

IRCAM Verb

FLUX:: Immersive

2/6/23

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1 IRCAM Verb

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ircam
Tools

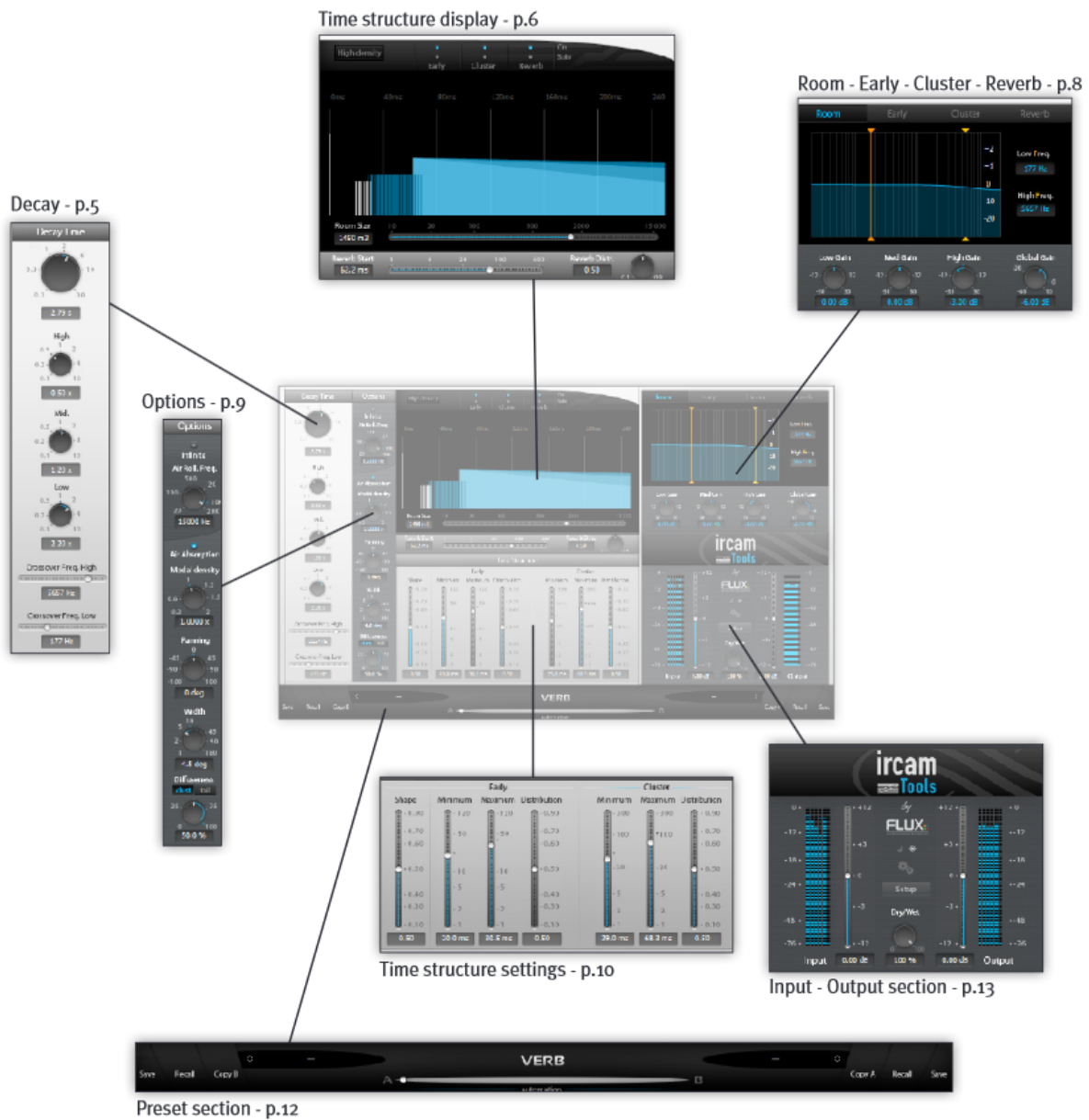
by

FLUX
sound and picture development

VERB

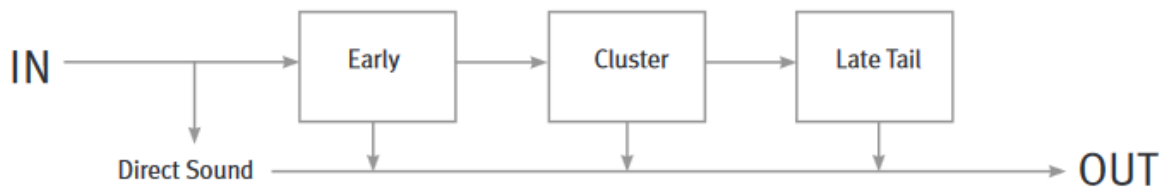
Room Acoustics & Reverberation processor





2 Quick start guide

The Verb is an algorithmic reverberation processor build in a modular way, with an recursive filtering based engine.



