Revision 0.6. - 290306
Preliminary Release.

[Users Guide]

Software Release 5.0

Applies to the following models:

MSD600M, MSD600M++,
MSD200C, MSD600C-III, MSD600C-5.1,
PT200C, PT600C-III, PT600C-5.1,
PT0660M and PT0660M-LS.

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PRELIMINARY Release

1 INTRODUCTION

Congratulations! By purchasing a Master Stereo Display (MSD) from DK-Technologies you have decided to actually "see what you hear!" We are certain that your Master Stereo Display will prove an invaluable tool in your daily work.

1.1 This Manual

To obtain higher software reliability and a simpler way to ensure flawless software support, tracking and in-field software upgrade ability, all newer models of the MSD family have been designed around a single common operating system (OS). During power-up the OS is being configured according to the hardware and functions found on the actual MSD model.

To reflect the flexibility of the OS, this manual will describe all functions found in the OS with no regard to how many of the functions are actually available in your MSD model. Some menu functions will vary from model to model because of the difference in input and output modules/interfaces in the unit.

This software manual covers the following MSD models:

MSD600M, MSD600M++,
MSD600C-III, MSD600C-5.1, MSD200C,
PT600C-III, PT600C-5.1, PT200C,
PT0660M and PT0660M-LS.

All trough this manual any of the supported models will be referred to simply as the MSD.

1.2 The formatting of this manual.

This manual uses different typefaces for different information.

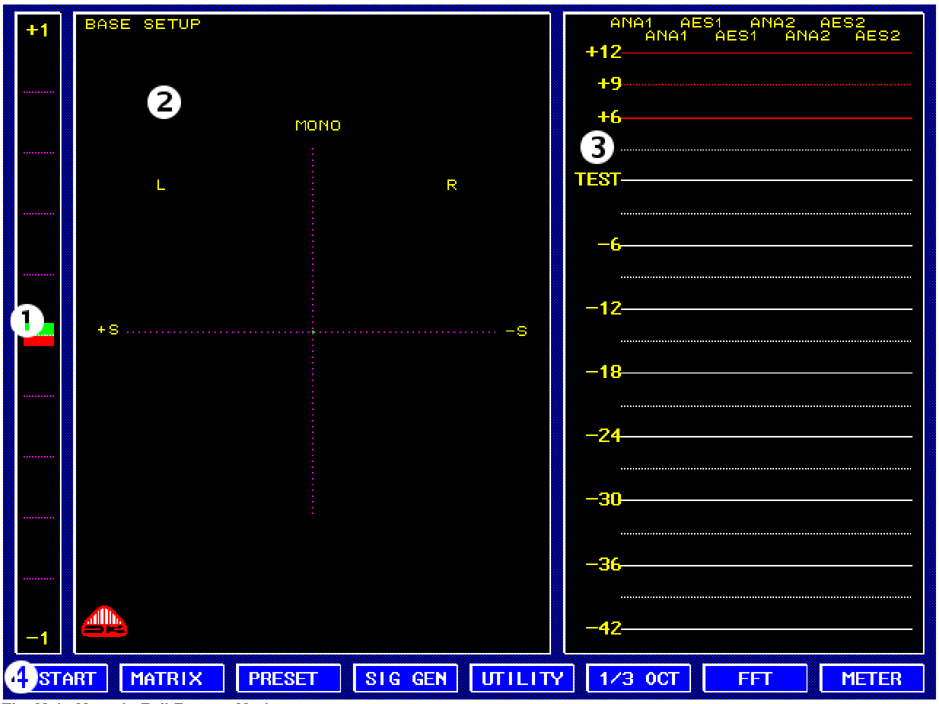
[INFO] – This is a key in a menu that can be pressed.

'**INFO**' - This is a message or item that can be selected on the LCD screen.

In the headlines above some of the sections of this manual are images of keys from the softkey menues. These images represent the necessary key order to reach a particular function from the main menu in 'Full Feature Mode'.

2 Full Feature Mode.

When the MSD is powered up for the first time the MSD will start in the ‘Full Feature Mode’



The Main Menu in Full Feature Mode.

In ‘Full Feature Mode’ the MSD will show the three main metering functions and the main menu.

❶: The Phase Correlation Meter.

❷: The Audio Vector Oscilloscope / Jelly-Fish™

❸: The Peak Programme Meter (PPM).

❹: The main softkey menu.

2.1 Navigating the softkey menus.

The MSD is controlled using the eight softkeys placed below the LCD display. These softkeys correspond to the menues in the lower part of the LCD display. These menues can have up to 8 different items or functions. The leftmost key in a menu will in most cases exit to the previous menu level.

An example is the tone generator menu, where it is possible to change the frequency and the output level of the built in sinus generator.

The sinus-generator menu.

Key ❶ will exit to the previous menu.

Key ❷ is a flag that can be set or cleared, enabling or disabling a particular function. If a flag is set, the function is enabled and the square will be green.

Key ❸ and ❻ (**[HERTZ]** and **[dBu]**) are just labels, they have no function.

Key ❹, ❺, ❼ and ❽ are two pairs of edit keys which will increase or decrease the value shown between them.

3 The Application Start Menu. START

3.1 Navigating the Application Start List.

START ➔ ▲ ▼ SELECT

The Application Start List.

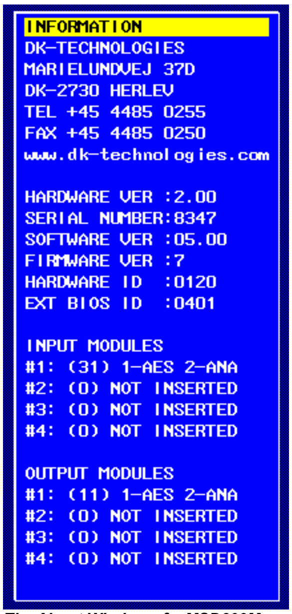
The 'Application Start List' is an easy way to access commonly used applications in the MSD.

To execute an application in the MSD e.g. the FFT-Analyser, use the **[▲] / [▼]** keys to move the green selection bar to the application that has to be executed and then press **[Select]**. If write protection is disabled *(see section 3.4)* while selecting an application then this application is the first to be executed after power up or a reset. If an application is selected as the default start up application then the particular item in the 'Application Start List' will be green instead of white. In this case the 'Application Start List' is the first to be shown.

The following applications are available from the start window.

START MENU	This Menu.	Section 3
FULL FEATURE MODE	Normal MSD operation.	Section 2
	Phase Correlation Meter.	Section 7.1
	Audio Vector Scope / Jelly-Fish™.	Section 7.2
	PPM-Bargraphs.	Section 7.3
EASY SHORT HAND	Quick Preset Selection. ***(Not implemented.)***	
DMU-1630 EMULATION		Section 9.1.1
SESSION LOGGING		Section 9.3
GRAPHIC LOUDNESS		Section 9.2
DCF77 RADIO CLOCK	Radio Controlled Real Time Clock.	Section 9.4
1/3 ANALYSER	The 1/3 Octave Analyzer.	Section 14.
FFT ANALYSER.	The FFT Analyzer.	Section 13.
PRESET LIST	Preset selection menu.	Section 4.
MATRIX COMPACT	The Standard Audio Matrix menu.	Section 5.
MATRIX EXTENDED	The Extended Audio Matrix menu.	Section 6.
TONEGENERATOR	Tonegenerator Menu.	Section 8.
METERING MENU	PPM bargraph configuration menu.	Section 9.

3.2 The About Window. START ➔ ABOUT



The About Window of a MSD600M++.

The MSD have been designed using FLASH-Memory. Besides holding the OS this FLASH memory also stores other vital information used by the MSD.

In addition to being used as a product verification tool the information found in the **[ABOUT]** window is critical when it comes to updating the MSD with new features as they are released by DK-Technologies. Software updates are the most common way to enable new functions in the MSD. To enable new hardware functions not yet implemented, the firmware that configures the hardware of the MSD can in rare cases also be updated. For information regarding the latest software and firmware releases please visit our website at: www.dk-technologies.com.

When contacting technical support please provide the information found in this window. This will greatly help us with a fast and accurate response.

Additionally if the MSD is modular, a list of installed input and output modules are shown.

3.3 LCD Backlight intensity. START ➔ DARKER BRIGHT

To adjust the LCD display's brightness, use the keys **[Darker]** or **[Bright]**. Adjustments are made in eight steps.

To store this setting please see section 4.21 on how to save a preset. Please note that the backlight setting is a global setting that affects all eleven presets.

3.4 Write protection. START ➔ ENABLE

To avoid accidental changes to the presets the MSD is fitted with a Write-Protect function. Pressing the **[Enable]** key toggles the MSD write protection. If the key is greyed out, the MSD is write protected and changes to the presets cannot be saved.

The MSD Write-Protect function is also used to prevent other critical functions like the **[Reset]** function to be accessed or changed by a mistake

3.5 Reset. START ➔ RESET

The reset function works like a Power-up of the unit, generating a hardware reset followed by a total initialisation of all hardware and software related functions.

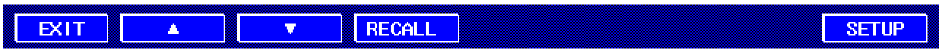
Just like a power-up the **[RESET]** function will force the MSD to load all settings from the default startup preset. Since the power supply for the MSD often is placed “out of reach”, the reset function is a very handy way to force the MSD trough a Power-up sequence.

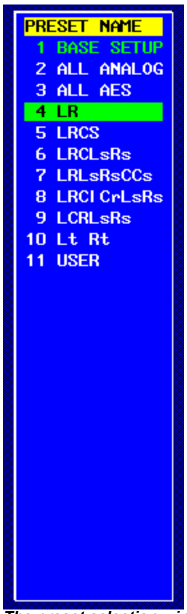
The **[RESET]** key is considered a critical function and is therefore greyed out until the write protection is disabled. *(Section 3.4.)*

Do not mistake this RESET function with the same named menu functions in the METER UTILITY DMU-1630 scale.

This is NOT a factory reset, the MSD is only restarted.

4 Preset selection menu. PRESET





The preset selection window.

To make the daily operation of the MSD easier the MSD has a total of eleven possible presets, where all are user-definable. By establishing a number of presets designed for a specific purpose, these presets can easily be recalled for a specific job. All the presets will be fully-set with parameters such as inputs, PPM scales, oscilloscope modes (surround or stereo), reference levels, colour coding etc.

Depending on the MSD model a certain number of the available presets are used as Factory Presets. The remaining preset locations will be used as USER PRESETS. Each MSD model will have its own set of factory presets reflecting the functionality of the specific model.

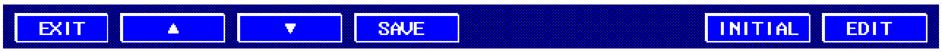
For easier setup, the configuration of the MSD (all 11 presets) can be uploaded and edited in the Windows® program called DK-Matrix. Please refer to the software manual for DK-Matrix for further details.

The Preset Selection Menu can easily be accessed from the 'Application Start List' by selecting the item 'Preset List'. The Preset Selection List can also be accessed from the 'Main Menu' in 'Full Feature Mode' by pressing the **[Preset]** key.

4.1 Recall a Preset. PRESET ➔ ▲ ▼ RECALL

To recall a preset, use the **[▲]** / **[▼]** keys to move the green selection bar to the desired preset and then press the [Select] key.

4.2 Preset Setup. PRESET ➔ SETUP



When the MSD write protection are disabled the **[Setup]** key is enabled. Please see section 4.4 on how to disable the write protection. Pressing the **[Setup]** key will bring up the 'Preset Setup Menu'.

4.2.1 Save a Preset. PRESET ➔ SETUP ➔ ▲ ▼ SAVE

To save a preset use the **[▲]** / **[▼]** keys to move the green selection bar to the preset which shall be replaced by the new settings.

When the save key is pressed, the selected preset will be replaced by the new settings, and the MSD write protection will be enabled. The setup menu will also be exited.

Please note that if preset 1 the 'Base Setup' has to be replaced, the **[Save]** key has to be held down for about 10 seconds.

4.2.2 Default Startup Preset. PRESET ➔ SETUP ➔ INITIAL

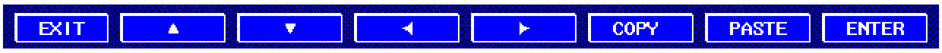
When the MSD is powered up or restarted it will by default load preset 1 the 'Base Setup'. It is possible to select another startup preset by moving the green selection bar to another preset and press the **[Initial]** key. When a preset is selected as the default startup preset the item in the 'Preset List' will be green.

4.2.3.1 Rename a Preset. PRESET ➔ SETUP ➔ EDIT

The preset selection window in edit mode.

By pressing the **[Edit]** key a preset can be renamed to a more describing name.

When the **[Edit]** key is pressed the edit menu will appear and the green Selection Bar will become pink showing a flashing cursor below a character in the preset name.

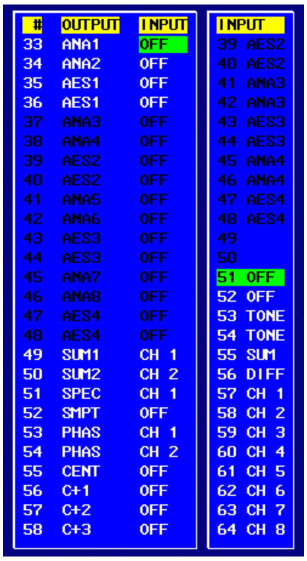


To select a new character at this position use the **[▲]** / **[▼]** keys to step through the different characters. Use the **[◄]** / **[►]** keys to move the cursor to the next position in the preset name. The preset name can consist of up to 10 alpha-numeric characters. The name of the currently selected preset will always be shown in the top left corner of the 'Audio Vector Scope' window.

To save the changes to the preset name use the **[Enter]** key. The changes are saved to the flash memory immediately, as a result it is not necessary to save the preset again.

Press **[Exit]** to abandon the changes.

5 The Compact Audio Matrix. MATRIX



The Matrix Status Window with the source window open.

The OS has been built around an audio matrix. This means that any of the 32 input channels or 16 output channels are not hard wired to any specific function in the MSD. This means that even left and right channels in an AES-3 signal can be routed separately; they can even be converted to an analogue signal if the MSD is fitted with output modules.

Selecting **'Matrix Compact'** from the **'Application Start List'** or pressing the **[MATRIX]** key from the **'Full Feature Mode'** will open the matrix status window at the left side of the screen. The MSD always looks at the signal routing from a DESTINATION (output) point-of-view.

The matrix status window is divided into three columns. The first column is the number of the destinations (outputs) in the matrix. There are a total of 96 destinations in the matrix where 16 of them (number 33 to 48) are physical analogue and digital outputs from the MSD. The rest of the destinations are outputs to software functions. I.e. the FFT-Analyser or a PPM-bar.

The second column shows the name of the destination, this name can be edited with the Windows® based software called DK-Matrix. Please refer to the DK-Matrix manual for further information.

The third column is the source (input) for the destination.

To set a new x-point (route a source to a destination), it is necessary to move the green cursor in the input column to the appropriate destination. Only white lines in the matrix window can be selected. In this case destinations from number 37 to 48 are black. The reason for this is that this particular MSD has only been fitted with one output module.

When a destination has been selected using the **[▲]** / **[▼]** or **[page up]** / **[page down]** keys, a new window will appear when the select key is pressed. This window is the matrix source window. There are a total of 88 sources in the matrix, where 48 of them (number 1 to 48) can be physical inputs but only up to 32 at a time. An example of this could be if there is mounted an analogue input module in input slot number 1 then the module will register as input number 33 to 36 and then input number 1 to 8 will not be available.

5.1 Audio Matrix Destination List.

The following is a list of destinations in the matrix:

Destination Number	Destination Name	Description.
1..32	DM 1 .. DM31	Reserved for future use.
33..48		Physical outputs from the MSD. Analogue and digital AES-3.
49..50	SUM1 / SUM2	Destinations for the sum and difference amplifier. See section 6.4.3 for further details.
51	SPEC	Destinations for the FFT and 1/3 Octave Analyser. See section 10 for further details.
52	SMPT	Destinations for the internal SMPT-Timecode decoder.
53..54	PHAS	Destinations for the phase correlation meter and the stereo audio vector oscilloscope. See section 7.2 for further details.
55..62	CENT, C+1..C+7	Destinations for the Jelly-Fish™. See section 7.2 for further details.
65..96		Destinations for the PPM-Bargraphs. See section 7.3 for further details.

Special attention must be taken to the PPM destinations. All 32 possible PPM Destinations are fed back into the Matrix. This will make it possible to use the PPM destinations as sources from the 'Matrix Source Window'. Using this feature configuring the Matrix can be simplified by only assigning the physical Input to a single destination (the PPM) even though the same signal are used in several places. Please see the configuration of the applied factory presets which all are using this type of PPM signal feedback.

