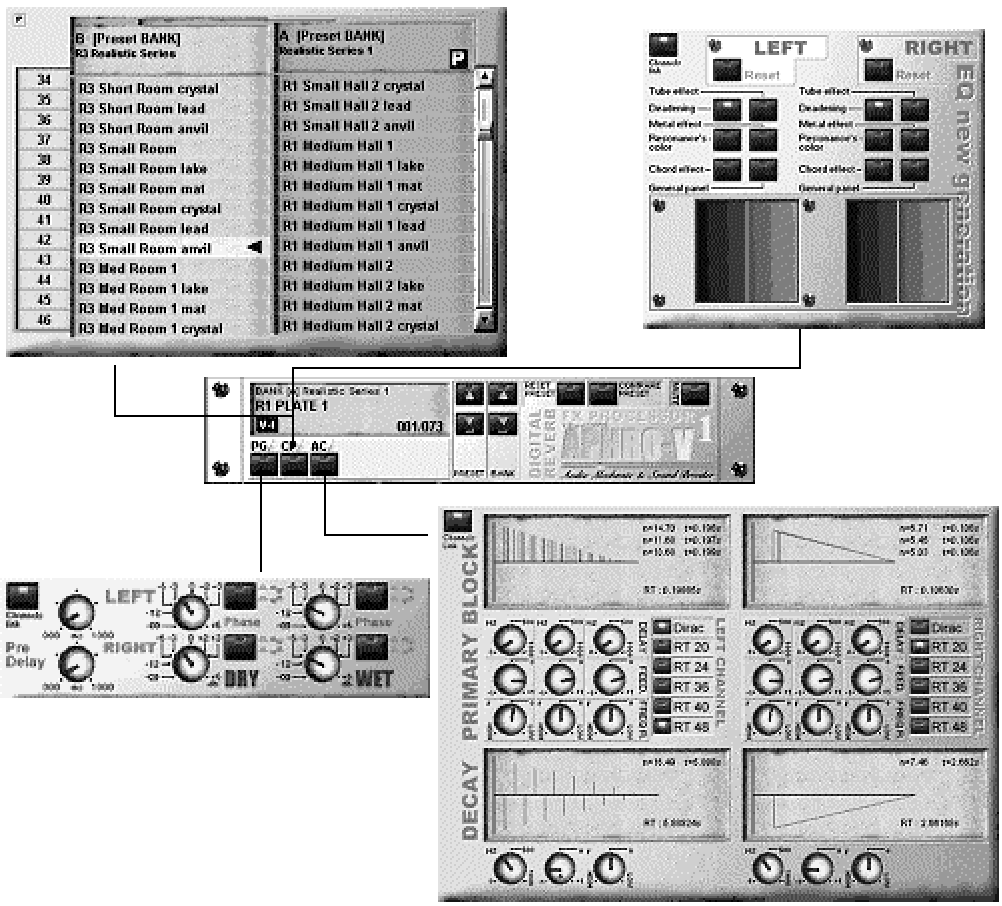
Aphro-V1

Digital reverb & fx processor..



Summary

English translation by Muriel François
Thanks to Bill Corley for his help and valuable advice.

1 May 1999

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Specifications:

APHRO-V1 DX Plug-in for PC

PC configuration	Intel Pentium© 133MHz (min).
Memory configuration	HD 5Mo Free, RAM 16Mo (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
Output pin (data type)	As Input
Output pin (frequencies accepted)	As Input

frequencies normes	**11025Hz**	**22050Hz**	**44100Hz**	**48000Hz**
Speed (stereo)	15% (CPU*)	25% (CPU*)	50% (CPU*)	55% (CPU*)
Memory access (stereo)	2.7 Mb/s	5.5 Mb/s	11.1 Mb/s	12.0 Mb/s
Memory used (stereo)	0.9 Mb	1.7 Mb	3.5 Mb	7.7 Mb

Features	
Factory preset bank	internal presets. : **Realistic series 1** (73 presets)
PGi	pre-delay and Gain interface.
CPi	Color Panel Interface.
Preset and Bank	Interface for presets and banks processing.

(*) CPU time measured with Wavelab 2.0© for a 16bits stereo sound on INTEL PENTIUM© 200Mhz with 60ns RAM.

Introduction :

Aphro-V1 is a real time Digital Effects Processor. First 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.

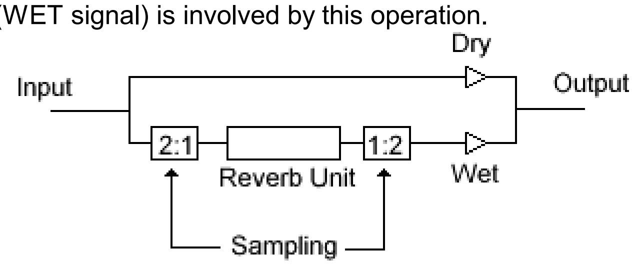
The handling philosophy is made simple and practical thanks to a wide range of preset, which requires the user to select a preset matching the best desired effect, and then to use the different interfaces in order to adjust it, according to his conveniences.

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 especially to reach Multi Dialog Box, in order to get the expected result.

Quick or not

Aphro-V1 includes two versions : **Aphro-V1** and **Aphro-V1 Quick** (a **Q** on the right bottom of the main interface allows you to make the difference.).

The only difference between both of them stands internally, in the manner of processing the signal. The **Quick** version 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 usefull 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.

