DLM 260 / DLM 480

DIGITAL LOUDSPEAKER MANAGEMENT

OPERATING MANUAL

The DLM series are DSP based 2 inputs x 6 output for 260 model and 4 inputs x 8 output for 480 model digital loudspeaker management processor, ideally suited for fixed installations and live events.They combine the functions of a multitude of conventional products in a compact 1 U unit with extensive PC remote control capabilities.

Routing is freely assignable from the two/four inputs to the six/eight outputs. Input and output equalisation is available with a choice of 16 filter options. A total of 31 IIR EQ is available for each input or 13 IIR EQ + FIR; eight IIR EQ for each output or four IIR EQ + FIR is also available.Many options for crossover type include IIR from 6 to 48dB/oct, FIR and Linear Phase with constant group delay. Input and output delay is available to time align components in a system; RMS and Peak limiters will protect your speakers from damage due to excessive power being applied.

On input paths, the processing chain sees in cascade: input gain / delay / noise gate / dynamic loudness / filtering section / RMS compressor.

On output paths the processing chain sees in cascade: output gain / delay / polarity / filtering section / RMS compressor / peak limiter.

The processor features 32 user memory locations to store popular configurations and settings can be exported to a file via the dedicated PC application, either as a back-up or to store more favourite configurations if the 32 on-board presets are not sufficient.

A very useful function is the ability to recall entire output channel configurations and a comprehensive library of output files for FBT products is available making it very quick ans easy to create exactly the configuration you need. PC connectivity is available via a front panel USB port but for larger systems with several processors can use Ethernet or RS 485 network connections.

FEATURES:
- Professional 2/4-input, 6/8-output system processors for a wide range of system applications
- Comprehensive, sophisticated processing options including EQ, Gate, Delay and limiting to optimise and protect a system
- Fully assignable matrix routing of both inputs to any output with attenuation option on each routing mode
- Two/four balanced XLR inputs with up to 12dB of gain, polarity invert, 600ms of delay, 10 band parametric EQ with a choice of 16 filter types individually selectable for each band
- Six/eight balanced XLR outputs with 128dB gain, polarity invert, 200ms delay, high and low pass filter; 8 band parametric EQ with up a choice of 16 filter types, indipendent program and peak limiters
- Convenient front panel controls and LCD display
- One press mute buttons for all inputs and outputs
- Front panel USB port for quick and easy PC connection
- Networking option using either Ethernet or RS485 for larger systems
- Output channel preset import function to quickly and easily compile a system to suit any application utilising a library of FBT product channel presets
- 32 user presets to store system configurations
- Universal switch mode mains PSU accepts 90 to 240V - 50/60Hz

GENERAL INFORMATIONS

SUMMARY

Where affixed on the equipment or package, the barred waste bin sign indicates that the product must be separated from other waste at the end of its working life for disposal. At the end of use, the user must deliver the product to a suitable recycling centre or return it to the dealer when purchasing a new product. Adequate disposal of the decommissioned equipment for recycling, treatment and environmentally compatible disposal contributes in preventing potentially negative effects on the environment and healt and promotes the reuse and/or recycling of equipment materials. Abusive product disposal by the user is punishable by law with administrative sanctions.

MASTER CARTON

PAPER

PAPER DISPOSAL

20
PAP
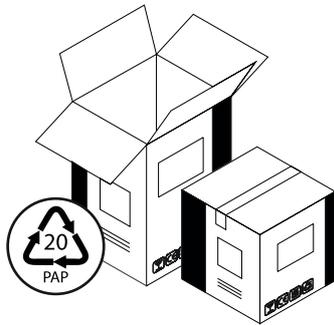
INTERNAL PROTECTIONS

POLYETHYLENE

PLASTIC DISPOSAL

4
LDPE
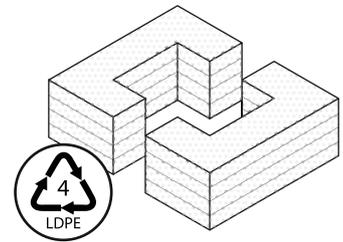
BAG

POLYETHYLENE

PLASTIC DISPOSAL

4
LDPE
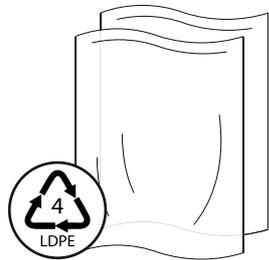
MANUAL

PAPER

PAPER DISPOSAL

22
PAP

Avoid release to the environment
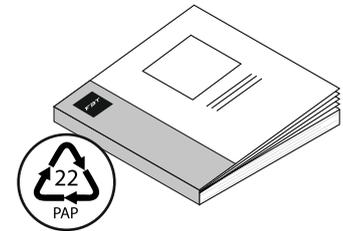
Check the instructions of your municipality
Separate the components and arrange them properly

IMPORTANT SAFETY INSTRUCTIONS

The lightning flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated dangerous voltage within the product's enclosurem, that may be of sufficient magnitude to constitute a risk of electric shock to persons.

The exclamation point within an equilateral triangle is intended to alert the user of the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

• Read these instructions
• Keep these instructions
• Heed all warnings
• Follow all instructions
• Do not use this apparatus near water
• Clean only with dry cloth
• Do not block any ventilation openings
• Install in accordance with the manufacturer's instructions
• Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat
• Do not defeat the safety purpose of the polarized or grounding type plug. A polarized plug has two blades with one wider than the other; a grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet
• Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles and the point where they exit from the apparatus
• Unplug this apparatus during lightning storms or when unused for long periods of time
• Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power supply cord or plug is damaged, liquid has been spilled or objects have fallen inot the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
• This apparatus shall not be exposed to dripping or splashing. No object filled with liquid such as vases, shall be placed on the apparatus
• Do not overload wall outlets and extension cords as this can result in a risk of fire or electric shock
• This apparatus has been designed with Class I construction and must be connected to a mains socket outlet with a protective earthing connection (the third grounding prong).
• This apparatus has been equipped with a rocker-style AC mains power switch. This switch is located on the rear panel and should remain readily accessible to the user
• The mains plug or an appliance coupler is used as the disconnect device, so the disconnect device shall remain readily operable

WARNING: To reduce the risk of fire or electric shock do not expose this appliance to rain or moisture.

CAUTION: Internal lithium battery. Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type.

WARNING: This equipment has been designed be installed by qualified professionals only ! there are many factors to be considered when installing professional sound reinforcement systems, including mechanical and electrical considerations, as well as acoustic coverage and performance. FBT strongly recommends that this equipment be installed only by a professional sound installer or contractor.

CAUTION

Risk of electric shock, do not open.
Risque de choc electrique ne pas ouvrir.

To reduce the risk of electric shock do not remove cover (or back), no user serviceable parts inside, refer servicing to qualified personnel.
Pour eviter les risques de choc electrique ne pas enlever le couvercle, aucun entretien de pieces interieures par l'usager. Confier l'entretien au personnel qualifie.

AC POWER INPUT

The AC voltage provided by the audio processor with built-in global voltage operation must be within ±10% of the specified line voltage (90/240V~). The third pin (grounding pin) on the power cord that comes with the package is a necessary safety component; please do not try to disable the ground connection by using an adapter or other methods.

SIGNAL INPUT AND OUTPUT

Analog signal input:
Like all digital signal processing equipment, the signal level supplied to the unit must be appropriate to avoid working in a low signal-to-noise ratio or distortion. The DLM series processors can accept up to +20dBu analog signal level, which is greater than the linear output level of most mixers, so the impact of this type of problem is reduced. When performing equalization processing, pay attention to the boosted gain value not to be too large, so as to avoid the gain of the unit from causing digital clipping (of course, you can use the built-in anti-clipping function to prevent this event from happening). It must be noted that the maximum input level in the specification is a clipping level, not a safe practical level. It must be ensured that the clipping level is not lower than the next device in the signal chain and a certain margin must be left during use.

For each input channel (A/B/C/D) is a female XLR connector. Even channel numbers are for analogue inputs only.
The HOT, "+" or "in phase" connection should be made to pin 2 of the XLR connector.
The COLD, "-" or "out of phase" connection should be made to pin 3 of the XLR connector.
Pin 1 of the XLR connector is internally connected to the chassis.

INPUT XLR BALANCED CONNECTION

PUSH

1 Ground,shielding
2 Signal (in phase +)
3 Signal (out of phase -)

INPUT XLR UNBALANCED CONNECTION

PUSH

1 — 3 Ground,shielding
2 Signal
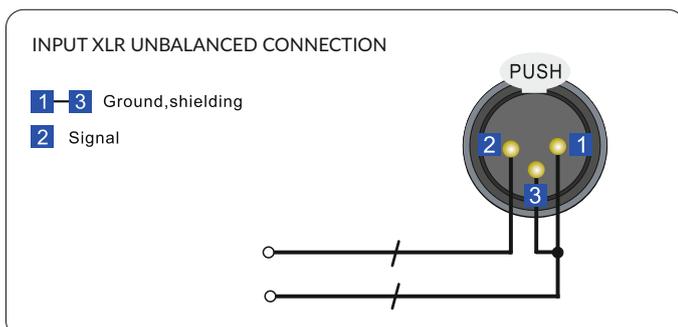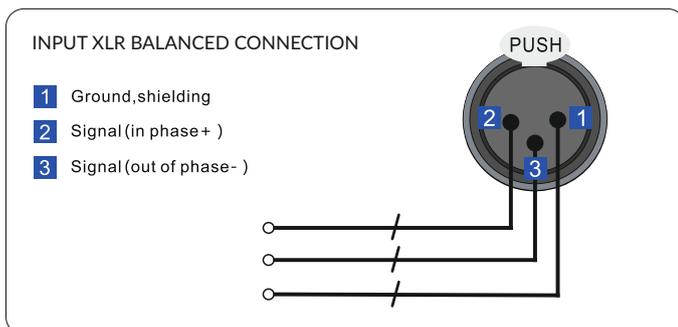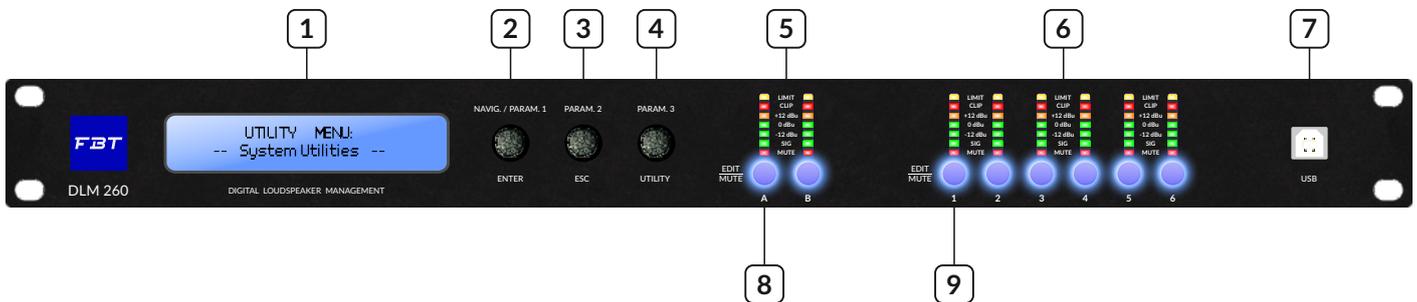
1) 2 * 24 alphanumeric display

2) PARAM.1/ENTER: the knob is used for the selection of the main pages of the several menus; when pressed the button function a lower level of the menu is entered or an edited parameter's value is confirmed.

3) PRAM.2/ESC: the knob is used for changing the values or the status of a parameter edited within the specific menu page, accessed through the PARAM.1/ENTER controls; when pressed the button function a higher level of the menu is entered or an edited parameter's value is NOT confirmed.

4) PARAM.3/UTILITY: the knob is used for changing the values or the status of a parameter edited within the specific menu page, accessed through the PARAM.1/ENTER controls; when pressed the button function a utility menu is entered for the DLM 260 / DLM 480 general setting (the navigation within the several "utility" menus is done using the PARAM.1/PARAM.2 knobs.

5) INPUT SIGNAL LEVEL METERS: these meters display the Input signal level before the Input Gain: MUTE does not affect the displayed level; the MUTE status is shown by the "mute led" which turns red when the related channel is muted; when the SIG led is on, it means the input signal reaches the -40dBu; -12dBu, 0dBu, +6dBu, +12dBu represent the actual RMS value of the signal; the CLIP/OVER led is on to indicate that the signal is exceeding the maximum level before the A/D analog-to-digital conversion OR that an internal Overflow is occurring due to excessive added gain on the signal path (normally the Overflow is controlled by an internal process and doesn't bring to signal distortion, but need to decrease the signal level along the path); the LIMIT light will turn on when the Input RMS Compressor is operating a dynamic control.

6) OUTPUT SIGNAL LEVEL METERS: these meters display the Output signal level before the output Gain, therefore after the all processes: MUTE affects the displayed level; the MUTE status is shown by the Mute led which turns red when the related channel is muted; when the SIG led is on, it means the input signal reaches the -40dBu; -12dBu, 0dBu, +6dBu, +12dBu represent the actual RMS value of the signal; the OVER led is on to indicate that an internal Overflow is occurring due to excessive added gain on the signal path (normally the Overflow is controlled by an internal process and doesn't bring to signal distortion, but need to decrease the signal level along the path); the LIMIT light will turn on when the Output RMS Compressor or/and the Output Peak Limiter is/are operating a dynamic control.

7) USB type B interface for PC connection

8) INPUT CHANNEL SELECTION KEY: short press this key to edit the parameters of the input channel processes (Gain, Polarity, Delay, Parametric EQ / RMS Compressor) or general parameters as the channel name. Press and hold for three seconds to mute the current channel.

9) OUTPUT CHANNEL SELECTION KEY: short press this key to edit the parameters of the output channel processes (Gain, Polarity, Delay, Xover, Parametric EQ / RMS compressor) or general parameters as the channel inputs to output channel Matrix. Press and hold for three seconds to mute the current channel.
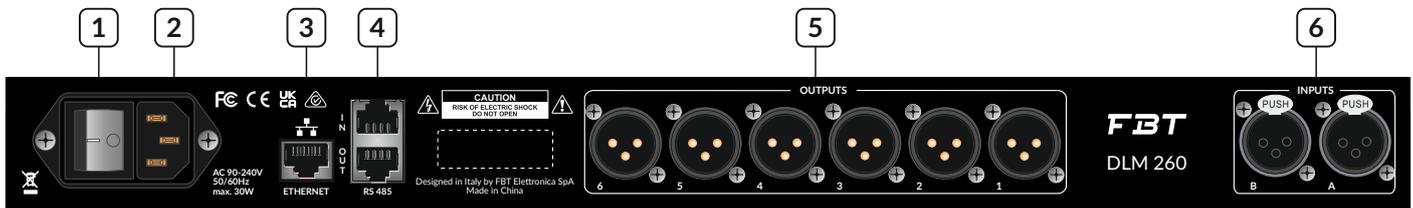
1) ON/OFF power switch.

