MultiDAC software

USER'S MANUAL

V2.5

INDICE

Jan 2002

0.- ACKNOWLEDGEMENT

Thank you for choosing the **MultiDAC** digital audio controller as your digital processor for your applications.

Before starting to work with your new **MultiDAC** processor, please read this manual about the Configuration software carefully in order to make the most of all its features, get the best results and use your PA equipment in a safe and correct way.

All the products manufactured by VMB Española S.A. have been designed using the latest and best materials available on the market together with the most up-to-date technology. They are being checked in the most difficult working conditions in order to assure their correct performance in any ambience.

The VMB engineering department works day after day to offer the best products that fulfil the needs of the market. If you have any doubt or comment about the performing of one of our products, do not hesitate to contact us.

VMB Española S.A. thanks you for relying on the **MultiDAC** audio controller. We stay at your disposal for any collaboration. If you have any suggestion regarding the functioning of the Configuration software or the processor, please inform us: we are open to any suggestion that could improve our products.

To get more information about VMB products, to contact us, to obtain any manual or to download the latest version of the software go and visit our web site:

www.vmb.es

1. - Introduction :

The **MultiDAC** *digital audio controller* is a digital audio processor with 2 inputs and 4 outputs, completely configurable by its user. It has been designed to perform like:

- A complete 3 and/or 4 ways PA mono processor (global equalisation, delay unit, crossover, individual out equalisation, limiter and an independent noise gate per way).
- A complete 3 ways PA mono processor with sub-bass output through Auxiliar send.
- 2-ways stereo processor
- 2-ways stereo processor with mono sub-bass + Aux send.
- 4 outputs distribution system with different equalisation, delay and levels.
- Stereo paragraphic equalizer
- Stereo delay line with equalisation
- Stereo compressor/limiter
- Etc…

The optimal configurations of all VMB´s audio systems as well as free memories for the user have been enclosed in the **MultiDAC** processor. Using the **MultiDAC** software, the user will be able to modify or configure ALL the parameters of the **MultiDAC** processor.

Subsequently, it is possible from the processor to change the configuration and modify the parameters to adjust to any installation (like delays, gains and polarities) whereas the parameters that characterise the equipment (equalisation, crossover and limiters) remain hidden to make sure that no one can change them and that the equipment security is guaranteed. For more information read the instruction manual of the processor.

MultiDAC software has been completely developed by VMB Española S.A. If you want to get the latest improved version of this software with its new functions, do not hesitate to contact us or to visit our Web site: www.vmb.es. As the whole DSP Operating System is stored in the **MultiDAC** processor in flash memory, you can download it as many times as you want . This way, you will have whenever you want the latest version and up-to-date improvements of the processor without having to contact the technical service.

VERY IMPORTANT

NOTE ABOUT VERSION 2.5 : The **MultiDAC** Software Version 2.5 with the DSP version 2.5 has some process improvements that are not compatible with older verisons. If you use Software V2.5 with processors with DSP version older than V2.5, the software will detect it and force the user to update the DSP version to V2.5. It the user updates the DSP version to V2.5, then ALL MEMORIES HAVE TO BE RE-SAVED. OTHER WAYS DIFFERENT FERQUENCY RESPONSES AND LEVELS WILL BE OBTAINED THAT COULD DAMAGE THE SOUND SYSTEM. NEVER USE SOFTWARE VERSION OLDER THAN V2.5 WITH PROCESSORS WITH DSP VERSION OLDER THAN V2.5. If you doubt or have some questions, please contact your dealer or direct with VMB Española S.A.

1.1 - General features

MultiDAC processor includes two 32 bit floating-point DSPs (Digital Signal Processors) with 40 bits internal resolution that guarantee an extreme dynamic range superior to 700 dBs (this is the reason why it is completely unsaturated in what internal operations are concerned). Round-off errors are as lowest as possible. These DSPs can develop up to 120 million mathematic operations per seconds providing a great processing capacity. All these algorithms have been developed to provide the best precision and the smallest round-off errors in the calculations. As a result, the best sound fidelity and transparency free of noise is achieved.

The CRYSTAL AD and DA converters used are 24bits ones with a dinamic range of 117dB. This guarantees a clean and free-distortion sound with a negligible noisefloor level, making **MultiDAC** one of the processors on the market with better characteristics. The analog components have been chosen and optimized to diminish noise and distortions.

The 19 user memories and the DSP operating system are stored in flash memory: this allows to actualise them. Therefore, the processor can be actualised from the **MultiDAC** software and it is possible to use the last version of the software.

The processor is provided with a LCD display and buttons from which it is possible to change memory or modify some parameters of the current memory as well as to copy the configuration of one processor to another one through a serial wire without using the computer.

There is a possibility of protecting memories to prevent them from being deleted or changed by mistake. The keyboard of the processor can also be locked and the input can be protected from the keyboard thanks to a password.

From the **MultiDAC** Software the user is able to configure all the parameters storing them in the processor with a simple serial interface.

2.- Software Installation

2.1.- Setup MultiDAC software

Before setting up the **MultiDAC** processor software, check that your computer complies with this minimum requirements such as:

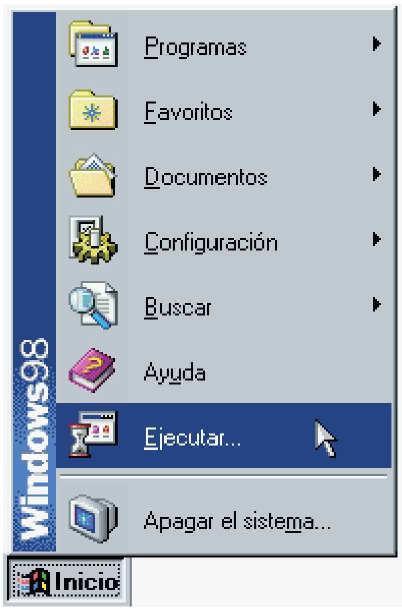
- Pentium 133 Mhz.
- 16 Mb of RAM memory.
- 5 Mb of free memory in the hard disk.
- One or several available COM prot (9-pin sub-D connector)
- Windows™ 95 / 98 / NT / 2000 / Me / XP.

If your computer complies with all of these requirements or exceeds them, the software will be set and will work without any problem.

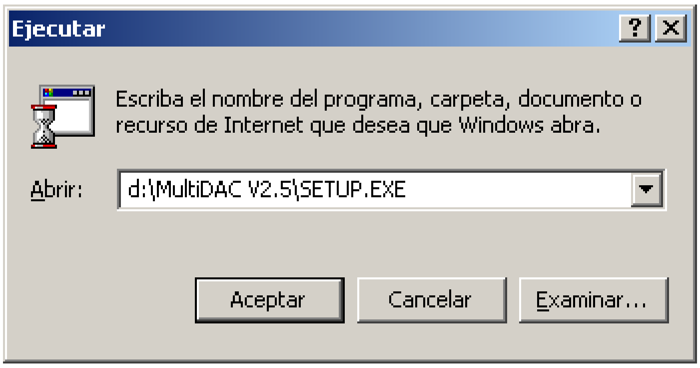
This software has been designed to run under Windows ™ 95/98/NT/2000/Me/XP. To set up your copy of this program, follow the instructions mentioned below.

Before starting with the setting up, make sure that all the current applications are switched off.

Insert the floppy disk provided with the processor in the floppy- disk drive of your computer, normally called A:



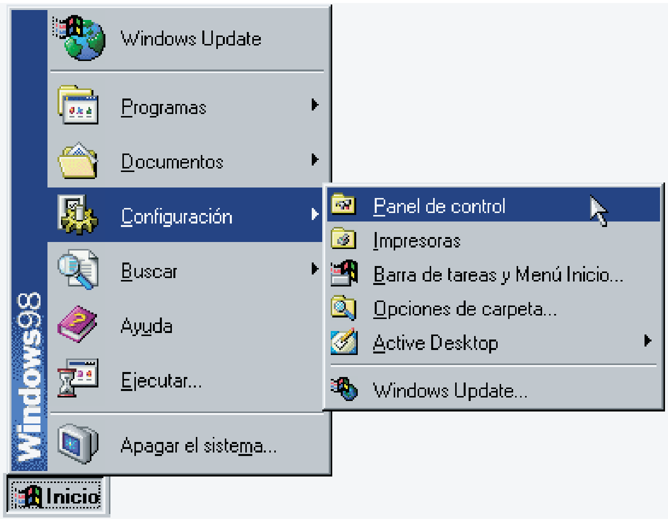
In the Start menu select the EXECUTE option…
In the window that appears, in the "Open" blank space write: *D:\MultiDAC V2.5\\Setup.exe* and press Enter.



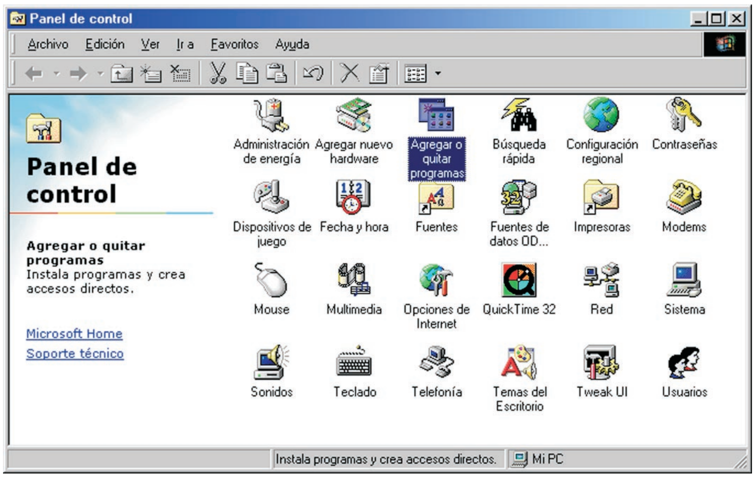
Few seconds later the installation software will start. From now on follow the instructions displayed on the screen.

Another way of setting up the **MultiDAC** software consists in using the “Add/Delete programs” in the CONTROL PANEL window.

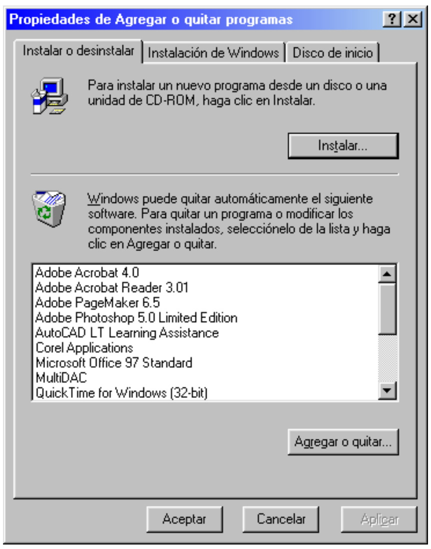
Proceed as follows:



From the START MENU OF Windows ™, select the CONFIGURATION option and then the CONTROL PANEL one.
The corresponding window with the same name will appear.

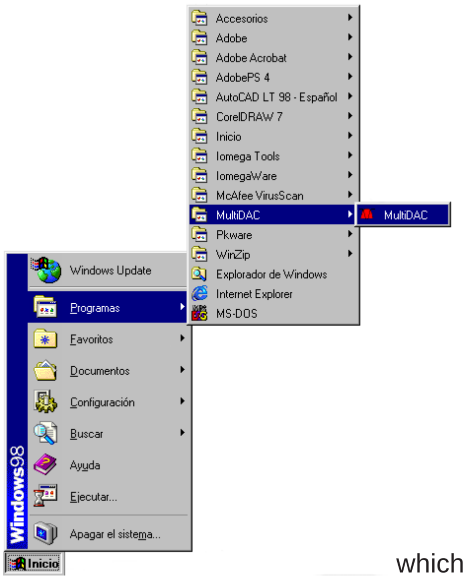


Select the “Add/Delete programs” icon and double-click on it.

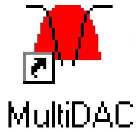


Once in this window, select the Install option to execute the software setting up program. Follow the instructions of the program for the setting up.

2.2.- Getting the MultiDAC software started

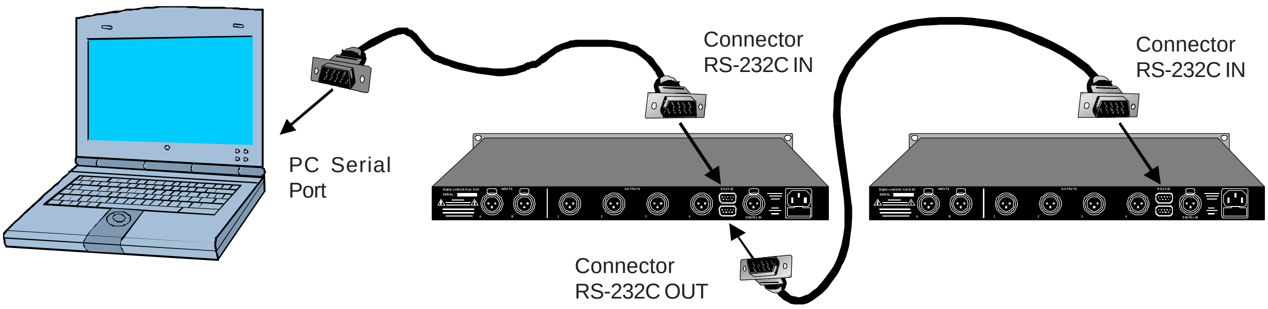


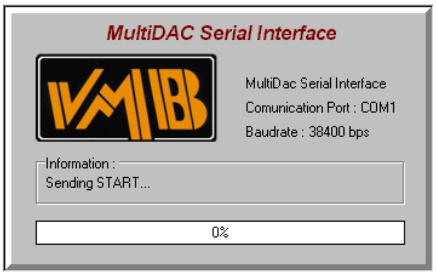
The installation program creates an input in the program list of the Windows START menu. This new input is called **MultiDAC**.
To start the program, select PROGRAMS in the START menu. Once in the PROGRAMS window, select **MultiDAC** by clicking on its icon.
If you have created a direct access to the program from the main screen, just click twice on the icon.



The first window to appear asks you to indicate the PC serial port to which your **MultiDAC** processor is connected. To answer you can select one of the three proposed options:
- Working without connection (Offline)
- Processor connected to COM1.
- Processor connected to COM2.

If you want to work with your processor connected to the computer in order to configure it, take the provided serial wire and connect one of its ends to the selected serial port (COM1 or COM2) and the other one to the rear part of the **MultiDAC** in the RS-232C IN connector. Once selected the corresponding option, confirm the selection the OK function key. Is is possible to control several processors in parallel with the same configuration, bridgeing the output of the processor RS-232C OUT with the RS-232C IN of the next one with an identical serial cable.





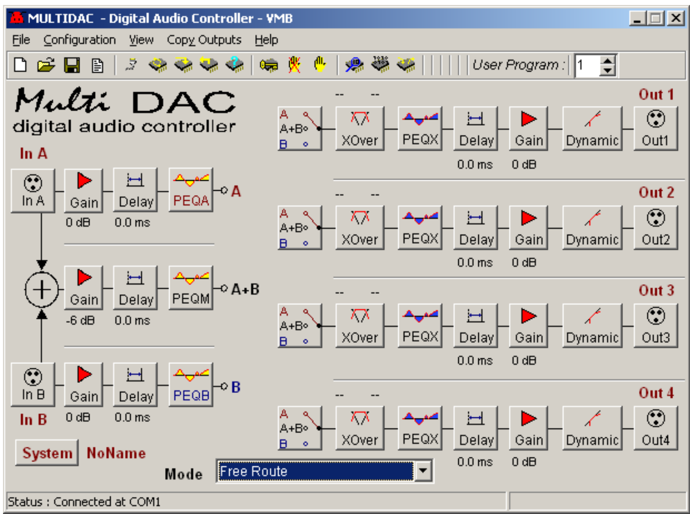
Selecting COM1 or COM2 makes the communication with the processor effective; the communications screen will then appear. If the communication has been set correctly, this screen will disappear immediately giving way to the main screen of the program. If there is any problem in the communication, an error message will appear.

This message informs you that the communication with the processor has not been achieved. In this situation you have to check that the selected serial port is the correct one and is still vacant and that the processor is in its main menu.

CAUTION

The processor must be in its main menu to communicate with the computer.

When the communication has been achieved, you get to main menu of the MultiDAC Software which shows the process scheme.



2.3.- Software removing.

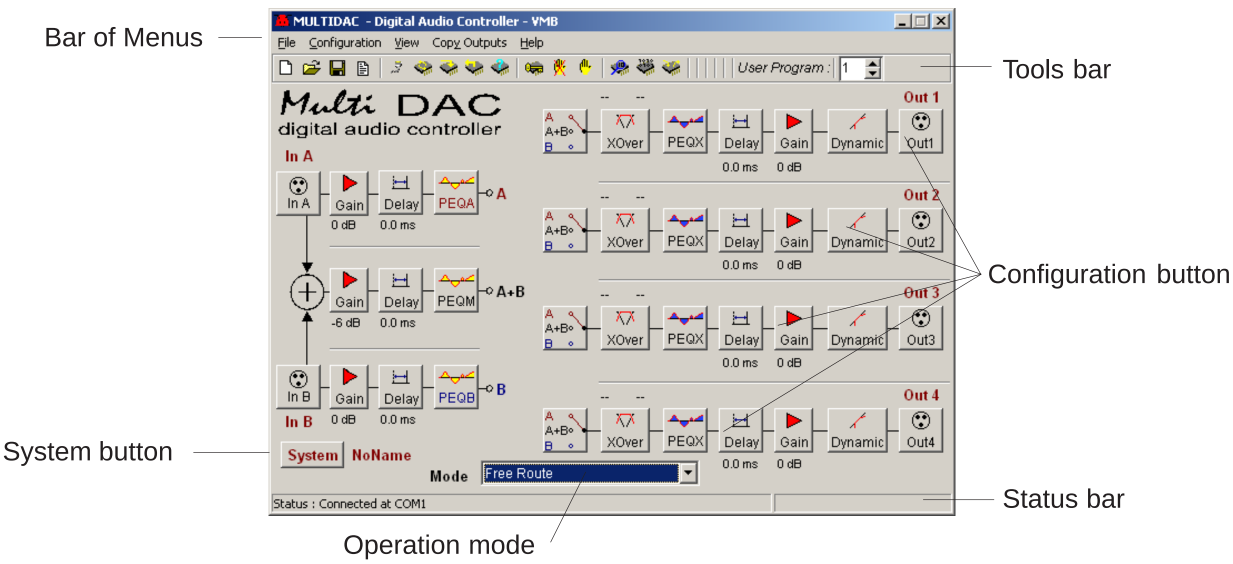
To remove the MultiDAC software, follow the same steps of any other program. From the Control Panel, go to Add/Delete Program. Select the MULTIDAC and press the button Add/ Delete…

It is recomended to keep a safety copy of all the configuration files that have been created (with extension *.dac) to be able to use them at a later date.