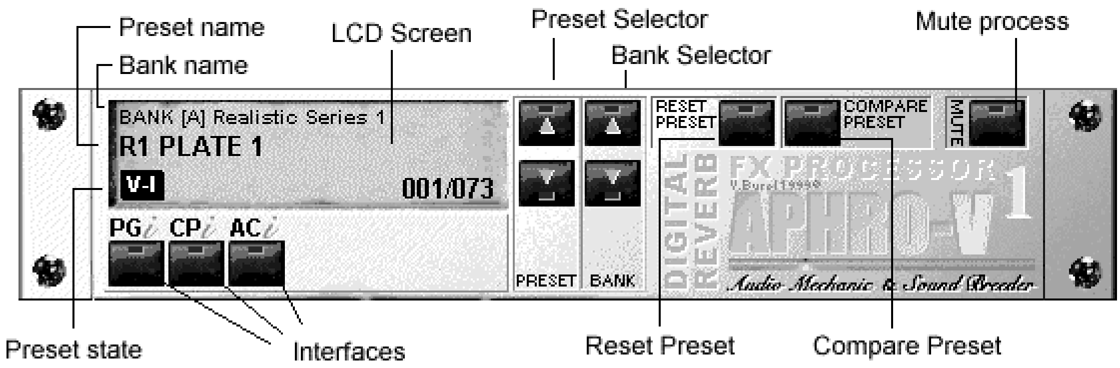
The **Aphro-V1** interface is quite alike the one we could have on a hardware unit. One of the ergonomic problem recently encountered concerned especially buttons behaviors : Are they physical buttons that could be only modified by the user ? Or are they cursors for parameters setting and visualization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

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, being small, enables the user to have all the useful functions at his disposal. It essentially means changing **presets** with both "up" and "low" buttons or activating an interface by pressing one of the "**PGI**" or "**CPI**" 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-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-V2** is able to deal with presets coming equally from **APHRO-VI** or **APHRO-V2**.

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 (**PGi, CPi, ACi**). This multi-interfaces system provides a minimal use of screen space and a thematical grouping of controls and parameters.

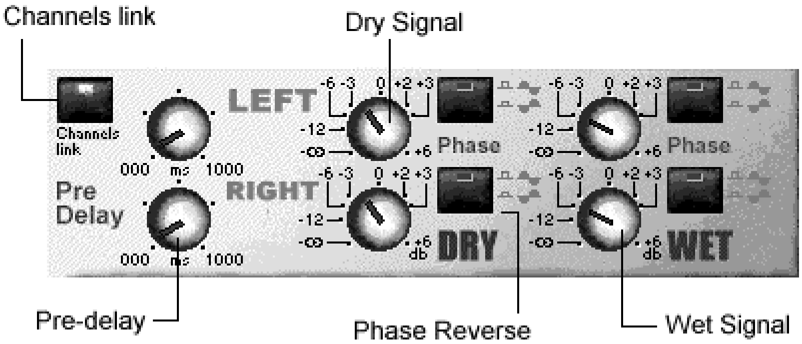
A third interface may be selected by "clicking" directly on the LCD display. A dialog box specialized on managing **banks** and **presets** will then appear

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**.

PGi (pre-delay and gain interface)

This interface allows you to check the basic parameters of the effect. For both left and right channels, be aware you can edit different parameters ("right-click" on the mouse when set on a rotary button).

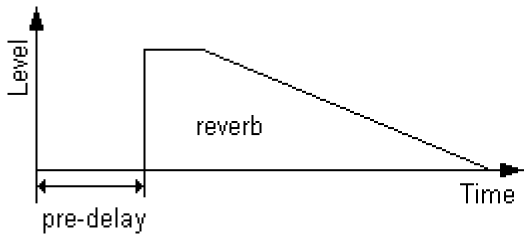


Channels Link:

Channels Link defines the running mode of the rotary buttons as regards this dialog box: if the button is pressed, both channels are linked. For instance, a modification of gain on the left channel will affect the right channel in a same way and vice-versa. (If the **plug-in** is on monophonic reproduction, only the left channel parameters are used by the effect).

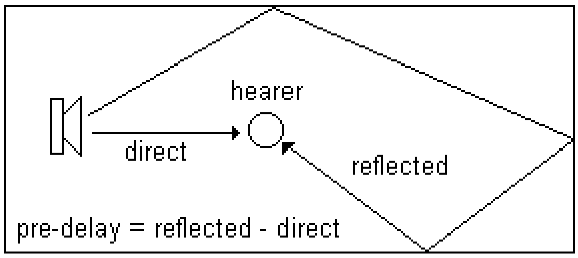
Pre-Delay : time before effect.

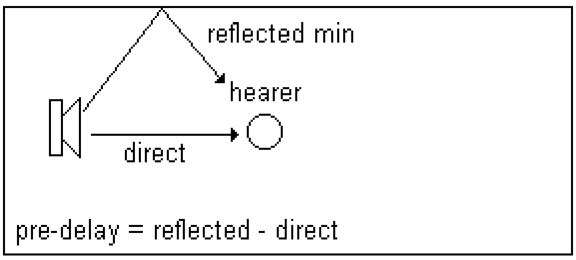
Because a reverberation always occurs after the direct sound, the **pre-delay** is an essential parameter. It will help define the lapse of time which partly characterizes the size of the hall.



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.





