent on one or two delay lines (according to the settings you have chosen). If you chose two delay lines, the left output will go in the first delay line, and the right one on the second delay line.

Each one of those two delay lines have the basic characteristics of a classical delay : delay time (repetition between two echoes), and feedback amount (level difference of two consecutives echoes).

We also added other processing (filter and distortion) inside the loops of the delay. Thus, signal is regularly reprocessed and the effect amplifies.

When there are two lines of delay, it is possible to « cross » their feedbacks. In crossed mode, the output signal of the first line is put in the second line, and the output signal of the second line is put in the first line. Any variation between « straight » and « crossed » mode can be obtained.

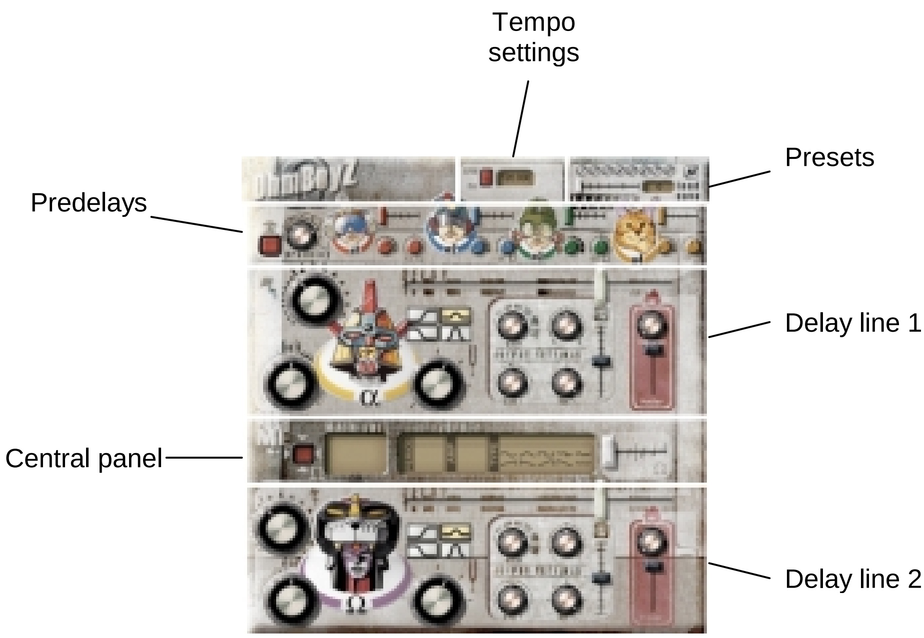
Lastly, the final output of the lines can be balanced and mixed with the predelay output.


Delay
time
Predelay
Delay
Filter
Disto
High
shelf
Pan
Mix
Type / Cutoff /
Reso / Q
Level /
Type
Cutoff /
Level
Feed-
back
Other delay
line
Sustained
Loop
Feedback
separation

3. User interface

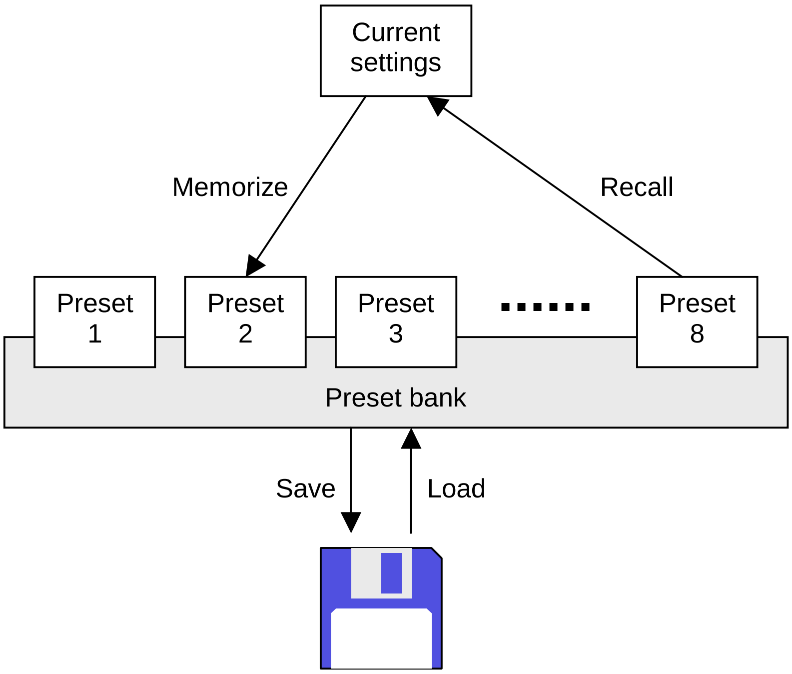
The user interface is divided into several distinct parts :