5.2 Audio Matrix Source List.

The following is a list of sources in the matrix:

Source Number	Source Name	Description.
1..48		Physical inputs to the MSD. These inputs could either be analogue, digital or even embedded audio in a HD-SDI signal.
49..50		Not used.
51..52	OFF	These two inputs can be used to disable a destination in the matrix. I.e turn off a PPM-bar.
53..54	TONE	Sources from the internal signal generator. See section 8 for further details.
55..56	SUM / DIFF	Sources from the sum and difference amplifier. See section 6.4.3 for further details.
57..88	CHxx	Sources from the PPM-bars. As the only destination in the matrix the PPM-bars can also be used as a source. These would normally be routed to the Phase Correlation Meter and Vector Scope / Jelly-Fish™.

5.3 Route signals in the 'Audio Matrix'.

Example 1.
If the MSD is equipped with a standard input and a standard output module which both have an analogue stereo pair and a digital AES-3 connection, the MSD can be used as a digital to analogue converter.

The procedure to configure the matrix for this functionality can be divided into three steps.

Step 1:

Press the **[MATRIX]** key from the **'Full Feature Mode'** and the Matrix Status Window will appear.

In the Matrix Status Window, use the **[▲]** / **[▼]** or **[page up]** / **[page down]** keys to move the green cursor to destination #33 which is the first analogue audio channel on the output module.

Step 2:

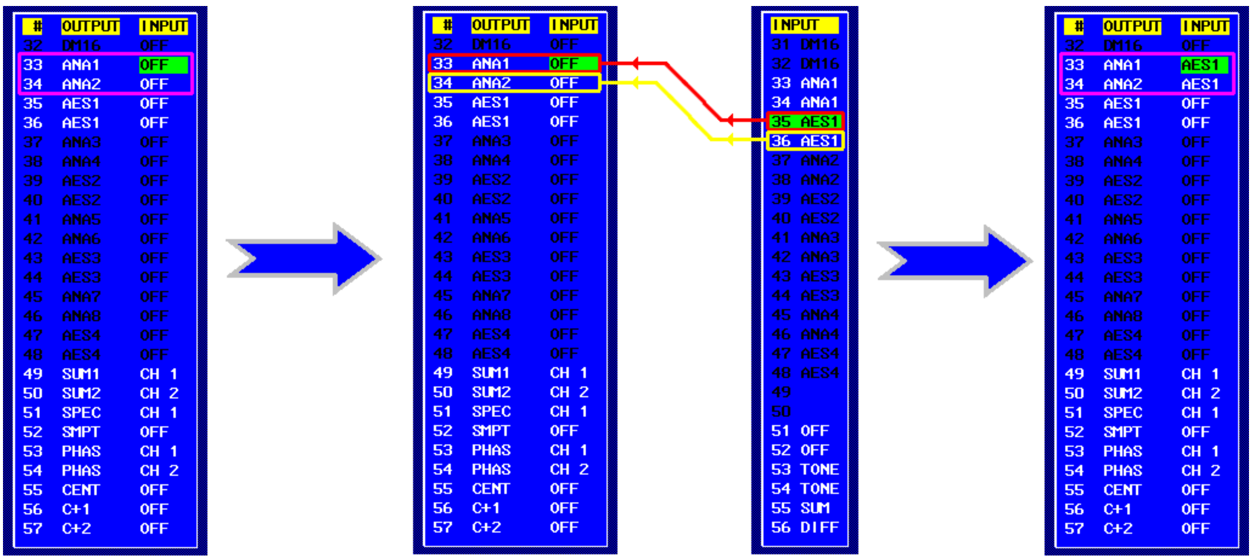
To route the digital signal from the input module to this output press the **[SELECT]** key and the Matrix Source Window will appear. In this window move the green cursor to source #35. This source is the first channel in the digital AES-3 stream if the module in input slot #1 is a standard input module. Press **[SELECT]** to set the X-Point. The first channel in the digital AES-3 stream has now been routed to the first analogue output.

It is now necessary to route the second channel in the digital AES-3 stream to the second analogue output. Proceed as in step 1 and 2 but select destination #34 and source #36 instead.

Step 3:

When both channels have been routed, the MSD will function as a digital to analogue converter.

Step 1. The Matrix Status Window. **Step 2.** Select source in the Matrix Source Window. (Window at the right.) **Step 3.** New sources selected.



Example 2.

To extend the setup in example 1 it could be interresting to monitor the digital input level on the PPM bars.

Step 1.

In the Matrix Status Window, use the **[▲]** / **[▼]** or **[page up]** / **[page down]** keys to move the green cursor to destination #65 which is the first PPM-Bargraph.

Step 2:

To route the digital signal from the input module to this PPM-Bargraph press the **[SELECT]** key and the Matrix Source Window will appear. In this window move the green cursor to source #35. If the module in input slot #1 is a standard input module this source is the first channel in the digital AES-3 stream. Press **[SELECT]** to set the X-Point. The first channel in the digital AES-3 stream has now been routed to the first PPM-Bargraph.

It is now necessary to route the second channel in the digital AES-3 stream to the second PPM-Bargraph. Proceed as in step 1 and 2 but select destination #66 and source #36 instead.

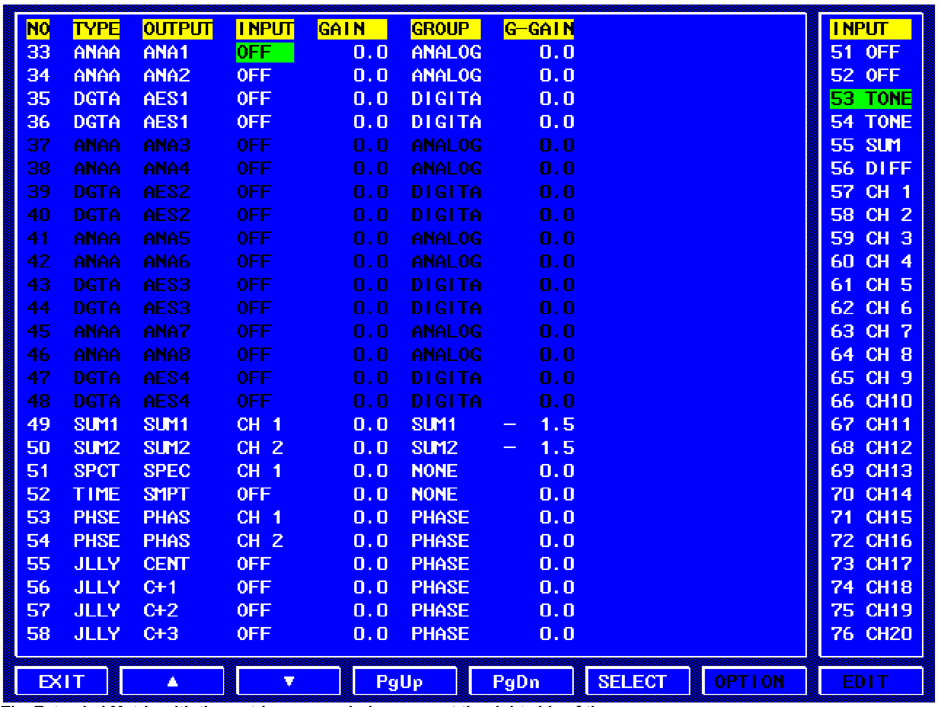
Step 3:

When both channels have been routed, the MSD will now show the digital AES-3 stream on PPM-Bargraph 1 and 2.

6 The Extended Audio Matrix. MATRIX ➔ EXTEND

As the name implies the ‘Extended Audio Matrix’ is an extended version of the ‘Compact Audio Matrix’. The ‘Extended Audio Matrix’ has all the same functions as the ‘Compact Audio Matrix’ plus a lot more features.

The ‘Extended Matrix’ window is divided into seven columns.
The first column is a fixed number of the destinations (outputs) in the matrix.
The second column is the type of output. I.e. analogue, digital, Jelly FishTM.
The third column shows the name of the destination.
The fourth column is the source (input) for the destination.
The fifth column is the output gain for the destination.
The sixth column is the group the destination belongs to.
The seventh column is the global output gain. (Master Group)



NO	TYPE	OUTPUT	INPUT	GAIN	GROUP	G-GAIN
33	ANAA	ANA1	OFF	0.0	ANALOG	0.0
34	ANAA	ANA2	OFF	0.0	ANALOG	0.0
35	DGTA	AES1	OFF	0.0	DIGITA	0.0
36	DGTA	AES1	OFF	0.0	DIGITA	0.0
37	ANAA	ANA3	OFF	0.0	ANALOG	0.0
38	ANAA	ANA4	OFF	0.0	ANALOG	0.0
39	DGTA	AES2	OFF	0.0	DIGITA	0.0
40	DGTA	AES2	OFF	0.0	DIGITA	0.0
41	ANAA	ANA5	OFF	0.0	ANALOG	0.0
42	ANAA	ANA6	OFF	0.0	ANALOG	0.0
43	DGTA	AES3	OFF	0.0	DIGITA	0.0
44	DGTA	AES3	OFF	0.0	DIGITA	0.0
45	ANAA	ANA7	OFF	0.0	ANALOG	0.0
46	ANAA	ANA8	OFF	0.0	ANALOG	0.0
47	DGTA	AES4	OFF	0.0	DIGITA	0.0
48	DGTA	AES4	OFF	0.0	DIGITA	0.0
49	SUM1	SUM1	CH 1	0.0	SUM1	- 1.5
50	SUM2	SUM2	CH 2	0.0	SUM2	- 1.5
51	SPCT	SPEC	CH 1	0.0	NONE	0.0
52	TIME	SMPT	OFF	0.0	NONE	0.0
53	PHSE	PHAS	CH 1	0.0	PHASE	0.0
54	PHSE	PHAS	CH 2	0.0	PHASE	0.0
55	JLLY	CENT	OFF	0.0	PHASE	0.0
56	JLLY	C+1	OFF	0.0	PHASE	0.0
57	JLLY	C+2	OFF	0.0	PHASE	0.0
58	JLLY	C+3	OFF	0.0	PHASE	0.0

INPUT
51 OFF
52 OFF
53 TONE
54 TONE
55 SUM
56 DIFF
57 CH 1
58 CH 2
59 CH 3
60 CH 4
61 CH 5
62 CH 6
63 CH 7
64 CH 8
65 CH 9
66 CH10
67 CH11
68 CH12
69 CH13
70 CH14
71 CH15
72 CH16
73 CH17
74 CH18
75 CH19
76 CH20

The Extended Matrix with the matrix source window open at the right side of the screen.

6.1 Set X-Point in ‘Extended Audio Matrix’.

MATRIX ➔ EXTEND ➔ ▲ ▼ SELECT

To set a X-Point in the ‘Extended Audio Matrix’ use the **[▲]** / **[▼]** keys to move the green cursor to the preferred destination. When the destination has been selected use the **[Select]** key to open the Matrix Source Window. When the ‘Matrix Source Window’ are open then use the **[▲]** / **[▼]** or **[page up]** / **[page down]** keys to move the cursor to the preferred source and the press **[Select]** to set the new X-Point or press **[Exit]** to cancel.

6.1.1 Special option for the PT0660M Series.

MATRIX ➔ EXTEND ➔ WHEL ▲▼

The PT0660M has a special selection wheel on the front panel. This wheel can be used to move the cursor around in the ‘Extended Matrix’ window and the **[WHEL]** key will change the direction of the wheel. (Up/down or left/right) Pressing the wheel when the cursor is in the output collumn the matrix source window will open.

6.2.1 Input Options. MATRIX ➔ EXTEND ➔ OPTION

Depending on the type of physical inputs in the MSD there are in the ‘Matrix Source Window’ different options available.

To access the input options first select the desired source in the ‘Matrix Source Window’ using the green selection bar and then press the **[OPTION]** key. If the option key is greyed out no options is available for that particular source.

Note for PT0660M users: Pressing the wheeler will have the same effect as pressing the **[OPTION]** key. The **[#1]** key will in most cases exit to the previous menu.

The following is a list of input modules with corresponding input options.
8 Channel Analogue Input Module (8A) – No options available.

Analogue Input Module (11) – Analogue – No options available.
Analogue Input Module (11) – Digital – No options available.

Analogue Input Module (31) – Analogue – No options available.
Analogue Input Module (31) – Digital – AES Input Options. 6.1.2.1.1.

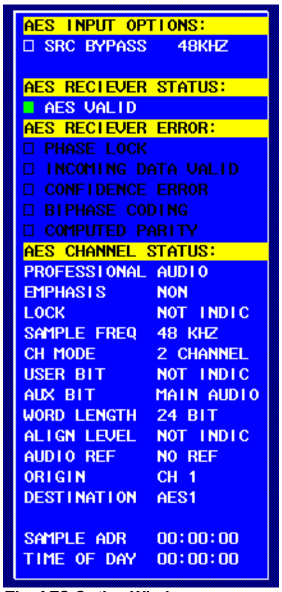
2 Channel AES Input Module (22) – AES Input Options. Section 6.1.2.1.1.

4 Channel AES Input Module (24) – AES Input Options. Section 6.1.2.1.1.

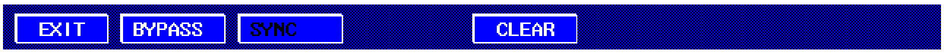
SDI-DeEmbedding Module (A2) – SDI Input Options. Section 6.1.2.2

HD SDI-DeEmbedding Module (E2) – HD SDI Input Options. Section 6.1.2.3.

6.2.1.1.1 AES Input Options. OPTION



The AES Option Window.



The AES Input Option Window is not only used to control the AES Sample Rate Converter Bypass (SRC Bypass), but it is also an AES Receive and Status analyser.

The AES Option menu is a “sticky” menu. This means that the displayed Receive and Input statuses are not dynamically updated. The MSD reads the current status when entering this option Menu and will update the displayed information only after re-entering the menu or pressing the **[CLEAR]** key.

Since the AES options are highly hardware dependent not all MSD models covered in this manual will support these functions.

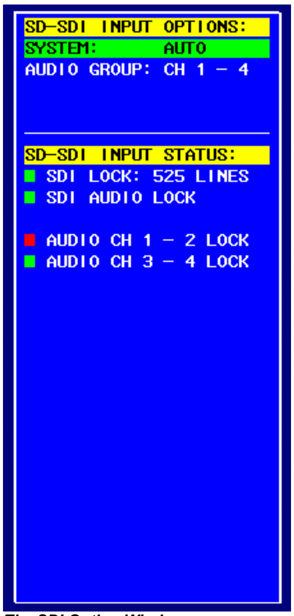
6.2.1.1.2 SRC Bypass and External Sync. BYPASS SYNC

To bypass the Sample Rate Converter (SRC) on one of the AES sources, first select the desired AES source in the 'Matrix Source Window' using the green selection bar and then press the **[Bypass]** key to bypass it.

The SRC's on all the AES Inputs can be individually bypassed. When bypassing a SRC it is important to remember that the incoming AES signal must be synchronised to the MSD. In most situations it is not recommended to bypass the SRC due to the risk of generating a synchronisation error. Such errors will generate full-scale "clicks" in the audio stream. Without audibly monitoring the measured AES signal, these clicks could be mistaken for a real audio signal.

The MSD will be delivered from the factory with the SRC on.
Please see section 11.5.1 for further information on external synchronisation.

6.2.1.2 SDI Input Options. OPTION



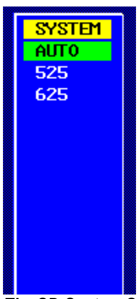
The SDI Option Window.

If the MSD is fitted with a SDI De-Embedding module it is possible to extract (de-embed) the audio and use it in the MSD as any other digital audio source.

Since the SDI options are highly hardware dependent not all MSD models covered in this manual will support these functions.

There can be a total of 16 channels embedded in a SDI signal. These audio channels are grouped into four groups of four channels. Only one group of channels can be de-embedded at a time.

6.2.1.2.1 SDI System select.



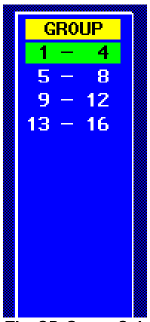
The SD-System Select Window.

The video system is normally detected automatically, but it can be forced to be either 525-lines NTSC or 625-lines PAL.

To select between video systems use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the item called '**SYSTEM**' and then press the **[SELECT]** key.

From the system select menu, use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the desired system and then press the **[SELECT]** key. If the correct video system is detected the '**SDI Lock**' flag in the SDI option Window will turn green.