This block diagram view explain the basics of this engine, to explain the time structure of the reverberation tail:

- The direct signal is pushed first. It represents the direct sound, aka. the sound that comes first to the listener.
- Then, follow a first generator of early reflexions, called here EARLY. Theses early reflexions are particularly important as they describe the immediate spatial environment around the sound source: walls, floor, roof. Theses reflexions are depending on the source position : they are panned in the space.
- A second delay generator follows the early stage, called CLUSTER. Theses early reflexions comes a bit later, and have seen their density increased in comparasion to the first early. They can be shown as a transition stage between the early and the late tail generator. In a standard configuration, theses reflexions are seen as a common part of all the space : they are not localized. You can make it directive using the diffuseness parameter.
- A final LATE TAIL stage, synthetize the reverb tail. Most of the time, acoustical description of this tail shows it as an dense and homogeneous material in the reverberated space (you can change that too with the same diffuseness parameter).

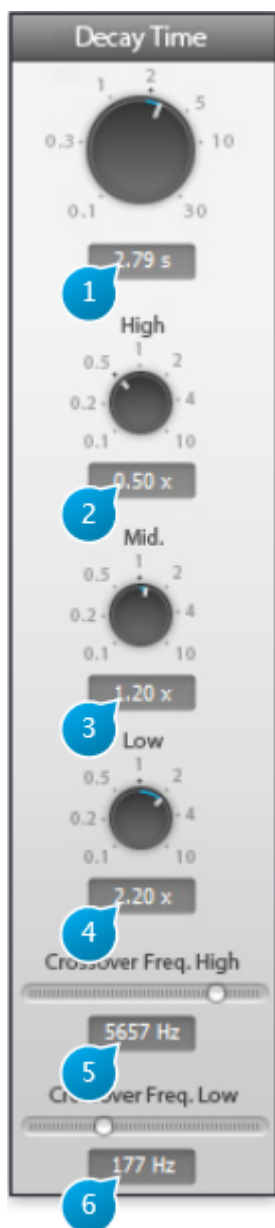
Understanding this time structure, and the associated listening feeling is really important for an enhanced approach to any reverberation system.

For a first quick approach, you can:

- set the reverberation time with the main decay time
- set the room basics characteristic with the room size meta-parameter

- set some room filters, to adapt the tail to your current needs.

3 Decay



3.1 (1) Decay time

This is the duration of the reverberation ‘tail’ in seconds, in other words how long it takes for the reverberated sound to vanish away. In more technical terms, this is sometimes referred to as the RT60 factor, which is the time at which the response of the reverberation to an input signal goes below -60dB of attenuation.

Please bear in mind that the master decay and high/mid/low controls are interactive, which means that the same audible result can be attained with different settings. This is intentional, as this allows you to get to the result faster and in a manner suited to your personal habits. Generally speaking, it might be more convenient to adjust the master decay time using the resulting sound as a guideline, then fine tune using high and/or low decay controls and leave the mid decay at the default setting. On the other hand, if you specifically to concentrate on the mids, for example to create a ‘hollow room’ sound, it’s easier to focus on the mid decay control, leaving hi and mid decay untouched.

3.2 (2) Decay High

Adjusts relative decay time of high frequencies specifically. It is expressed as a ratio of the global decay time setting explained above. Default setting is 0.5, meaning the high frequencies decay faster than the main decay time. This behavior is quite typical of natural spaces, where high frequencies are easily absorbed (by furniture, carpet, etc...) before the lowest ones. Roughly speaking, for a given master decay time, increasing this ratio increases the sense of liveliness of the acoustic space, whereas decreasing it deadens it.

The correspondingly affected band frequencies are determined by the Crossover freq. high control setting.

