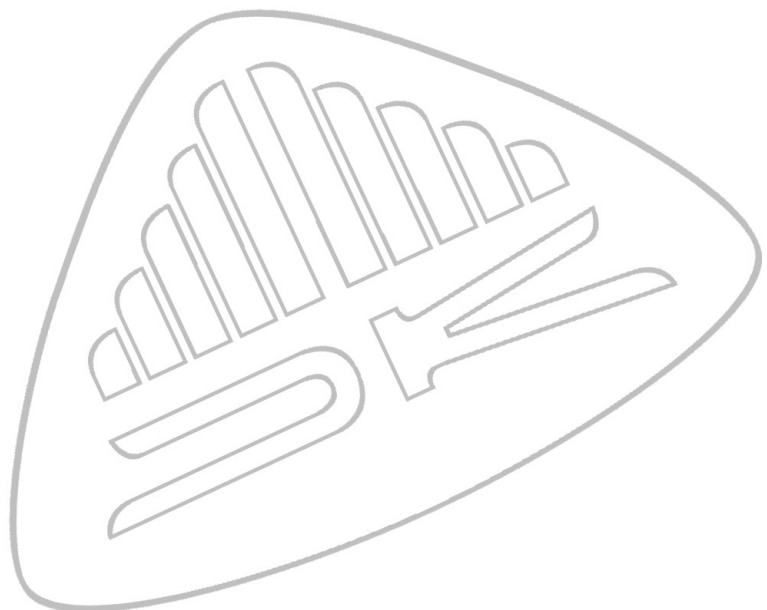# [Users Guide]
Software Release 5.4

Applies to the following models:

**MSD600M++, PT0600M,
PT0660M and PT0660M-LS.**

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Software manual for MSD software version 5.4.

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++, PT0600M
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.

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

`'ABOUT'` - This is a name of a specific function or item in the MSD.

In the headlines above some of the sections of this manual are images of keys and menu items shown on the LCD screen. These images represent the necessary key order to reach a particular function from the main menu in `'MSD Native Style'`.

Some functions can be reached by different key combinations and menus, but the headline will always describe how to reach the function from the `'MSD Native Style'`.

| START | MATRIX | PRESET | SIG GEN | UTILITY | 1/3 OCT | FFT | METER |

The main menu in 'MSD Native Style'.

2   MSD Native Style.

When the MSD is powered up for the first time the MSD will start in the `'MSD Native Style'`
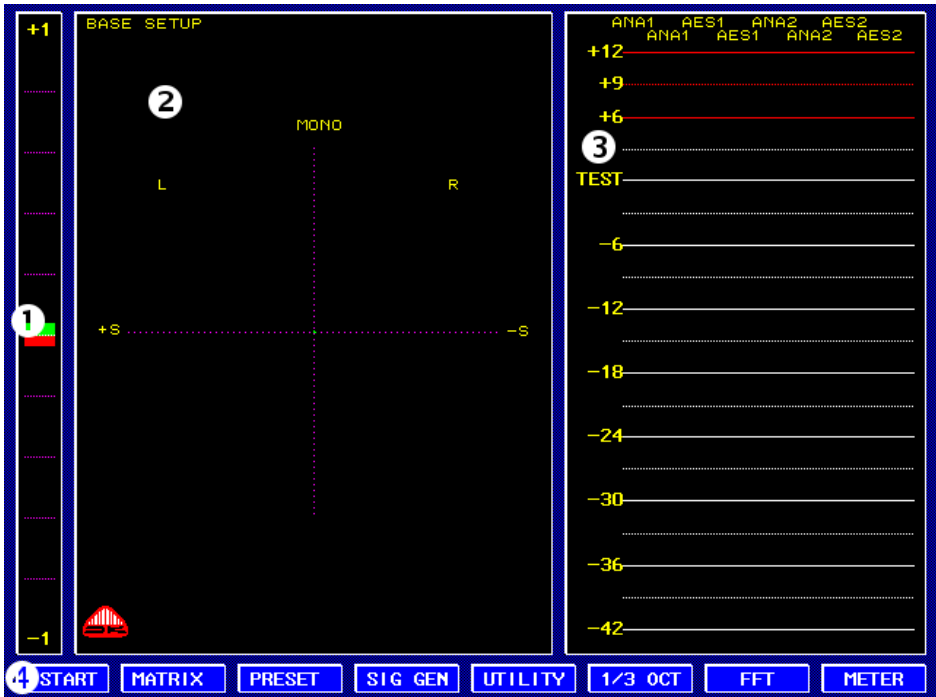
The Main Menu in 'MSD Native Style'.

In `'MSD Native Style'` the MSD will show the three main metering functions and the main menu.

❶: The Phase Correlation Meter.

❷: The Audio Vector Oscilloscope / JellyFish™ / StarFish™.

❸: The Peak Programme Meter (PPM).

❹: The main softkey menu.

Please see section 5.1 for a detailed description of the main metering functions.

2.1   Navigating the softkey menus.

The MSD is controlled using the eight softkeys placed below the LCD display. These softkeys correspond to the menus in the lower part of the LCD display. These menus 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 sinus 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 **[dBFS]**) can contain functions related to the keys on the right of it.

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 List. `START`

`SELECT` `▲` `▼` `SHOWALL`   `INITIAL` `ABOUT`

3.1   Navigating the Application Start List.

`START` ➔ `SELECT` `▲` `▼`

```
APPLICATION START LIST
PROGRAM LIST (STANDBY)
CONTROL PANEL
AUDIO GLOBALS
SCREEN SETTINGS
SYSTEM INFORMATION
SECURITY SETTINGS
ERROR REPORT
AUDIO MENU MODES
MSD NATIVE STYLE
PRESET SHORT HAND
PRESET LIST DETAILED
MATRIX COMPACT
MATRIX DETAILED
AUDIO UTILITIES
SIGNAL GENERATORS
BLITS SEQUENCER
DCF77 RADIO CLOCK
AUDIO ANALYSERS
1/3 OCTAVE IEC
FFT 1024 POINTS
AUDIO LEVEL TOOLS
RLB-LU METERING
SONY DMU1630 EMULATION
GRAPHICAL LOUDNESS
VIDEO METER TOOLS
```

The Application Start List.

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

This list is divided into different groups/folders that can be expanded or collapsed when selected.

To execute an application in the MSD, use the **[↑]** / **[↓]** keys to move the green selection bar to the desired application and press **[Select]**. Use the same method to expand or collapse a group/folder highlighted in yellow.

The following applications is available from the Application Start List.

APPLICATION START LIST		
PROGRAM LIST (STANDBY)	Standby Function.	Section 3.1
CONTROL PANEL	(Folder)	
AUDIO GLOBALS	Setup of global reference levels.	Section 4.1
SCREEN SETTINGS	Setup of the LCD display and VGA connector.	Section 4.2
SYSTEM INFORMATION	Setup of various global settings.	Section 4.3
SECURITY SETTINGS	Setup of the MSD Lock Code.	Section 4.4
ERROR REPORT	Access the 'Software Trap Report'	Section 4.5
AUDIO MENU MODES	(Folder)	
MSD NATIVE STYLE	Normal MSD operation.	Section 5.1
	Phase Correlation Meter.	Section 5.1.1
	Audio Vector Scope/JellyFish™/StarFish™.	Section 5.1.2
	PPM bargraphs. (Maximum 32).	Section 5.1.3
PRESET SHORT HAND	Quick preset selection.	Section 5.2
PRESET LIST DETAILED	Preset Selection and Configuration Menu.	Section 5.3
MATRIX COMPACT	The Standard Audio Matrix Menu.	Section 5.4
MATRIX DETAILED	The Extended Audio Matrix Menu.	Section 5.5
AUDIO UTILITIES	(Folder)	
SIGNAL GENERATORS	The signal generator menu.	Section 6.1
BLITS SEQUENCER	Black & Lane's Identification Tones for Surround	Section 6.4
DCF77 RADIO CLOCK	DCF77 Radio Controlled Real time Clock.	Section 6.2
AUDIO ANALYSERS	(Folder)	
1/3 OCTAVE IEC	The 1/3 Octave Analyser.	Section 7.1
FFT 1024 POINTS	The FFT Analyser.	Section 7.2
AUDIO LEVEL TOOLS	(Folder)	
RLB METERING	Loudness according to ITU 1770.	Section 8.2
SONY DMU1630 EMULATION	Sony DMU1630 emulation	Section 8.1
GRAPHICAL LOUDNESS	Short term loudness measurement	Section 8.3
VIDEO METER TOOLS	(Folder) Not implemented in the MSD600M++ Series	

3.1.1   MSD Standby mode. `START` ➔ `PROGRAM LIST (STANDBY)`

When selecting the menu item `'Program List (Standby)'` the MSD will go into a standby mode where the display is turned off. The routing functions in the audio matrix will still be active.

3.1.2   Default Startup Application. `START` ➔ `INITIAL`

If write protection is disabled *(see section 3.2.1)* the **[INITIAL]** key will become active. Selecting this key will set the currently highlighted application as the default startup application. This application is the first to be executed after power up or reset. If `'Program List (Standby)'` is selected, the MSD will start up in the `'Application Start List'` not the Standby mode.

3.2   The Information Window. `START` ➔ `ABOUT`

`EXIT`   `ENABLE` `RESTART`

```
INFORMATION
DK-TECHNOLOGIES A/S
MARIELUNDVEJ 37D
DK-2730 HERLEV
TEL +45 4485 0255
FAX +45 4485 0250
www.dk-technologies.net

HARDWARE VER :2.00
SERIAL NUMBER:8347
PASSWORD    :B5R3
SOFTWARE VER :05.11
FIRMWARE VER :7
HARDWARE ID  :0120
EXT BIOS ID  :0402

INPUT MODULES
#1: (31) 1-AES 2-ANA
#2: (0) NOT INSERTED
#3: (0) NOT INSERTED
#4: (0) NOT INSERTED

OUTPUT MODULES
#1: (11) 1-AES 2-ANA
#2: (0) NOT INSERTED
#3: (0) NOT INSERTED
#4: (0) NOT INSERTED
```

The About Window of a MSD600M++.

The MSD has 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 `'Information'` 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.

Every new MSD running software version 5.0 or later is registered at our web site *(SWA-login)*. Using the MSD serial number and the corresponding 4 character password shown in the `'Information'` Window, it is possible to login and download the latest software for the MSD. It is also possible to download the original software and any customized preset configurations the MSD was delivered with. We encourage you are as an end-user to login to this web page and register your personal contact information.

When contacting technical support please provide the information found in the `'Information'` 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 in the `'Information'` Window.

3.2.1   MSD Write Protection. `START` ➔ `ABOUT` ➔ `ENABLE`

To avoid accidental changes to the presets and other critical functions the MSD is fitted with a Write-Protect function.*(Please see section 5.2 for further information about presets.)* Pressing the **[Enable]** key toggles the MSD write protection. If the key is greyed out, the MSD is write protected and changes to the MSD cannot be saved, and some functions can not be accessed.

This Write-Protect function can be locked with a four digit security code to prevent unauthorised changes to the meter. Please see section 4.4 for further information about the `'Security Settings'`.

3.2.2   MSD Restart. `START` ➔ `ABOUT` ➔ `RESTART`

The restart 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 **[RESTART]** 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 restart function is a very handy way to force the MSD trough a Power-up sequence.

The **[RESTART]** key is considered a critical function and is therefore disabled until the MSD Write-Protection is disabled. *(Section 3.2.1.)*

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

3.2.3 MSD Factory Reset.

A full factory reset can not be performed locally on the MSD.

To perform a full factory reset it is necessary to use the Windows® software DK-Matrix to download a new preset package using a serial port.

DK-Matrix, standard factory preset packages and the MSD firmware is supplied on a CD-Rom with the MSD.

If it should be necessary to update the software installed in the MSD please follow the instructions in the DK-Matrix users guide.

DK-Matrix will also make it easier to configure the MSD to specific needs. It is even possible to directly remote control the MSD using a serial port.
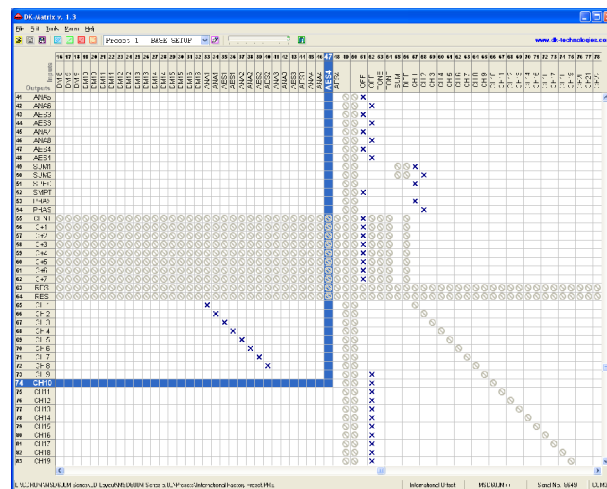
Screenshot of the Windows® software DK-Matrix.

If the MSD was delivered with a customized preset configuration it is possible to login to the SWA page described in section 3.2 and download the original preset.

4   The MSD Control Panel.

The MSD Control Panel is a group of functions that configures the global functionality of the MSD.

4.1   Audio Globals. START ➜ AUDIO GLOBALS

| EXIT | ▲ | ▼ | VALUE ▲ | VALUE ▼ | | ☐ EXT S | RESTART |

The audio globals define the reference levels and gain settings for the entire MSD and all the presets.

Please note that it is only possible to change settings in the Audio Globals menu if the **[ENABLE]** flag in the About menu is enabled.

To change a value move the cursor to the item you want to change with the **[↑]** / **[↓]** keys and change the value with the **[VALUE ↑]** / **[VALUE ↓]** keys.

Press the **[RESTART]** key to save the changes. This will also restart the meter.

4.1.1   Analogue FS Input.

Audio FS Input defines the global input reference level for the MSD.

The reference level can be adjusted between +12dBu and +24dBu.
Different reference levels are used in different regions of the world.

Default values:
International (EBU):   `'+18dBu EBU'`.

Germany (DE):   `'+15dBu DE'`

United States (US):   `'+24dBu US'`.

4.1.2   Analogue FS Output.

Audio FS Output defines the global output reference level for the MSD.
This setting can not be changed.

4.1.3   LEQ Summing Method.

The LEQ Summing method is the summing method used by the Loudness Applications. The summing methods can be set to "3dB DOUBBL", "ITU 400Hz" or to "ITU 1KHz". The "3dB DOUBBL" is used when measuring FLAT, Leq(M) or Leq(A). The "ITU 400Hz" and the "ITU 1KHz" is used when measuring Loudness according to the ITU BS1770 recommendation. Use the "ITU 400Hz" if the reference frequency should be 400Hz or use the "ITU 1KHz" if it should be 1KHz.
(See section 8.1 for RLB Loudness and section 8.3 for Graphical Loudness)

4.1.4   LEQ Flat 85 Equals.

This is the digital reference level when using the FLAT response in Graphical Loudness. Default `'-20dBFs US'`. (See section 8.3 for Graphical Loudness)

4.1.5   LEQ RLB Equals.

This is the digital reference level when using the RLB (ITU BS1770) response

in Graphical Loudness and RLB-LU Metering. Default **'-20dBFs US'**. (See section 8.1 for RLB Loudness and section 8.3 for Graphical Loudness)

4.1.6   LEQ(M) 85 Equals.

This is the digital reference level when measuring the Leq(M) in the Graphical Loudness. Default **'-20dBFs US'**. (See section 8.3 for Graphical Loudness)

4.1.7   LEQ(A) 85 Equals.

This is the digital reference level when measuring the Leq(A) in the Graphical Loudness. Default **'-20dBFs US'**. (See section 8.3 for Graphical Loudness)

4.1.8   Downmix Out Gain.

This is the overall output gain for the Surround Sound down mix function.
Default [dB]: **'-3.3'**.

4.1.9   Downmix CNT to L/R.

This is the Centre channels gain level when adding it to the Left and Right in the Surround Sound down mix function.
Default [dB]: **'-6.0'**.