6.2.1.2.2 SDI Audio Group selection.



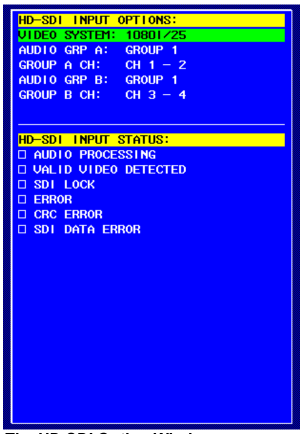
The SD-Group Select Window.

To select one of the four audio groups in the SDI video signal use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the item called '**AUDIO GROUP**' and then press the **[SELECT]** key.

From the group select menu, use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the desired audio group and then press the **[SELECT]** key. If the selected audio group contains valid audio, the 'SDI AUDIO Lock' flag will turn green. The 'AUDIO CH 1 – 2 LOCK' and 'AUDIO CH 3 – 4 LOCK' flag will also turn green if the respective channel pairs contain valid audio.

Note: If system select is set to AUTO detect and the selected audio group does not contain any valid AES audio, the MSD will try to detect the video system and the '**SDI Lock**' flag will alternate between 525 Line NTSC and 625 Line PAL.

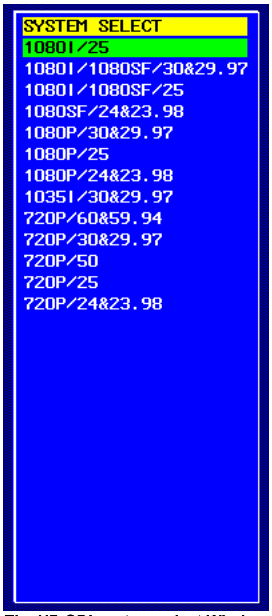
6.2.1.3 HD-SDI Input Options. OPTION



The HD-SDI Option Window.

If the MSD is fitted with a HD-SDI de-embedding module it is possible to de-embed the audio from the digital video signal. As with the standard SDI-signal there can be a total of 16 channels embedded in the video signal. These audio channels are grouped into four groups of four channels. Each group can be split up into two stereo pairs. It is therefore possible to de-embed a stereo pair from two different audio groups. Please observe that only two stereo pairs can be de-embedded at a time.

6.2.1.3.1 HD-SDI Video System.



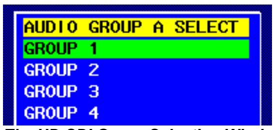
The HD-SDI system select Window.

The HD-SDI de-embedding module supports 13 different video systems. To select between video systems use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the item called '**VIDEO SYSTEM**' and then press the **[SELECT]** key.

From the '**SYSTEM SELECT**' window use the arrow **[▲]** / **[▼]** keys to select the preferred video system and then press **[SELECT]** to set the system or press **[EXIT]** to abandon the changes. Please note that the MSD can not automatically detect the video system.

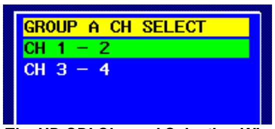
6.2.1.3.2 HD-SDI Audio Group selection.

Since it is possible to de-embed audio from two different audio groups in the HD-SDI signal, the group selection has to be done in two steps. If the HD-SDI de-embedding module in the MSD is mounted in input slot number 1, then the module is assigned to source number 33, 34, 35 and 36 in the input selection window in the audio matrix.



The HD-SDI Group Selection Window.

Step One: To assign a stereo pair from one of the four audio groups in the HD-SDI signal to source #33 and #34, use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the item called '**AUDIO GRP A**' and then press **[SELECT]**. In the group selection window use the arrow **[▲]** / **[▼]** keys to move the green selection bar to one of the four available audio groups and then press **[SELECT]** to assign the group or **[EXIT]** to abandon the changes.



The HD-SDI Channel Selection Window.

Step Two: To assign one of the two stereo pairs in the selected HD-SDI audio group use the arrow **[▲]** / **[▼]** keys to move the green selection bar to the item called '**GROUP A CH**' and then press **[SELECT]**. From the channel select window use the arrow **[▲]** / **[▼]** keys to select either '**CH 1 – 2**' or '**CH 3 – 4**'.

Use the same procedure for 'Audio Group B' to assign a stereo pair to audio source #35 and #36 in the audio matrix.

6.2.1.3.4 HD-SDI Input Status.

In the HD-SDI option window there are 6 status flags. The first three flags in the list are confirmation flags.

'**Audio Processing**' – If green, the HD-SDI signal contains a valid AES audio stream.
'**Valid Video Detected**' – If green, the HD-SDI signal contains data.
'**SDI Lock**' – If green, the HD-SDI signal is valid and the signal is locked.

The next three flags are warning flags.

'**AES Error**' – If red, the embedded AES audio stream can not be decoded correctly.
'**CRC Error**' – If red, the HD-SDI video stream is corrupted.
'**SDI Data Error**' – If red, the HD-SDI signal might not be connected.

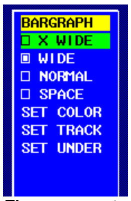
The warning flags in the HD-SDI option window are "sticky" flags. This means that the displayed warning flags are not dynamically updated. When an error occurs the corresponding warning flag is set, but it will only be cleared if the HD-SDI option window is closed or the **[CLEAR]** key is pressed.

6.3 Output Options. MATRIX ➔ EXTEND ➔ OPTION

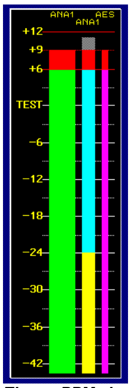
Depending on the type of physical outputs in the MSD there are in the 'Extended Matrix Window' different options available.

To access the output options from the 'Extended Matrix Window' use the **[◄] / [►]** keys to move the cursor to the column '**OUTPUT**' and then use the use the **[▲]** / **[▼]** keys to move the cursor to the desired destination and press the option key. If the option key is greyed out no options is available for that particular source.

6.3.1 Bargraph (PPM) Options. OPTION



The upper part of the bargraph options window.

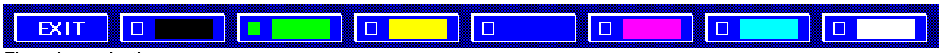


Three PPM bars with different widths and colours. The middle PPM bar has the 'Track' colour set to white.

To change the appearance of the individual 'Peak Programme Meter' bargraphs move the green selection cursor to a destination number between #65 and #96 column '`OUTPUT`' and press **[OPTION]**.

The bargraph option window has seven items. The '`X WIDE`', '`WIDE`' and '`NORMAL`' sets the width of the selected bargraph. 'Wide' is the default setting. The '`SPACE`' item will if checked, insert an extra space at the right side of the bargraph.

The colour of a bargraph can also be changed by using the colour selection menu. A bargraph is divided into three fields, the top part - 'overload', the bottom part - 'underload' and the middle part - 'normal'. The 'overload' part will always be red, but the colour of the 'underload' and 'normal' parts can be changed using '`SET UNDER`' and '`SET COLOUR`' items.



The colour selection menu.

'`SET TRACK`' will set the trace colour of the bargraph. When the trace colour is set to another colour than black, the background of the bargraph will show the highest bar reading until the the window is repainted or the **[CLEAR]** key in the **[METER]** menu is pressed.

6.3.2 Audio Vector Scope Options. OPTION

To change the settings for the 'Audio Vector Oscilloscope' and the Jelly-Fish™, move the green selection cursor to a destination number between #53 and #62 in the column '`OUTPUT`' and press **[OPTION]**.

Please note that the only option available the Jelly-Fish™ (destination #55 to #62) is the colour option. Please see section 6.3.2.3 for further information about this option.

6.3.2.1 Meter integration time. OPTION ➔ SLOW FAST

The integration time for the 'Phase Correlation Meter' can by selecting either **[SLOW]** or **[FAST]** be changed from 10mS. (slow) to no integration time (fast).

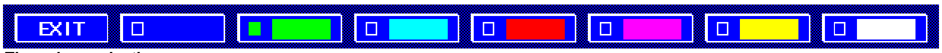
6.3.2.2 Meter Compressor. OPTION ➔ C-OFF GAIN ↑ 12 ↓

To obtain the best visual indication of signals in the 'Audio Vector Oscilloscope' the 'Audio Vector Oscilloscope' uses dynamic scaling. This is done by adjusting the input gain right before the metering function. The auto-adjustment are done so that the average inputlevel results in a indication that fills most of the window.
By selecting **[C-OFF]** the compressor is disabled and the input gain (and the size of the image) can be set manually by using the gain **[▲]** / **[▼]** keys.

Note for PT0660M users: If the compressor is off, the input gain for the 'Audio Vector Oscilloscope' can be adjusted using the wheeler. By pressing the **[#2]** key, the compressor can be enabled or disabled. Pressing the **[#1]** key will exit the options menu.

6.3.2.3 Scope Colour. OPTION ➔ SCOPCOL

The colour of the 'Audio Vector Oscilloscope' and the Jelly-Fish™ can be changed by entering the colour menu with the **[SCOPCOL]** key.



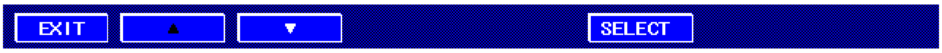
The colour selection menu.

6.4 Audio groups and Gain.

The physical outputs from the MSD can be assigned to groups. These groups can have different output gains. I.e. the digital group/outputs can be 6 dB higher then the analogue group/outputs. There are three different groups for the physical outputs: None, Master, Analogue OR Digital (depending on the type of physical output in the MSD).

Note: The analogue outputs can not be assigned to the digital group and vice versa.

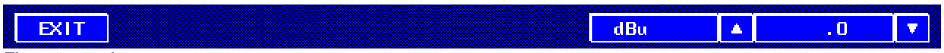
6.4.1 Assign Audio Group. OPTION



The upper part of the selection window for the audio groups. (Analogue outputs.)

To assign a physical output to a new Audio Group, move the cursor to the desired destination using the **[▲]** / **[▼]** keys. Now move the cursor to the group-column and the press the **[Option]** key. This will open the 'Group Selection' window. Use the **[▲]** / **[▼]** keys to select the group and then press **[Select]** to assign the new group or press **[Exit]** to cancel.

6.4.2.1 Adjust the Output Gain for a single output. OPTION



The output gain menu.

To adjust the output gain of a single output, use the **[◄] / [►]** keys to move the cursor to the column '**GAIN**' and then use the use the **[▲]** / **[▼]** keys to move the cursor to the desired output where the gain is to be adjusted. Press the **[Option]** key to enter the 'Output Gain' menu. In the 'Output Gain' menu use the **[▲]** / **[▼]** key to adjust the output gain and then press **[Exit]** when done.
The output gain can be adjusted in steps of 0,5dB between –12dB and +12,0dB.

6.4.2.2 Adjust Output Gain for an Audio Group. OPTION

To adjust the output gain of an Audio Group, use the **[◄] / [►]** keys to move the cursor to the '**G-GAIN**' column and then use the use the **[▲]** / **[▼]** keys to move the cursor to an output that is in the group where the gain is to be adjusted. Press the **[Option]** key to enter the 'Output Gain' menu. In the 'Output Gain' menu use the **[▲]** / **[▼]** key to adjust the output gain and then press **[Exit]** when done.

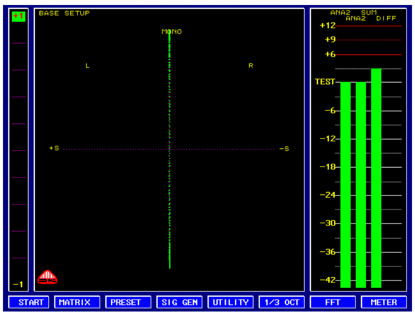
The output gain can be adjusted in steps of 0,5 dB between –12,0 dB and +12,0 dB.

Note for PT0660M users: Using the wheeler in the main menu in full feature mode will adjust the group gain for the Master group.

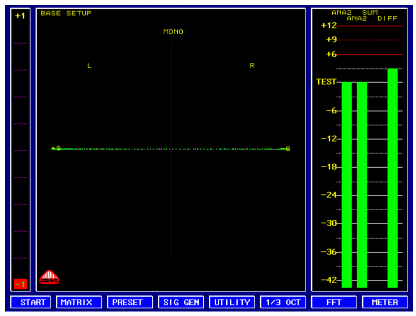
6.4.3 SUM and Difference amplifier. (M-S Mode.)

Even though most material today is produced in stereo, it is still important to have information about the mono compatibility of the stereo signal. There are different ways to measure this.

A way to do this, is to sum the two signals (left and right) together with a differential showing of the same signal pair. By showing these as PPM-bargraphs next to the stereo pair (M-S meters), it is possible to show the differential between the M (sum) and S (difference) and from that measure the mono compatibility. The further these two readings travel from each other, the likelihood is that the signal is going out of phase.



In the illustration above a 1 KHz sinewave with a phase difference of **0** degrees is applied to channel 1 and 2. When the signal is completely in phase the **SUM** of the signal will be doubled.

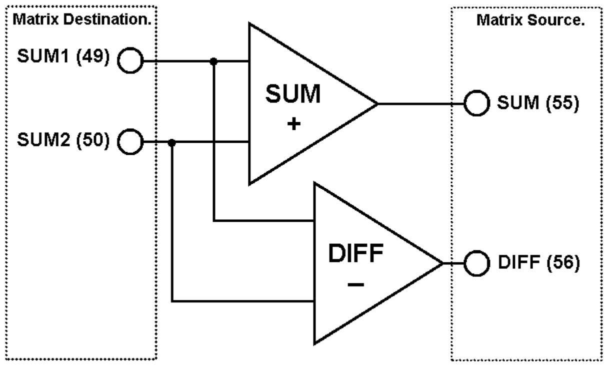


In the illustration above a 1 KHz sinewave with a phase difference of **180** degrees is applied to channel 1 and 2. When the signal is completely out of phase the **DIFFERENCE** of the signal will be doubled.

Please refer to section 7.1 and 7.2.1 for information on how to use the 'Phase Correlation Meter' and the 'Audio Vectorscope' to show the phase relationship between two channels.

6.4.3.1 Routing audio to the SUM and Difference amplifier.

Audio signals to and from the 'Sum and Difference Amplifier' is routed using the 'Audio Matrix'. The 'Sum and Difference Amplifier' has two inputs and two outputs. The inputs to the amplifier is found in the 'Matrix Destination List' as destination #49 ‘**SUM1**’ and #50 ‘**SUM2**.’ The outputs from the 'Sum and Difference Amplifier' is found in the 'Matrix Source List' as sources #55 ‘**SUM**’ and #56 ‘**DIFF**’.

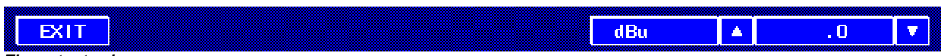


Equivalent schematic of the 'Sum and Difference Amplifier'.

6.4.3.2 Adjust output gain for the SUM and Difference amplifier.

As with any other output in the MSD, the output gain (amplification) for the 'Sum and Difference Amplifier' can be changed. As default, the SUM output from the amplifier will if the signals are of the same level and phase amplify with 3 dB.

To adjust the output gain for the 'Sum and Difference Amplifier', use the **[◄] / [►]** keys in the 'Extended Audio Matrix' to move the cursor to the ‘**G-GAIN**’ column, and then use the use the **[▲]** / **[▼]** keys to move the cursor to destination #49 ‘**SUM1**’. Press the **[Option]** key to enter the ‘Output Gain’ menu.



The output gain menu.

In the ‘Output Gain’ menu use the **[▲]** / **[▼]** keys to adjust the output gain and then press **[Exit]** when done. When done use the same procedure to adjust the output gain for destination #50 ‘**SUM2**’. The output gain for destination #49 and #50 must be set to the same level.

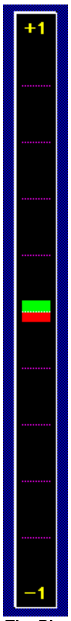
The output gain can be adjusted in steps of 0,5 dB between –6 dB and +6,0 dB.

If the amplification in the 'Sum and Difference Amplifier' should be lowered from +6dB to +3dB, the the ‘**G-GAIN**’ value for each destination should be set to -1,5 dB. The reason for this is that a +3dB amplification is 3dB lower than the standard +6dB and is therefore deducted from +6dB. Since the 3dB that has to be deducted has to be divided between two channels the result is -1,5dB for each channel.

7 The Main Metering Functions.

The Main Metering Functions consists of three different metering devices: The Phase Correlation Meter, The Audio Vector Scope / Jelly-Fish™ and the Peak Programme Meter (PPM).

7.1 The Phase Correlation Meter.



The Phase Correlation Meter in the left side of the main screen.

The ‘Phase Correlation Meter’ is of the so-called ‘Center-Zero’ type, and displays the phase relationship between two input signals. A stereo signal will ideally show **[0]** which indicates a random distributed phase, and therefore the maximum ambient effect. A mono signal will indicate a **[+1]**.

