

# DX1.5 Loudspeaker Management System



The Martin Experience

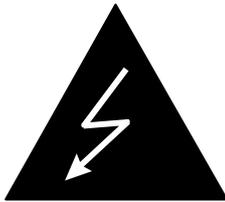
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# ! READ THIS FIRST !

## 1) Important Safety Information

-  This equipment must be earthed.
-  Do not expose to rain, moisture or excessive dust
-  No user serviceable parts inside, refer servicing to qualified service personnel.
-  Do not remove covers



-  Objects containing liquids, such as vases, must not be placed on this equipment.
-  Do not remove any protective earth or signal cable shield connections.
-  Only use this equipment with an appropriate mains cord.

In the USA the cord should comply with the requirements contained in the Standard for Cord Sets and Power Supply Cords, UL 817, be marked VW-1, and have a current carrying capacity not less than the marked rating of the apparatus.

## 2) Electrical compliance



An example of this equipment has been tested and found to comply with the following European and international Standards for Electromagnetic Compatibility and Electrical Safety:

Radiated Emissions (EU):	EN55013-1 (1996)
RF Immunity (EU):	EN55103-2 (1996)
	RF Immunity, ESD, Burst Transient, Surge, Dips & Dwells
Electrical Safety (EU):	EN60065 (1993)
Country of origin:	United Kingdom

## 3) Unpacking the unit

After unpacking the unit, please check it carefully for any damage. If any is found, immediately notify the carrier concerned - you, the consignee, must instigate any claim. Please retain all packaging in case you need to return the unit.



Please think of our environment.

When the product has reached the end of its useful life, please dispose of it responsibly through a recycling centre.

## 4) Introduction

The DX1.5 is a powerful DSP based audio processor, ideally suited for install applications, where they combine the functions of a multitude of conventional products in a compact 1U unit. To achieve this, the unit's have inputs and outputs which can be configured in a selection of basic crossover modes. The DX1.5 also offers a "free assign" mode, which allows completely flexible routing of any output from any combination of the inputs.

Each input has a gain control, variable delay and a further eight bands of fully parametric equalisation. The parametric filter bands have a large selection of different filter types available, including shelving.

Each output has a gain control, variable delay, high and low pass crossover filters, nine bands of fully parametric equalisation, polarity switching and, additionally, a fully featured limiter, and a final clip limiter. The crossover filters offer slopes of up to 48dB/Octave., with a variety of responses available.

Security lock-out is available for all controls.

The DX1.5 is also equipped with AES/EBU digital inputs, and includes a sample rate converter, capable of accepting anything from 32kHz up to 192kHz.

## 5) Features

Superb audio quality – carefully optimised double precision signal processing coupled with 24 bit conversion ensure a dynamic range in excess of 117dB. The high sampling rate of 96kHz means minimal filtering providing exceptional sonic purity with a bandwidth in excess of 32kHz.

A flexible input/output multi-mode format caters for any configuration, regardless of scale.

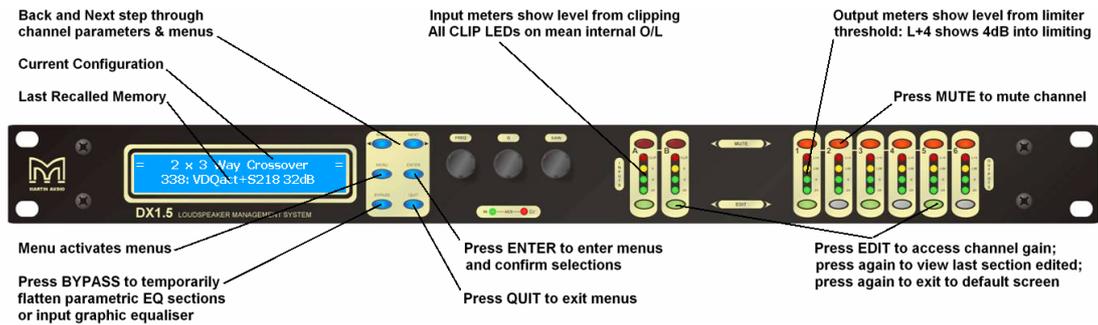
Both routing of inputs to outputs, and ganging (for editing) are completely flexible.

A completely new SHARC™-based DSP platform supplies phenomenal computational power, allowing the unit to provide not only multiple bands of standard parametric equalisation on every input and output, but an additional full spectrum graphic equaliser on each of the four inputs. This additional power also permits both program limiters and no overshoot clip limiters on each output.

Delay of up to 650mS may be independently set for each output, with an exceptionally fine minimum increment of 300nS, which corresponds to a distance change of 0.1mm!

The comprehensive standard specification also includes up to 255 memories.

## 6) DX1.5 Quick Reference



### Editing channels

Press channel's **EDIT** key. First press accesses that channel's gain. To scroll through channel's parameters, use the **BACK** and **NEXT** keys. The second press accesses last viewed parameter. Third press will drop back to the default screen.

### Accessing menus

Press the **MENU** key. Use the **BACK** and **NEXT** keys to select the sub-menu required, and enter the sub-menu using the **ENTER** key. This applies to all levels of menu. **ENTER** always confirms selections.

### The displayed menus and their contents

#### GLOBAL MEMORY Sub-menu

Recall/Store/Erased input, crossover or combination settings.

#### INPUT SECTION Sub-menu

Set up input ganging,

#### CROSSOVER Sub-menu

Set up or adjust crossover design, including routing and auto limiter setting. Also set up output ganging.

#### INTERFACE Sub-menu

Communications interface setup.

#### SYSTEM Sub-menu

Used to view unit's status, and select various global options such as PEQ 'Q' or bandwidth units, delay units, and output metering point (pre/post mute).