4.1.10   Master Volume SPL (Sound Pressure Level).

The wheel button on the PT0660M series can be used as a Master Volume Control. Please see section 5.5.5.1 for further information about assigning audio groups. (The Master Volume Control only affects destinations in the Master Group.)

The **'Master Volume SPL'** parameter defines the default startup output level for the master volume control.

If this setting is set to **'Individual'** the master volume setting can be stored individually for each preset, otherwise the setting will apply to all presets.

If assigned to a value the wheel button only works as volume control key otherwise it can also be used to operate the Meter.

Default values:

PT0660M: **'Individual'**
PT0660M-LS [dB}:   **'-70'**

4.1.11   Master Volume Step.

The Master Volume Step defines the size of each step when the level of the Master Volume is adjusted by the wheel button on a PT0660 series MSD.
Default [dB]: **'1.5'**

4.2   External Synchronisation. `☐ Ext S`

The **[Ext S]** key in the Audio Globals menu toggles the external synchronisation.
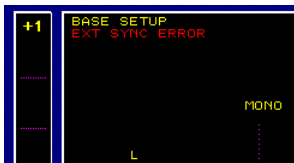
The top left corner of the MSD screen in MSD Native Style.

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 (Only on older meters, all new meters can sync on the AES/Analogue input module type 51). Selecting an external synchronisation source will force the MSD to follow the sample rate of the sync signal. **It is important to ensure that the external sync sources sample rate is 48kHz.**

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 Oscilloscope'** window. **If the MSD shows this warning the MSD will not function correctly.**

External Synchronization on the MSD600M++ series is selected from this menu. On all other models it is selected from the AES option menu.

4.3   Screen Settings   `START` ➜ `SCREEN SETTINGS`

`EXIT`   `DARKER`   `BRIGHT`   `SAVE`   `PRI 1`   `SEC 1`

4.3.1   LCD Backlight intensity. `DARKER` `BRIGHT`

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

4.3.2   LCD Screen Selection Setup. `PRI 1` `SEC 1`

This function is only supported by the PT0760M Series.

The **[PRI]** key will select which of four available screen setups is shown on the internal LCD screen. The **[SEC]** key will select the screen setup for the external monitor.

4.3.3   Saving the screen settings. `SAVE`

If the MSD Write Protection is disabled *(please see section 3.2.1)* the **[SAVE]** key will become active. Pressing this key will save the current screen settings.

4.4   **Security Settings.** `START` ➔ `SECURITY SETTINGS`

`EXIT` `EVRYBDY` `ENABLE` `0` `0` `0` `0` `SAVE`

From this menu it is possible to lock some of the functions found in the MSD.

The default code is **0000**.

If the MSD is locked, the MSD Write Protection can not be disabled and the state of the `'Function Lock'` (**[EVRYBDY]**) can not be changed. Please see section 3.2.1 and 4.4.1 for further information.

If the lock code has been lost it can be cleared using the Windows® software DK-Matrix. Please refer to the DK-Matrix User's Guide for further information about the subject.

4.4.1   **Function Lock. (Everybody).** `EVRYBDY`

APPLICATION START LIST
PROGRAM LIST (STANDBY)
SECURITY SETTINGS
MSD NATIVE STYLE
PRESET SHORT HAND

The Application Start List in locked mode.

If the key **[EVRYBDY]** is highlighted then all the functions in the MSD is available, however if it is greyed out, the `'Application Start List'` is limited to only four items.

Available functions when the 'Function Lock' is active.		
PROGRAM LIST (STANDBY)	Standby Function.	Section 3.1
SECURITY SETTINGS	Setup of the Write Protect Lock Code.	Section 4.4
MSD NATIVE STYLE	Phase Correlation Meter, Vector Oscilloscope and PPM-Bargraph. All in viewing mode only. (No changes can be applied.)	Section 5.1.1
PRESET SHORT HAND	Quick preset selection.	Section 5.2

Please note that the state of the **[EVRYBDY]** function can only be changed if the **[ENABLE]** key is highlighted.

4.4.2   **Lock Code.** `ENABLE` `0` `0` `0` `0` `SAVE`

If the **[ENABLE]** key is highlighted the MSD is unlocked.

4.4.2.1   **Lock the MSD.**

To lock the MSD make sure that the **[ENABLE]** key is highlighted (unlocked). When the **[ENABLE]** key is highlighted enter a four digit code and press the **[SAVE]** key.

The MSD is now locked with the new code.

4.4.2.2   **Unlock the MSD.**

To unlock the MSD make sure that the **[ENABLE]** key is greyed out (locked). When the correct four digit code is entered the **[ENABLE]** key is automatically highlighted and the MSD unlocked. Do not press the **[SAVE]** key, this will re-lock the MSD.

If a save function is performed where the MSD Write Protect is automatically enabled (e.g. a preset is saved) then the MSD Write Protect is automatically enabled the MSD is also re-locked with the current code.

4.4.2.3   **Disable or change the lock code.**

To disable the lock code, enter the code **0000** and press the **[SAVE]** key when the MSD is unlocked (the **[ENABLE]** key is highlighted).

If a code other than **0000** is selected, the code is changed and MSD is locked with the new code.

4.5   Error Report. START ➔ ERROR REPORT

```
SOFTWARE TRAP REPORT:

0000   NO ERROR RECORDED
```

```
EXIT                                    □ ETRAP  □ MONTR  RESTART    RESET
```
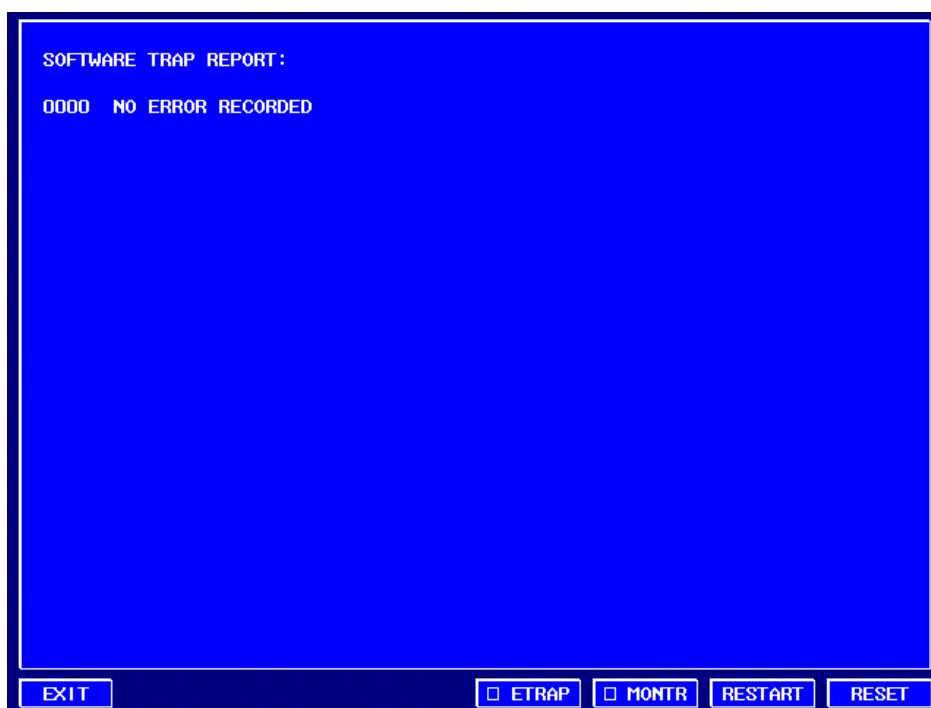
The **'Error Report'** is a tool used by DK-Technologies to detect errors.

If the MSD starts in this mode an error has been detected. To return to normal operation press **[RESET]** and then **[EXIT]**.

5   The Audio Menu Modes.

The **'Audio Menu Modes'** is a group of functions that is used in normal operation of the MSD.

5.1   The Main Metering Functions. *(MSD Native Style)*

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

Please note that if the **'Function Lock'** is active *(see section 4.4.1)*, the main menu in **'MSD Native Style'** is greyed out and no changes can be made.

5.1.1   The Phase Correlation Meter.

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 should 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 5.4 'The Compact Audio Matrix' on how to select the two source signals for the **'Phase Correlation Meter'**.

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

5.1.2   Audio Vector Oscilloscope or Surround Sound Monitoring.

The middle window in **'MSD Native Style'** has two functions. It can either be used as an **'Audio Vector Oscilloscope'** or as the JellyFish™ / StarFish™ Surround Sound Monitor.

The type of display is depending on the settings in the audio matrix.

Refer to section 5.4 'The Compact Audio Matrix' for a detailed description on how to navigate 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 JellyFish™ / StarFish™ surround sound display, then the sources for destination #53 and #54 (PHAS) will only affect the **'Phase Correlation Meter'**. This means it is possible to monitor the phase relationship between two channels independent of the JellyFish™ / StarFish™. If destinations #53 and #54 (PHAS) are set to 'OFF' then the **'Phase Correlation Meter'** and the **'Audio Vector Oscilloscope'** / JellyFish™ / StarFish™ will be hidden.

Note that the **'Phase Correlation Meter'** and **'Audio Vector Oscilloscope'** share the same destinations in the Audio Matrix (#53 and #54).

Please note that it is a good practice to route the signals to the **'Audio Vector Oscilloscope'** / JellyFish™ / StarFish™ through the **'Peak Programme Meter'** destination #65 to #96 (or #74 in surround mode), this way when changing the input to a PPM-Bargraph then the input to the **'Audio Vector Oscilloscope'** / JellyFish™ / StarFish™ will change accordingly. Appendix B.1 Factory Presets includes examples on how to configure the matrix.

The **'Audio Vector Oscilloscope'** uses 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'** **cannot** 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 **'Extended Audio Matrix'**. Please refer to section 5.5.4.2.2 Meter Compressor.

5.1.2.1   JellyFish™ or StarFish™.

The mode of the surround sound scope is configured in the **'Extended Matrix'**. Please see section 5.5.4.3 for further information about selecting between the JellyFish™ and StarFish™ surround sound scope .

5.1.2.2   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'`**.
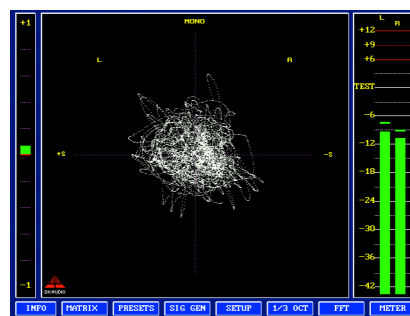
Ideal stereo signal.
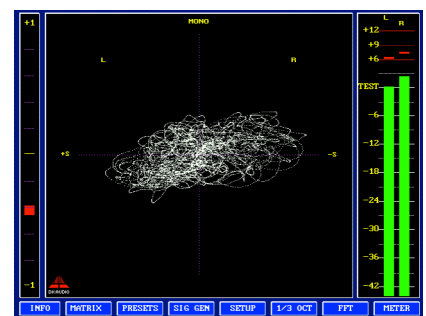
Reversed phase stereo signal.
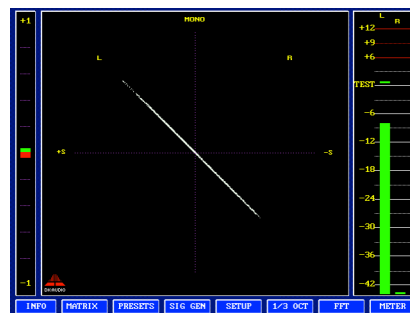
Left signal.
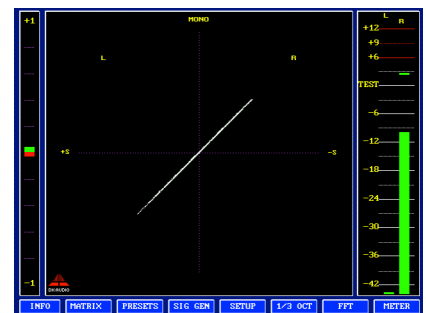
Right signal.
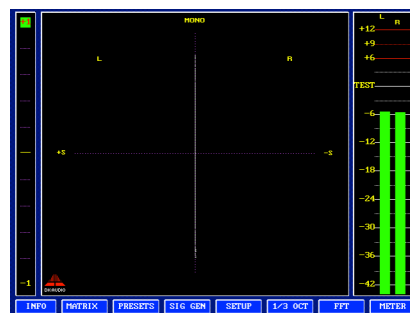
Mono signal.
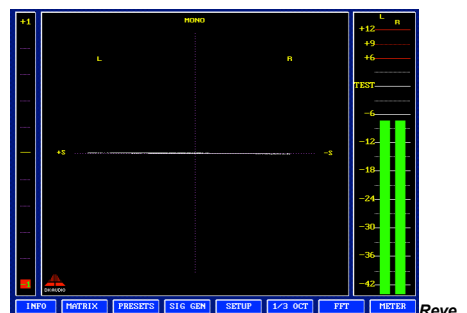
Reversed phase mono signal.

Please refer to section 5.4 'The Compact Audio Matrix' on how to select the two source signals for the **`'Audio Vector Oscilloscope'`**. Destination #53 and #54 in the audio matrix.

5.1.2.3   **The JellyFish™ Surround Sound Monitor.**

The MSD600M has a full Surround Sound monitoring function using the unique JellyFish™ image.
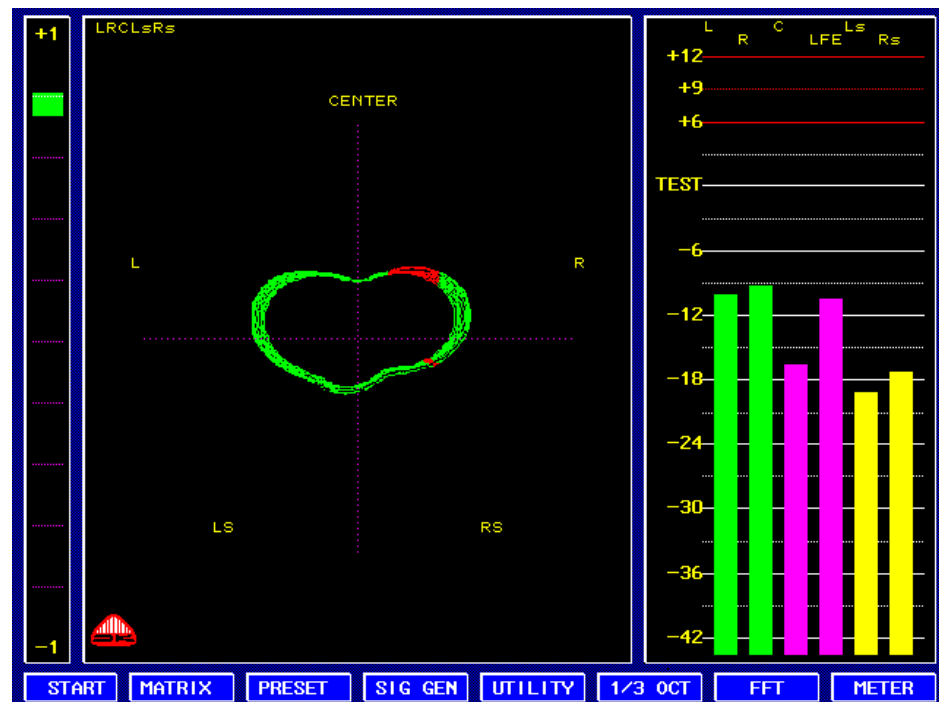
The JellyFish™ in 5.1 surround mode, indicating some phase errors.

The JellyFish™ figure represents the surround channels in a vector format. The JellyFish™ 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 JellyFish™ 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 JellyFish™.