3.- Edition with MultiDAC

3.1.- Main screen

When entering the **MultiDAC** software, the main working screen appears, showing the internal processing scheme. You can visualise the route followed by the audio signal from the input to the outputs, you will see the different proceeding blocs and the order in which it crosses them. From this main screen you can access directly to all the configuration parameters. All the elements you will find on it are the following:



If the software starts without being connected to the processor, the buttons of the Tool's Bar in relation with the processor communication will remain off and appear in grey colour. The Status bar will show: *Offline*.

Thanks to the **configuration buttons**, you can get to any configuration screen such as Gains, Delays, Equalisation,…

In the **tools bar** you can get to options such as Read, Create, Save configurations and all the options related with to the communication with the processor: connection, DSP Operating System updating, Password activating, keyboard locking and storing configuration in memories.

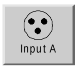
The **Status bar** informs you of the kind of communication established with the processor: Offline, COM1 or COM2.

The **Menu bar** gives access to the different menus:

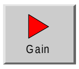
File (working with files and getting out of the program), Configuration (communication options), View (access to configuration screens), Copy Outputs (copy configurations from one output to other) and Help. The **System button** allows you to name the configuration you have created and to add more information such as the name of the project, installation, name of the engineer and comments. Thanks to the list of the **Operation Mode**, you can select the kind of configuration you want to use: 3 or4 way mono, 3 way mono with Aux., 2 way stereo, 2 way stereo with sub mono and Full range or Free route in order to be able to configure freely.

The following list explains and gives a detailed description of the configuration buttons.

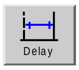
Configuration buttons



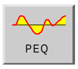
INPUT and **OUTPUT**: They give access to the window “**Input and Output configuration**“ in which it is possible to give a name to the inputs A and B, and outputs from the first to the fourth as well as give a name to the complete configuration. These names are also stored in the processor appearing subsequently on the display.



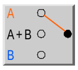
GAIN: It gives access to the window “**Input and Output Levels**“. Form this window you can configure all the input and output gains. You can also make them mute and invert the phase of outputs.



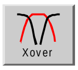
DELAY: It allows to configure input delays (by pressing the buttons located at the inputs) and output delays (with the output buttons). For each input you have at your disposal 290 ms. (about 100 meters) to use as a main delay line meanwhile you have 10 ms. (3,5 meters) to correct the position of a multiway equipment cabinets or transducers inside a cabinet and correct the phase difference for not being in the same vertical plane.



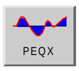
PEQ (global equalisation of inputs per channel).You access the configuration of equalisation parameters for inputs **A** and **B** and signal mono **M** internally generated. There are 6 filters in each input completely configurable; these are: Parametric, Shelving low 6dB and 12dB/oct, and shelving high 6dB and 12dB/oct, Low pass, Band pass, Reject band and first and second order Allpass. They will provide the global equalisation of the audio equipment or PA.



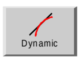
ROUTE. With this button, you will select from which input each output will take the signal: from the input **A**, **B** or mono **A+B**. If you select the **Operation mode** from the list, the Route buttons will get configured according to the selected mode. It is possible to modify the route only when selecting the **Free Route** mode.



XOVER (Crossover). Pressing this button you access the window corresponding to the Crossover configuration, where it will possible to configure the frequency dividing filters of each output. Filters available are: **Linkwitz-Riley**, **Butterworth**, **Bessel** and **Custom** (with resonance frequency and quality factor Q editable of each second order filter) up to **48 dB/oct**. It is also possible to let them in **Bypass** and have a full range output.



PEQX (output channel individual equalisation). You access the configuration for the individual equalisation parameters of each output channel after the CrossOver. There are 6 filters in each output with the same features as the one of PEQ. In this window you will see the effect of the different filters and the gains in the total input-output frequency response, as well as the effect of delays and the response of the loudspeakers until you will get the required electro-acoustic response.



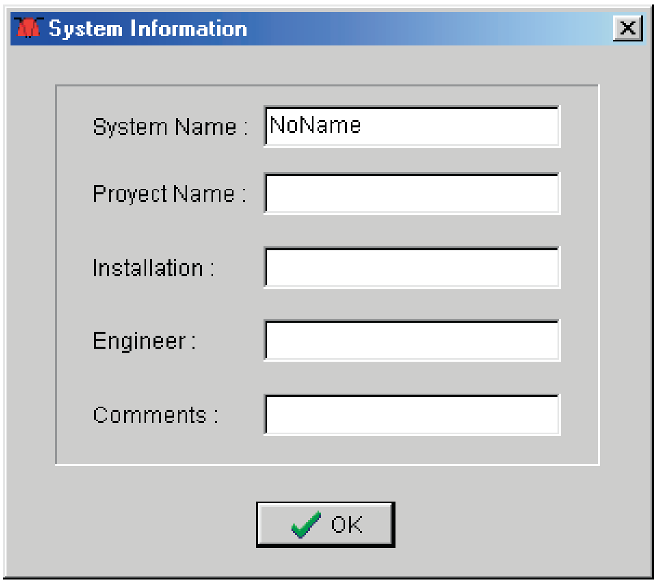
DYNAMIC. This button allows you to access the window “**Dynamic configuration - Noise Gate**”. In this window you will configure all the parameters of the compressor/limiter and individual noise gate per output such as thresholds, ratio, gain, C.R.I. knee and Attack and Release times.

SYSTEM. It gives access to a general identification window where you can give a name to the system in which you are working, the project it belongs to, the installation, the person in charge of it and also to introduce your comments.

3.2.- System Information

Now that you are familiar with the general meaning of the buttons of the main screen, you can begin to work with the program. Fist of all, to prevent future omissions in the configuration you will create, you have to give a name to each project, or equipment. You will achieve this by using the SYSTEM button.

Press this button. The “**System Information**” window appears:



You will find several blank sections to fill in:

System Name: The default name of the system is NoName. You have 12 characters at your disposal to enter a name that has to be quite significant for you to remind the configuration in the future. This name will be the one to appear on the processor’s display to identify the configuration.

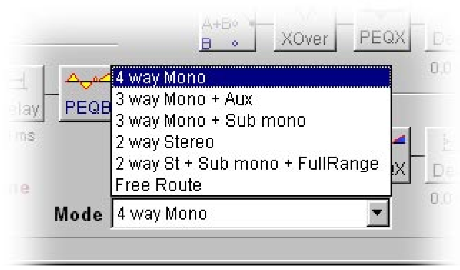
Project name: Name of the project it belongs to.

Installation: Name of the installation.

Engineer: Name of the engineer or technician in charge of creating the configuration.

Comments: Blank space for you to make a 40-character comment.

3.3.- Mode



Once the system that we are going to work with has been identified, operating parameters have to be defined. The first thing to do is to select the mode according to the system or installation to configure. Available modes are the following:

4 way mono: 4 way mono system. Configuration for PA equipment of 4 ways, which are normally, sub-bass, bass, mid and high and requiring the use of two processors to work in stereo. The signal comes in through Input A. As an example, we can mention the **C2-ARRAY** equipment from **VMB**.

3 Way mono + Aux.: 3 way mono system with the fourth output free for an auxiliar input. This configuration will be used in PA equipment of three ways (bass, mid and high) with input A. In this case, the fourth way remains free and its signal starts in input B with its equalisation, delay and independent dynamic in order to achieve through it an auxiliary consignment, to monitors for instance. The **VMB Concept C2** is an example of 3-way equipment.

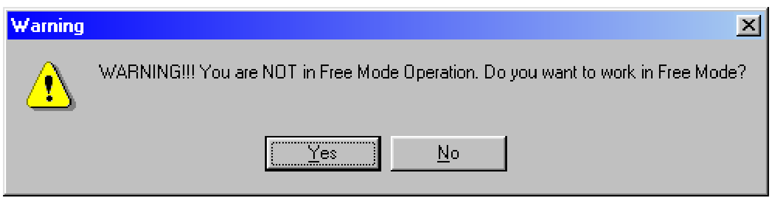
3Way mono + Sub mono: 3-way mono system with sub bass mono (in principle) in the fourth output. This configuration is similar to the previous one but instead of achieving an auxiliar send consignment you will achieve a sub bass mono.

2 way stereo: 2-way stereo system. Outputs 1 and 2 are respectively the bass and mid-high of input A; outputs 3 and 4 are the same for signal from input B. This configuration is recommended for installation using equipment similar to the **Pro** series or **Phase** series of **VMB**.

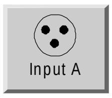
2 way stereo + Sub mono + Full range: 2- way stereo system with common sub-bass in mono and full range in other output. Configuration similar to the previous one but with sub bass in mono and a free output for if it is necessary to achieve another ausiliar send with it.

Free route: it is a free mode. In this case the user is free to chose signal source for each output among inputs **A**, **B** or **A+B**, each one with its gain, delay and independent equalisation. In this mode, Route buttons remain unlocked so that the technician can adjust the configuration to any installation. From this mode it is also possible to get all the others: these are pre-configuration of Route buttons for each output. This is the default mode when starting a configuration.

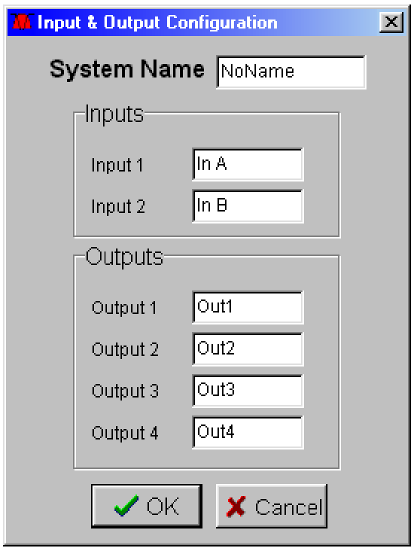
If you want to change the **Mode**, open the mode list and select the one you require: the Route buttons for each output will get configured automatically. If the mode you chose is different from the Free Route mode, the route buttons will remain locked and if you want to modify them the program will inform you that it will be necessary to come back previously to the **Free Route** mode.



3.4.- Input & Output Configuration



Once you have defined the mode you want t work with, you will start to configure each of the operating parameters of the processor. First, you have to identify the inputs and outputs you are going to use. For that, just press the button corresponding to the input or output of the channel you are interested in; the following window will appear:



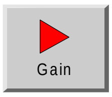
To enter a name in the blank spaces, position the mouse on said space: the writing cursor will appear. Enter the name with the keyboard: 12 characters for the name of the system and 4 for inputs and outputs. Go from one to the other pressing directly the tabulator.
These name will appear subsequently on the processor´s display in order to identify the configuration and inputs and outputs.

For instance in a configuration 4 way mono you would enter IN for input A, -- for B to indicate that you will not use it, and SUB, LOW, MID and HIGH respectively for outputs 1 to 4.

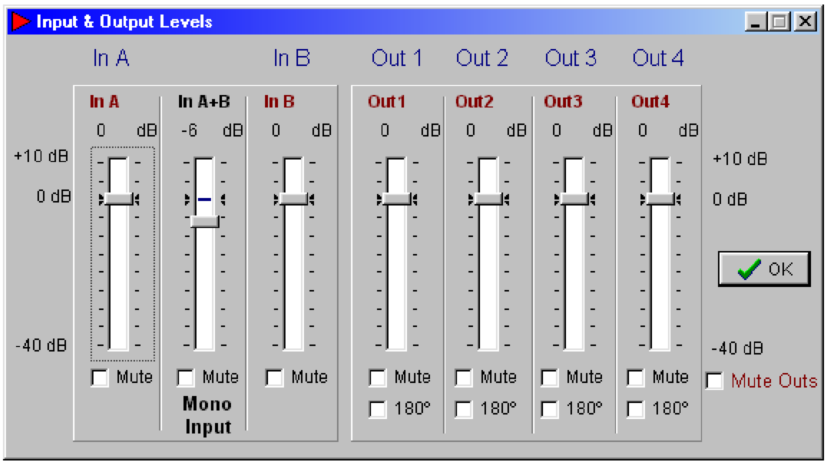
It is convenient to give clear names: since output connectors are numbered 1 to 4 in the rear part of the processor, you will not know which output is which and this way the loudspeakers would be in a dangerous situation (just think about what will happen to a high drive if you change the sub bass outputs for the high ones). When you finish, press OK to confirm the operation (or press cancel to cancel it), and you will come back to the main screen.

Once you have entered a name in each section, the indication of the corresponding button will change and the name you entered will appear below the icon as well as the name of the system will appear next to the System button. This way, you will have a clear positioning of way and/or output on the screen.

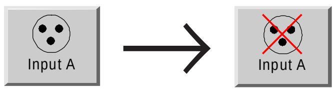
3.5. - Input & Output Levels



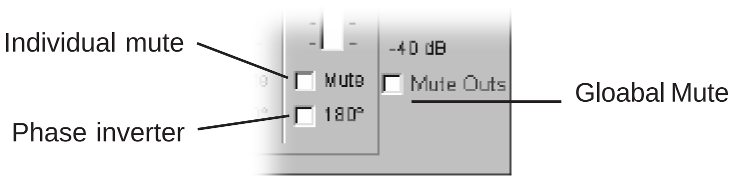
From now on, the mode and the ways you are going to work with are defined. Now you will configure input and output gains according to your needs. Pressing any Gain button and the following window will appear:



All the default gains are at 0 dB except the Mono A+B that is at -6 dB in order to divide the signal per two. You have at your disposal 10 dB gain and -40dB attenuation by 0.1 dB increases. For this, move up or down the potentiometers with the mouse. If you want a more precise control, you should use the keyboard Up and Down arrows from the keyboard to progress by steps of 0.1 dB. If you want the input and /or the output to be mute, click on the Mute space. You can also invert the output phase by activating the 180º selection box. When activating a mute space, the main screen change the corresponding input/output button showing a red cross:



No Mute Mute



The **input gains** allow us to adapt the optimum operating level to the level of the signal coming from the mixing console. If you see that you need a lot of mixer level to have the processor correctly excited, you will have to increase the gain of the inputs used, or decrease it in case you reach at once the maximum operating level with a very light signal from the console. This will depend on the sensitivity of the power amplifiers that are used and the gain or attenuation coming through the equalisation of each processor's output. In a normal situation, with typical sensitivities of power amplifiers (i.e. between 1 volt and ½ volt) and no exaggerated equalisations, inputs will remain at 0 dB.

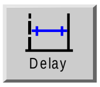
There are two reasons for modifying **output gains**: first, to adjust the output signal level to the connected power amplifier sensitivity. Secondly, to equal electrically the different sensitivities of each transductetrs, for instance, mid and high usually present distinct sensitivity and they have to be equalised acoustically.

To avoid any confusion, you will activate **Mutes** in inputs and outputs you won't use.

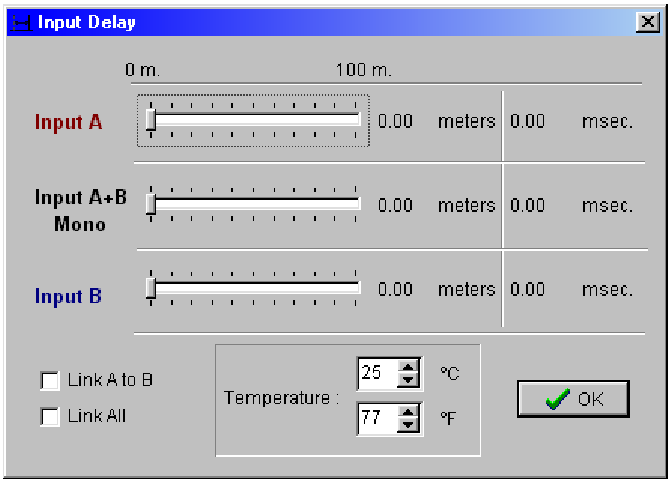
180º Phase inverter will be very useful to solve quickly any problem dealing with phase. In case there is any inverted cable, it won't be necessary to wire again. Also use it when it is necessary to complement the order of the **Crossover** filters in two, 180º, and obtaining phased crossing. It is also useful when the sub bass positioning is not in the same vertical plane as the remaining, leading to phase distortion that can be easily corrected by inverting the phase. Nevertheless, it is recommended to solve all these phase difference problems using output delays since you will have a much better adjustment (in millimetres -grades) and not in mid wavelength (180º).

3.6.- Input & Output Delay

The following button of input configuration is the one dealing with DELAY.



You will also find it in the output area just before the gains. Regarding the **inputs**, this button open the window "**Input Delay**". The configuration window is as follows:



There, you will configure the general delay for channels A, B and input mono A+B. For this, you have at your disposal three potentiometers that allow to adjust the different input delays centimetre by centimetre. When modifying any of them, the delay value expressed in metres and milliseconds will appear on the right side of the window.

If you want a precise adjustment in delays, select the corresponding potentiometer and then, using the right and left arrow keys of the keyboard, you will increase or decrease centimetre by centimetre.

If the selection box **LinK A to B**, is activated, you will make the delays for channels A and B equal and if you modify one of them the other one will perform in the same way. If the **Link All** space is activated, all the input delays will coincide. When coming back to the main screen, the value of delay you have entered will appear below the corresponding delay button of the main screen.