2) AC power input, standard C13 interface; please ensure that the grounding pin of the power supply is well grounded, otherwise an electric shock accident may occur.

3) The LAN control port supports TCP/UPD protocol, and the IP address is automatically obtained by DHCP by default.

4) RS485 protocol interface, providing 1 input and 1 output dual interface, which can be used to connect software and also be used for central control protocol transmission; more....the Figure has to be set below the following line where wiring is defined as fig:

RS485B
RS485A

5) ANALOG SIGNAL OUTPUT INTERFACE: processor audio signal output, maximum output level +18dBu, minimum load 100 Ohm.

6) ANALOG SIGNAL INPUT INTERFACE: processor audio signal input, maximum input level +20dBu, input impedance 20kOhm.

SYSTEM SETTINGS

Utility general menu contains 3 sub-menu system settings; under UTILITY MENU turn the PARAM1 navigation key to select System utilities.

| UTILITY MENU |
| System Utilities |

| UTILITY MENU |
| Program Utilities |

| UTILITY MENU |
| Network Utilities |
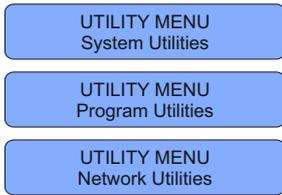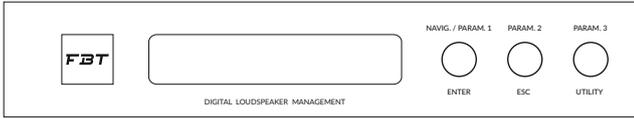
INPUT SOURCE SELECTION

Press PARAM1 to enter the secondary sub-menu where you can select Analog Input or built-in Signal generator.

| SYSTEM UTILITY |
| Input Source |

| Input Source |
| Source = Analog Inp |

| Input Source |
| Source = Signal Gen |
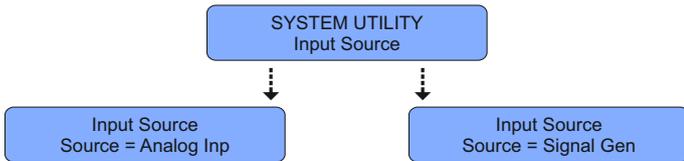
SIGNAL GENERATOR SETTINGS

Use the PARAM2 knob to select pink noise or white noise and use the PARAM3 knob to adjust the signal level from -30dBu to +10dBu with a step if 1dB

| Signal Generator | | Signal Generator |
| Type = Pink   L = -30dBu | | Type = White   L = -30dBu |

| Signal Generator | | Signal Generator |
| Type = White   L = -30dBu | | Type = White   L = +10dBu |
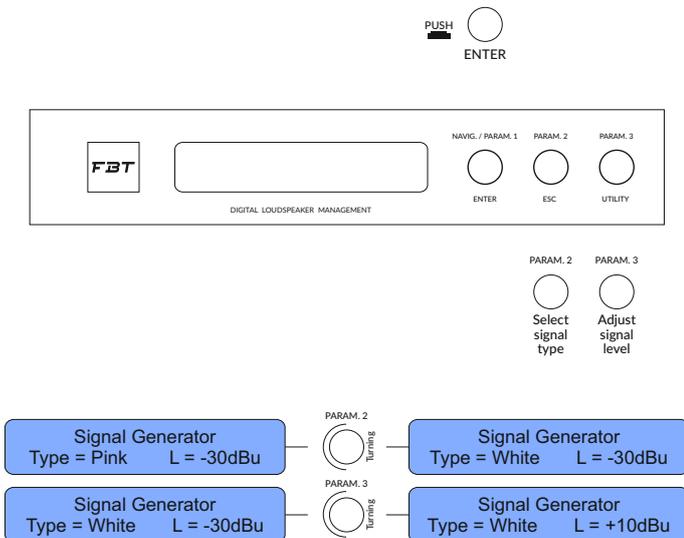
LINK INPUT

For the linked input channel modify the settings of any one of the linked channels and the other linked channels will synchronously modify the same value.

PARAM. 2 — Select input channel
PARAM. 3 — Turn on/off linkage

| Link Input | | Link Input |
| A=OFF   B=OFF | | A=OFF   B=OFF |

| Link Input | | Link Input |
| A=OFF   B=OFF | | A=OFF   B=ON |
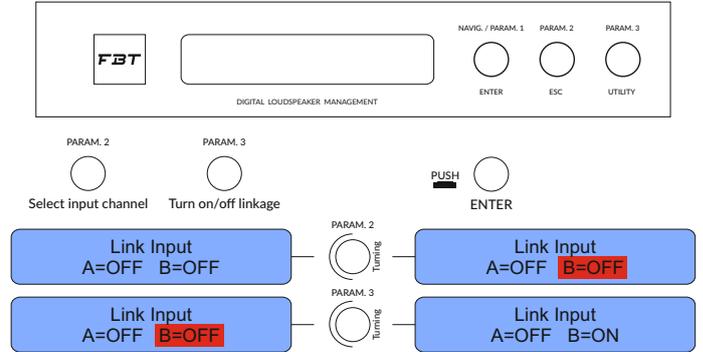
Turn the left and right knobs PARAM2 to select the AB channel, turn PARAM3 to the right to turn on the selected channel linkage, turn PARAM3 to the left to turn off the selected channel linkage.

LINKAGE OUTPUT

For the input channel after linkage, modify the settings of any linkage channel and the other linkage channels will simultaneously modify the same value.

PARAM. 2 — Select input channel
PARAM. 3 — Turn on/off linkage

| Link Output | | Link Output |
| 1=OFF 2=OFF 3=OFF 4=OFF | | 1=ON 2=OFF 3=OFF 4=OFF |

| Link Output | | Link Output |
| 1=ON 2=OFF 3=OFF 4=OFF | | 1=ON 2=OFF 3=OFF 4=OFF |

| Link Output | | Link Output |
| 1=ON 2=OFF 3=OFF 4=OFF | | 1=ON 2=ON 3=OFF 4=OFF |
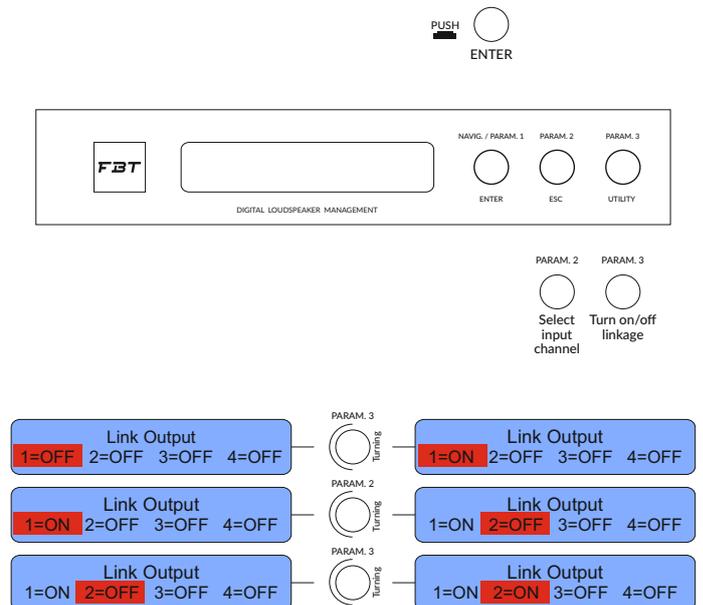
Knob PARAM2 selects the 1234 channnel, turn PARAM3 to the right to turn on and turn PARAM3 to the left to turn off the linkage.
For example, if you need to link output channels 3 and 4, the operation sequence is:
• Click the PARAM3 button
• Use the PARAM1 knob to select system
• Click PARAM1 to enter the submenu
• Rotate the PARAM2 to the right to select 3 channels
• Turn right PARAM3 to turn on
• Continue to rotate PARAM2 to the right to select 4 channel
• Turn right PARAM3 to turn on

RECALL MODE (MUTE SAFETY)

When this mode is turned on, the current channel mute state will be maintained regardless of reading any preset and only the muted state of the preset will be read when this mode is turned off.
This function can prevent the loud sound pressure from the loudspeaker from frightening the staff when switching presets.

KEEP LINKED CHANNELS LEVEL OFFSET (LINKED BY DCA MODE)

Allows the fader to be link in equal proportions in different positions. When the proportion of the input or output channels needs to be adjusted as a whole during use, the default linkage function will make the gain of each channel change along with the first changed channel. If you do not want to use it if the gain ratio changes, you can use this function, similar to the DCA/VCA groups.

Protect the output from distortion Keep the signal after the processor A/D will not be distorted and ensure that no distortion occurs in the processor regardless of any improvement.

When the channel linkage is not turned on, this function is not available.
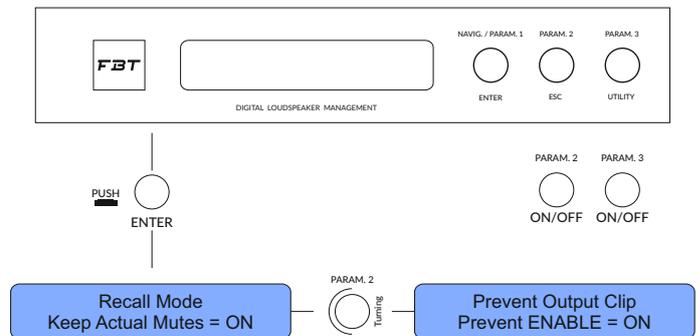
VIEW FIRMWARE VERSION
You can check the firmware version of the current processor, you can go to the our website to download the latest version of the firmware to get function updates and bug fixes.
Among them FP is the front panel firmware and MB is the main board firmware.

> SYSTEM UTILITY
> Firmware Version

> Firmware Version
> FP = 1.1.1   MB = 0.9
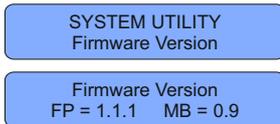
Note :
• The firmware version can be upgraded online
• Back up all presets before the firmware upgrade
• During the upgrade keep the power supply intact. If the upgrade fails midway, irreversible data loss may occur.

PROGRAM (PRESET) SETTINGS
Under UTILITY MENU turn the PARAM1 knob to select program utilities.

> UTILITY MENU:
> Program Utilities
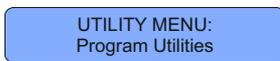
READ PRESET
Can read previously saved presets from this unit (before pressing read, please save the current preset, otherwise the current modification will be lost) Add a box.
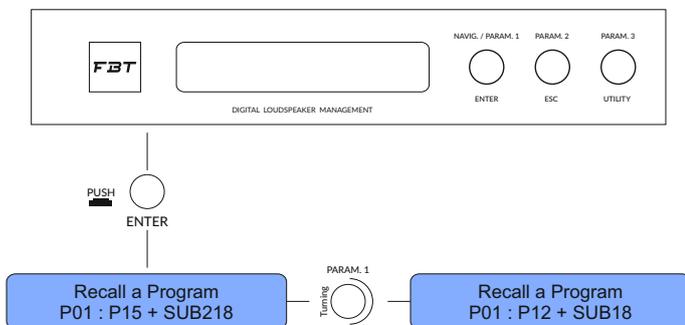
For example read the preset 3 from the machine
Press Utility
• Turn the PARAM1 knob to the right
• Go to Program Utilities and press PARAM1
• Press PARAM1 to recall a program
• Turn the PARAM1 knob to the right
• Choose P03
• Press PARAM1 to wait for the machine to finish reading

SAVE PRESET
Save the current input/output /routing and the entire state of the filter to the machine.
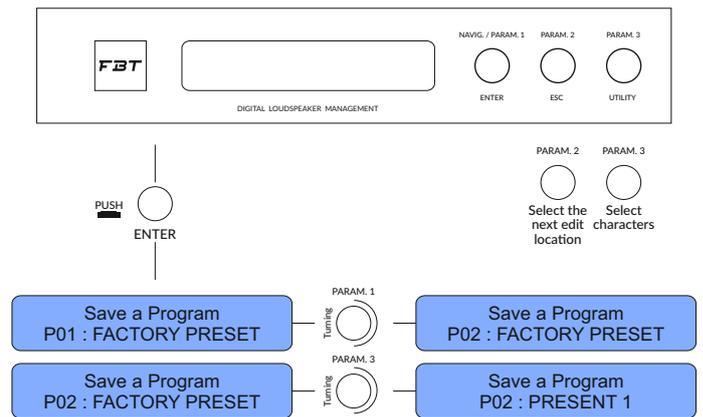
For example, if some changes have been made to the channel and need to be saved in the processor, follow the steps below to save:
• Click the PARAM3 button
• Use the PARAM1 knob to select program
• Press PARAM1 to enter the submenu
• Rotate PARAM1 to the right to select save a program
• Press PARAM1 to enter the next level
• Rotate PARAM1 to the right to select the preset position (such as P01)
• Press PARAM1 to confirm

• Enter the name, PARAM1 select the character position, PARAM2 select the character from the gallery below, the maximum allowable 16 characters

	!	″	#	$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

• After the modification is completed, click PARAM1 to save.

NETWORK OPTIONS

Under the settings, turn the PARAM1 knob, the last option is the network setting, the network setting is used to connect to the processor management software and provides 3 different interfaces, namely USB, RS485, LAN; among them the USB connection is the easiest and the installation driver can be directly connected to the PC; RS485 needs to set the serial port number, ID, baud rate, etc.. and need to be consistent with the PC settings; LAN is suitable for multiple processors or long distance connections. Through the local area network, stable connection quality and high connection speed can be obtained.

The DLM serie processors provide three connection methods: USB, RS485, TCPIP and you can choose any one of them to connect without choosing.

The RS485 serial port can be set in the network option menu and the network settings of the machine can be viewed.