The labels used in the 'background'-setting for the JellyFish™ 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 assigned PPM-bargraphs relates to the JellyFish™. The demonstrated format is taken from factory preset 6 'LRCLsRs' which is designed for monitoring a 5.1 surround sound setup.
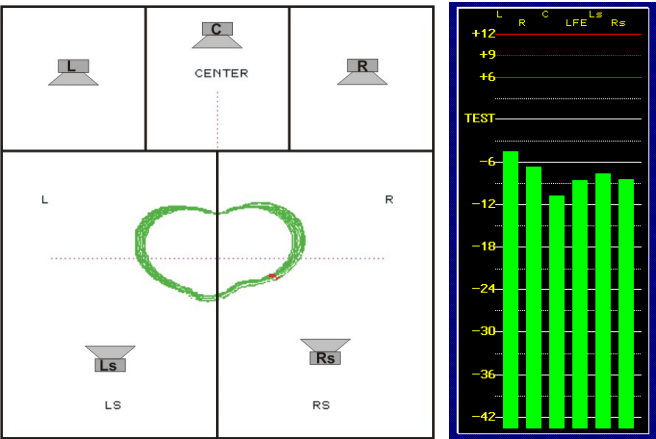
The JellyFish™.
Each of the 5 areas illustrate a physical input source,
labelled as the corresponding PPM Bargraph.

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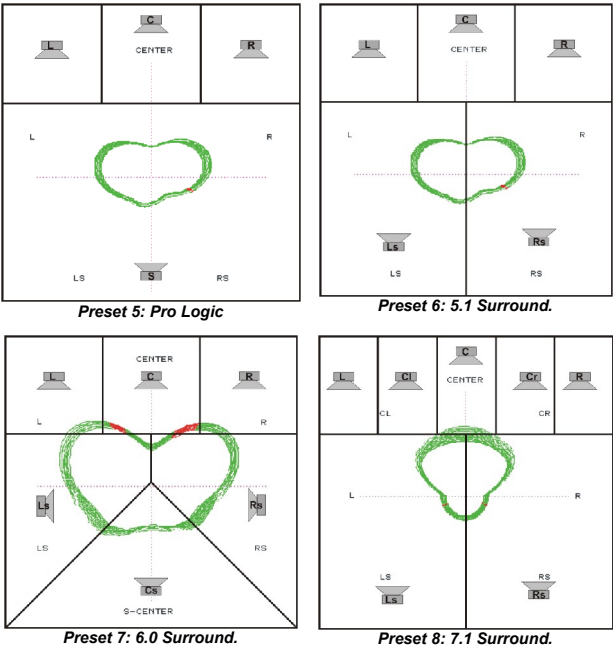
To set the MSD for a given surround sound format, the specific Matrix destinations (#55 to #62) have to be assigned to a source . The number of destinations assigned determines which surround sound format that is selected.

Preset 6 – Default 5.1 Surround Sound Setup.

Physical Audio Source		PPM-bargraphs		JellyFish™		
Source 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 JellyFish™ destinations #60 to #62 are set to OFF.
When routing audio from the physical inputs to the JellyFish™ 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 JellyFish™. Please note that when a PPM-bargraph is used as a source the source number is 8 less then the destination

number for the same PPM-bargraph. In this case the PPM-bargraph #3 which have the destination number 67 relates to source #59. Source #59 is then routed to the JellyFish™ center channel (destination #55).

This might seem like a long way around a simple solution, but the advantage is that the JellyFish™ surround sound monitor 'follows' the PPM-bargraphs so if a new physical source is needed for a particular PPM-bargraph then there is no need to select a new source for the JellyFish™.

The audio signals assigned to the JellyFish™ are assigned clockwise starting with the center channel.
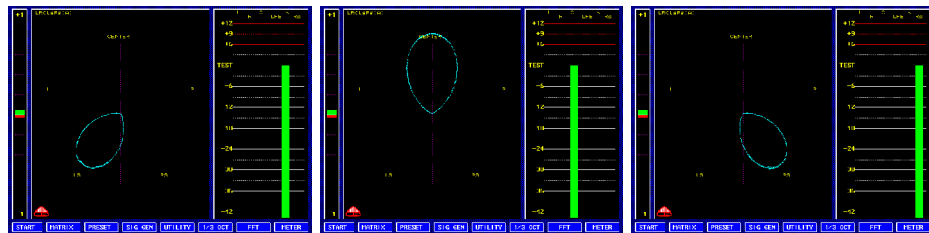
A sinus signal in the left surround channel. *A sinus signal in the center channel.* *A sinus signal in the right surround channel.*
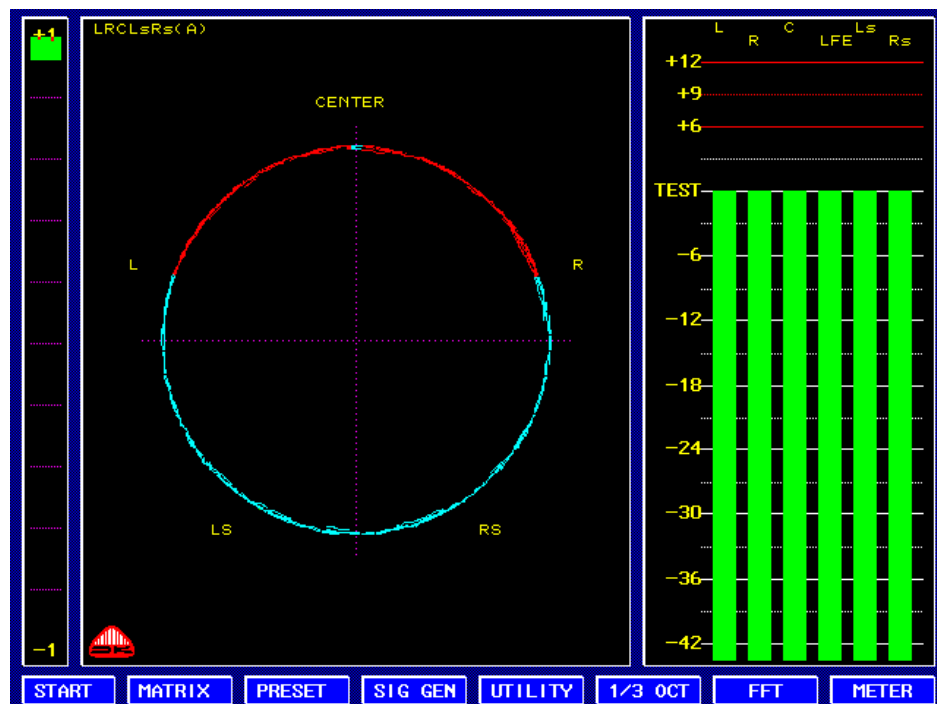
A circular-shaped figure will indicate a perfect surround signal. The red areas in the circle indicates that the center channel is 180 degrees out of phase with the left and right channels.

A complete comparison list for all the surround sound formats can be found in Appendix B '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 B 'Factory presets'.

To monitor the complex phase relationships in a common surround sound signal the JellyFish™ is identifying any phase problems accurately by a colour change to red in the relevant vector in the JellyFish™ 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 JellyFish™ 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 JellyFish™ monitor.

5.1.2.4   StarFish™ Surround Sound Monitor.

The StarFish™ is configured in the same way as the JellyFish™ so please refer to section 5.1.2.2 for information about configuring the StarFish™.

The StarFish™ visualizes surround sound in an intuitive way that is simple to understand and easy to read. It gives an accurate view of the volume of the sound coming from the loudspeakers, as it is perceived by the listener. Further, in the same display it shows the correlation between the sound channels indicating to the audio engineer of possible problems.
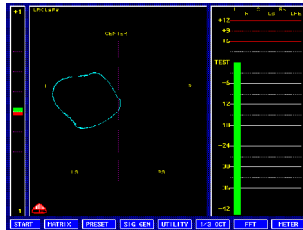
To illustrate how the StarFish™ works, let us start with the display of a single tone in the left loudspeaker (L). In the picture to the left the characteristic display extends from the centre towards the upper left of the display. The size of the figure depends on the volume of the tone.

When we at the same time also switch on the same tone to the left surround (Ls) loudspeaker, we obtain the display in the picture to the right. It extends from the centre to the upper left and lower left of the screen, indicating that the sound is coming from both left hand loudspeakers. Where the two fields meet the colour of the contour is yellow to indicate that the signals are correlated, as

the two channels carry the same signal and are in phase. In the middle left side of the screen the figure has its largest extension from the centre to indicate that the sound comes from a direction between the two loudspeakers.
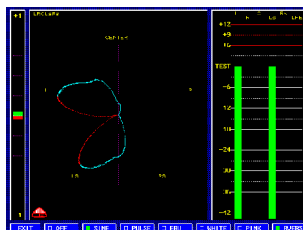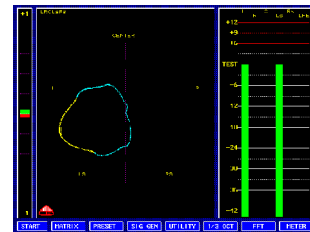
If we still apply the same tone to the L and Ls loudspeakers, but reverse the phase to one of the loudspeakers, we obtain a completely different display as shown in the picture to the left. Because of the opposite phases the signals cancel each other between the two loudspeakers. The contour of the display changes to red between L and Ls to indicate the negative correlation between the channels.

In the picture to the right we apply two different tones with the same level to the L and Ls loudspeakers. On the display the volume is shown as equal in the two loudspeakers, and the contour is blue to indicate that the tones are uncorrelated.
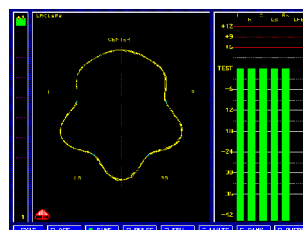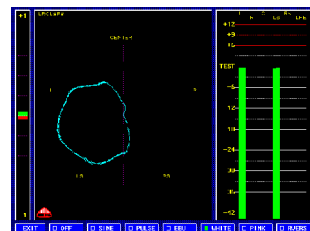
When we extend these principles to the 5 channels in the 5.1 surround sound signal and apply the same tone to all the channels, we obtain the characteristic star-shaped display from which the StarFish™ has got its name. When a real surround sound signal is applied to the audio meter, the shape and colour of the StarFish™ changes continuously like a jellyfish to indicate that the volume and the correlation of the audio signal.

A complete comparison list for all the surround sound formats can be found in appendix B.

5.1.2.4.1   Surround sound stereo downmix.

When the StarFish™ is active, a stereo downmix of all the surround channels is automatically created. This stereo downmix is available in the audio matrix as source #49 `'LMIX'` and #50 `'RMIX'` and can be routed to any destination in the audio matrix. The parameters used to make the downmix can be found in the Audio globals menu. See section 4.1.8 and 4.1.9.

5.1.3   The Peak Programme Meter (PPM). `METER`

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

`EXIT` `SCALE` `SET REF` `■ PEAK` `☐ HOLD` `☐ FLEX` `CLEAR` `☐ 20dB`

The `'Peak Programme Meter'` is designed for direct measurement of the quasi-peak level of complex electrical signals occurring in the transmission of music and speech. This is achieved without varying the sensitivity of the device in order 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 is configured in the audio matrix. When PPM-bargraphs are enabled or disabled the `'Audio Vector Oscilloscope'` window is automatically scaled to fill the remaining part of the screen. See section 5.4 for information on how to route audio channels to the PPM-bargraphs. (Destinations #65 to #96.) Please observe that if the JellyFish™ / StarFish™ is active the maximum number of active PPM-bargraphs is 16.

Please refer to section 5.5.4.1 for information on how to change the appearance of the PPM-bargraphs.

5.1.3.1   Scale selection. [METER] ➔ [SCALE]

[EXIT] [■ I] [□ IIA] [□ IIB] [□ DIN] [□ VU] [□ DMU1] [□ DMU2]

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

All scales can be changed using the Windows® software 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 scale package that is installed as standard.

| Type I – Nordic (+20 dB Active.) | Type IIA - BBC | Type IIB - EBU | DIN | VU | DMU1 | DMU2 |
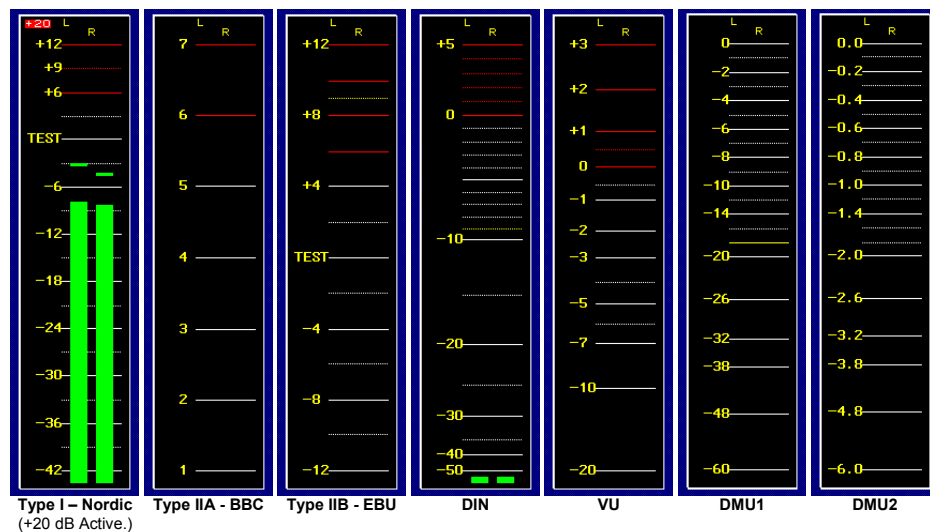
5.1.3.2   PPM Set Reference Level. [METER] ➔ [SET REF]

If write protection is disabled *(section 3.2.1)* the **[SET REF]** key will become active. This function will make it possible to set the analogue and digital reference levels for the PPM-Scales individually from the rest of the MSD. This setting will only affect physical analogue and digital inputs in the MSD. The default setting is an offset of 0 dB for both analogue and digital signals.

To change the reference level for the PPM-Scales press the **[SET REF]** key and the reference level menu will appear.

[EXIT] [SAVE] [ANALOG] [▲] [00] [▼] [DIGITAL] [▲] [00] [▼]

Use the **[↑]** / **[↓]** keys to change the analogue and digital reference levels individually. When done press the **[SAVE]** key to save the changes to the preset. Each of the 11 presets can have it's own setting.

5.1.3.3   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.

5.1.3.4   **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]** is enabled and **[HOLD]** is active then the peak segments will remain at the highest reading until the **[CLEAR]** key is pressed or another menu has been accessed.

5.1.3.5   **Flex Mode.** `METER` ➜ `☐ FLEX`

The **[FLEX]** key will enable or disable the flex-mode. In flex-mode each PPM-bargraph can have its own ballistic properties independent of the currently selected scale.

These properties are set in the extended matrix. Please see section 5.5.4.1.1 for more information.

5.1.3.6   **PPM Gain. (+20dB)** `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 in section 5.1.3.1 on the previous page.

5.2   Presets.

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 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.

5.2.1   Preset Short Hand. [START] ➔ PRESET SHORT HAND

| START | BASE SE | ALL ANA | ALL AES | LR | LRCS | LRCLsRs | LRLsRsC |

The `'Preset Short Hand'` menu is an easy way to quickly toggle between presets.

In the `'Preset Short Hand'` menu the first seven presets in the MSD are assigned to softkeys. Pressing a softkey in this menu will immediately switch to the corresponding preset. Use the **[START]** key to return to the `'Application Start List'`.

5.3   Preset List Detailed. PRESET

| EXIT | ▲ | ▼ | RECALL | | SETUP |
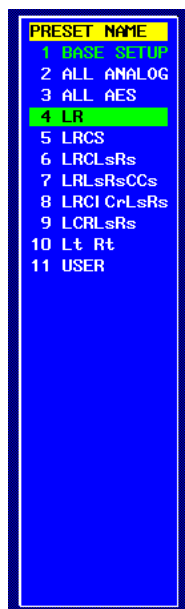
The preset selection menu can easily be accessed from the `'Application Start List'` by selecting the item `'Preset List'`. The `'Preset Selection List'` can also be accessed from the `'Main Menu'` in `'MSD Native Style'` by pressing the **[PRESET]** key.

```
PRESET NAME
 1 BASE SETUP
 2 ALL ANALOG
 3 ALL AES
 4 LR
 5 LRCS
 6 LRCLsRs
 7 LRLsRsCCs
 8 LRClCrLsRs
 9 LCRLsRs
10 Lt Rt
11 USER
```

The preset selection window.