- Presets
- Tempo setting
- Predelays
- Two delay lines
- Central panel

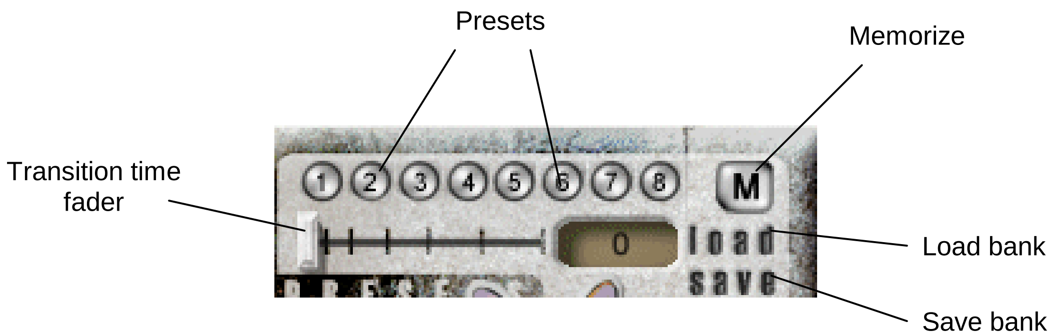


3.1 Presets

Most Ohm Force plug-ins have the preset feature ; it might not look alike, but it has the same functionalities. It enables you to manage your sound settings. A bank of eight slots enables you to memorize your settings, and can be saved on your hard disk. Those banks are multi-platforms ; therefore, you can use your presets on another computer or with another sequencer.



You can, when applying a preset, add a transition time, during which the buttons are going to turn slowly to go from the previous setting to the one you have chosen.



- **Presets / Memorize**

To memorize the effect current setting in a preset, click once on the M button ; it will light on. Then, click on button of the preset in which you want to memorize the effect.

To apply a preset, make sure the M button is off (to turn it off, click it once). Then, click on the preset you want to execute.

- **Transition time**

This fader enables you to vary the time the plug-in will take to go from a sound to another when a preset is executed. The time measured in seconds is displayed beside. By default, this duration is to 0 ; thus the preset application is immediate.

- **Load / Save Bank**

Those two buttons will help you save and load your preset banks on the hard disk for a later use. The 8 presets are memorized or loaded at once. During disk loading, the current setting is not modified.

3.2 Tempo setting

Various parameters (such as delays or LFOs) are time values which are expressed under time units. From a musical point of view, it is more easy to express them as beats. Since the beat duration depends on the musical tempo (called BPM, i.e. number of beats per minute), the tempo has to be adjustable. This part of the interface is dedicated to this task.

- **Display and tempo control**

When the host (a sequencer for instance) generates the tempo, you cannot control it from the plug-in which will only display the value that the host gives it. That is why the tempo control will not be displayed on the GUI.

