Aphro-V1.5

Digital reverb & fx processor..

English version
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Summary

Specifications :

APHRO-V1.5 DX Plug-in for PC

PC configuration	Intel Pentium© III 500 MHz (min).
Memory configuration	HD 10Mo Free, RAM 64Mo (min)
Display configuration	Display config. : 800x600 color 16bits (min).
Input pin	mono or stereo
Input pin (data type)	16bits integer or 32bits FLOAT IEEE32
Input pin (frequencies accepted)	All above 11025Hz
Output pin	As Input

frequencies	**44.100Hz**	**48.000Hz**	**88.200Hz**	**96.000Hz**
Speed (stereo)	15% (CPU*)	18% (CPU*)	30% (CPU*)	35% (CPU*)
Memory access (stereo)	11.4 Mbyte/s	12.5 Mbyte/s	22.8 Mbyte/s	25.0 Mbyte/s
Memory used (stereo)	4.7 Mbyte	5.1 Mbyte	9.4 Mbyte	10.2 Mbyte

Features	
Factory preset bank	internal presets. : **Realistic series 1** (73 presets)
Main interface	delay, gain wet dry, global size, time, damping.
CPi	Color Panel Interface (color and parametric equalizer).
ACi	Algorithm control interface (Early reflections et decay control)
Preset and Bank	Interface for presets and banks management.

(*) Max CPU time measured with Steinberg Wavelab© 4.0 for a 16bits stereo sound on INTEL PENTIUM© III 500Mhz.

APHRO-V1.5 VS3 Plug-in for Pyramix© Merging technologies

Pyramix configuration	Mykerinos MB2 (min).
Memory configuration	HD 10Mo Free, RAM 64Mo (min)
Display configuration	Display config. : 800x600 color 16bits (min).
Input pin	mono or stereo (channel selectable on multibuses)
Input pin (data type)	32bits FLOAT IEEE32
Input pin (frequencies accepted)	All above 11025Hz
Output pin	As Input

frequencies	**44.100Hz**	**48.000Hz**	**88.200Hz**	**96.000Hz**
Speed (stereo)	13% (CPU*)	16% (CPU*)	16% (CPU*)	16% (CPU*)
Memory access (stereo)	11.4 Mbyte/s	12.5 Mbyte/s	11.4 Mbyte/s	12.5 Mbyte/s
Memory used (stereo)	2.9 Mbyte	3.2 Mbyte	2.9 Mbyte	3.2 Mbyte

Features	exactly like native version (see Aphro-V1.5 Direct-X - VST)

(*) Max CPU time measured with Mykerinos MB3.

SUB SAMPLING TABLE (and allocated memory) **:**

Sample Rate	Aphro-V1.5 Quick	Aphro-V1.5 Normal	Aphro-V1.5 96
44.100 Hz	**2 :1** (1.47 MB)	**1 :1** (2.93 MB)	**1 :1** (2.93 MB)
48.000 Hz	**2 :1** (1.60 MB)	**1 :1** (3.19 MB)	**1 :1** (3.19 MB)
64.000 Hz	**4 :1** (1.06 MB)	**2 :1** (2.13 MB)	**1 :1** (4.26 MB)
96.000 Hz	**4 :1** (1.59 MB)	**2 :1** (3.19 MB)	**1 :1** (6.38 MB)
128.000 Hz	**8 :1** (1.06 MB)	**4 :1** (2.13 MB)	**2 :1** (4.25 MB)
196.000 Hz	**8 :1** (1.63 MB)	**4 :1** (3.25 MB)	**2 :1** (6.51 MB)
256.000 Hz	**8 :1** (2.13 MB)	**8 :1** (2.13 MB)	**4 :1** (4.25 MB)
384.000 Hz	**8 :1** (3.19 MB)	**8 :1** (3.19 MB)	**4 :1** (6.37 MB)

Introduction :

Aphro-V1.5 is a real time Digital Effects Processor. Second element of a high quality effects processor series called **Aphro-Vx**, **Aphro-V1** is specially created to simulate sonorous atmosphere and room effects, in a realistic way.

The handling philosophy is made simple and practical thanks to a wide range of preset, which requires the user to select a preset matching the best desired effect, and then to use the different interfaces in order to adjust it, according to his conveniences.

Grouping parameters by theme, gave us the idea of creating a modular and ergonomic user interface. The Main interface (front of our « engine ») enables you to control the regular parameters (pre-delay, mixing, reverb size...) and also to reach other Dialog Box : CPi and ACi and also the preset bank manager by clicking on the LCD.

Quick or not ?

Aphro-V1.5 includes two versions on native PC : **Aphro-V1.5** and **Aphro-V1.5 Quick** (a **Q** on the right bottom of the main interface allows you to make the difference.). For VS3 Version Pyramix, Aphro-v1.5 is delivered in 3 plug-ins : **Aphro-V1.5**, **Aphro-V1.5 Q** and **Aphro-V1.5 96**.

The difference between all version stand internally, in the sub-sampling processing according the samplerate of your audio digital workstation. On native PC, the **Quick** version always works on half sampling rate (half bandwidth if you prefer). As it is described on the schema right below, the effect only (WET signal) is involved by this operation.
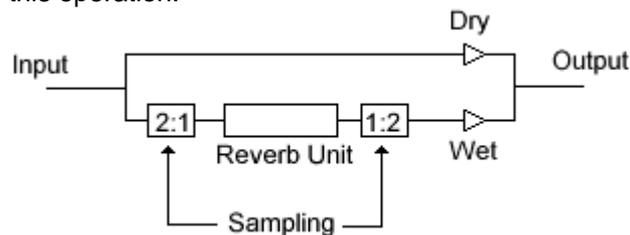
this process allows to divide process time by 2 (but also divides the effect bandwidth by two, sound being less acute.), this could be useful when limited by computer power. Moreover, when working at 96Khz, the quality of the effect calculation at 48Khz (half of it) will remain more than acceptable, for a low CPU resource.

Interface Philosophy.

The **Aphro-V1.5** interface is quite alike the one we could have on a hardware unit. One of the ergonomic problem recently encountered concerned especially buttons behaviors : Are they physical buttons that could be only modified by the user ? Or are they cursors for parameters setting and visualization ?

By default, all your settings are saved in the current **preset**. Changing a **preset** displays all buttons positions, according to new **preset** parameters. Well, buttons are used to modify and display parameters value for each **preset**. If this method has got its own advantages, keep parameters (level and color for example) from a **preset** to another is made impossible...

That's why now, all dialog boxes have a contextual menu (right click on the interface) which will allow you to require each button how to keep its own value from a **preset** to another.

Main Interface.

The main interface, allows to set the regular parameters of the effect : selecting **presets** with both "up" and "low" buttons, adjusting the **wet/dry** balance, setting the reverb size etc... Also the main user interface let you open preset bank manager (Click on LCD) or activating an interface by pressing one of the "**CPI**" or "**ACI**" buttons.
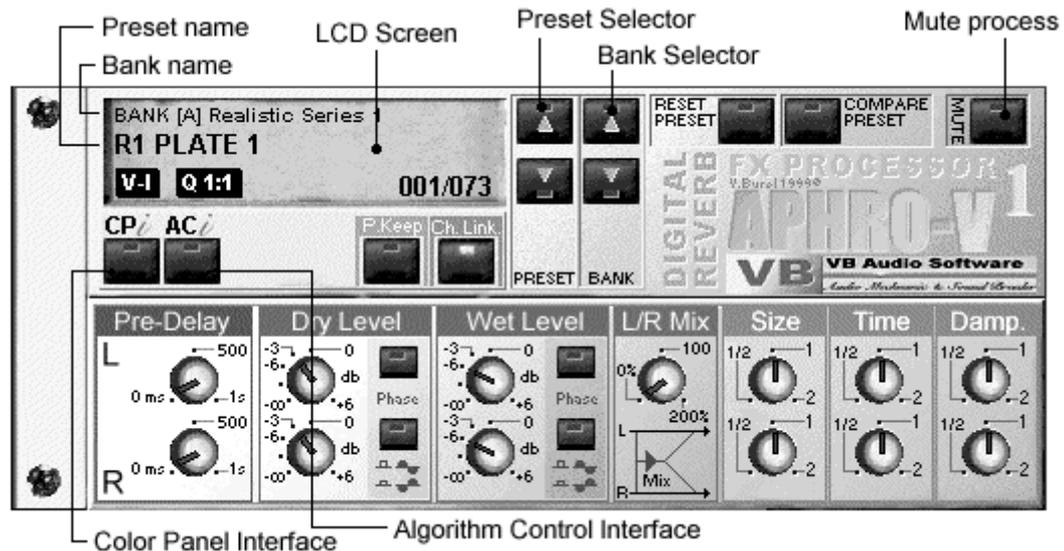
LCD Display :

The LCD display shows current information about the reverb effect running. At the top, you can see the selected **preset** bank, underneath in bold, the name of the preset (a bank - **BANK**- is a group of **presets**, a **preset** is a group of parameters which entirely defines a reverb effect).

On the lower left side of the screen, **V-I** indicates the type of **preset**. The parameters are compatible with several reverb algorithms. **APHRO-VI** is able to use and read a preset initially made for **APHRO-V1.5** or **APHRO-V2** but the sound may be different. In any case, the reverberation will be different. That's why **V-II** will appear on the screen when the selected **preset** does not fit with **APHRO-VI** format. On the contrary, **APHRO-V1.5** is able to deal with presets coming from **APHRO-VI**.

On the lower right side, you will find the reference number of the activated **preset** in the current **bank** and the total number of **presets** in the current **bank** (xxx/xxx).