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:

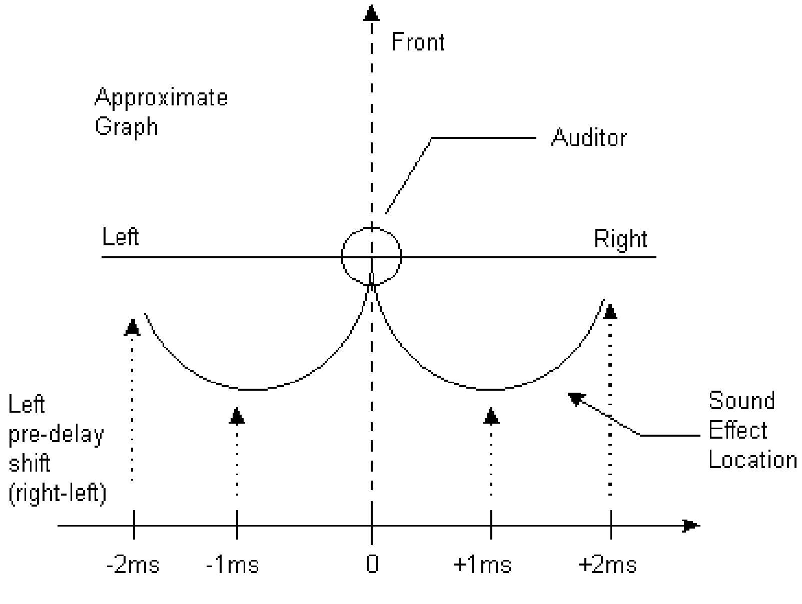
If you make both channels independent by pushing up the **channels Link** button, you can severally define the right **pre-delay** time from the left one. This allows us to induce time-shifts between the left and right channels.

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.

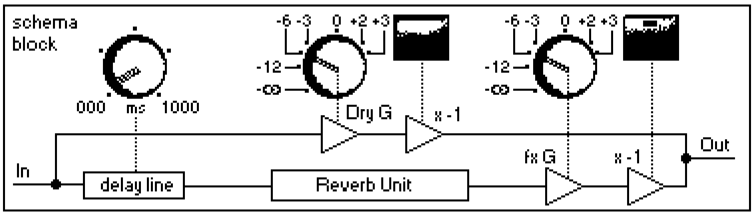
This effect is not very stable as it depends on the nature of the original sound, its stereophonic effect, its spectral density...

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:



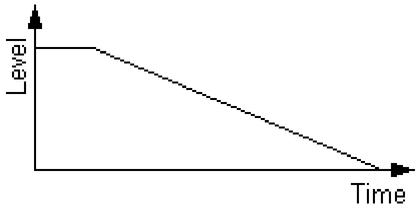
These two parameters will enable you to balance between direct sound (**Dry**) and reverb effect (**Wet**). The aim is to adjust both parameters (**Dry** and **Wet**) in order to obtain the expected ratio and the optimal output level (what is to say the output level as close as possible to the 0db level, 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 ≠ 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.

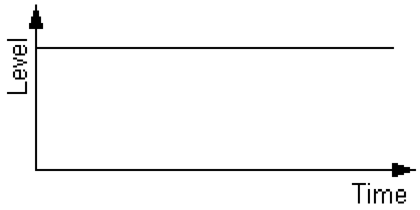
Gain reaction of the reverb effect.

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:

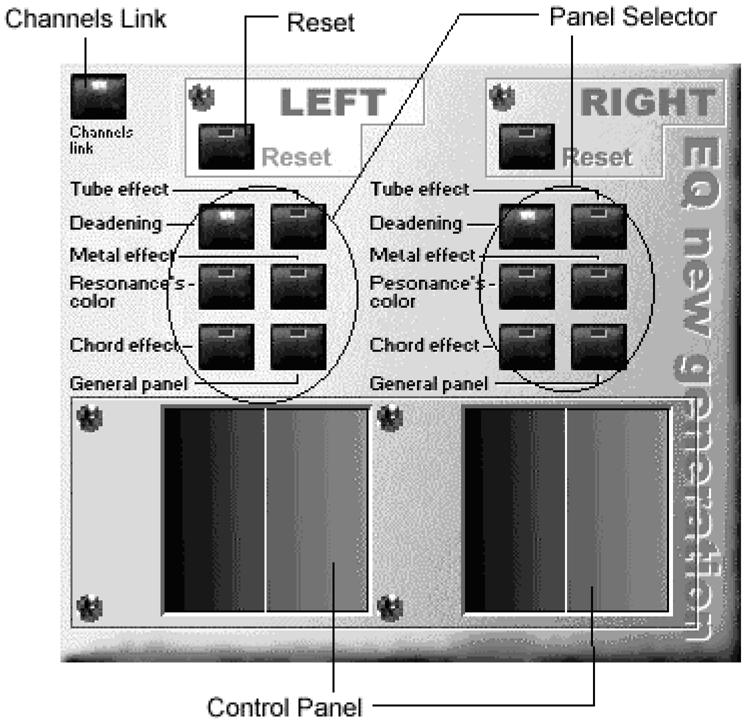
The input signal (**Dry**) like the reverb signal (**Wet**) can be displaced of **180°** (channel by channel) by pressing the push buttons connected to the rotary ones.

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...

CPi (Color Panel interface)

This interface permits to control the reverb effect "**colour**" for the left and right channels.