#### SECURITY Sub-menu

Used for locking various operations of the unit, using a 4 digit code.

### Notes

Crossover & input settings may be stored independently via the **Global Memory sub-menu**.

The output meters show level, in dB from the limiter threshold, and the input meters show level from clipping the A-D converters, pre-gain and all EQ.

The high and low pass crossover filters are defined independently on each output channel.

To access the limiter attack and release parameters, select **AutoLimiter TimeCst: No** when designing a crossover.

To swap parametric filter units between bandwidth ('BandW') and 'Q', enter **System Sub-menu**, select **Filter Q / Bandwidth**, and select required readout units.

To swap delay time units, enter **System Sub-menu**, select **Delay Time / Distance**, and select required readout units.

If you press an **EDIT** key the corresponding channels routed to / from that channel will flash.

## 7) Loading factory presets into your Martin Audio 2in/6out DX1.5 Loudspeaker Management System



A zip folder containing loader software and current DX1.5 binary files is available from the accompanying User Guides CD and from [www.martin-audio.com](http://www.martin-audio.com). Extract the contents to your hard disk. Do not run the programme from the zip folder.

### Warnings

Loading presets will overwrite previously loaded factory presets - and any changes to them (e.g. channel gain, delay, primary limiter and ClipLim settings).

To make a permanent record of changes to the user-editable parts of a preset, you must save the preset in one of the lower number memory locations.

Make sure you have a full set of .bin files before proceeding so that you can reload your original presets if you need to.

If you are not familiar with the Martin Audio DX1.5, please read the DX1.5 User Guide.

You should have extracted the following files to a suitable folder on your hard drive:

**MartinFlashLoader.exe** – The binary preset loader software

### General presets

**MARTN15RLib5.bin** – The general family of binary presets for most regular Martin Audio loudspeakers and stage monitors. Line array systems have separate preset families - see below...

### Line array presets

Note that we recommend the 4in/8out remote controllable DX2 for more complex systems (e.g. band-zoned Longbow + W8LD + W8LS).

**MARTN15RLB02.bin** – Binary presets for W8L Longbow

**MARTN15RLCDLC3.bin** – Binary presets for W8LC + W8LCD

**MARTN15RLM21802.bin** – Binary preset for W8LM/LMD + WS218X/W8LS

**MARTN15RLMLX01.bin** – Binary preset for W8LM/LMD + WLX

**MARTN15RLMMX03.bin** – Binary presets for W8LM/LMD+WMX

(Note that preset libraries may have been updated since this user guide was published. Updated presets may show slight title variations)



You will need a 9 pin male-to-female, fully wired, pin-to-pin RS232 cable

**Note: this lead should be pin-to-pin – not the null modem type.**

## To load the latest presets into your DX1.5

1. Connect the 9-pin male to 9-pin female DEE type serial lead from your computer to the external socket on the rear of the unit.



2. On the DX1.5 use the blue *Menu*, *Back*, *Next* and *Enter* keys to go into the Interface Sub-menu and then into **External Interface**.

3. Once entered, setup as follows:

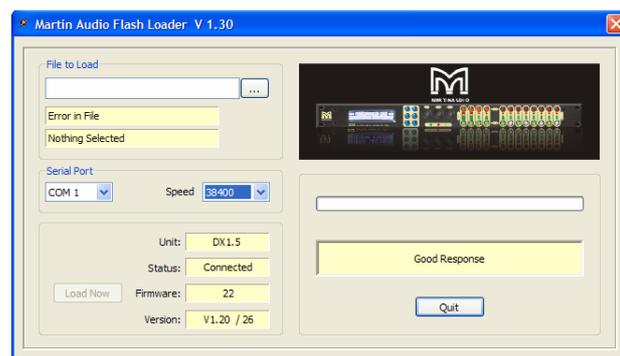
**Serial Speed:** 38400

Press *Enter* to confirm

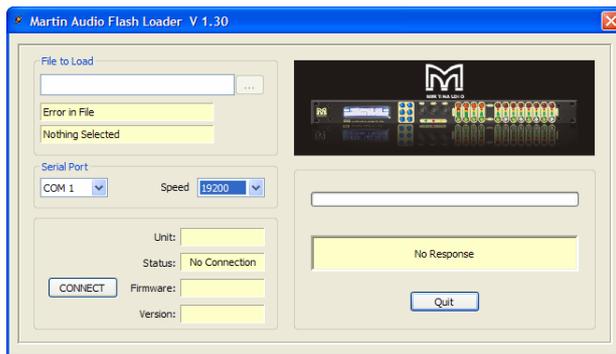
4. Run the programme **MartinFlashLoader.exe**.

5. Set the Serial Port **Speed** to 38400.

The bottom right hand window should indicate **Good Response**.



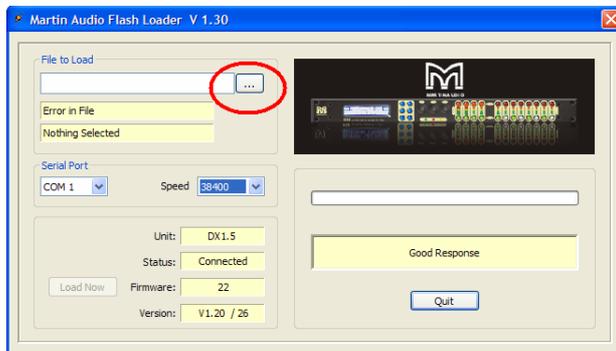
If the **Serial Speed** number is not set to 38400 to match the DX1.5 speed, the software lower right window may indicate **No Response**.