Interfaces:

Underneath the LCD display, you will find all the different specialized user interfaces. To see these interfaces, just press the right button (**CPi, ACi**). This multi-interfaces system provides a minimal use of screen space and a thematical grouping of controls and parameters. The **CPi** dialog box allows you to tweak the color of the reverberation by different type of equalizer. The **ACi** Dialog Box will let you adjust the early reflections and echo-decay.

A third interface may be selected by "clicking" directly on the LCD display. A dialog box specialized on managing **banks** and **presets** will then appear

Reset Preset:

Each preset has two memory locations, a static one which can't be modified and a dynamic one which will memorize all your modifications. For each preset, you can go back to the initial state (the one included in the static memory) by pressing on the **Reset/Preset** button.

Compare:

Each time you use/select a **preset**, it is copied in a small memory block which can be recalled by pressing on the **compare button**. The compare function enables you to exchange the current and previous state of the preset... You can use it as many times as you want in order to compare the "before - after" **presets**.

CH LINK :

Defines the way you manage stereo knob on the main user interface : If this button is pushed on (On per default), the both channels are linked. Modifying the Left Gain for example, involve the same modification on the Right channel and vice versa. (If the plug-in works in Mono, only the Left channel parameters are used by the processing).

P.KEEP (Paramaters Keep)

This button makes appear a popup menu containing the list of the parameters on the main user interface : Pre-delay, WET, DRY, L/R Mix etc... By checking items in this menu, you can decide to keep the state of the knob related to this parameter when changing preset. This very practical option allows you, for example, to keep the WET / DRY balance from a preset to an other and then compare them quickly.

PGi (pre-delay and gain interface)

This interface allows you to manage the primary parameters of the effect. For the Left an Right channel. Note that you can move knob or edit parameter by different ways (right click on a knob to make appear the option popup menu).
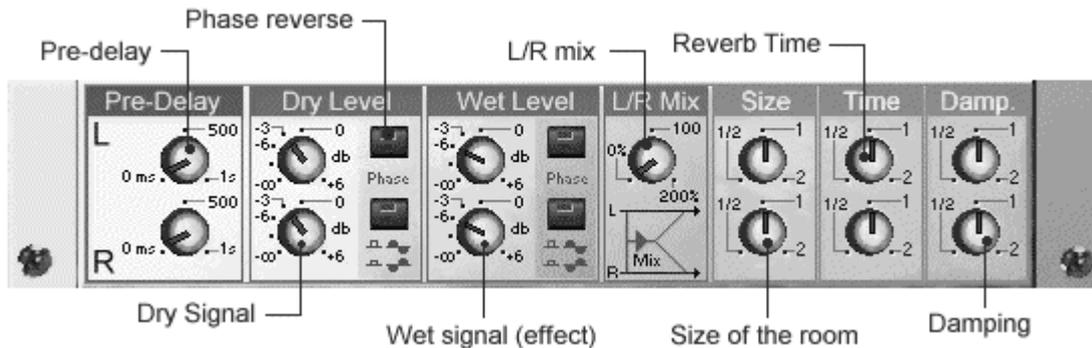
Pre-Delay : time before effect.

Because a reverberation always occurs after the direct sound, the **pre-delay** is an essential parameter. It will help define the lapse of time which partly characterizes the size of the hall.

The reverberation effect is caused by the reflection of various sound waves coming from the original source, reflected from the walls or surfaces around it. A lapse of time is needed for the sound to reach the nearest reflectors before we can hear the reverberation.

We must also wait for the sound to feed back to the hearer. As demonstrated in the opposite diagram, the pre-delay time is not easy to establish with only parameters concerning the size of the hall, the respective positions of the sources as well as the hearer's position are needed.

pre-delay = reflected - direct

pre-delay = reflected - direct

Usually we want to find the shortest reflections in order to gain the most realistic pre-delay value.

For a realistic adjustment of this time parameter, we need only remember that it has to last all the more since the hall is vast. Considering sound speed (about 330m/s) we can roughly determine the pre-reverberation time with the early reflections distance and consider that 10 ms equals about 3.3 m.

Pre-Delay and Spatialization Effect:

If you make both channels independent by pushing up the **channels Link** button, you can severally define the right **pre-delay** time from the left one. This allows us to induce time-shifts between the left and right channels.

If this interval is less than 5 ms, we realize a setting effect of the reverb sound in the hearer's azimuthal plan, especially concerning medium frequencies.

This effect is not very stable as it depends on the nature of the original sound, its stereophonic effect, its spectral density...

With this method, we can give the hearer the impression (especially if he has headphones on) that he can hear the reverb sound behind him, on the left side or on the right side by inducing slight time-shifts between the left and right channels.

If the interval increases (10 ms), the effect becomes diffuse and gives the hearer the impression the sound comes from everywhere and nowhere at the same time (impression all the more important since the reverb effect is long).
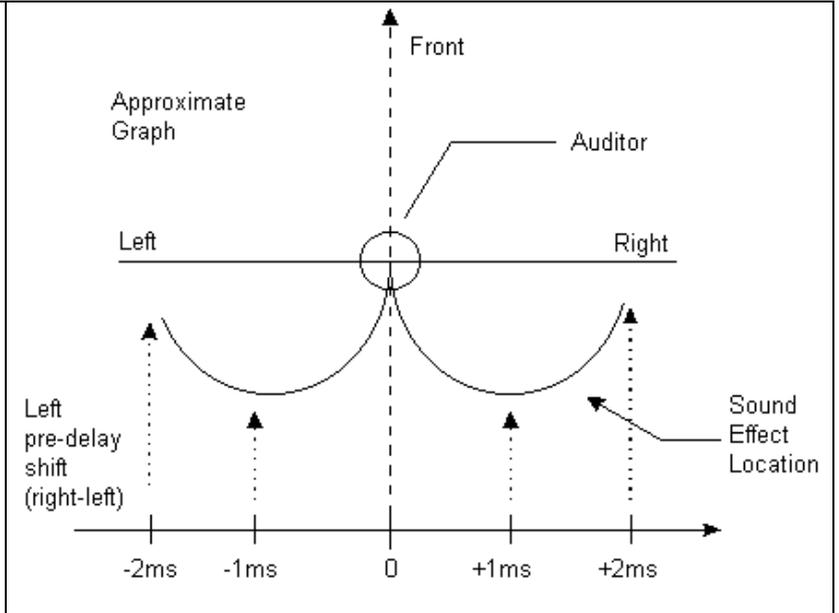

Dry & Wet Gain:
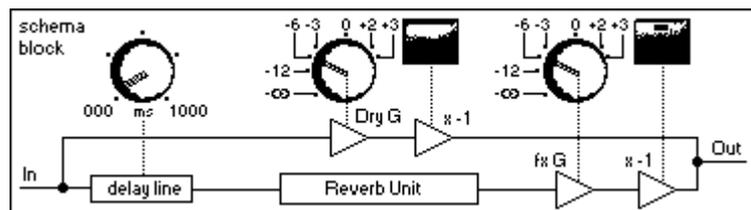
These two parameters will enable you to balance between direct sound (**Dry**) and reverb effect (**Wet**). The aim is to adjust both parameters (**Dry** and **Wet**) in order to obtain the expected ratio and the optimal output level (what is to say the output level as close as possible to the 0db level, without clipping).

In order to help you in this task, and although taking into consideration that the output signal can be limited by many effects, you should know that all presets have been set up to obtain a 0db input/output ratio. In fact, the pure effect has the same level as the input signal.

The output of the effect is composed of the addition of the input signal (**Dry**) and reverb signal (**Wet**). **0db + 0db ¹ 0db** but **0db + 0db < +6db**. This is why we sometimes set both gains on -6db. This theoretical value allows to avoid clipping (input signal <0db). However it does not correspond to all using cases. For a not too condensed sound, the main output level can be really too weak.

Gain reaction of the reverb effect.

Let's consider opposite cases of impulsive signal and permanent signal. (Taking into account that the **Dry** and **Wet** levels are both set on **0db**).
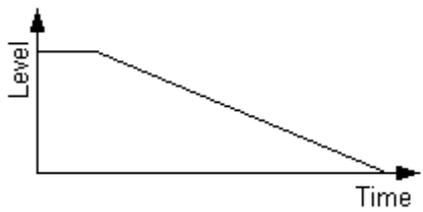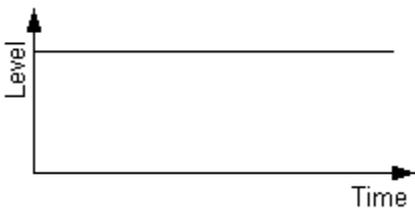
For an impulsive input signal (drum sound for example), the direct sound (**Dry**) does not overlap the reverb sound (**Wet**). The Dry one , when it is not void, is summed up to a null Wet signal. The reverb effect appears always after the direct sound, thus if the direct sound is brief enough, we will hear both **Dry** and **Wet** sounds at different times. Also, the reverb sound will be added to a null direct sound (**Dry**) because over. In that case, **0db + 0db = 0db**.

In an impulsive system, the reverb level has time to decrease normally according to the selected reverberation length.

In a permanent system, (an organ sound for example), we get the following results :

The addition of signals (**Dry** and **Wet**) is greater than **0db** because in this case, signals are overlapping each other within time. Moreover, the reverb effect level has not the time to decrease, because there is always a sound to feed the reverb effect and give it the aspect of a constantly volumetric signal. It is only if the input signal drops to zero that the reverb effect will begin decreasing.

If the original sound is very dense, as with an intense bass/drum rhythm for example, you can consider, in case of long reverb effects, that the reverb level remains more or less constant in time. You should take this into account when setting the **Dry** and **Wet** levels.