A signal with reversed phase components will indicate in the area between **[0]** and **[–1]** and when the indicator moves into the ‘non mono compatible’ area the colour will change to red for easy identification. Never allow negative indication if the signal is going to be reproduced in mono.

Only major phase components are considered so input signals below a predefined threshold will force the indicator towards zero.

If the indicator is showing zero it can have different meanings e.g. the signal is an ideal stereo signal, the signal is only in one of the two channels (left or right), or there might be no signal at all. The ‘Phase Correlation Meter’ is used in conjunction with the ‘Audio Vector Oscilloscope’ and the ‘Peak Programme Meter’.

Please refer to section 5 ‘The Compact Audio Matrix’ on how to select the two source signals for the ‘Phase Correlation Meter’.

7.2 Audio Vector or Surround Sound Monitoring (Jelly-Fish™)

The middle window in ‘Full Feature Mode’ has two functions. It can either be used as an ‘Audio Vector Oscilloscope’ or as the ‘Jelly-Fish™’ Surround Sound Monitor.

The type of display is depending on the settings in the ‘Audio Matrix’.

The destinations in the ‘Audio Matrix’ from #53 to #62 determines in which mode the middle window is used.

When a source is selected for destination #53 and #54 (PHAS) only, then the middle window will show the ‘Audio Vector Oscilloscope’. If sources are selected for #55 to #62 then the middle window will show the Jelly-Fish™ and then the sources for destination #53 and #54 (PHAS) will only affect the ‘Phase Correlation Meter’. This means that it is possible to monitor the phase relationship between two channels independent of the ‘Jelly-Fish™’. If destinations #53 and #54 (PHAS) are set to ‘OFF’ then the ‘Phase Correlation Meter’ and the ‘Audio Vector Oscilloscope’ / ‘Jelly-Fish™’ will be hidden.

Note that the Phase Correlation meter and Audio Vector Oscilloscope is sharing the same destinations in the Audio Matrix (#53 and #54).

Refer to section 5 ‘`The Compact Audio Matrix`’ for a detailed description on how to navigate the Audio Matrix.

Please note that it is a good practice to route the signals to the ‘Audio Vector Oscilloscope’ / ‘Jelly-Fish™’ through the ‘Peak Programme Meter’ (destination #65 to #96 / #74 in surround mode), this way when changing the input to a

PPM-Bargraph then the input to the ‘Audio Vector Oscilloscope’ / ‘Jelly-Fish™’ will change accordingly. Please see section B.1 'Factory Presets' for examples on how to configure the matrix.

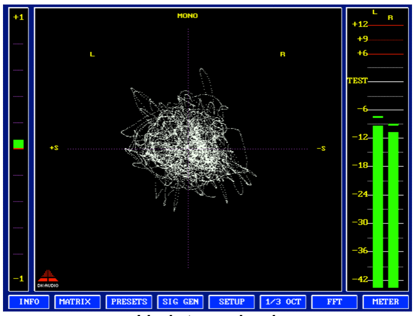
The ‘Audio Vector Oscilloscope’ is using dynamic auto adjustment (scaling) on the horizontal and vertical scales. Auto scaling enables the display window to ‘follow’ the average input signal level, resulting in a readout that fills most of the window most of the time. Because of this dynamic scaling it is important to notice that the ‘Audio Vector Oscilloscope’ can not be used to monitor the actual level of the signal but only the relative value.

This auto-adjustment feature (referred to as the ‘Meter Compressor’ or just ‘C-OFF’) can be enabled or disabled in the ‘Audio Matrix’. Please refer to section 6.3.2.2 ‘Meter Compressor.’.

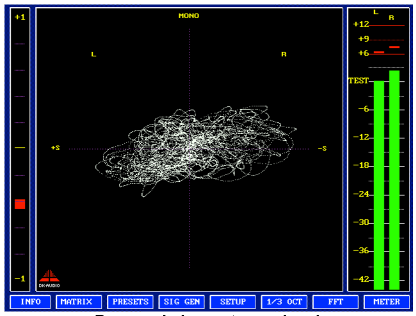
7.2.1 The Audio Vector Oscilloscope

The ‘Audio Vector Oscilloscope’ is also known under the names ‘Stereo Image Monitor’ or ‘Goniometer’. It is based on continuous graphic illustration of a stereo signal in the Lissajous-format.

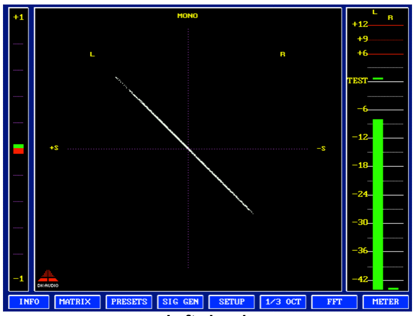
If phase and amplitude is randomly distributed, the signal is an ideal stereo signal. Normally this will only happen with a live recorded signal. Applause from a live-recorded audience is an excellent example of a true stereo signal. The figure of a true stereo signal should be represented on the ‘Audio Vector Oscilloscope’ as a perfect circle, or rather a ‘ball’ See the illustrations below on examples of different signal types as they appear on the ‘Audio Vector Oscilloscope’, and note the relationship between the representations on the ‘Phase Correlation Meter’ and the ‘Audio Vector Oscilloscope’.



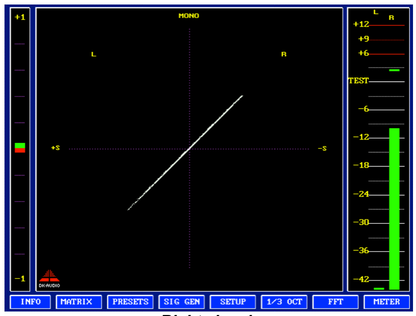
Ideal stereo signal.



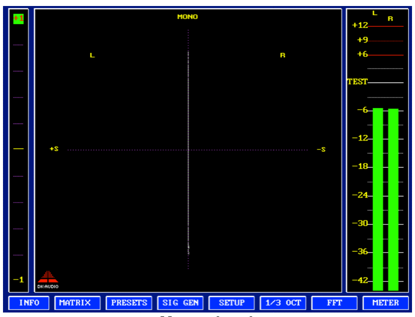
Reversed phase stereo signal.



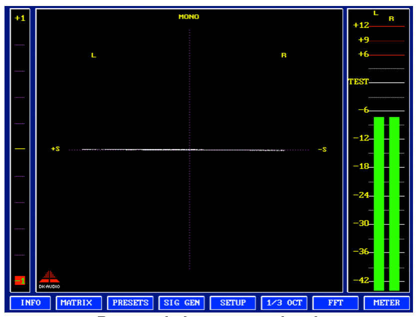
Left signal.



Right signal.



Mono signal.

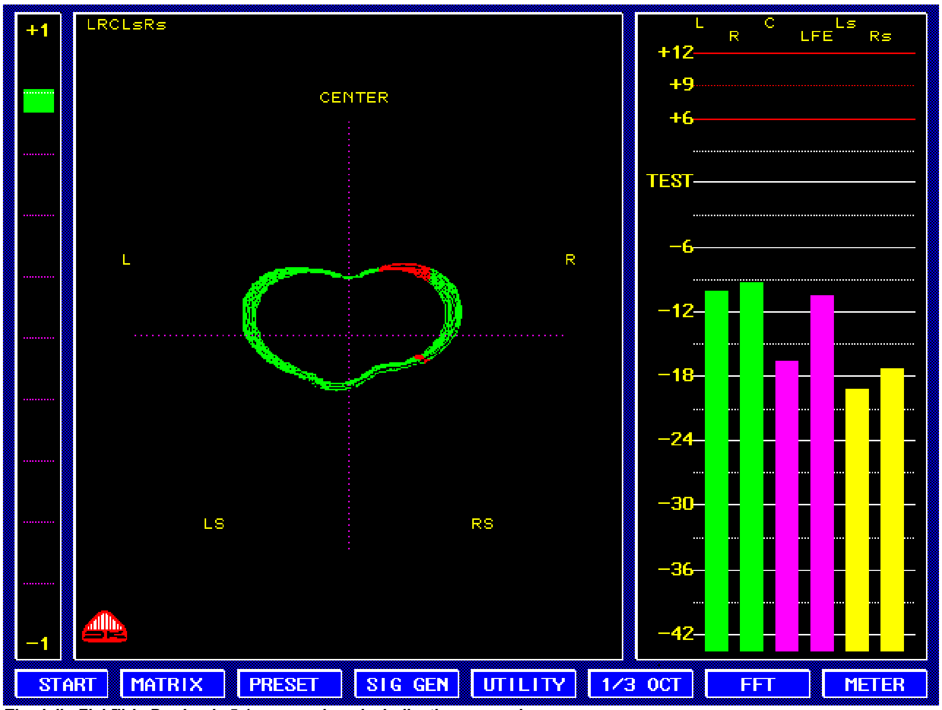


Reversed phase mono signal.

Please refer to section 5 ‘The Compact Audio Matrix’ on how to select the two source signals for the ‘Audio Vector Oscilloscope’. Destination #53 and #54.

7.2.3 The Jelly-Fish™ Surround Sound Monitor.

The MSD600M has a full Surround Sound monitoring function using the unique Jelly-Fish™ image.



The Jelly-Fish™ in Pro-Logic 5.1 surround mode, indicating some phase errors.

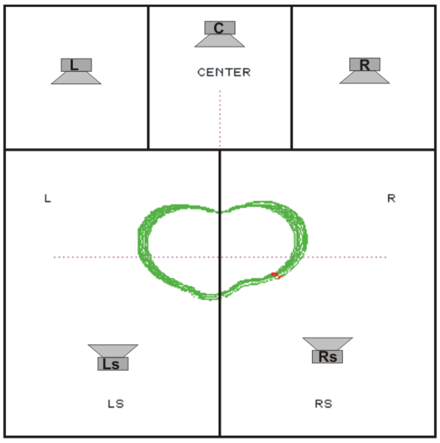
The Jelly-Fish™ figure represents the surround channels in a vector format. The Jelly-Fish™ can be used to monitor all standard surround sound formats up to 7.1. Including Pro-Logic, 5.1, 6.0, 7.1. In addition a pseudo-surround sound decoding mode of a stereo signal is also provided.

The Jelly-Fish™ monitor was designed to provide a fast and intuitive way of visualizing the surround sound field and its complex phase relationships.

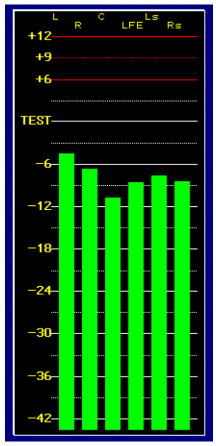
Depending on the format used (automatically detected based on the number of sources applied to the surround sound monitor), the MSD will select an appropriate 'background'-setting for the Jelly-Fish™.

The labels used in the 'background'-setting for the Jelly-Fish™ does not represent the placement of the speakers in the surround sound format, but are merely meant to indicate a specific direction in the surround sound field.

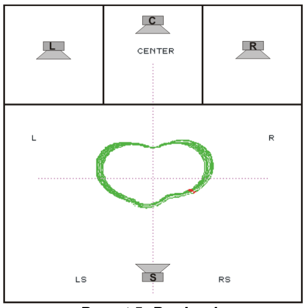
The following illustrations shows how the Physical Audio Inputs relates to assigned PPM-bargraphs and the Jelly-Fish™. The demonstrated format is taken from factory preset 6 'LRCLsRs' which is designed for monitoring a 5.1 surround sound setup.



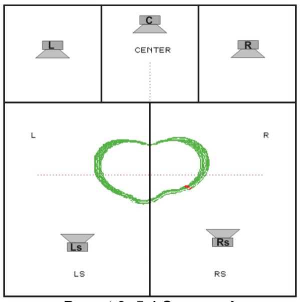
The Jelly-Fish™.
Each of the 5 areas illustrate a physical input source, labelled as the corresponding PPM Bargraph.



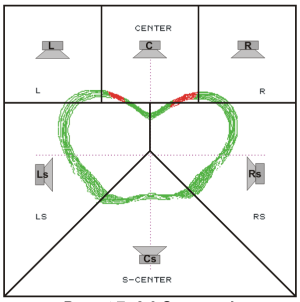
The Peak Program Meter.
Each Bargraph correspond to a Physical Audio source.



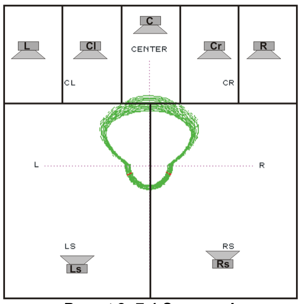
Preset 5: Pro Logic



Preset 6: 5.1 Surround.



Preset 7: 6.0 Surround.



Preset 8: 7.1 Surround.

To set the MSD for a given surround sound format, the specific Matrix destinations (#55 to #62) have to be assigned to a source . The number of destinations assigned determines which surround sound format that is selected.

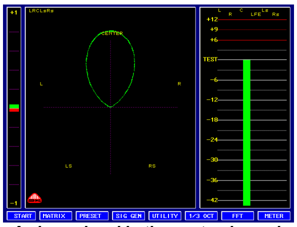
Preset 6 – Default 5.1 Surround Sound Setup.

Physical Audio Source		PPM-Bargraphs		Jelly-Fish™		
Destination Number	**Matrix Name**	**Destination Number**	**Matrix Name**	**Destination Number**	**Matrix Name**	**Display Name**
#37	C	#67	CH 3	#55	CENT	CENTER
#34	R	#66	CH 2	#56	C+1	R
#41	Rs	#70	CH 6	#57	C+2	RS
#38	Ls	#69	CH 5	#58	C+3	LS
#33	L	#65	CH 1	#59	C+4	L

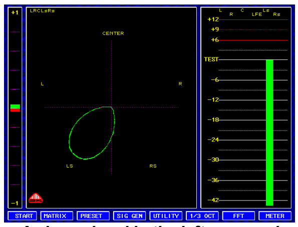
In the above example the Jelly-Fish™ destinations #60 to #62 are set to OFF.

When routing audio from the physical inputs to the Jelly-Fish™ it is necessary to route the audio through the PPM-Bargraphs. This means that in the above example the physical audio input #37 is routed to PPM-bargraph #3 which is destination #67. Since the PPM-bargraphs is routed back into the ‘Audio Matrix’ they can be used as sources for the Jelly-Fish™. In this case the PPM-bargraph #3 has the source #59. Source #59 is then routed to the Jelly-Fish™ center channel (destination #55). This might seem like a long way around a simple solution, but the advantage is that the Jelly-Fish™ surround sound monitor ‘follows’ the PPM-bargraphs so if a new source is needed for a particular PPM-bargraph then there is no need to select a new source for the Jelly-Fish™.

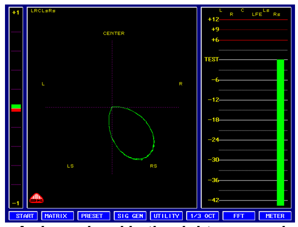
The audio signals assigned to the Jelly-Fish™ is assigned clockwise starting with the center channel.



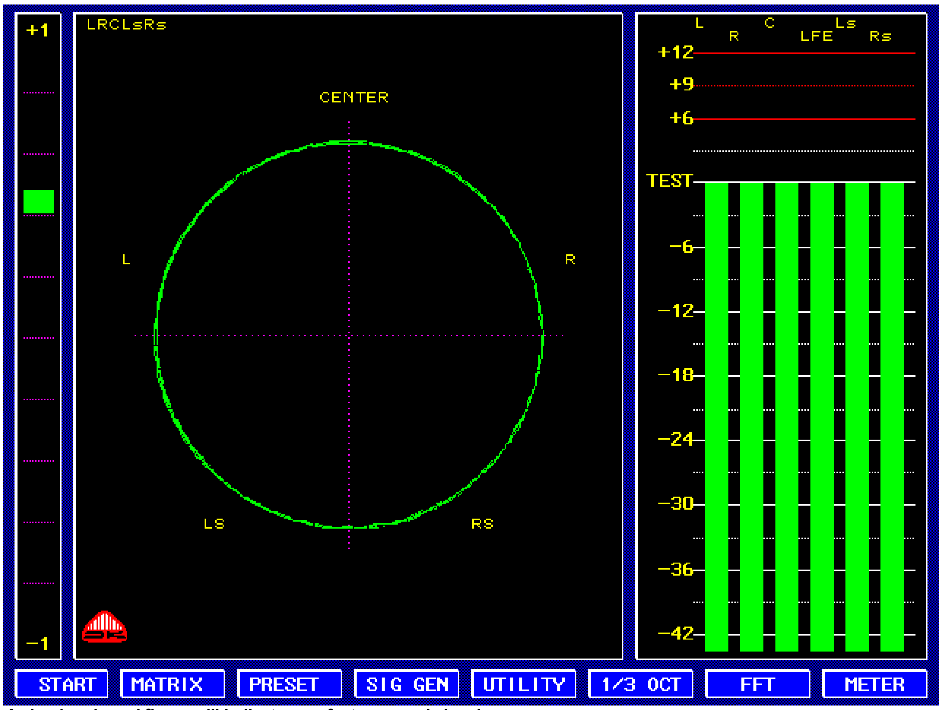
A sinus signal in the centerchannel.



