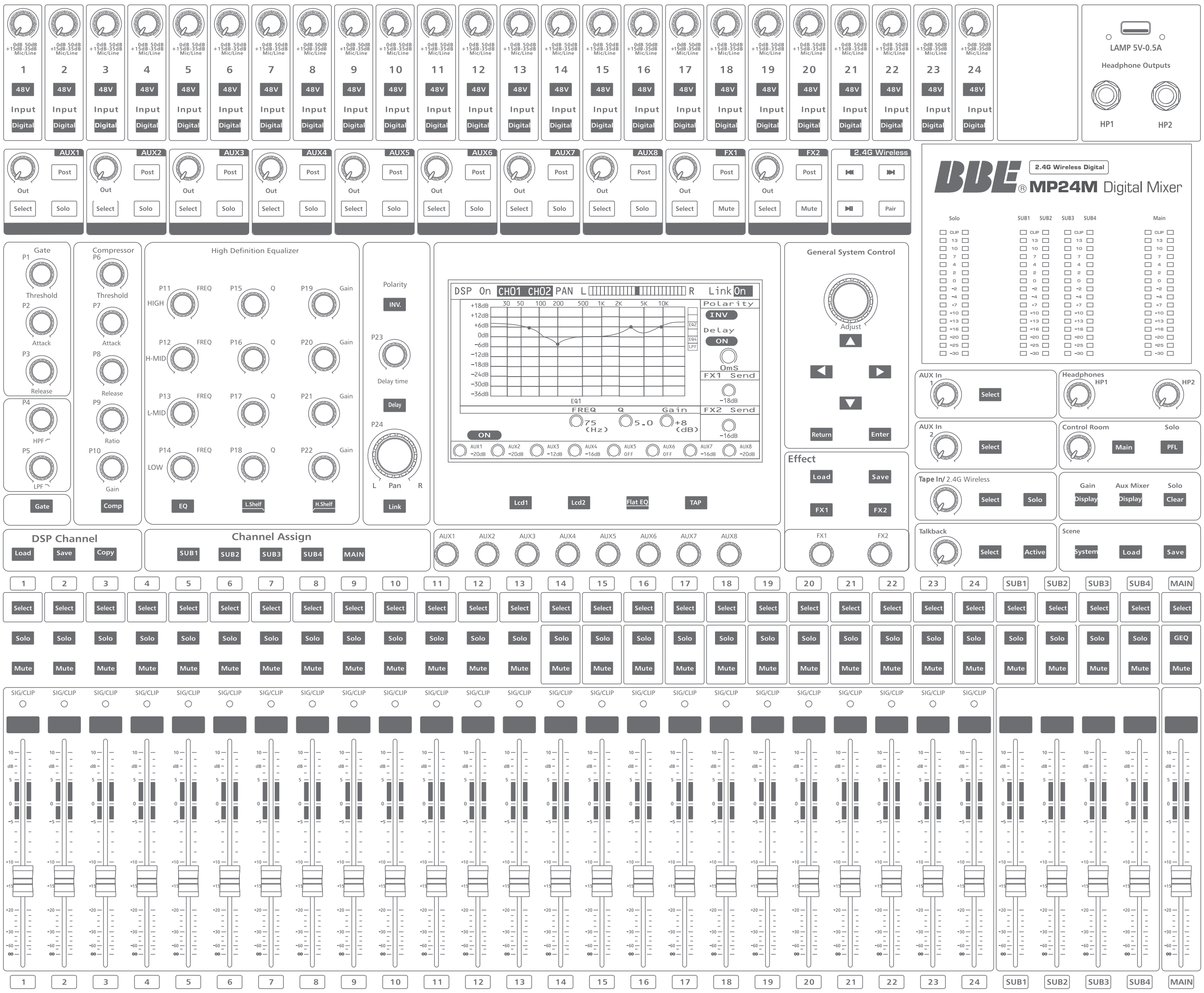
User´s Manual

BBE® MP24M

24-Channel Digital Mixer

Featuring the BBE Sonic Maximizer® Processor



Important Safety Instructions

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

This symbol, wherever used, alerts you to the presence of un-insulated and dangerous voltages within the product enclosure. These are voltages that may be sufficient to constitute the risk of electric shock or death.