Phase Inversion:

The input signal (**Dry**) like the reverb signal (**Wet**) can be displaced of **180°** (channel by channel) by pressing the push buttons connected to the rotary ones.

If you displace one of the channels of the direct sound (**Dry**), you obtain a "**quasi-surround**" effect with an attenuation of low frequencies due to the phase inversion. If you displace one of the reverb effect channels (**Wet**), you obtain then an equal effect, but only on the reverberation (in the case of a short reverberation PLATE), or also a softened, diffuse, reverb effect, in the case of a long reverberation (HALL)...

NB : This effect depends on the original nature of the sound. For example, for a minimum effect, the original sound must be stereophonic, but does not need to possess two totally independent channels...

L/R MIX Knob

All reverberation effects of the Aphro-Vx series are truly stereo. The generated reverb on left channel is fully independent from the reverberation generated on the right channel. Then this reverberation effect respects the original mixing and the stereo image of the direct sound (Original Input Signal). However In some cases, this behavior might be not adapted, that's why the L/R Mix control allows you to add a monophonic sound (composed be Left and Right input signal), on the Left and on the Right channel in the

same time. The Reverberation becomes more dense and the reverberated field more homogeneous, especially if the sound comes exclusively from one channel only : Left Channel or Right Channel exclusively.
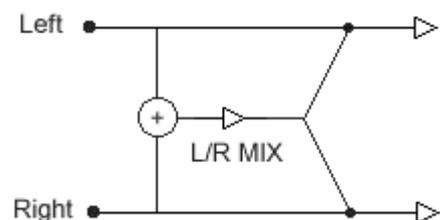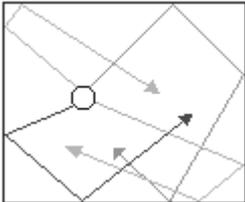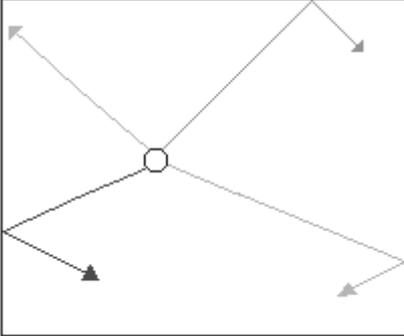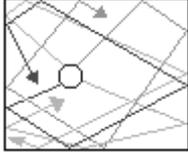
Reverberation Time (RT) :

The reverberation time is a parameter difficult to handle since the perception of this duration depends on many other parameters. A Bright Reverberation or a powerful one in the medium frequencies will appear longer that the same cut at 1Khz. Also the treble attenuation (damping) is a critical point in the way you will feel the duration of the reverb. Finally the type of the direct sound (input signal), its spectrum identity and it's density (percussive sounds like a single drum or more continuous sound like a philharmonic orchestra) make you appreciate the reverb duration differently.

The Time or Room Size control of the usual reverberation effect tend to misrepresent widely the original **preset**, even to make it incoherent without having the possibility to change something here. On the opposite, **Aphro-V1.5** permits an accurate and a precise management in a more direct relation with the physical phenomena involving changes and modifications of the Room Size or reverberation duration. This achievement has been made by considering not only one parameters but two : the size of the room and the resonance rate.

SIZE (of the room)

On a given preset, this knob allows you to modify the size of the room. Means to move away or to move nearer the walls of the room. This operation is made by keeping the time constant : the energy of the current reverberation is saved. Like a gas endowed with a given pressure, expanding the space containing this gas involves a pressure decreasing even when decreasing this space gives a pressure increasing. The reverberation, for a given resonance, has a similar behavior. Increasing the space room makes lose energy, means losing sound volume. On the opposite, makes the room smaller and smaller increases the number of reflections, so increase the resonance power, so finally means getting a louder effect.

Let's explain more to let you understand how Aphro-V1.5 manages this parameter.

| For a given preset, we consider a virtual room where there is an amount of reflections according an amount of energy.	Increasing the size of the room will increase the distance to cover by the sound before being reflected by the walls. Consequently the sound will lose more energy before being reverberated and then won't be able to provide the same audio volume than in a smaller room.	Naturally, decreasing the size of the room will bring more reflections, more energy and then more audio volume.
 Initial room	 Bigger Room	 smaller Room

TIME (of the reverberation)

The reverberation time is here considered as a resonance rate or a reflections amount, or even like a reverberation energy. Increasing this rate means increasing the energy of the reverberated field for a given room with a given size. To argue again by analogy with gas behavior, let say that it's like increasing the pressure in a given space without modifying the size of this space (by heating or by adding an amount of gas).

This parameter (TIME), in relation to the parameter of the size of the room (SIZE), allows to create consistent spaces or even unrealistic, but in a precise way and with the maximum control. Or we can decide to tweak the reverberation time according the size of the room in a realistic way. Or we can create impossible spaces and atmospheres, for example by setting a disproportionate reflections rate to the size of the room.

CPi (Color Panel interface)

This interface permits to control the reverb effect "**colour**" for the left and right channels. In the **Aphro-V1.5** version, this Dialog Box include an other part containing 2 bands parametric EQ for each channel, placed on the output of the reverberation effect.
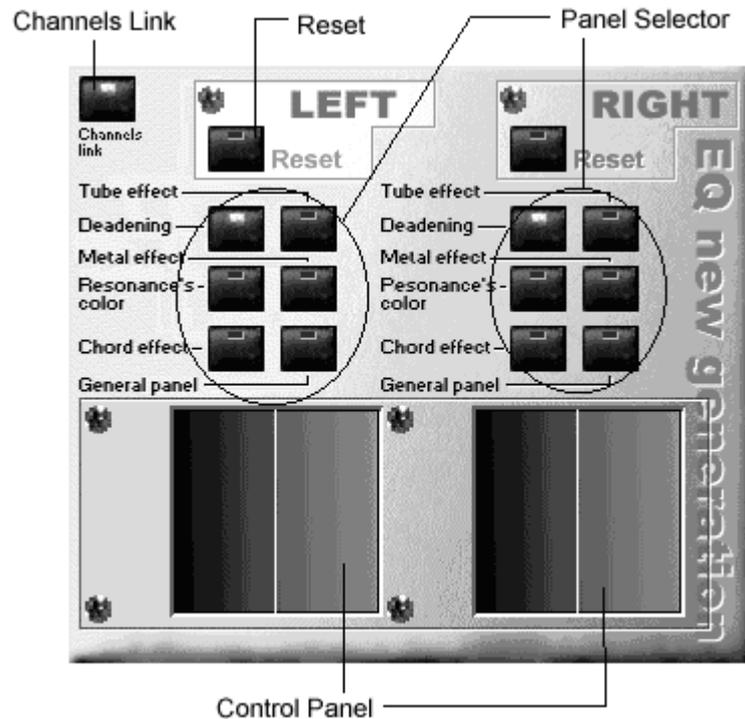

Channels Link:
Channels link defines the running mode of the push buttons and the control panels as regards in this dialog box. If the button is pressed, both channels are linked. A change on the left channel leads to the same change on the right channel and vice versa. (If the **plug-in** is on monophonic mode, only the left channel parameters are used by effect).

Panel Selector:
The six control panels (**Deadening, Resonance's Color, Chord effect, Tube effect, Metal effect and the General panel**) allow to control the same filter, but on a different way. You should be aware of how each panel is supposed to react before selecting the panels, which will give the desired effect.

Deadening panel:

The shadow panel can be compared to a **shelving**, very efficient in the elimination of the medium and high frequencies.
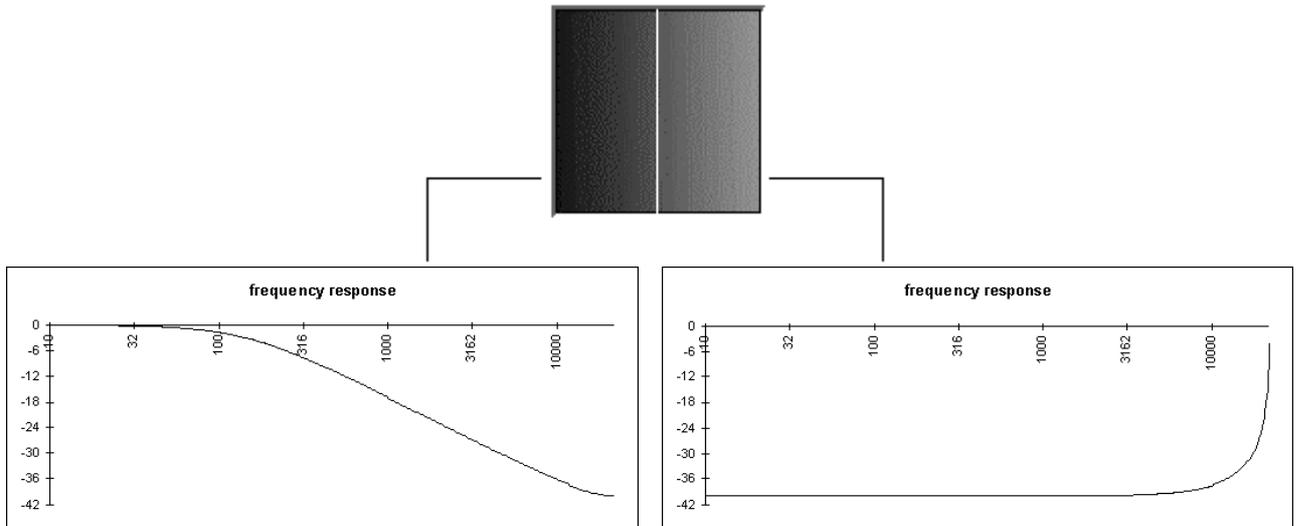
Resonance's Color Panel:

This control panel is one of the most efficient in the checking of the reverberation color modification (resonance of the hall) and in the addition of more or less pleasant frequencies in order to simulate some kind of successful reverberations. Realistically speaking, each hall, each place, has its own frequency response. This is why the Resonance's color Panel helps adapting the Hall effect to the user's convenience...