> UTILITY MENU:
> Network Utilities

SET RS485 ID NUMBER

When multiple devices are connected via RS485 protocol, they need to be set to different ID numbers and cannot be repeated. After clicking PARAM1, rotate PARAM2 to select ID numbers from 1 - 32.
For example, the steps to set 485ID to 3 are:

• Click the PARAM3 button
• Use the PARAM1 navigation knob to select Network
• Click PARAM1 to confirm
• Rotate PARAM1 to the right to select RS485 ID
• Click PARAM1 to confirm
• Rotate PARAM2 to the right to select 03
• Click PARAM1 to confirm

> NETWORK UTILITY
> RS-485 ID

> RS-485 ID
> SET ID=01

PARAM. 1
Tuning

> RS-485 ID
> SET ID=03

DEVICE NAME

> NETWORK UTILITY
> Device Name

ENTER
PUSH

> Device Name
> Name = DLM 260

The model of the machine is displayed by default, if you need to modify it please follow the following order:
• Click the PARAM3 button
• Use the PARAM1 knob to select Network
• Click PARAM1 to confirm
• Rotate PARAM1 to Device Name
• Click PARAM1 to confirm
• Rotate PARAM2 to the right to select the edit character position
• Rotate PARAM3 to the right to select characters
• Rotate PARAM2 to modify the character position and PARAM3 to select the character; the maximum allowable 16 characters

	!	″	#	$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{	|	}	→	←

• Enter to confirm the modification and return

IP ADDRESS
Only the IP Address is displayed here and the default is automatic acquisition and the real address will be displayed after the connection is successful. It cannot be modified on the panel.

```
NETWORK  UTILITY          ENTER          IP  Address
   IP Address              ( )        IP  Addr = 000.000.000.000
                          PUSH
```

SUBNET MASK
The display is 255.255.255.0 which is only for display and cannot be modified on the panel.

```
NETWORK  UTILITY          ENTER          Subnet Mask
  Subnet Mask             ( )        SubMask = 000.000.000.000
                          PUSH
```
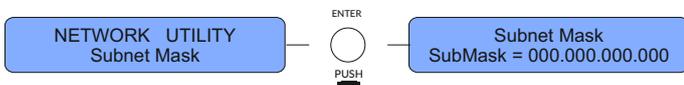
GATEWAY
Only for display, if you need to modify, you need to use the software to modify online

```
NETWORK  UTILITY:                        Gateway
    Gateway                          Gatway = 000.000.000.000
```

LAN MODE
The LAN mode defaults to DHCP and the panel is only used as a display. It can be modified to DHCP or a fixed IP address through software. When the number of devices exceeds a certain number, that is, there are too many DHCP servers in the LAN, network congestion may occur, so it is recommended to >5 pcs processors; when connecting at the same time use the fixed IP method to connect.

```
NETWORK  UTILITY:                        LAN  Mode
   LAN  Mode                          Mode = DHCP
```

SETTING OF INPUT CHANNEL
Edit input channel
Click the EDIT button below the processor level meter to edit the processing module of the input channel, long press to mute the current channel.

The level meter of the input channel shows the pre-fader level, that is, it is not affected by mute; the level meter of the output channel shows the post-fader level and the level is not displayed after the mute.

CHANNEL NAME EDITING
Click the edit button and the first item displayed is the channel name edit.

```
CH - A        Name      ENTER      CH - A        Name
Name = CH - A             ( )      Name = CH - A
                         PUSH

CH - A        Name      PARAM. 3   CH - A        Name
Name = CH - A             ( )      Name = PH - A
                        Turning

CH - A        Name      PARAM. 2   CH - A        Name
Name = PH - A             ( )      Name = PH - A
                        Turning

CH - A        Name      PARAM. 3   CH - A        Name
Name = PH - A             ( )      Name = PA - A
                        Turning

CH - A        Name      PARAM. 3   CH - A        Name
Name = PA - A             ( )      Name = PA - L
                        Turning
```
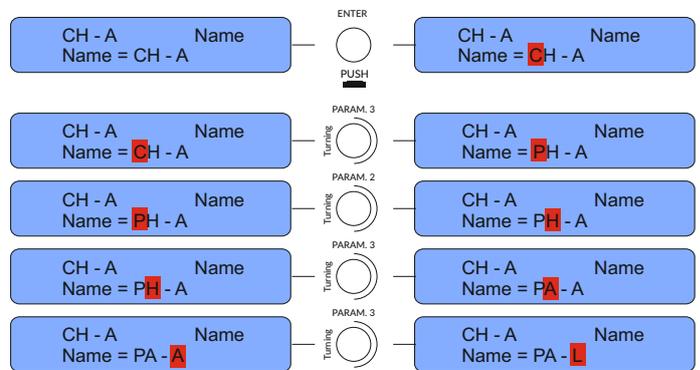
The default display, if you need to modify, please follow the following order:
• Click the Edit button to edit the channel
• Click PARAM1 to confirm
• Rotate PARAM2 to the right to select the edit character position
• Rotate PARAM3 to the right to select characters
• Rotate PARAM2 to modify the character position and PARAM3 to select the character ; the maximum allowable 6 characters.

	!	"	#	$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{			}	→	←

• Click PARAM1 to save after modification

NOISE GATE
The noise gate can appropriately suppress the obvious background noise caused by the accumulation of front-end equipment or improper system settings. On the current page click PARAM1 to edit the overall state of the noise gate and rotate PARAM2 / PARAM3 to open or close the noise gate.

| CH - A   N. Gate
Enable = OFF | PARAM. 2
Turning | CH - A   N. Gate
Enable = ON |
| CH - A   N. Gate
Threshold = -80 | PARAM. 2
Turning | CH - A   N. Gate
Threshold = -45 |
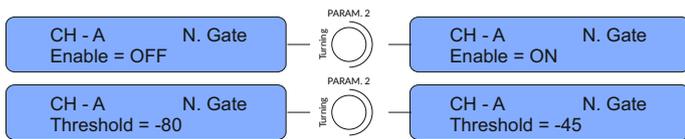
The threshold can be adjusted from -80 dBu to -45 dBu through PARAM2.

| CH - A   N. Gate
Rel Time = 1ms | PARAM. 2
Turning | CH - A   N. Gate
Rel Time = 1.0 sec |
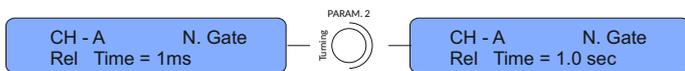
The startup time can be adjusted from 1 ms to 1000 ms rotating PARAM2.

| CH - A   N. Gate
Atk Time = 1ms | PARAM. 2
Turning | CH - A   N. Gate
Atk Time = 1.0 sec |
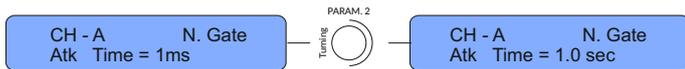
The release time can be adjusted from 1 ms to 1000 ms by rotating PARAM2.

DYNAMIC LOUDNESS BOOSTER
In the input channel the working principle is based on the equal loudness curve of the human ear to compensate and enhance the ultra-low frequency and ultra-high frequency.
When the signal strength is low, the boost ratio is relatively large; when the signal strength is high, the boost ratio is relatively small or even zero.
The maximum allowable boost gain value can be set separately for UHF and UHF and the setting of starting speed is also provided.

| CH - A   Loudness
Enable = OFF | PARAM. 2
Turning | CH - A   Loudness
Enable = ON |

When adjusting the maximum allowable gain the adjustment step of the knob PARAM2 is 1 dB and the adjustment step of PARAM3 is 0.1 dB.

| CH - A   Loudness
Lo - MaxBoost = 0.0 dB | PARAM. 2
Turning | CH - A   Loudness
Lo - MaxBoost = +10.0 dB |
| CH - A   Loudness
Hi - MaxBoost = 0.0 dB | PARAM. 2
Turning | CH - A   Loudness
Hi - MaxBoost = +10.0 dB |
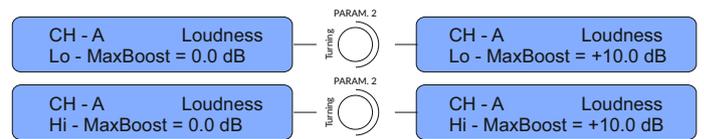
There are three levels of start-up time adjustable, namely fast, medium and slow.

| CH - A   Loudness
Atk Speed = SLOW |
| PARAM. 2
Turning |
| CH - A   Loudness
Atk Speed = MID |
| PARAM. 2
Turning |
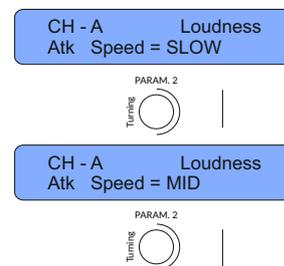
GAIN
Controllable gain range from -18 to +12 dB, 0.1 d B step

| CH - A   Gain
Gain = -18.0 dB |
| PARAM. 2
Turning |

FBT
DIGITAL LOUDSPEAKER MANAGEMENT
NAVIG. / PARAM. 1   PARAM. 2   PARAM. 3
ENTER   ESC   UTILITY

| PARAM. 2
Turning |
| CH - A   Gain
Gain = +12.0 dB |
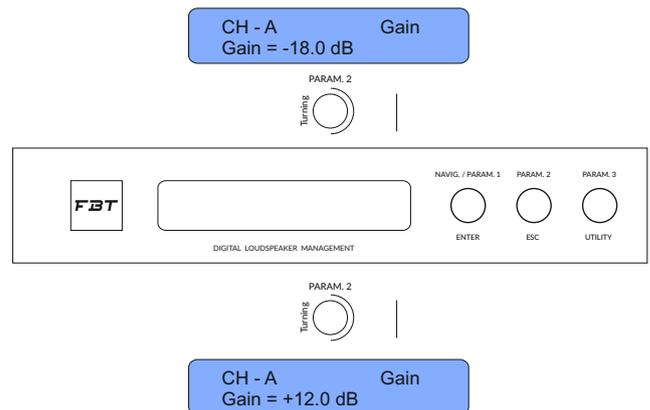
POLARITY
The overall polarity can be selected as positive polarity or -180° reverse polarity, which can be used to match the phase of the loudspeaker or correct the overall inversion caused by the wrong connection of the signal cable.

| CH - A   Polarity
Polarity = Normal | PARAM. 2
Turning | CH - A   Polarity
Polarity = Inverted |

DELAY

The input part can provide a maximum delay of 340 milliseconds, which can be used to delay the alignment of the tower speaker and the main amplifier speaker with a maximum distance of 115 meters.
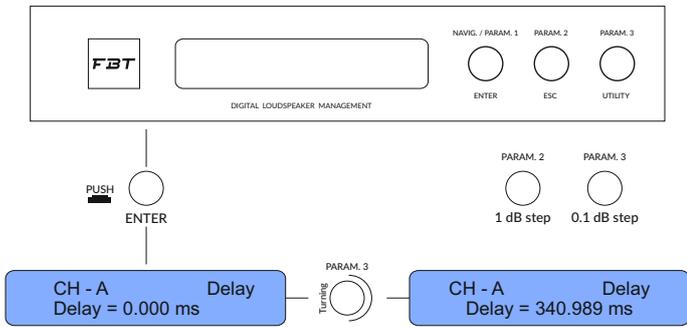
COMPRESSOR

RMS compressor reduce the start threshold to use the compressor to control the dynamics of the input signal. Using a lower start time (such as 5ms) and a higher release time (such as 1000ms) can firmly control the input signal within a certain range, of course, this will bring an obvious sense of compression (flatness).

Setting different start/release times in combination with music types will bring different listening feelings, and make up can also be used to compensate for the loss of loudness; and with a variable knee setting, the default 0 is the hard knee and the threshold is the compression starting point; 100% is the soft knee when the compression starting point is lower than the set threshold and gradually exceeds the threshold to achieve full effect.

Under high compression ratios, it is easy to hear unnatural and abrupt sounds when using hard knees. On the contrary, using soft knees can make the compression transition smoothly and bring a natural sense of hearing.
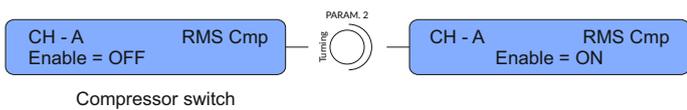
CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Enable = OFF	Turning	Enable = ON

Compressor switch

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Threshold = -10.0 dBu	Turning	Threshold = +20.0 dBu

The start threshold can be adjusted from -10dBu to +20dBu.

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Ratio = 2:1	Turning	Ratio = 32:1

The compression ratio can be adjusted from 2:1 to 32:1

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Knee = 0%	Turning	Knee = 100%

The soft and hard inflection point can be adjusted among which 0% is the hard inflection point and 100% is the soft inflection point.

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Rel Time = 100ms	Turning	Rel Time = 15s

The adjustable range of the release time is: 100ms to 15000ms.

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Atk Time = 0.1ms	Turning	Atk Time = 1s

The start-up time can be adjusted from 0.1ms to 1000ms.

CH - A RMS Cmp	PARAM. 2	CH - A RMS Cmp
Makeup = 0.0 dB	Turning	Makeup = +12.0 dB
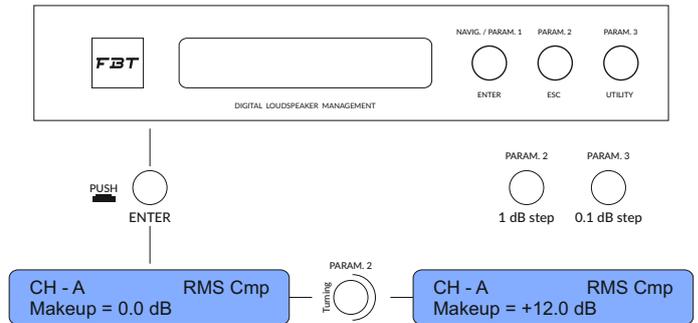
Provides automatic compressor settings and automatically sets the compression start threshold

EQUALIZER MODE

Provide 4 combinations; when multiple PEQs are needed, 31 bands of PEQ can be provided at most.