A sinus signal in the left surround channel.



A sinus signal in the right surround channel.



A circular-shaped figure will indicate a perfect surround signal.

A complete comparison list for all the surround sound formats can be found in Appendix XX ‘Factory Presets’.

When monitoring a surround sound signal it is always recommended to use one of the factory presets(preset #5-9) as a template for MSD settings. In that way a problem free guide on how to "wire up" the MSD can be ensured when following the instructions found in Section X.X 'Factory presets'.

To monitor the complex phase relationships in a common surround sound signal the Jelly-Fish™ is identifying any phase problems accurately by a colour change to red in the relevant vector in the Jelly-Fish™ image. The width of the red colour spot indicates the degree of phase error.

To monitor the exact phase relation between any two signals in the Jelly-Fish™ these signals can be routed to the separate 'Phase Correlation Meter' which is destination #53 and #54.

When using a factory preset on a surround sound format using a separate Low Frequency Energy "LFE" channel (format 5.1 and 7.1) the MSD600M will include this LFE channel on the PPM-bargraph but since the LFE channel does not contain any frequency information that can be directional determined it is NOT included in the Jelly-Fish™ monitor.

7.3 The Peak Programme Meter (PPM). 

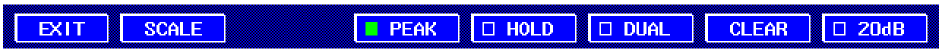
The 'Peak Programme Meter' is designed for direct measurement of the quasi-peak level of complex electrical signals ocuring in the transmission of music and speech, without varying the sensitivity of the device, to achieve optimum technical utilization of the transmission channel, or the recording medium.

For this purpose a full-wave rectifier is used, and the integration time is chosen to obtain an amplitude as high as possible, without overloading the transmission link for a period long enough to give rise to audible non-linear distortion of the programme. The return time is relatively long in order to avoid unnecessary viewer fatigue.

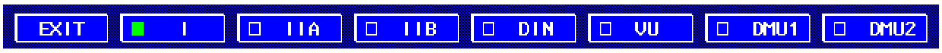
The 'Peak Programme Meter' is capable of showing up to 32 input channels simultaneously. The number of visible channels are configured in the Audio Matrix. When PPM-bars are enabled or disabled the Audio Vector Scope window is automatically scaled to fill the remaining part of the screen. See section 4 for information on how to route audio channels to the PPM bars. (Destinations #65 to #96.) Please observe that if the Jelly-Fish™ is active the maximum number of active PPM-bargraphs is 16.

Please refer to section 6.3.1 for information on how to change the appearance of the PPM bargraphs.

The configuration of the 'Peak Programme Meter' is done from the **[Meter]** menu.



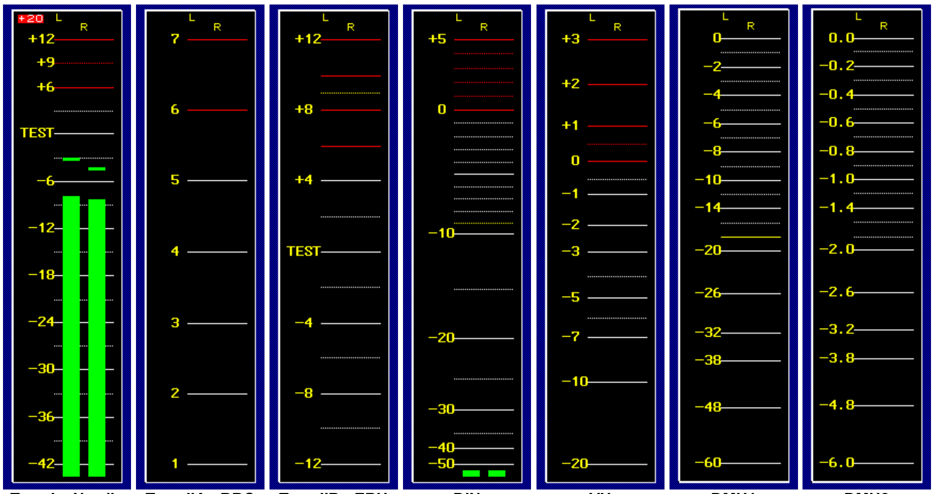
7.3.1 Scale selection. METER ➔ SCALE



The MSD has 7 different PPM scales installed and they are available from the submenu **[Scale]**.

All scales can be changed using the Windows™ programme DK-Scale. Please refer to the DK-Scale documentation for information on how to do this.

The installed scale package differs from region to region e.g. United States, Germany or International for the rest of the world. Below is the 7 scales from the international scalepackage that is installed as standard.



Type I – Nordic (+20 dB Active.) Type IIA - BBC Type IIB - EBU DIN VU DMU1 DMU2

7.3.2 PPM Peak. METER ➔ PEAK

The **[PEAK]** function activates the ‘flying’ peak indicators above each PPM-bargraph. Press the **[Peak]** key to toggle this function on or off. This function works closely with the **[HOLD]** function described below. The peak indicators will remain at their highest peak value for two seconds, then either fall back or go to a new highest level.

7.3.3 Dual Peak Indicators. METER ➔ DUAL

It is possible to show the PPM meter with two different ballistics for the same signal. While the selected scale with its standard PPM peak indicators will retain its original ballistics, another set of PPM peak indicators will show the true digital peak (regardless of the selected scale).
To enable the dual ballistics function use the **[DUAL]** key to toggle the function on or off.

The true digital peak indicators may be selected without the normal peak indicators, and this in fact may be advisable for most applications.

7.3.4 PPM Peak and Hold. METER ➔ ☐ HOLD CLEAR

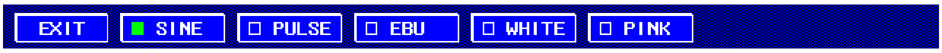
Using the **[Hold]** function it is possible to check the maximum signal level during, or especially after, a recording session. Thus you can secure that the levels does not exceed any pre-determined limits.

If **[PEAK]** or **[DUAL]** is enabled and **[HOLD]** is active then the peak and dual segments will remain at the highest reading until the **[CLEAR]** key is pressed.

7.3.5 Input Gain METER ➔ ☐ 20dB

The input sensitivity of the PPM meter can be increased by +20dB by selecting the **[20dB]** key. This function is used to obtain the best possible dynamic range and accuracy of measurement from weak signals. A red +20 indicator in the top left corner of the PPM meter window indicates that this function is ON. Please see the Type I scale on the previous page.

8 The Signal Generator. SIG GEN



The Tone Generator menu controls all signal generator functions. To access this menu press **[SIG GEN]** in 'Full Feature Mode'.

Please note that the level and frequency for Pulse, EBU, Pink and White Noise Test signals are set in the **[SINE]** menu.

The Test Signal Generator enables a low distortion signal to be fed into the Audio Matrix just like any other external audio signal. This means that the Signal Generator via the Audio Matrix can be set to output a test signal on any physical output available on a MSD model.

The Signal Generator shows up in the Input Matrix line number #53,54 (Tone) and can be treated just like any other source. In this way it is also possible to feed the output from the Signal Generator directly to the PPM Meter or to any of the other functions found in the MSD. For the MSD Models with no physical outputs it is still possible to use the Signal Generator as an internal test generator.

The active Test Signal is indicated by the highlighted green square.
To disable the signal generator on a destination, just select another source for that destination.

8.1 Sine Wave Test Tone Generator. SIG GEN ➔ SINE

Enter the **[SINE]** menu to adjust the frequency and the amplitude for any of the available test tones. Setting the frequency and amplitude is done by the arrow **[▲] / [▼]** keys. The frequency can be set in the range of 31Hz to 19952Hz in predefined steps. The amplitude adjustments (measured in dBu) are made in steps of +/-0.1dB, but adjustment speed can be increased by a factor ten (+/-1.0dB) by pressing the **[x10]** key. If active, the **[x10]** highlighted with a green square.

Note for PT0660M users: It is possible to adjust the frequency and output level using the wheeler. Pressing the wheeler determines which value to be adjusted. The **[#2]** key above the wheeler will toggle the **[x10]** option. The **[#1]** key above the wheeler will exit the sine wave setup menu.

8.2 Pulse Modulation Tone Generator. SIG GEN ➔ PULSE

The PULSE function changes the sine wave signal from the first channel (#53) from the Sine Wave Generator into a 'pulse' (chopped sine wave). The second channel (#54) from the Sine Wave Generator is not affected. The 'Pulse' and 'Pause' duration times is set in milliseconds using the arrow **[▲] / [▼]** keys. Single millisecond increment adjustments are made by 'clicking' these keys. For greater increments, keep the key pressed down.

The amplitude of the pulse signal is adjusted in the **[SINE]** menu. Please refer to section 8.1 for information on how to do this.

Note for PT0660M users: It is possible to adjust the pulse and pause timing using the wheeler. Pressing the wheeler determines which value to be adjusted. The **[#1]** key above the wheeler will exit the pulse setup menu.

8.3 EBU Test Signal. SIG GEN ➔ □ EBU

For radio, TV and video production the EBU test signal identifies the right channel of a stereo signal by switching the right signal off for 3 seconds, then on for 1 second. To enable this function in the MSD press the **[EBU]** key.

Please note that the source signals must be selected in the proper LEFT/RIGHT order in the Audio Matrix. Source #53 is the left channel and Source #54 is the right channel.

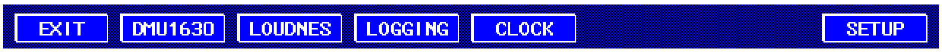
The amplitude and frequency of the EBU signal is adjusted in the **[SINE]** menu. Please see section XX.X on how to do this.

8.4 White & Pink Noise Test Signals. SIG GEN ➔ □ WHITE □ PINK

The Test Signal Generator is also able to generate both White and Pink Noise test signals.

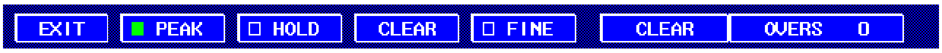
The amplitude of these signals is adjusted in the **[SINE]** menu. Please refer to section 8.1 for information on how to do this.

9 Utility Functions. UTILITY



From the **[UTILITY]** menu four special functions are available. DMU-1630 Digital Audio Peak Meter, The Graphic LEQM, The session Log and a DCF-77 Radio Controlled Clock. Please note that the utility functions only support up to 8 input channels.

9.1.1 DMU-1630 Digital Audio Peak Meter. UTILITY ➔ DMU1630



The MSD has included the DMU-1630 scale. The DMU-30 scale is a Digital True-Peak scale with a range of -60dB to 0dB.

Just like any other scale selected from the **[SCALE]** menu the DMU-1630 scale offers **[PEAK]**, **[DUAL]**, **[HOLD]** and **[RESET]**. The DMU-30's **[RESET]** function is equal to the **[CLEAR]** function used by any of the 7 internal scales in the **[METER]** menu. Please refer section 9.2, 9.3 and 9.4 for a description of these functions.

By pressing the **[FINE]** key the resolution of the DMU-1630 scale can be reduced by a factor of ten for accuracy. If selected, the scale now has a range of -6.0dB to 0.0dB. If disabled, a yellow line on DMU-1630 scale will indicate the full scale level for the **[FINE]** mode. Because of the high resolution LCD display used on the MSD, the actual scale resolution is VERY high. Far better than the original DMU-30's 0.1dB.

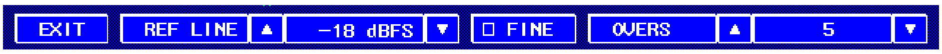
In **[FINE]** mode a yellow indicator can in either the top or bottom of the PPM window indicate if a channel is in overload or underload.

Please see section 11.2 for information on how to change the reference-level for the **[FINE]** mode.

The FINE mode differentiates from the standard DMU-2 scale by the capability of setting the Reference level. The standard DMU-2 scale reference level is fixed on digital full-scale.

The OVERS function will count the number of times the input samples reached 'Digital Full-Scale'. To Clear and restart this counter press the **[CLEAR]** key.

9.1.2 Setting Reference and Overs for the DMU-1630 Scale. UTILITY ➔ DMU1630



Setting the DMU-1630 reference-level and OVERS-Reference count is considered a critical setting and can therefore only be done when the MSD Write-Protection are disabled. Please see section 3.4 on how to disable the MSD Write-Protection.

When the MSD Write-Protection are disabled the **[DMU1630]** key in the **[UTILITY]** menu will now access the DMU-1630 Setup menu instead of the standard DMU-1630 menu.

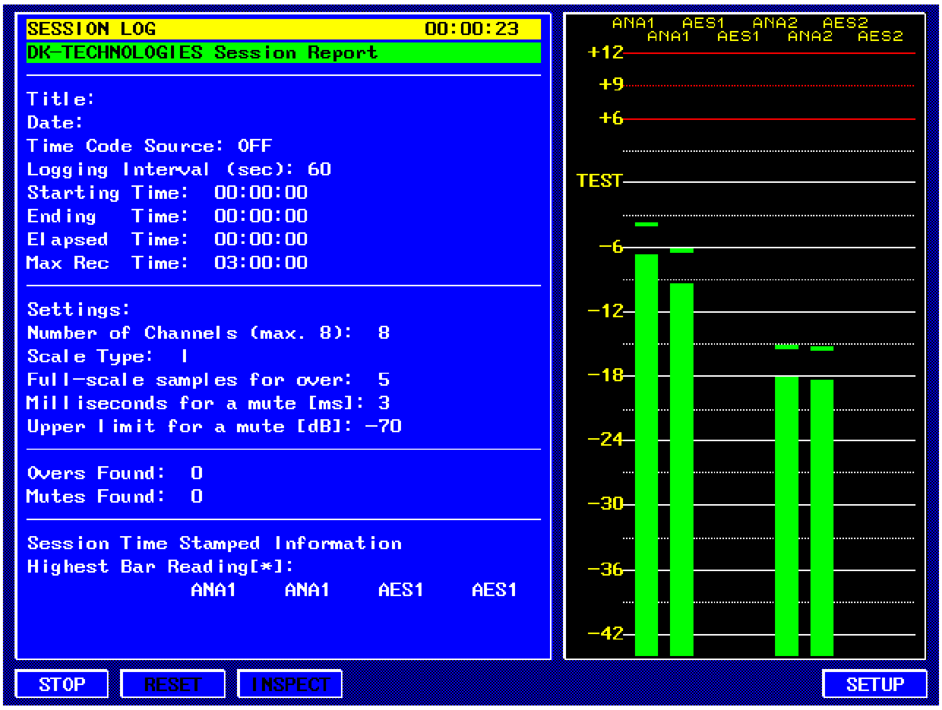
To set the DMU-1630 Reference level, indicated by a yellow line in the DMU-1630 PPM window, use the arrow **[▲] / [▼]** keys. The Reference level can be set in 1dB intervals in the range from –20dB to 0dB full scale. The original DMU-30 was limited to –20dB to –10dB in 2dB steps.
This Reference level is used to decide the top (full-scale) level in the **[FINE]** mode. In this way the **[FINE]** key is used to zoom a selected area of the PPM window. E.g. if the reference-level is set to –18dBFS then -6.0dB in the **[FINE]** mode corresponds to –24dBFS.

To set the OVERS Reference count use the OVERS-arrow **[▲] / [▼]** keys. This determines how many continuous digital full-scale samples that will trigger an OVER. In this way very short spikes hitting the digital full-scale can be filtered out, only leaving longer more damaging clipping to be registered. The OVERS can be set in the range from 1 to 15 samples.

Do not forget to enable the MSD Write-Protection after finishing the settings.

When leaving the DMU-1630 Menu the MSD will change the scale back to the one selected in the METER/SCALE Menu. To keep a DMU-1630 PPM scale select the DMU-1,DMU-2 scale from the SCALE Menu.

9.3 Session LOG. UTILITY ➔ LOGGING



The Session Log is capable of logging information about an audio signal. The sessionlog will at a predefined interval log the levels of up to 8 channels. These log entries will be timestamped either by an internal time code or by an external SMPTE time code signal applied to an audio input. Included in the Session Log is information about which scale is used, the number of PPM-Bargraphs that is monitored, If an overload or a mute has been detected this information will also be included in the Session Log.