The **Resonance's Color Panel** is a subjective tool that should not be avoided by the hearer to obtain a good result. « So, It's up to you! »
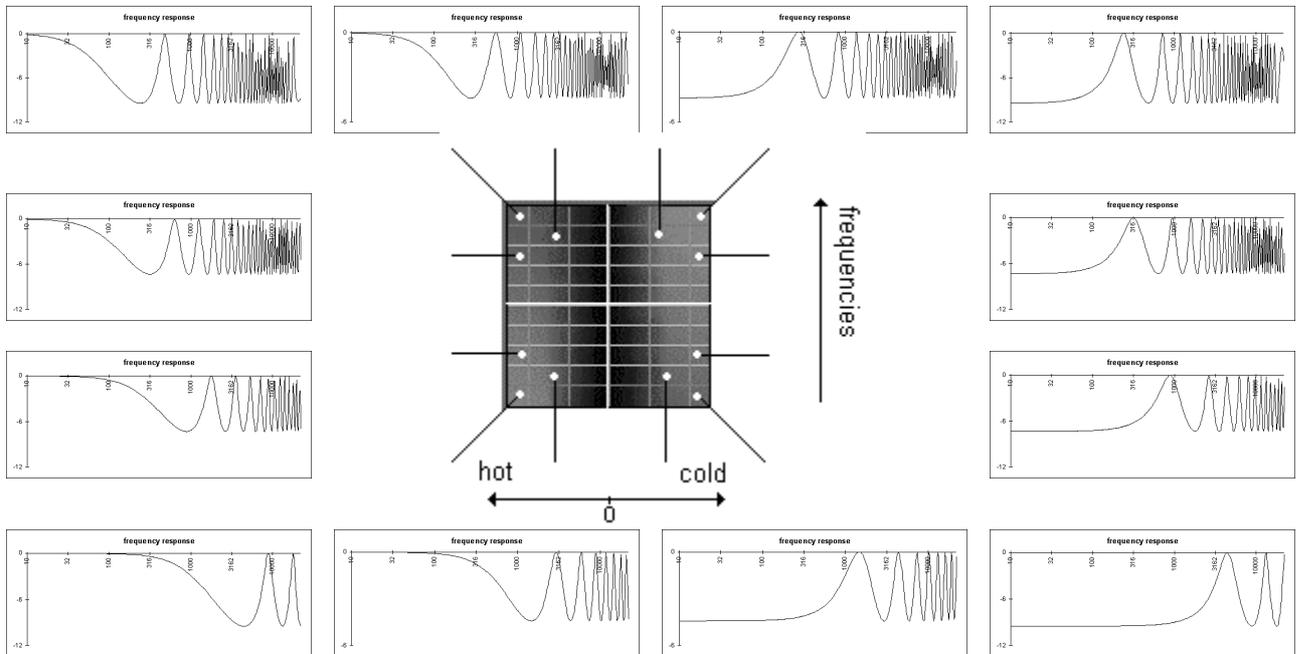
Chord & Tube effect

You can appreciate the originality of these two control panels: on the one hand the **Chord effect** is especially used to make drums sounds sound like those of a guitar, the **Tube effect** on the other hand makes these sounds sound like a draft passing through a tube.

Be aware that if you move the mouse down on the panel, the filter changes into a resonating filter for 2 or 3 frequencies. It sometimes helps getting an interesting equalization effect.
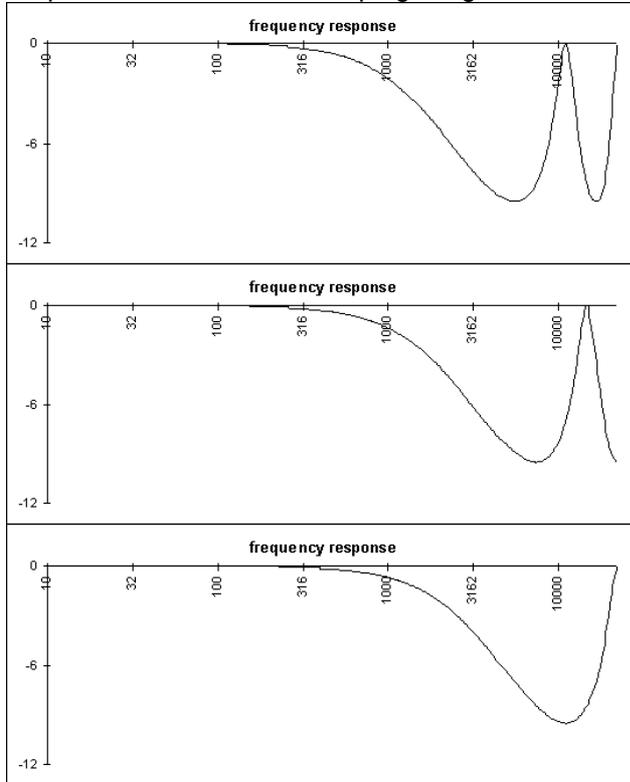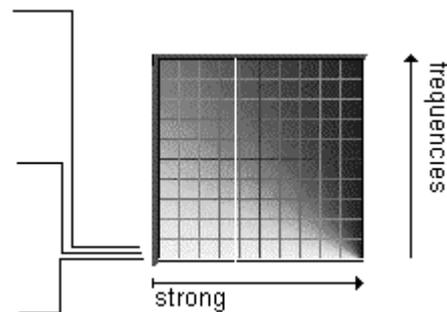
Chord effect

More selective than the low pass filter connected to the **deadening**, it also allows frequency peaks to appear in the high sounds.
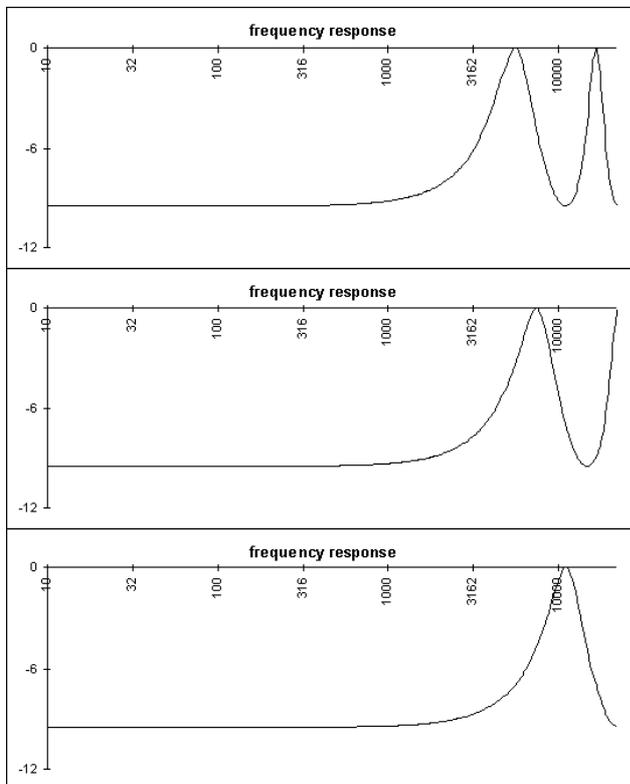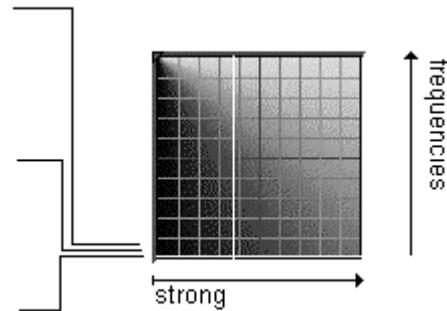
Tube effect

More selective than the high pass filter connected to the **deadening**, it also allows frequency peaks to appear in the medium sounds.

Metal effect
This effect gives a slight tremolo in the medium and bass frequencies. It is better if you can carefully listen (**Dry** muted) this effect before using it, because it is difficult to hear it on long reverb effects.

General Panel
This control panel allows a complete parameter input of the filter, but is not precise enough. If the first five control panels modify only a limited and selective range of the filter parameters, the **General Panel**, indeed, controls the whole parameters range. The first five control panels are then included in this latter one.

Output parametric Equalizer
Aphro-V1.5 now includes 2 usual parametric EQ cells on the output of the effect. This allows to control the color of the reverberation in a better way, and also to correct the frequency response, for example to remove some bad frequency.