3.3 (3) Decay Mid

Adjusts relative decay time of medium frequencies specifically. Default setting is 1.

The mid range is where the human ear is most sensitive, and roughly corresponds to the frequency spectrum of the human voice. The correspondingly affected band frequencies are comprised in between the Crossover freq. high and low control frequency values .

3.4 (4) Decay Low

Adjusts relative decay time of low frequencies specifically. Default setting is 1.

In most real acoustic spaces, low frequencies reverberate freely in the sense that wall materials barely affect the low-frequency response, except if specifically adapted materials have been employed such as bass-traps and anything that acts as a tuned resonator. Generally room size and shape is what influences the low-frequency reverberation content the most, so one could say the default setting corresponds to a space with no low-frequency specific acoustic treatment.

The correspondingly affected band frequencies are determined by the Crossover freq. low control setting.

3.5 (5) Crossover Freq High

Sets the frequency above which reverberation time is determined by the Decay high setting, expressed in Hertz(Hz).

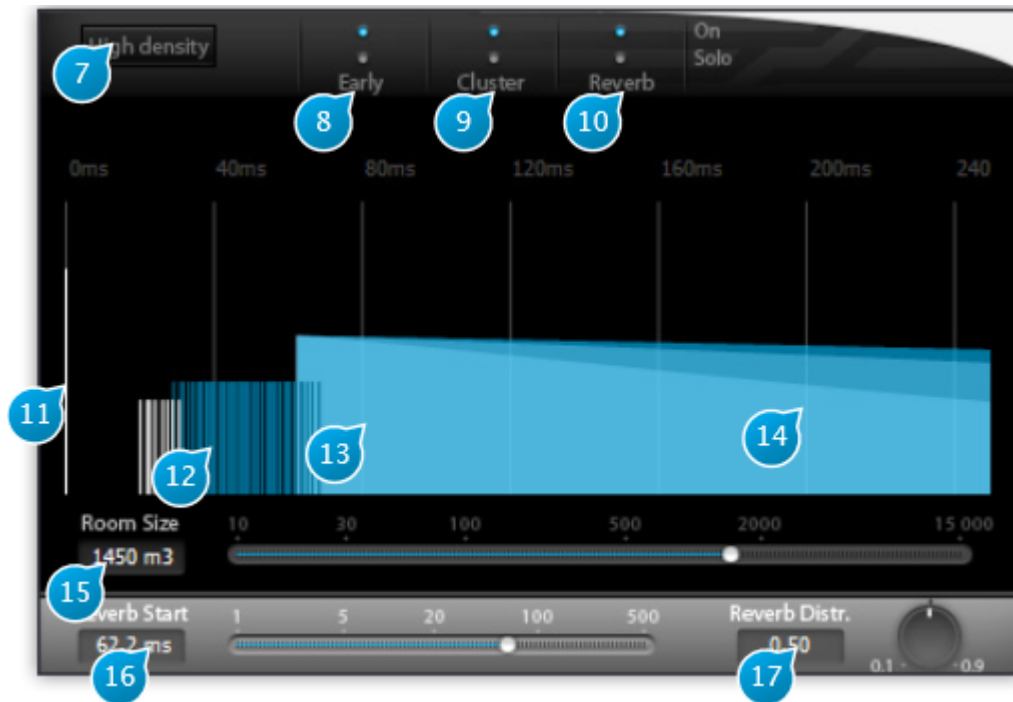
Default value: 5657 Hz

3.6 (6) Crossover Freq Low

Sets the frequency below which reverberation time is determined by the Decay low setting, expressed in Hertz(Hz).

Default value: 177 Hz

4 Time Structure Display



4.1 (7) High density

Toggles between standard and high density reverberation engine.

High density gives better quality at the expense of a little more CPU consumption, by increasing the size of the feedback network used to compute the reverberation.

Please note that this affects not only the sound quality of the reverberation, but can also change its character and tonal qualities somewhat, especially at certain settings, so, as always, you should let your ears be the judge as to which is best for a particular situation.

4.2 (8) Early On / Solo

These buttons are part of a mini-mixer console for the reverberation engine where each channel is fed by one of the reverberation sections.

Here these standard mute and solos controls belong to the early reflection section channel.

They allow you to isolate it or temporarily suppress a section of the reverberation so you can exactly evaluate the influence that this specific section has on the overall reverberation sound, for example when fine-tuning is required.