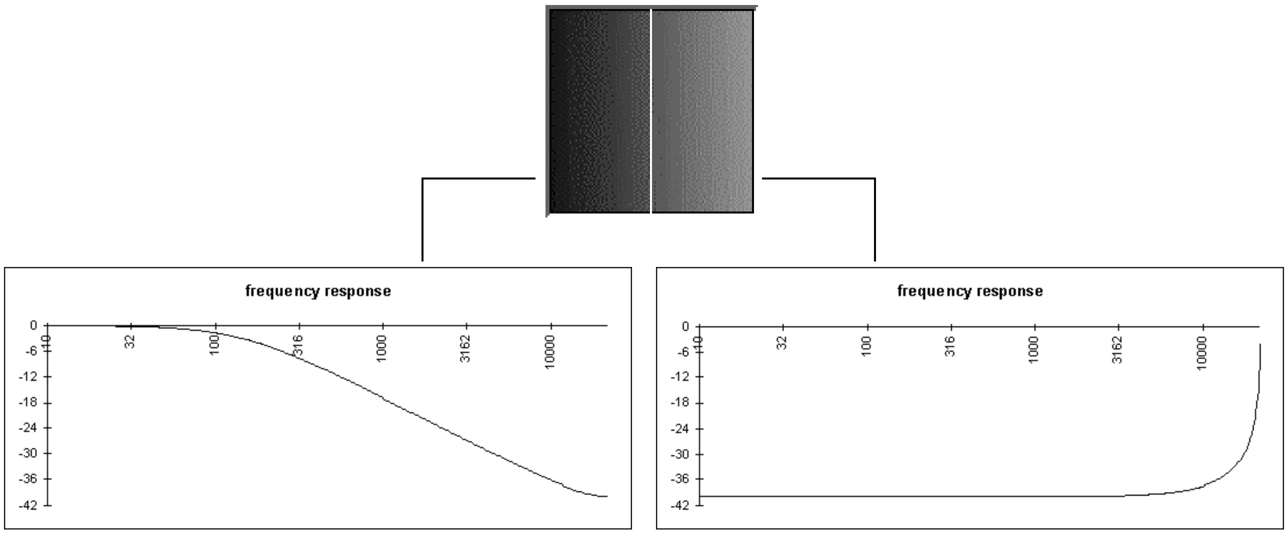
Channels Link:

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:

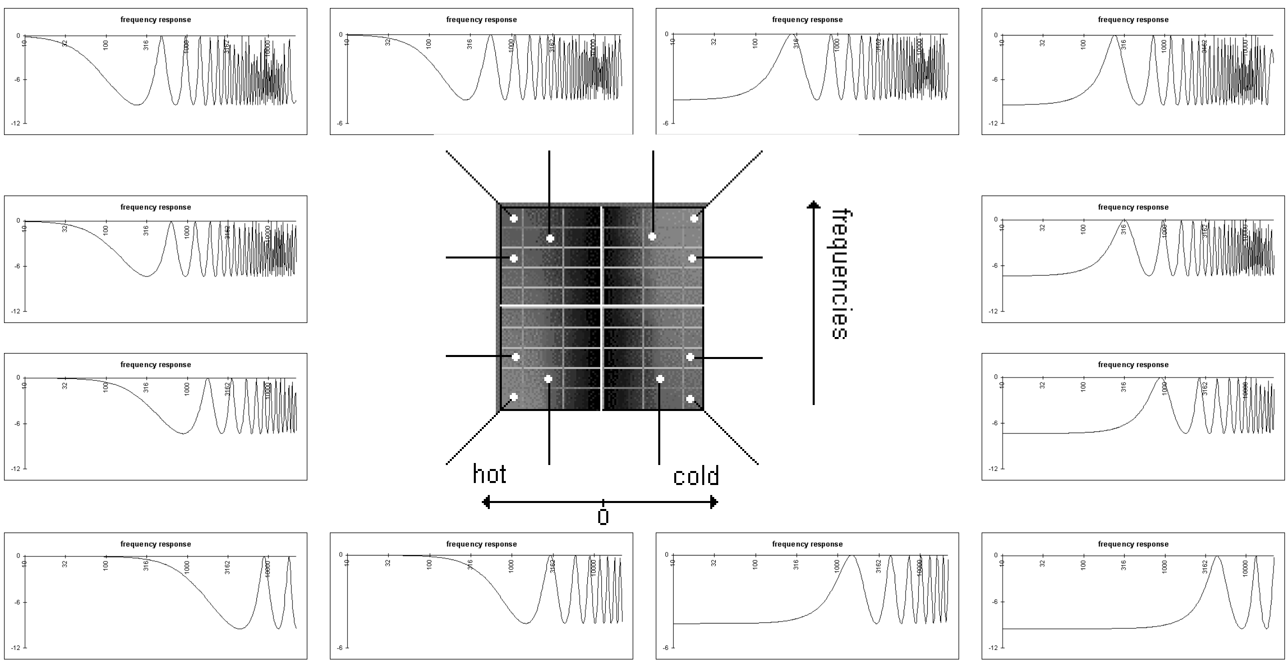
The six control panels (**Deadening, Resonance's Color, Chord effect, Tube effect, Metal effect and the General panel**) allow to control the same filter, but on a different way. You should be aware of how each panel is supposed to react before selecting the panels, which will give the desired effect.

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...

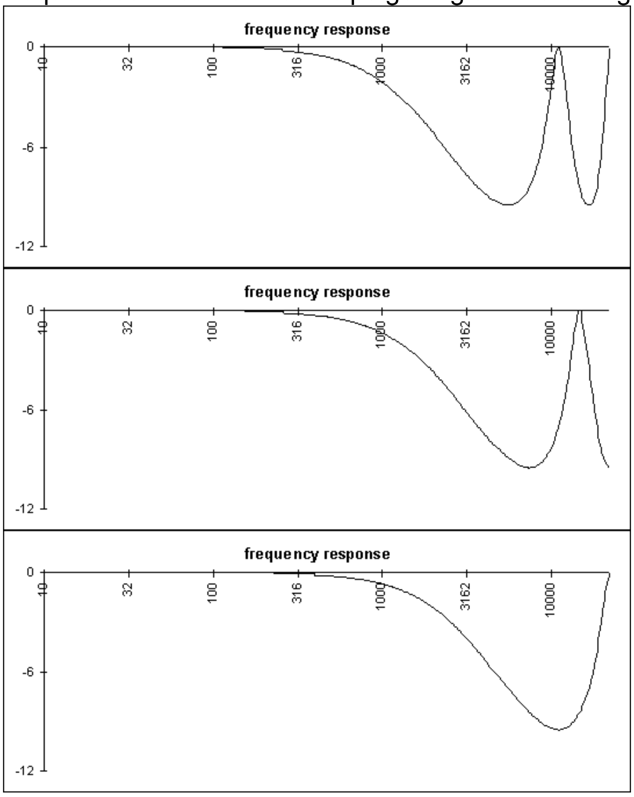
The **Resonance's Color Panel** is a subjective tool that should not be avoided by the hearer to obtain a good result. " So, It's up to you! "