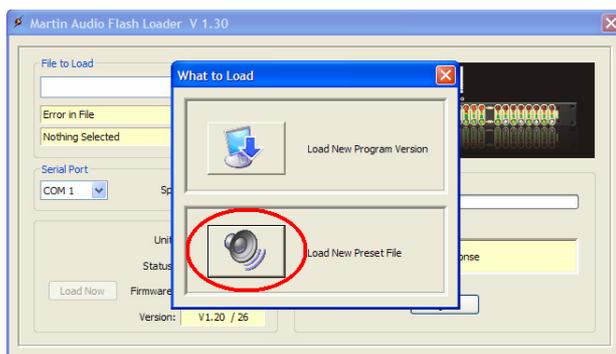
Make sure that the Serial port **Speed** on your pc screen is set to 38400 to match the DX1.5.

If that fails to fix the problem, try adjusting the **Com (Serial Port) number** and clicking the **Connect** button again. It is usually quicker to do this by trial and error as Com port numbering can be a mystery to the average PC user!

6. Click on the **File to Load** button (circled)

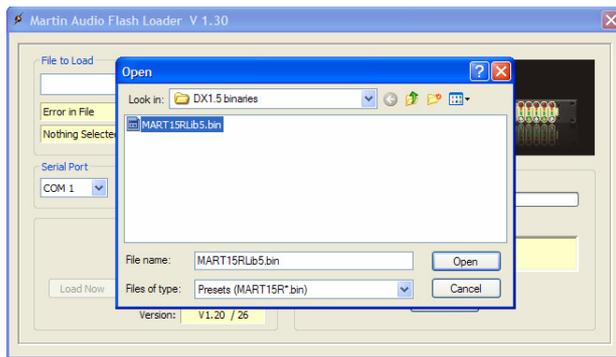


7. Click on the **Load New Preset File** button (circled)

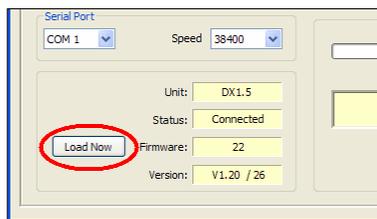


8. Browse to find the files you extracted earlier.

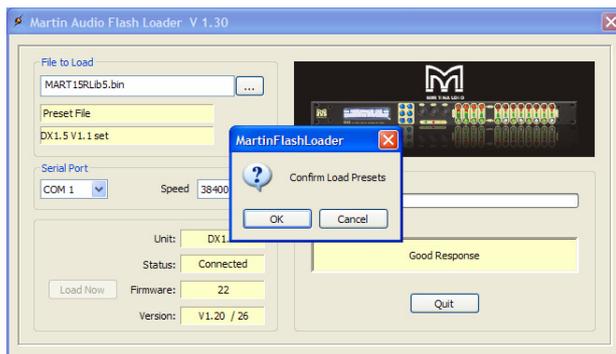
Click on the preset required (to highlight it) and click **Open**.



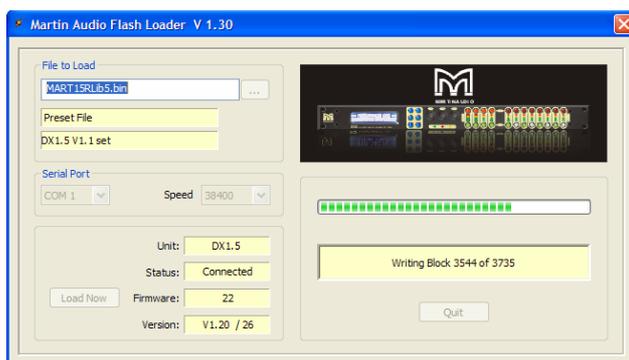
9. Click **Load Now** (circled)



10. The programme will communicate with the DX1.5 and ask for confirmation.



Click **OK**.



You'll see the green block writing progress bar proceeding from left to right

and

on the DX1.5 front panel you'll see graphics shuffling from right to left...followed by an indicator test (all LEDs on)...and a short delay...and then a wake-up procedure.

## 8) Selecting a specific preset

Each binary file will load a family of presets into your DX1.5.

**Please note that the DX1.5 will retain its current preset until instructed to change to one of the newly loaded ones.**

Change the DX1.5's settings to one of the newly loaded presets as follows:

Press the blue *Menu* button, then the *Back* and *Next* buttons to find the **Global Memory Sub Menu** and press *Enter*.

Use the *Back* and *Next* buttons to find the **Recall a Memory** and press *Enter*.

Use the *Back* and *Next* buttons to find the **Crossover only** and press *Enter*.

Browse for the appropriate preset using the *Back* and *Next* buttons or the **Frequency** knob.

Note that the DX1.5 will default to its previous condition if the above operations are not completed and no buttons are pressed for 20 – 30 seconds.

### Input parameters

Input parameters are not usually specified within standard Martin Audio presets – hence the **Crossover only** selection above.

When configuring a new system always ensure that there are no input gain, base delay or input eq settings left on the input sections by previous users.

Press the input **EDIT** buttons and scroll through the parameters for that input using the blue *Back* and *Next* buttons. See section 14) *Editing Audio Parameters – Input Channels* for operating details...

## 9) Amplifier power, amplifier gain, DX1.5 primary limiter and ClipLim settings

### Amplifier power

We suggest you specify an amplifier capable of meeting the loudspeaker's peak power rating – as long as you use the correct Martin Audio limiter recommendations.

### Primary limiters

The primary **Limiter** parameters protect the loudspeaker from rms heating effects whilst the correct amplifier power specification and **ClipLim** settings protect the loudspeaker from transient overloads and clipping.

Primary **Limiter** thresholds are factory preset for new products and recommended in the user guide CD controller spreadsheets for older products. Factory preset primary **Limiters** are set for power amplifiers with the pro-audio voltage gain\* standard of 32dB (x40). Spreadsheet recommendations for older products show a range of primary **Limiter** settings depending on your chosen amplifier gain/sensitivity.