9.3.1 Session Log Time Stamp setup.

When an entry is added to the Session Log, the entry is time stamped. The Time Code used is generated from either an internal free running time code starting from 00:00:00 when the **[RESET]** key is pressed or from an external SMPTE time code signal applied to an audio input on the MSD.

The source for the SMPTE time code reader determines which mode is selected (internal or external). The selected source for the SMPTE time code reader is listed by the ‘Time Code Source’ in the session log window.

When the source is set to ‘OFF’, the MSD will generate the internal free running time code otherwise the time code will be read from the external SMPTE signal applied to an audio input.

To select between the internal free running time code or the external SMPTE time code signal applied to an audio input, enter the Audio Matrix and select a source for the SMPT destination #52. If the selected source is not a valid SMPTE Time Code signal the time code will not run. The internal free running time code generator will be enabled if the source for the SMPT destination #52 is ‘OFF’ source #51 or #52.

9.3.2 Logging Interval. UTILITY → LOGGING → SETUP

The Logging Interval can be set in the range of 1 second to 240 seconds (4 minutes).

To set the 'Logging Interval' enter the setup menu by pressing the **[SETUP]** key. In the setup menu use the arrow **[▲] / [▼]** keys to move the green selection bar to the field called **'Logging Interval (sec)'** and press the **[SELECT]** key. The 'SECONDS' menu will now appear. From this menu use the arrow **[▲] / [▼]** keys to set the interval in seconds. Press **[EXIT]** when done.

Below the Logging Interval is listed relevant logging information including start time, end time, elapsed time and the maxavailable logging time. The maximum logging time is limited by the MSD's internal available data memory, the selected logging rate plus the number of channels selected. The longer time between each log entry the longer maximum logging time. For a two channel system with a logging time set to one minute (60 Seconds) the maximum logging time will be 12 hours equal to 720 log entries.

9.3.2 Trigger Events. UTILITY → LOGGING → SETUP

When the Session Log is adding entries to the log it always logs the highest PPM-Bargraph reading since the last log entry. In addition to that the 'Session Log' also checks for **mutes** and **overs** (digital full-scale samples).

For the mute trigger event it is possible to set a threshold for which the signal must be below before it can trigger an event. In addition to that the period in which the signal is below this threshold can be set. This way short breaks in music or in an interview can be filtered out, only letting longer muted periods of the source signal trigger the event. The mute threshold can be set in the range of -80 to –40 dB. The duration of a mute can be set from 1 to 20 mSec.
To set the threshold for the mute press the **[SETUP]** key and use the arrow **[▲] / [▼]** keys to move the green selection bar to the field called **'Upper limit for a mute [dB]'** and press the **[SELECT]** key. From this menu use the arrow **[▲] / [▼]** keys to set the interval in seconds. Press **[EXIT]** when done. To set the duration of the mute use the same procedure as for setting the threshold but select **'Milliseconds for a mute [ms]'** instead.

The trigger event for 'overs' can be configured to filter out short peaks in the signal by only triggering on longer peaks where the signal is 'clipping'. The number of samples that triggers an 'over' can be set from 1 to 15. If all 'overs' should be logged, just set the trigger event to 1. To set the number of full-scale samples that triggers 'overs', use the same procedure as for setting the threshold for a mute, but select **'Full-scale samples for over'** instead.

9.3.3 Create and Inspect a Session Log.

UTILITY → LOGGING → SETUP → STOP RESET

A Session Log is created simply by pressing the **[RESET]** key in the Session Log Menu. Pressing the **[STOP]** key will stop the logging and make it possible to inspect the Session Log Entries. As soon as the logging is started a time code is shown on the yellow bar in the top of the session log window. During the logging the highest reading will be marked with a **[*]**.

When the logging is stoped press the **[INSPECT]** key to enter inspect mode where it is possible to scroll the green selection bar through the log entries. When logging is stoped the total amount of 'mutes' and 'overs' will be calculated and the timecodes will be added to the end of the log.

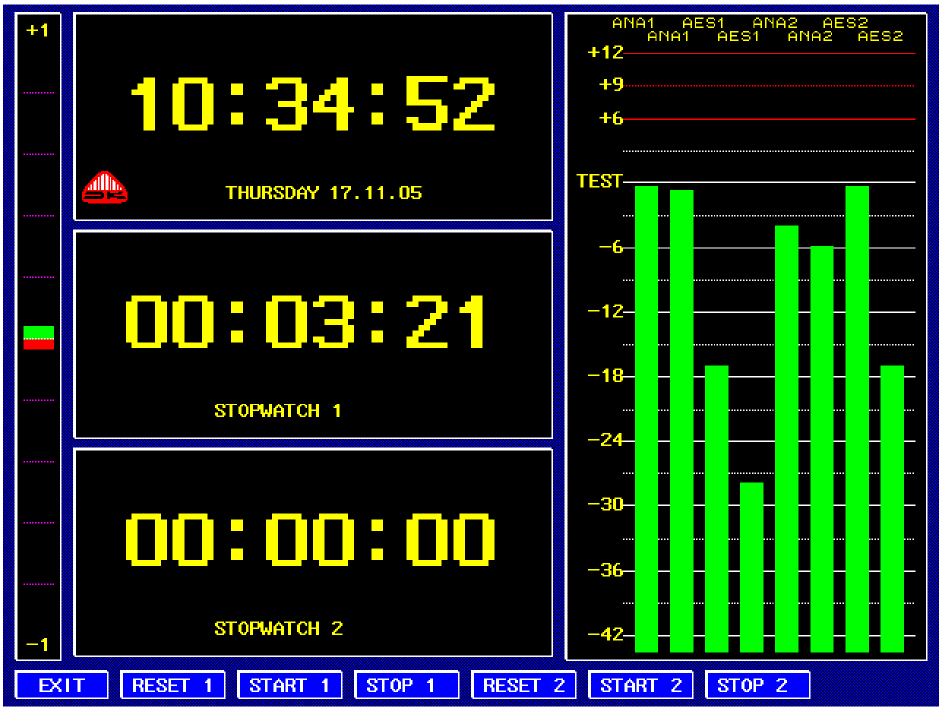
In the inspect mode it is possible to press the **[PRINT]** key. When this key is pressed the session log is transmitted on the RS232 port on the MSD. It is possible to connect a serial printer directly to this port. Another option is to use a terminal program on a computer.

The communication settings are as follows:

- **9600 Baud.**
- **8 Data Bits.**
- **No Parity.**
- **1 Stop Bit.**
- **No Handshake.**

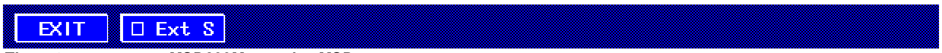
9.4 DCF-77 Real Time Clock. UTILITY ➔ CLOCK

If the MSD is fitted with a General Purpose I/O Module it is possible to use the MSD as a DCF-77 Real Time Clock. Please see appendix A.1.8 for further details about the General Purpose I/O Module. When the **[CLOCK]** function is enabled the MSD will show three small windows on the screen. If a DCF-77 Receiver is connected to pin 19 on the General Purpose I/O Module, the MSD is able to decode this signal and show the time and date on the screen in the topmost window. When the clock window is selected for the first time after a reboot the clock will count seconds from zero. Within two minutes the clock will be synchronized to the DCF-77 time signal. If the **[CLOCK]** menu is exited after synchronization the MSD will still show the correct time. The MSD can only be synchronized while the **[CLOCK]** menu is selected. The MSD has also included two stopwatches which is controlled using the **[START]**, **[STOP]** and **[RESET]** keys.



9.5 SETUP. UTILITY ➔ SETUP

The **[SETUP]** menu accesses advanced functions in the MSD. This menu will change according to the hardware.

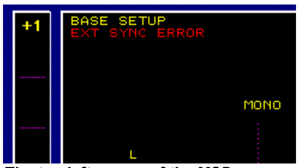


The setup menu on a MSD600M++ series MSD

9.5.1 External synchronization. UTILITY ➔ SETUP ➔ □ Ext S

The **[SETUP]** menu on the MSD600M++ calls a menu from where it is possible to select external synchronisation.

The MSD family has been designed to run on a fixed internal sample rate at 48kHz. However, in certain situations it is required to synchronise the MSD against an external source. On the MSD600M++ this source can be an AES signal connected to the dedicated External Sync Input located on the Utility Module. All other models supported by this manual it is possible to synchronise to any of the incoming AES signals. Selecting an external synchronisation source will force the MSD to follow the sample rate of the sync signal. It is therefore important to ensure that the external sync sources sample rate does not exceed 48kHz.

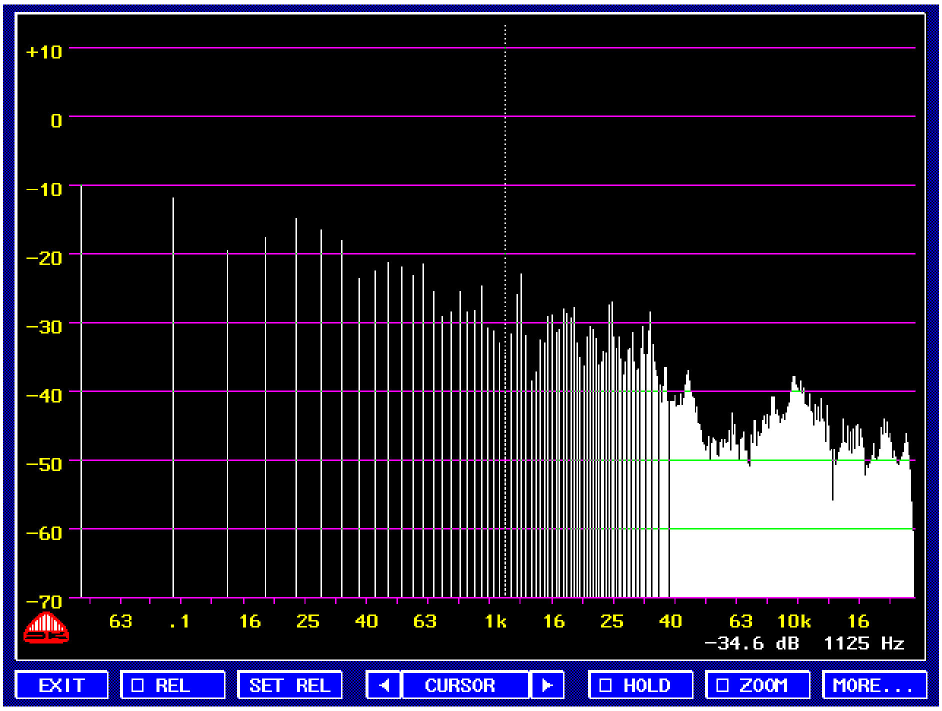


The top left corner of the MSD screen in full feature mode.

If the MSD is not able to synchronize to the selected signal the warning 'EXT SYNC ERROR' will be shown in the top left corner of the 'Audio Vector Scope' window. If the MSD shows this warning the MSD will not function correctly.

External Synchronization on the MSD600M++ is selected from the **[SETUP]** menu available from the **[UTILITY]** menu. On all other models it is selected from the AES option menu.

10 FFT Spectrum Analyser. FFT

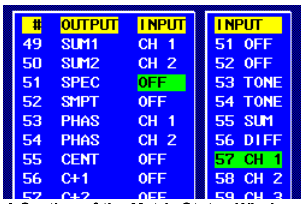


Most spectrum analysers used in sound engineering are based on analogue filtering techniques. For many years this has been the only practical solution for real-time analysers at a sensible size and cost. However, modern Digital Signal Processing (DSP) used by the MSD Family have made it possible to implement the complex FFT (Fast Fourier Transform) algorithm, which radically out performs its analogue counterpart many times over. For example, the number of frequency bands has been increased from the traditional 27 or 31, to a massive 512, and the dynamic range to 80dB. These improvements make it possible to analyse noise, distortion(IM) and frequency response in a far more detailed and accurate way than ever before.

To enter the FFT Spectrum Analyser mode, select **[FFT]** key from the Main Menu in Full Feature Mode or select the '**`FFT-Analyser`**' from the start menu.

Frequency is indicated at the bottom of the screen from 48Hz to 20kHz, with signal level indicated on the left hand side in 10dB intervals from -70dB to +10dB.

10.1 Routing Audio to the Analyser Applications.



A Section of the Matrix Status Window with CH 1 (PPM Bar #1) selected as source for the Analyser Applications.

Both Spectrum Analysers (1/3 Octave and FFT) available in the MSD shares the same destination in the Audio Matrix (#51 Spec). In that way when toggling between the 1/3 Octave and the FFT analyser it is not needed to set-up the Matrix more than once.

Please refer to section 5 'The Compact Audio Matrix' for information on how to route signals in the Audio Matrix.

10.2 Relative Offset Display. FFT → □ REL SET REL ◄ CURSOR ►

The Display Relative **[REL]** function works in conjunction with the Set relative **[SET REL]** function and the CURSOR Select function. When applying the display relative function, by pressing the **[REL]** key, the MSD tries to shift the whole FFT curve up or down relative to the **[SET REL]** level until the cursor level reading is equal to zero dB. The relative level is defined as the measured level at the cursor location at the exact time when pressing the **[SET REL]** Key. Use the **[◄] / [►]** keys to move the cursor to a specific bar of the FFT. The numeric frequency of the selected bar and the actual measured level is displayed in the bottom right corner of the FFT display.

The **[SET REL]** key will only manipulate the on-screen data and should generally not be used to raise a weak signal, since accuracy will be lost.

The resolution of the spectrum analyser is far higher than that of the screen resolution. Consequently, not all analysis results are displayed directly on screen. However, the frequency cursor can be used to obtain this 'hidden' data since it will display the exact frequency resolution in the numeric field display. Note that the numeric readout corresponds to the selected function. The numeric values always follow the actual data on the screen, i.e. if any of the functions REL and REF have been selected the numeric readout will reflect this.

10.3 Hold Maximum Level. FFT → □ HOLD

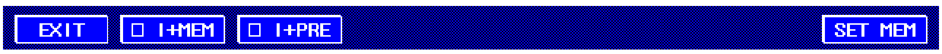
The FFT HOLD function will disable the curve fall-back function, and thus show the maximum value of the level within each frequency band since the **[HOLD]** key was activated.

10.4 FFT Zoom Function. FFT → □ ZOOM

The full frequency range of the spectrum analyser can be changed by a factor of ten. This means that the display will alter from showing a range of 46Hz to 23.531kHz, to showing only the lower frequency range of 4Hz to 2.353kHz. When activating this function by pressing the **[ZOOM]** key, a much more detailed look at the important low frequency signals is obtained.

10.5 Relative Memory FFT Measurements.

FFT → MORE... → OP-MODE



The ‘OP-Mode’ menu is used to display the spectrum analysis relative to a temporary or predefined reference curve. The relative display functions can all be active at the same time. To access the ‘OP-Mode’ menu from the ‘FFT-Main Menu’ press **[More…]** and then **[OP-MODE]**.

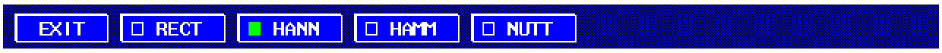
When measuring against any of the two reference curves the FFT result is displayed relative to this curve, frequency bar by frequency bar. In that way if measuring an Input signal relative to itself it will result in a flat response around 0 dB. (this is simply done by first pressing the **[SETMEM]** key follow by the **[I+MEM]** key.

The memory function is sensitive to the Write-Protection setting. If Write-Protection is on the **[SET MEM]** can only be used for temporary storage. The Analyser will loose the stored data each time the function is exited. This is very useful when making repeat measurements like the testing of room acoustics, loudspeakers etc. With Write-Protect Off the SETMEM function will save the reference curve in the MSD's FLASH memory saving the reference curve even after re-powering the unit.

The Predefined reference curve **[I+PRE]** is a pre-emphasis curve 'lifting' the data readout with +3dB/Octave, resulting in a visual appearance of a 1/3 Octave analyser. This lift provides a closer relationship between the aural and the visual judgement of the musical audio spectrum.

Both reference curves can be enabled at the same time.

10.6 FFT Window Functions. FFT ➔ MORE... ➔ WINDOW



Since the spectrum analyser is based on the Fast Fourier Transform principle, the sampled audio data needs 'windowing' (An in-depth mathematical explanation is not possible within the scope of this manual, we would therefore recommend that a detailed explanation of windowing be retrieved from other technical literature. However, a basic explanation of the windowing concept is as follows):

After the Audio is sampled and entered in a digital buffer with 1024 stages the data is processed by the FFT algorithm and finally the results are displayed. While FFT processing the 1024 samples, a new set of data are recorded. No data is lost. Since the 1024 samples used for these calculations are a window of continuous audio samples, the beginning and the end of the 1024 samples need some 'smoothing' in order not to develop erroneous results. This data smoothing is known as 'windowing'.