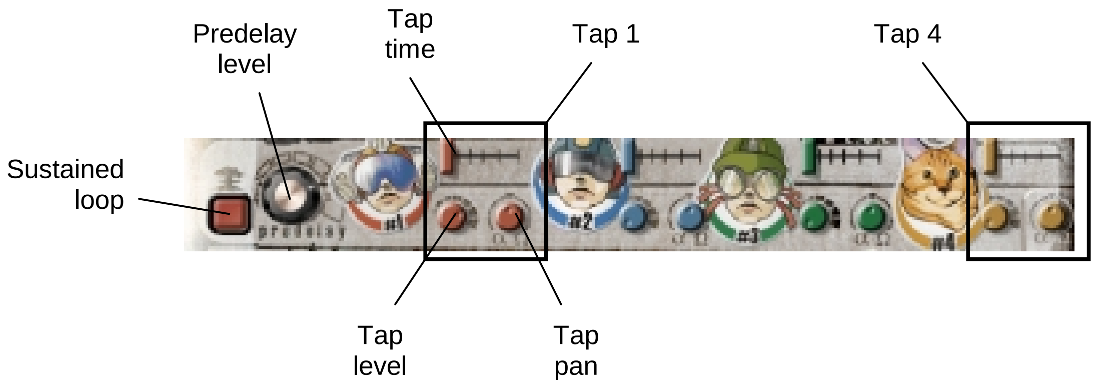
If it is on the GUI, you can control the tempo by clicking on the tempo control or by entering a value on the keyboard. Tempo is a parameter therefore, even if the sequencer does not « give » tempo to its plug-ins, you can control tempo with parameter automations anyway (if your sequencer support that feature).

- **Time unit**

It can be useful to be given the temporal parameter times in seconds. The switch button enables you to display the time values either in beats or in seconds.

3.3 Predelay zone

In this zone, you can control the predelay parameters. Each of the four group of buttons corresponds to a « tap ».



3.3.1 For each tap

- **Tap time**

It is the delay of each sound replica.

- **Tap level**

It is the volume of the tap. 0 dB is the volume of the original sound.

- **Tap pan**

It is a panoramic potentiometer which controls :

- The mix from the predelay outputs. It works like a standard stereo balance.
- The inputs of the delay lines. When the two delay lines are activated, a pan set on the left will direct the sound in the first line, and a pan set on the right will direct it in the second line.

3.3.2 Others

- **Predelay level**

It only controls the mix of the taps in the final output signal. You can control their volume. The normal volume is 0 dB.

- **Sustained loop**

By clicking on that magic button, you will generate a sustained loop (infinite echo). A sustained loop is the following :

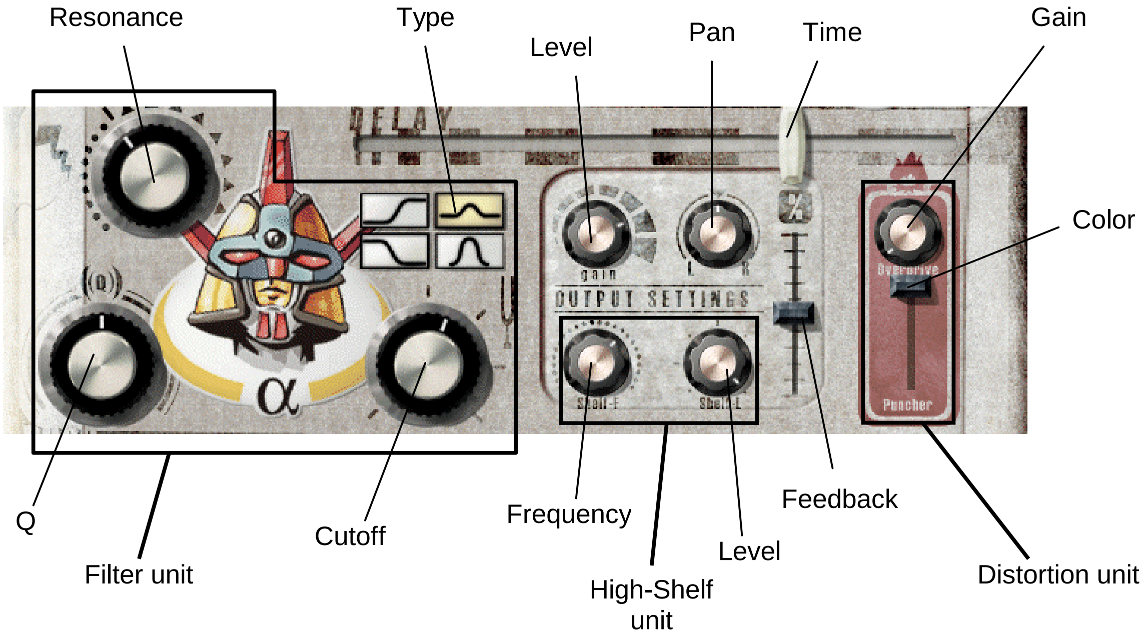
- Sound input is muted
- Feedback of the two delay lines is set to 100%