Note that this equalization is applied only on the reverberation effect (WET signal), the direct sound (DRY signal) is not concerned by this equalization.
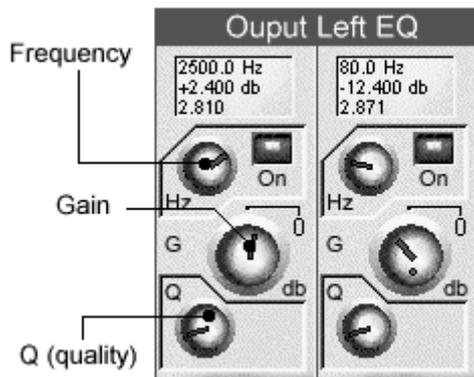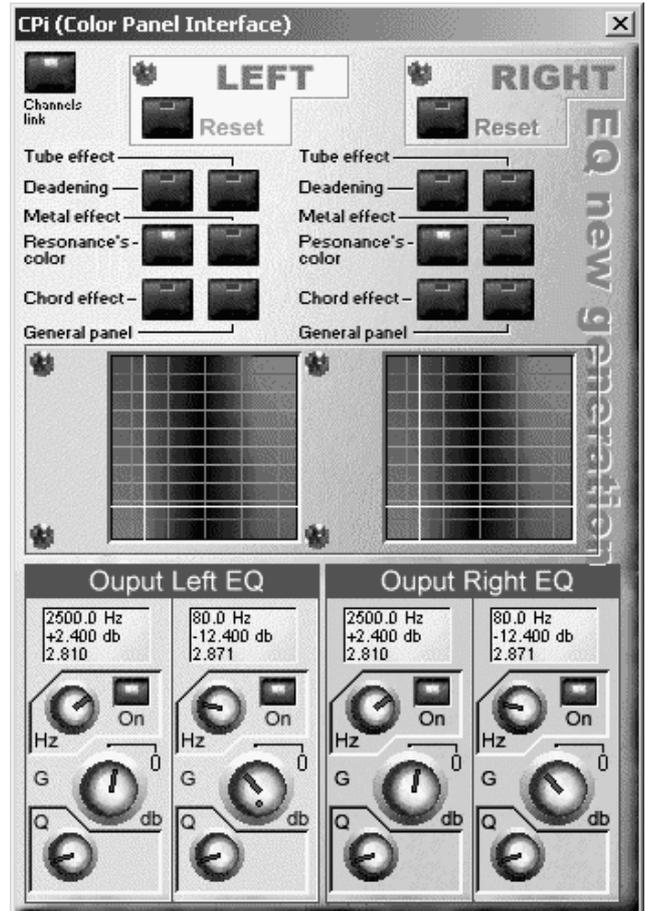
The Parametric cell is regular, and includes 3 knobs to let you set the frequency, the wanted gain and the quality of the filter. Per default the quality parameter is set for removing bad resonance.

The small LCD above knob display the current parameter values of the equalizer : Frequence (Hz), gain (db). And Q (quality).

An On/Off switch (Off per default) will let you enable or disable this filter.

ACI (Algorithm Control Interface)
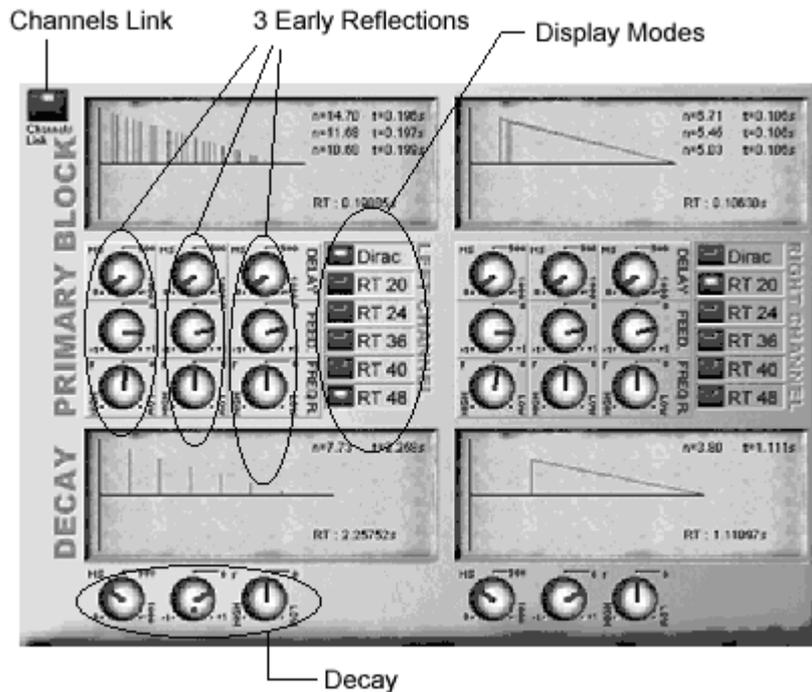
This interface allows to manage the 3 early reflections of the reverberation and one **Decay**. This two parameters block control respectively two processing block as shown on the diagram below
.
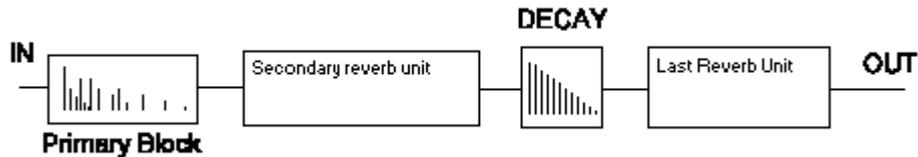
The way of organizing the processing units means that modifying the 3 early reflections, or setting the **decay**, is going to change the sound and the reverb integrity of the **preset**.

Theoretically speaking, if this interface enables, directly or not, to modify attack and length of the reverberation, and to create echo and different special FX, practically speaking, the result will only be confirmed by a carefully hearing.
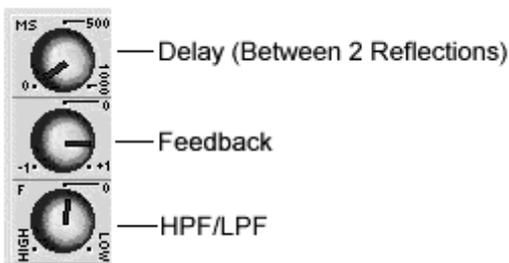
Each filter has three parameters and can generate many echoes (many taps).

**Delay** is the time between each reflection.

Feedback determines the number of reflections, the number of successive echoes (separated by the same time, given by the **Delay**). **Feedback** can have negative values, which make a phase reverse on each tap.

HPF/LPF is a kind of shelving, echo after echo. It is, in fact a damping on frequency response of the echeos in time.

Understanding LCD.

Dirac Mode — Echo 1, Echo 2, Echo 3, Echo n..., n=7.73, t=2.258s, nb echo, RT : 2.25752s, Total Time (fonction of RT)

Time Mode (Dirac Button Off) — Delay, Total Time, n=3.80, t=1.111s, RT : 1.11097s
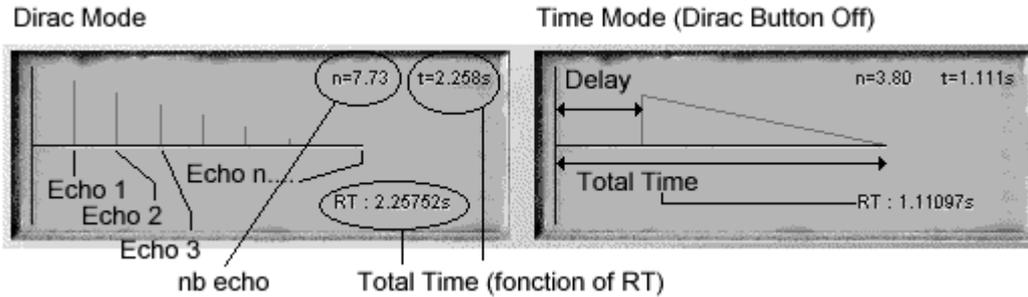
Each reflection, thanks to the **feedback** parameter, can create a lot of successive echoes. LCD screens give in real time the global time and the number of tap for a given RT.
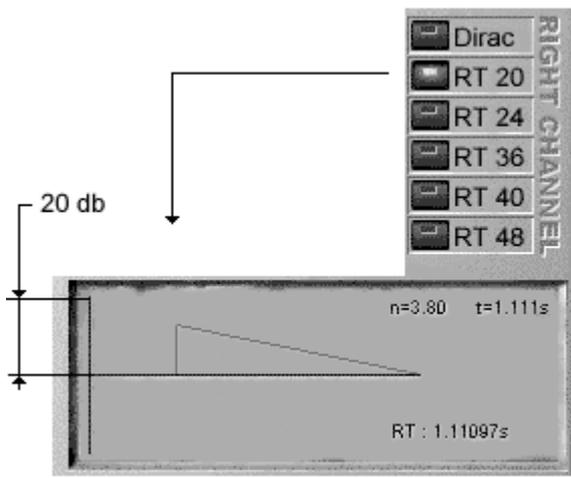
RTxx

The reverb time is function of the number of echoes on the one hand, and on the other hand, of the limit from which we can consider we cannot hear this echo anymore.

RT20 for example, defines the needed time for the reverberation to lose 20db. It is a norm to get a coherent total time.

According to the manner we set the reflections parameters, we need to know their response curves with different accuracies. It is exactly what the push buttons column does.

PRESET & BANK

When "clicking" directly on the LCD display of the main interface, the dialog box below will appear and enable you to manage your **presets** and **banks**.
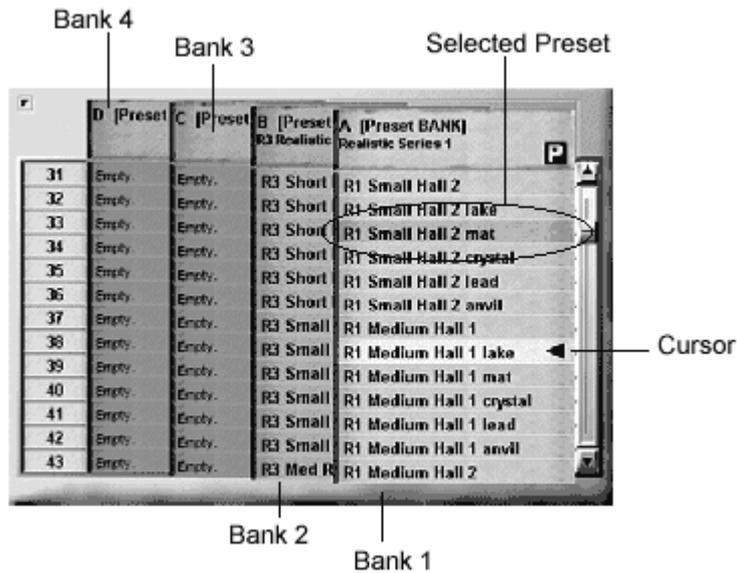
This dialog box is composed of a new type of user control, helping to get four visible columns, each one representing a **bank**. Each column can be brought forward, (just like a sliding panel, "click" on it) or removed ("click" on the title an move the mouse).