There is also an option called **TEMPERATURE**. With this option you can enter the approximate operating temperature (degree Farhenheit/Celsius) for the calculation of the speed of sound to be as precise as possible. To calculate the **speed of sound Vs** according to the temperature, apply the following formula:

$$\mathbf{Vs} = 20.06 \times \sqrt{(273+{}^{o}C)}$$

ºC corresponds to the temperature in degree Celsius. From this formula and considering an average temperature of 20ºC (68ºF) we obtain the following useful ratios:

Delay in milliseconds = Distance in meters x 2.192

Delay in milliseconds = Distance in foot x 0.955

Distance in meters = Delay in milliseconds x 0.456

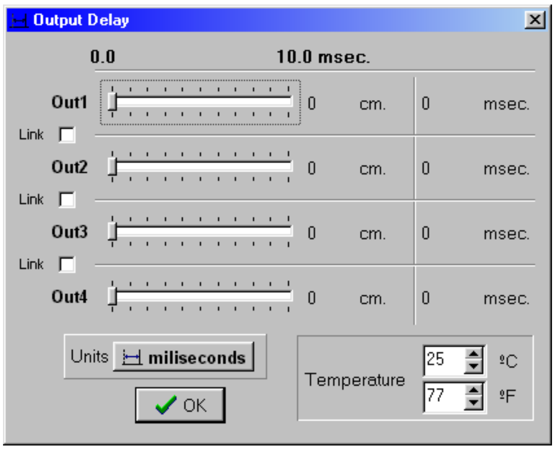
Distance in foot = Delay in milliseconds x 1.047

Keep in mind that if you want to change from degrees Celsius to degrees Fahrenheit you have to apply the following formula:

$$ºC=(ºF-32)x5/9.$$

In each input there is an available delay up to **100 meters** (291.23 milliseconds at 20ºC). These delays will be used as main delay lines in case the audio equipment performs like an audio reinforcement equipment in a big concert quite far from the stage.

There is also a delay button in **outputs**. The following window appears to allow you to configure output delays.



Now you have the 4 potentiometers of the 4 outputs at your disposal. On their right you can see the delay that has been entered in centimetres and milliseconds.

With the button Units, you can make the potentiometer to work in milliseconds or in centimetres. You can operate by steps of 0.5 cm using the left and right arrows. You can also adjust the temperature in the same way than in the input delays.

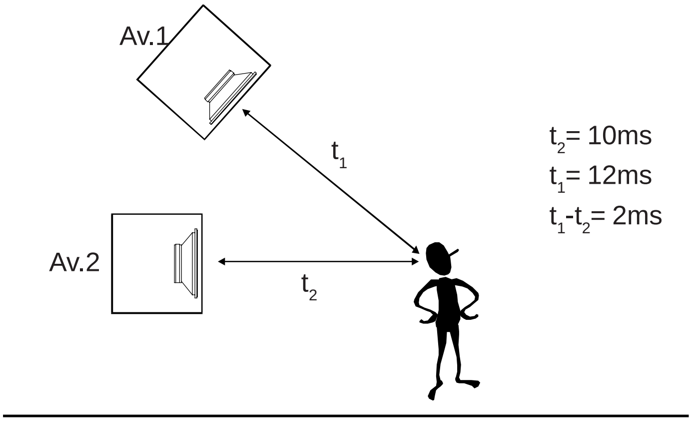
On the left you will find 3 spaces for **Link** selection that allow to unify all the adjacent output delays.

For instance, if you select the last Link, delay of output 3 and 4 will act in unison and if you modify one of them the second one will act in the same way.

Once delays are configured, they will appear expressed in milliseconds on the main screen below each output delay button.

In each output you have up to **10 milliseconds** at your disposal (3.6 meters at 20ºC). These delays will have different uses as you will see below.

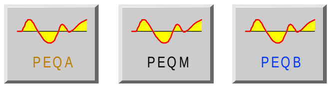
- Correction of the vertical position of cabinets in a multi way equipment. This correction is useful when cabinets are not in the same vertical plane or when part of the equipment (for instance, mid high frequencies) is elevated and that the remaining equipment (low frequencies) is on the floor, with the resulting difference in wave paths. Anyway, this difference in wave paths will have to be calculated and correcting entering the corresponding delays in the nearest ways.



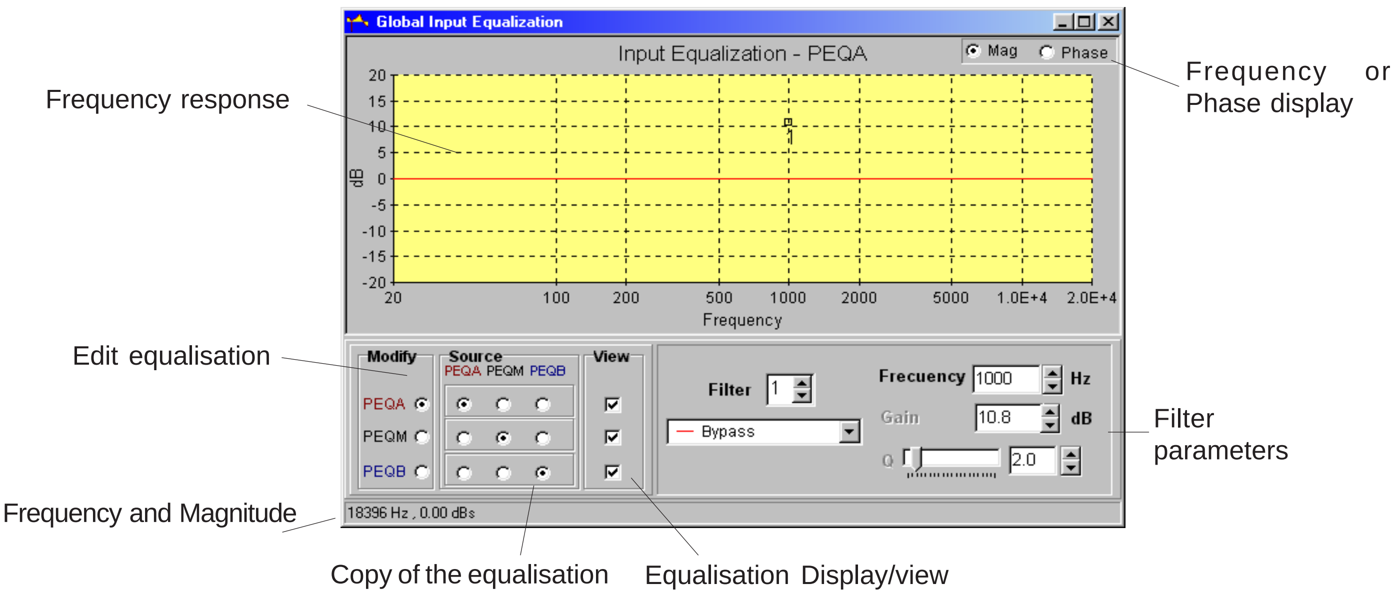
- Precise phase adjustment among the transducers of a multi way equipment. Normally, the positioning of the different transducers of equipment does not coincide exactly in its vertical plane (for instance, a 12-inch loudspeaker and a 2-inch drive in the same cabinet). Furthermore, the order of filters and the phase response of each transducer do not usually coincide either. In this case, you will be able to correct this defect with a 0.5-cm precision (less than 8º at 1500 Hz) until obtaining an output completely in phase with the 2 transducers and having a good link and a coherent wavefront.
For this, it is recommended to use the measurements of each transducers taken in the same conditions and to go to the PEQX screen to see the effect in the final electroacoustic response, seeing at the same time the response of the two ways and its total amount response, where you will change the delay until you get a completely closed cross response.
It is also possible to do it observing frequencies closed to the cross in a spectrum analyser (with pink noise as an excitation signal), changing the delay until you see that the response closes and that the cross is in phase.

3.7.- PEQ - Global Equalisation

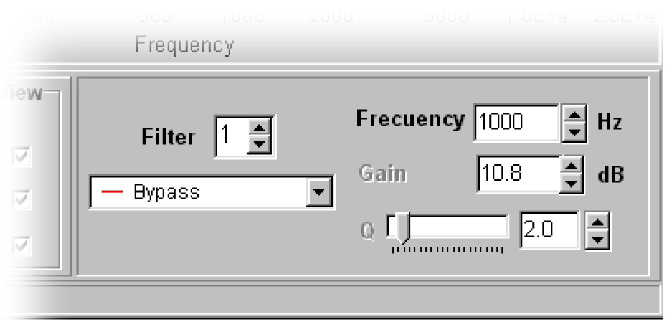
After the input delays let's see the buttons PEQ (channel input equalisation), PEQA (input A), PEQB (input B) and PEQM (Mono A+B), respectively in red, blue and black color characters to differentiate them.



When pressing any of these buttons the following window will appear. In this window you will give the Global equalisation to each input. Here are the different parts of the window:



The frequency response is shown on the upper part. Equalisation curves that are activated in the **View** Spaces (Equalisation display) will appear each one in its own colour (PEQA in red, PEQB in blue and PEQM in black). 6 filters completely configurable are available in each input.

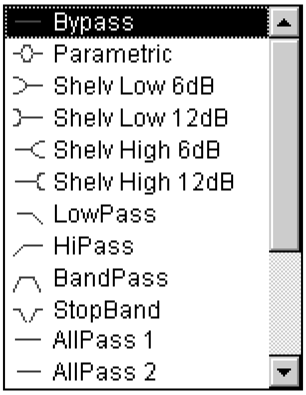


To modify or edit an equalisation follow these instructions:

- In **Modify**, select the equalisation to edit or modify and make sure that the corresponding **View** option is activated.

- A square mark will appear on the equalisation curve showing a number that indicates which of the 6 available filters is being edited. This number will also appear on **Filter** in the filter parameter panel. If you modify the **Filter** valueyou will be able to go through the 6 filters available in each input, updating the remaining parameters to the values of each filter. All the filters are Bypass by default with a 0-dB gain and a 1000 Hz starting frequency.

- Once the filter to be changed has been selected, you have to chose which kind of filter you want to work with. Open the list **Type of Filter**; you will see the, following options:



- **Bypass:** Filter with no effect.
- **Parametric:** Parametric filter that allows to adjust the gain or the attenuation, the performing frequency and the quality factor Q. The value of Q has been defined as a relation between the centre frequency and the bandwidth and the points of mid-gain.
- **Shelv Low 6dB:** Shelving filter to attenuate or emphasize low frequencies by a 6 dB/oct. increase.
- **Shelv Low 12dB:** Shelving filter to attenuate or emphasize low frequencies by a 12 dB/oct. increase.
- **Shelv High 6dB:** Shelving filter to attenuate or emphasize high frequencies by a 6 dB/oct. increase.
- **Shelv High 12dB:** Shelving filter to attenuate or emphasize high frequencies by a 12 dB/oct. increase.
- **LowPass:** Second orden lowpass filter of Butterworth type.
- **HiPass:** Second order hipass filter of Butterworth type.
- **BandPass:** Bandpass filter with frequency, Q and gain variable.
- **StopBand:** Stopband filter with frequency and Q variable.
- **Allpass 1:** First order AllPass. It shifts 90 degrees at the selected frequency.
- **Allpass 2:** Second order Allpass filter. It shifts 180 degrees at the selected frequency with a phase change controlled by the Q value.

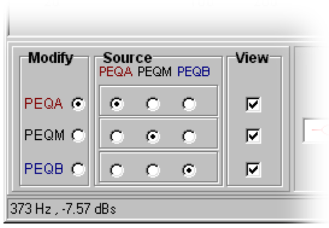
- According to the kind of filter you have chosen, the parameters to be modified in each case will get activated: **Frequency**, **Gain** and **Q**. All the parameters can be edited from the keyboard by entering the data numerically and updating them pressing Enter, up and Down arrows and exiting from the control.

- It is also possible to modify the **Frequency** and **Gain** values with the **mouse**: just place the mouse near the filter mark until the icon becomes a hand.

Click without releasing the button of the mouse in order to see the filter moving to the place you want.

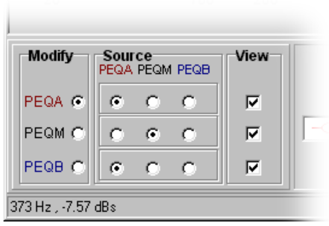
- If you need another filter, just go to another one in Filter. To modify any of them you will also go to Filter until getting to the required filter.
- You can also see the designed equalisation phase going from **Mag** to **Phase**.
- In the information bar situated at the bottom of the screen, there is always an indication about the frequency in which is located the mouse and the magnitude in dB or phase in degrees of the current equalisation.

Once you have achieve the desired equalisation in a channel, you can copy easily this equalisation on



another channel without having to copy one by one the programmed filters. For this, go to the option Source on the bottom left of the PEQ screen. Here you can associate to each input the equalisation you want: PEQA, PEQB or PEQM.

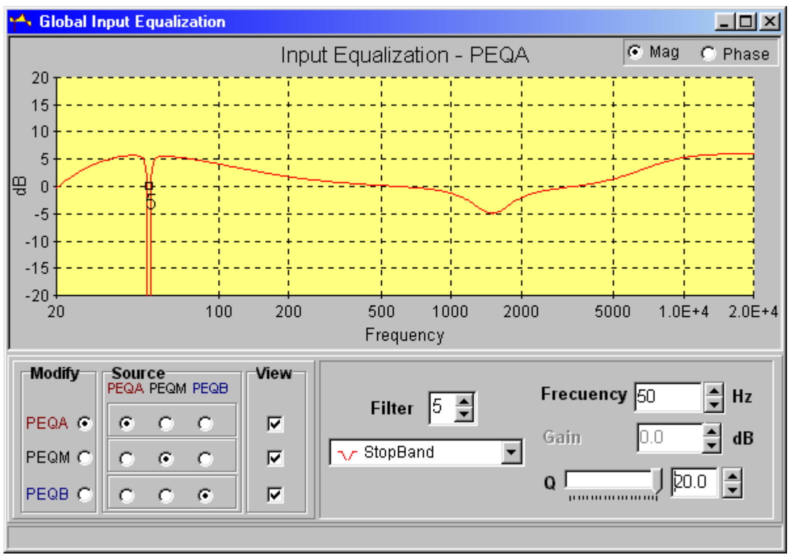
At the beginning, each one is associated to itself but, for instance, if you work in stereo you will require PEQA and PEQB to be exactly the same. If you click on the Source line of PEQB in the first option on the left which is actually PEQA (see the illustration),



you will associate PEQB to PEQA, showing it in the main screen too, PEQA appearing in red colour in both inputs.

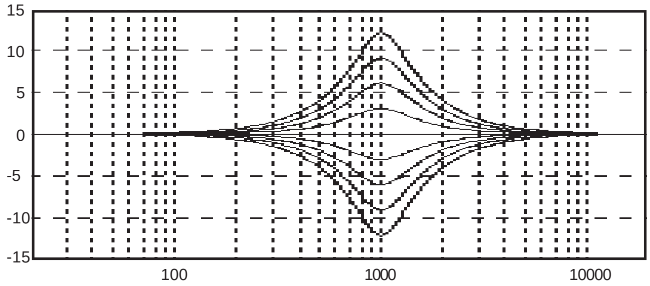
Thanks to this easy selection box you will also make each input to use the equalisation you want. The input equalisation of the Modify line will be assigned to the selected one in the Source column.

Here is an example of equalisation with a screen including 5 filters:

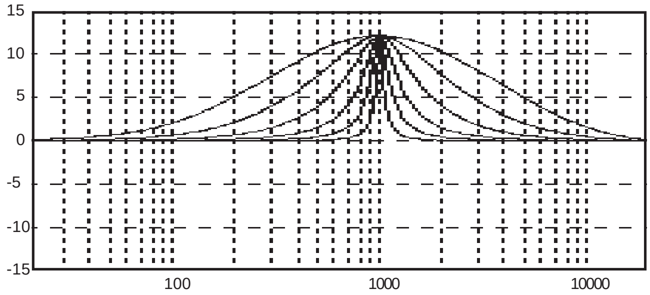


The frequency response of Paramtetric and Shelving filters availables are shown on the following diagrams:

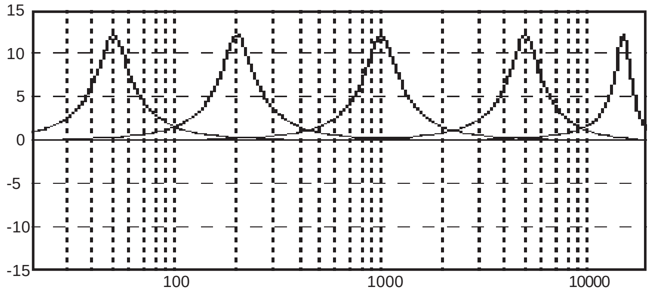
Parametrics
G=12, 9,6,3,0,-3,-6,-9,-12
f=1000
Q=1



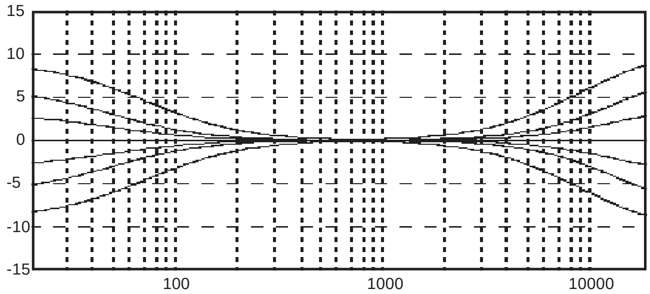
Parametrics
Q=0.25,0.5,1,2,4,10
f=1000
G=12



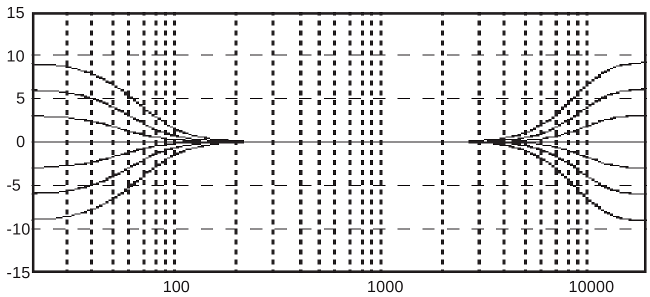
Parametrics
f=50,200,1000,5000,15000
Q=2
G=12



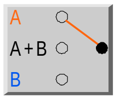
Shelving 6dB/oct
f=50,10000
G=9, 6, 3, 0, -3, -6, -9



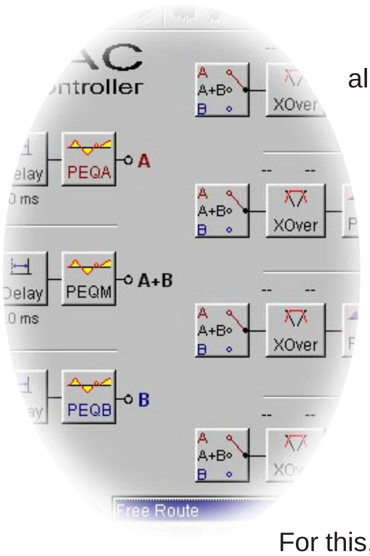
Shelving 12dB/oct
f=50,10000
G=9, 6, 3, 0, -3, -6, -9



3.8.- Route



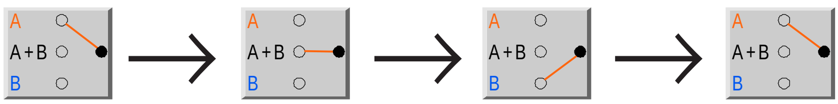
After the equalisation section PEQA, PEQB and PEQM, the signal appears in points named **A**, **B** and **A+B** as can see on the illustration. This point represents the end of the input section and the beginning of the output sections.