5.3.1   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 **[RECALL]** key.

5.3.2   Preset Setup. `PRESET` ➔ `SETUP`

`EXIT` `▲` `▼` `SAVE` `SAVE AS` `INITIAL` `EDIT`

When the MSD write protection is disabled the **[SETUP]** key is enabled. Please see section 3.2.1 on how to disable the MSD Write Protection. Pressing the **[SETUP]** key will bring up the **'Preset Setup Menu'**.

5.3.3.1   Save a Preset. `PRESET` ➔ `SETUP` ➔ `SAVE`

To save a preset in the current preset location press the **[SAVE]** key. When the preset is saved the MSD write protection will automatically be enabled. The setup menu will also be exited. If the green selection bar is highlighting another preset than the current, the preset is still saved in the current preset location. To save the preset at another preset location, please see section 5.3.3.2 for further information.

5.3.3.2   Save a Preset to another memory location.
`PRESET` ➔ `SETUP` ➔ `▲` `▼` `SAVE AS`

To save a preset to another preset location than the current, use the **[↑]** / **[↓]** keys to move the green selection bar to the preset which shall be replaced by the new settings.

When the **[SAVE AS]** 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 the input and output names in the audio matrix will not be replaced.

5.3.4   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 using the **[↑]** / **[↓]** keys, and then press the **[INITIAL]** key. When a preset is selected as the default startup preset the item in the **'Preset List'** will be highlighted in green.

5.3.5   Rename a Preset. `PRESET` ➜ `SETUP` ➜ `EDIT`

```
PRESET NAME
 1 BASE SETUP
 2 ALL ANALOG
 3 ALL AES
 4 LR
 5 LRCS
 6 LRCLsRs
 7 LRLsRsCCs
 8 LRCl CrLsRs
 9 LCRLsRs
10 Lt Rt
11 USER
```
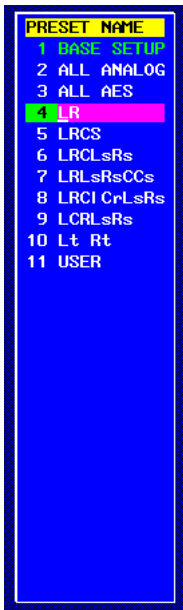
The preset selection window in edit mode.

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 the first character in the preset name.

| EXIT | ▲ | ▼ | ◄ | ► | COPY | PASTE | ENTER |

To select a new character at the current cursor 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 Oscilloscope'` window.

Use the **[COPY]** key to copy the entire preset name into memory, and then at a later time use the **[PASTE]** key to paste it into another preset.

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.

To abandon the changes press **[EXIT]**.

5.4   The Compact Audio Matrix. *(Matrix Compact.)* `MATRIX`

`EXIT`   `▲`   `▼`   `PgUp`   `PgDn`   `SELECT`   `SAVE`   `EXTEND`

#	OUTPUT	INPUT	INPUT
33	ANA1	OFF	39 AES2
34	ANA2	OFF	40 AES2
35	AES1	OFF	41 ANA3
36	AES1	OFF	42 ANA3
37	ANA3	OFF	43 AES3
38	ANA4	OFF	44 AES3
39	AES2	OFF	45 ANA4
40	AES2	OFF	46 ANA4
41	ANA5	OFF	47 AES4
42	ANA6	OFF	48 AES4
43	AES3	OFF	49
44	AES3	OFF	50
45	ANA7	OFF	51 OFF
46	ANA8	OFF	52 OFF
47	AES4	OFF	53 TONE
48	AES4	OFF	54 TONE
49	SUM1	CH 1	55 SUM
50	SUM2	CH 2	56 DIFF
51	SPEC	CH 1	57 CH 1
52	SMPT	OFF	58 CH 2
53	PHAS	CH 1	59 CH 3
54	PHAS	CH 2	60 CH 4
55	CENT	OFF	61 CH 5
56	C+1	OFF	62 CH 6
57	C+2	OFF	63 CH 7
58	C+3	OFF	64 CH 8

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. (Please see section 5.4.3.1 for an example on how to achieve this.)

Selecting `'Matrix Compact'` from the `'Application Start List'` or pressing the **[MATRIX]** key from the `'MSD Native Style'` 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. e.g. the FFT-Analyser or a PPM-bargraph.

The second column shows the name of the destination, this name can be changed in `'The Extended Audio Matrix'` (see section 5.5.3) or with the Windows® based software 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 the illustration to the left 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. As an example of this, if an analogue input module was mounted in input slot number 1 then the module would register as input number 33 to 36, then input number 1 to 8 would not be available. But if a 4 Channel Digital Input module (8 mono) was installed in input slot 1 instead, then input number 1 to 8 would be available but not input number 33 to 36. Please see section 5.4.2 for further information about how physical input channels are assigned in the audio matrix.

Please see the examples in section 5.4.3 on how to route the audio signals (set x-points) in the matrix. If the MSD Write Protection is disabled *(section 3.2.1)* it is possible to quickly save the new x-points by pressing the **[SAVE]** key. It is therefore not necessary to save the whole preset as described in section 5.3.3.1.

5.4.1 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 5.5.5.3 for further details.
51	SPEC	Destinations for the FFT and 1/3 Octave Analyser. See section 7 for further details.
52	SMPT	Destinations for the internal SMPT-Timecode decoder. See section 6.3 for further details.
53..54	PHAS	Destinations for the phase correlation meter and the stereo audio vector oscilloscope. See section 5.1.1 for further details.
55..62	CENT, C+1..C+7	Destinations for the JellyFish™ / StarFish™. See section 5.1.2.2 for further details.
65..96		Destinations for the PPM-Bargraphs. See section 5.1.3 for further details.

Special attention must be paid 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 is used in several places. Please see appendix B where all of the applied factory presets are using this type of PPM signal feedback.

5.4.2   Audio Matrix Source List. (Module Assignment.)

Input modules can be placed in two categories: A 4-Channel module (mono) (e.g. SDI De-Embedder (E2) or a standard analogue input module (31)) and a 8 Channel Module (e.g. a 4-CH AES Module. (24))

The four input slots correspond to two different areas in the matrix depending on whether it is a 4 or 8 channel module that is installed.

Assignment of channel numbers in the audio matrix.		
	8-Channel Input Module. Module ID's: 24, 64	**4-Channel Input Module.** Module ID's: 11, 22, 31, 51, A2, E2
	Source Number.	**Source Number.**
Input Slot #1	1 to 8	33 to 36
Input Slot #2	9 to 16	37 to 40
Input Slot #3	17 to 24	41 to 44
Input Slot #4	25 to 32	45 to 48

The module ID's is shown in the 'Information Window' on the MSD.

The input modules and assignments can be combined and this will give many different combinations. The inputs are normally renamed from factory but if a new module is installed in a particular input slot the names has to be changed manually.

It is therefore necessary to know what type of channels the input module support. e.g. a standard analogue input module (31) has two analogue channels and two digital. The first two channels on that module are the analogue and the last two are the digital. If this module is placed in slot #2 the analogue channels will be assigned to input channel 37 and 38 and the digital channels are assigned to input channels 39 and 40. If the same module is placed in input slot #4 the channel assignment are 45 and 46 for the analogue channels and 47 and 48 for the digital channels.

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 from a SDI video signal.
49..50	LMIX / RMIX	Stereo downmix of surround sound routed to the StarFish™. See section 5.1.2.3 for further details.
51..52	OFF	These two inputs can be used to disable a destination in the matrix. E.g. turn off a PPM-bargraph.
53..54	TONE	Sources from the internal signal generator. See section 6.1 for further details.
55..56	SUM / DIFF	Sources from the sum and difference amplifier. See section 5.5.5.3 for further details.
57..88	CHxx	Sources from the PPM-bargraphs. As the only destination in the matrix the PPM-bargraphs can also function as a source. These would normally be routed to the Phase Correlation Meter and Vector Scope / JellyFish™ / StarFish™.

5.4.3   **Routing signals in the 'Audio Matrix'.**
5.4.3.1   **Example 1 – The MSD as a D/A converter.**

If the MSD is equipped with a standard input (#31 or 51) and a standard output (#11) 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.
(In this example the input module is placed in input slot #1 and the output module is placed in output slot #1.)

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

Step 1:

Press the **[MATRIX]** key from the `'MSD Native Style'` 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 in output slot #1.

Step 2:

To route the digital signal from the input module to this destination 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.
Select the destination in the Matrix Status Window.

Step 2.
Select source in the Matrix Source Window. (Window at the right.)

Step 3.
The MSD now functions as a D/A converter.
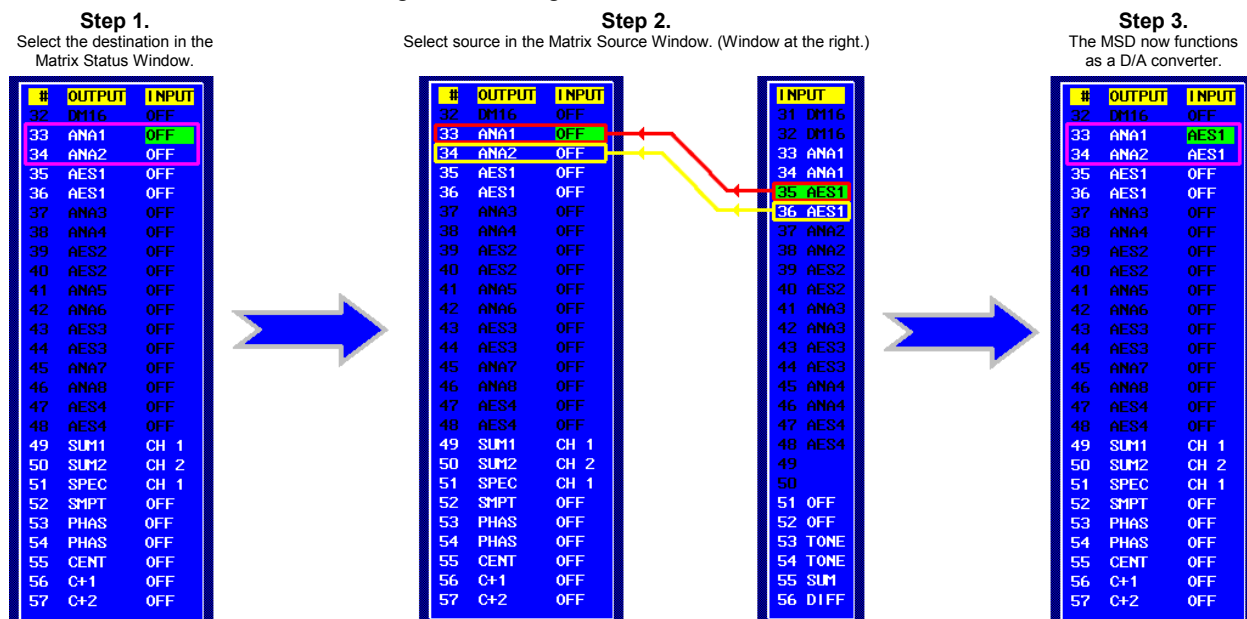
This example can also be reversed so the MSD will function as an analogue to digital converter. To do this, select destination #35 and #36 in step 1, and in step 2 select source #33 and #34.

5.4.3.2   **Example 2 – Route signals to the PPM-bargraphs.**

To extend the setup in example 1 it could be interesting to monitor the digital input levels on the PPM-bargraphs.

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.

Step 1.	**Step 2.**	**Step 3.**
Select the destination in the Matrix Status Window.	Select source in the Matrix Source Window. (Window at the right.)	The AES3-Stream from the input module in slot #1 can now be monitored on the PPM-bargraphs.
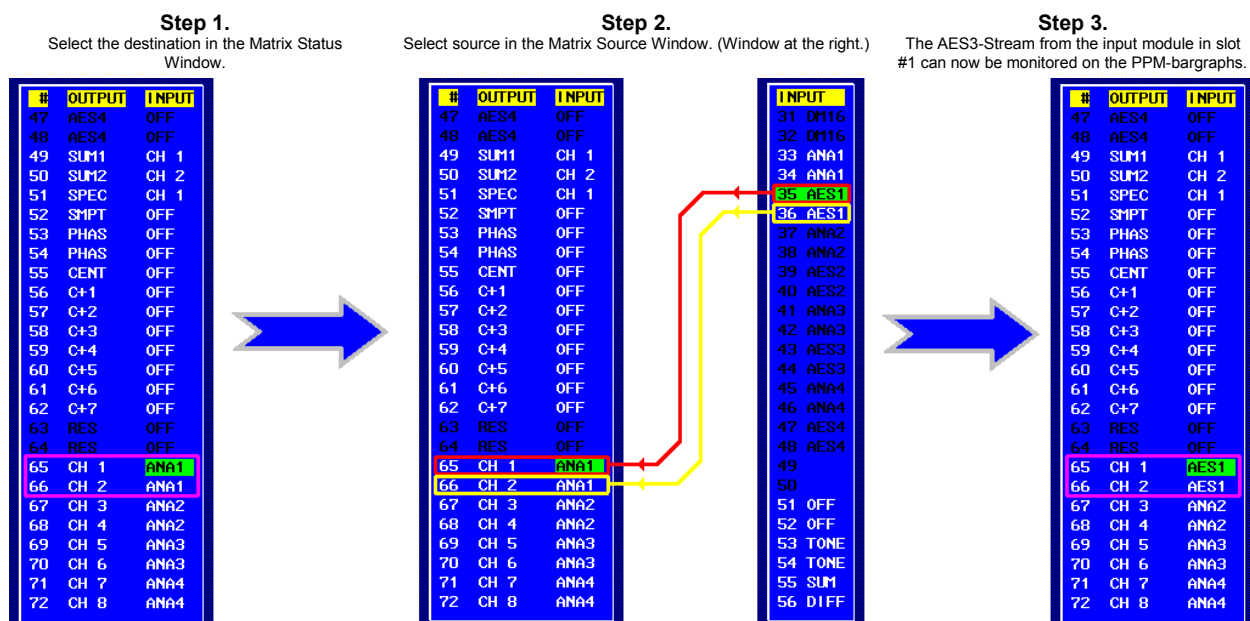
5.4.3.3   **Example 3 - The Audio Vector Oscilloscope.**

We can extend the setup in example 2 even further. To see the phase relationship between channel 1 and 2 in the AES-3 stream we can route these channels to the Audio Vector Oscilloscope.

Step 1.

In the Matrix Status Window, use the **[↑]** / **[↓]** or **[page up]** / **[page down]** keys to move the green cursor to destination #53 which is the left channel in the Audio Vector Oscilloscope.

Step 2:

To route the digital signal from the AES3-stream to this destination press the **[SELECT]** key and the Matrix Source Window will appear.

In example 2 the AES3-stream was routed to PPM-bargraph 1 and 2. This makes it possible select the AES3-stream from two different sources in the audio matrix.

Option 1: Select the AES3-stream directly from the Input module as it was done in example 2. (Source #35 and #36).

Option 2: Since a PPM-bargraph can function both as a destination and a source, any signal routed to a PPM-bargraph can be extracted from the PPM-bargraph and reused in the audio matrix.

The result is that it is possible to select the same AES3-stream which is available at the source numbers described in option 1, are also available at source #57 and 58 which is the output from the PPM-bargraph #1 and #2.

The advantage of using a PPM-bargraph as a source for the **'Audio Vector Oscilloscope'** is that if it was necessary to change the source for the PPM-bargraphs, the **'Audio Vector Oscilloscope'** would automatically show the new signal applied to the PPM-bargraphs.

In this example we will go with option 2. In the **'Matrix Source Window'** move the green cursor to source #57. Press **[SELECT]** to set the X-point. The first channel in the digital AES-3 stream has now been routed to the left channel in the **'Audio Vector Oscilloscope'**.