These parameters values are not lost, they are just bypassed. Click again to get the original sound back.

3.4 The two delay lines

Depending on the mode you have chosen, one or two delay lines are displayed. They have the same functions and each of one can be divided as follows :

- The delay and its mix
- Resonant filter
- Distortion
- High-shelf filter



3.4.1 Delay and mix

It is the basic part of the line, where the most important functions are.

- **Time**

This parameter gives the time between two echoes. It gets shorter as you slide the cursor on the left.

- **Feedback**

It is the regeneration rate of the delay : the ratio between the volume of an echo and the volume of the next one. When this ration is 50 %, it means that the volume is twice less loud. When it is 100 %, there is an infinite echo ; sound level won't decrease.

- **Level**

It is the volume of the line in the final output.

- **Pan**

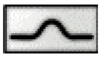
Gives the stereophonic position of the line in the final output.

3.4.2 Resonant filter

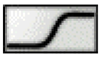
The filter is the first effect inside the loop of the line. It controls the damping of the sound and can accentuate some frequency bands.

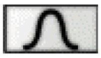
- **Type**

You can select one of these four filters :

 **Peak** It increases (wha-wha effect) or decreases (notch style) a particular frequency range without changing the others. The bandwidth is given by Q, and the peak height depends on resonance. When the resonance is set to 0 dB, it means that the sound won't be modified.

 **Low-pass** It is a filter which passes frequency components below a cutoff frequency and rejects other frequency components. When resonance is increased (over –3 dB), a peak is generated at the cut-off frequency, and therefore intensify it.

 **High-pass** It is similar to the low-pass filter, except that it rejects the low frequencies.

 **Band-pass** It passes all frequency components inside a selected band of frequencies and rejects all frequency components outside. Like the peak filter, the bandwidth is given by Q. Resonance has only a volume effect.

- **Cutoff**

It controls the filter cut-off frequency.

- **Resonance**

Resonance gives the gain of the signal at the cut-off frequency. It is an exact value for the peak and the band-pass filters, but it is a bit less accurate for low-pass and high-pass filters.

The resonance level has to be manipulates very carefully : if it is too high, the delay line has an auto-oscillation effect that you might not want. To prevent this, the resonance level plus the feedback level must be under 0 dB. For instance, if the feedback level is 50 % (-6dB), the resonance level should not be over 6dB.

- **Q**

This parameter is used by the peak and the band-pass filters. It gives the bandwidth. When Q is low, the band is wide and the filter is less selective than when Q is high (thin band).

When the band is thin, its sound volume is not very loud. It could be therefore necessary to increase the resonance level.

3.4.3 Distortion

Distortion occurs right after the sound is filtered. It can degrade or boost the sound, as you wish.