With the **Route** buttons you will select which input signal is taken by each output allowing you to obtain any operating configuration you want:

- 4-way mono processor with all the outputs in the input A, B or A+B.
- 3-way mono processor with any signal input plus the fourth independent output.
- 2-way stereo processor with inputs in A and B.
- 2-way stereo (input A and B) processor with sub-bass in mono (A+B) plus the fourth independent output.
- Audio distribution system with each output taking the signal from A, B or A+B according to the needs.
- Etc.

For this, each time a **Route** button is pressed, it will go from A to A+B, from A+B to B and from B to A.



You will be able to see for each output which input is providing it with the signal depending on the switch position.

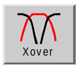
To modify the signal route with the **Route** buttons, <u>the Operating **Mode** has to be in **Free Route** mode</u>, otherwise the program will show pre-established positions for the converters according to the selected configuration.

These pre-configurations are: (the input taken by each output is indicated):

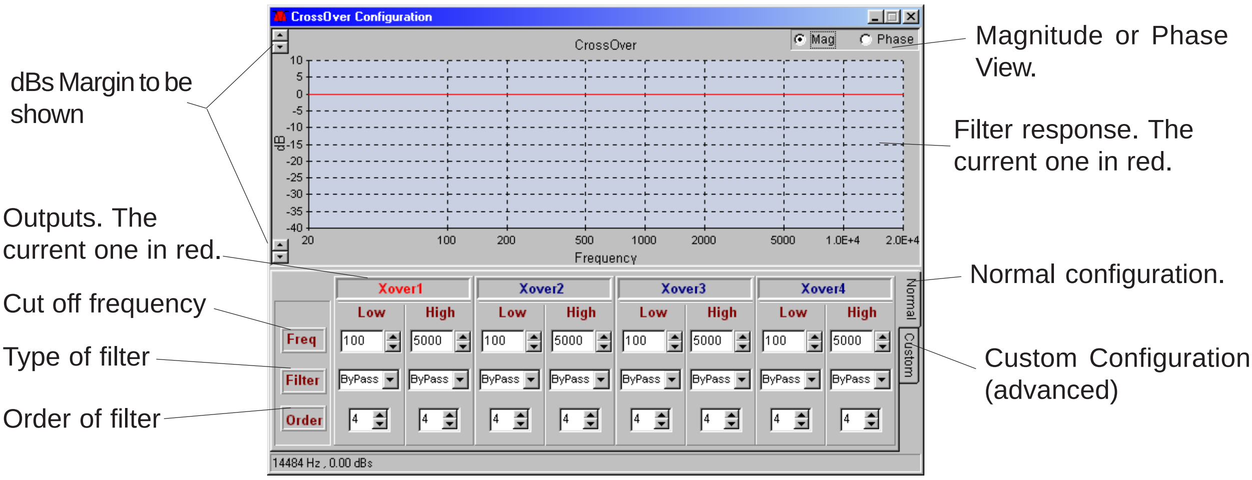
- **4 Way mono** : Out1 A, Out2 A, Out3 A, Out4 A.
- **3 Way mono + Aux** : Out1 A, Out2 A, Out3 A, Out4 B.
- **3 Way mono + Sub mono** : Out1 A, Out2 A, Out3 A, Out4 A+B.
- **2 Way stereo** : Out1 A, Out2 A, Out3 B, Out4 B.
- **2 Way stereo + Sub mono + Full Range** : Out1 A, Out2 B, Out3 A+B, Out4 A.
- **Free route** : Free mode.

3.9.- Crossover Configuration

Once the input signal (A, B or A+B) for each output has been selected with the **Route** button, you will find the separating filters for crossover bands.



When pressing the Xover button of any output, the window "**Crossover Configuration**" appears showing the following elements:



Normal configuration

The four columns from **Xover1** to **Xover4** represent respectively the Crossover filter configuration of the four outputs. Each of them is divided into **Low** and **High** in order to differentiate between **Low** cut-off (Hipass filter) and **High** (Lowpass filter) in which you will configure the cut-off frequencies **Freq.**, the type of filter used for the Crossover **Filter** and the order of each filter **Order** to adjust the curve.

To edit the values just go to the desired edit box and enter values with the keyboard or change them with the mouse with the Up and Down arrows beside the numerical value. The change of value will be effective when pressing Enter or leaving the edit box.

The output we are editing will appear in red at any time as well as the corresponding phase or frequency response curve and the XOver label. Just like in PEQ, the information bar shows the frequency and magnitude or phase of the current crossover filter wherever the mouse's position.

Through **Normal** configuration (that you can select on the right-hand side of the screen), you can design standard and traditional crossover filters. The following filters and values are available:

• Bypass	Without filter
• Linkwitz-Riley	12, 24 and 48 dB / Octave
• Butterworth	6, 12, 18, 24, 30, 36, 42 and 48 dB / Octave
• Bessel	6, 12, 18, 24, 30, 36, 42 and 48 dB / Octave
• Custom	Customizable filter. Custom configuration.

To select to the crossover High-pass and Low-pass filter of each way, select the desired option from the scrolling bars **Filter** and **Order** (the resulting filter slope is 6*order dB / octave).

If you do not want to activate the crossover or one of its filters, just leave the kind of filter **Filter** in **Bypass** where desired. If you do not want it to have any effect on the bandwidth output signal just leave **Low** and **High** in **Bypass**.

The **Low** and **High** frequency cutoffs of the filters are defined in **-3 dB** for **Butterworth** and **Bessel** filters and in **-6 dB** for **Linkwitz-Riley**. In case you use the same kind of filter and order in the superior cutoff of one way and an inferior cutoff of another one with the same frequency cutoff, and if you want to obtain a correct electric add response, it will be necessary to invert the phase of the way with lowest frequencies. These cases are the following:

- Butterworth of 2nd and 6th order.
- Linkwitz-Riley of 2nd and 6th order.
- Bessel of 2nd, 3rd and 4th order.

The ideal total electric response is obtained in the Linkwitz-Riley case. Butterworth is showing a 3 dB boost or cut, and Bessel a +/- 2 dB boost or cut.

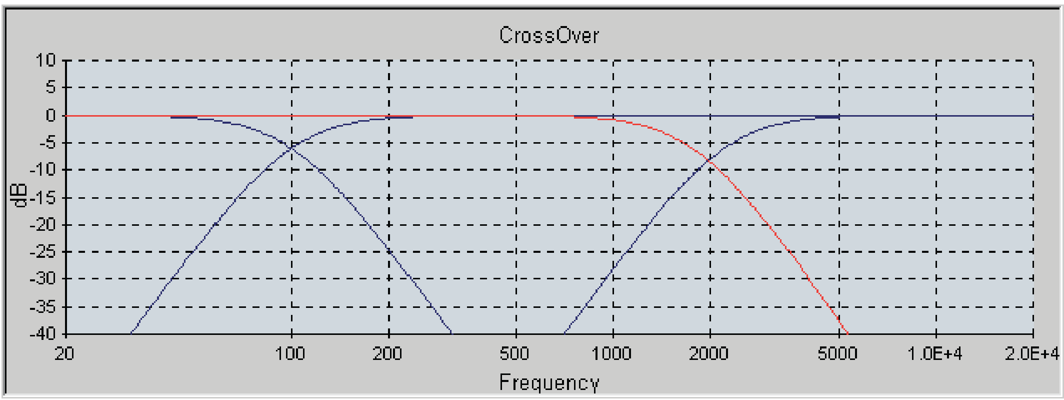
Nevertheless, you will obtain a response improvement in the crossover area since some typical possible phase difference problems are being solved in this kind of configuration. Anyway, **the response of the transducer has also an influence on the final phase response, do not forget it**.

In the case of Bessel, the high cut-off frequency is limited by design restrictions depending on the order of the filter. This limits are (from first to eigth order) : 20000, 18800, 16500, 14900, 13650, 12550, 11650, and 10800.

It is also possible to get nearly flat links with some ripple in the crossing zone using the same kind of crossover filter and order but modifying band frequency cutoffs, lowering a little bit the Lowpass filter cutoff and increasing the Hipass cutoff.

For instance, for a 1000Hz link with Bessel filters of 4th order, you should bring down the Lowpass filter to 875 Hz and bring up the Hipass one to 1250 Hz. You will check all this in the **PEQX** screen enabling the crossover effect and visualising only two outputs of the crossover, enabling the **Add** respons.

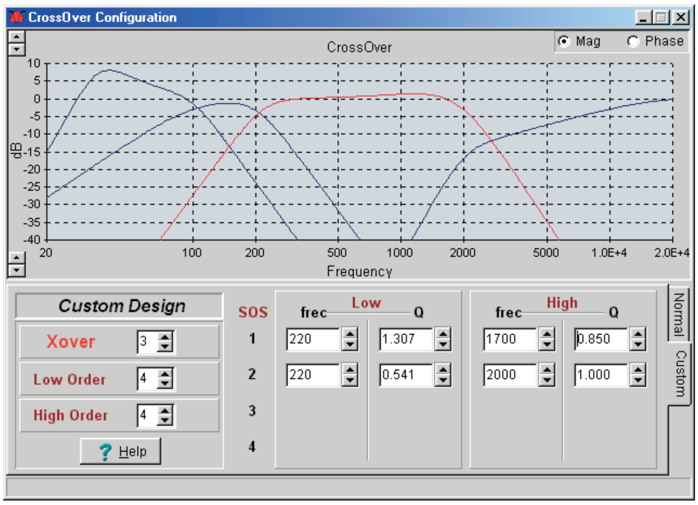
Since the Low and High cutoffs of each crossover filter can be adjusted independently, you will feel free to create **asymmetric crossovers** with different frequencies, slopes and type of filters which is necessary in many configurations. In the following screen you will see an example of symmetric crossover at100Hz with Linkwitz-Riley 4th order filters (with link at 100 Hz -6dB), an one asymmetric over 2000, with the low cut at 1750 Hz and the high cut at 2200 Hz, both of Linkwitz-Riley type. With this, the electrical link is at 1950 Hz with -8.3 dB. In general, this kind of asymetric crossovers will achieve the final electroacustical filter objective.



Custom Configuration

MultiDAC incorporates a **Custom** crossover Mode completely customizable much more versatile than the **Normal** Mode. It allows to customise crossvover filters taking into account the own response of the transducers.

CAUTION:
THE CUSTOM MODE IS AN ADVANCED WORKING MODE REQUIRING GREAT ENGINEERING KNOWLEDGE ABOUT HIGH ORDER ELECTRONIC FILTERS AND THEIR LINKAGE WITH THE FINAL ACOUTOELECTRIC RESPONSE. IF THE USER HAS NOT THIS KNOWLEDGE, IT IS HIGHLY RECOMMENDED TO WORK IN NORMAL MODE AND USE THE PEQX EQUALISATION FOR FINAL CORRECTIONS.



In the **Custom** mode, the engineer who is adjusting a multi-way sound equipment has the possibility to **configure** in each crossover filter the **resonant frequency** and **Q** (Quality factor) of each filter of 2nd order that define the whole filter.

Therefore, the obtained crossover filters can be much more complex and allow to adjust its response much closer to the own response of the transducer without using more PEQX filters in the final correction. The final filter order is then smaller with the consequent advantages in what phase response and group delay are concerned.

To work in **Custom** mode, it will be necessary to set the kind of **Filter** from **Normal** Mode to **Custom** otherwise the program will not permit modifications in parameters. It will only show on screen current values of frequency cutoff and Q of each filter of second order in the selected option (Butterworth, Linkwitz-Riley or Bessel). In case the order of some filters is odd, the program will also show the value of the frequency cutoff at -3 dB of the first order filter, while the Q box disappears.

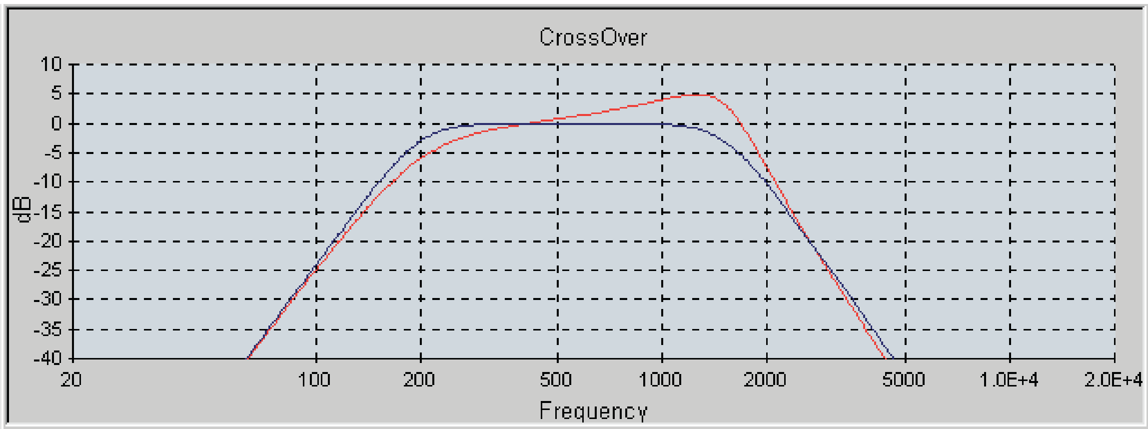
Once the filter is set as **Custom** mode, you can go to the Custom tab and modify the values of each filter of second order independently. You will also be able to modify the values of each filter in **Low Order** (low cutoff, Highpass) and **High Order** (high cutoff, Lowpass) as well as selecting the crossover filter to be modified in the **Xover** box. Just as in the Normal Mode, the activated crossover filter will appear in red and the information bar will show Frequency and Magnitude or Phase values of the current crossover filter wherever the mouse is over.

As it is possible to modify independently each frequency and each Q of second order filters of the whole filter, responses that can be obtained can be much more adapted to the own response of the transducers than when using the **Normal** mode. It may be convenient to start the design of a filter in the Custom mode with the frequencies and Q values of a Normal filter such as Butterworth, Linwitz-Riley or Bessel, change to Custom Mode and modify those values according to your needs.

Here is an example: just think in a medium way between 200 Hz and 1700 Hz working for mids in a horn. After analysing the loudspeaker response, we note that this one is showing a decrease in the high part of its response from 1000Hz and we want to extend this response up to 1700Hz for the treble driver cut at a higher frequency. Also the low band has a boost that will need to be attenuated. Starting from a 4th order Butterworth filter, we can arrive from the Butterworth response to the desired modifying these values:

Starting values (Butterworth)	
frec	Q
200 200	1.307 0.541
1700 1700	1.307 0.541

Custom final values	
frec	Q
200 200	0.9 0.541
1250 1500	0.89 2.1

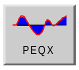


As you can see, the Custom response gets much more adapted to the loudspeaker with no need of using parametric correcting filters in PEQX. At the same time, since the complete filter order is smaller, you obtain a better response in frequency, phase and lower group delay.

When using correctly the crossover **Custom** configuration together with the possibility of importing measurement data from transducers in **PEQX**, you will be able to see directly on screen the final electro-acoustical response. You will also adjust correctly orders, resonance frequencies and Q of each crossing filter as well as you will achieve the final adjustment of the PEQX equalisation on the final response: this is a much more rapid and correct adjustment.