This symbol, wherever used, alerts you to important operating and maintenance instructions.

Please read.

Protective Ground Terminal

AC mains (Alternating Current)

AC mains (Alternating Current)

ON: Denotes the product is turned on.

OFF: Denotes the product is turned off.

WARNING
Describes precautions that should be observed to prevent the possibility of death or injury to the user.

CAUTION
Describes precautions that should be observed to prevent damage to the product.
Disposing of this product should not be placed in municipal waste but rather in a separate collection.

WARNING
Power Supply
Ensure that the mains source voltage (AC outlet) matches the voltage rating of the product. Failure to do so could result in damage to the product and possibly the user. Unplug the product before electrical storms occur and when unused for long periods of time to reduce the risk of electric shock or fire.

External Connection
Always use proper ready-made insulated mains cabling (power cord). Failure to do so could result in shock/death or fire. If in doubt, seek advice from a registered electrician.

Do Not Remove Any Covers
Within the product are areas where high voltages may present. To reduce the risk of electric shock do not remove any covers unless the AC mains power cord is removed. Covers should be removed by qualified service personnel only.

No user serviceable parts inside.

Fuse
To prevent fire and damage to the product, use only the recommended fuse type as indicated in this manual. Do not short-circuit the fuse holder. Before replacing the fuse, make sure that the product is OFF and disconnected from the AC outlet.

Protective Ground
Before turning the unit ON, make sure that it is connected to Ground. This is to prevent the risk of electric shock.
Never cut internal or external Ground wires. Like wise, never remove Ground wiring from the Protective Ground Terminal.

Operating Conditions
Always install in accordance with the manufacturer's instructions.
To avoid the risk of electric shock and damage, do not subject this product to any liquid/rain or moisture.
Do not use this product when in close proximity to water.
Do not install this product near any direct heat source.
Do not block areas of ventilation. Failure to do so could result in fire.
Keep product away from naked flames.

IMPORTANT SAFETY INSTRUCTIONS
Read these instructions
Follow all instructions
Keep these instructions. Do not discard.
Heed all warnings.
Only use attachments / accessories specified by the manufacturer.

Power Cord and Plug
Do not tamper with the power cord or plug. These are designed for your safety.
Do not remove Ground connections!
If the plug does not fit your AC out let seek advice from a qualified electrician.
Protect the power cord and plug from any physical stress to avoid risk of electric shock.
Do not place heavy objects on the power cord. This could cause electric shock or fire.

Cleaning
When required, either blow off dust from the product or use a dry cloth.
Do not use any solvents such as Benzol or Alcohol.
For safety, keep product clean and free from dust.

Servicing
Refer all servicing to qualified service personnel only.
Do not perform any servicing other than those instructions contained within the User's Manual.