When this control has the "**Focus**", you can move inside it by using the keyboard cursors. Copy/Paste a preset is possible by combining **CTRL+C/CTRL+V**

The selection of a preset is made in two phases, two mouse clicks.
1 - Select the new preset (either with the mouse or the keyboard)
2 - Confirm the selection ("click" a second time on the mouse or press <**ENTER**> or <**SPACE**> - if the software HOST does not interfere...).

Contextual menu, right click:

A right click on a bank opens the following POPUP menu:

Divided into five sections, this menu will enable you to create your **banks** of **presets**, to load or save a **bank**, to modify the name of a **preset**, and copy/paste (CLIPBOARD).

Note the following information:

■ The factory bank (in the **bank A** when loaded) is not a data file, but is stored directly within the program.

■ Loading a **bank** in the memory will erase the previous version.

■ You can load a **bank** instead of the factory bank.

■ When you create a **bank**, the number of presets is selected at the start by the position of the cursor.

■ The number of **presets** in a **bank** cannot be changed.

■ The creation of a new **bank** erases the previous one.

■ To copy presets (**CTRL-C**) use the clipboard in order to copy **presets** from a **plug-in** to another.

## Preset Classification

Generally, **presets** are classified by 6 **(lake, mat, crystal, lead anvil)**. Each one describe a different behaviour on frequency response (such a pre-defined DAMPING effect).

(normal)	Natural
lake	bright
mat	dark
crystal	no attenuation on high frequencies (not natural)
lead	muffled
anvil	only bass are reflected

ANNEX A

## Explanation of the controls for the users interface.
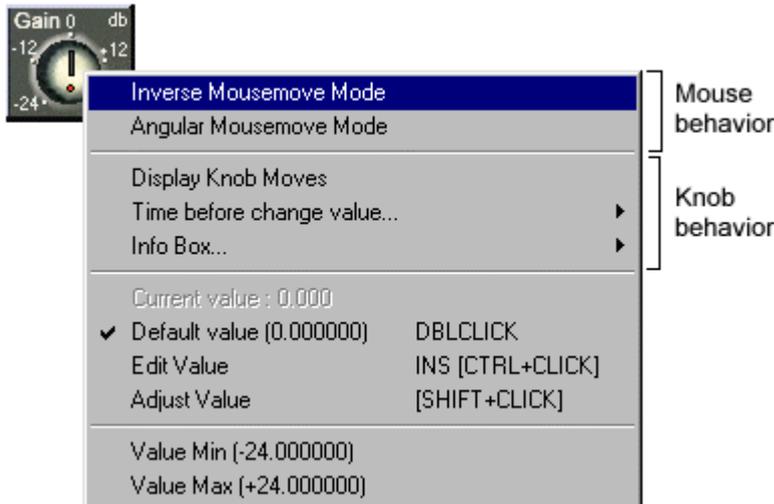
The buttons (**Rotary and Slider**) are part of the contextual menu that can attribute a use to the buttons, and to have access to specific interfaces which can set the parameters to which the buttons are attached.

The gray contextual menu shows the current value and the min. and max. levels attached to the buttons.

With certain controls, the menu shown can be completed by a list of pre-defined values.

The mouse enables the user to turn the knobs to the right as if it was a real knob being turned by hand, turn downwards for a clockwise movement. This is all done by default, when the mouse moves down, the value goes up.

If the **inverse mousemove mode** is selected then the knob becomes « normal », this means that when the mouse goes up, so does the value.
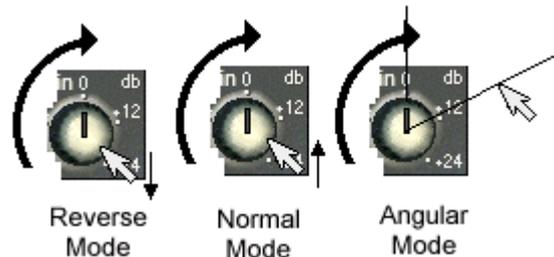
The third selection is the Angular mode. This means the level will move to the angle the mouse has made with knob.
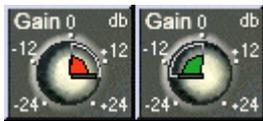
If the user selects the **Display Knob Moves** option, the knob will show the difference between the current value and the default value. The red if it has been increased and green if it has been reduced. This option can be practical to show instantly which knobs have been used.

You can also set a reaction time for the modification of the knob position. This will allow the mouse to move the button in a small manner so that it can change value.

By default, the time is fixed at **0ms** so that the button will react quickly.
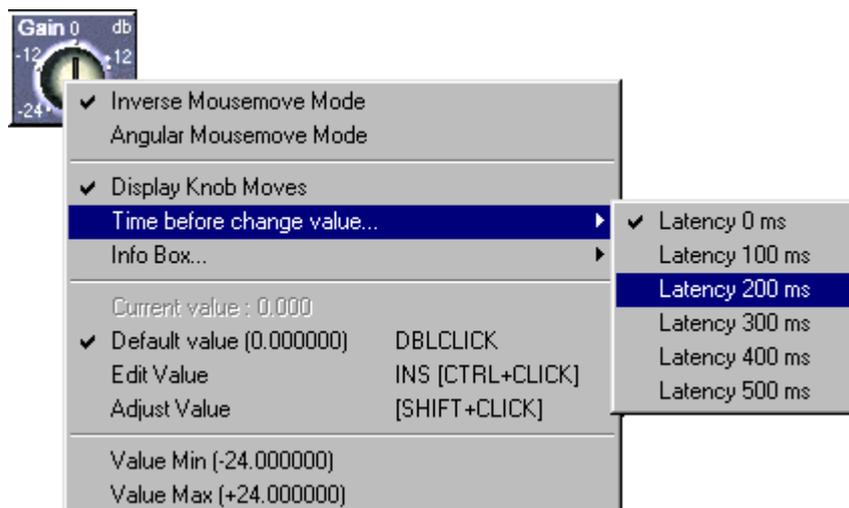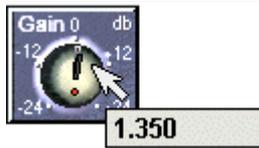
When the user changes the button value, a small box will appear to show the current corresponding value of that button. This box can appear where you want it to, in any corner on the control. See the **Info Box** option on the contextual menu.

CTRL+CLICK or MIDDLE CLICK will make a little edit box appear, so that you can type in the value that you want to put on a parameter. Close this box by pressing ESC or ENTER.

If the button controls a parameter of time, it makes the edit box a lot more important as you can see in the display below.

Here you can edit the time with several units, on a musical scale or even on a video unit. If all the conversions are done in Real time, the **Musical Time** makes the exception where it can keep the last parameters of the tempo or note for a moment, or together for the duration. The button **Apply** allows the times to be changed here.

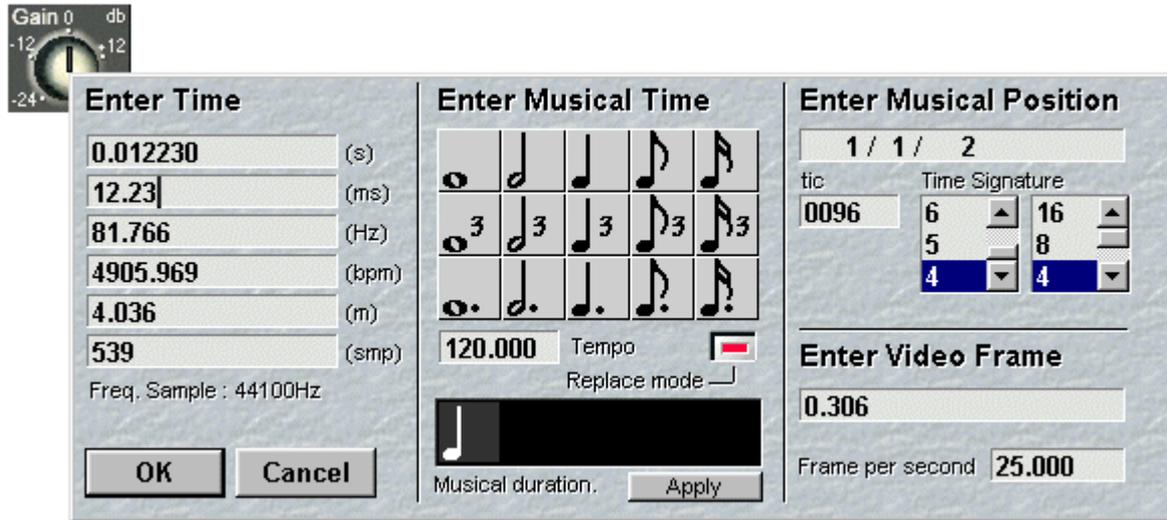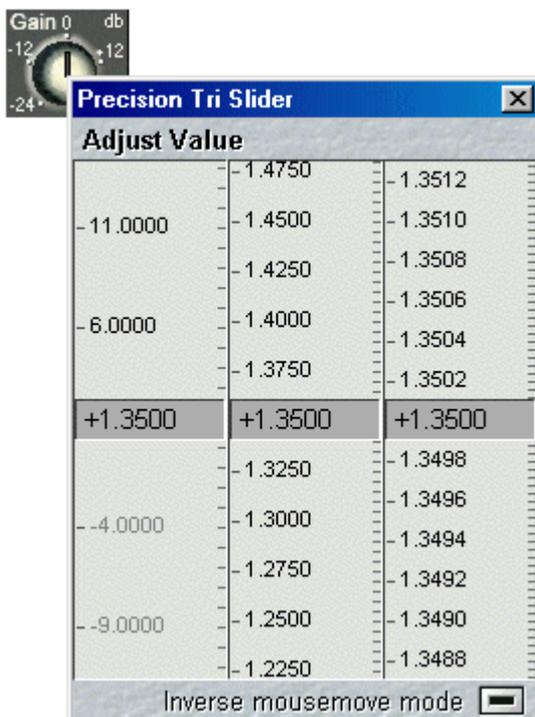Finally, the **Adjust value** option that can be used by pressing SHIFT-CLICK will produce the **Precision Tri Slider** which will allow the user to adjust in real time the value of every button with maximum precision.