\* We recommend using power amplifier voltage with known gain settings if at all possible. Manufacturers' *sensitivity* settings are not very useful as the resulting gain figures will vary from model to model depending on their power specifications causing spectral imbalance when using mixed amplifier set-ups.

If you have to use a power amplifier set to a different gain standard, make absolutely sure that you adjust the 32dB standard factory preset or spreadsheet recommendation as follows:

Power amplifier gain	Primary limiter threshold adjustment from 32dB preset standard
23dB	Add 9dB (e.g. +4dBu becomes +13dBu)
24dB	Add 8dB
25dB	Add 7dB
26dB (Crown standard)	Add 6dB (e.g. +4dBu becomes +10dBu)
27dB	Add 5dB
28dB	Add 4dB
29dB	Add 3dB
30dB	Add 2dB
31dB	Add 1dB (e.g. +4dBu becomes +5dBu)
32dB (Industry standard)	Martin Audio preset standard
33dB	Decrease by 1dB (e.g. +4dBu becomes +3dBu)
34dB	Decrease by 2dB
35dB	Decrease by 3dB
36dB	Decrease by 4dB (e.g. +4dBu becomes 0dBu)
37dB	Decrease by 5dB
38dB	Decrease by 6dB
39dB	Decrease by 7dB
40dB	Decrease by 8dB (e.g. +4dBu becomes -4dBu)

### ClipLims

The **ClipLim** settings protect the loudspeaker from transient overloads (and protects against amplifier clipping if the amplifier is capable of supplying the loudspeaker's rated peak power).

All Martin Audio loudspeakers are designed to professional AES power ratings that allow for 6dB signal peaks above the rated AES power. This means that Martin Audio loudspeakers have peak power handling capabilities that are 6dB above (4x) their AES power ratings – when the correct controller settings are used.

Note that the ClipLims default to **Medium 2dB Above** the primary limiter thresholds if not set up. To take advantage of your power amplifier's and loudspeaker's peak capability, set the **ClipLims** as follows:

Subwoofers	Slow 6dB Above
Low mid/LF	Medium 6dB Above
High mid/HF	Fast 6dB Above

If you have specified an amplifier smaller than the loudspeaker's peak power rating you may need to reduce the **ClipLim** settings by a dB or two to prevent amplifier clipping as it runs out of headroom.

You must, of course, set the recommended primary **Limiter** threshold before adjusting the ClipLim level to prevent amplifier clipping.

If you end up setting the **ClipLim** as low as **2dB Above** in an attempt to prevent significant amplifier clipping, and the amplifier still clips, you probably need a bigger amplifier. If budgets don't allow for a bigger amplifier you may need to reduce the primary **Limiter** threshold settings to prevent the amplifier from clipping at the 2dB Above ClipLim setting.

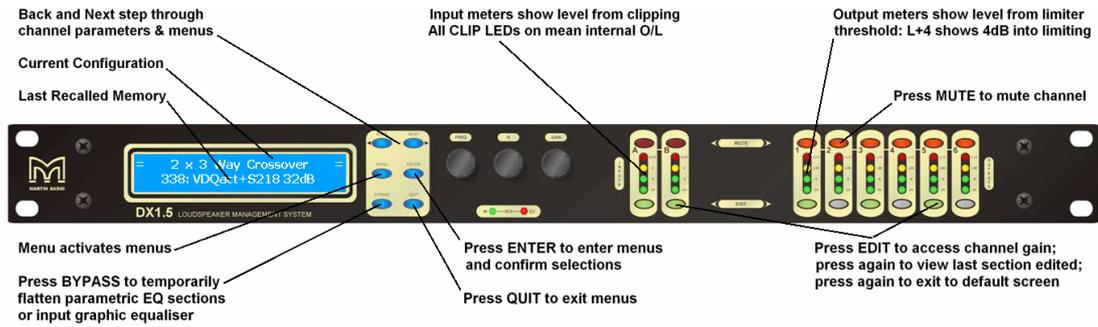
### System configurations

Your Martin Audio DX1.5 Loudspeaker Management System must be correctly patched.

There are a large number of Martin Audio factory presets and configurations and these are being expanded all the time. It is always sensible to check your DX1.5 output channel allocations for the particular preset you have chosen to use before patching your system. This will ensure that the appropriate band is being routed to each loudspeaker section.

You can do this by pressing each yellow output channel **EDIT** button and noting the output channel allocations (e.g. Low, MidHi) indicated on the DX1.5's display panel.

## 10) Front Panel functions (l-r)



### LCD Screen

By default, the screen shows the current routing configuration on the top line and the name of the last recalled memory on the bottom line. All other parameters and menus are shown as they are edited.

### Control Keys – Parameter selection and adjustment

**NEXT** key - moves forward through list of parameters.

**BACK** key - moves back through list of parameters.

**MENU** key activates the main menu – a second press selects the last menu edited – a third press selects the last menu item. In this way, three presses on **MENU** from the default screen will jump back to the last parameter adjusted.

Different menus are selected using the **BACK** and **NEXT** keys, or with the **FREQ** encoder.

**ENTER** key enters the chosen menu, confirms selections, and changes filter types when editing parametric sections.

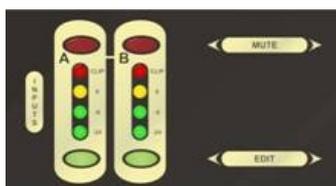
**BYPASS** will flatten the currently selected parametric sections. Note that, for safety reasons, it is not possible to bypass the high and low pass filter sections.

**QUIT** exits menus back to the default screen.

**FREQ, Q & GAIN** (velocity sensitive encoders) adjust the relevant parameters as displayed on the screen.



The two status LEDs indicate that **AES** inputs have been selected and **Error**.