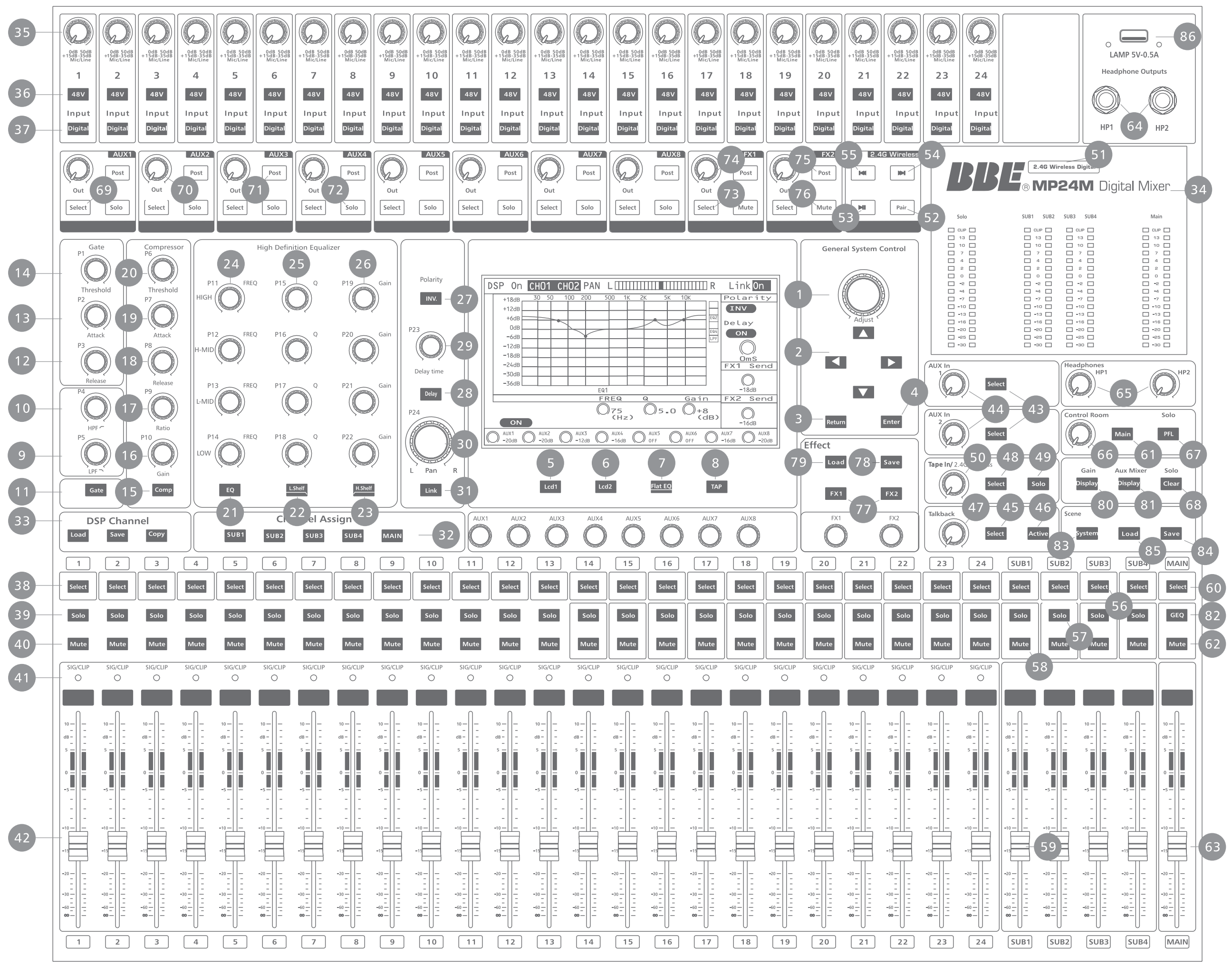
PORTABLE CART WARNING

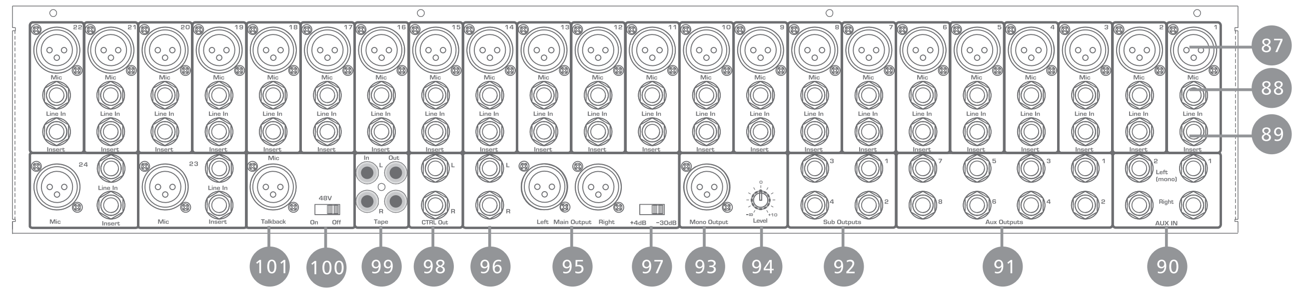
Carts and stands - The component should be used only with a cart or stand that is recommended by the manufacturer.
A component and cart combination should be moved with care. Quick stops, excessive force, and uneven surfaces may cause the component and cart combination to overturn.

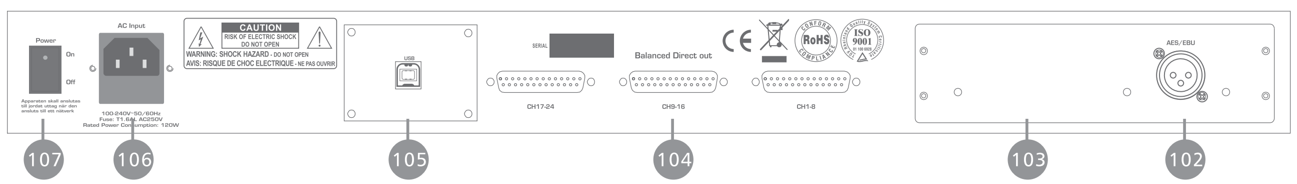
Table of Contents

Visit www.BBEsound.com for more information about other BBE® products.

Console Controls And Rear Panel Connections







Control and Rear Panel Connection Index

1

Introduction

Thank you for purchasing the BBE MP24M digital mixer, the only digital mixing system available with BBE® Sonic Maximizer® processing. With features like twenty-four high-head-room, class A MASS® microphone preamps; processing with a virtual rack of fourteen 31-band graphic EQs; channel compressor, gate, delay, polarity reverse, DSP effects; eight AUX sends and four Subgroups all with full DSP processing; accurate LED metering; load/save/copy for all mixer settings; dedicated Talkback mic input and much more, the BBE MP24M is designed to create an unparalleled audio mixing experience. It is simple and intuitive to operate even though it is a powerful mixing system.