Each one of these graphs can be turned using the mouse to great precision. With a factor of 100 from one slider to another, you can play in Real time for the parameters that demand precision. For example, time, a frequency or feedback.

This dialogue box isn't MODALE and will stay on screen just until you decide to close it. There is the possibility of keeping the interface on the screen and selecting another button to work on in the same way.

By default the mouse will give direction to the movement of the sliders. The little push button on the bottom right will make the reverse effect happen. This means that when the mouse rises so will the values and vice versa.

Reverberation (some thoughts)

Publication of the web site of the French Cubase VST club : ECVST (www.espace-cubase.org)

Introduction

The reverberation of a sound (also called "room effect") is the result of the sound's interaction with its diffusion environment. In other words, the sound will be different depending on where it is given forth and listened to. You can for example do the experience with a tambourine. Try it in all the rooms of the house, from the bathroom to the garage, crossing the garden... The sound changes, its timbre and its duration too. Well, it's funny.
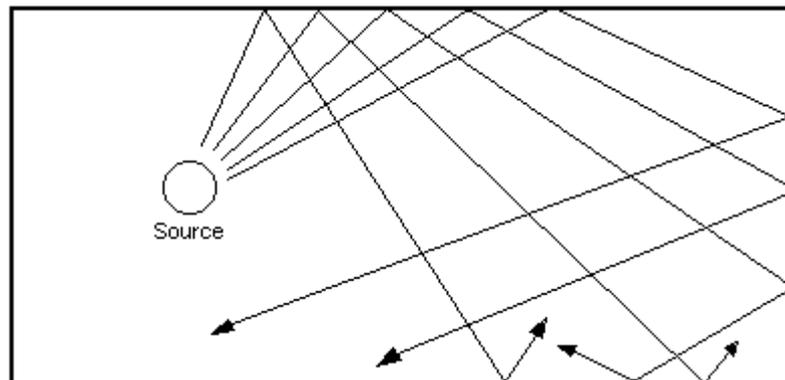
In fact, a sound has to be considered as an array of acoustic rays that go in all possible directions. The rays are then reflected, absorbed, refracted, depending on the nature of the bodies (solid, gaseous, liquid) they meet on their trajectory. If the sound doesn't come across an obstacle or a change in the ambient environment (a warm air stream in a cold room for instance), there won't be any reverberation. It's almost the case with open field sound system, or in an anechoic chamber.

The « reverb »

The goal of a reverberation effect is to simulate a room effect, to recreate the sonic atmosphere of a more or less known environment : small room, train station hall, church, concert hall, etc... However, these reverberation effects are often used for different objective uses : thickening a sound, giving it some space depth, putting it in front, masking defects, drowning it, giving colour... Each and every user has a precise idea about what is a reverb, and that's what make it a special plug-in. It can almost be compared to an instrument, in the way that each reverb owns an unique colour, an unique behavior.

The fact that there is no norms, no standards... leads to a consequent variety of effects, all grouped under the denomination "reverb". It seems then quite difficult to do an objective study of the different products on the market. There will always be a specific use in which the qualities and defaults of a product will come through.

However, to stay in the general direction of using the reverberation effect for what it is supposed to do, that is recreating a room atmosphere, we somehow have to compare it to the facts. The comparison is to be done against a real concert hall, a cathedral, an open garden, etc... And I invite you to wander in reverberating places so that you can hear it for yourself. Thus the choice of a reverberation effect becomes restrictive and difficult., as regards to the conditions of use in one hand, and to the ear on the other hand, which has a natural propensity for getting accustomed to mediocrity.

Quality of a digital reverberation.
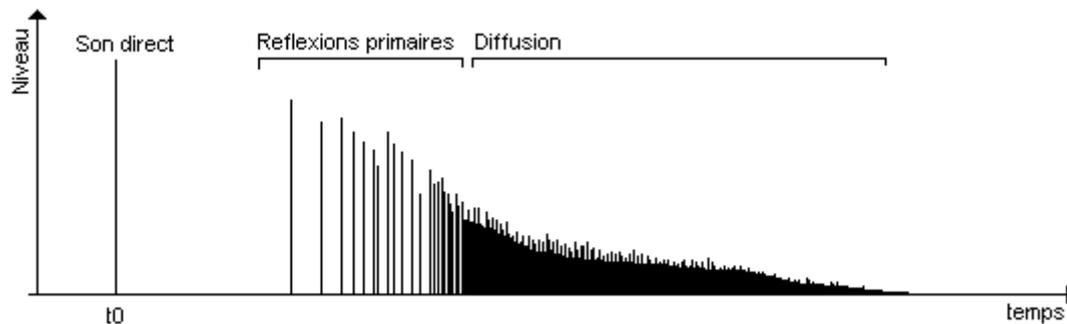
This graph shows the volume curve of a reverberated Dirac peak impulsion. A Dirac impulsion is a very short sound, of which the length is in theory only one sample long at time t0, where its value is maximum. That peak is represented as the first bar on the graph (t0). When the sound peak passes through a natural or digital reverberation the result above is obtained. The early reflections (or echoes) can be clearly seen; they form the series of distinct lines following the peak. The rest of the curve represents the densification of the reflections, called diffusion. The strength of the phenomenon decays over time, until the total loss of acoustic energy.

To simulate a reverberation effect in the digital realm, we thus have to create a series of echoes that will be densified over time, all the while decaying in volume. A reverb that doesn't make the sound more dense is of a bad quality, or maybe it's simply an echo, multi-echo or multi-tap effect. To make your mind on this phenomenon, you can try your favorite reverb with percussive sounds (short sound with a strong attack). Let one or two seconds go between each blow and then listen closely to the whole sound reverberating. It's one of the only objective tests one can do to judge a reverberation. However, though this test is necessary, it's not enough to determine the quality of a particular effect.

In fact, a reverb is not only a succession of echoes. On more, the behavior is not the same with an impulsion (as shown on the last graph) and a continuous sound, such as a musical piece (a full audio track with bass, drums, flute, guitar, piano, voices, etc.). It's not the same either with low frequencies and high frequencies. In brief, it gets complicated.

To show this complexity, let's use the case of a church and a cathedral that would be 100 times greater than the church. These buildings are usually made of stone, therefore materials that are highly reflective. As you guess the church will give back a generous amount of high frequencies in its reverberation (think of a "ttsssss"). However, if we make the church grow, high frequencies won't get more present, they will tend to do just the contrary. Why ? It's because high frequencies are not very energetic when compared to low frequencies. Consequently, the distance to travel before meeting an obstacle gets too long for high frequencies, which have lost a lot of energy getting there, bumping through the molecules of air before arriving to the wall.

The examples of that kind are numerous, and foreshadow a complete and complex interaction between all parameters of a reverberation. It's the way the effect creators took these interactions into account, that give each reverb its own sound, the natural of it, the coldness or the warmth, the color. What a poem...

Caracterization of a reverberation.

We have to understand that a real-time reverberation effect is not really a simulation, it's the simulation of a simulation. I'll explain... When objects are created in 3D video, a texture can be applied to them, and then they can be lighted with a number of spots. Be it for a ray tracing or a ray tracing algorithm, the softwares that can light a 3D object do a simulation of the behavior of the lights in the 3D scene. In the case of ray tracing, the software simulates the progression of the important light rays and their reflections to calculate the resulting lightning and shadows.

In case of sound, the algorithms of digital reverberations don't simulate the acoustic behavior of sounds, but rather carry out an optimization of the simulation process through a highly simplified algorithm. But alas ! It's thus far from the physical phenomenon of reverberation of a sound in a room. This alternate process is opted for because of the enormous difficulties in putting in relation the physical parameters of a room and

the parameters of the simulated effect. So, while we wait for softwares that are able to use the architects' plans as an input parameter in real-time treatment (this can be conceived today, but forget for a real-time treatment on PC), we'll expose the usual parameters used in a simulation.

The broad family of reverbs

Presets are often classified by type, which are essentially defined by the shape of the amplitude curve of the first reflections. That curve is often linear, but sometimes the program ads a "liveness" button that modifies the slope. All variations within these types are possible of course, and other classifications could be brought...