**Inputs A & B** - meters show dB from clipping point of the analogue to digital converters.

**MUTE** buttons illuminate when pressed and mute audio for that input channel.

**EDIT** buttons illuminate yellow when pressed. The buttons access gain on first press, last viewed parameter on second press, then exit on third press.

**Outputs 1-6** - meters show dB below limit (Green), Limit (yellow) and Limit + 4dB (Red).

**MUTE** buttons illuminate when pressed and mute audio for that output channel.

**EDIT** buttons illuminate yellow when pressed. The buttons access gain on first press, last viewed parameter on second press, then exit on third press.

## 11) Rear Panel functions (l-r)



**Power Switch** - turns the unit's mains supply off and on.

**Mains Fuse** - located in a finger-proof holder adjacent to the mains inlet. (A spare fuse is also located inside this holder)

**Important** - Always replace the fuse with the correct type and rating as shown on the rear panel legend.

**Mains Inlet** - connected via a standard IEC socket.

**External** - 9 pin D-type connector for preset loading via RS232 PC link. See later for DX1.5 preset loading instructions.

**Outputs 1 - 6** – a 3 pin XLR socket is provided for each output. All are fully balanced, pin 2 hot, 3 cold, 1 screen.

**Recessed AES input switch** - to select AES digital inputs. Red LED will illuminate in the hole when AES inputs are selected, along with the corresponding front panel indicator.

**Inputs A & B** – a 3 pin XLR socket is provided for each input. Both are fully balanced – pin 1 = screen, pin 2 = hot, pin 3 = cold. Input A is also used as a 2-ch AES digital audio input when AES mode is selected.

## 12) Operating your DX1.5

### Start-up procedure

Switching on the unit will display a brief message detailing the unit type and software version running and all LEDs will briefly illuminate. The unit will then begin its countdown to the wake-up procedure, during which time the audio will fade up to the level last set. Metering will begin to operate when the fade-up starts.

### Preliminary Set-up

The procedure below should be followed when first installing a DX1.5 unit.

Design your crossover! To do this, press **MENU**, and use the **BACK** or **NEXT** key to select '**Crossover sub-menu**' and then press **ENTER**. Use the **BACK** or **NEXT** key to select '**Design a crossover**' and then press **ENTER**. Finally, use the **BACK** or **NEXT** key to select the desired routing and follow the set-up wizard to finalise your design.

Note that when in a menu, **ENTER** is always used to confirm selections. The current selection is marked with an asterisk '\*'.

Use the **EDIT** keys on each output channel with the **BACK** and **NEXT** keys to select the high pass filters, low pass filters, parametric equalisers etc. Note that when designing a new crossover, the high and low pass filters will be set to default values.

Use the **EDIT** keys on each input channel with the **BACK** and **NEXT** keys to select the gain, delay and parametric equalisers available on each input.

Note that if no action is taken in menu mode, the unit will return to normal 'default' mode after about twenty seconds. Repeat the above directions to return to menu mode.

### 13) Routing Options and Processing Blocks

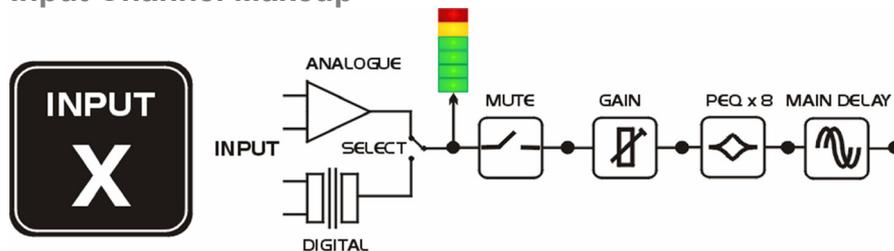
Due to the completely new DSP platform, the routing possibilities within the DX1.5 have been made completely flexible, with a matrix available allowing any combination of inputs to be routed to any output. The additional DSP power has permitted the inclusion of more processing blocks, even considering the extra inputs and outputs, and the doubling of sample rate.

To reduce set-up time and aid usability, several standard configurations are available as described in a later section.

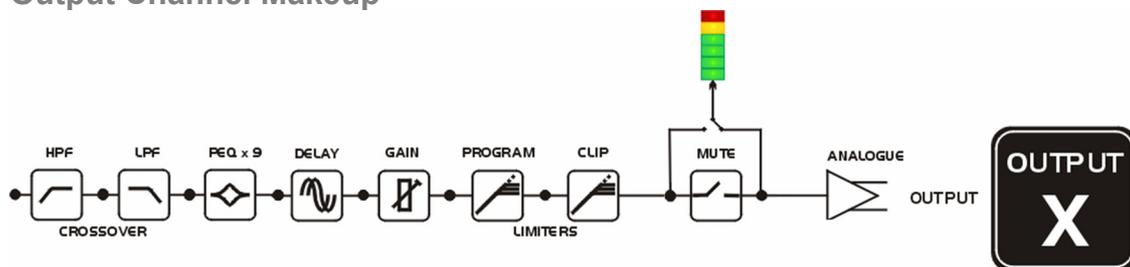
This section will outline the processing blocks available in relation to the signal path, and explain the various options for routing, including the "Free Assign" mode, which opens up completely flexible channel routing.

The diagrams below show the processing available on inputs and outputs – ignoring matrix routing.

#### Input Channel Makeup



#### Output Channel Makeup



#### Preset Routing Configurations

In addition to the ability to assign any combination of inputs to any output, a number of preset configurations are provided, for use when designing a crossover from scratch.

These have the advantage of suggested settings for the high and low pass filters to useful basic starting points, to filter the different outputs as appropriate for the chosen configuration. These may be freely modified afterwards should they not suit the requirements exactly.