It is now necessary to route the second channel in the digital AES-3 stream to the right channel in the **'Audio Vector Oscilloscope'**, proceed as in step 1 and 2 but select destination #54 and source #58 instead.

Step 3:

When both channels have been routed, the MSD will now show the digital AES-3 stream on the **'Audio Vector Oscilloscope'**.

Step 1.	**Step 2.**	**Step 3.**
Select the destination in the Matrix Status Window.	Select source in the Matrix Source Window. (Window at the right.)	PPM channels #1 and #2 is now monitored by the **'Audio Vector Oscilloscope'**.
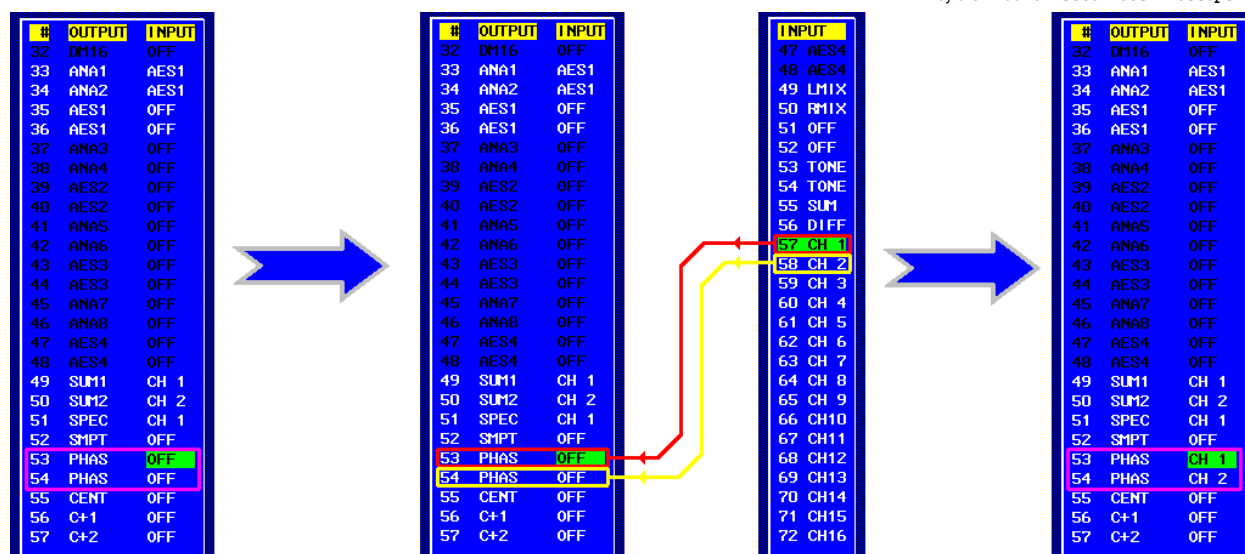
5.4.3.4   **Example 4. - Disabling outputs** *(Hiding a PPM-bargraph)*.

At some point it might be necessary to disable a destination. In this example a PPM-bargraph will be disabled, and by that, hiding it on the LCD screen.

The following procedure can be applied to all destinations in the audio matrix.

Step 1.
In the Matrix Status Window, use the **[↑]** / **[↓]** or **[page up]** / **[page down]** keys to move the green cursor to the destination that should be disabled. In this example destination #72, PPM-bargraph number 8.

Step 2:
To disable this destination press the **[SELECT]** key. The Matrix Source Window and the source select menu will appear.

The Matrix Source Select Menu.

There are two ways to disable the destination.
Option 1: The easiest way is to press the **[OFF]** key. When this key is pressed source #51 (Off) is automatically routed to the selected destination.

Option 2: The other way to disable the destination is to use the **[↑]** / **[↓]** or **[page up]** / **[page down]** keys to move the green cursor to source #51 or #52 and then press the **[SELECT]** key. There is no difference between source #51 and # 52.

Step 3:
The destination is now disabled. In this case the PPM-bargraph is hidden on the LCD screen. If there are one or more active PPM-bargraphs at the right of the PPM-bargraph, it will not be hidden but will still be disabled.

Step 1.
Select the destination in the Matrix Status Window.

Step 2.
Select source #51 or #52 in the Matrix Source Window or press the **[Off]** key in the 'The Matrix Source Select Menu'.

Step 3.
The PPM-bargraph has been hidden.

5.5   **The Extended Audio Matrix.** *(Matrix Detailed)* 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 Audio 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. e.g. analogue, digital, Jelly Fish™.
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 group output gain.

NO	TYPE	OUTPUT	INPUT	GAIN	GROUP	G-GAIN
33	ANAA	ANA1	ANA1	0.0	NONE	0.0
34	ANAA	ANA2	ANA1	0.0	NONE	0.0
35	DGTA	AES1	OFF	0.0	NONE	0.0
36	DGTA	AES1	OFF	0.0	NONE	0.0
37	ANAA	ANA3	OFF	0.0	NONE	0.0
38	ANAA	ANA4	OFF	0.0	NONE	0.0
39	DGTA	AES2	OFF	0.0	NONE	0.0
40	DGTA	AES2	OFF	0.0	NONE	0.0
41	ANAA	ANA5	OFF	0.0	NONE	0.0
42	ANAA	ANA6	OFF	0.0	NONE	0.0
43	DGTA	AES3	OFF	0.0	NONE	0.0
44	DGTA	AES3	OFF	0.0	NONE	0.0
45	ANAA	ANA7	OFF	0.0	NONE	0.0
46	ANAA	ANA8	OFF	0.0	NONE	0.0
47	DGTA	AES4	OFF	0.0	NONE	0.0
48	DGTA	AES4	OFF	0.0	NONE	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

| EXIT | ▲ | ▼ | ◄ | ► | SELECT | WHEL ▲▼ |

The Extended Matrix.

5.5.1   **Special option for the PT0660M Series.** WHEL ▲▼

The PT0660M and PT0660M-LS has a special selection wheel on the front panel. This wheel can be used to move the cursor around in the **'Extended Audio 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'** column the **'Matrix Source Window'** will open.

5.5.2   Set X-Point in 'Extended Audio Matrix'.

[MATRIX] ➜ [EXTEND] ➜ [▲] [▼] [◄] [►] [SELECT]

The 'Matrix Source Window'.

To set a X-Point in the `'Extended Audio Matrix'` use the **[↑]** / **[↓]** keys to move the green cursor to the `'Input'` column. Then 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'` is 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.

Please note that the MSD Write Protection must be enabled, otherwise the **[Select]** key will be replaced by a **[Edit]** key that will make it possible to rename the selected source.

5.5.3   Rename a Source or Destination.

[MATRIX] ➜ [EXTEND] ➜ [▲] [▼] [◄] [►] [EDIT]

The top left corner of the extended matrix window in edit mode.

To rename a source or destination in the `'Extended Audio Matrix'` use the **[>]** / **[◁]** keys to move the green cursor to the `'Input'` or `'Output'` column. By pressing the **[Edit]** key, the edit menu will appear and the green cursor will become pink showing a flashing cursor below the first character in the name.

[EXIT] [▲] [▼] [◄] [►] [COPY] [PASTE] [ENTER]

To select a new character at the current cursor position use the **[↑]** / **[↓]** keys to step through the different characters. Use the **[>]** / **[◁]** keys to move the cursor to the next position in the name. The name can consist of up to 4 alpha-numeric characters.

Use the **[Copy]** key to copy the entire name into memory, and then at a later time use the **[Paste]** key to paste it into another name.

To save the changes to the 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.

To abandon the changes press **[Exit]**.

5.5.3   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. For information about the `'Matrix Source Window'` see section 5.5.2.

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. 5.5.3.1.

2 Channel AES Input Module (22) – AES Input Options. Section 5.5.3.1.

4 Channel AES Input Module (24) – AES Input Options. Section 5.5.3.1.

SDI De-Embedding Module (A2) – SDI Input Options. Section 5.5.3.2.

5.5.3.1   AES Input Options. `MATRIX` ➜ `EXTEND` ➜ `OPTION`

```
AES INPUT OPTIONS:
□ SRC BYPASS   48KHZ

AES RECIEVER STATUS:
■ AES VALID
AES RECIEVER ERROR:
□ PHASE LOCK
□ INCOMING DATA VALID
□ CONFIDENCE ERROR
□ BIPHASE CODING
□ COMPUTED PARITY
AES CHANNEL STATUS:
PROFESSIONAL AUDIO
EMPHASIS       NON
LOCK           NOT INDIC
SAMPLE FREQ    48 KHZ
CH MODE        2 CHANNEL
USER BIT       NOT INDIC
AUX BIT        MAIN AUDIO
WORD LENGTH    24 BIT
ALIGN LEVEL    NOT INDIC
AUDIO REF      NO REF
ORIGIN         CH 1
DESTINATION    AES1

SAMPLE ADR     00:00:00
TIME OF DAY    00:00:00
```

The AES Option Window.

EXIT	BYPASS	SYNC		CLEAR

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.

5.5.3.1.1   SRC Bypass and External Sync. `BYPASS`   `SYNC`

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 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.

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 **[OPTION]** key to open the `'AES Input Option Window'`, in this window press the **[BYPASS]** key to bypass the SRC.

The MSD will be delivered from the factory with the SRC enabled.

The input modules MSD600M-INPUT/1 or MSD600M-INPUT/2 can external synchronise the MSD by pressing the **[SYNC]** key.

5.5.3.2   SDI Input Options. `MATRIX` ➔ `EXTEND` ➔ `OPTION`

```
SD-SDI INPUT OPTIONS:
SYSTEM:        AUTO
AUDIO GROUP: CH 1 - 4


SD-SDI INPUT STATUS:
■ SDI LOCK: 525 LINES
■ SDI AUDIO LOCK

■ AUDIO CH 1 - 2 LOCK
■ AUDIO CH 3 - 4 LOCK
```
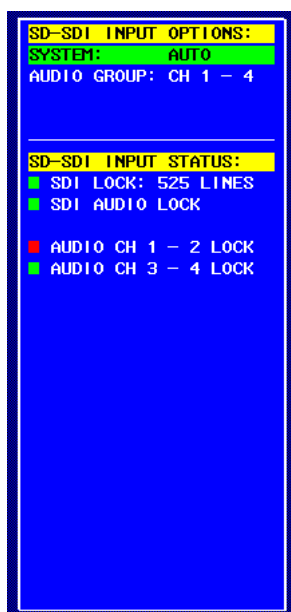
The SDI Option Window.

If the MSD is fitted with a SDI De-Embedding module it is possible to extract (de-embed) the audio from the SDI signal 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.

5.5.3.2.1   SDI System select. `OPTION` ➔ `SYSTEM:        AUTO`

```
SYSTEM
AUTO
525
625
```
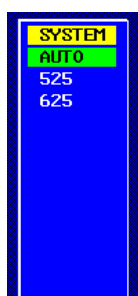
The SD-System Select Window.

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.

5.5.3.2.2   **SDI Audio Group selection.** `OPTION` ➔ `AUDIO GROUP: CH 1 - 4`
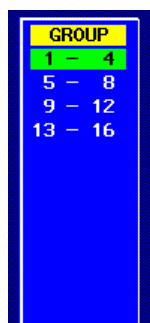
The SD-Group Select Window.

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 in the '`SDI Input Options Window`'. 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.5.5.4   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 **[↑]** / **[↓]** 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 destination.

5.5.4.1   **Bargraph (PPM) Options.** `OPTION`

The upper part of the bargraph options window.

To change the options for 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 eight 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 selected 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.
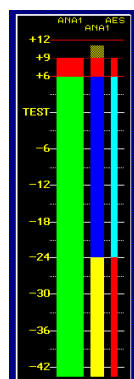
Three PPM bargraphs with different widths and colours. The middle PPM bargraph has the 'Track' colour set to yellow.

'`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 window is refreshed or the **[CLEAR]** key in the **[METER]** menu is pressed. Please note that this option is not available if the flex-mode *(section 5.5.4.1.1)* is active.

5.5.4.1.1   Dual Ballistics. *(Flex Mode)* `OPTION` ➜ `SET DYNAM`

As described briefly in *(section 5.5.4.1)* each PPM-bargraph can have its own ballistic properties, independent of the current scale. These properties are set in the menu `'Set Dynam'`.

Move the green selection bar to `'Set Dynam'` and press the **[Select]** key. The `'Dual Ballistics'` menu will appear.

From this menu 6 different options is available.

Available options in the `'Dual Ballistics'` **menu.**	
None	The ballistics for the PPM follows the default for the selected scale.
I	The ballistics follows IEC 268-10 I.
IIA	The ballistics follows IEC 268-10 IIA.
DIN	The ballistics follows DIN45XXX.
PEAK	The PPM shows digital true peak.
MCDY	The ballistics follows IEC 268-17 VU. *(This makes it possible to emulate the McCurdy ATS-100 meter in the PPM-window.)*

5.5.4.2   Audio Vector Oscilloscope Options. `OPTION`

`| EXIT | □ SLOW | ■ FAST |          | □ C-OFF | ▲ | 00 | ▼ | SCOPCOL |`

To change the settings for the `'Audio Vector Oscilloscope'`, move the green selection cursor to destination number #53 or #54 in the column `'OUTPUT'` and press **[OPTION]**.

5.5.4.2.1   Meter integration time. `OPTION` ➔ `■ SLOW` `□ FAST`

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

5.5.4.2.2   Meter Compressor. `OPTION` ➔ `□ C-OFF` `GAIN ↑ 12 ↓`

To obtain the best visual indication of signals in the `'Audio Vector Oscilloscope'` JellyFish™ / StarFish™ the `'Audio Vector Oscilloscope'` JellyFish™ / StarFish™ uses dynamic scaling. This is done by adjusting the input gain right before the metering function. The auto-adjustment is done so that the average input level 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.

5.5.4.2.3   Scope Colour. `OPTION` ➔ `SCOPCOL`

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

`| EXIT | □ | ■ | □ | □ | □ | □ | □ |`

The colour selection menu.

5.5.4.3   JellyFish™ / StarFish™ Options. `OPTION`

`| EXIT | □ SLOW | ■ FAST | □ JELLY | ■ STAR | □ C-OFF | ▲ | 00 | ▼ |`

To change the settings for the JellyFish™ / StarFish™, select a destination number between #55 and #62 in the column `'OUTPUT'` and press **[OPTION]**.

Please see section 5.5.4.2.1 for information about the **[SLOW]** and **[FAST]** keys (Meter integration time).

Please see section 5.5.4.2.2 for information about the **[C-OFF]** key. (Meter Compressor.)

5.5.4.3.1   Select JellyFish™ or StarFish™. `OPTION` ➔ `□ JELLY` `■ STAR`

The **[JELLY]** and **[STAR]** keys select between the JellyFish™ and StarFish™ mode for the surround sound oscilloscope. StarFish™ is the default setting.

5.5.5   **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 cannot be assigned to the digital group and vice versa.

5.5.5.1   **Assign Audio Group.** `OPTION`

| EXIT | ▲ | ▼ | | SELECT |

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 then 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.

5.5.5.2.1   **Adjust the Output Gain for a single output.** `OPTION`

| EXIT | | | dBu | ▲ | .0 | ▼ |

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 **[↑]** / **[↓]** 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.

5.5.5.2.2   **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 **[↑]** / **[↓]** 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 **'MSD Native Style'** will adjust the group gain for the Master group. While adjusting the group gain a small window on the screen will show the setting.

5.5.5.3   **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.
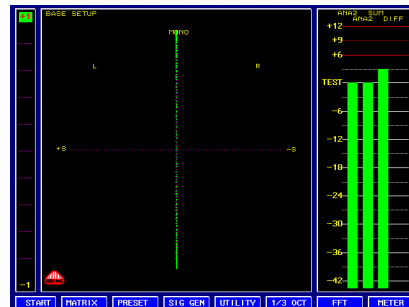
In the illustration above a 1 KHz sine wave with a phase difference of **0** degrees is applied to channel 1 and 2. When the signal is completely in phase the **SUM** of the signal will be raised.
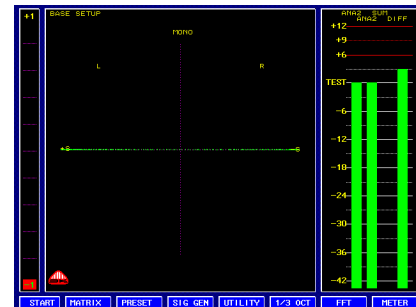
In the illustration above a 1 KHz sine wave with a phase difference of **180** degrees is applied to channel 1 and 2. When the signal is completely out of phase the **DIFFERENCE** of the signal will be raised.