Chord & Tube effect

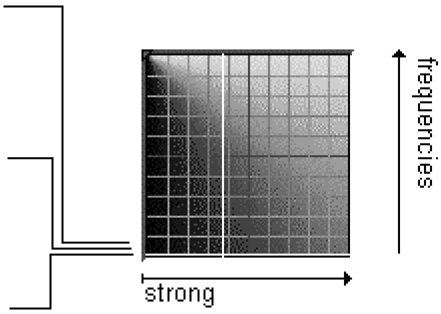
You can appreciate the originality of these two control panels: on the one hand the **Chord effect** is especially used to make drums sounds sound like those of a guitar, the **Tube effect** on the other hand makes these sounds sound like a draft passing through a tube.

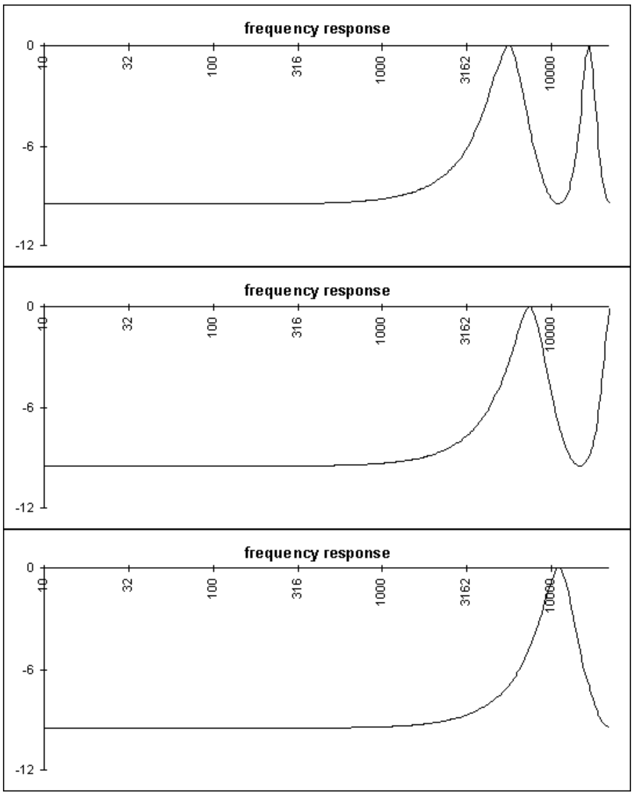
Be aware that if you move the mouse down on the panel, the filter changes into a resonating filter for 2 or 3 frequencies. It sometimes helps getting an interesting equalization effect.



Chord effect

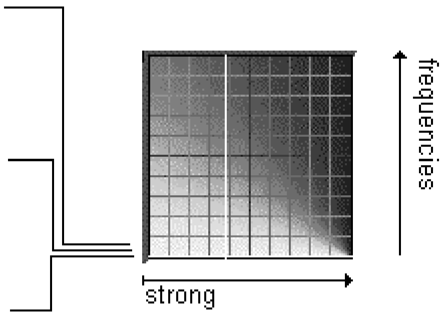
More selective than the low pass filter connected to the **deadening**, it also allows frequency peaks to appear in the high sounds.





Tube effect

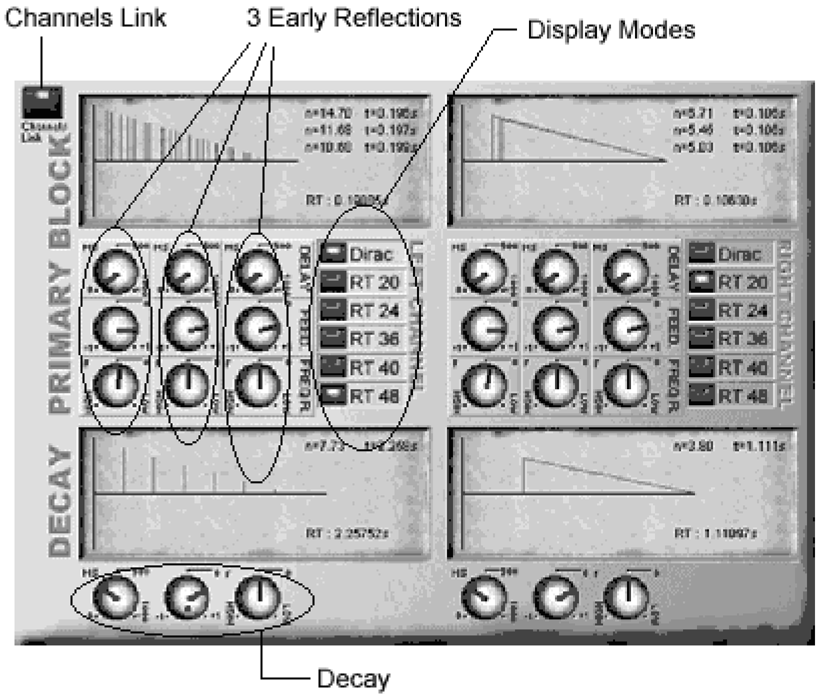
More selective than the high pass filter connected to the **deadening**, it also allows frequency peaks to appear in the medium sounds.