PLATE Type. The amplitude curve of the early reflections is, as the name implies, like a plate. This type corresponds well enough to small rooms.	 Reflexions primaires — Type Plate
ROOM Type The early reflections decay steadily and simulate the reverberation in a somewhat big room, such as a small concert hall for example.	 Reflexions primaires — Type Room
HALL Type The decaying of the early reflections is irregular, letting you ears think that there's a complex environment with multiple obstacles. It's better suited to the simulation of train stations and big buildings' interiors	 Reflexions primaires — Type Hall
REVERSE Type This was the rage in the 80's; it's usually followed by a GATE. It's very tricky to use this type to simulate a real space... Maybe a room coupled with a small room from which the sound is emitted ?	 Reflexions primaires — Type Reverse
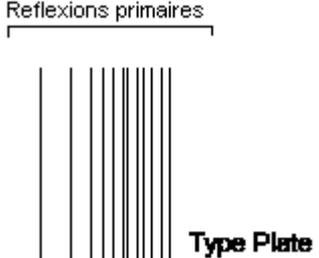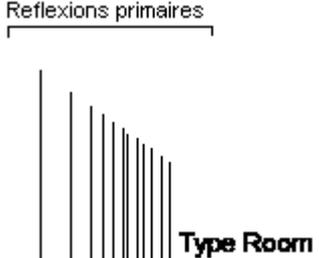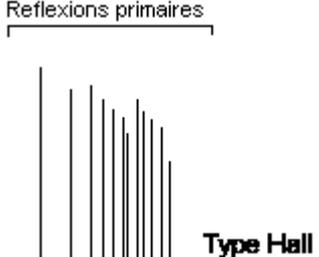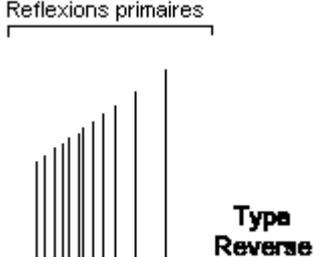
Note : the amplitude curve of the early reflections is not enough to define a PLATE reverb or a ROOM reverb. Others parameters come in play such as the time between the reflections, the type of diffusion, etc...

Parameters of the reverberation effects.

Because of the diversity of the algorithms, many parameters can vary from a software to an other, they cannot easily be normalized. But they might have the same effect on the sound as those described below.

Dry and Wet or the balance (mix).

The "dry" parameter is usually the gain of the source, and is often called "direct" for a reverberation. The wet parameter is the gain of the effect, that is the reverberated sound. "Direct field" and "reverberated field" are terms that can be used too.

The "dry" and the "wet" can be combined in a single parameter, then called "mix", that controls the balance between the direct and the effect. Though It might be convenient, it's not very realistic. That's because the dosage of the dry and the wet sound contributes to the perception of the listener's position in the simulated room. The farther the listener is, the more the reverberation will be present to his ears, the reverse being true. However, let's not forget that the reverberated field has a homogenous amplitude over the whole volume of the room. This observed property implies that if the listener moves away from the sound source, the dry sound will be less present but the reverberation (wet sound) amplitude will NOT be higher.

Pre-delay

Pre-delay is the time it takes for the sound to touch the first important obstacles in the room. For example, if the first wall to meet is at 10 meters from the source the pre-delay will have a value between 30 and 50 milliseconds. A lot of reverberation effects forget about this parameter and it's unfortunate because it's the only one to have a clear relation to the physical phenomenon that interests us : the room effect.

Early Reflections

When the sound is reflected a first time (that is, at the beginning of the reverberation, when the sound waves meet their first reflecting obstacle), we get of series of first reflections, known as early reflections.

Few softwares let the user control the early reflections, essentially because not everybody can use them appropriately. Unless you have a visual appreciation of these reflections it's almost impossible to treat them correctly. Furthermore, a modification of the early reflections implies modifying the whole of the effect, diffusion included, and this to a degree usually not anticipated by the creator. In one word, you cannot be able to modify only the third reflection, and only this precise one.

Reverb time (or RT)

The reverb time makes it possible to shorten the length of the reverberation, that is to say, normally, to change indirectly the size of the room. One will keep in mind that a modification of that parameter gives as a result an uncontrolled modification of the colour, density, sound texture, essentially because of the technology of the algorithm itself, and the incapability to put the parameters of this algorithm in touch with a reverb time without changing the integrity of the sound. The value of this parameter, sometimes indicated into seconds or milliseconds, corresponds to nothing. On more, this parameter makes often it possible to transform a Plate reverb into a Hall reverb and vice versa ; then, the user could believe that he has just to reduce the time of a Hall reverb to obtain a Plate reverb, that is a featured nonsense.

Let's seize the opportunity to talk about Mr Sabine who defined reverb time (I think so) to be the time that takes a reverberated space to loose a certain number of decibels. One talk also about RT60 that is the time taken by a reverberation to loose 60dB, but one could also talk about RT40 or RT20. When such a parameter is implanted in a digital effect, the problem is the perception of that time. That is to say, in certain cases, it is not out of the question that you could perceive a RT40 preset of 5 seconds, for example, less long as an other RT40 of 2 seconds. Besides, this parameter was in the past considered as a sufficient parameter in the building of concert halls or auditoriums. Obviously, it has been quickly noticed that two halls having the same RT60 didn't have of necessity the same acoustic results, the same reverb.

Perception of reverb time is a psycho-acoustic phenomenon, that is to say it depends on subjective parameters. A reverb that is strong in high frequencies would always appeared longer than the same one being cut above 5 KHz. In permanent sound, you aren't be able to hear the reverb when its level get down 24 dB under the direct signal. So, the most the direct sound will be stronger than the reverberated sound, the less this last one would be perceived as long. Etc, etc...

In short, this parameter will make it possible to you to change the length of the reverb, but it should be manipulated without excess if you want your sound to be coherent and of quality.

Damping

This parameter must be put in connection with the type of used material in the room. The more the materials are reflective, the less muffled are the reflections. In the opposite direction, the reflections would be more quickly muffled if, for example, the walls of the room are covered up of tissue (absorbing material).

"Damping" could be also assimilated to a frequency of response or to the soften rate of the high frequencies. But in fact, it is more subtle, because "damping" conditions really the absorption rate of the high frequencies in time for the entire reflections of the sound.

The adjustments of this parameter are rather the same for any machines. However, any unit can be assigned to it.

Frequency of response (Freq Rep, Colour, brightness)

It is often about a low-pass filter that is more or less performing, it is set before or after the reverb unit, to give more or less high frequencies.

Use of a digital reverb.

As we already told, the reverb effects can be used by different ways and for some very various purposes too. We will forget about the chapter of synthesis or re-synthesis (yes, reverb can also transform the sound), it-s up to you to use your creative imagination...

Use for room simulation

For this type of use, in post-production video or to create sound atmosphere for example, the effect should be handled with discernment by using logic and the properties of sound. Let's imagine that there's a concert in a hall, the listener is just in front of the source of the sound. In this case, the source of the sound must be stronger than the reverberated field... Except if he is at 40 meters away from the flute player, who is depth in the background of the cathedral, not too far from the saloon... Let's keep in mind that the reverberated field can't produce more energy than the direct sound. Reverb is a passive phenomenon, walls, sides and obstacles don't give energy to the sound, they preserve it at the most, a little bit as a capacitor.

Adapt the preset to the place. If the spot is a garage or a saloon, or if it is an open place like a courtyard, a "Train-station Hall" effect will be inappropriate and vice versa, if we have still the purpose to respect the energy logic of the sound and to adapt the intensity of the reverb in consequence. If two people whisper in a spacious hall, the reverb will be very low because the level of the speech will be very low, thus, this speech will be reverberated very low. On the contrary, a mortar shot in the closing center of the crypt of the basilica St-Pierre de Rome would be "heavily" reverberated. Moreover, one avoids to add the sound effects like this in very reverberatable places like the subway, big train-station halls, etc., by increasing the number of sonorous sources. Some amplifiers are set every 5 meters, for example, and produce a little acoustic power in order to make it possible to the sound to loose a maximum of energy before being reflected, so reverberated.

The use of a reverb effect on an already reverberated sound could be conceived, but only with high-quality effects. One will avoid to use the same preset from a treatment to an another one. As for a work in multitracks, using several different reverbs on the audio tracks could be done only with high-quality machines. On the contrary cases, one will obtain undesirable effects like resound, variations of phase, tube effect, distortion, and sometimes divergence of the signal (that is to say that the volume of the sound increases till a total distortion and full disappearance.

Moderated use or how to have a big sound

Reverbs give to the sound some frequencies, some harmonics, some sonorous space, some power too. In short, there's a reason why organs are often located in churches, the building is then used as a sound box being a real part of the instrument.

In a lot of cases, a well-chosen reverb improve the quality and the depth of the sound. To do so, discreetly, one will adapt the preset to the sound. The more the sound will be short and aggressive (drums, for example), the more one will use a short reverb, and vice-versa. A variation on the right and left pre-delays will make it possible to give more sonorous space, in other words, to "enhance" the final sound.

The purpose is to not be able to distinguish the reverb effect from the original sound.

In mastering

Generally, and except any particular case like making a "live" album from a studio record, the reverbs applied to the sound are often at a very low level (-12 to -40 dB) compared to the level of the direct sound. This gives beauty to the silent parts and adds insensibly a certain heat and unity to the sound.

Sometimes, an album is treated with a small reverb to unify its sound, and in order to simulate the sonorous ambiance of the room where would have been played the songs.

Conclusion.

Digital reverb effects give some satisfying solutions to the ear, in the creation of sonorous spaces and room effects. The quality of these effects depends hardly on the number of the filters used in the algorithm. So, don't be surprised to notice that the best of them request a lot of CPU resources.

Except the technical part of a reverb effect, that is besides not determinate (that is to say that there 's no wonder solution, no miraculous algorithm), the quality of a reverb depends on the feeling of the person who adjusted the sounds. This work can't be done in a mathematical way, systematically, it still remains the sensible part when one elaborates a reverb effect. For this reason, a reverb is an instrument, with its own characteristics and its specific colour...