## Free Assign Routing

If none of the preset configurations are appropriate to the required system setup, it is possible to manually select the routing of the crossover. This is achieved through the **Crossover Menu > Design a Crossover**.

Pressing **ENTER** will start the crossover design wizard, with the first option being to choose the routing.

The display will show, for instance

**Design a Crossover ->**  
**Routing = 2 X 3 WAY**

or whatever the current configuration is set to. Press **BACK** until the display shows

**Design a Crossover ->**  
**Routing = Free Assign**

And then press **ENTER**. The **EDIT** key will illuminate for output 1, as will any relevant input **EDIT** keys, showing which inputs are feeding output 1. The display will also detail the current combination of inputs feeding this output. To change the routing for any output, press its **EDIT** key, and then choose the required input channel combination by just pressing the input **EDIT** keys as appropriate. The input combinations can also be stepped through in turn by pressing **NEXT**, or **BACK**.

To complete the procedure, press **ENTER**.

The screen will show:

**Design a Crossover ->**  
**Clear Settings : No \***

Press **ENTER** again.

The wizard will continue, and if the routing has been changed, all outputs will be muted on exit.



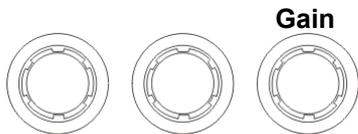
2) Press EDIT input key  
to select/deselect input

1) Press EDIT to show  
input routing selection

## 14) Editing Audio Parameters – Input Channels

### Input Gain

The range of the control over the input gain is  $-40\text{dB}$  to  $+6\text{dB}$  in  $0.1\text{dB}$  steps.



The display will show, for instance

```
InA Input A Gain  
Input Gain = +6.0dB
```

### Base Delay

The maximum available delay between any input and output is  $650.00\text{mS}$ . For example, if the input delay on channel A is set to  $500\text{mS}$ , the maximum available output delay for any output fed from input A will be  $150\text{mS}$ .

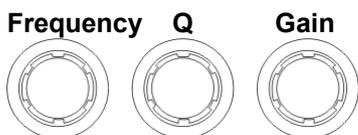


The display will show, for instance

```
InA Input A Delay  
Base Delay = 0.00mS
```

### Input Parametric EQ

There are eight bands of parameter equalisation available on every input. The behaviour of each individual band can be changed to a variety of different filter shapes, including high and low shelves, notch, and band-pass. Changing the filter type is achieved by pressing **ENTER** during editing any particular band.



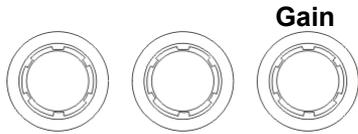
The display will show, for instance

```
InA Input A PEQ:1<>  
1k00Hz Q=3.0 0.0dB
```

## 15) Editing Audio Parameters – Output Channels

### Output Gain

The range of the control over the input gain is  $-40\text{dB}$  to  $+15\text{dB}$  in  $0.1\text{dB}$  steps.

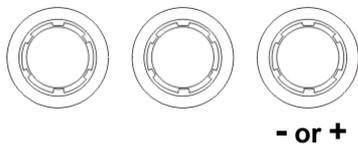


The display will show, for instance

```
OP1 Output 1 Gain
Output Gain = +6.0dB
```

### Output Polarity

The polarity (or phase) of each output may be switched individually as below.

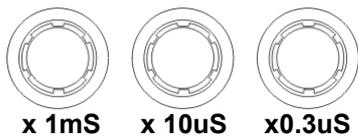


The display will show, for instance

```
OP1 Output 1 Polar.
Polarity = [+]
```

### Output Delay

The maximum available delay between any input and output is  $650.00\text{mS}$ . For example, if the input delay on channel A is set to  $500\text{mS}$ , the maximum available output delay for any output fed from input A will be  $150\text{mS}$ .



The display will show, for instance

```
OP1 Output 1 Delay
Delay = 0.000mS
```

### Output High Pass Filter

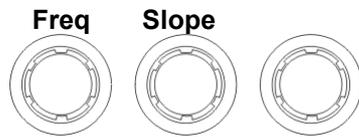
The high pass crossover filter on each output has a frequency range of  $<10\text{Hz}$  up to  $32\text{kHz}$  in  $1/36^{\text{th}}$  Octave steps.

If you try to set the high pass filter to a higher frequency than the low pass (which would be pointless and result in no output), the message High/Low Freq. Overlap! will be displayed. Note that to access the  $48\text{dB}/\text{Octave}$  filters, parametric bands 6 & 7 need to be bypassed, or set to  $0\text{dB}$ .

If they are not, the message

**Bypass PEQ's 6 & 7 To Access 48dB Slopes**

will be displayed.



The display will show, for instance

**OP1 Output 1 HPF /~~  
<10Hz Linkw-Riley 48dB**

### Output Low Pass Filter

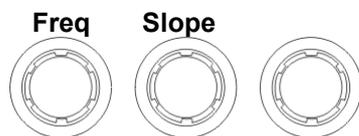
The low pass crossover filter on each output has a frequency range of 35.1Hz up to >32kHz in 1/36<sup>th</sup> Octave steps.

If you try to set the low pass filter to a lower frequency than the high pass (which would be pointless and result in no output), the message High/Low Freq. Overlap! will be displayed. Note that to access the 48dB/Octave filters, parametric bands 8 & 9 need to be bypassed, or set to 0dB.

If they are not, the message

**Bypass PEQ's 8 & 9 To Access 48dB Slopes**

will be displayed.



The display will show, for instance

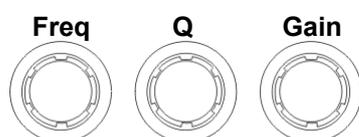
**OP1 Output 1 LPF ~~~\|  
>32kHz Linkw-Riley 48dB**

(Note that 2 bands each will be lost when using 48dB slope crossover filters, resulting in a maximum of 5 bands of EQ when both high and low pass are set to 48dB/Octave)

### Output Parametric EQ

There are nine bands of parametric equalisation available on every output. The behaviour of each individual band can be changed to a variety of different filter shapes, including high and low shelves, notch, and band-pass.