CH - A   EQ Mode
Mode = 31PEQ
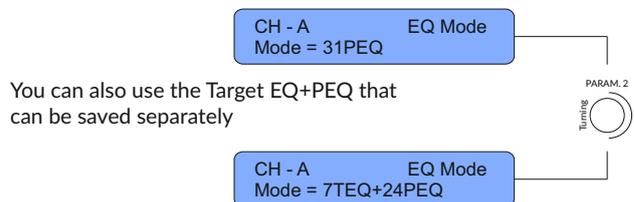
You can also use the Target EQ+PEQ that can be saved separately

PARAM. 2

CH - A   EQ Mode
Mode = 7TEQ+24PEQ

FIR FILTER

The input channel allows the use to use a 512 taps FIR filter, its operating sampling rate is 48kHz. But do not do any processing for the signal after 24000Hz, keep it as it is.

You can connect the software to edit the FIR filter, import or generate FIR high/low/band pass filters according to the guide.

When PEQ mode is selected as with FIR filter, this option can be turned on.

```
CH - A              FIR E/N
FIR Enable = OFF
```

PARAM. 2
Turning

```
CH - A              FIR E/N
FIR Enable = ON
```

```
CH - A              FIR E/N
FIR Enable = N / A
```
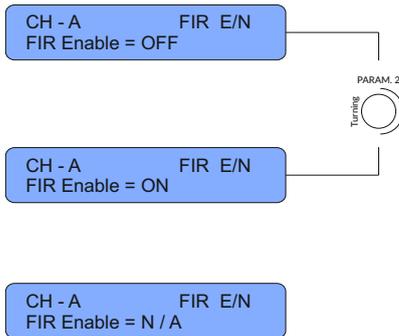
EQUALIZER STATE

PEQ overall on/off, TEQ overall on/off


```
CH-A              EQ E/N
TEQ E/N=OFF  PEQ E/N=OFF
```

PARAM. 2
Turning

```
CH-A              EQ E/N
TEQ E/N=ON  PEQ E/N=OFF
```

```
CH-A              EQ E/N
TEQ E/N=ON  PEQ E/N=OFF
```

PARAM. 3
Turning

```
CH-A              EQ E/N
TEQ E/N=ON  PEQ E/N=ON
```
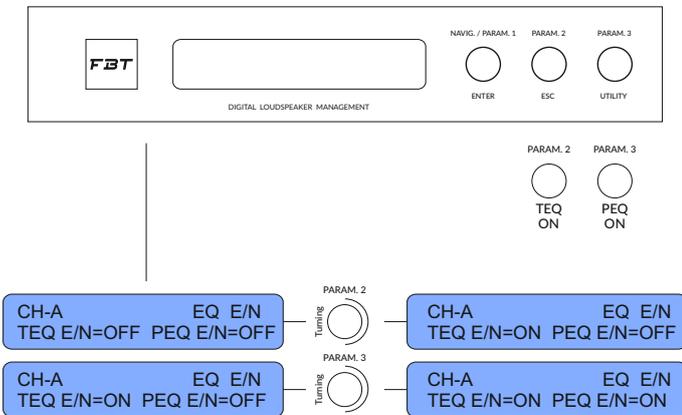
Note: when PEQ mode is selected as full PEQ, TEQ is not available, that is, N/A displayed

```
CH - A              EQ E/N
TEQ E/N=N/A    PEQ E/N=OFF
```

TEQ (TARGET EQ)

Types include: BELL, HiShvQ, LoShvQ, LP Q, HP Q.

```
CH - A              TEQ - 01
E / N = ON      Type = BELL
```

```
CH - A              TEQ - 01
Frequency = 4000Hz
```

```
CH - A              TEQ - 01
Gain = 0.0dB      Q = 1.30
```
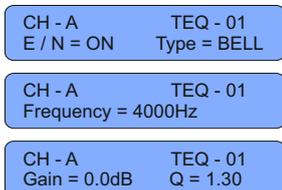
PEQ 1-31

Specific turning of each band of parametric equalization, after Enter, you can rotate PARAM1 to select the filter type, you can select high and low Shelf / Bell / High and Low pass/all pass/notch, etc, rotate the PARAM1 button to enter the specific parameter adjustment, PARAM2 = gain, PARAM3 = Q value, Frequency and so on.

To add a high-self Q with a frequency of 5506 Hz, a bandwidth of 5.1 and a gain of -5dB to input channel B, follow the steps below.

- Click the Edit button of input B
- Use the PARAM1 knob to select PEQ1
- Click PARAM1 to enter the edit
- Rotate PARAM3 to the right to select HiShvQ
- Continue to rotate PARAM1 right to select frequency adjustment
- Turn PARAM2 to the right to adjust the frequency to 5500Hz, use PARAM3 to fine-tune to 5506Hz
- Click PARAM2 to return
- Right-turn PARAM1 to adjust gain and Q value
- Use PARAM2 to adjust gain and PARAM3 to adjust Q value
- Complete and return

```
CH-A              PEQ - 01          CH-A              PEQ - 01
E / N = ON   Type = BELL            E / N = ON   Type = HiShv Q
```
PARAM. 3
Turning

Types include: BELL, HiShv1, HiShv2, HiShvQ, LoShv1, LoShv2, LoShvQ, LP 1st, LP 2nd, LP Q, HP 1st, HP 2nd, HP Q, Notch, AllPs1, AllPs2.

```
CH-A              PEQ - 01          CH-A              PEQ - 01
E / N = ON   Type = HiShv Q         Frequency = 27 Hz
```
PARAM. 1 Turning

```
CH-A              PEQ - 01          CH-A              PEQ - 01
Frequency = 27 Hz                   Frequency = 5500 Hz
```
PARAM. 2 Turning

```
CH-A              PEQ - 01          CH-A              PEQ - 01
Frequency = 5500 Hz                 Frequency = 5506 Hz
```
PARAM. 3 Turning

```
CH-A              PEQ - 01          CH-A              PEQ - 01
Frequency = 5506 Hz                 Gain = 0dB      Q = 2.90
```
PARAM. 1 Turning

```
CH-A              PEQ - 01          CH-A              PEQ - 01
Gain = 0dB      Q = 2.90            Gain = -5.0 dB      Q = 2.90
```
PARAM. 2 Turning

```
CH-A              PEQ - 01          CH-A              PEQ - 01
Gain = -5.0 dB      Q = 2.90        Gain = -5.0 dB      Q = 5.10
```
PARAM. 3 Turning

EDIT OUTPUT CHANNEL

EDIT button click into the lower level processor can edit the table processing module output channels.

Long press the Edit button to mute the current channel, the level meter of the output channel displays the post-fader level and does not display the level after mute.

CHANNEL NAME

Channel name edit.

Click the Edit button and the first item displayed is the channel name edit; the default display, if you need to modify, please follow the following order:

- Click the channel edit button
- Click PARAM1 to confirm
- Rotate PARAM2 to the right to select the edit character position
- Rotate PARAM3 to the right to select characters
- Rotate PARAM2 to modify the character position, PARAM3 to select the character; the maximum allowable 6 characters.

	!	"	#	$	%	&	'	(	)	*	+	,	-	.	/	0	1	2	3	4	5	6	7
8	9	:	;	<	=	>	?	@	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O
P	Q	R	S	T	U	V	W	X	Y	Z	[	¥	]	^	_	`	a	b	c	d	e	f	g
h	i	j	k	l	m	n	o	p	q	r	s	t	u	v	w	x	y	z	{			}	→	←

- Click PARAM1 to save after modification

Push

INPUT MATRIX

The input matrix allows the user to mix 2 input signals to any output channel in any ratio. The knob PARAM1 selects the AB input source and the PARAM3 adjusts the gain amount, which can be adjusted from -30 to 0 dB.

PARAM. 2 — On/Off
PARAM. 3 — Gain

CH-1 Source A:0dB B:OFF C:OFF D:OFF	ENTER PUSH	CH-1 Source CH - A = ON Gain = 0dB
CH-1 Source CH - A = ON Gain = 0dB	PARAM. 1 Turning	CH-1 Source CH - B = OFF Gain = 0dB
CH-1 Source CH - B = OFF Gain = 0dB	PARAM. 2 Turning	CH-1 Source CH - B = ON Gain = 0dB
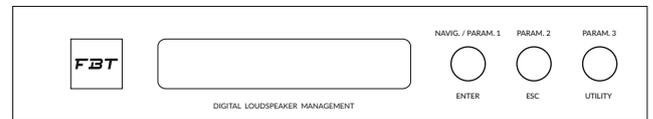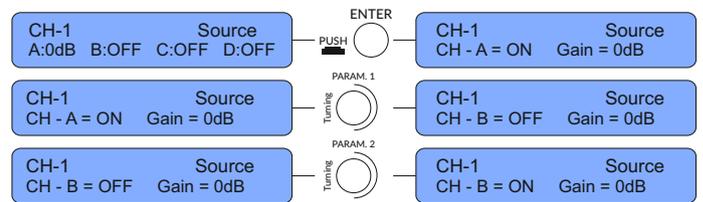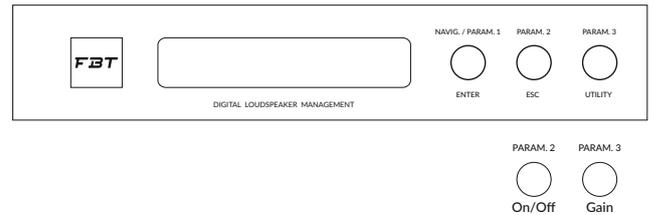
The default route of processor is channel AB(CD) pass-through output 1,2,3,4,5,6,(7,8). In actual use it may be necessary to assign one input signal to multiple output channels, such as routing input channel A to output 1/2 (since output channel 1 has sent 0dB from input A by default, so directly start from channel 2).

- Press the select button for output 2
- Select the PARAM1 navigation button to go to the source screen
- Press PARAM1
- To rotate PARAM1, select B
- Rotate PARAM3 to adjust the gain of input B to minus infinity
- To rotate PARAM1 , select A
- Rotate PARAM3 to adjust the gain of input A to 0dB
- Press PARAM2 to complete

15

GAIN

The controllable gain range is from -18 to +12dB in 0.1dB steps.

| CH-A Gain
Gain = 0.0 dB | Turning
PARAM. 2 | CH-A Gain
Gain = -18.0 dB |
| CH-A Gain
Gain = 0.0 dB | Turning
PARAM. 2 | CH-A Gain
Gain = +12.0 dB |
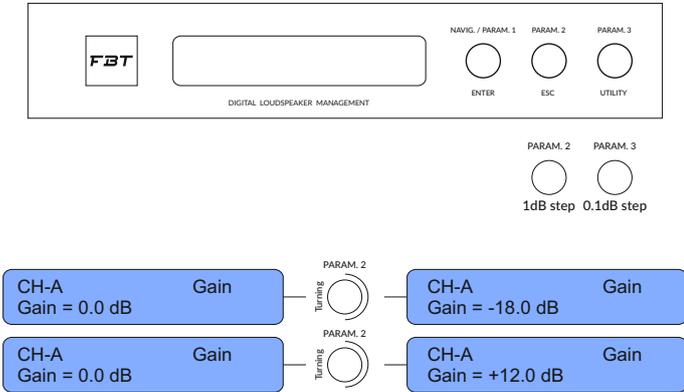
POLARITY

The overall polarity can be selected as normal polarity or -180° reverse polarity, which can be used to match the phase of the loudspeaker or correct the overall inversion caused by the wrong connection of the signal cable.

| CH-1 Polarity
Polarity = Normal | Turning
PARAM. 2 | CH-1 Polarity
Polarity = Inverted |
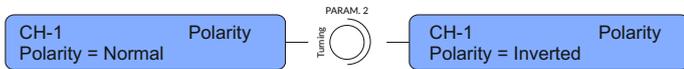
DELAY

The output part provides a maximum delay of 170 milliseconds, which can be used to align the time between the multi-way units, in a step of 10.4 microseconds / 0.01 milliseconds.

| CH-1 Delay
Delay = 0.000ms | Turning
PARAM. 3 | CH-1 Delay
Delay = 169.989ms |
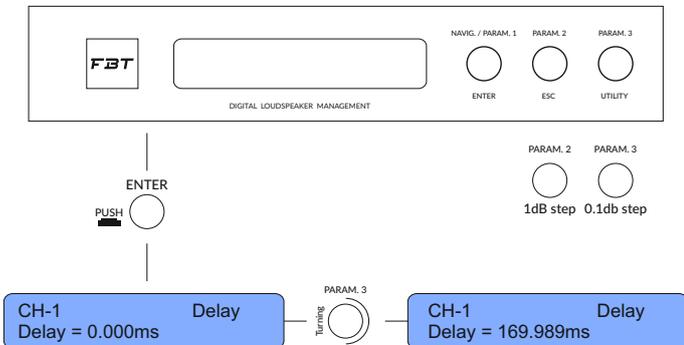
RMS COMPRESSOR

It is mainly used to limit the RMS power of the unit. It needs to cooperate with the AES power provided by the unit manufacturer and the amplification factor of the power amplifier to calculate the threshold; the attack time is often determined by the period corresponding to the frequency of the high-pass filter and the release time is always set to 16 times the startup time.

For example, the AES power (2h) of a HF driver is 100 watts, the impedance is 16 Ohms, the crossover point is 1000Hz and the amplification factor of the power amplifier is 40dB, then according to $P=U^2/R$, the maximum input voltage of the unit is 40 volts, divided by the power amplifier magnification is 100 times, the voltage limit should be activated when the signal level is 0.4v, 0.4v is converted to $20\log(0.4/0.775)$ to get -5.84 about -6dBu, that is, the treshold is -6dBu; the cycle corresponding to the crossover point 1000 if it is 1ms, then the start time can be set to 1ms and the release time is 100ms.
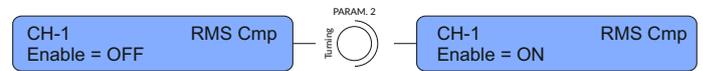
| CH-1 RMS Cmp
Enable = OFF | Turning
PARAM. 2 | CH-1 RMS Cmp
Enable = ON |

Compressor switch

| CH-1 RMS Cmp
Threshold = +20.0dBu | Turning
PARAM. 2 | CH-1 RMS Cmp
Threshold = -6dBu |

The attack time can be adjusted from 0.1ms to 1000ms.

| CH-1 RMS Cmp
Atk Time = 10ms | Turning
PARAM. 2 | CH-1 RMS Cmp
Atk Time = 1ms |

The start threshold can bge adjusted from -10dBu to +20dBu

| CH-1 RMS Cmp
Rel Time = 100ms |
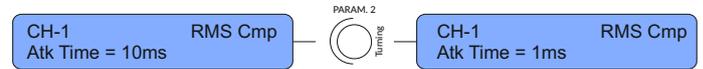
Release Time can be adjust from 100ms ~ 15

| CH-1 RMS Cmp
Ratio = 32:1 |

The compression ratio can be adjusted from 2:1 to 32:1

```
CH-1                    RMS Cmp
Knee = 0 %
```

The soft and hard inflection point can be adjusted among which 0% is the hard inflection point and 100% is the soft inflection point.

```
CH-1            RMS Cmp          PARAM. 2          CH-1            RMS Cmp
Makeup = 0.0 dB                  Tuning           Makeup = +12.0 dB
```
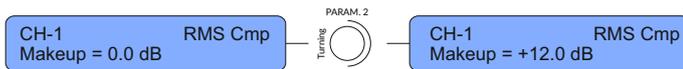
Compression compensation can be set range: 0 ~ 12dB

PEAK LIMIT

It is mainly used to limit the peak signal and protect the woofer from mechanical damage caused by the voice coil movement exceeding the linear stroke. If you need to accurately set the stroke/frequency and voltage curve of the unit, you can consult the manufacturer of the unit or measure and obtain data by yourself.