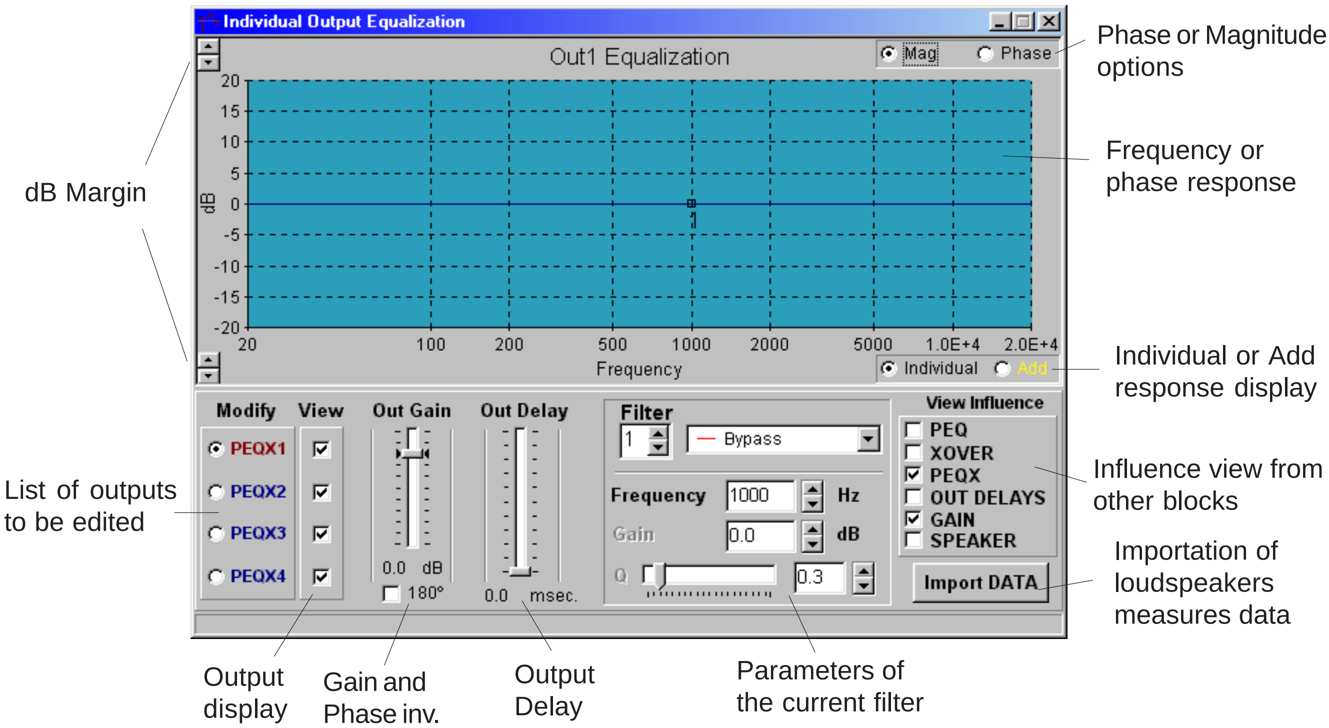
3.10.- PEQX - Output individual equalisation

Subsequently to the **Xover** crossover there is in each output an independent equalisation section called **PEQX** to which you access pressing the button:



You have at your disposal **6 filters** completely configurable per output with the same features than in **PEQ**. Furthermore, you will see on the same screen the accumulated effect of all the processing blocks on the final response such as **PEQ** (input equalisation), **XOVER** (crossover), **PEQX** (output equalisation), **Gains** and even the effect of **Output Delays** and the own **electro-acoustical response** if you import measures from external programs with **Import Data**.

The different elements of the screen are as follows:



To edit the equalisation of each output, select the corresponding output in **Modify** and make sure that its **View** option is activated. Just as in **Xover**, the current equalisation curve will appear in red and the remainder in blue. The working method will the same as in PEQ: select the filter to edit or modify with **Filter** selection box, select the kind of filter from the scrolling list and modify the parameters of **Frequency**, **Gain** and **Q** numerically or with the mouse.

In the View Influence frame you will have the possibility to see the influence of all the other processing blocks on the final response: input equalisation **PEQ**, crossover **XOVER**, output equalisation **PEQX**, output delays **OUT DELAYS**, **GAINS**, and even the own response of the transducer **SPEAKER** if it has been entered in **Import DATA.** This way, <u>you will be able to see the TOTAL electric input-output response of the processor or the final electro-acoustical response if the speaker's data are entered</u>.

In this screen you will modify the output gain of each way with the bar **Out Gain** to make equal the sensitivities of the different connected amplifiers and transducers.

Modifying the gain in this section is the same as doing it from the **Input & Output Gains** screen.

The same happens when modifying the output delay **Out Delay**: you can modify it in this section or in the window **Output Delays**. You will modify this delay to solve phase problems between ways using the option **Add** instead of **Individual View**.

When activating **Add**, you will see on the screen in yellow the individual response of all the outputs with the **View** option activated together with the **Add response**, taking into account both the **magnitude** and the **phase** of all the outputs with the option **View** activated.

If the **Speaker** option is activated with loudspeaker data, you will see the acoustoelectric response of the two ways implicated in the cross. You will also see how is the response in the cross area with the possibility to change the output delay little by little (with up an down arrows) and see how is affected the frequency response until you get the crossing area closed (with the option Delay activated). It is also easy to see the effect of inverting the phase on the response around the cross when activating or not the 180º box.

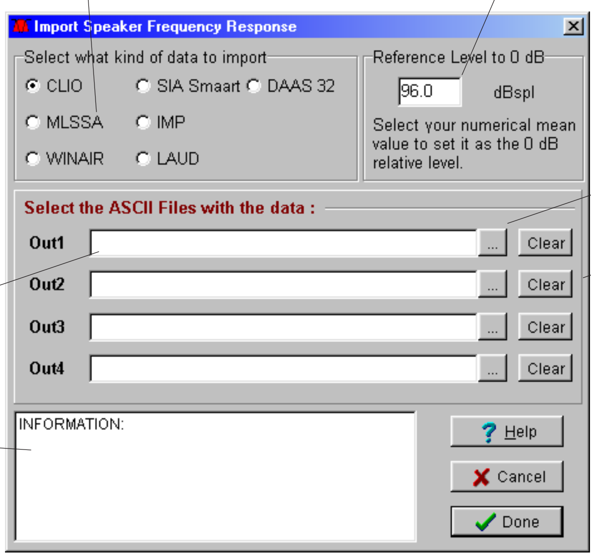
Import Data

One of the most interesting features that differentiates MultiDAC from other products is the possibility to work with real data from the own transducers or loudspeakers. The sound engineer or technician can include this response at the end of the processing and see directly on screen the **final electro-acoustical response** and even equalize over it until to obtain the required final response which can be plane, loudness or whatever.

To enter data it is necessary to have or obtain exported data in an **ASCII format** from one of the following standard data systems:

- **CLIO**
- **MLSSA**
- **SIA Smaart Pro / Smaart Live**
- **LAUD**
- **IMP**
- **WINAIR**
- **DAAS 32**
- **CALSOD**
- **LoudSpeaker Lab**

Pressing the **Import Data** button in the window will allow you to enter ASCII files with the data of the transducer connected to each output:



It is important to adjust correctly the **reference level to 0 dB**. In this box you will enter the average value that you consider you have in the data files. Normally, this value will get adjusted according to the medium way in case of a multi-way equipment or the value of mid frequencies in full-band measures. This value will subsequently be subtracted from all the measures to remain centred around 0 dB on screen. For instance, if you note that mid average is around 96 dBspl, you will take this value as reference. If later on you note some variations, you will modify this value entering again in the Import Data window and updating the reference value until you get the response more or less centred around 0 dB.

The way to obtain files exported in **ASCII** vary from one system to another one. Here is the process and file format accepted in each case:

CLIO :

Make the MLS or sinusoidal measure using preferably the sampling frequency of 51200 and a smoothing of 1/6 Octave for MLS and at least a 1/24 Octave resolution for a Sinusoidal. Once the standard is achieved, export it in ASCII format pressing Shift+F2: this will generate a file with the name you want and an extension *.txt* that **MultiDAC** will be able to read.

The accepted file format is the following:

```
Freq        dB     Phase
20.0       84.7    87.81
20.3       84.5    89.54
20.5       84.4    91.46
20.8       84.3    93.55
21.1       84.3    95.84
21.3       84.3    98.33
21.6       84.4   101.02
 ..         ..       ..
```

MLSSA :

In MLSSA, to get the ASCII file, you will need to make the measure, transfer to FFT to get the response in frequency **Transfer Function Magnitude** and visualise it with some smoothing. Once the frequency response is on screen, go to the option ***T****ransfer,* ***E****xport,* ***B****ode*, select the phase (normally ***A****ctual-phase*), select a resolution of at least 1/6 Octave and give a name to the file, save it with a *.txt* extension and with a format that can be read by **MultiDAC**.

```
"Transfer Function Bode Plot - dB volts/volts (eq)"
     "Hz"  "Mag (dB)"      "deg"
 18.42571,  8.800242,   1.53268
 36.85141,  12.55392,  -120.637
 55.27712,  9.681987, -62.79101
 73.70283,   11.5648,  123.8057
 92.12853,  12.57369,  102.3705
 110.5542,  18.3404,  121.5294
 221.1085,  12.62453, -41.98371
 257.9599,  5.280994, -60.50087
    ...        ...         ....
```

WINAIR :

Select MLS as analysis signal and once made the measures, go to the **Storage** section, enter the name of the file with an *.txt* extension and press the Save button to save it in the directory where the WinAir program is to be found. The saved format must coincide with the following one:

```
WinAIR saved file
Samples per second     44100
  Number of points      1024
       Hz   Amplitude    Phase
   21.533  210.38184     0.00
   43.066   51.68376     1.40
   64.600  116.45122    -1.27
   86.133   42.01191     0.18
  107.666  121.01157     1.14
  129.199   27.09770    -0.13
  150.732   15.25970     0.10
  172.266   19.68722     1.40
  193.799   46.94252    -0.96
  215.332   30.90430     0.24
  236.865   53.93416    -0.58
  258.398   16.91670    -0.79
  279.932   20.37465     1.50
     ...       ...        ...
```

SIA Smaart Pro / Smaart Live:

Sia Smaart Pro Real-Time Module is provided with the option *ASCII Save* in the *File* menu. It is recommended to use 44100 or 48000 as the sampling frequency and a FFT size superior or equal to 1024 points, as well as make the measure with an Averages number greater than 25 in order to have a stable response. The measures must be done using the program in **Transfer Function Mode**.

The measure can be done using White Noise with FFT analysis or Pink Noise if you use a 1/3 or 1/6 Octave analysis. It is also possible to achieve it by using music with a great spectral content and much averages in the measure. The measures must be done introducing in one channel of the sound card the direct signal and in the other one the measure signal of the acoustic system with a microphone and amplified up to line level. In this case it is convenient to adjust the delay of the signal with the option Delay Locator.

In the Sia Smaart case, the file format heading can be changed (lines starting with point and comma) depending on the options selected by the user. Nevertheless, standard format is always the same and must coincide with:

```
; SIA-Smaart Pro Real-Time Module Trace Output
; This File Created On Tuesday September 26, 2000, At 16:32:59
;
; Comment:
;
;
;
; --------------- Transfer Function Trace, Not Smoothed ---------------
;
; Comment: "No Comment"
; Sample Rate: 48000
; FFT: 1024
; Frequency Resolution: 46.9
; Data Window: Welch
; Y +/-: 0.0
;
;
; Frequency (Hz)     Magnitude (dB)  Phase (degrees)
0.0   3.69  0.00
46.9  2.32  2.60
93.8  -2.12 -45.42
140.6 1.49  -77.71
187.5 3.18  -63.61
234.4 -5.06 -7.78
281.3 -0.36 3.60
328.1 6.02  -19.82
375.0 0.54  -6.87
 ..   ..    ..
```

CALSOD / DAAS 32 / LoudSpeaker Lab:

For use the measures made with **DAAS 32** or **LoudSpeaker Lab**, it is necessary to export the files in **CALSOD** format with ***.spl** extension for **DAAS 32** and ***.inf** for **LoudSpeaker Lab**, with the frequency, magnitude and phase information. The format must be like:

```
CALSOD
Frequency resp. NoName               0000                ,
2
0.0
0.0
1.0
0
   13.33   33.80   58.70
   14.41   34.30   58.40
   15.58   35.00   57.90
   16.84   35.70   57.30
   18.20   36.70   56.40
   19.67   38.00   55.30
   21.27   39.30   53.50
```

IMP / LAUD : option not available.
LMS : option not available.
OTHER FORMATS : Please contact with VMB Española S.A.

General considerations to realize the measures

It is very important to realize correctly the measures of the transducers for the subsequent results to be reliable and valid. The following list can be considered as a useful guide to obtain good measurement files.

1 - Make the measures of every way at the same distance (several meters) and with the microphone situated at the height you consider to be the central listening point. This way, the magnitude and phase information is the same for all the measures.

2 - The measure equipment has to be equally configured in every measure, i.e. no change has to be made in signal gains, sensitivity, sampling frequency, etc. from one measure and another one.

3 - It is recommended to make a smoothing in the measure (between 1/3 and 1/6 Octave) in order to obtain measures with fewer peaks. Furthermore, keep in mind that the functioning of the hear is similar to the one of an spectrum analyser of about 1/3 to 1/5 Octave.

4 - Measure introducing directly the signal to the power amplifier you are going to use and in normal conditions so as to include in the measure the gain of each amplifier and not to have to recalculate gain a posteriori.

5 - In case you measure with MLS (Maximum Length Sequence) and windowing the temporal response in order to avoid reflections in the measure, take the same temporal window in all the measures and with a starting time common to all of them. This way you will have the same information of phase in all of them and the final electro-acoustical Add will be correct.

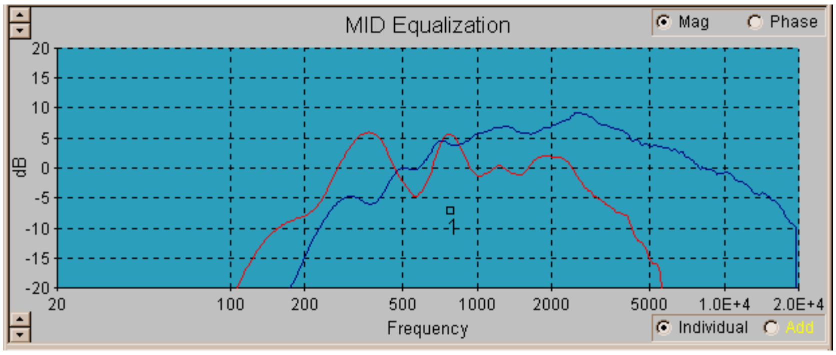
6 - In high and mid frequencies, measures will be nearly always correct in case you measure with MLS avoiding reflections in the measure when windowing time. For low and mid-low frequencies, unavoidable problems in measure reflection and stationary waves will falsify the measure in what phase and magnitude are concerned. In these cases, measures are always to be interpreted because if you want reliable measures you should use an anechoic chamber or achieve them in free field. This matter is more problematic in indoors than in the open air.

7 - Store all the files exported to ASCII in the same directory with significant names for you to remember which measure it is.

8 - In case you use the MultiDAC as a bandwidth equaliser, it is possible to achieve the measure of all the system and equalise on it in the section of input equalisation PEQ to correct the general response.

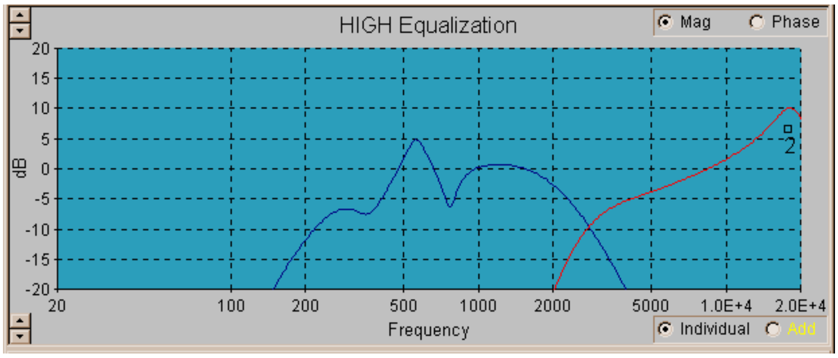
In the folowing screens you will see an example of adjustment of mid and high ways of an equipment using measures from transducers.

First, you will only see the measure, natural response of the loudspeakers in its cabinet and with its corresponding amplifier. The only activated option is the SPEAKER one. You will observe that mid frequencies present a very irregular response and the high driver sensitivity is about 6 dB superior and present a natural fall of 6 dB/oct above 4kHz.



Natural response of Mids and High

Working on measures and according to them, mids are cut at 250hz underneath and at 2 kHz above with Linkwitz-Riley 4th order 24dB/oct filters and high at 2500Hz with Linkwitz-Riley 6th order 36dB/oct filters. Subsequently the equalisation of each output is corrected as follows:



XOVER + PEQX response

Mid:

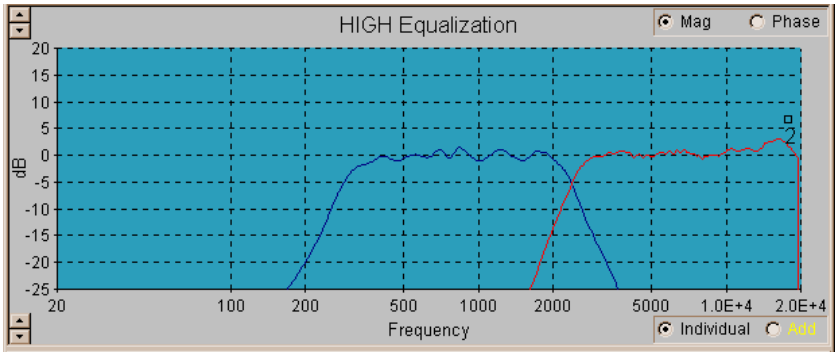
The following filters are used:
1º: Parametric f=365 Hz, G= -6dB, Q=2.4;
2º: Parametric f=558 Hz, G= +6.9dB, Q=3.4;
3º: Parametric f=763 Hz, G= -7.7dB, Q=5;
4º: Shelving High 6dB/oct f= 1307 and G=+6dB.
Output Gain –0.7 dB.

High:

1º Shelving High 6dB/oct f= 12kHz, G=+12dB;
2º: Parametric f=17870, G= +6.5dB, Q=0.7
with an Output Gain of –1.7dB.

The electric response of the processor would be the one shown on the illustration X-over response + PEQX.