The MSD system is implemented with four different windowing functions listed below:

Rectangular Window. RECT
Samples are multiplied by a factor of one (i.e. by themselves). This window function is merely a method of disabling windowing of the data. The Rectangular Window provides the highest selectivity, but the lowest dynamic range.

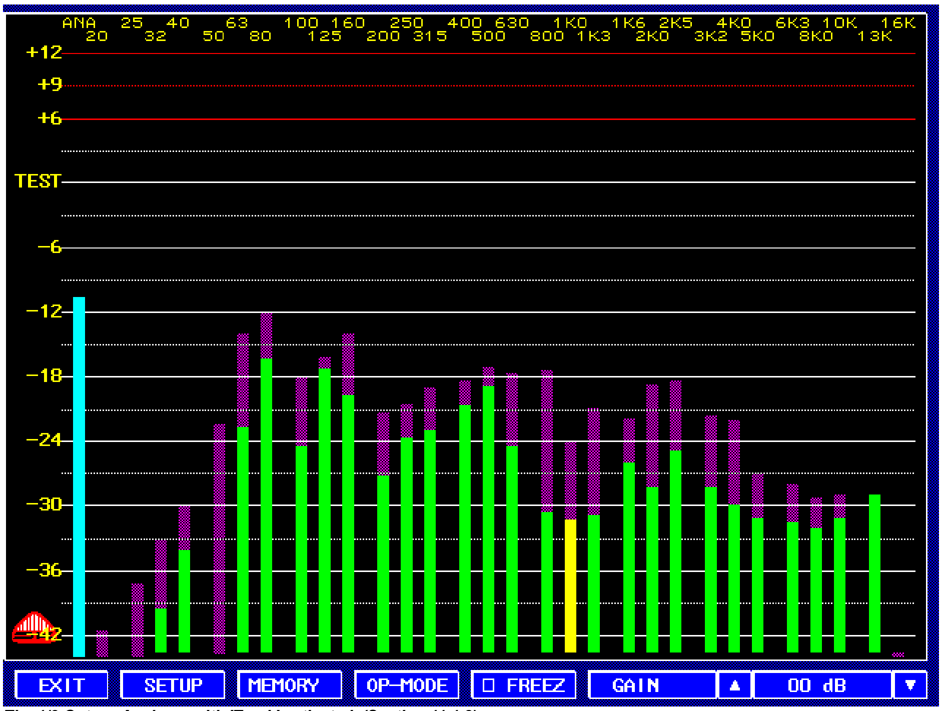
Hanning Window. HANN
The Hanning Window is useful in most applications and should be selected as the default. The Hanning Window provides high dynamic range, but low selectivity.

Hamming Window. HAMM
The Hamming Window is ideal for applications where the requirement for the dynamic range is known. The Hamming Window compromises between selectivity and dynamic range.

Nuttall Window. NUTT
The Nuttall Window is especially useful for distortion measurements. The Nuttall Window provides the highest dynamic range and compromises the selectivity.

The recommended window function for most applications is the HANNING window.

11 1/3 Octave Spectrum Analyser. 



The 1/3 Octave Analyser with 'Track' activated. (Section 11.1.2)

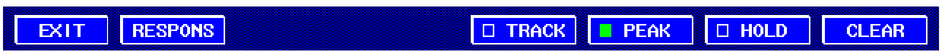
In the analysis of sound, it has always been important to describe the frequency content of a signal. Traditionally, this has been done using a filter bank containing a number of filters with relative bandwidths, for example 1/3 Octave. The signal being filtered passes through one filter at a time. Once the signal for a filter has been detected and its value determined, then the next filter is reviewed.

The left most bar on the 1/3 Octave Analyser (blue) is an input level indicator for the Analyser. Since the Input signal contains the energy from the entire 30 band combined, this bar will normally show a higher value than any of the separate 1/3 Octave bands.

To separate the individual bands easier, the 1kHz band has been coloured yellow.

Please refer to section 10.1 'Routing Audio to the Analyser Applications.' for details about selecting the source signal for the Analyser in the Audio Matrix.

11.1 Ballistic Setup for the 1/3 Octave Analyser. 

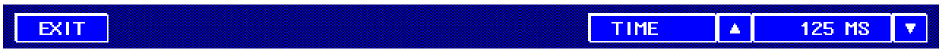


To obtain improved flexibility of the 30 band 1/3 Octave Analyser it is possible to adjust several of the Meter Ballistics. As well as the Meter Response time it is also possible to set the Meter Peak/Hold functions.

To alter the ballistics of the Analyser enter the **[SETUP]** Menu from the ‘1/3 Octave Base Menu’.

11.1.1 Response time for the 1/3 Octave Analyser.

FFT → SETUP → RESPONS



The analysers response time is set by the **[RESPONS]** key. Pressing this key opens a new menu from where it is possible to enter the response time from 125 milli seconds to 1000 milli seconds in steps of 125 milli seconds.

11.1.2 Measuring modes.

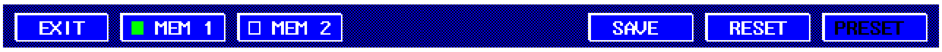
FFT → SETUP → TRACK PEAK HOLD CLEAR

To measure the maximum output level from any of the 30 bands over time, the **[TRACK]** function combines this feature together with the standard display mode. When engaged the maximum output from each band will now be displayed as a purple coloured bar behind the normal 'live' bar.

Combined with or separate from the TRACK function the Analyser also have a normal "flying" peak indicator with a very long release time (holds the peak indicator for approximate 2 seconds). The **[PEAK]** function can also be used in conjunction with the **[HOLD]** function enabling a classic Peak/Hold function. The displayed peak indicators can be cleared by pressing the **[CLEAR]** key. Both the **[HOLD]** and **[CLEAR]** functions can only be used when the **[PEAK]** function has been selected.

11.2 Measurement 'Snap-Shot's' of the 1/3 Octave Analyser.

FFT → SETUP → OP-MODE



Adding a Memory Function to the 1/3 Octave analyser greatly expands the usability of the analyser, giving the user the power of saving actual measurements and using these again as reference for new measurements. The **[MEMORY]** Menu is used in conjunction with the **[OP-MODE]** menu. (See next section).

To save a measurement, first select the memory number **[MEM 1]** or **[MEM 2]**. Then press the **[PRESET]** key to store the current measurement. The stored values are "SNAP-SHOTS" taken at the exact time the **[PRESET]** key has been hit.

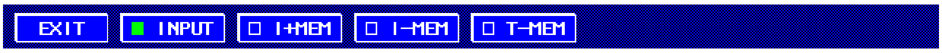
Just like a preset the two Memory locations are saved in the FLASH Memory. The memories save the content even after a power down. When entering the 1/3 Band Octave Analyser these two Memory "presets" are loaded into normal working memory of the MSD. This memory will however NOT be saved when leaving the Analyser. Since the **[SAVE]** key saves the measurement into this memory, any stored measurements done with this function will be lost when leaving the Analyser Menu.

As opposed to the **[SAVE]** key the **[PRESET]** key will save the snap-shot into the FLASH memory. The **[SAVE]** function will normally be used when the two memory locations already contain valuable data used by the operator, and the need of an additional temporary snap-shot is present. In such a case the **[SAVE]** key will temporary overwrite the Memory locations, but only until exiting the Analyser Menu. When entering the Analyser Menu again the FLASH saved memory snap-shots created by the **[PRESET]** key will once again be ready for use.

To erase **both** saved FLASH memory locations press the **[RESET]** key.

The Memory Snap-Shot currently used in the selected Operation Mode is the one highlighted in the MEMORY Menu.

11.3 Operation Modes for the 1/3 Octave Analyser. FFT → OP-MODE



The 1/3 Band Octave Analyser can be operated in four different modes. Beside analysing the Input Signal specified by the Audio Matrix, it is also possible to modify the measured Input signal in relation to the two saved Memory Snap-shots. The **[OP-MODE]** Menu is used in conjunction with the **[MEMORY]** menu. (See previous section).

To use the Analyser in Normal Mode displaying the analysed results from the INPUT signal press the **[INPUT]** key.

Selecting the **[I+MEM]** or **[I-MEM]** results in displaying the current measured Input signal Added (I+MEM) or subtracted (I-MEM) with the selected Memory Location.

To temporarily display the content of the selected Memory Location press the **[TMEM]** key. This key can be used to verify any stored measurements.

11.4 Using the Freeze Function. FFT → FREEZ

To momentarily freeze the Analyser display the **[FREEZ]** key can be selected. This function can be very useful when a non-continue Input signal have to be saved before further analysed, such as being modified with a measurement previously saved in one of the two memory locations.

Note: The freeze function is cleared when the Analyser Menu is exited.

11.5 Using the Gain Function. FFT → GAIN ▲ 00 dB ▼

To maximise the dynamic area of the Analyser the Input signal can be amplified up to +20dB using the **[GAIN]** arrow keys **[▲] / [▼]**. This function is mainly used for two different applications. One is to “normalize” a measurement in such a way that an actual measured average level for a specific band (or band’s) is set to match a predefined level on the Meter Scale. An example is to set the GAIN until the average level of a given band is around the 0dB mark on the scale. The Actual measured average value can now be determined by subtracting the entered Gain value from 0dB.
Another useful application for the Gain function is to be able to measure very low level signals. Typical the noise level of a system. By applying the total of +20dB Gain on the Input Signal it is possible to analyse signals levels down to -74dBu.

The scale used by the 1/3 Band Octave Analyser follows the one selected for the PPM Meter. Please see section x.x in this manual on how to change scale.

Please note that if using the VU and LEQ(m) scales which both have a very long integration time the 1/3 Octave Analyser will not work properly.

Appendix A Hardware Specifications.

A.1 MSD600M++ Base Unit.

PPM Analogue References:
Indication: 0 dBu
Input voltage: 1.55 V

PPM Scales:
Dynamic response:
Pflichtenheft 3/6: 3 ms / -3 dB
IEC 268-10: 5 ms / -2 dB
IEC 268-17: VU: 300 ms
Return (fallback) time:
Pflichtenheft 3/6: 20 dB / 1.5 s
IEC 268-10: 20 dB / 20 s
Division of scales:
Type I: -42 dB to +12 dB
Type IIA: +1 dB to +7 dB
Type IIB: -12 dB to +12 dB
Type DIN: -50 to +5 dB
Type VU: -20 dB to +3 dB
Type DMU-I: +60 dB to 0 dB
Type DMU-2: -6.0 dB to 0 dB

Phase Correlation Meter:
Indication range: +1 to –1

Audio Vectorscope:
Automatic gain offset range: 30 dB
Phase error between channels: none

LCD Display:
Resolution in dots: 640 x 480
Pixel size: 0.2 mm
Lifetime: 50.000 hours
Contrast ratio: 100:1
Viewing area: 135 x 100 mm
Luminance: 300 cd/m^2

Power Supply:
Supply voltage range: 12 V to 24 V DC
DC power consumption: approx. 18 W at 12 V DC nominal supply
Safety: according to IEC 65

Environmental Conditions:
Temperature range: 0 ^oC to 45 ^oC

Cabinet Dimensions:
MSD600M++, desktop version:
    Width: 186 mm plus mounting nuts 4 mm
    Height: 144 mm without mounting racket
    Depth: 50 mm without connectors

PT0600M, rackmount version.
    Width:
    Height:
    Depth:

A1.1 Utility Module w/ RS232 communication.

Type Number: MSD600M-Utility/1

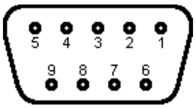
Description: Utility module with RS232 communication, VGA output and external synchronisation.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Connector type: 9 Pin Female D-Sub and 15 Pin Female High Density D-Sub.

Pin Configuration:

9 Pin D-Sub:

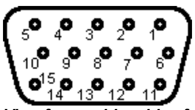


View from solder side of the 9 pin MALE D-Sub connector.

Signal Name.		Pin number.
Power +VCC	12-24 volt DC.	4
Power Ground		5*)
RS232	TX	2*)
RS232	RX	3*)
AES Synchronisation	Hot	8
	Cold	9
	Ground	1

*) *If connecting to a 9 pin serial port (D-Dub) then pin 2, 3 and 5 must be connected to the same pin numbers on the PC. (RX and TX should not be switched.)*

15 Pin D-Sub. (VGA)



View from solder side of the 15 pin MALE High Density D-Sub connector.

Signal Name		Pin number.
Red		1
Green		2
Blue		3
Ground		5
Ground		6
Ground		7
Ground		8
Ground		10
H-Sync		13
V-Sync		14

A.1.1 2 Channel Analogue Input Module.

Type Number: MSD600M-Input/1

Description: Input module with 2 analogue audio inputs and 1 AES-3 input.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Analogue inputs:
Maximum input level: +24 dBu.
Sample rate with internal sync: 48 kHz.
Sample rate range with external sync: 32 kHz to 50 kHz.
Bit resolution: 24 bits.
Frequency range within ±0.3 dB: 30 Hz to 20 kHz.
Passband ripple: ±0,002 dB.
Group delay: less than 0.82 msec.
Dynamic range, A-weighted: more than 103 dB.
Crosstalk at 1 kHz: less than –96 dB.
Signal-to-noise ratio: typical 93 dB.
Nominal input impedance: larger than 20 kohm.

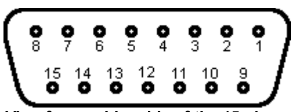
Digital input:
The digital input is equipped with a sample rate converter to synchronize the input to the internal clock. The sample rate converter may be by-passed. In this case the base unit should be synchronized externally by an AES-3 signal applied to the sync input on the utility connector.

Sample rate range: 30 Hz to 100 kHz.
Internal sample rate: 48 kHz.
Bit resolution: 24 bits.
Group delay: maximum 1.75 msec.
Passband ripple: ±0.008 dB.
Total harmonic distortion and noise: typical –103 dB at 1 kHz.
Dynamic range: larger than 120 dB.
Nominal input impedance: 110 ohm.

Possible placement: Priority C[*)], Input slot #1, #2, #3 and #4.
*) ***Modules with higher priority must be placed in input slots with a lower number, starting with input slot #1. Modules with the same priority can be interchanged.***

Connector type: 15 Pin Female D-Sub.

Pin Configuration:



View from solder side of the 15 pin MALE D-Sub connector.

Signal Name.		Pin number.
Analogue Channel 1	Hot	15
	Cold	8
	Ground	7
Analogue Channel 2	Hot	14
	Cold	6
	Ground	13
AES-3	Hot	9
	Cold	2
	Ground	1
	Ground	4
	Ground	10

A.1.2 8 Channel Analogue Input Module.

Type Number: MSD600M-Input-8A/O

Description: Input module with 8 analogue inputs.

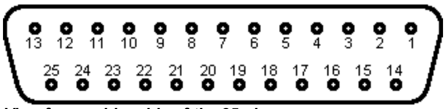
Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Analogue Inputs:
Maximum input level: +24 dBm
Sample rate with internal sync: 48 kHz
Sample rate range with external sync: 32 kHz to 50 kHz
Bit resolution: 24 bits
Frequency range within ±0.3 dB: 30 Hz to 20 kHz
Passband ripple: ±0,002 dB
Group delay: less than 0.82 msec
Dynamic range, A-weighted: more than 103 dB
Crosstalk at 1 kHz: less than –96 dB
Signal-to-noise ratio: typical 93 dB
Nominal input impedance: larger than 20 kohm

Possible placement: Priority A[*)], Input slot #1, #2, #3 and #4.
*) ***Modules with higher priority must be placed in input slots with a lower number, starting with input slot #1. Modules with the same priority can be interchanged.***

Connector type: 25 pin Female D-Sub

Pin Configuration:



View from solder side of the 25 pin MALE D-Sub connector.

Signal Name.		Pin number.
Analogue Channel 1	Hot	14
	Cold	2
	Ground	1
Analogue Channel 2	Hot	3
	Cold	16
	Ground	15
Analogue Channel 3	Hot	20
	Cold	8
	Ground	7
Analogue Channel 4	Hot	9
	Cold	22
	Ground	21
Analogue Channel 5	Hot	17
	Cold	5
	Ground	4
Analogue Channel 6	Hot	6
	Cold	19
	Ground	18
Analogue Channel 7	Hot	23
	Cold	11
	Ground	10
Analogue Channel 8	Hot	12
	Cold	25
	Ground	24

A.1.3 2 Channel Digital Input Module.

Type Number: MSD600M-Input-2D/O

Description: Input module with 2 digital inputs.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

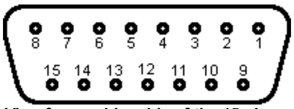
Digital Inputs:
Dual AES-3 Input Module. The inputs are equipped with sample rate converters to synchronize the inputs to the internal clock. The sample rate converters may be by-passed. In this case the base unit should be synchronized externally by an AES-3 signal applied to the sync input on the utility connector.