4.3 (9) Cluster On / Solo

Same as above, acting on the cluster section of the reverberation engine.

4.4 (10) Reverb On / Solo

Same as above, acting on the reverberation tail section of the reverberation engine.

4.5 (11) Direct signal

The grey bar at the start of the reverberation pictogram represents the direct sound send at the input of the plug-in. In the time structure of the reverberation, it is the first element that is heard.

4.6 (12) Early

Overall representation of the early reflections distribution.

Vertical bars roughly indicate at what time locations (horizontally) and levels (bar height) these early reflections occur.

4.7 (13) Cluster

See 12.

4.8 (14) Reverb

Shows a graphical representation of the reverberation tail part of the engine. The decay curves of the high, mid and low bands, which are controlled by the decay time settings, are superimposed in different colors and can rapidly be assessed and checked.

Also see 12.

4.9 (15) Room size

This parameter is a meta parameter that allow you to quickly perform an homogeneous set of parameters for the early reflexions part (early + cluster). Theses part are particularly important to achieve the “room” feeling of the desired space.

It adjust the time structure of the whole reverberation (early-min, early-max, cluster-min, cluster-max, reverb-start). This is a key control for quick settings, before a detailed fine tune with each parameters.

4.10 (16) Reverb Start

The time at which the latest part (diffuse part) of the reverberation section starts to be heard, in milliseconds. This is the delay between the dry signal and the beginning of the late reverberated signal. Please note that this setting does not affect the time characteristics of the early and cluster sections. It is however not possible to move the reverberation start time before the first early reflections.

4.11 (17) Reverb Distr.

Reverberation tail distribution controls the way in which reverberation tail ‘spikes’ are scattered in time.

5 Options



5.1 (28) Infinite

When activated, the signal is recirculated indefinitely inside the reverberation engine. Best suited for special effects such as “deep-freezing” the signal, or if you’re looking to create something a little less conventional than a fade-out for the end of your track.

5.2 (29) Air Roll Freq

Roll-off frequency for the air absorption simulation via a low-pass filter. Signal content above this frequency vanishes faster.

5.3 (30) Air Absorption

Simulates the frequency-dependent absorption of air, where high frequencies roll-off quicker than low-frequencies with respect to distance. You’ve most probably noticed this real-world phenomenon when you’re far away from a concert venue and only able to hear the bass, and gradually start to hear the whole mix as you get closer.

5.4 (31) Modal density

Scales the modal density with respect to the current setting, which is internal to the plug-in engine, and depends on other parameters such as reverberation time, etc.

The modal density governs the frequency “smoothness” of the verb engine. Increasing this setting reduces the graininess of the reverberation. Adjust to taste, depending on the source material and desired result.

5.5 (32) Panning

Virtual source panning direction offset relative to input channels, in degrees.

In mono-to-stereo mode, this acts as a standard pan control, adjusting L/R direction of the source.

In a stereo-to-stereo channel configuration, this controls allows one to gradually remap the input

channels to each virtual source. In N-to-N surround configuration, the input channels are gradually remapped to their closest neighbors, in a circular, carousel fashion.

5.6 (33) Width

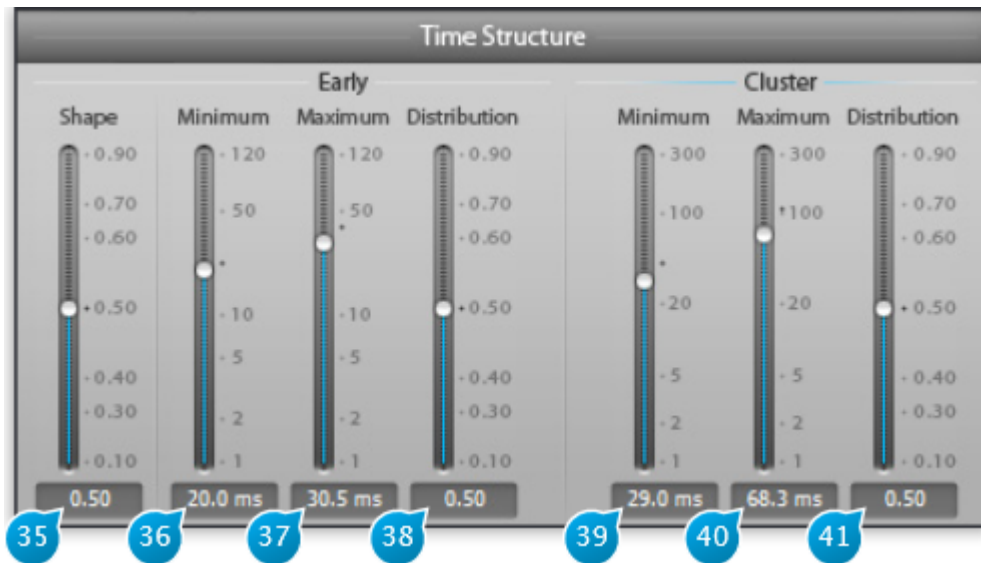
Panning width of the input channel - virtual source remapping described above.