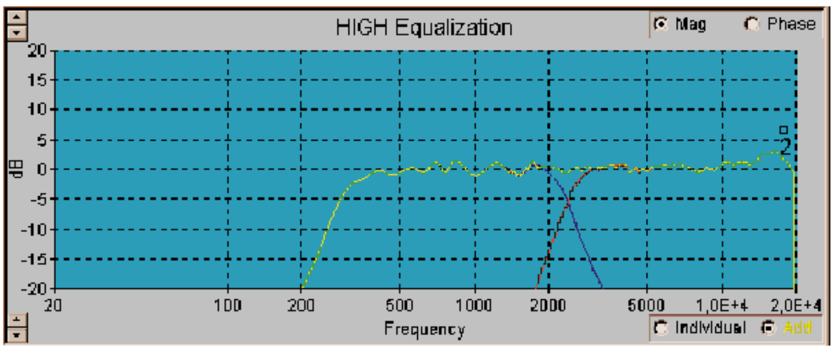
If now you can see also the response of the two loudspeakers and gains, the whole electro-acoustical response remains exactly as it appears on the right hand screen.



Electro-acoustical response in Mids and High

Finally, you have to adjust the high Delay for the crossing acoustic add (2000 to 3000 Hz) to be correct. Activate the option Add and the one to see DELAY; slowly adjust the Output Delay, reducing the response to 0.06ms.

The final response of both ways is the one shown.

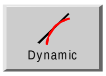


Final Electro-acoustical response

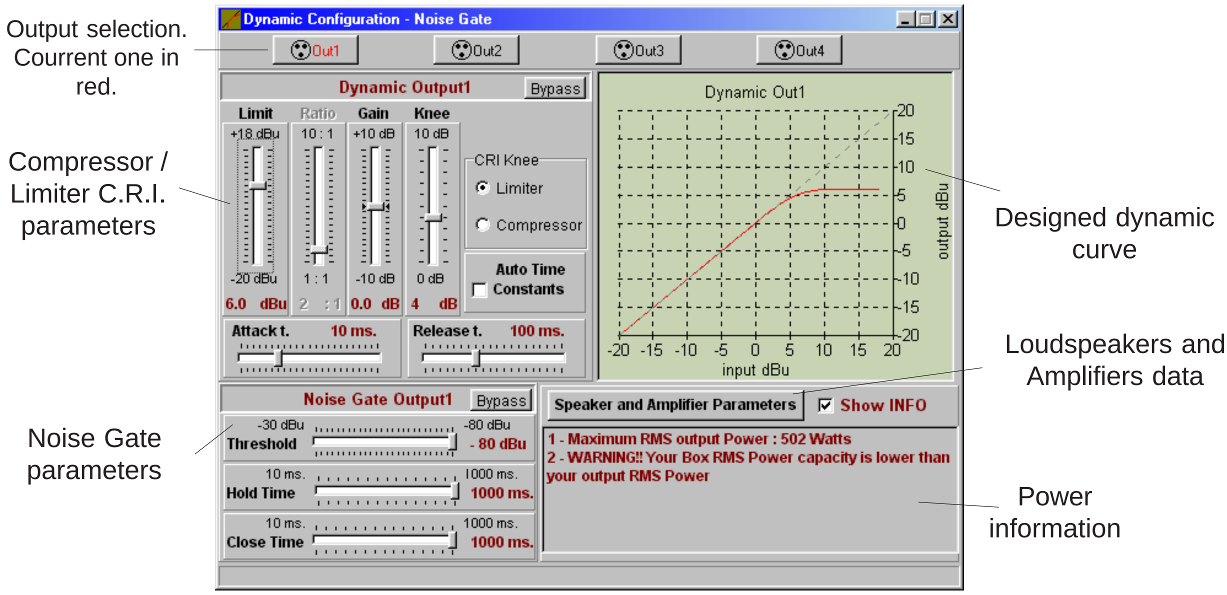
3.11.- Dynamic Configuration & Noise Gate.

Subsequently to the PEQX section about Output Individual equalisation, you will find more information about the *Output Delay* mentioned in section 3.6 and the Out Gains also explained in section 3.5 *Input and Output gains*. Once the signal arrives to this point, it already presents a global equalisation, general delay, Route selection, crossover, output equalisation and gain. The last missing thing is the dynamic control (compressor/limiter) and an independent noise gate per each output.

This way, you will get a final dynamic control and will be able to limit the power provided to each output. To access to this part, just press the button **Dynamic**:



The **Dynamic Configuration-Noise Gate** window will appear pressing any button of **Dynamic**. From this window you will configure the dynamics and noise gates of the four outputs.

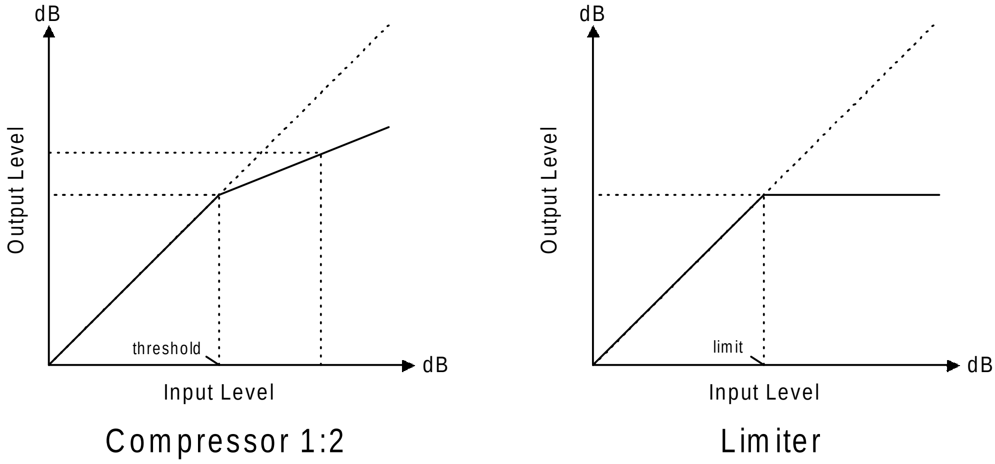


Pressing the top- hand buttons you will access to the configuration of each output. The button in red characters corresponds to the output you are configuring: all the values of the different parameters (limit, ratio, gain, knee, attack and release times, etc…) appear on potentiometers. On the right hand chart is the **dynamic curve** that express the relation there is between the input signal level in dB and the output one.

Compression and Limitation C.R.I. (Continuous Ratio Increment)

Through the use of compressors and limiters what we want is to modify the signal dynamic. We will use compressors to reduce the Dynamic range from a certain threshold and limiters for not to let the signal exceed a preset level.

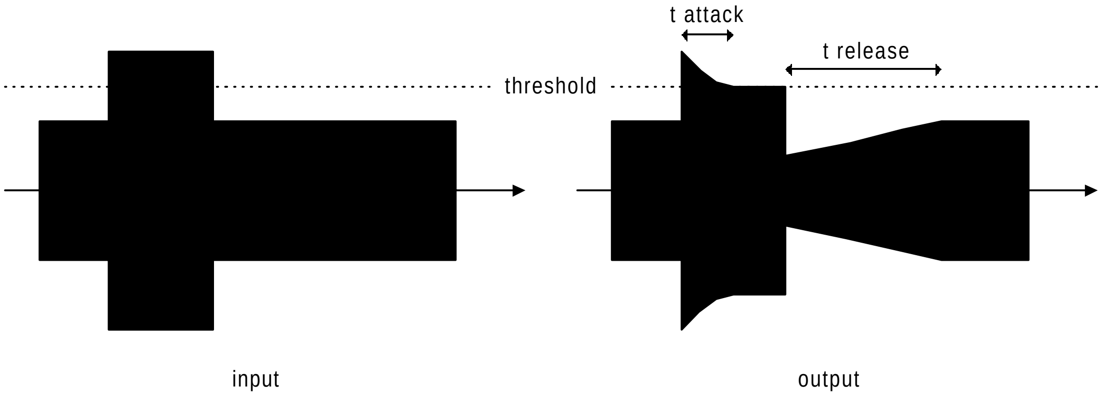
The behaviour of compressors and limiters is described by their Dynamic Curve in which the output signal level is linked to the input signal level.
These curves are shaped as follows:



In **Compressor**, the output signal follows the input one while it is below the **threshold** level. Once it exceeds it, the output lessens according to the selected **ratio**. On the diagram the ratio is 1:2 which means that once the signal exceeds the threshold, for every 2 dB of input increase, there will be 1 dB of increase in the output. The **limiter** behaves as a compressor of 1:∞ ratio, and does not allow that the output level exceeds the threshold **limit**.

In both situations their behaviour is ruled by attack and release time constants that controls the time behaviour. The following diagram shows the functioning of a limiter in time and we can see which is the function of each time.

The input signal appears on the left and the limiter output one on the right.



The **attack time** corresponds to the time the compressor/limiter takes acting and lets the transient of the signal go in without lessening them. Very short attack times lead to lose many transient in the signal loosing the punch sensation. By the contrary, long times let too much signal level go in which leads to some loss in the use of the compressor/limiter.

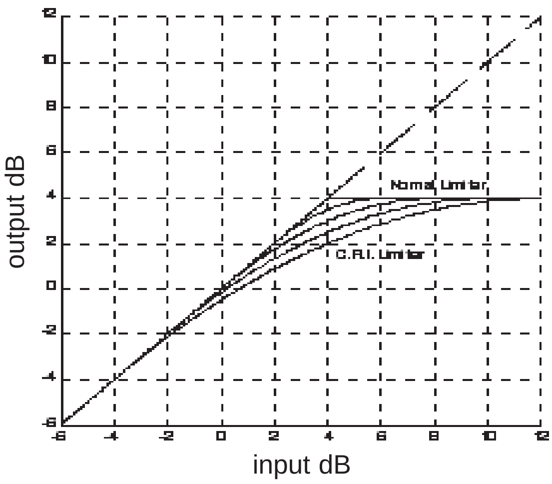
The **release time** is the one that goes in once the input signal comes under the threshold until the compressor/limiter stops acting. Short release times allow to retrieve the original signal level too rapid which lessens when faced to the arrival of higher level signal. This effect introduces much pumping and even distortion in extreme cases producing unpleasant sound. In the opposite case, excessively long release times, we won't be able to retrieve signal level and we will loose dynamics.

The **adjustment of these times** is very important for the final sound quality. As a basic rule of selection of times, these will depend on the frequency range included in the input signal. For low and sub-low frequencies, attack times will be included between 16 and 10 milliseconds, for mid frequencies between10 and 5 milliseconds and for high frequencies less than 5 milliseconds. In all the cases the release times will be much more superior to the attack ones, between 10 and 20 times superior. If the signal to compress or to limit corresponds to all the bandwidth, keep in mind that almost all the signal energy is situated in the low frequencies: you will have to provide attack times around 10 milliseconds.

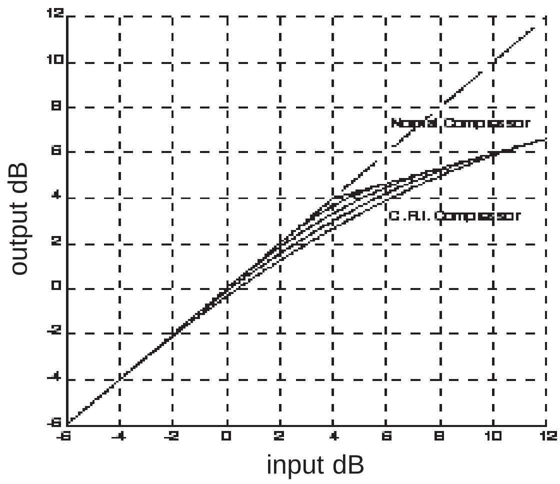
With traditional compressor and limiters, the compression or limit input is very punctual (it happens when the signal passes the threshold): this can be clearly noted when they are performing affecting too much the natural dynamic of music.

To avoid this problem, above all in multi-way P.A. applications, VMB Española S.A. has developed in its first digital processor C2-DAC, Continuous Ratio Increment **C.R.I.** dynamic curves in which the threshold changes from a define angle to a continuous curve that links perfectly the linear area of compressors or limiters with a 1:1ratio with the compression or limit area in a configurable dBs range. In these situations, the Knee of compressors or limiters changes continuously within the dBs range required. This progressive compression and limit effect applied to multi-way systems allows to get a dynamic control much more transparent since the headroom or the available signal margin does not stops with the entry of limiters; the typical stifling effect of traditional limiters disappears.

C.R.I. curves used in the MultiDAC are the same as the ones shown on the following graphs:



C.R.I. Limiter



C.R.I. Compressor

The **C.R.I. Limiter** graph shows a normal limiter at +4dB facing C.R.I. Limiters at +4dB with a Knee of +/-2, 4, 6 and 8 dBs. You can see that the dynamic control is more progressive when there is still **headroom** once the limiter is acting. In the Compressor C.R.I. situation, the threshold is set in +4 dB and the ratio is 1:3; an increment of 3 dB in the input will provide 1 dB in the output. You will also observe the same effect of gradual control, the compressor input being much less noticeable since its starting attenuation is much smaller.

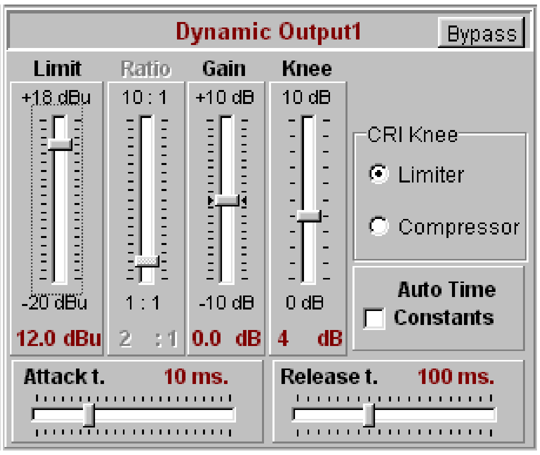
The **greatest advantage of using C.R.I. Limiters** is that it has an inherent compressor working as a **limiter depending on the input signal level**; this avoids the need to include a previous compressor to reduce the dynamic. C.R.I. Limiters behave like a compressor updating the dB range required from a 1:1 ratio to 1:∞ with the Knee. The result is a more progressive and transparent control that maintains **headroom** even when it is already acting. It is exactly the same with the compressor C.R.I. in which the ratio changes from 1:1 to 1:n, n being the new selected ratio.

Distorsion in Compressors and Limiters

A negative effect resulting from the use of compressors and limiters is that they introduce some distortion in the signal since they control the gain in a dynamic way. The smallest attack and release times, the greatest distorsion.

As far as this matter is concerned, **MultiDAC** presents a great advance since the processing algorithms of the DSPs signal developed by VMB Española S.A. allows to have at your disposal a dynamic control (for both the compressor and/or the limiter) with **0% distortion**, **completely free of distortions even with great levels of attenuation.** This unique feature, together with dynamic curves C.R.I. provide the technician with the most transparent and available gradual dynamic control.

Compressor / Limiter Parameters



In the **CRI Knee** Panel, you will decide if you want to use a C**ompressor** or a **Limiter**. Remember that a Limiter C.R.I. includes a continuous compressor that finishes in a limiter. If you select Limiter, **Limit** will appear on the left potentiometer showing the limit value and the **Ratio** potentiometer will remain deactivated: when acting as a limiter, the ratio is 1:inf. If you select **Compressor**, Limit will change to **Threshold** and the ratio will be activated. By acting on potentiometers you will modify the **Limit** values (from left to right): Limit signal level or **Threshold, Ratio**: the ratio of the compressor; **Gain**: gain introduced by the compressor or limiter; **Knee**: C.R.I. control indicating the dB margin in which the knee is acting. In every case the current value is shown in red below each potentiometer.

Modifying the **Knee** will allow to change from a limiter or normal compressor with Knee=0dB to a C.R.I. one with 10 dB units of performing margin.

In case of using limiters, it is recommended to give gain with Gain and let it at 0dB otherwise the limit value selected in Limit will be affected by the Gain value, value that is then added to the whole dynamic curve.

Each Dynamic section can be omitted if you activate the Bypass button: the dynamic control will remain unchanged, using the compressor as well as the limiter. In case you deactivate it, the Bypass button will appear in red.

Once the levels and thresholds of the compressor or limiter have been modified, you will adjust their **time constants**. For the attack time you will modify the **Attack time** potentiometer and for the release time the **Release time** one. The set time value will appear in red and in milliseconds. Available times in milliseconds are as follows:

Attack Time: 0.1, 0.2, 0.5, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 12, 14, 16, 18, 20, 25, 30, 35, 40, 45, 50, 60, 70, 80, 90, 100, 120, 140, 160, 180, 200, 250, 300, 350, 400, 450, 500, 600, 700, 800, 900, 1000.

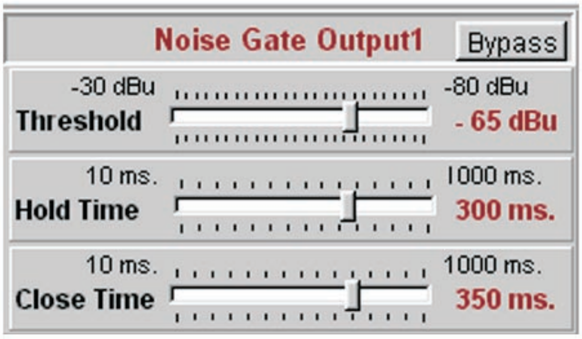
Release Time: 1, 2, 3, 4, 5, 10, 20, 30, 40, 50, 60, 70, 80, 90, 100, 125, 150, 175, 200, 250, 300, 350, 400, 450, 500, 600, 700, 800, 900, 1000, 1250, 1500, 1750, 2000, 2500, 3000, 4000, 5000, 10000.

There is a possibility to use automatic time constants when selecting **Auto Time Constants**: Attack and Release time controls remain deactivated. In this case the program selects automatically Attack and release time constants **according to the crossover inferior frequency cutoff** of the way. The following chart shows the constants selected by the program:

Inferior Frequency Cutoff	Attack Constants	Release Constants
Bypass	12 ms.	200 ms.
20-62 Hz.	16 ms.	200 ms.
63-124 Hz.	12 ms.	150 ms.
125-249 Hz.	7 ms.	100 ms.
250-499 Hz.	5 ms.	80 ms.
500-999 Hz.	4 ms.	60 ms.
1000-1999 Hz.	3 ms.	40 ms.
2000-20000 Hz.	2 ms.	30 ms.