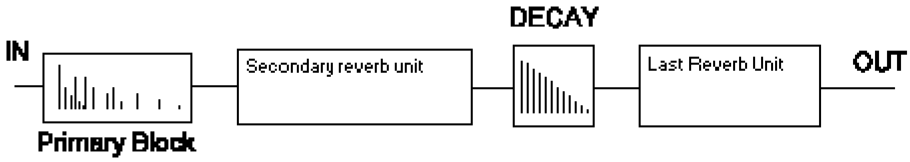
Metal effect
This effect gives a slight tremolo in the medium and bass frequencies. It is better if you can carefully listen (**Dry** muted) this effect before using it, because it is difficult to hear it on long reverb effects.

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.

ACI (Algorithm Control Interface)

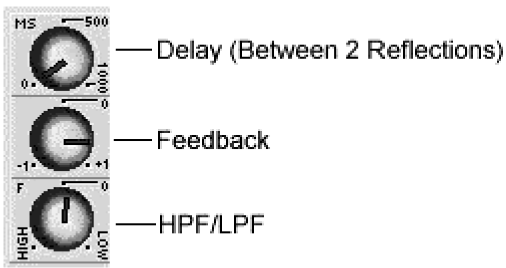


This interface allows to manage the 3 early reflections of the reverberation and one **Decay**. This two parameters block control respectively two processing block as shown on the diagram below
.



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.



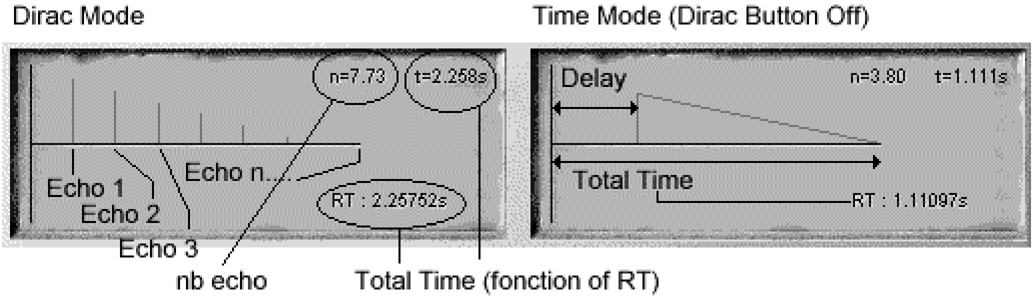
Each filter has three parameters and can generate many echoes (many taps).

Delay is the time between each reflection.

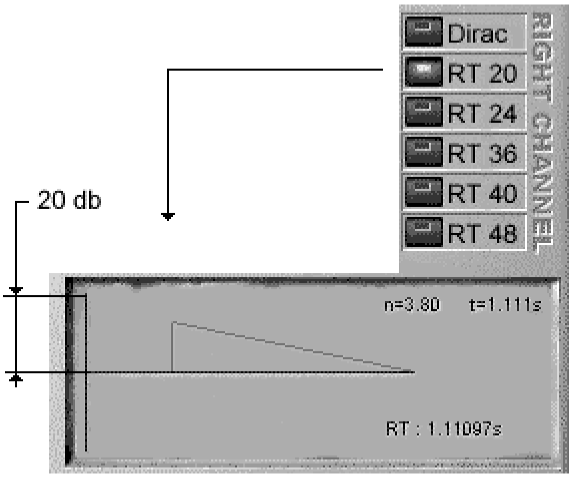
Feedback determines the number of reflections, the number of successive echoes (separated by the same time, given by the **Delay**). **Feedback** can have negative values, which make a phase reverse on each tap.

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.



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

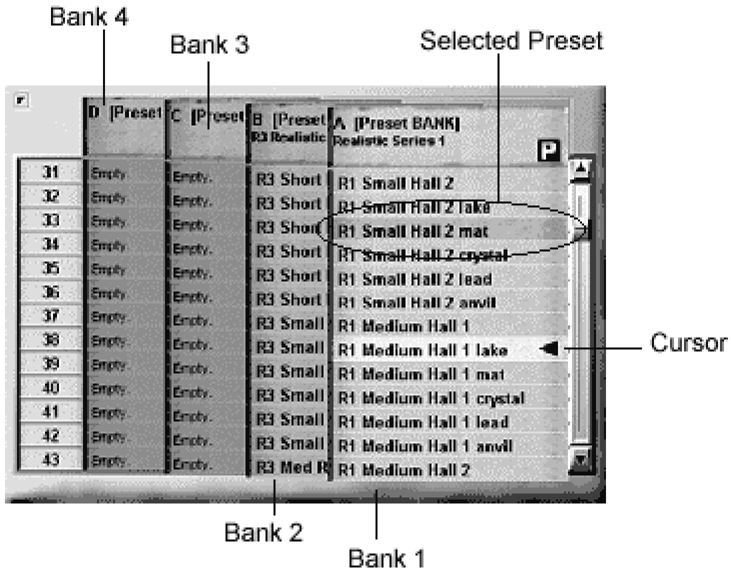
The reverb time is function of the number of echoes on the one hand, and on the other hand, of the limit from which we can consider we cannot hear this echo anymore.

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**.