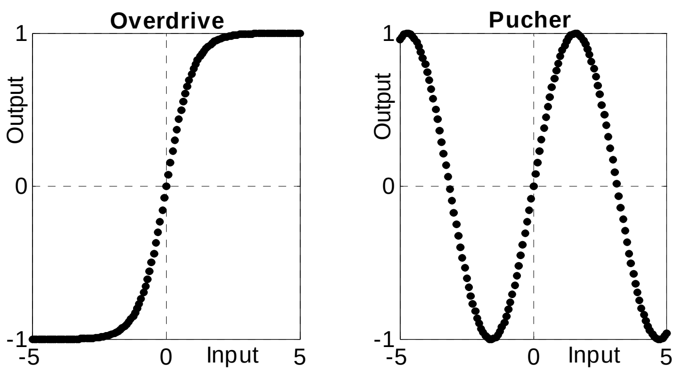
- **Gain**

It is the control which regulates the amount of distortion. When it is set to 0, the sound has not been changed.

- **Color**

This fader shows the curve which damage the sound. According to the curve you choose, the effect is different.

- on overdrive, a saturation is put into the sound. It gives it a large amount of sharp tones.
- on puncher, the sound gets more powerful. The new tones are added in a different manner than on overdrive. With a high gain and an input sound with many sharp tones, it is possible to obtain a white noise.



3.4.4 High-shelf filter

This filter slightly attenuates the sharp tones (first order low-pass filter) ; this is often useful when distortion is used.

- **Frequency**

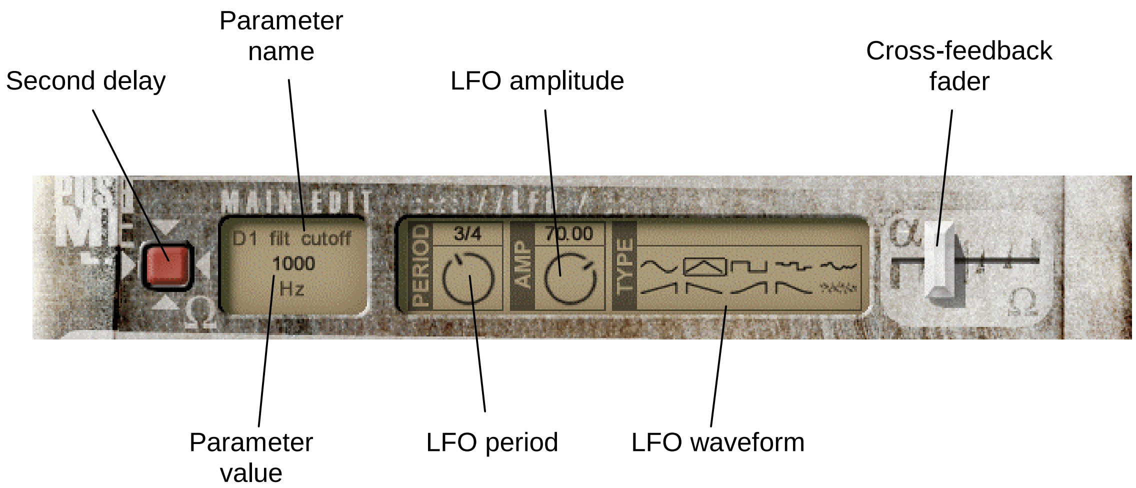
Frequency above which the sound begins to be attenuated.

- **Level**

It is the shelf level, that is how many decibels lessen the high frequencies. 0 dB means the high-shelf does not modify the sound.

3.5 Central panel

This panel has various functionalities :



3.5.1 Parameter

This contextual zone depends on which parameter you have chosen (it has a yellow outline). Indeed, it would not be simple if all the parameter digital values were displayed on the interface at once.

- **Name**

Name of the selected parameter.

- **Value**

It is the parameter value expressed with the selected unit. You can edit this value by clicking on it.

3.5.2 LFO

Like the Parameter zone, the LFO zone is activated only when a parameter has been selected, and when there is an associated LFO.

- **Period**

It the time taken by an LFO oscillation.

- **Amplitude**

It is the amplitude of the oscillations. 0 % means that the LFO does not affect the sound.

- **Waveform**

This parameter affects the shape of the oscillations. The forms on the left have a classical shape (triangle, saw-tooth, square...). The three forms on the right are various random oscillations.