Noise Gate

Further to the Compression or Limiter, MultiDAC incorporates an independent **Noise Gate** per output. You also have at your disposal the option to cancel it by pressing the Bypass button that will appear in red if the noise Gate is deactivated. The parameters you can modify are the following, the selected value appearing in red on the right of the potentiometer:

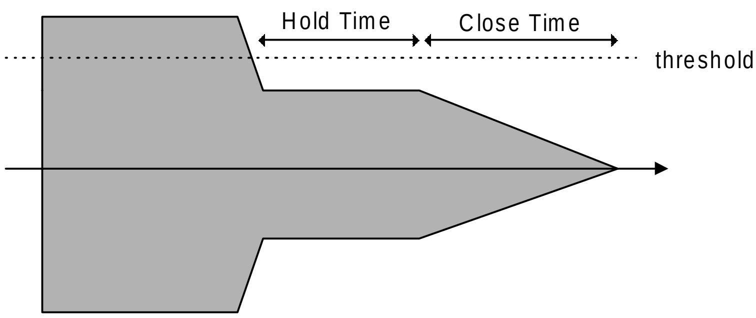


Threshold: threshold below which the noise gate gets activated. It can be configured within a range from –30dBu to –80dBu, normal values being situated between –60 and –70 dBu.

Hold Time: It indicates the time during which the signal has to be below the Threshold level to consider activating the noise gate. Its function is to avoid the noise gate to activate whenever the signal passes below the threshold level. You can vary between 10 and 1000 milliseconds. Times inferior to 250 ms are not recommended.

Close Time: It is the time the noise gate needs to act completely. The noise gate closes progressively during this time after having been below the threshold Hold Time. In case the signal exceeds the threshold, the noise gate opens instantaneously. Values are included between 10 and 1000 milliseconds. For the closing to be smooth, do not lower the Close time below 300 milliseconds.

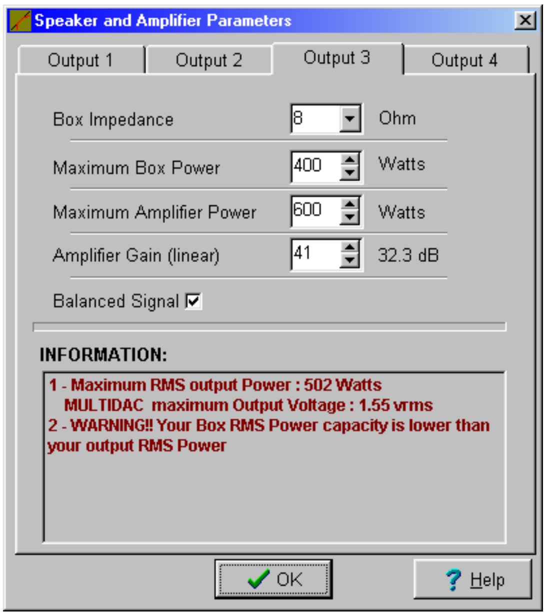
The next figure shows the audio signal zone together with the threshold and Hold and Close times. As you can see, the gate closing is progressive within the time you configure; the total time for the gate to get closed once the signal falls below the threshold is Hold Time + Close Time.



Noise Gate

Speaker and Amplifier Parameters

In the Dynamic Configuration – Noise Gate window you will find a button labelled **Speaker and Amplifier Parameters** through which you will access to the following window:



This window is a guide to adjust limiters according to amplifiers and speakers used in each output of the processor. The program will always indicate in the first line of the **INFORMATION** frame, the maximum RMS output power that will be provided to the output according to the maximum output signal level of the processor, the amplifier Gain, the cabinet impedance and the kind of signal (balanced or not). This data will be indicated like this: **Maximum RMS Output Power : X Watts**. This **INFORMATION** frame is the same as in the Dynamic screen if you activate the option **Show Info**. In this window you will be able to configure the following data:

- Box Impedance : Box or speaker impedance connected in this output. You have at your disposal 2, 2.67, 4, 8 and 16 Ohms.

- Maximum Box Power : Maximum RMS Power allowed by the box or speaker connected in the output. You will configure from 25 to 3200 watts by steps of 25 till 1000 watts and steps of 100 from 1000 watts. According to this data, the program will inform you in the **INFORMATION** window that the amplifier output power is superior to the admitted one. The following message will then appear: ***WARNING!! Your box RMS Power capacity is lower than your output RMS Power.***

- Maximum Amplifier Power: This is the maximum amplifier output power PER CHANNEL. You have at your disposal from 100 to 3200 watts by steps of 25 till 1000 watts and steps of 100 to 3200 watts. In case the maximum output level of the processor multiplied per the amplifier gain and according to the connected box impedance, you will need more power in the output. The following warning message will appear in the **INFORMATION** window:
WARNING!! Your Power Amplifier RMS is lower than your total RMS Power needs.

- Amplifier Gain (linear): If this data is shown in GdB you will need to change it to Linear G using the formula: G=10^(GdB/20). Variation range goes from 10 to 70. This data is necessary to know the RMS power that you want to provide the box with.

- Balanced Signal : You will activate this option every time the signal used is balanced. Make sure it is because with the balanced signal the output power is four times greater than with the signal unbalanced.

It can also be useful to know the output signal level in RMS volts that are getting out of the processor, above all if the available data of the amplifier are those of input Sensitivity. There are some amplifiers that do not have anti-clip internal circuitry and it is not convenient to overload them because in the one hand, it increases the output RMS power and in the other hand the final sound quality gets worse as the distortion increases. This information will always appear in the second line to the **INFORMATION frame: MultiDAC maximum Output Voltage: X Vrms.** This data always concerns a balanced output. In case of using unbalanced signal you will have half.

The formulas used by the program to obtain the RMS output power according to grabbed data are as follows:

Balanced signal :

$$\text{RMS Output Power} = \frac{(\text{Vout} \times \text{Gain})^2}{\text{Impedance}} \quad \textbf{watts}$$

Unbalanced signal :

$$\text{RMS Output Power} = \frac{(0.5 \times \text{Vout} \times \text{Gain})^2}{\text{Impedance}} \quad \textbf{watts}$$

Vout is the maximum balanced output of the processor; **Gain** is the linear amplifier gain and **Impedance** is the box or speaker impedance connected to the amplifier output.

WARNING:

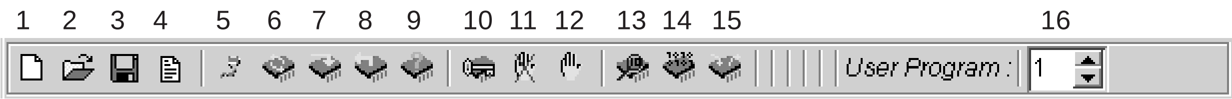
VMB ESPAÑOLA, S.A. IS NOT RESPONSIBLE FOR THE VERACITY AND ACCURACY OF THE INFORMATION PROVIDED BY MANUFACTURERS REGARDING ADMITTED POWER AND LOUDSPEAKER IMPEDANCE AND GAIN, SENSITIVITY AND POWER OF POWER AMPLIFIERS. THIS IS THE REASON WHY THE COMPANY WILL NOT ASSUME ANY RESPONSIBILITY IN THE ADJUSTEMENT OF LIMITERS THAT MAY DAMAGE EQUIPMENTS, EXCEPT THE ONES CONFIGURED BY VMB ESPAÑOLA S.A. IN ITS OWN SYSTEMS. THIS MANUAL DEALING WITH LIMITER ADJUSTMENT IS JUST A HELP BUT DOES NOT GUARANTEE THAT RESULTS ARE THE REQUIRED ONES SINCE EVERYTHING DEPENDS ON DATA VERACITY. SHOULD YOU HAVE ANY DOUBT ASK DIRECTLY TO THE MANUFACTURER OF SPEAKERS, BOXES, AMPLIFIERS AND ALWAYS ADJUST LIMITERS BELOW THEIR MAXIMUM ADMITTED LEVEL.

4.- MultiDAC configuration

4.1.- Tools Bar

Once all the parameters of the processor are configured for a system, it will be necessary to store the hardware configuration in the **MultiDAC** so that it remains ready to be used exactly as you programmed it.

You can access to all the options from the **Menu** bar as well as the buttons of the toolbar. Theses buttons are the ones described below. If you place the mouse on any button during few seconds, a small description of it will appear in a Hint.



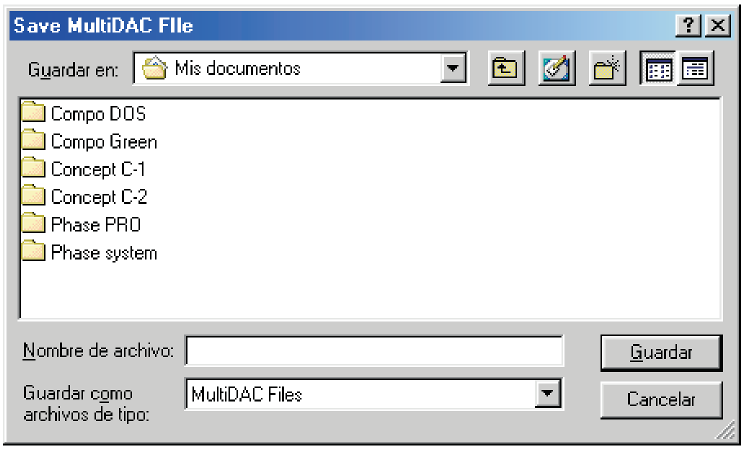
1.- ***New File*** : Button used to start all the parameters in order to create a new configuration.
2.- ***Open File*** : Open a configuration stored in the PC.
3.- ***Save File*** : Save the current configuration in the PC.
4.- ***Generate Report*** : Generates an automatic report of the actual configuration.
5.- ***Connect MultiDAC*** : Connect to the MultiDAC processor.
6.- ***Reset MultiDAC*** : Reset the processor. Reboot the starting process.
7.- ***Update System*** : Update the DSP Operating System.
8.- ***Get System*** : Obtain the Operating System from the MultiDAC.
9.- ***Get Version*** : Obtain the DPS software version.
10.- ***Set Password*** : Configure the password to protect the keyboard.
11.- ***LOCK Keyb*** : Lock the keyboard of the processor. Keyboard unavailable.
12.- ***UNLOCK Keyb*** : Unlock the keyboard. Keyboard available.
13.- ***Read MultiDAC Programs*** : Obtain the configuration of the MultiDAC.
14.- ***Store Configuration*** : Store configuration in the MultiDAC.
15.- ***Change Config*** : Change the MultiDAC configuration.
16.- ***User Memory*** : Memory number in which you store the configuration.

Some buttons may appear inactivated. Depending on whether you work in Offline mode or connected to **MultiDAC**, all the buttons in relation with communications will be shown activated or not.

4.2.- Saving and Retrieving Configuration Files

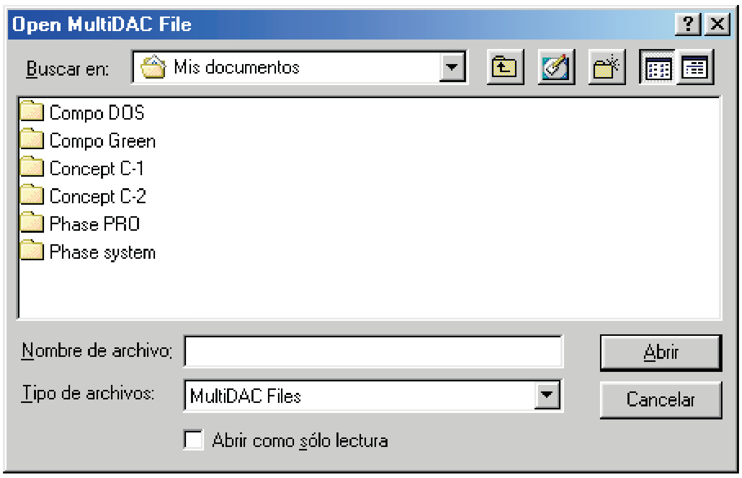
Saving Configurations

Whenever you want you can store the current configuration pressing the **Save File** button in the toolbar (button 3) or selecting the option **Save** or **Save As** in the menu File. **MultiDAC** files have the following filename extension: ***.dac** that has to be respected for if you want to retrieve configurations later on.. It is highly recommended to save any advance in configuration every few minutes in order to avoid error or problems inherent in Windows. It is also recommended to save changes in configurations with significant names so as to compare later on. It is very useful to fill in the Comments field in the System window with information relative to the system or so that when retrieving it subsequently you will know what it is about. It is also recommended to organise correctly configuration files ***.dac** in directories. When saving a configuration, the typical window of Save File will appear.

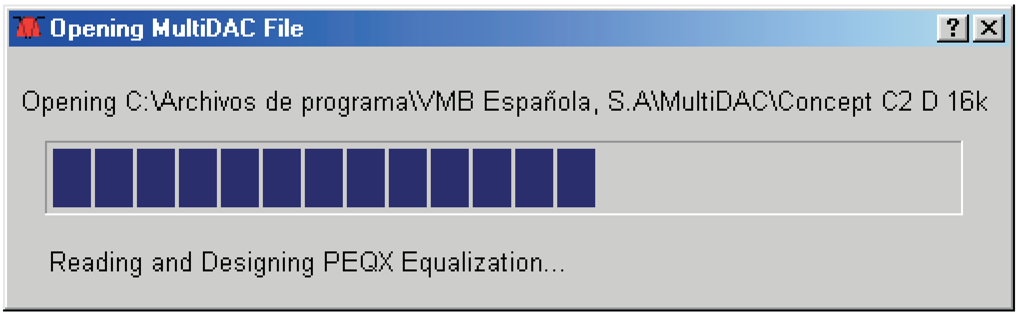


Retrieving Configurations

Saved configurations can be retrieve later on through the **Open File** button or selecting the option **Open** in the **File** menu. The program will show the **Open** File window for you to indicate the file **.dac** you want to open. If necessary, change the directory until you get to the required configuration.



Once you have selected the file ***.dac** you want to open, a new window will appear with a progressive bar and a legend indicating that part of the file is being opened and designed at the same time. In case there is an error, it will stop and indicate the cause of the error.



If you want to initiate a new configuration from the very beginning there is one already activated, you will have to press button number 1 **New File**, option that is also available in the **File** menu, option **New**. The program will always ask you before loosing the current configuration whether you want to save it in the hardware or not, through the following window:

If pressing **Yes** you will access the **Save** window to enter the name **.dac** of the file you want to store. If you chose **No**, you will start a new configuration without storing the existing one and if you select the **Cancel** button nothing will be done.

4.3.- MultiDAC Configuration

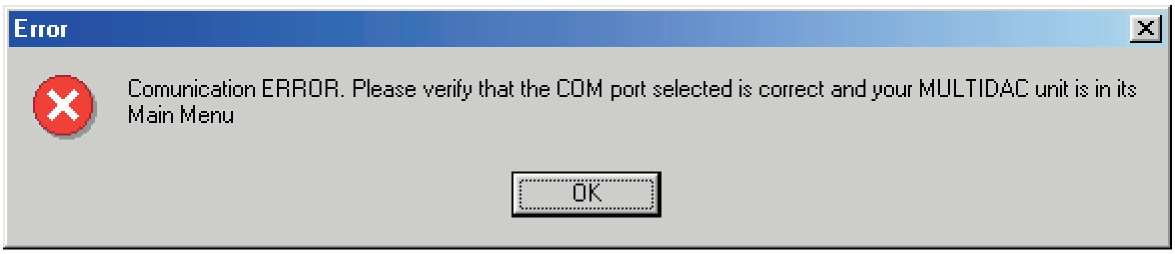
To configure the **MultiDAC** processor with all the programmed parameters, it must be connected to the computer. If you have started the program in **Offline** mode, you will be able to connect subsequently pressing the button **Connect MultiDAC** in the toolbar. The initial window of the connecting program will then appear again.

Select the serial port Com1 or Com2 that is available and connect the provided serial cable as it is indicated in the paragraph 2.2 (page 9). If the communication is correctly established, it will appear briefly on the Communication screen before making way for the main screen of the program with the **MultiDAC** processing sheme.

If the serial port selected for the Com1 r Com2 connection is already used for another device such as the mouse or modem, the program will inform you with the following window:

You should then change to another serial port. If there is anyone left, you will have to deactivate or remove the device that is using it.

If the serial port is available but the cable has not been correctly connected, the program will try unsuccessfully to established the communication with the processor and will show the following window indicating that the cable is not correctly connected or that the processor is not in the main menu. Take in mind that the **<u>MultiDAC processor must be in the main menu in order to be able to establish the communication</u>.**

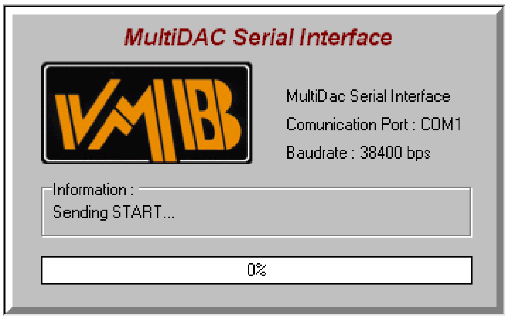


Once the connection with the processor is established, all the buttons of the toolbar that are linked with communications will be activated and you will be able to program your configurations in the processor.

Once the required configuration is achieved, select one of the 19 memories so as to store it. For this, change the value of the **User Program** check box until you reach the one you want.

Press the button **Store Configuration** to program the selected memory. The communication is established with a safe protocol designed by **VMB Española S.A**. with a speed of **38400 bauds.**

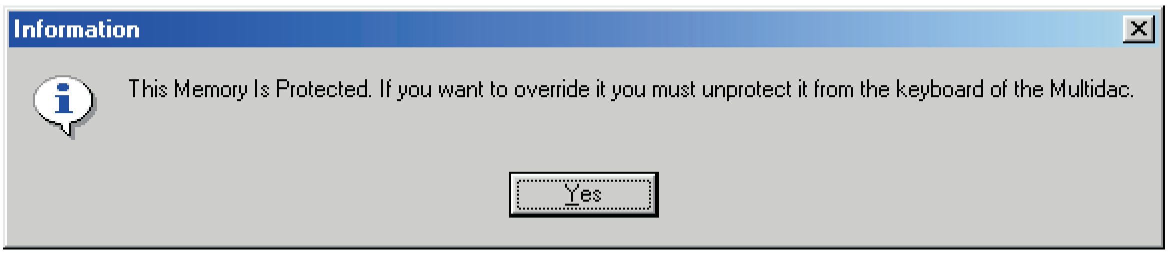
While the communication is being established, the **MultiDAC Serial Interface** window appears. It indicates in the **Information** frame the situation of the communication (starting, checking, communications and finishing) and in the progressing bar the percentage of the transmission.