Changing the filter type is achieved by pressing **BYPASS** to bypass the filter and then pressing **ENTER** during editing any particular band.



The display will show, for instance

**OP1 Output 1 PEQ:1<>**  
**1k00Hz Q=3.0 0.0dB**

### Output Limiter

The limiter on each output has adjustable attack and threshold, with a release time that is selectable to be a multiplier of the attack time. For example, as shown below, the attack time is 2mS and release is “x16” so 32mS.

The attack and release times can be automatically linked to the high pass filter frequency, so that they are set to correct values for the output’s frequency range. If this feature is enabled, the display will show Automatic T/C in place of the attack and release times.

Selection of automatic time constants is through the Design a Crossover wizard, in the Crossover Sub-Menu.

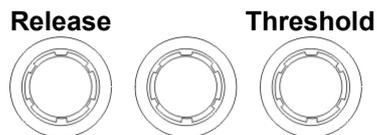


The display will show, for instance

**OP1 Output 1 Limiter**  
**Atk=2.0mS Rel=x16 +22dB**

### Output “D-Max” (ClipLim)

The clip limiter is designed to sit at a threshold just above the standard limiter and has a look ahead attack so that its threshold can never be exceeded. The release time can be automatically linked to the high pass filter frequency. If this feature is enabled, the display will show Rel. = Auto in place of the release time. Selection of automatic time constants is through the Design a Crossover wizard, in the Crossover Sub-Menu.



The display will show, for instance

**OP1 Output 1 ClipLim**  
**Rel.=Medium 6dB Above**

## 16) Input Ganging and Output Ganging

The method of linking inputs or outputs together during editing is achieved in the same way, so only crossover (output) ganging will be explained here. Having selected Crossover Ganging from the menu under the Crossover Sub-Menu, the current ganging set-up will be displayed.

This will either be a preset selection as would be useful in a standard crossover configuration – for example

### <-Crossover Ganging Ganging=1+4 2+5 3+6

...would be a logical ganging arrangement if the crossover was set up as a 3 x 2 way – linking the control and adjustment of all “Low” outputs together, all “Mid” outputs together and all “High” outputs together.

However, if the crossover has not been set up with a preset routing configuration, then it may be required to set up the ganging to compliment this configuration. This is achieved using the Free Assign mode.

This is selected from the preset ganging choices, which are:

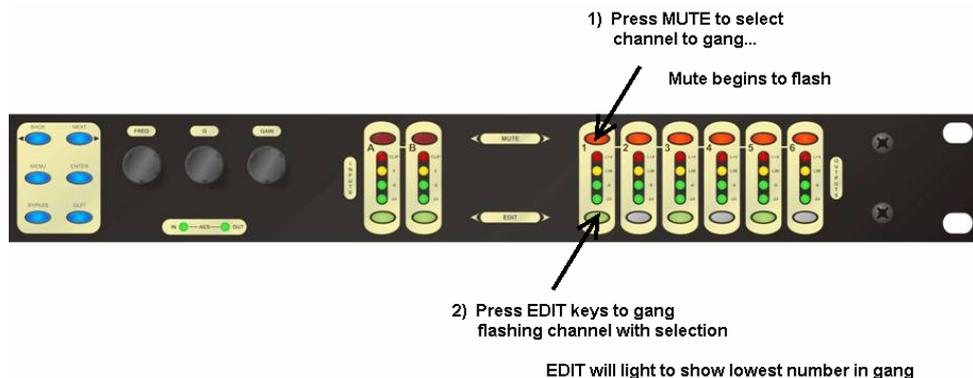
- Ganging = None
- Ganging = Free Assign
- Ganging = 1+2+3+4+5+6
- Ganging = 1+4 2+5 3+6
- Ganging = 1+3+5 2+4+6
- Ganging = 1+3 2+4

Selecting **Free Assign** and then pressing **ENTER** will begin the process of ganging outputs together using the following simple rules:

All outputs are ganged to the lowest number – so to gang 3 & 5, 5 must be selected and then ganged to 3.

Outputs cannot share more than one ganging set – so for example output 3 cannot be ganged to 2 and 4 unless they are ganged together as well. (Effectively 3 and 4 are ganged to 2 in this case)

With these rules in mind, selecting and setting up gangs is quite straightforward.



Press a **MUTE** key to choose the output to gang – its LED will begin to flash, and an **EDIT** key will illuminate to show which output it is currently ganged with.

To change this selection, just press another **EDIT** key, remembering that gangs work from the highest to lowest number. So, to gang outputs 1 and 5, press **MUTE** 5 then **EDIT** 1 – the display will show

### <-Crossover Ganging Gang Output 5 with 1

Ganging is cleared by selecting **Ganging=None** from the initial choices given above.

## Input ganging

The Input Ganging procedure is identical to the crossover ganging, selectable under the Input Sub-Menu.

## 17) Memory Structure

The DX1.5 has its memories split into sections, allowing independent recall of crossover settings (i.e. all parameters associated with outputs), and input settings.