For example, using the ** model woofer, it is found that the stroke of the unit reaches the limit of 12mm at 103 Volts. At this time, the amplifier magnification is 38dB, 103/80=1.2875v, 20log (1.2875/0.775)=4.4, that is, the peak compressor's the startup threshold is 4.4dBu.
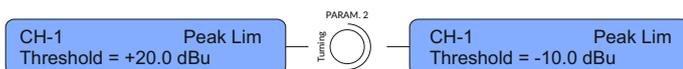
```
CH-1            Peak Lim         PARAM. 2          CH-1            Peak Lim
Enable = OFF                     Tuning           Enable = ON
```

Peak limiter switch

```
CH-1            Peak Lim         PARAM. 2          CH-1            Peak Lim
Threshold = +20.0 dBu            Tuning           Threshold = -10.0 dBu
```

The threshold can be adjusted from -10 dBu to +20 dBu

```
CH-1            Peak Lim         PARAM. 2          CH-1            Peak Lim
Rel Time = 1.0s                  Tuning           Rel Time = 100ms
```

The adjustable range of the release time is 100ms to 15000ms

```
CH-1            Peak Lim         PARAM. 2          CH-1            Peak Lim
Atk Time = 1ms                   Tuning           Atk Time = 1sec
```
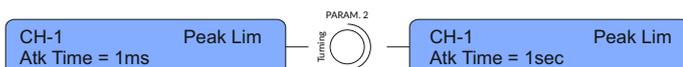
The attack time can be adjusted from 0.1ms to 1000ms

X-OVER MODE

The DLM 260 series has built-in three kinds of crossover filters.

• Traditional IIR filter, including three types of Linkwiz-Raylei/Bessel/Butterworth with a maximum slope of 48dB / oct.

• Linear phase IIR filter, using FBT exclusive algorithm, retains the shape of the Linkwiz-Raylei/Bessel/Butterworth filter types without phase distortion, maintaining the linear phase and controlling the time delay at the same time in a reasonable range, it can directly replace the traditional IIR filter for speaker frequency division and only need to align the delay.

• FIR finite impulse response filter, allowing users to customize the window function in the software to generate high-slope, steep filter shapes and also accept FIR convolution generated by external third-party software based on measurement , which can realize complex filter shapes and multiple sections, the parameters are equalized and the phase is corrected at the same time.

Only three modes are displayed here. The specific parameters are adjusted in the software high and low pass filter options, which cannot be adjusted on the front panel.

```
CH-1                    Out Mode
Mode=Xover - IIR + 8PEQ
```

IIR mode

```
CH-1                    Out Mode
Mode=Xover - DLM + 8PEQ
```

Linear phase mode

```
CH-1                    Out Mode
Mode=Xover - FIR + 8PEQ
```

FIR mode

```
CH-1                    Out Mode
Mode=Xover - IIR+FIR+4PEQ
```
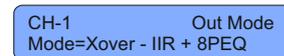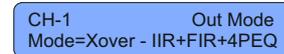
IIR + FIR mode (only 4 band of PEQ can be used in this mode)

The software contains 4 combinations of filter forms, including IIR / LIN (Linear phase filter) / FIR / IIR+FIR four combinations are optional. When a single IIR / LIN / FIR mode is selected , a total of 8 band PEQ can be used, while when IIR+FIR filter is selected, only 4 band PEQ can be used.

FIR FILTER
The output channel allows the user to use a 512-taps FIR filter and its operating sampling rate is 48kHz, but it does not do any processing for the signal beyond 24000Hz, keep it as it is (for the FIR filter status switch in the previous X-over mode).
Select the mode with FIR filter to display ON/OFF.

CH-1 FIR E/N		CH-1 FIR E/N
FIR Enable = OFF	Turning PARAM. 2	FIR Enable = ON

Otherwise it shows unavailable N/A

CH-1   FIR E/N
FIR Enable = N / A
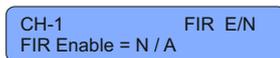
HIGH PASS FILTER
In IIR mode the type, slope, frequency and other specific parameters of the high pass filter can be sdjusted.
Click PARAM1 to edit.

CH-1 HPF		CH-1 HPF
F = 20Hz Bypass	ENTER PUSH	Frequency 20Hz
CH-1 HPF		CH-1 HPF
Frequency 20Hz	Turning PARAM. 2	Frequency 20000Hz

Traditional filter categories include: Bessel, Butterworth, Linkwitz-Riley, with a maximum slope of 48dB / oct.

CH-1 HPF		CH-1 HPF
Slope = Bypass	Turning PARAM. 2	Slope = -48dB LR

It additionally contains NXF (Notched X-over Filter), the slope range is -40 ~ -75dB/oct.

CH-1 HPF		CH-1 HPF
Slope = -48dB LR	Turning PARAM. 2	Slope = -40dB NXF
CH-1 HPF		CH-1 HPF
Slope = -40dB NXF	Turning PARAM. 2	Slope = -75dB NXF

When the combination of IIR and FIR filter is selected, the highest slope of IIR filter is -24dB/oct and only 4 band PEQ are available.

CH-1 HPF		CH-1 HPF
Slope = Bypass	Turning PARAM. 2	Slope = -24dB BS

LOW-PASS FILTER
In IIR mode the type, slope, frequency and other specific parameters of the low-pass filter can be adjusted.
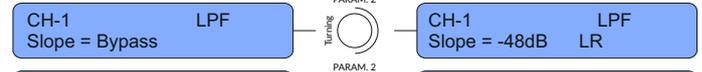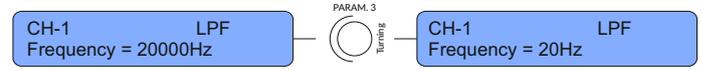
CH-1 LPF		CH-1 LPF
F = 20000Hz Bypass	ENTER PUSH	Frequency = 20000Hz
CH-1 LPF	PARAM. 3	CH-1 LPF
Frequency = 20000Hz	Turning	Frequency = 20Hz
CH-1 LPF	PARAM. 2	CH-1 LPF
Slope = Bypass	Turning	Slope = -48dB LR
CH-1 LPF	PARAM. 2	CH-1 LPF
Slope = -40dB NXF	Turning	Slope = -75dB NXF

When the combination of IIR and FIR filter is selected the highest slope of IIR filter is -24dB/oct and only 4 band PEQ are available.

CH-1 LPF		CH-1 LPF
Slope = Bypass	Turning PARAM. 2	Slope = -24dB BS
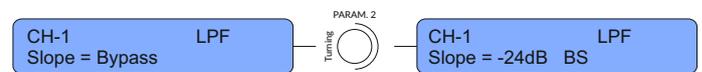
When the LIN (linear phase filter) or FIR filter type is selected the high/low pass filter cannot be adjusted on the panel, only by software.

CH-1 HPF		CH-1 LPF
Highpass is Not Actived	Highpass is Not Actived
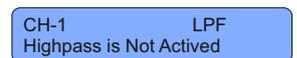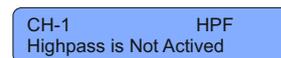
EQUALIZER ON STATE
PEQ overall on/off

CH-1 EQ E / N		CH-1 EQ E / N
Enable = OFF	Turning PARAM. 2	Enable = ON
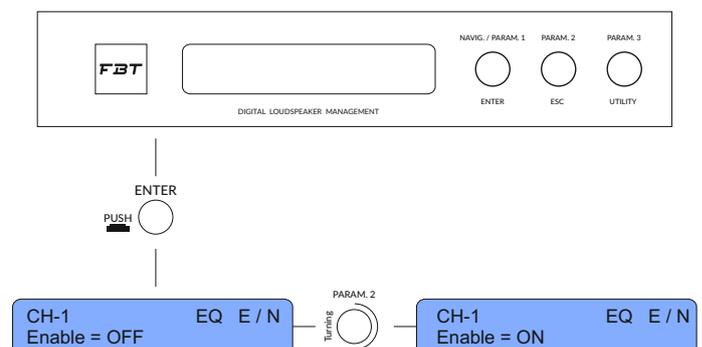
PEQ 1-8

Specific turning of each stage of parameter equalization, after enter, you can rotate PARAM1 to select filter type, you can rotate high and low Chevron/Bell/high and low pass/all pass/notch, etc; rotate PARAM1 navigation key to enter specific parameter adjustment, PARAM2=gain , PARAM3=Q value, frequency and so on.

- Press the Edit button of Output 2
- Rotate the PARAM1 to select PEQ1
- Press PARAM1
- Rotate PARAM3 to the right to select All Ps2 filter
- Press PARAM2
- Continue to rotate the PARAM1 selection frequency adjustment to the right
- Rotate PARAM2 to adjust the frequency to 5600Hz, use PARAM3 to fine tune to 5606Hz
- Press PARAM2 to return
- Rotate PARAM1 right to adjusts the Q value
- Complete return

CH-2 PEQ - 01	PARAM. 3	CH-2 PEQ - 01
E / N=ON Type=BELL	Turning	E / N=ON Type=AllPs2

Types include: BELL, HiShv1, HiShv2, HiShvQ,LoShv1,LoShv2, LoShvQ, LP1st, LP2end, LP Q, HP 1st, HP 2nd, HP Q, Notch, AllPs1, AllPs2.

CH-2 PEQ - 01	PARAM. 1	CH-2 PEQ - 01
E / N = ON Type = AllPs2	Turning	Frequency = 31 Hz
CH-2 PEQ - 01	PARAM. 2	CH-2 PEQ - 01
Frequency = 31 Hz	Turning	Frequency = 5600 Hz
CH-2 PEQ - 01	PARAM. 3	CH-2 PEQ - 01
Frequency = 5600 Hz	Turning	Frequency = 5606 Hz
CH-2 PEQ - 01	PARAM. 1	CH-2 PEQ - 01
Frequency = 5606 Hz	Turning	Gain = 0.0dB Q = 2.9
CH-2 PEQ - 01	PARAM. 3	CH-2 PEQ - 01
Gain = 0.0dB Q = 2.9	Turning	Gain = 0.0 dB Q = 4.80
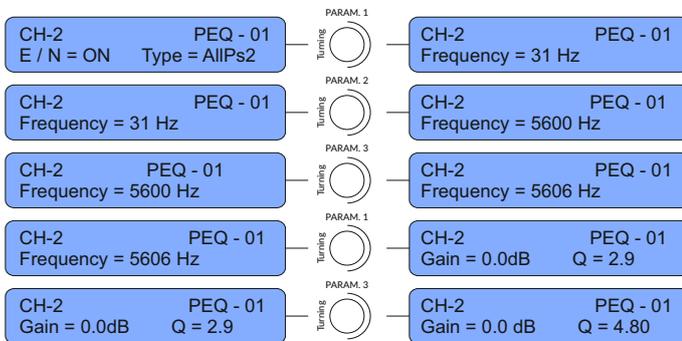
OUTPUT MATRIX

The output matrix allows the user to mix output channel signal to any physical output channel in any ratio. The knob PARAM1 selects the source and the PARAM3 adhusts the gain amount, which can be adjusted from -30 to 0dB.

For example, to mix output channel 1.2 to physical output 2, you need to do the following:

- Click the edit button below output 1
- Rotate PARAM1 key to the end of CH1 routing
- Rotate PARAM1 to select output channel 2
- Rotate PARAM3 to adjust the gain to 0dB
- Click PARAM2 to return
- Click the EDIT button below output 2 and it will automatically appear on the CH2 routing interface
- Rotate PARAM1 to select output channel 1
- Rotate PARAM3 to adjust the gain to 0dB finish

CH-1 Routing	ENTER	CH-1 Routing
1:0dB 2:OFF 3:OFF 4:OFF	PUSH	OUT1=ON Gain=0dB
CH-1 Routing	PARAM. 1	CH-1 Routing
OUT1=ON Gain=0dB	Turning	OUT2=OFF Gain=0dB
CH-1 Routing	PARAM. 2	CH-1 Routing
OUT2=OFF Gain=0dB	Turning	OUT2=ON Gain=0dB
CH-1 Routing	PARAM2	CH-1 Routing
OUT2=ON Gain=0dB	PUSH	1:0dB 2:0dB 3:OFF 4:OFF
CH-1 Routing	EDIT CH2	CH-2 Routing
1:0dB 2:0dB 3:OFF 4:OFF	PUSH	1:OFF 2:0dB 3:OFF 4:OFF
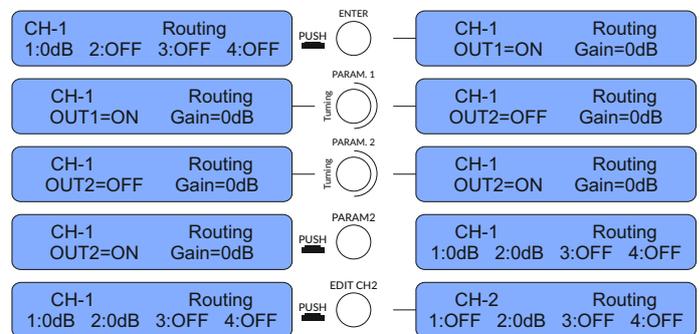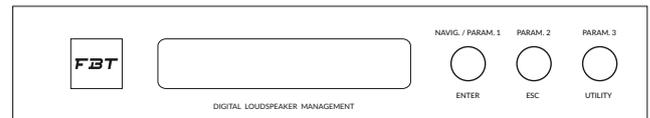
PEQ 1-8
Minimum system requirements for running DLM 260/480 software:
Operating system   Microsoft Windows
CPU   Single core 2.0 GHz
Running memory   2 GB
Storage space   1 GB
Need Microsoft Net framework 4 or higher and Microsoft Visual C++2015-2022
The DLM series provided 3 kinds of control interfaces, namely USB, RS485, TCP IP and there is no need to set up the connection switching of the three ways.