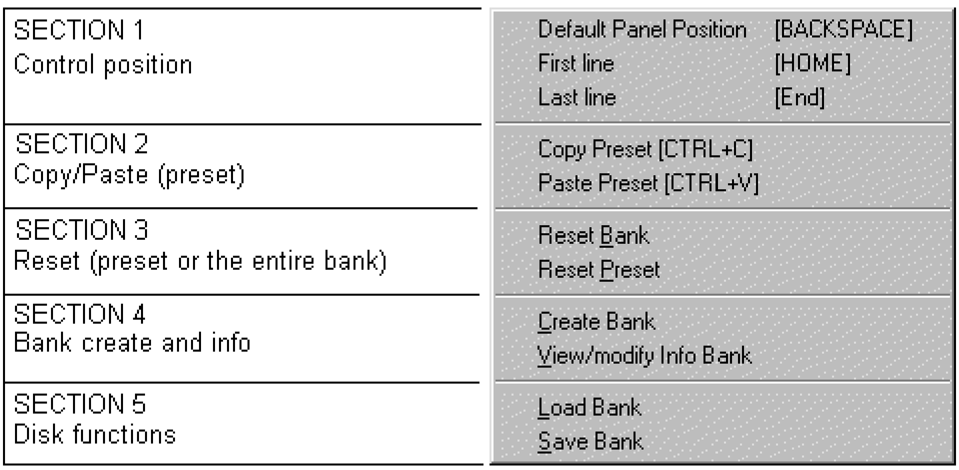
This dialog box is composed of a new type of user control, helping to get four visible columns, each one representing a **bank**. Each column can be brought forward, (just like a sliding panel, "click" on it) or removed ("click" on the title an move the mouse).

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.

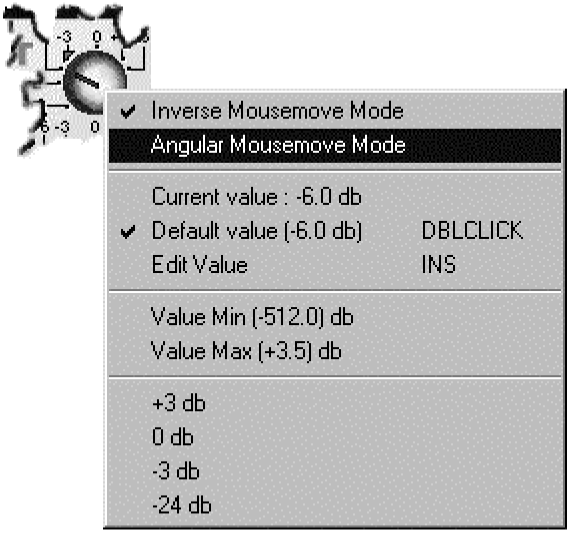
Preset Classification

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

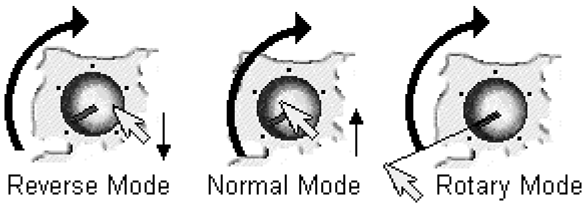
Rotary buttons use.



Rotary controls have a contextual menu (right-click the mouse) which helps getting information about the parameter connected to the button :

- current value.
- default value.
- extreme values.

The low part of the menu has a list of pre-defined values. The high part of the menu helps selecting a running mode of the rotary button.

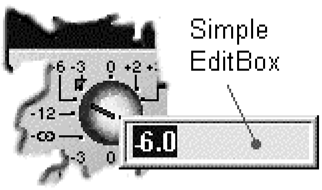


Rotary buttons can be used according to three different modes.

The first one is the default mode which requires the user to couple forces to turn the button.

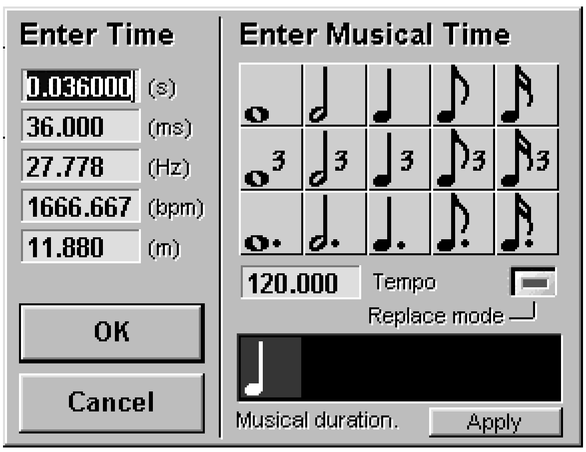
The second one is the classical mode which, with an up and down mousemove, makes the value increase or decrease.

The third one is the angular mode. You define the button position with the angle corresponding to the mouse position.



The rotary buttons values can be edited (click MIDLE, or left-click + CTRL, or left-click + SHIFT).

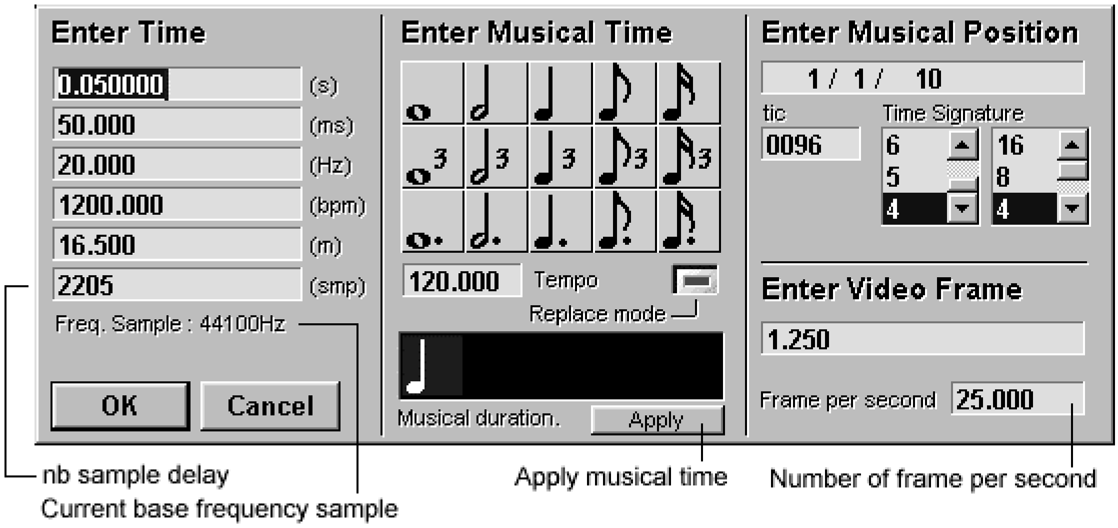
<ESC> to cancel <RETURN> to validate.