 **Sine** It is the default waveform.

Triangle	Oscillation go and returns linearly between the two extreme points.
Square	LFO stays a half-period on the maximum point, then the other half-period on the minimum point.
Ramp up	Oscillation go linearly from the minimum point minimum to the maximum one.
Ramp down	Like Ramp up, but in the other direction.
Cos up	A bit like Ramp up, but LFO go and arrive more gently at the extreme points (a kind of shelf).
Cos down	Like Cos up, but in the other direction.
Random	At each period beginning, LFO takes a random values which keeps constant until period end.
Brown noise	LFO value changes randomly, combining wide but slow moves with small and fast oscillations. With a very long period, this kind of LFO is perfect to give a parameter a quasi-human random variation.
White noise	LFO value changes randomly very quickly. Period doesn't affect the movements.

3.5.3 Miscellaneous

- **Second delay**

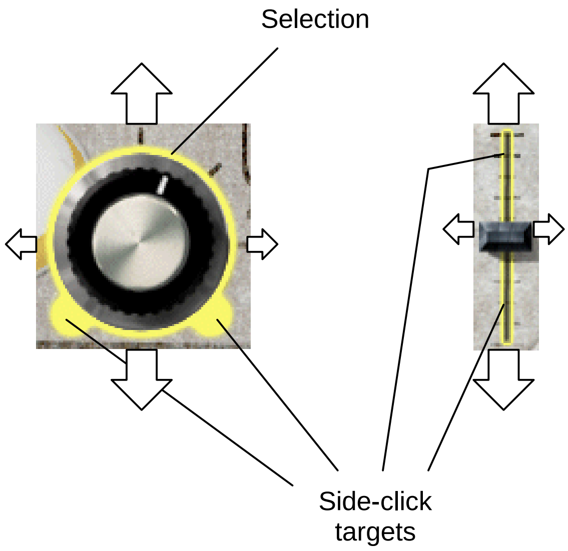
Click on this button to activate or cancel the second delay line.

- **Cross-feedback**

This fader does not work when the second delay line is activated. It regulates the separation of the feedbacks. When it is on the left, the two line are independent, and when it is on the right, they are crossed.

3.6 Using knobs and faders

All the knobs and the faders work the same way. There are two modes : direct action or slide-clicks.



3.6.1 Direct action

You can catch the button by giving a long click on it (on the mobile zone for the fader) and move the mouse on the right or on the left, keeping the button pressed.

Actually, each button has a direction : the knobs move vertically (you do not have to move your mouse around the button) and the faders move according to their orientation. If you move the mouse in the same direction, the move will be quick. But, if you move your mouse in the orthogonal direction, then the move will be slow and therefore very accurate.

For the « notch » buttons, like the temporal parameters buttons, you can chose a value which is not on the « notch » by using the orthogonal movement (accurate).

3.6.2 Side-clicks

The button is divided into two zones on which you can click to turn the button on the right or on the left. For the fader, those two zones are on both sides of the moving part. For the knob, they are at 4:30 and 7:30 on the button.

The button will move slowly if you give a long click on these zones without moving the mouse. This can help slightly and quickly adjust a parameter value.

If you click on this zone then move your mouse without releasing it, the button will move automatically, and keep moving even after having stopped clicking. When you move the mouse away from where you have clicked, the movement of the button will get faster. To stop that move, just click again.

3.6.3 The buttons of the delay lines

To facilitate those manipulations when using two delay lines, you can chose to apply your moves to each one of the two lines. For instance you can affect one parameter of delay 2 and delay 1 at the same time.

To do so, you have to click on the button with right click of the mouse (Control + click for Macintosh systems). The parameters of both lines take the same value.

If you give a long press on the Shift key and click on the right mouse button, both buttons move at the same time but keep their own original gap. For instance, when the original value of the first button is 10 % and the original value of the second button is 50 %, if you higher the value of the first button up to 30 %, you higher the value of the second button up to 70 % at the same time.

You can also combine the right click to the control key (the Apple key on Macintosh). It will reverse the move of the slave button.