CONNECTING TO A DEVICE

• Using a network cable to connect it is the most direct and safe way of connecting. Use a network cable to connect the PC's network port directly to the processor's Ethernet port and make sure that the computer's IP is set to automatically get connection (DHCP).
In order to get things working follow this actions sequence: Settings---Network and Internet---Ethernet---Change adapter options---Select the currently used adapter---Properties---Internet Protocol version 4 (TCP/IPv4)---Select automatic get IP address.
After the network cable connection is confirmed, double click the DLM series software, the device will be automatically discovered and a pop-up window will prompt.

• Use RS485 protocol connection through USB to RS485 device, you can get the serial port of RJ45 port and connect directly to RS485-IN/OUT on the rear panel. The RS485 input and output on the rear panel are designed with loop-out design. When only one device is connected can be plugged into RS485-IN/OUT at will.
When using the USB interface to connect, you need to pay attention to the following points:
- When your PC operating system is Windows XP or Windows 7, you need to install the STM32 virtual COM driver so that the processor can be correctly identified and connected.
- When your PC operating system is Windows 8 or above, there is usually no need to install a driver and the system will correctly identify the processor (if the driver is not installed automatically , install the STM32 virtual COM driver manually).

RTA + Signal Generator

Analog IN A
Analog IN B
Analog IN C
Analog IN D

A/D

A
B
C
D

Meter — Noise Gate — Gain — Mute — DYN EQ

31band PEQ
24PEQ+7TEQ
13PEQ+FIR
6 PEQ+7 TEQ+FIR

RMS Comp — Polariaty — Level — Delay

Routing Matrix

IIR — 8PEQ
LINEAR Phase — 8PEQ
FIR — 8PEQ
IIR+FIR — 4PEQ

HP / LP — Gain — RMS Comp — Meter — PEAK Limiter — Delay — Polariaty — Mute

Routing Matrix

Routing Matrix

D/A

Analog OUT 1
Analog OUT 2
Analog OUT 3
Analog OUT 4
Analog OUT 5
Analog OUT 6
Analog OUT 7
Analog OUT 8

CONNECT WITH USB
When using the USB interface, directly connect to the USB port of the PC with a B-type cable , add USB to the software and select the corresponding model.

USB

USB

USE NETWORK CABLE TO CONNECT
When using the network interface to connect use a category 5/6 network cable to connect the Ethernet port of the processor to the network adapter interface of the PC, adjust the IPV4 address option of the corresponding network adapter in the Windows network settings to automatically obtain DHCP and then open the software, the current processor can be found automatically.

CAT5/6 Cable
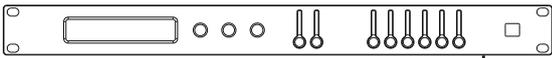
When using only one processor you can use DHCP to connect to the processor. The specific methods are as follow:
The PC does not need to change the settings by default, directly connect the Ethernet port of the PC and the processor with a network cable and open the software in the CD to automatically connect.
When using multiple processors you can use a fixed IP to reduce the error rate. You need to set the PC to the same network segment as the processor but different IP addresses.

LAN

1

2

1

2
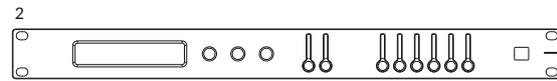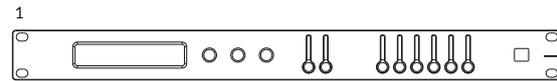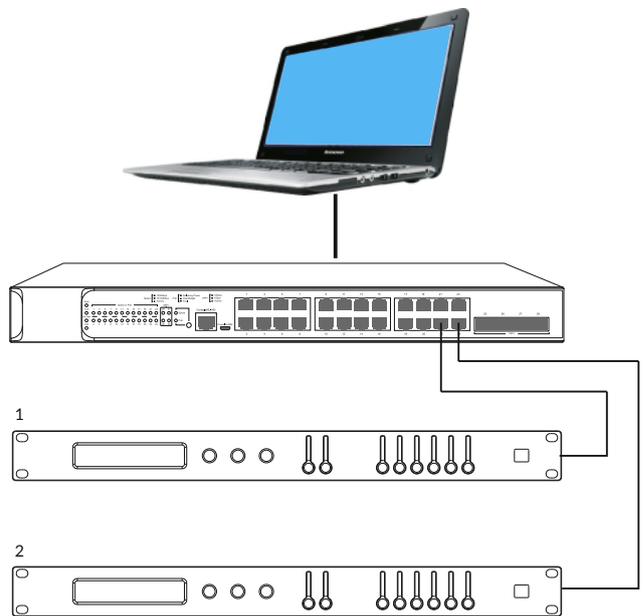
USE RS485 TO CONNECT

The use of RS485 often requires a patch cord:
- Traditional PC will provide DB9 serial port which needs to use DB9 to RJ45 conversion cable to adapt.
- Generally current household models do not provide serial ports, so USB to serial converters are needed and DLM series provide USB to serial port.

FIRMWARE UPGRADE

When FBT releases a new version of the firmware, users can download the latest firmware from the official website.

You can use TCP/IP to upgrade the firmware to ensure the stability of the power supply during the upgrade process. In case of a power outage, you may need to return to the factory for repair.
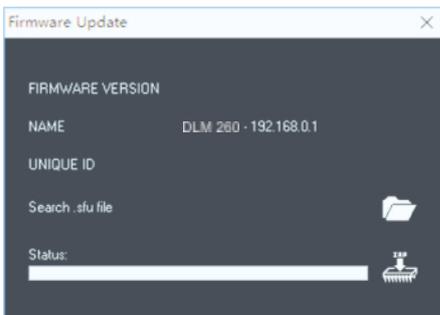
ADVANCED FEATURES
- Group control Overall marshalling control .
  The simultaneous marshalling control of multiple processors simplifies a lot of work processes for on-site debugging, can quickly respond to emergencies and reduces the amount of repetitive work.
- To use the marshalling function it is recommended to use a switch to connect multiple processors with a network cable and then modify the IP address of each to a fixed IP and use a different IP address.
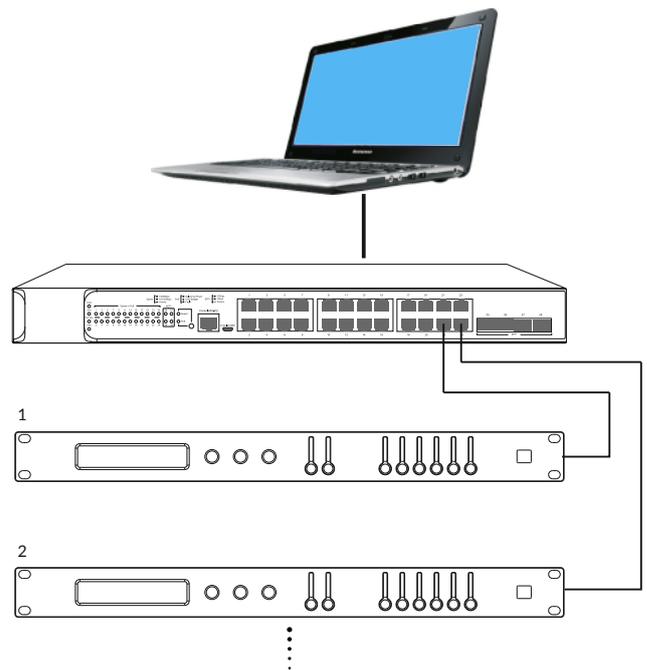
Connect the processors that need to be grouped online in the software and choose to add grouping.
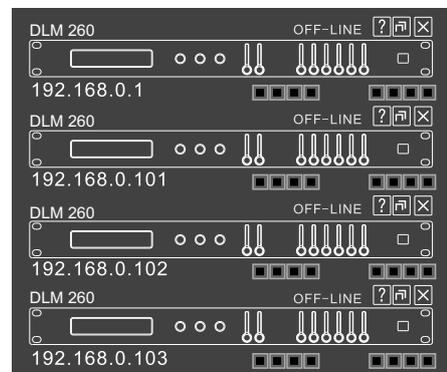
Tick the input channels that need grouping control, you can freely group the input channels, you can create a new group to control the remaining ungrouped channels, click Add after the selection is complete.

At this time there is a small window for group control, you can control the overall gain, mute, monitor level,etc...
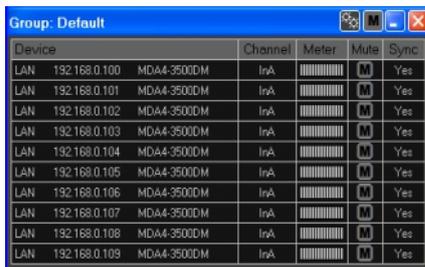
HOME PAGE OVERVIEW

Enter the main page of the software, the upper left corner respectively shows: New / Import / Save project

Add device

For the case of a known machine with a fixed IP address / USB / RS485, use to add the corresponding device for online operation.

Find device

For the situation where the IP address of the device is unknown and the port number of 485 is inknown, you can select the search function to quickly find the corresponding device.

Grouping

You can group input channels of the same model using any connection medium, including but not limited to gain, polarity, delay and PEQ.

In the upper right corner of the software, Chinese and English bilingual switching is provided, click the red box to switch.
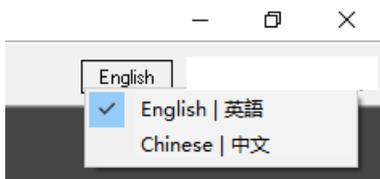
SOFTWARE MAIN INTERFACE

On the main interface of the processor the overall status of the audio paths routing and the order of the signal processing modules are displayed. Note: the signal processing modules here cannot be directly clicked to access, need to be modified separately in the corresponding input or output channels.
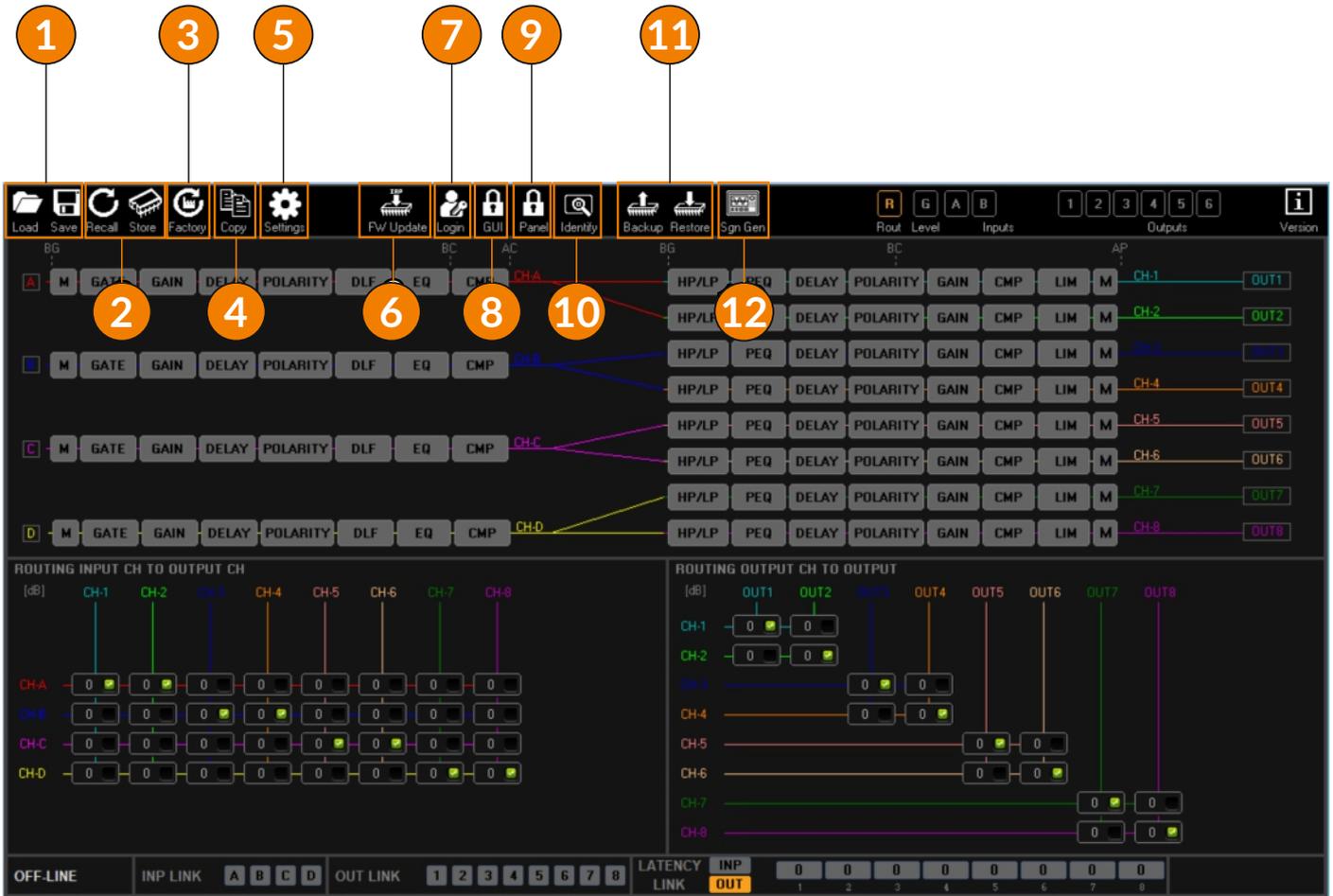
Load & Save: you can save a single preset as a PC file or load a preset from a PC file.

Recall & Store: read the preset from the device; store the current preset to the device.

Use the **Factory (reset)** button to clear the current preset to the factory default state without affecting other presets stored in the machine.

FBT

4

Copy

CHANNEL FUNCTIONS COPY: select (highlight) the channel to be copied, and select the function to be copied at the same type, then the required function can be copied to the corresponding channel.

For example, copy all the contents of output 1 channel to output 3 , just follow the diagram.

5

Settings

SETTINGS: LAN, RS485, channel label, group setting, recall mode, user authority, administrator password, output anti-overflow and other settings.