5.7 (34) Diffuseness

Determines the spatial width of the reverberated signal part, one could also say it changes the directional information of the reverberation, or the ability of the listener to locate the spatial origin of the this signal. In a real-life space, this would corresponding to how non-symmetric, irregular and complicated the shape of the room would be.

When engaged, separate cluster and tail reverberation push-buttons determine which section is affected by the diffuseness parameter. A zero setting equates to maximum localization while a 100-percent setting gives full diffuseness and no localization.

6 Time Structure Settings



6.1 (35) Early shape

Governs the amplitude rise or fall of early reflections. The default setting of 0.5 corresponds to early reflections all having the same level. This mimics an acoustical space where reflective surfaces are all located at roughly the same distance to the listener.

Below 0.5 early reflections decay with time, above 0.5 they rise with time. Early reflections of decreasing level would be typical of a space where most of the reflective surfaces are grouped at a range closest to the listener.

6.2 (36) Early Min

Early reflections minimum time, i.e. the time at which the early reflections start to appear, in milliseconds. This is the analogous of the ubiquitous “pre-delay” setting found on most reverberation processors. It represents the time between the direct sound and the first early reflection.

6.3 (37) Early Max

Early reflections maximum time, i.e. the time at which these cease to appear.

6.4 (38) Early Distribution

Early reflections distribution. Determines the way early reflections are scattered in time, inside the Early Min. / Early Max. interval. The default setting of 0.5 corresponds to regularly spaced reflections, above these are more grouped towards the Early Max. value, and vice-versa.

6.5 (39) Cluster Min

Cluster minimum time. See Early Min.

> Please keep in mind the cluster is fed with the input of the early reflections processor section, as is shown accordingly on the display.

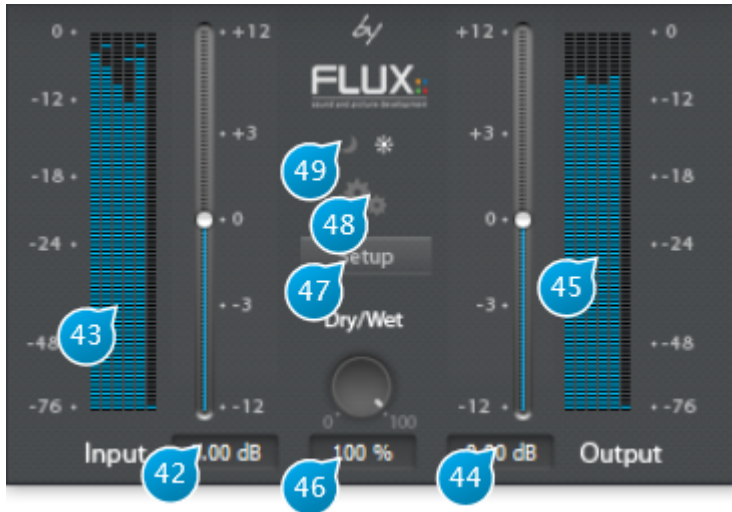
6.6 (40) Cluster Max

Cluster maximum time. See Early Max.

6.7 (41) Cluster Distribution

Cluster distribution. See Early Distribution.

7 Input Output Section



7.1 (42) Input Gain

Adjusts the level of the signal fed to the plug-in, in dB increments.

7.2 (43) Input level meter

Shows the current level of the input signal after applying input gain, in dB FS (decibel Full Scale).

7.3 (44) Output Gain

Used to trim the output signal and possibly avoid any overloading of the signal in the rest of the signal-chain.

7.4 (45) Output level meter

Shows the current level of the input signal after applying output gain, in dB FS(decibel Full Scale).