Please see section 5.5.3.2 for information about setting the output gain for the sum and difference amplifier.

Please refer to section 5.1 and 5.2.1 for information on how to use the `'Phase Correlation Meter'` and the `'Audio Vector Oscilloscope'` to show the phase relationship between two channels.

5.5.5.3.1   **Routing audio to the SUM and Difference amplifier.**

Audio signals to and from the `'Sum and Difference Amplifier'` is routed using the `'Audio Matrix'` *(Compact or Extended)*. The `'Sum and Difference Amplifier'` has two inputs and two outputs. The inputs to the amplifier are found in the `'Matrix Destination Window'` as destination #49 `'SUM1'` and #50 `'SUM2'`.' The outputs from the `'Sum and Difference Amplifier'` are found in the `'Matrix Source Window'` as sources #55 `'SUM'` and #56 `'DIFF'`.
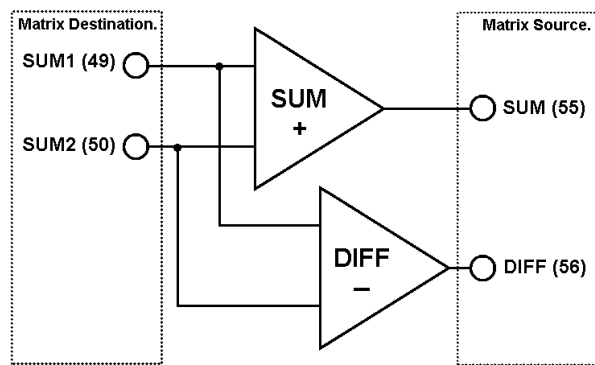
Equivalent schematic of the 'Sum and Difference Amplifier'.

5.5.5.3.2   **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 be amplified 3 dB if the signals are of the same level and phase.

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 –6,0 and +6,0 dB.

If the amplification in the `'Sum and Difference Amplifier'` should be lowered from +6dB to +3dB, 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.

6   Audio Utilities.

6.1   The Signal Generator. `SIG GEN`

`EXIT` `☐ OFF` `■ SINE` `☐ PULSE` `☐ EBU` `☐ WHITE` `☐ PINK` `☐ RVERS`

The `'Signal Generators'` menu controls all signal generator functions. To access this menu press **[SIG GEN]** in `'MSD Native Style'` or select `'Signal Generators'` from the `'Application Start List'`.

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

The `'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 outputs from the `'Signal Generator'` are found in the `'Matrix Source Window'` in row number #53 and #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.

6.1.1   Disable The Signal Generator. `SIG GEN` ➔ `☐ OFF`

To disable the signal generator select the **[OFF]** key.

6.1.2   Sine Wave Test Tone Generator. `SIG GEN` ➔ `■ SINE`

`EXIT` `■ x10` `HERTZ` `▲` `1000` `▼` `dBFS` `▲` `-18.0` `▼`

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 dBFS) are made in steps of +/-0.1dB, but adjustment speed can be increased by a factor of 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.

6.1.3   Pulse Modulation Tone Generator. `SIG GEN` ➔ `☐ PULSE`

`EXIT`   `PULSE` `▲` `250` `▼` `PAUSE` `▲` `250` `▼`

The PULSE function changes the sine wave signal from the first channel (#53) from the `'Sine Wave Test Tone Generator'` into a 'pulse' (chopped sine wave). The second channel (#54) from the `'Sine Wave Test Tone 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 and the frequency for the pulse signal are adjusted in the **[SINE]** menu. Please refer to section 6.1.2 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.

6.1.4   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 6.1.2 on how to do this.

6.1.5   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 6.1.2 for information on how to do this.

6.1.6   Reverse the phase of the Signal Generator. `SIG GEN` ➔ `☐ RVERS`

The **[RVERS]** key will when active reverse the phase 180° of source #54. *(Right output from the signal generator)*.

6.2   **DCF-77 Real Time Clock.** [START] ➔ [DCF77 RADIO 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 selected 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 current time and date in the topmost window in the center of the screen. 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 DCF-77 Realtime Clock menu is exited after synchronization, the MSD will still show the correct time. The MSD can only be synchronized while the DCF-77 Real Time Clock menu is selected. The MSD has also included two stopwatches which is controlled using the **[START]**, **[STOP]** and **[RESET]** keys.
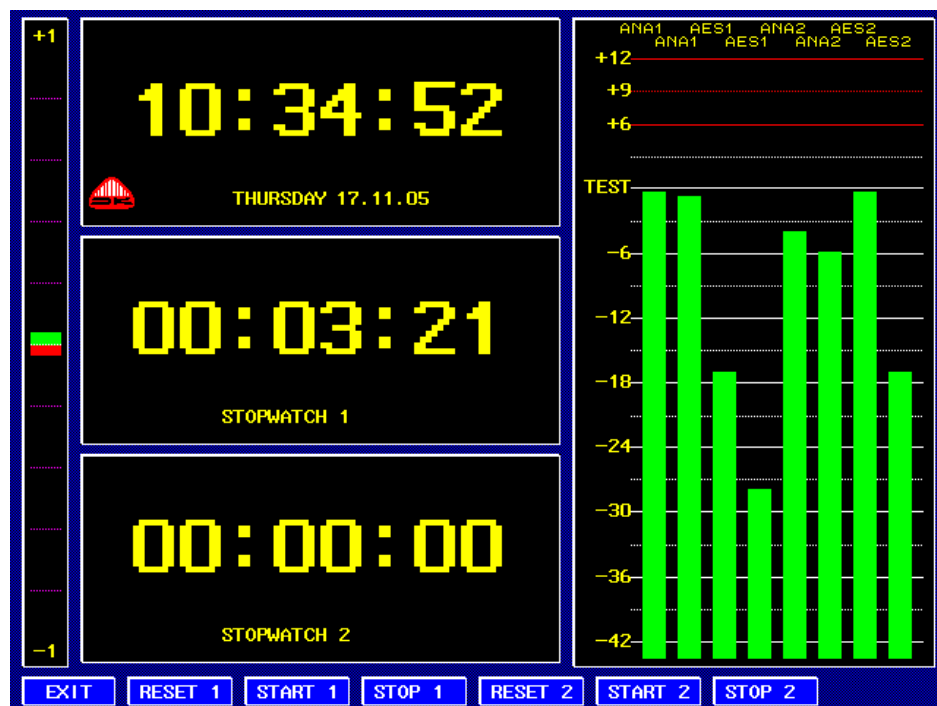
6.3   SMPTE Timecode Reader.

In the MSD there is incorporated a SMPTE timecode reader that supports the most common SMPTE / EBU timecode formats.

To enable the timecode reader an audio signal containing a valid SMPTE timecode must be connected to an analogue input on the MSD, and a x-point must be set for destination #52 (SMPT) in the audio matrix. See section 5.4 **'Compact Audio Matrix'** for further information about configuring the audio matrix.

When a physical audio source has been routed to destination #52 (SMPT) in the audio matrix, the timecode window will in MSD Native Style become visible below the **'Audio Vector Oscilloscope'** / JellyFish™ / StarFish™. If destination #52 (SMPT) is set to **'OFF'** then the timecode reader is disabled and the time code window will be hidden.
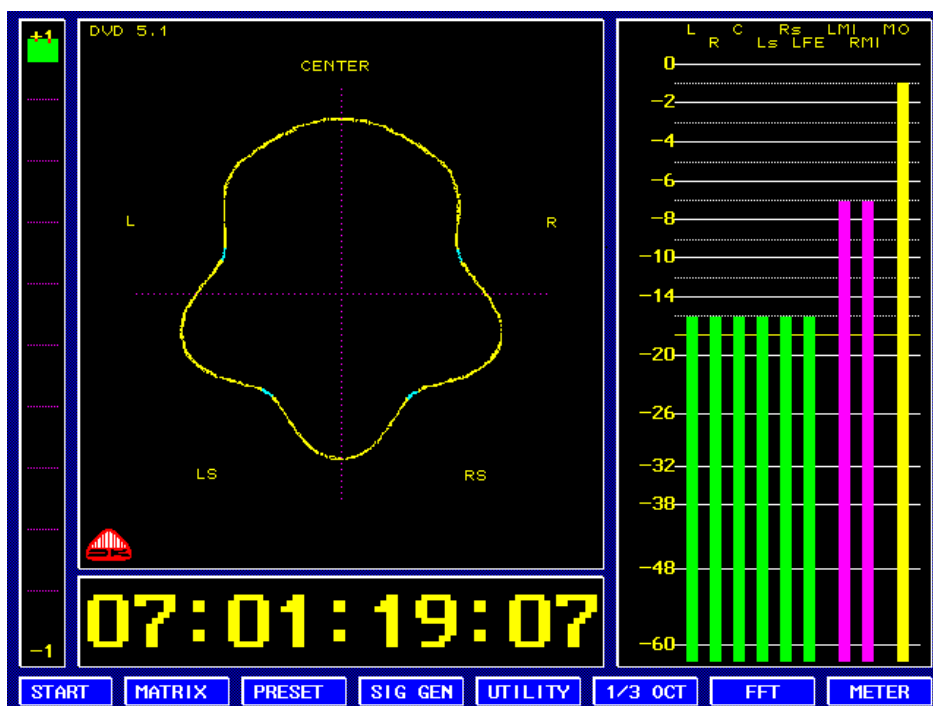
The SMPTE time code shown in a window below the StarFish™ window in the MSD Native Style.

The timecode signal can used by the **'Graphical Loudness'** application. The application will use an internal free-running timecode if no source is selected for the timecode reader. Please see section 8.2 and 8.3 for further details.

This timecode signal can also be used in conjunction with the Windows® software DK-Levelread. DK-Levelread is a programme that is capable of monitoring audio levels of up to 16 audio channels and generating a log file that can be opened either by the Adobe® Reader® or Microsoft® Excel.
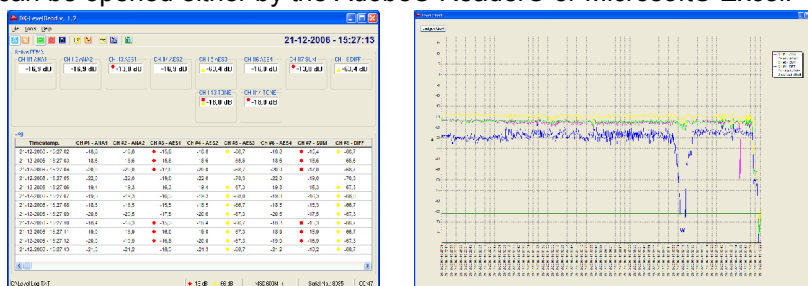
Screenshots of the Windows® software DK-LevelRead.

6.4 BLITS (Black & Lane's Ident. Tones for Surround)

BLITS (Black & Lane's Identification Tones for Surround) provides channel identification tones for all channels within a 5.1 Surround Sound signal.
This also helps to provide information in a stereo downmix – channel presence/ absence. The BLITS tones can be used at the start of a programme to help identify channels, in OB trucks for channel identification back to the studio and MCR as well as lineup for storage devices.

The frequencies used are based on the international music standard and are L=880Hz, R=880Hz, C=1,320Hz, LFE=82.5Hz, Ls=660Hz & Rs=660Hz.
The arrangement of the tones is designed to provide sequential and easy to read displays on bargraph meters.

It also provides additional controls to enable the user to change the frequency, the level and the duration of each step.
If the sequence is modified it can be saved to a Preset in the meter.

Please note that the BLITS Generator needs output modules installed in the MSD to work.
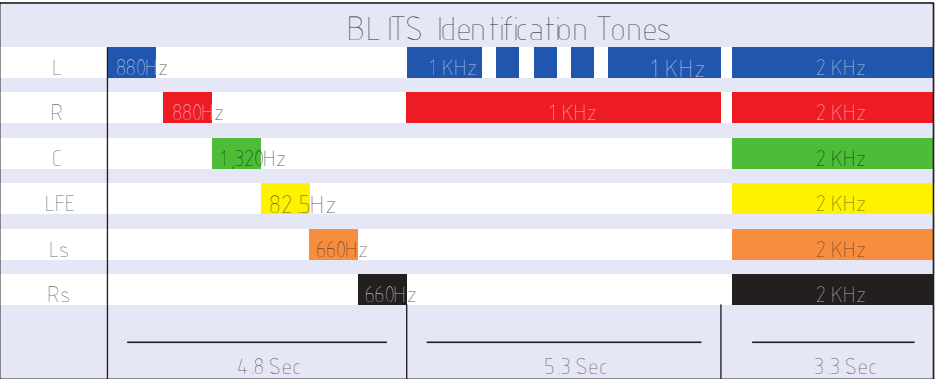
The BLITS sequence.

6.4.1 Configuring the BLITS Generator [EDIT]

BLITS generator in edit mode with the first channel selected.

The BLITS Generator is configured to use the AES outputs as default, but can easily be changed to use the analogue outputs.

Please note that it is only possible to change the settings in the BLITS generator if the **[ENABLE]** flag in the About menu is enabled.

To change a channel move the cursor to the channel and use the **[↑]MODIFY[↓]** to change output channel. Press the **[SAVE]** key to save the settings.

It is also possible to modify each step in the sequence. To do this move the cursor to the step your want to change (1 to 22) with the keys **[↑] / [↓]**. Then place the cursor on the item you want to change ('TIME S', 'FREQ HZ' , 'LEVEL', 'LEFT', 'RIGHT', 'CNTR', 'LFE', 'LSUR', 'RSUR', 'AUX1' or 'AUX2') with the **[←] / [→]** keys.
Use the modify keys **[↑]MODIFY[↓]** to change the value and press **[SAVE]** to save the changes.

6.4.2   Operating the BLITS Generator.

NO	TIME S	FREQ HZ	LEVEL	LEFT AES1	RHGT AES1	CNTR AES2	LFE AES2	LSUR AES3	RSUR AES3	AUX1 AES4	AUX2 AES4
01	.6	880	−18dBFS	ON	..	..	..	..	..	..	..
02	.2	880	−18dBFS	..	..	..	..	..	..	..	..

EXIT   EDIT   START   STOP

The BLITS generator is started by pressing the **[START]** key and will run in a loop until the **[STOP]** key is pressed.

When the **[STOP]** key is pressed the generator will finish the ongoing sequence.

If only one sequence is needed the **[STOP]** key should be pressed just after the **[START]** key.

7   Audio Analysers.

The MSD contains two types of Spectrum Analysers. A 1/3 Octave Spectrum Analyser and a FFT Spectrum Analyser.