LAN SETTINGS: DLM series IP address defaults to DHCP, if you need to adjust it to a fixed IP, you need to modify it in.
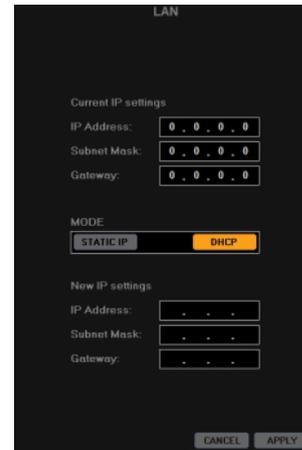
RS 485: Set the COM ID of RS485, the default is 01, the maximum is 32.

CHANNEL LABEL: you can set different labels for the input and output channels (involving the front panel display, only English characters and Arabic numerals are supported).

GROUP RESET: after using the group, the last saved group information is still in the input channel. If you need to remove it, you need to reset the required channels in the group reset.
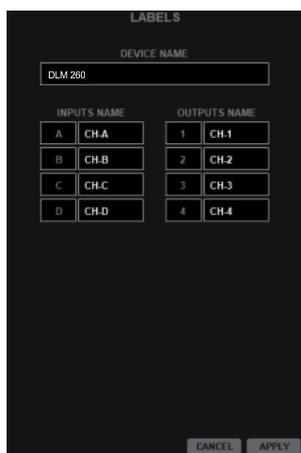
RECALL MODE: when recall presets from the device, if you need to keep the mute state by current setting, you can turn on this function.
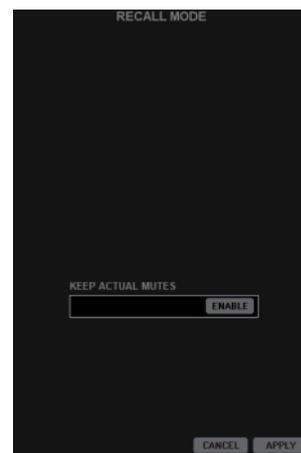
USER AUTHORITY SETTING: when logging in as an administrator, part or all of the output channel functions of the machine can be locked to ensure the integrity of the preset. The administrator password is 111111 by default. Click lock and log out of the administrator account. The locked items will be grayed out and cannot be modified in the software or the front panel. When the parameters are locked and hidden, the hidden area will not be visible in the software or the front panel.

For example: now the administrator has logged in, select all locked items and hide them.

After logging out of the administrator account, the output part / routing part becomes empty.

29
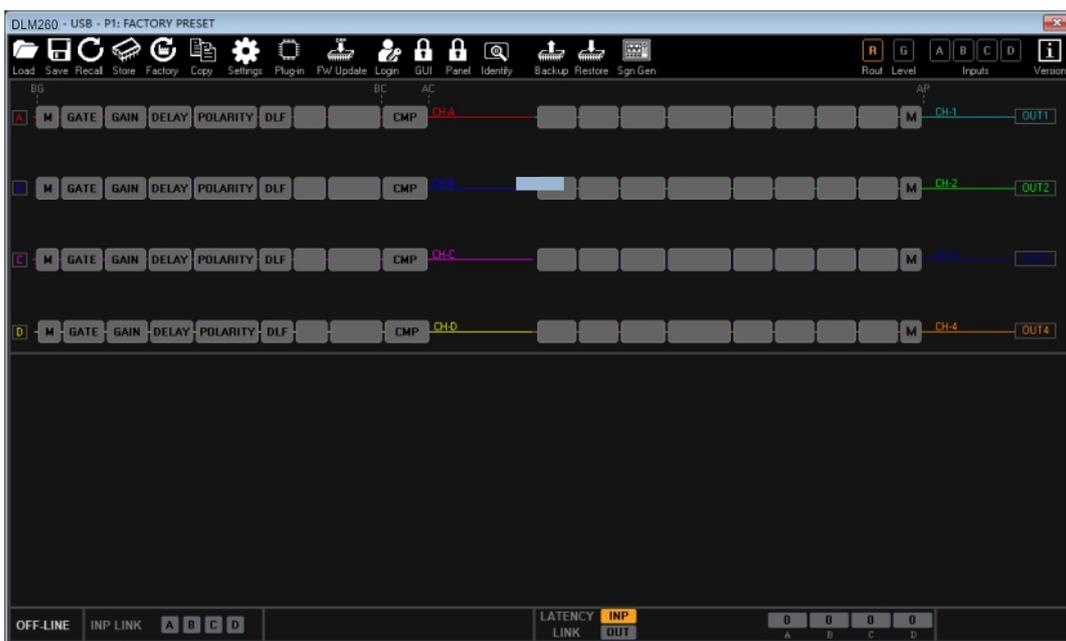
ADMINISTRATOR SETTINGS: modify the administrator password
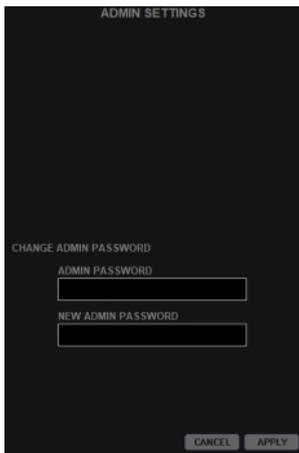
OUTPUT SETTING: the output setting includes a hidden Anti-Clip limiter: Max input level is +20dBu and Max output level is +18dBu, this means that when input is going over +18dBu, if no Limit process on output, the output will be clipped up to +2dB in excess.
The Anti-Clip Limiter is limiting up to +2dBu the output when input is exceeding
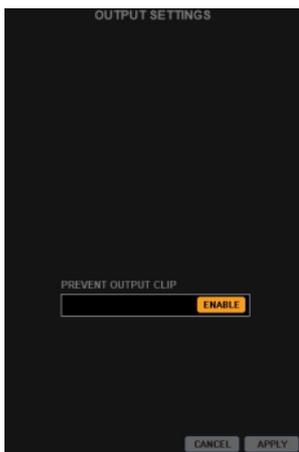
The same applies to dual-channel, that is, two channels can be applied with plug-ins and the corresponding FIR with a sampling rate of 96kHz needs to be used, and the FIR generated by a third party can be imported here (Note: the FIR filter of the input and output channels runs at 48kHz sampling rate, the audio stream is processed by FIR through down-sampling and then up-sampled back to the DSP processing chain. Similarly, the FIR convolution generated by AEQ is also running at a sampling rate of 48kHz , so here 96k FIR can only be imported from the file).

When selecting RTA and advanced signal generator.
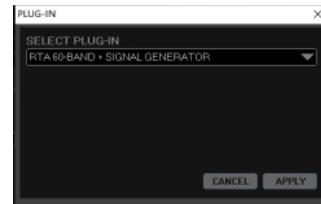
The signal generator and RTA icons will appear.

This is the main interface when the advanced generator and RTA are not selected

6

FW Update

The DLM series provide firmware upgrade function, users can experience the latest version without leaving home, including new plug-ins, bug fixes, new functions, etc... At present , this function needs to be connected via a network cable.

To get the latest firmware you can visit FBT official website.

When using the USB/RS485 connection clicking on the firmware upgrade will pop up such a prompt.
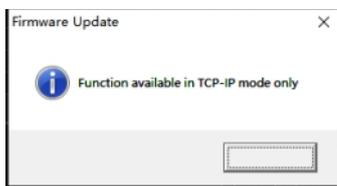
When obtaining the latest firmware IAP firmware package, click Firmware Upgrade. After the serial number and firmware version of the processor are correctly identified, select the correct firmware file "****".sfu.
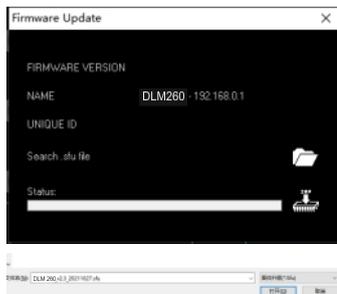
Then Click
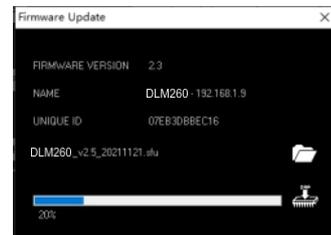
When the progress bar is completed the machine will automatically restart and the upgrade is complete.
TIPS: The upgrade process takes about 1 minute. If the progress bar freezes or fails to enter the system after the upgrade, please retry the above steps to ensure that the machine's power supply and PC software are running normally during the upgrade process. If the upgrade fails, you need to contact your FBT sales staff.

7

Login

LOGIN: Administrator login, the administrator has the highest management authority of this

8

GUI

To lock the **Software Interface** you need to enter a user password.

9

Panel

The device panel lock need to enter the user password, the front panel is unavailable after being locked and it can be restored after restarting the processor. When the DLM processor is online the front panel and software can be adjusted at the same time instead of locking the panel.
When you have to leave your position in front of the PC sw

10

Identify

After pressing the identification button the level lights on the front panel of the corresponding device will all light up and flash for 10 seconds, which is used to quickly find the currently edited one among multiple devices. And every time the PC software is operated the three white

11

Backup

Restore

For the copy of all presets and functions of the whole machine, all 32 preset and network settings can be quickly imported into the new machine, which is very convenient for the migration or backup of the whole machine for engineering companies and OEM customers.

12

Sgn Gen

Advanced generator provides a choice of insert position, which can be selected before the input channel processing (after A/D), after the input channel processing or output (before D/A) when the input channel is selected after processing or output (before D/A), when select INPUT PRE-PROC, there are two noises: pink noise / white noise.
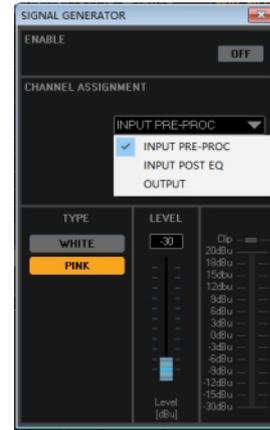
When selecting input post EQ or output, different filter types can be selected, including filtered white noise, filtered pink noise, sine wave, sweep sine wave, etc.., which can only be sent to 1 channel of the input/output.

FBT

SIGNAL ROUTING PART

SIGNAL FLOW CHART: Indicate the processing path after analog signal input A/D conversion, including routing status, plug-in status, etc..It is only for display and cannot be clicked to enter the modification. If you need to make changes you can make detailed modifications in the input and output interface.
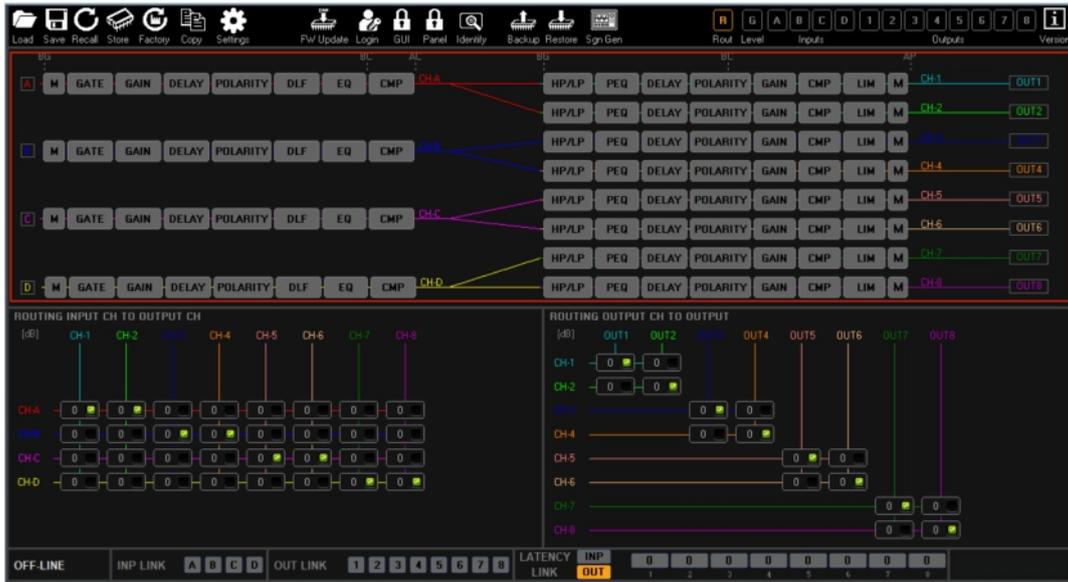
In the input matrix you can arbitrarily route the signal from the input processing channel to the output processing channel. The default sending volume is 0dB which is equivalent to group output. When choosing different sending volume it is equivalent to real Mixing Matrix sending.
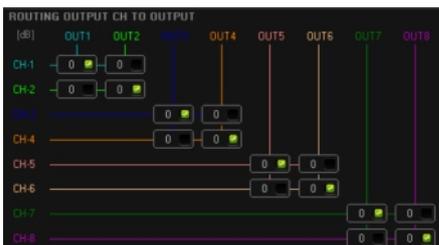
Similarly, the output matrix is responsible for routing the output channel to the physical output channel. The clever use of input and output routing can achieve powerful functions that cannot be achieved by conventional processors, such as multi band compression.

In example, use the processor A channel input, the output channel 1 is connected to a 2 way passive speaker and the route is a shown in the above figure. The output channel uses a DLM Linear phase filter to perform a virtual X-over on the speaker (it is better to know the crossover point), so that different compressor activation thresholds can be set for the high and low drivers and, at the same time, this setting will not destroy the original amplitude and phase of the speaker can add some color or an extra protection as required.

The specific method is to assume that the output channels need to be linked. When the output channel linkage is turned on, turn on the "keep linked CHs level offset" at the red circle, so that the output channel can be linked while maintaining the original ratio.

The channel level page displays the input post-A/D and output pre-D/A signal levels of all channelsa, corresponding to the physical level meter on the front panel, where you can intuitively adjust the channel level and mute.

A new function called "keep linked CHs level offset" has been added. The conventional linkage channel will link the level faders together. For some active speakers the output level ratio should remain the same. When this happens select "keep linked CHs level offset".