7.5 (46) Dry/Wet

When used as insert effect, one can dial the right amount of “wet”, reverberated signal with respect to the “dry”, untreated input signal.

The default 100-percent wet setting is mostly intended for the typical and preferred use in a send-effect configuration.

7.6 (47) Setup

Toggles the display of the routing matrix, where the user can adjust the routing between input channels and virtual sources.

7.7 (48) Setting

Gives access to a sub-menu where you can either select the I/O configuration, namely the input channel count followed by the output channel count, for hosts that support dynamic I/O configuration or display the credits page.

The exact I/O combinations available depend on your actual audio hardware and host configuration.

7.8 (49) Day - Night

Toggles between two interface schemes, which, as the name implies, are best suited to high or low light environments respectively. In a dimly-lit studio environment, switching to the nighttime scheme with its darker color palette and lower contrast will minimize eye-fatigue when doing long sessions.

8 Preset Section



8.1 (50) Save

Saves a snapshot of the current settings for future use.

Short description and assorted comments can be provided, which comes in especially handy when sharing presets with other users, when the preset is part of a large preset bank, or to identify the author and source.

Entering a descriptive keyword is a good practice to be able to quickly sort your presets, according to character, the type of space they simulate (e.g. hall, room, etc.), and the intended usage (e.g. voice, percussion, guitar, etc.)

A preset can be locked to prevent any further editing.

To re-save your preset under a new name, open the preset manager by clicking the corresponding (A/B) preset slot, then select New, enter a name for your preset, and finally press Save.

8.2 (51) Recall

Recall the settings from the currently selected preset, overwriting any current settings of the plug-in. The sub-menu which appears allows to recall at your choice:

- all parameters
- all parameters but setup: intended for when your particular speaker configuration is different from that of the preset's author (typically stereo)
- all parameters but setup and dry/wet mix: useful in a mix setting when comparing and choosing presets

8.3 (52) Copy B

Copy current settings to the second parameter slot (B). To try out a variation of the current settings without erasing the reference, press this button, switch to B and adjust your parameters of choice, then switch or morph between A and B.

When copying a preset to a slot, the morphing slider will automatically fly to the corresponding slot.

8.4 (53) Preset Name

Displays the current preset name, if any. Clicking the associated button (up&down arrows) brings up the preset manager.

8.5 (54) Morphing A B

Gradually morphes parameters from A to B slots.

The parameter set associated with the current morphing slider position can be saved as a preset. In addition, when the morphing slider is in an intermediary position, any edit made to a parameter switches the slider back to slot A or B, whichever is closest to the current position.

8.6 (55) Automation

Enabling the Automation control switch makes the morphing slider exposed and available for automation read.

When engaged, keep in mind only the morphing slider value is used for automation, and other parameter values are ignored. This behavior is intended and necessary to prevent any parameter conflicts that would otherwise occur.

As a consequence of this, you need to make sure the Automation switch is engaged when mapping the morphing slider mapped to a control surface hardware knob or slider. On the opposite, when not engaged, the plug-in will listen for any parameter automation, except the morphing slider.

9 Preset Management

9.1 From the Plug-in interface

9.1.1 A-B Sections

A plug-in features two preset sections : A & B. Clicking on the slot of a specific section reaches the shared preset bank.

From the preset management window you can select the preset you want to recall in the specific preset section.

9.1.2 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.

9.1.3 Recall

Once a preset is selected from the preset list it must be explicitly loaded into the section A or the section B by using the recall button. A preset is effective only after it has been recalled. Double-clicking on the preset name from the list, reloads the preset into the selected slot.