If the rotary button is associated to time length, the edition is made with this dialog box.

Very efficient, it enables you to define a time according to five different units.

The right part (MUSICAL TIME) allows to use a note value, simple or complex, according to tempo.

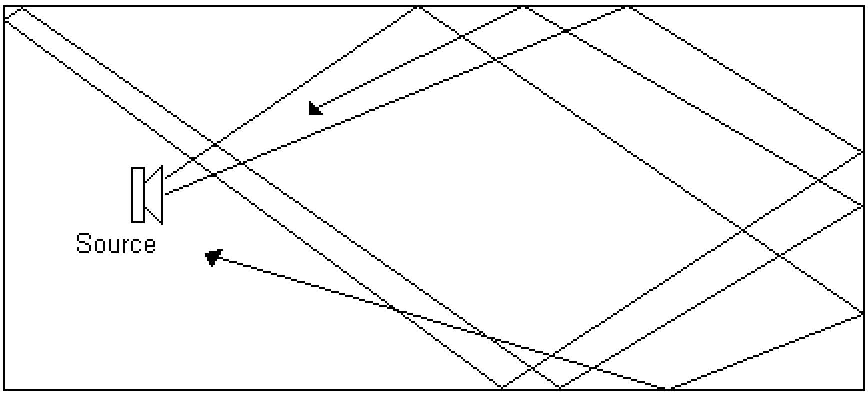

Enter Time
0.050000 (s)
50.000 (ms)
20.000 (Hz)
1200.000 (bpm)
16.500 (m)
2205 (smp)
Freq. Sample : 44100Hz
OK
Cancel
Enter Musical Time
120.000
Tempo
Replace mode
Musical duration.
Apply
Enter Musical Position
1 / 1 / 10
tic
0096
Time Signature
6
5
4
16
8
4
Enter Video Frame
1.250
Frame per second
25.000
nb sample delay
Current base frequency sample
Apply musical time
Number of frame per second

ANNEX B

Reverberation (some thoughts)

In theory, let's say that the reverberation of a sound is the result of reflections, refractions, diffractions of sound waves sent in a surrounding way, which reflects, propagates and absorbs them. In actual practice and in order to use a reverb effect realistically speaking, we will take into account the following issues:

1 - There is a reverberation if the sound coming from the original source is supposed to meet obstacles, surfaces, walls or anything which can be used as a reflector, propagator or absorber.



2 - Sound cannot be continuously reflected, because not only it loses its energy when moving (the further you are from the source, the weakest the sound) but it also loses energy when meeting obstacles (walls, chair, hearer...). Consequently the volume of a reverb effect decreases within time.

3 - Obstacles, even if they are reflecting ones, are supposed to absorb a quantity of the energy of the direct sound, and modify the spectrum in phase or frequency of this sound. One of the consequences of this phenomenon is that the different frequencies are not reverberated the same way. Generally, high frequencies have a reverberation time shorter than bass frequencies. Reverberation color belongs to those complex parameters, difficult to handle because of numerous factors: position of the source, size and shape of the hall, materials used in that room...

4 - The reverb area in a hall can be considered as constant everywhere (in terms of sound volume). This issue is essential in the moving simulation of a hearer within a hall having a fixed original source (See **APHRO-V2**).

5 - The relation between the response frequency of the reverberation and the size of the hall is not straight-line. We can say that the reflecting materials condition the bright or muted aspect of the reverberation. In fact, it is far more complex. It is true that the less absorbing the materials are (concrete, stone, metal) the higher the frequencies are within the reverberation. But, high frequencies are less energy giving than low frequencies, that is to say they die down faster when covering a long distance. In other terms, if the hall is too vast, the distances to cover (for any kind of sound reflections) become too long for the high frequencies to be intensively reverberated (this effect is very efficient with the **APHRO-VI** presets - see "SMALL HALL" and "LARGE HALL"). On the contrary, in a small hall, with an acute reverb effect, reflections are overlapping each other up to changing and alternating the frequency response of the sound. In fact, it is as if the sound was going through many comb filters. In that case, **APHRO-VI** presets are set up to reproduce this effect (see PLATE).

6 - The length of a reverb effect depends on the size and structure of the room as well as on psycho-acoustic phenomena: input/output ratio, “ tone ” and “ range ” of the reverberation, decreasing curve and etc... However, according to these various parameters and realistically speaking, APHRO-VI parameters will be used with a sense of distinction. For example with the preset "PLATE" corresponding to a small room reverberation, if you slightly increase the pre-delay, at an outside yard, the hearer will not hear the direct sound any more, the effect being more like a clear echo. (Pre-delay at 260 ms). In any case, always ask yourself what would happen in reality. You will find better solutions in this way than by using complex calculations. Aphro-V1 does the calculations leaving the creation of realistic reverbs to your ear and good judgment.