There are, therefore, three types of memory available

**INPUT ONLY**

**CROSSOVER ONLY**

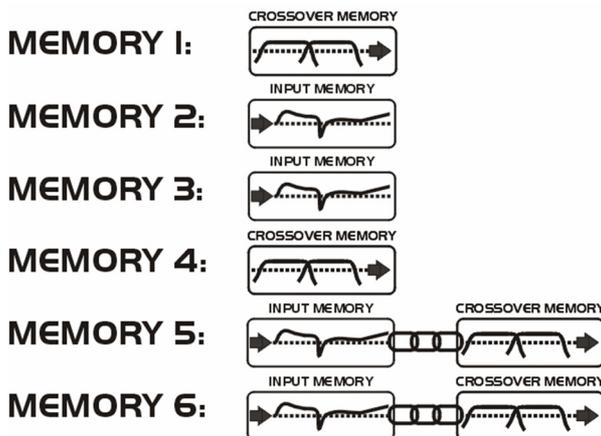
**INPUT & CROSSOVER.**

These appear in the **GLOBAL MEMORY Sub Menu**, and its operation warrants a little more explanation.

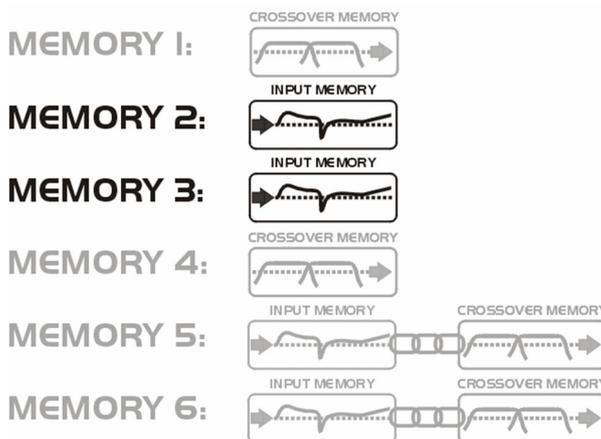
Selecting to Store or Recall using the Global Memory option offers the possibility of storing various combinations of the available memory types, and these are selected using the **BACK** and **NEXT** keys.

To explain how this all works, please consider the following example.

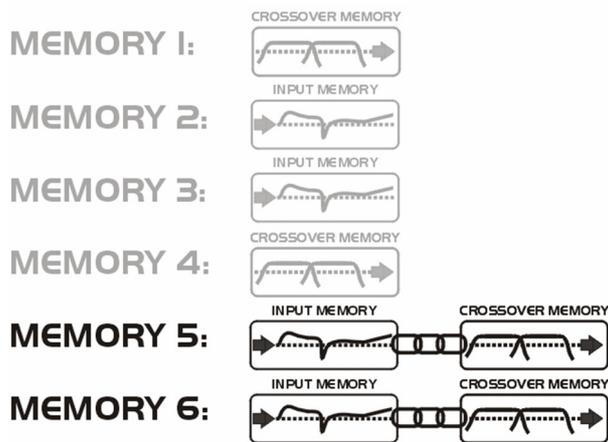
There are 6 memories stored in the unit with various combinations of crossover only, input only and input & crossover memories. Different memory locations contain different combinations.



If it is required to recall a location that contains, for instance, input only memories this will limit the selection to memories 2 & 3.



Recalling, for instance, input & crossover memories will limit the selection to memories 5 & 6.



Note that storage and erasure of memories does not follow quite the same rules, being simpler in its operation.

Selecting **Input and Crossover** during a Store will skip any memories that have other combinations in them.

Selecting Erase for any combination will show only locations that have EXACTLY that combination – it is not possible to erase just one part of a combination memory.

The DX1.5 has 256 memory locations, but these are dynamic in nature – obviously a memory containing Input and Crossover settings takes up more space than one containing just Input settings.

## 18) Loading new software via a pc

The unit's internal software may be updated via the RS232 port ONLY, one unit at a time using the pc loader software. Your Martin Audio distributor or dealer will let you know if updates are necessary.

## 19) AES Inputs

DX1.5 units have AES input facilities (via input XLR A) built in as standard. This allows the unit to receive digital audio directly

Input selection is via a recessed switch on the rear panel of the unit, near output 1. A red LED inside this aperture illuminates to show that the AES digital inputs have been selected. A complimentary LED on the front panel also illuminates.

The rear panel switch controls the rear panel LED directly, whilst the front panel one is via the processor, allowing it to relay a little more information.

If it is flashing, this means that AES inputs have been selected but have not locked. Once a stable AES signal is being received, it will be permanently illuminated.

### AES Diagnostics and Status Information

Also under the **AES/EBU Sub Menu** is the AES Status Information option, which can be used to check the incoming sample rate(s) and confirm that the data is being received correctly.

Pressing **ENTER** will first show something like

```
AES Device Status  
V1: E V2: E
```

This display shows the correct operation of the AES transmitters V1 & V2. The letter after each is the silicon version (and is of no importance to the user).

Pressing **ENTER** again will show

```
AES Device Status  
V : 96k0
```

This display shows the status of the AES receiver. The sample rate the unit has been able to lock to is shown, or **UNLOCKED** will be displayed in its place. The unit will lock to sample rates from 32kHz up to and including 192kHz.

The unit's own processing sample rate is 96kHz, and AES output data is always at 96kHz. Internal sample rate converters will translate all incoming rates to 96kHz – one converter for each AES input. This allows the two input streams to be at different rates if necessary.

## 20) Warranty

Martin Audio DX1.5 Loudspeaker Management Systems are warranted against manufacturing defects in materials or craftsmanship over a period of 1 year from the date of original purchase.

During the warranty period Martin Audio will, at its discretion, either repair or replace products which prove to be defective provided that the product is returned in its original packaging, shipping prepaid, to an authorised Martin Audio service agent or distributor.

Martin Audio Ltd. cannot be held responsible for defects caused by unauthorised modifications, improper use, negligence, exposure to inclement weather conditions, act of God or accident, or any use of this product that is not in accordance with the instructions provided by Martin Audio. Martin Audio is not liable for consequential damages.

This warranty is exclusive and no other warranty is expressed or implied. This warranty does not affect your statutory rights.