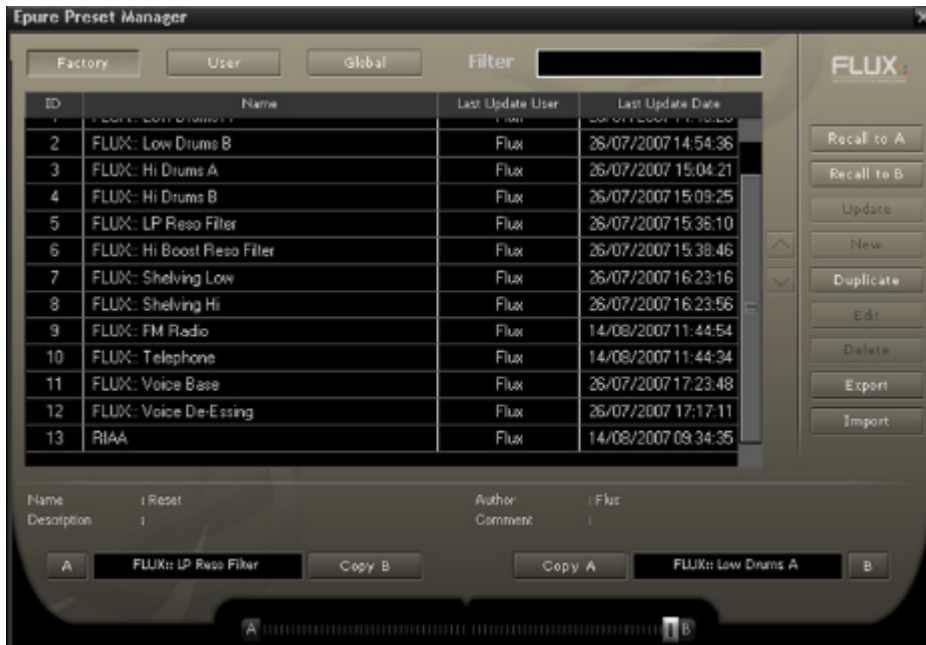
9.1.4 AB Slider

This horizontal slider has no unity nor specific value display. It allows to morph current settings between two loaded presets. A 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 save as a new preset.

9.2 From the Preset Management Window

The Preset Management Window features three preset banks: * The Factory bank gathers presets that can't be edited by users. * The User bank is dedicated to the users presets. * The Global bank features presets for A, B and morphing sections. A single global preset includes A and B section content and the morphing slider position.

A Preset can directly be recalled into the preset section selected by the morphing slider position, by double-clicking on its name on the list. The preset lists can be filtered. This filter is applied to any preset information such as name, description, author, comments or key words.



9.2.1 Recall A

recalls the selected preset into the corresponding section.

9.2.2 Recall B

recalls the selected preset into the corresponding section.

9.2.3 Copy A and Copy B

buttons allow to easily create a variation around a preset.

9.2.4 Update

allows to save the current settings for the selected preset.

9.2.5 New

creates a new preset in the list.

9.2.6 Duplicate

creates a new preset in the list from the selected one.

9.2.7 Edit

gives access to the specific windows which allows to change preset name, description, key words.....

9.2.8 Delete

suppresses the selected preset.

9.2.9 Export

creates a file reflecting the content of the preset bank.

9.2.10 Import

adds existing presets into the preset bank.

9.2.11 Ordering arrows

orders the presets into the list.



Please Provide Preset Informations

Name
My Old Compressor

Description
4 ratio

Comment
Bass

Keyword
Electric bass, Kick

☒ Locked

Ok Cancel

The preset protection if engaged, allows only its original modification author to uncheck and edit. So you can protect your presets in a multi-user configuration. Protected presets can only be modified using the session of their creator. If used in another user session they can only be imported or deleted.

10 Credits

Spatialisateur and Spat~ are trademarks of Ircam and Espaces Nouveaux.

Design of digital signal processing algorithms and implementation in Max: Jean-Marc Jot (Espaces Nouveaux / Ircam).

Objective and perceptual characterization of room acoustical quality: Jean-Pascal Jullien, Olivier Warusfel, Eckhard Kahle.

Additional contributions by Gerard Assayag, Georges Bloch, Martine Marin, v@ronique Larcher, Guillaume Vandernoot and Khoa-Van Nguyen.

C++ development code : Thibaut Carpentier , Remy Muller; additional contributions: Gael Martinet.

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Collection Manager: Frederick Rousseau

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IRCAMTOOLS

Collection Manager for IRCAM: Frederick Rousseau

Collection Manager for Flux:: Gael Martinet

IRCAMTOOLS SPAT, VERB

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FLUX::

Head software engineering: Gael Martinet

Developpers: Gael Martinet, Samuel Tracol, Siegfried Hand

Designer: Nicolas Philippot

Contributions: Lorcan McDonagh

11 Specifications

11.1 Availability

IRCAM Verb is available in:

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

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

11.2 Processing

IRCAM Verb provides :

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

11.3 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.

11.4 Software License Requirements

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

11.5 Compatibility

All major native formats are supported

11.5.1 Windows – 10, in 64 bits only.

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

11.5.2 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*

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