7.1   1/3 Octave Spectrum Analyser. `1/3 OCT`

The 1/3 Octave Analyser with 'Track' activated. (Section 7.1.1.2)
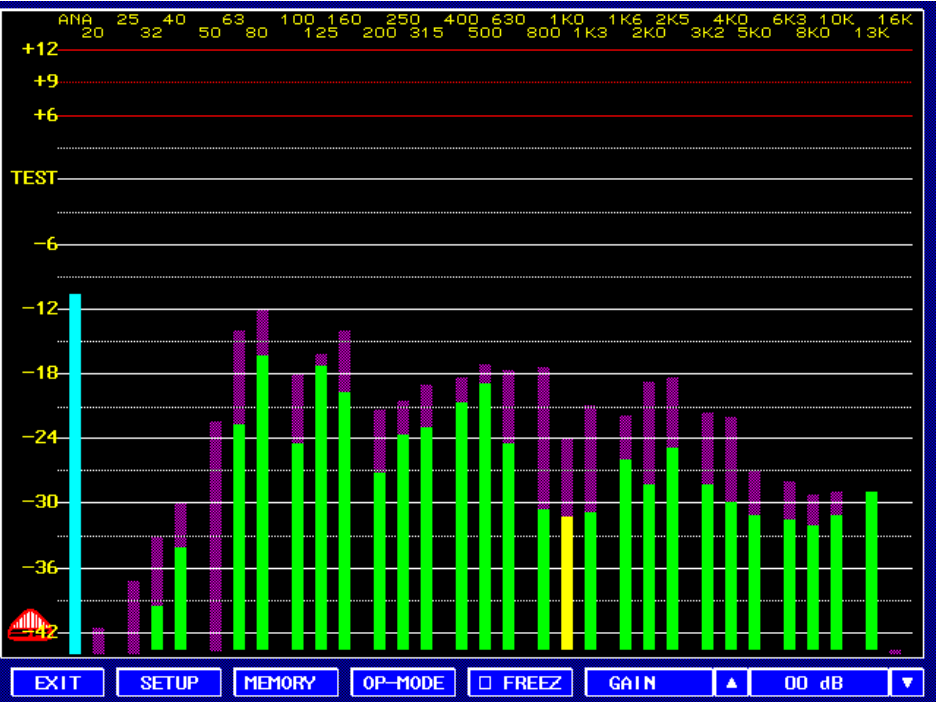
In the analysis of sound, it has always been important to describe the frequency content of a signal. Traditionally, this has been done using a filter bank containing a number of filters with relative bandwidths, for example 1/3 Octave. The signal being filtered passes through one filter at a time. Once 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.

7.1.1   Routing Audio to the Analyser Applications.

A Section of the Matrix Status Window with CH 1 (PPM bargraph #1) selected as source for the Analyser Applications.
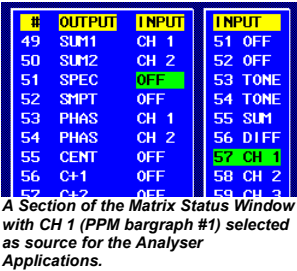
Both Spectrum Analysers (1/3 Octave and FFT) available in the MSD share the same destination in the Audio Matrix (#51 Spec). In that way when toggling between the 1/3 Octave and the FFT analyser it is not needed to set-up the Matrix more than once.

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

7.1.2   **Ballistic Setup for the 1/3 Octave Analyser.** `1/3 OCT` ➔ `SETUP`

`EXIT` `RESPONS`   `☐ TRACK` `■ PEAK` `☐ HOLD` `CLEAR`

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'.

7.1.2.1   **Response time for the 1/3 Octave Analyser.**
`1/3 OCT` ➔ `SETUP` ➔ `RESPONS`

`EXIT`   `TIME` `▲` `125 MS` `▼`

The analyser 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.

7.1.2.2   **Measuring modes.**
`1/3 OCT` ➔ `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 enabled 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 has 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.

7.1.3   **Measurement 'Snap-Shot's' of the 1/3 Octave Analyser.**
`1/3 OCT` ➔ `SETUP` ➔ `OP-MODE`

`EXIT` `■ MEM 1` `☐ MEM 2`   `SAVE` `RESET` `PRESET`

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.

7.1.4   Operation Modes for the 1/3 Octave Analyser.

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.

7.1.5   Using the Freeze Function. `1/3 OCT` ➜ `☐ FREEZ`

To momentarily freeze the Analyser display the **[FREEZ]** key can be selected. This function can be very useful when an interrupted Input signal has 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.

7.1.6   Using the Gain Function. `1/3 OCT` ➜ `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 5.1.3.1 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.

7.2   FFT Spectrum Analyser. `FFT`
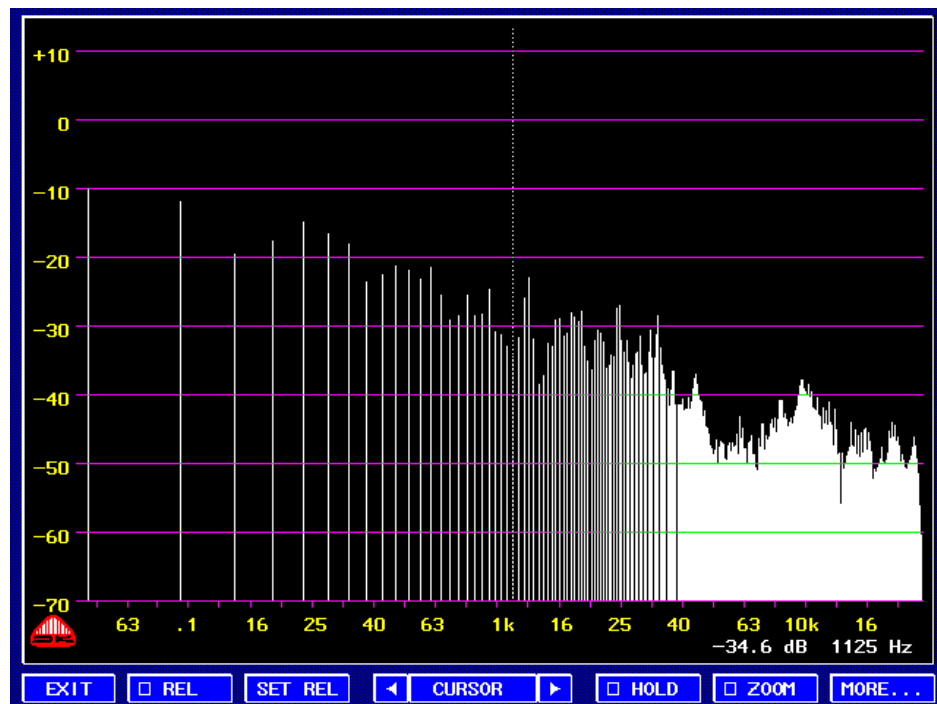
Most spectrum analysers used in sound engineering are based on analogue filtering techniques. For many years this has been the only practical solution for real-time analysers at a sensible size and cost. However, modern Digital Signal Processing (DSP) used by the MSD Family have made it possible to implement 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 1024, 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 **'MSD Native Style'** or select the **'FFT-Analyser 1024 Points'** from the **'Application Start List'**.

Frequency is indicated at the bottom of the screen from 48Hz to 23.521kHz, with signal level indicated on the left hand side in 10dB intervals from -70dB to +10dB.

Please refer to section 7.1.1 'Routing Audio to the Analyser Applications.' for details about selecting the source signal for the Analyser in the Audio Matrix.

7.2.1   Rel. 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.

7.2.2   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.

7.2.3   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.

7.2.4   Relative Memory FFT Measurements.
` FFT ` ➔ ` MORE... ` ➔ ` OP-MODE `

` EXIT ` ` □ I+MEM ` ` □ I+PRE `   ` SET MEM `

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-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 MSD Write Protection setting. If MSD Write Protection is active the **[SET MEM]** can only be used for temporary storage. The Analyser will lose 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.

7.2.5   FFT Window Functions. `FFT` ➔ `MORE...` ➔ `WINDOW`

`EXIT` `☐ RECT` `■ HANN` `☐ HAMM` `☐ NUTT`

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.
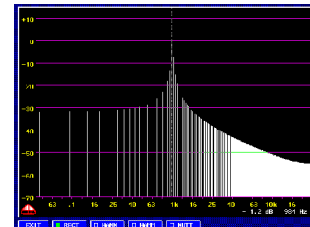
Hanning Window. `☐ HANN`
The Hanning Window is useful in most applications and should be selected as the default. The Hanning Window provides high dynamic range, but low selectivity.
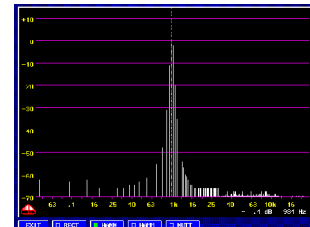
Hamming Window. `☐ HAMM`
The Hamming Window is ideal for applications where the requirement for the dynamic range is known. The Hamming Window compromises between selectivity and dynamic range.
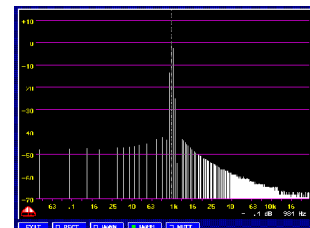
Nuttall Window. `☐ NUTT`
The Nuttall Window is especially useful for distortion measurements. The Nuttall Window provides the highest dynamic range and compromises the selectivity.
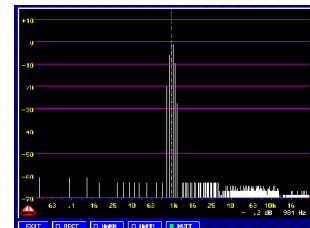
The recommended window function for most applications is the HANNING window.

8   Audio Level Tools.

Audio Level Tools is a group of functions that extends the functionality of the MSD. Please note that the functions found in this group only support up to 8 PPM channels.

These functions can also be found in the **[UTILITY]** menu in MSD Native Style

8.1.1   DMU1630 Digital Audio Peak Meter. START ➔ SONY DMU1630 EMULATION

| EXIT | ■ PEAK | □ HOLD | CLEAR | □ FINE | CLEAR | OVERS 0 |

The DMU1630 is a Digital True-Peak Meter function with a zoom function. It ranges of 0dBFs to -60dBFs.
DMU1630 is an emulation of the digital meter DMU1630 from Sony.

Just like any other scale selected from the **[SCALE]** menu the DMU1630 scale offers **[PEAK]**, **[HOLD]** and **[CLEAR]**. The DMU-30's **[CLEAR]** function is equal to the **[CLEAR]** function used by any of the 7 internal scales in the **[METER]** menu. Please refer section 5.1.3.3 and 5.1.3.4 for a description of these functions.

By pressing the **[FINE]** key the resolution of the DMU1630 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 DMU1630 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 8.1.2 for information on how to change the reference-level for the **[FINE]** mode.

The DMU1630 Scale with different signal levels.

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.
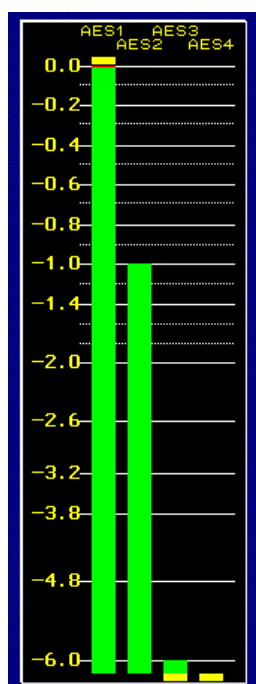
The DMU1630 Scale in FINE mode and the same signal levels as above.

8.1.2   Setting Reference and Overs for the DMU1630 Scale.

START ➔ SONY DMU1630 EMULATION

EXIT | REF LINE | ▲ | −18 dBFS | ▼ | ☐ FINE | OVERS | ▲ | 5 | ▼

Setting the DMU1630 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 3.2.1 on how to disable the MSD Write Protection.

When the MSD Write Protection are disabled the **[DMU1630]** item in the **'Application Start List'** menu will now access the DMU1630 Setup menu instead of the standard DMU1630 menu. To set the DMU1630 Reference level, indicated by a yellow line in the DMU1630 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.

The reference level is saved immediately when the DMU1630 Setup menu is exited. However the OVERS Reference count needs to be saved in the preset an can therefore be different in all the presets. Please see section 5.3.2 for further details about saving a preset.

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

When leaving the DMU1630 menu the MSD will change the scale back to the one selected in the **[METER]** / **[SCALE]** menu. To keep a DMU1630 PPM scale select the DMU-1 or DMU-2 scale from the **[SCALE]** menu. Please see section 5.1.3.1 for further details.

8.2   **RLB-LU METERING (RLB-LU)** `START` ➔ `RLB-LU METERING`

The ITU Recommendation BS.1770 specifies the algorithms to measure audio programme loudness. The RLB-LU application implements these algorithms to display the loudness of a Stereo signal as well as for a Surround Sound Signal.

Different working modes of the meter are selectable, including:
· Fast mode for real time control of loudness.
· Integrating mode for measuring the loudness of a recorded section or the complete recording.

Applications
There are many applications, where it is necessary to measure and control the perceived loudness of audio signals.
Examples of this include television and radio broadcast applications, where the nature and content of the programme material changes frequently. In these applications the audio content can continually switch between music, speech, and sound effects. Such changes in the content of the programme material can result in significant changes in subjective loudness.
Moreover, various forms of dynamic processing are frequently applied to the signals, which can have significant effect on the perceived loudness of the signal.
The matter of subjective loudness is also of great importance to the music industry and in production of commercials, where dynamic range processing is commonly used to maximize the perceived loudness.

Loudness Measurements
It is thus well recognized that loudness metering is required for broadcast applications.
The algorithm recommended by ITU and implemented in the RLB-LU application is based on an extension of the Leq(RLB) algorithm to cover stereo and multichannel audio signals. In this the Leq is an unweighted mean-square measure, while the RLB involves a high pass frequency weighting curve referred to as the Revised Low-frequency B-curve.
Loudness is a perceptual property of an audio signal, when it is reproduced acoustically. It is a complex non-linear function of amplitude, frequency and bandwidth.
Present audio level meters measure the level of audio signals expressed as the amplitude of the signal-either the rms voltage of an electrical signal or the sound pressure of an acoustical signal.
This level is an objective property, which is independent of frequency and bandwidth and is measured linearly in volts, if electrical, or Pascals, if acoustical.
For the purpose of broadcasting, loudness can also be measured as an electrical property, assuming a fixed electro acoustic gain for reproduction. This assumption is the basis for the broadcast loudness meter. The reproduction level, that has been assumed in the home, is 60 dBA, a level found to be typical for television viewing at home.

The RLB-LU application has two main operating modes for different purposes:

Fast mode for instantaneous monitoring of the loudness level. This mode is used in production, post-production and presentation. The programme level should be set so that on typical dialogue the meter displays 0 LU on average.

Integrating mode for a single figure overall loudness. This mode is used for quality control, mainly at programme ingestion, programme emission and in post-mortem analysis. The single number output from this mode allows clear and unambiguous information for loudness matching and gain setting. This mode makes it easy to secure consistent loudness over different programmes

and short sequences.

Important! You can set the 0 LU reference in the `'AUDIO GLOBALS'` Read more about this in section 4 The MSD Control Panel. In here you should also set the `'LEQ SUMMING METHOD'` to "ITU 400Hz" or to "ITU 1KHz" depending on what frequency you reference level is defined at.

RLB-LU Metering

The RLB-LU Metering function have been included in the software for MSD600M++ series since software version 5.3.

RLB label will change to red if peak overload.

Graphical representation of the Loudness (LU)

Scale in LU according to the ITU recommendation

LU (Loudness Unit), The colour indicates the sign of the Loudness. Purple is a value above 0 LU. Green is a value below 0 LU.
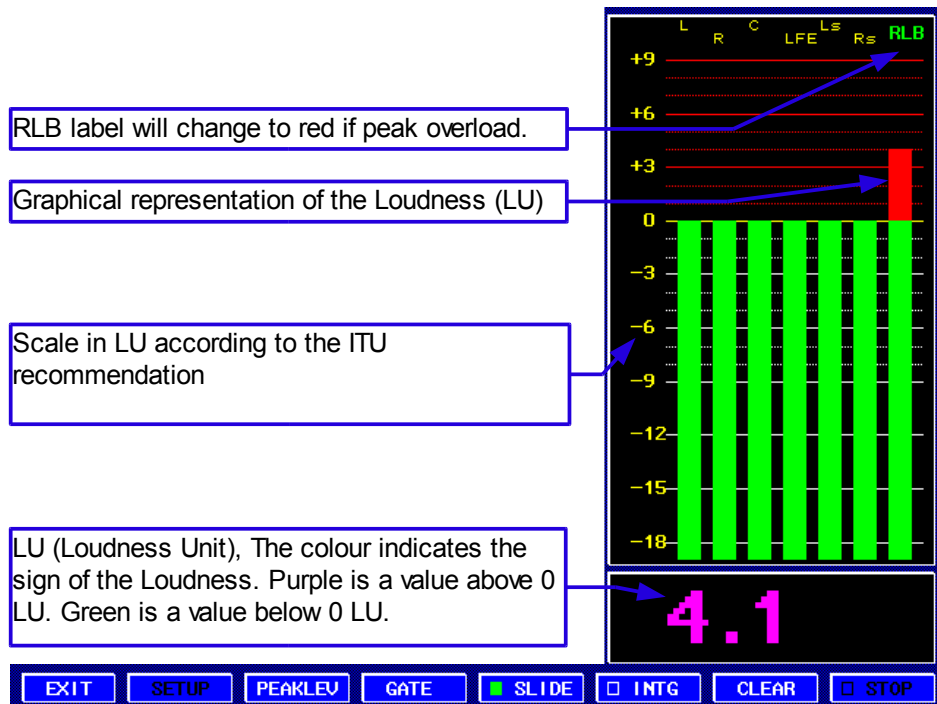
Fast Mode [SLIDE]
The Fast mode can be enabled by pressing the **[SLIDE]** key. This will also open the **[WINDOW]** menu where it is possible to set the width of the "sliding" window (Integration time) from 0 to 9 Seconds.