Once the transmission is finished, the processor automatically changes to the new memory so that it can check acoustically the achieved configuration.

It may happen that the memory in which you want to write is already occupied. In this case, you will be asked if you want to replace it.

It is also possible that the memory you want to use is not only occupied but also **protected**. In this case, the program will warn you and if you still want to use it, you will have to deactivate the protection from the processor. For this, see the instruction manual of the **MultiDAC** processor.



It is also possible to **retrieve configurations** already stored in the processor. Now, the configuration data are sent to the computer by the processor. First you will select the memory you want to read with **User Program** and then press the button **Read MultiDAC Programs**. Once the data have been transmitted, the program will design all the filters and parameters: they will all appear on the corresponding screen.

Thanks to the button Change Config, you will be able to change remotely the activated memory of the processor changing for the activated number in User Program.

4.4.- Updating the DSP Operative System.

One of the main advantages of **MultiDAC** is the use of Flash data memory that can be electrically programmed. The DSP Operating System as well as all the configuration memories are stored in this kind of memory: **MultiDAC** is a product that can be updated by its user. There is then no need of sending the product back to the provider or set new EPROM which would mean opening the device, remove the former EPROM and place correctly the new one…

This way, any new version of the product with the latest improvements or new options will be available. No matter when the user got the processor, he/she will always have the latest version.

There are two ways of checking which version is set in the processor: first, from the processor in the second menu of information, DSP Version, and second from the software MultiDAC by pressing the button with the processor activated. Updated versions will always have an increased number.

WARNING :

IF THE DSP OPERATING SYSTEM UPDATING PROCESS DOES NOT FINISH CORRECTLY THE PROCESSOR CAN REMAIN USELESS. ALWAYS CHECK THAT THE CABLES AND CONNECTIONS ARE CORRECTLY PLUGGED.

DO NOT SWITCH OFF THE POWER WHILE THE PROCESS IS ON.

The program will inform you that it is a tricky operation and give you the possibility to go back or go on with the updating through a warning window.

In case you go on with the updating process, a window will appear for you to enter the new version of the DSP Operating System, files with name and extension **mdac***.asc**, *** being the reference of the version. Once the file ***.asc** with the new version has been selected, the window **MultiDAC Serial Interface** will appear to inform you about the updating progress. Once the process is finished, the processor **MultiDAC** will reboot automatically with the new version.

If there is any problem or disconnection during the updating process, the processor won´t probably start correctly the next time you initialise it. If that happens switch it off and the switch it on by pressing at the same time the three keys: Up and Down arrows and Enter. This way, the processor will start but the DSP are not initialised and no memory is loaded. The processor can communicate with the computer from which you will try to enter the new DSP Operating System.

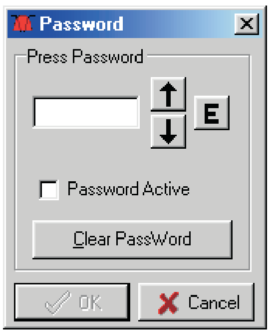
It is also possible to get the Operative System from any processor and save it in the hard disk drive with a format **.asc**. For this, connect the processor to the computer and press the button **Get System**: the version and the program will be transferred from the processor to the computer.

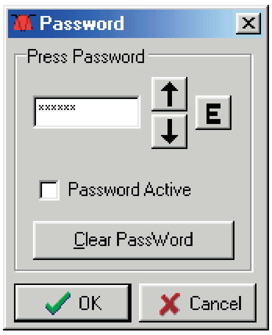
4.5.- Security.

It is often convenient to block the access to the controls of the processor in order to unauthorised people from modifying parameters like in fixed installations, or when renting a sound equipment to prevent people from modifying configurations.

There are two levels of protection of the processors **MultiDAC**: a partial one through a password and a complete one that blocks the keyboard.

To use a password to access any menu of the MultiDAC, you need to enter it previously. For this, press the button and the window **Password** will appear.





The password has to include 5 characters entered with the keys of the processors. For this, you will press the appearing keys, an asterisk appears for each key pressed. Once done, you can activate the password selecting the box **Password Active** and **Ok**. If you want to start again with the entering the password, use the button Clear Password. Once activated, you will be asked for the password any time you want to access the menus. If you do not enter it you will not be able to modify or change anything. If you want to deactivate it from the processor, you will have to go to the menu 1 **Configuration** in the option **Password** and change it for OFF.

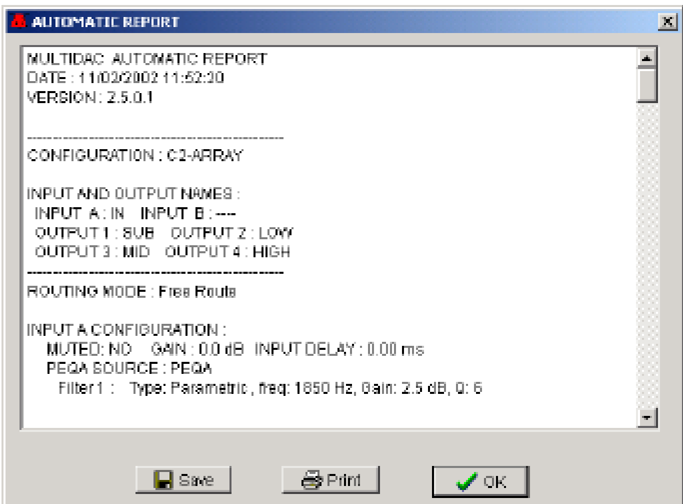
Another way to lock the processor is blocking the keyboard. When activating this option, the processor will show a message of Keyboard Locked before any attempt of accessing the menus. In this case it will be necessary to connect the **MultiDAC** to the computer to unlock it. This is a good solution for if you do not want anybody to change parameters or configurations. When press the button LOCK Keyb the following warning window will appear:

By pressing the OK button you will lock the keyboard and with Cancel you won't do anything. When the keyboard is locked and you want to unlock it, press the button UNLOCK Keyb. Now the message is:

You will unlock the keyboard by pressing OK. By pressing Cancel the keyboard will remain locked.

4.6.- Automatic Configuration Report

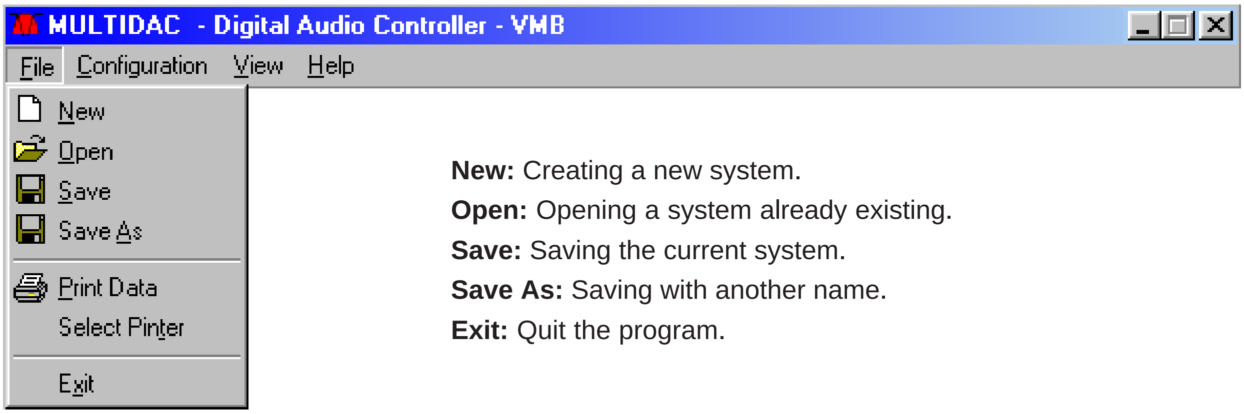
With the button is possible to generate an automatic detailed report of the actual configuration. All parameters of the configuration will be written to save to an ASCII text file or to print.



5.- ACCESS FROM THE MENUS

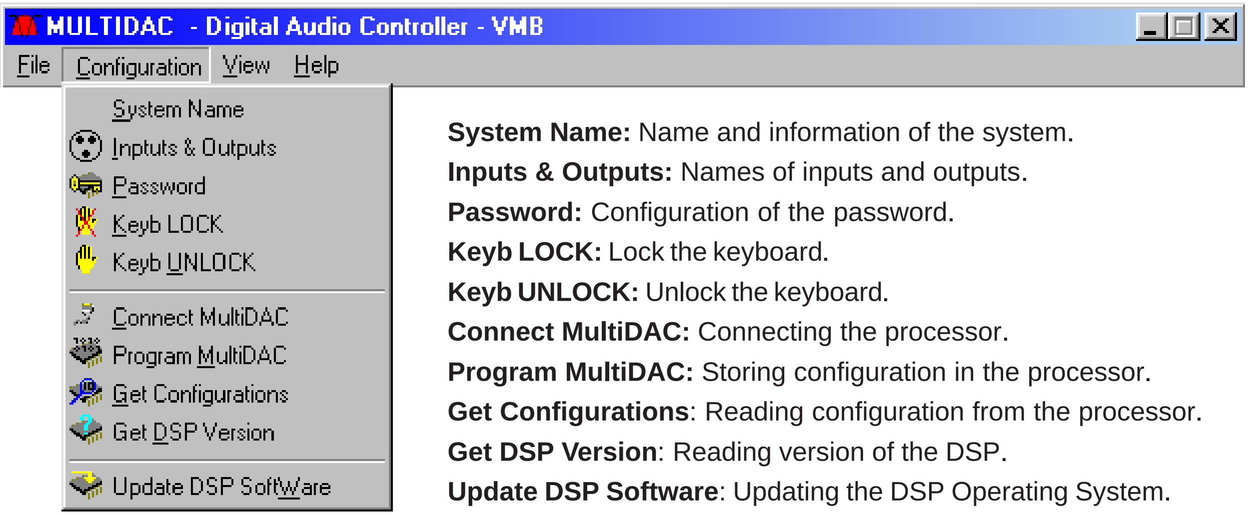
In the main window, you have a series of menus that allow you to access directly to every function as if you were in the Tools bar. These menus are:

5.1.- File Menu:



New: Creating a new system.
Open: Opening a system already existing.
Save: Saving the current system.
Save As: Saving with another name.
Exit: Quit the program.

5.2.- Configuration Menu:



System Name: Name and information of the system.
Inputs & Outputs: Names of inputs and outputs.
Password: Configuration of the password.
Keyb LOCK: Lock the keyboard.
Keyb UNLOCK: Unlock the keyboard.
Connect MultiDAC: Connecting the processor.
Program MultiDAC: Storing configuration in the processor.
Get Configurations: Reading configuration from the processor.
Get DSP Version: Reading version of the DSP.
Update DSP Software: Updating the DSP Operating System.

5.3.- View Menu: Direct access to program configuration windows.

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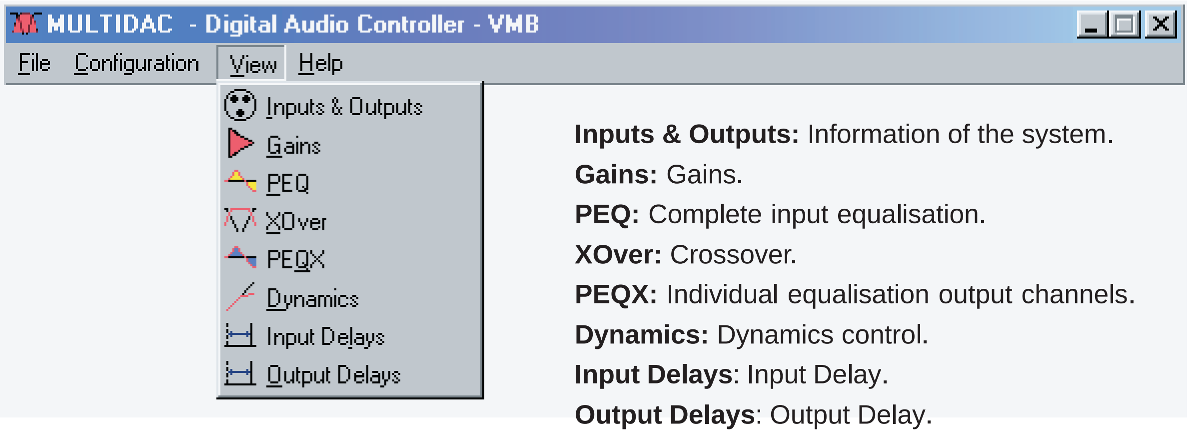
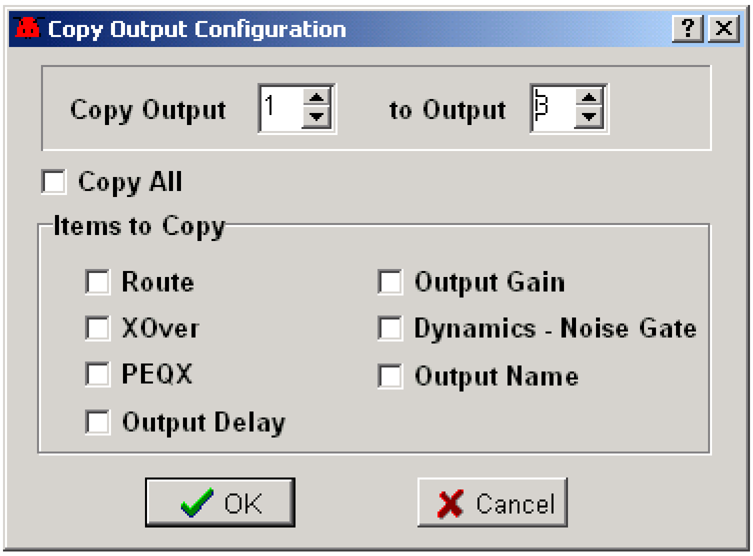


Inputs & Outputs: Information of the system.
Gains: Gains.
PEQ: Complete input equalisation.
XOver: Crossover.
PEQX: Individual equalisation output channels.
Dynamics: Dynamics control.
Input Delays: Input Delay.
Output Delays: Output Delay.

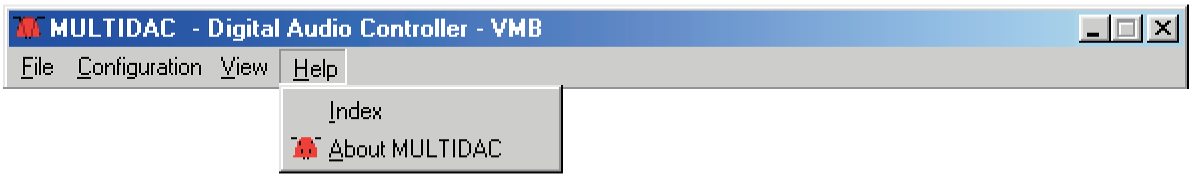
This menu gives access to the same configuration windows than the equivalent buttons of the main menu.

5.4.- Copy Outputs Menu- Copy parameters between outputs.

At the menu **Copy Outputs** the showed window will appear. From this window the user could copy some or all parameters from one output to another. At **Copy Output** the original output will be selected, and at **to Output,** the destiny output to receive the new parameters. If **Copy All** is enabled then all parameters will be copy. At **Items to Copy** is possible to select individually wich parameters will be copied: **Route**, **XOver, PEQX**, **Output Delay**, **Output Gain** , **Dynamics - Noise Gate** y **Output Name**.

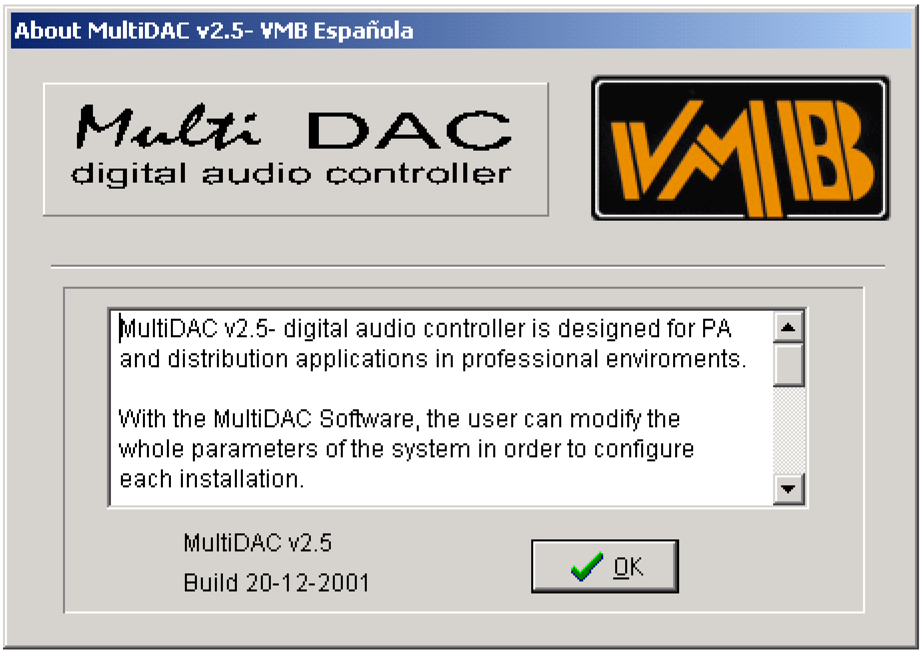


5.5.- Help Menu:



Index: Thematic index of help.

About MULTIDAC: Information about the available version of MULTIDAC..



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