Sample rate range: 30 Hz to 100 kHz
Internal sample rate: 48 kHz
Bit resolution: 24 bits
Group delay: maximum 1.75 msec
Passband ripple: ±0.008 dB
Total harmonic distortion and noise: typical –103 dB at 1 kHz
Dynamic range: larger than 120 dB
Nominal input impedance: 110 ohm

Possible placement: Priority B*), Input slot #1, #2, #3 and #4.
*) *Modules with higher priority must be placed in input slots with a lower number, starting with input slot #1.*
Modules with the same priority can be interchanged.

Connector type: 25 pin Female D-Sub

Pin Configuration:



View from solder side of the 15 pin MALE D-Sub connector.

Signal Name.		Pin number.
AES-3 Channel 1	Hot	2
	Cold	9
	Ground	1
AES-3 Channel 2	Hot	11
	Cold	3
	Ground	10
	Ground	4
	Ground	12

A.1.4 4 Channel Digital Input Module.

Type Number: MSD600M-Input-4D/O

Description: Input module with 4 digital inputs.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

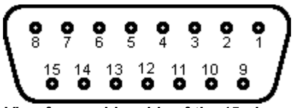
Digital Inputs:
Quad AES-3 Input Module. The inputs are equipped with sample rate converters to synchronize the inputs to the internal clock. The sample rate converters may be by-passed. In this case the base unit should be synchronized externally by an AES-3 signal applied to the sync input on the utility connector.

Sample rate range: 30 Hz to 100 kHz
Internal sample rate: 48 kHz
Bit resolution: 16 bits
Group delay: maximum 1.75 msec
Passband ripple: ±0.008 dB
Total harmonic distortion and noise: typical –103 dB at 1 kHz
Dynamic range: larger than 120 dB
Nominal input impedance: 110 ohm

Possible placement: Priority A*), Input slot #1, #2, #3 and #4.
*) ***Modules with higher priority must be placed in input slots with a lower number, starting with input slot #1. Modules with the same priority can be interchanged.***

Connector type: 25 pin Female D-Sub

Pin Configuration:



View from solder side of the 15 pin MALE D-Sub connector.

Signal Name.		Pin number.
AES-3 Channel 1	Hot	2
	Cold	9
	Ground	1
AES-3 Channel 2	Hot	11
	Cold	3
	Ground	10
AES-3 Channel 3	Hot	15
	Cold	8
	Ground	7
AES-3 Channel 4	Hot	14
	Cold	6
	Ground	13
	Ground	4
	Ground	12

A.1.5 SD-SDI Deembedding module.

Type Number: MSD600M-Input-SDI/4

Description: 4-Channel SDI Input Module

Supported by: MSD600M++, PT0600M, PT0660M, PT0660M-LS

Digital Inputs:
4-Channel SDI Input Module. SDI video input with re-clocked loop-through output. The SDI module de-embeds one of the four audio groups which contain 4 audio channels.

Input Format: 270 Mbps SDI component video. Complies with CCIR656 and SMPTE 259M.

Return loss: larger than 25 dB from 1 to 270 MHz
Deembedding delay: 312 µsec corr. to 26 audio samples.

Nominal input impedance: 75 ohm with re-clocked loop-through.

Possible placement: Priority B[*1)], Input slot #1, #2, #3 and #4[*2)].

[*)] ***Modules with higher priority must be placed in input slots with a lower number, starting with input slot #1. Modules with the same priority can be interchanged.***

[2)] ***It is only possible to insert TWO SD-SDI modules in a baseunit.***

Connector Type: 2 X BNC

A.1.6 HD-SDI Deembedding module.

A.1.7 2 Channel Analogue Output module.

Type Number: MSD600M-Output/1

Description: Output module with 2 analogue audio output and 1 AES3 output.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

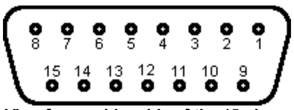
Analogue Outputs:

Maximum output level: more than +18 dBm at 600 ohm
Sample rate with internal sync: 48 kHz
Sample rate range with external sync: 32 kHz to 50 kHz
Bit resolution: 24 bits
Frequency range within ±0.3 dB: 30 Hz to 20 kHz
Passband ripple: ±0,007 dB
Group delay: less than 0.21 msec
Dynamic range, A-weighted: more than 101 dB
Crosstalk at 1 kHz: less than –96 dB
Signal-to-noise ratio: typical 93 dB
Nominal output impedance: less than 5 ohm

Possible placement: Priority A*), Input slot #1, #2, #3 and #4.
*) ***Modules with higher priority must be placed in output slots with a lower number, starting with output slot #1. Modules with the same priority can be interchanged.***

Connector type: 15 Pin Female D-Sub.

Pin Configuration:



View from solder side of the 15 pin MALE D-Sub connector.

Signal Name.		Pin number.
Analogue Channel 1	Hot	15
	Cold	8
	Ground	7
Analogue Channel 2	Hot	14
	Cold	6
	Ground	13
AES-3	Hot	9
	Cold	2
	Ground	1
	Ground	4
	Ground	10

A.1.8 Channel General Purpose I/O Module.

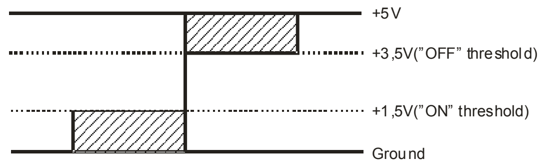
Type Number: MSD600M-I/O-23

Description: General purpose module with 23 I/O channels for remote control.

Supported by: MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

I/O connections:

The General Purpose I/O Module provide an easy way of interfacing with the MSD for remote control and other purposes. The I/O pins are TTL compatible with a strong pull-up and input protection resistors.

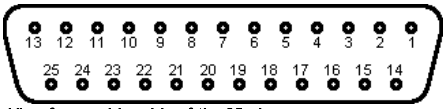


TTL Voltage Levels.

Possible placement: Output slot #4*).
*) *Only one General Purpose I/O module can be fitted in a MSD.*

Connector type: 25 Pin Female D-Sub.

Pin Configuration:



View from solder side of the 25 pin MALE D-Sub connector.

Function.	Direction.	Pin number.
Recall Preset #1.	Input.	13
Recall Preset #2.	Input.	25
Recall Preset #3.	Input.	12
Recall Preset #4.	Input.	24
Recall Preset #5.	Input.	10
Recall Preset #6.	Input.	23
Recall Preset #7.	Input.	11
DMU-1630 Fine. *(See chapter 9.1.1)*	Input.	20
Softkey #1.	Input.	2
Softkey #2.	Input.	15
Softkey #3.	Input.	3
Softkey #4.	Input.	16
Softkey #5.	Input.	4
Softkey #6.	Input.	17
Softkey #7.	Input.	5
Softkey #8.	Input.	18
Unused.		22
Unused.		9
Unused.		21
Unused.		8
DCF-77 Input. *(See chapter 9.4)*	Input.	19
Unused.		6
Unused.		7
Power +5V (Max. 100 mA.)	Output	1
Ground		14

B Factory Presets.

B.1.1 MSD600M Preset #1 - Base Setup.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: The 'Base Setup' is the default preset the MSD loads when it is booting. Please refer to section 4.2.2 for information on how to select another default startup preset. Preset #1 is write protected and requires a special procedure to be changed. Please refer to section 4.2.1 for information on how to do this.

The base setup is configured to show all information from the input modules in slot #1 and #2. The Audio Vector Scope and Phasemeter monitors the PPM Channel 1 and 2.

Depending on the type of installed modules this preset can vary from MSD to MSD.

If the MSD is fitted with two or more analogue input modules then the preset is configured as the following table shows.

Preset #1 - Base Setup.

	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	51	OFF
	56	C + 1	51	OFF
	57	C + 2	51	OFF
	58	C + 3	51	OFF
	59	C + 4	51	OFF
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	ANA1
	66	CH 2	34	ANA2
	67	CH 3	35	AES1
	68	CH 4	36	AES2
	69	CH 5	37	ANA3
	70	CH 6	38	ANA4
	71	CH 7	39	AES3
	72	CH 8	40	AES4

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.2 MSD600M Preset #2 – All Analogue.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: The 'All ANA' preset is based on all the analogue inputs in the MSD. The Audio Vector Scope and Phasemeter monitors the PPM Channel 1 and 2.

Depending on the type of installed modules this preset can vary from MSD to MSD.

If the MSD is fitted with '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #2 – All Analogue.				
	Destination Number.	**Destination Name.**	**Source Number.**	**Source Name.**
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	51	OFF
	56	C + 1	51	OFF
	57	C + 2	51	OFF
	58	C + 3	51	OFF
	59	C + 4	51	OFF
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	ANA1
	66	CH 2	34	ANA2
	67	CH 3	37	ANA3
	68	CH 4	38	ANA4
	69	CH 5	41	ANA5
	70	CH 6	42	ANA6
	71	CH 7	45	ANA7
	72	CH 8	46	ANA8

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.3 MSD600M Preset #3 – All Digital.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: The 'All DIGITAL' preset is based on the digital inputs in the MSD. The Audio Vector Scope and Phasemeter monitors the PPM Channel 1 and 2.

Depending on the type of installed modules this preset can vary from MSD to MSD.

If the MSD is fitted with '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #3 – All Digital.				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	51	OFF
	56	C + 1	51	OFF
	57	C + 2	51	OFF
	58	C + 3	51	OFF
	59	C + 4	51	OFF
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	35	AES1
	66	CH 2	36	AES2
	67	CH 3	39	AES3
	68	CH 4	40	AES4
	69	CH 5	43	AES5
	70	CH 6	44	AES6
	71	CH 7	47	AES7
	72	CH 8	48	AES8

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.4 MSD600M Preset #4 – 'LR' Stereo Preset.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: The 'LR' preset is based on the first two analogue inputs in the MSD. The Audio Vector Scope and Phasemeter monitors the PPM Channel 1 and 2.

Depending on the type of installed modules this preset can vary from MSD to MSD.

If the MSD is fitted with '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #4 – LR				
	Destination Number.	**Destination Name.**	**Source Number.**	**Source Name.**
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	51	OFF
	56	C + 1	51	OFF
	57	C + 2	51	OFF
	58	C + 3	51	OFF
	59	C + 4	51	OFF
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	ANA1
	66	CH 2	34	ANA2
	67	CH 3	52	OFF
	68	CH 4	51	OFF
	69	CH 5	51	OFF
	70	CH 6	51	OFF
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.5 MSD600M Preset #5 – 'LRCS' Pro-Logic Surround Sound.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'LRCS' preset the Jelly-Fish™ monitors a four channel surround sound signal applied as discrete channels to the first two analogue input modules. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses five vectors in the Jelly-Fish™. The centerchannel are coupled to the 'CENT' vector (#55). The surround channel are coupled to both the 'C+2' and 'C+3' vectors. (#57 and #58).

Depending on the type of installed modules this preset can vary from MSD to MSD.

If the MSD is fitted with '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #5 – LRCS				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	59	CH 3
	56	C + 1	58	CH 2
	57	C + 2	60	CH 4
	58	C + 3	60	CH 4
	59	C + 4	57	CH 1
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	L
	66	CH 2	34	R
	67	CH 3	37	C
	68	CH 4	38	S
	69	CH 5	51	OFF
	70	CH 6	51	OFF
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.6 MSD600M Preset #6 – 'LRCLsRs' 5.1 Surround Sound. (Analogue)

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'LRCLsRs' preset the Jelly-Fish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses five vectors in the Jelly-Fish™.

The vectors in the Jelly-Fish™ are assigned clockwise in the order C, R, Rs, Ls, L. Since the LFE channel does not contain any directional information, this channel is not assigned to the Jelly-Fish™.

If digital sources has to be monitored, use factory preset #9 where the sources for the destinations #65 to #70 are changed.

If the MSD is fitted with three '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #6 – LRCLsRs

	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	59	CH 3
	56	C + 1	58	CH 2
	57	C + 2	62	CH 6
	58	C + 3	61	CH 5
	59	C + 4	57	CH 1
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	L
	66	CH 2	34	R
	67	CH 3	37	C
	68	CH 4	38	LFE
	69	CH 5	41	Ls
	70	CH 6	42	Rs
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.7 MSD600M Preset #7 – 'LRLsRsCCs' 6.0 Surround Sound.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'LRCLsRs' preset the Jelly-Fish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses six vectors in the Jelly-Fish™.

The vectors in the Jelly-Fish™ are assigned clockwise in the order C, R, Rs, Cs, Ls, L.

If digital sources has to be monitored, the sources for destinations #65 to #70 has to be changed.

If the MSD is fitted with three '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #7 – LRLsRsCCs				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	61	CH 5
	56	C + 1	58	CH 2
	57	C + 2	60	CH 4
	58	C + 3	62	CH 6
	59	C + 4	59	CH 3
	60	C + 5	57	CH 1
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	L
	66	CH 2	34	R
	67	CH 3	37	Ls
	68	CH 4	38	Rs
	69	CH 5	41	C
	70	CH 6	42	Cs
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.8 MSD600M Preset #8 – 'LRClCrLsRs' 7.1 Surround Sound.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'LRClCrLsRs' preset the Jelly-Fish™ monitors a 7.1 surround sound signal applied as discrete channels to the four analogue input modules. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses seven vectors in the Jelly-Fish™.

The vectors in the Jelly-Fish™ are assigned clockwise in the order C, Cr, R, Rs, Ls, L, Cl. Since the LFE channel does not contain any directional information, this channel is not assigned to the Jelly-Fish™.

If digital sources has to be monitored, the sources for the destinations #65 to #72 has to be changed.

If the MSD is fitted with four '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #6 – LRCLsRs				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	63	CH 7
	56	C + 1	60	CH 4
	57	C + 2	58	CH 2
	58	C + 3	62	CH 6
	59	C + 4	61	CH 5
	60	C + 5	57	CH 1
	61	C + 6	59	CH 3
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	L
	66	CH 2	34	R
	67	CH 3	37	Cl
	68	CH 4	38	Cr
	69	CH 5	41	Ls
	70	CH 6	42	Rs
	71	CH 7	45	C
	72	CH 8	46	LFE

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.9 MSD600M Preset #9 – 'LRCLsRs' 5.1 Surround Sound. (Digital)

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'LRCLsRs' preset the Jelly-Fish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses five vectors in the Jelly-Fish™.

The vectors in the Jelly-Fish™ are assigned clockwise in the order C, R, Rs, Ls, L. Since the LFE channel does not contain any directional information, this channel is not assigned to the Jelly-Fish™.

If analogue sources has to be monitored, use factory preset #6 where the sources for the destinations #65 to #70 are changed.

If the MSD is fitted with three '2 Channel Analogue Input Modules' then the preset is configured as the following table shows.

Preset #9 – LRCLsRs				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	59	CH 3
	56	C + 1	58	CH 2
	57	C + 2	62	CH 6
	58	C + 3	61	CH 5
	59	C + 4	57	CH 1
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	35	L
	66	CH 2	36	R
	67	CH 3	39	C
	68	CH 4	40	LFE
	69	CH 5	43	Ls
	70	CH 6	44	Rs
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

B.1.10 MSD600M Preset #10 – 'Lt Rt' Pseudo-Surround Sound.

Applies to : MSD600M, MSD600M++, PT0600M, PT0660M, PT0660M-LS

Description: In the 'Lt Rt' preset the Jelly-Fish™ monitors a stereo signal applied to the first analogue input module. The Phasemeter monitors the PPM Channel 1 and 2.

The Jelly-Fish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses five vectors in the Jelly-Fish™.

The vectors in the Jelly-Fish™ are assigned clockwise in the order C, Rt, S, S, Lt. Please note that C and S are derived from the sum and difference of the Lt and Rt.

If analogue sources has to be monitored, use factory preset #6 where the sources for the destinations #65 to #66 are changed.

If the MSD is fitted with a '2 Channel Analogue Input Module' then the preset is configured as the following table shows.

Preset #10 – Lt Rt				
	Destination Number.	Destination Name.	Source Number.	Source Name.
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
Jelly-Fish™	55	CENT	59	CH 3
	56	C + 1	58	CH 2
	57	C + 2	60	CH 4
	58	C + 3	60	CH 4
	59	C + 4	57	CH 1
	60	C + 5	51	OFF
	61	C + 6	51	OFF
	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	33	Lt
	66	CH 2	34	Rt
	67	CH 3	55	C
	68	CH 4	56	S
	69	CH 5	51	OFF
	70	CH 6	51	OFF
	71	CH 7	51	OFF
	72	CH 8	51	OFF

Destinations 1 to 48 and 73 to 96 are set to OFF (#51).

C. Software Assurence.

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