Integrating Mode [INTG]
The Integration mode can be enabled by pressing the **[INTG]** key. To start the measurement press the **[START]** key and to end it press the **[STOP]** key. The average Loudness from the start to stop will then be displayed.

Threshold [GATE]
If the audio is under the **[GATE]** level the calculation of the Loudness will stop and first start again if one of the channels exceeds the **[GATE]** level. This is use full when measuring e.g. Golf matches and other types of material with long pauses in the audio.

Peak Level [PEAKLEV]
The [PEAKLEV] is the level of the peak indicator (The RLB label on top of the Loudness bar). If a sample in one of the input channels exceeds the Peak Level the RLB label with change colour from green to black with red background. The level can be set between -2 and -9 dBFs.

Channel Setup Menu [SETUP]

This Menu is only active if the **[ENABLE]** in the **[ABOUT]** menu is enabled.
In this menu you need to specify the surround channels (Ls and Rs) if you are measuring the Loudness for a 5.1 surround signal. According to the ITU BS1770 recommendation the surround channels should be gained by approximately 1.5 dB.
In this menu you can also save the settings by pressing the [SAVE] key.

8.3   Graphical Loudness. `START` ➜ `GRAPHICAL LOUDNESS`

The Graphical Loudness is a function for short term Loudness measurements especially suitable for commercials and trailers.

The Graphical Loudness consists of four different windows.
The PPM/Graph window shows each channel as a bar and a sum bar shows the overall loudness. It also shows the graphical presentation of the loudness over time.

Under the PPM/Graph window are tree windows. The first is a watch showing the elapsed time.

The next one is the overall loudness measurement presented as a numerical number.

The last window shows witch Loudness mode (FLATRMA, LEQ-RLB. LEQ-M or LEQ-A) is chosen and when the measurement will start and stop or for how long the last measurement did run.

To learn more about the LEQ-RLB and the ITU recommendation read chapter 8.2 RLB LU metering.

`OP-MODE`

8.3.1   Operations Modes.

The Graphical Loudness can work in tree different modes, **[FLATRMS]**, **[LEQ-RLB]**, **[LEQ-M]** and **[LEQ-A]**.

The **[FLATRMS]** is a Loudness measurement with no weighting of the signal. It calculates the RMS of each channel. The Loudness is then the sum of all the channels.

The **[LEQ-RLB]** is a Loudness measurement conforming with the ITU BS1770 Integration mode.

The **[LEQ-M]** is a Loudness measurement with a CCIR-468 weighting of the signal. The **M** is short for movie and indicates that this mode is mainly used for movies, trailers and commercials.

The **[LEQ(A)]** is a Loudness measurement with a A weighting of the signal.

8.3.2   Channel order setup. `OP-MODE` ➔ `SETUP`

The **[SETUP]** key is only active if the **[ENABLE]** in the **[ABOUT]** menu is enabled.
In order to get the right Loudness measurement the gain/levels setting for each channel needs to be specified.
The menu works in different modes depending of the selected operation mode

8.3.2.1   Setup for FLATRMS

`EXIT`   `WEIGHT` `▲` `L` `▼` `0.0 dB`

Select the calibrations gain for each channel by pressing the arrows **[↑]** / **[↓]** to the right of **[WEIGHT]** and then select the gain you need for the chosen channel (**[0.0dB]**, **[+1.5dB]** or **[LFE 85]**).

A normal setting when using the FLATRMS:
L: `'0.0 dB'`
R: `'0.0 dB'`
C: `'0.0 dB'`
LFE: `'LFE 85'`
Ls: `'+1.5 dB'`
Rs: `'+1.5 dB'`

Please check with your specification when specifying the gain for each channel in your setup.

See section 4.1.4 for reference level setup.

8.3.2.2   Setup for LEQ-RLB

`EXIT`   `WEIGHT` `▲` `L` `▼` `0.0 dB`

Select the calibrations gain for each channel by pressing the arrows **[↑]** / **[↓]** to the right of **[WEIGHT]** and then select the gain you need for the chosen channel (**[0.0dB]** or **[+1.5dB]**).

A normal setting when using the LEQ-RLB:
L: `'0.0 dB'`
R: `'0.0 dB'`

C: **'0.0 dB'**
Ls: **'+1.5 dB'**
Rs: **'+1.5 dB'**
(LFE: **'0.0 dB')**

Please check with your specification when specifying the gain for each channel in your setup.

See section 4.1.5 for reference level setup.

8.3.2.3   Setup for LEQ-M

| EXIT | TRIP | ▲ | 85 dB | ▼ | CALIBRAT | ▲ | L | ▼ | LEV 85 |

Select the calibrations level for each channel by pressing the arrows **[↑]** / **[↓]** to the right of **[CALIBRAT]** and then select the level you need for the channel (**[LEV 85]**, **[LEV 82]** or **[LFE 85]**).

A normal setting when using the Leq(M) on surround sound:
L: **'LEV 85'**
R: **'LEV 85'**
C: **'LEV 85'**
LFE: **'LFE 85'**
Ls: **'LEV 82'**
Rs: **'LEV 82'**

Please check with your specification when specifying the calibration level for each channel in your setup.

The **[TRIP]** is the overload threshold indication level. When the measured Leq(M) value exceed the TRIP level the number in the numeric display will turn red. The TRIP level can be adjusted with the arrows **[↑]** / **[↓]** to the right of **[TRIP]**. The TRIP level can be set between 78 and 93. The default level is 85.

See section 4.1.6 for reference level setup.

8.3.2.3   Setup for LEQ-A.

| EXIT |

See section 4.1.7 for reference level setup.

8.3.2   Operating the Graphical Loudness.

| EXIT | OP-MODE | ■ DYNAM | □ REFON | SET REF | RESET | START | STOP |

The measurement starts when the **[START]** key is pressed and stops when the **[STOP]** key is pressed (or if it has been running to maximum measurement time. The maximum measurement time is longer than 170 seconds dependent on the number and width of the channels.)
The second time a measurement is started it will automatically stop at the same time as the previous measurement.
If the meter is fed with a SMPTE time code (LTC) signal it will start and stop at the same locations as the first measurement. See section 6.3 regarding the SMPTE time code decoder.

The saved start and stop locations can be reset by pressing the **[RESET]** key.

8.3.3 Save Reference Graph. `☐ REFON` `SET REF`

A loudness graph can be saved by pressing the **[SET REF]** key.
A saved graph can be `☐ DYNAM` shown as a white graph by activating the **[REFON]** (Reference   On) key.

8.3.4 Dynamic Graph.

The dynamic graph is the sum of each channel plotted each second. This graph can be shown by activating the **[DYNAM]** key.

Appendix A   Specifications.

A.1   MSD600M++ Base Unit.

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 (fall-back) 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 Vector Oscilloscope:
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
Safety: according to IEC 65

Environmental Conditions:
Temperature range: 0 ^{o}C to 45 ^{o}C

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

A.1.1   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:
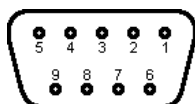
View from solder side of the 9 pin MALE D-Sub connector.

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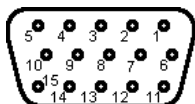
View from solder side of the 15 pin MALE High Density D-Sub connector.

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 Analogue and 1 AES 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.
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.

Module ID: 31.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

This is a ***4-Channel Input Module***.
Please see chapter 5.4.2 for information about how the channels will be assigned in the Audio Matrix.

Connector type: 15 Pin Female D-Sub.

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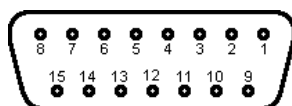
View from solder side of the 15 pin MALE D-Sub connector.

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 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 dBu
Sample rate range with external sync: 32 kHz to 50 kHz
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.

Module ID: 64.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

This is a *8-Channel Input Module*.
Please see chapter 5.4.2 for information about how the channels will be assigned in the Audio Matrix.

Connector type: 25 pin Female D-Sub

Pin Configuration:
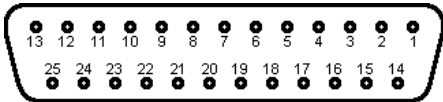
*View from solder side of the 25 pin
MALE D-Sub connector.*

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 AES3 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.

Module ID: 22.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

This is a *4-Channel Input Module*.
Please see chapter 5.4.2 for information about how the channels will be assigned in the Audio Matrix.

Connector type: 25 pin Female D-Sub

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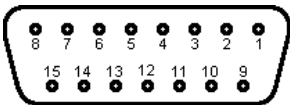
View from solder side of the 15 pin MALE D-Sub connector.

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 AES3 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.

Module ID: 24.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

This is a *8-Channel Input Module*.
Please see chapter 5.4.2 for information about how the channels will be assigned in the Audio Matrix.

Connector type: 15 pin Female D-Sub

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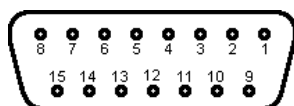
View from solder side of the 15 pin MALE D-Sub connector.

Signal Name.		Pin number.
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 De-embedding 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
De-embedding 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.*

Module ID: A2.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

This is a *4-Channel Input Module*.
Please see chapter 5.4.2 for information about how the channels will be assigned in the Audio Matrix.

Connector Type: 2 X BNC

A.1.6   2 Analogue and 1 AES3 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

Specifications:

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.

Module ID: 11.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

Connector type: 15 Pin Female D-Sub.
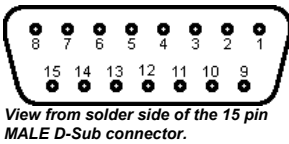
Pin Configuration:

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.1.7   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.


```
                                            +5V
                                            +3,5V("OFF" threshold)

                                            +1,5V("ON" threshold)
                                            Ground
```

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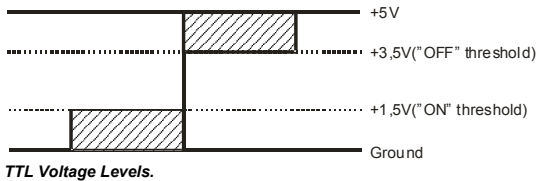
Possible placement: Output slot #4[*)].
[*)] *Only one General Purpose I/O module can be fitted in a MSD.*

Module ID: 10.
The module ID can be viewed in the 'Information Window' on the MSD. Please see section 3.2 for further information.

Connector type: 25 Pin Female D-Sub.

Pin Configuration:
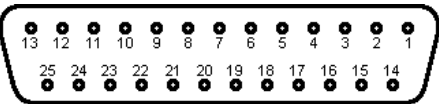
*View from solder side of the 25 pin
MALE D-Sub connector.*

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
DMU1630 Fine. *(See chapter 8.1.1)*	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
Peak Hold Reset	Input.	22
Unused.		9
Unused.		21
Unused.		8
DCF-77 Input. *(See chapter 6.2)*	Input.	19
Unused.		6
Unused.		7
Power +5V (Max. 100 mA.)	**Output**	1
Ground		14

Appendix 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 5.3.4 for information on how to select another default startup preset.

The base setup is configured to show all information from the input modules in slot #1 and #2. The **'Audio Vector Oscilloscope'** and **'Phase Correlation Meter'** 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 only analogue input modules *(ID 31)* 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
JellyFish™ / StarFish™	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 Oscilloscope'` and `'Phase Correlation Meter'` 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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
		**Preset #2 – All Analogue.**		
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
JellyFish™ / StarFish™	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 Oscilloscope'` and `'Phase Correlation Meter'` 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 only analogue input modules *(ID 31)* 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
JellyFish™ / StarFish™	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 Oscilloscope'` and `'Phase Correlation Meter'` 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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
Preset #4 – LR				
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
JellyFish™ / StarFish™	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 JellyFish™ / StarFish™ monitors a four channel surround sound signal applied as discrete channels to the first two analogue input modules. The **'Phase Correlation Meter'** monitors the PPM Channel 1 and 2.

The JellyFish™ / StarFish™ is configured using rows 55 to 62. In order to show the surround sound information, this setup uses five vectors in the JellyFish™ / StarFish™. The center channel 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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
		Preset #5 – LRCS		
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
JellyFish™ / StarFish™	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 JellyFish™ / StarFish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The **'Phase Correlation Meter'** monitors the PPM Channel 1 and 2.

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

The vectors in the JellyFish™ / StarFish™ 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 JellyFish™ / StarFish™.

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 only analogue input modules *(ID 31)* 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
JellyFish™ / StarFish™	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 JellyFish™ / StarFish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The **'Phase Correlation Meter'** monitors the PPM Channel 1 and 2.

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

The vectors in the JellyFish™ / StarFish™ 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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
Preset #7 – LRLsRsCCs				
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
JellyFish™ / StarFish™	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 JellyFish™ / StarFish™ monitors a 7.1 surround sound signal applied as discrete channels to the four analogue input modules. The **`'Phase Correlation Meter'`** monitors the PPM Channel 1 and 2.

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

The vectors in the JellyFish™ / StarFish™ 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 JellyFish™ / StarFish™.

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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
Preset #6 – LRCLsRs				
Phase	53	PHAS	57	CH 1
	54	PHAS	58	CH 2
JellyFish™ / StarFish™	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 JellyFish™ / StarFish™ monitors a 5.1 surround sound signal applied as discrete channels to the first three analogue input modules. The **'Phase Correlation Meter'** monitors the PPM Channel 1 and 2.

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

The vectors in the JellyFish™ / StarFish™ 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 JellyFish™ / StarFish™.

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 only analogue input modules *(ID 31)* then the preset is configured as the following table shows.

	Destination Number.	Destination Name.	Source Number.	Source Name.
Preset #9 – LRCLsRs				
Phase	53	PHAS	57	CH 1
Phase	54	PHAS	58	CH 2
JellyFish™ / StarFish™	55	CENT	59	CH 3
JellyFish™ / StarFish™	56	C + 1	58	CH 2
JellyFish™ / StarFish™	57	C + 2	62	CH 6
JellyFish™ / StarFish™	58	C + 3	61	CH 5
JellyFish™ / StarFish™	59	C + 4	57	CH 1
JellyFish™ / StarFish™	60	C + 5	51	OFF
JellyFish™ / StarFish™	61	C + 6	51	OFF
JellyFish™ / StarFish™	62	C + 7	51	OFF
Peak Programme Meter	65	CH 1	35	L
Peak Programme Meter	66	CH 2	36	R
Peak Programme Meter	67	CH 3	39	C
Peak Programme Meter	68	CH 4	40	LFE
Peak Programme Meter	69	CH 5	43	Ls
Peak Programme Meter	70	CH 6	44	Rs
Peak Programme Meter	71	CH 7	51	OFF
Peak Programme Meter	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 JellyFish™ / StarFish™ monitors a stereo signal applied to the first analogue input module. The **'Phase Correlation Meter'** monitors the PPM Channel 1 and 2.

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

The vectors in the JellyFish™ / StarFish™ 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 only analogue input modules *(ID 31)* 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
JellyFish™ / StarFish™	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).