INPUT PART

Input channel processing includes noise gate, DLf, equalizer, compressor, delay.
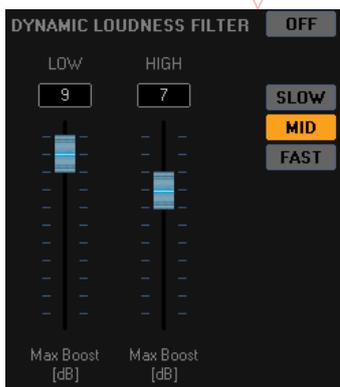
PROCESS 1 DYNAMIC LOUDNESS FILTER:

Working principle: according to the human ear equal loudness curve when the sound pressure is low, the ultra-low and ultra-high frequencies are increased; as the sound pressure level increases, the increase ratio approaches 0, so that a more average sound pressure level can be obtained Sense of hearing.

How to set: First set the maximum allowable boost level, the default is 9dB for low frequency, 7dB for high frequency and the maximum boost is 10dB. At the same time there are three start-up speeds to choose from to adapt to different styles of music.

PROCESS 2 VARIABLE EQUALIZER CONFIGURATION

CONTAINS FOUR CATEGORIES.

1

31 BANDS OF PEQ. PEQ with enough bands the default Q value is 10.5, which is equivalent to 31 bands of graphic equalization (GEQ), in fact, the frequency Q value of each band of equalization and even the filter type can be changed.

2

7 BANDS TARGET EQ AND 24-BANDS PEQ. The design purpose of target EQ is that users can create a small preset that is different from the overall presentation, which can be used for room acoustic correction and stored and recalled independently, which is very convenient.

3

13 BANDS PEQ+512-TAPS FIR FILTER. When you need to use FIR filter on the input channel, you can select this option to provide users with FIR correction at the input, which is suitable for AEQ room

Note: This FIR filter is 48kHz sampling rate 512 taps. Since the overall sampling rate is 96kHz , the sampling rate will down to 48kHz before FIR and after processing, the sampling rate will be re-extended back to 96kHz.

4

6-BANDS PEQ+7-BANDS TARGET EQ+512 TAPS FIR FILTER. The most complete type, can do almost all the EQ configurations you need.

There are as many as 16 types of filters. In addition to the most conventional Bell filter, it also includes high-shev/low-shev/high-low-pass/notch/all-pass, etc.., which can be selected as well.

PROCESS 3 COMPRESSOR: Conventional compressors will provide: threshold, attack time, ratio, release time, gain compensation, soft and hard

The compressor is a very important part of audio processing. It can change the timbre of our output audio or make the sound stable.

The Delay alignment option is provided in the lower right corner of the software, which can be used to align the delays generated by FIR filters with different latency applied to the different channels.

On

OUTPUT SECTION

The overall overview of the output section including crossover, polarity, delay, gain, equalizer, RMS compressor, peak limiter, hard limiter, etc..
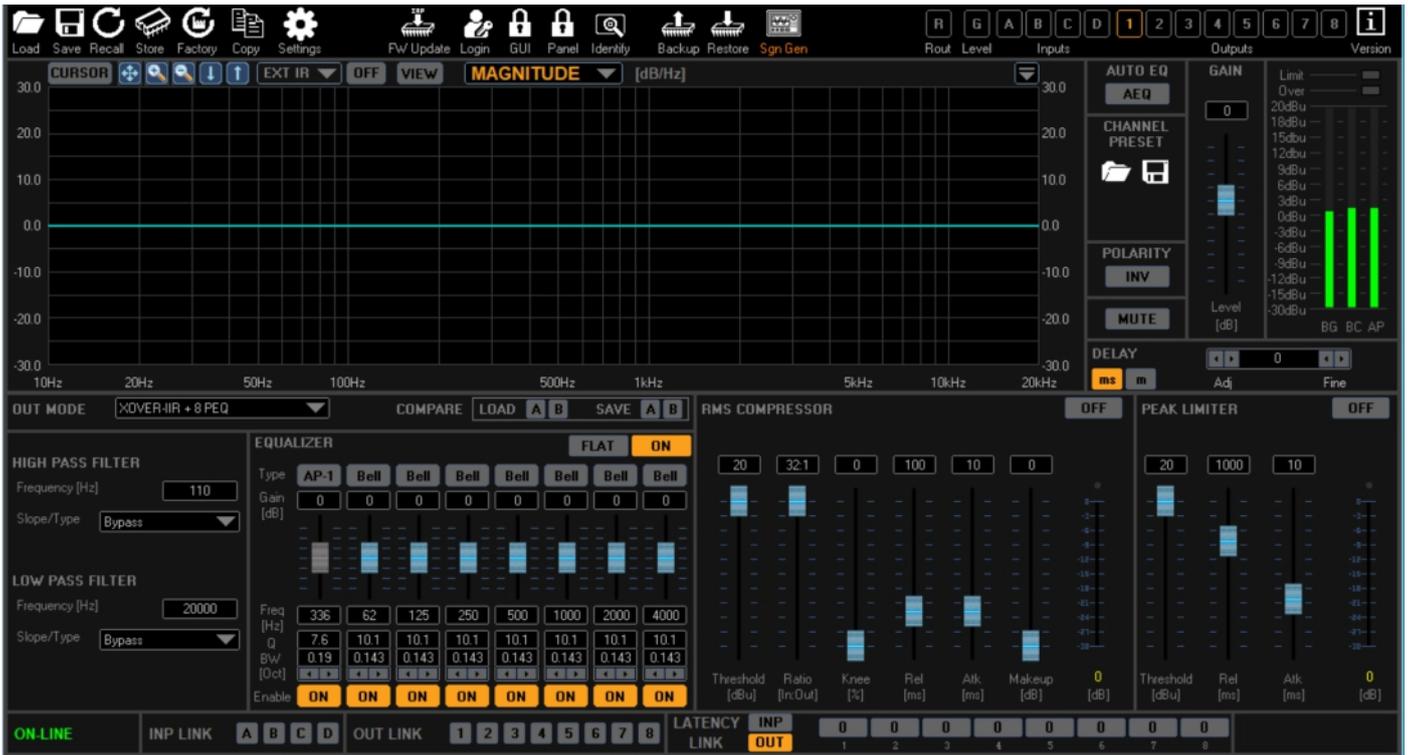
Process 1.   Linear phase filter / FIR filter / FIR + IIR

The conventional IIR filter maintains the slope of the analog filter, but will produce a phase deviation. The higher the order, the more the phase shift, which in turn causes magnitude problems.

In the high and low pass filters of IIR we provide three types of traditional Butterworth / Linkwitz-Riley / Bessel slopes ranging from -6dB/oct to -48dB/oct.

The Linear Phase Filters replicate exactly the slope of the analog filter (LR24 / LR48), there by keeping the phase Linear and easily coupling the phase at the crossover point.
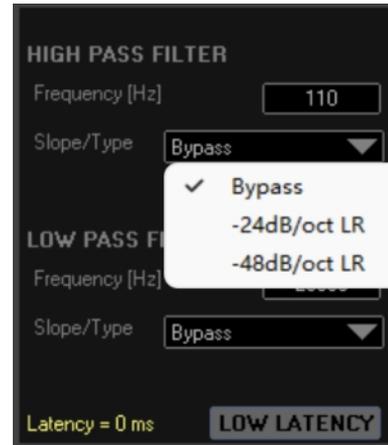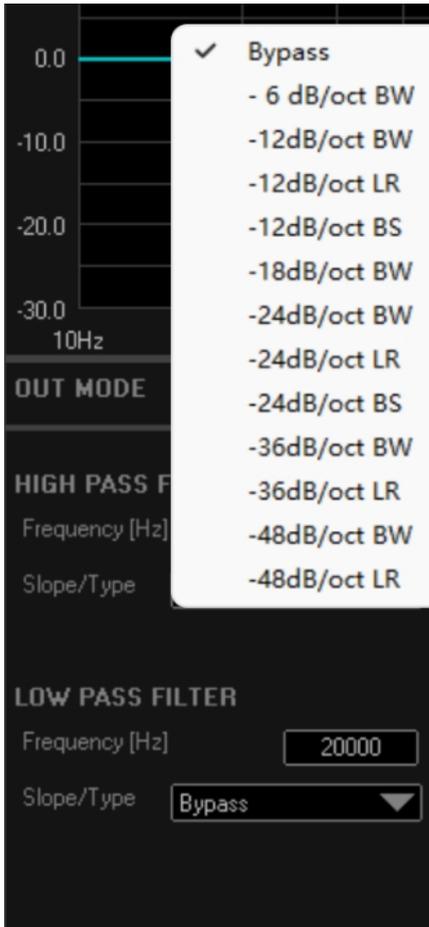
After the low latency mode is turned on, the delay caused by the latency coming with Linear Phase Filters can be further reduced, but the disadvantage is that the in band ripple close to the knee will increase from the default ± 0/01% to ±0.5dB will be generated at the crossover point. However this is not enough to affect the hearing.

For the FIR filter two modes can be selected, FIR+8 segment PEQ, FIR+IIR high and low pass with a slope of up to -24dB + 4 segment PEQ.

You can use a simple FIR filter +8 segment PEQ or you can choose a FIR+IIR mixed mode +4 segment PEQ. There are always a variety of solutions that are suitable for your use scenario.

Click Edit to enter the FIR editing options

Provides three guides of high-pass / low-pass / band-pass, you can enter the frequency that needs to be band-passed according to your needs, choose more or less taps, slopes from -20dB to -120dB/oct, and multiple window function types. Click save at the red circle (here we use band pass filter as an example).

If you need to use external FIR coefficient import, you can select "EXT" in the red circle here, click enter to import FIR coefficient file.

External FIR supports three format of FIR coefficients, namely ".csv", ".txt", ".saf". The first two file formats can be generated by mainstream FIR convolution software and the ".saf" file can be saved and generated from AEQ in the processor. It should be noted that the FIR filter in the input and output part of the DLM series works at a sampling rate of 48kHz and a maximum of 512 taps.
The FIR filter in the plug-in works at a sampling rate of 96kHz and a maximum of 1536 taps.
Compatible FIR convolution software has been tested: rePhase, FIR Designer, FIR Capture, Filter hose.

The output part provides three levels of compression limiter settings, namely RMS compressor, which can be used to set the compression option of the average level over a period of time. When the compression ratiois high, it is the limiter.

The peak limiter has a very low attack time and quickly suppresses the burst signal. It can be used to limit the maximum displacement of the speaker voice coil in conjunction with the X-max-voltage data given by the speaker manufacturer.
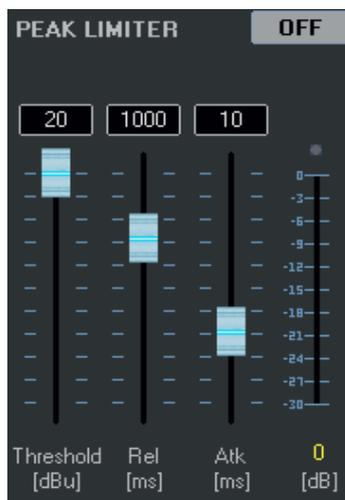
GENERAL

Input Impedance	20K Ohm balanced
Output Impedance	100 Ohm
Max. Input Level	+20dBu
Max. Output Level	+18dBu
Crosstalk	< -95dB
Sample Rate	48kHz
Signal to Noise Ratio	> 116dB (A weighted)
Noise Floor	< -95dB (A weighted)
Common Mode Rejection Ratio	60dB
Frequency Response	20Hz - 20kHz +0.3dB
THD (+4dBu 1kHz)	< 0.003%
Mains Voltage Range	90 - 240Vac / 50-60Hz
Nominal Power Consumption	30W

CONNECTORS

Audio Input	2 x 3 (4 x 3) pin female XLR
Audio Output	6 x 3 (8 x 3) pin male XLR
Ethernet	shielded RJ45, Dynamic or Static IP
RS485	2 x RJ45
USB	1 x USB "B"
Mains	3 pin IEC

PHYSICAL

Height	1U (44mm) (1.75")
Width	482mm (19")
Depth	230mm (9.1")
Weight	3.0kg (6.6 lb)

PROCESSING

Signal Generator	White/Pink noise - Level range: -30dBu +10dBu
Input & Output Gain	-18dB +12dB, step 0.1dB
Noise Gate	Threshold: -80dBu -45dBu
	Attack time: 1ms 1000ms
	Release time: 1ms 1000ms
Dynamic Loudness Filter	Gain range: 0dB -10dB
	Attack speed: fast/medium/slow
Parametric EQ	Input channels up to 31 optional types of EQ
	Output channels up to 8 optional types of EQ
Optional Filter types	Bell classic and constant Q filter, 1st order high Shelf filter, 2nd order high Shelf filter, Variable Q high Shelf filter, 1st order low Shelf filter, 2nd order low Shelf filter, Variable Q low Shelf filter, 1st order low-pass filter, 2nd order low-pass filter, Variable Q low-pass filter, 1st order high-pass filter, 2nd order high-pass filter, Variable Q high-pass filter, Band pass filter, notch filter, 1st order all-pass filter, 2nd order all-pass filter with variable Q value
Center Frequency	Adjustable within the frequency range of 20Hz-20kHz with a step accuracy of 1Hz
Q value / Bandwidth	The Q value range of Bell filter is 0.4 - 128, the step is 0.01, the range of the Q value of the Chevron/highpass/lowpass filter is 0.1 - 5.1 and the step is 0.01, the value range of bandpass /notch filter Q is 4 - 104 and step is 1
EQ Gain range	-15dB +15dB
IIR Crossover Filter	Butterworth slope: 6/12/18/24/36/48dB per octave, Bessel slope: 12/24dB per octave, Linkwitz-Riley slope: 12/24/36/48dB per octave
Linear Phase Filter	Linkwitz-Riley: 24/48dB per octave
FIR Crossover Filter	Type: high pass/low pass/band pass/external import Taps range: 256 - 512, slope range 21 - 120dB per octave Time window type: Rect/Sinc/Keiser/Hanning/Hamming/Blackman/Blackman-Harris/Blackman-Nuttal/Nuttal/Keiser-Bessel/Sine
RMS Compressor	Starting threshold range: -10dBu +20dBu
	Compression ratio range: 2 32:1
	Soft and hard knee: 0 - 100%
	Attack time: 0.1ms - 1000ms
	Release time: 10ms - 15000ms
	Gain compensation: Max 12dB
Peak Limiter	Threshold range: -10dBu +20dBu
	Attack time: 1ms - 1000ms
	Release time: 10ms - 3000ms
Delay	The adjustable delay time of each input channel + output channel is 452ms, step accuracy 0.0104ms
FIR Filter	Each input channel and output channel can import FIR filter with 48kHz sampling rate and 512 taps