We suggest that you use this manual to familiarize yourself with the features and applications of your BBE MP24M before using.

Summary of Features

- Features BBE's Sonic Maximizer® processing
- 24 ultra-high headroom BBE MASS® Class A, Twin Servo mic preamps
- 2 Stereo AUX line Inputs
- 4 Subgroups
- Dedicated Mono Output with Variable Level Control
- 2 Digital Effects Engines Featuring BBE's Proprietary FX Library
- 8 AUX Sends
- 15 Total Analog Outputs (L/R Stereo Mains, Mono Main, 1-4 Subgroups, 1-8 AUX Sends)
- Assignable AES/EBU Output
- 7-inch full-color LCD display
- 100 Scenes can be Stored for Recall Anytime
- MP MIX Software Allows Remote Mixing from Laptop PC (Direct USB or Via Wi-Fi Router)
- High-definition 24 bit/48 kHz A/D D/A converters
- 32-bit floating-point SHARC DSP Processors
- Dedicated Talkback mic input, fully assignable, w/EQ
- High and Low Pass Filters
- Full Function Gate and Compressor/Limiter
- 4-band fully parametric EQ on every input and output
- Phase Reverse
- Channel Delay
- 31 Band 1/3 Octave Graphic EQ Available on all analog Outputs
- Odd/Even Input Channels can be Linked
- Odd/Even AUX Sends and Sub-Groups can be Linked
- 2.4G Wireless Connectivity for Stereo Playback
- 100mm Long Throw Faders

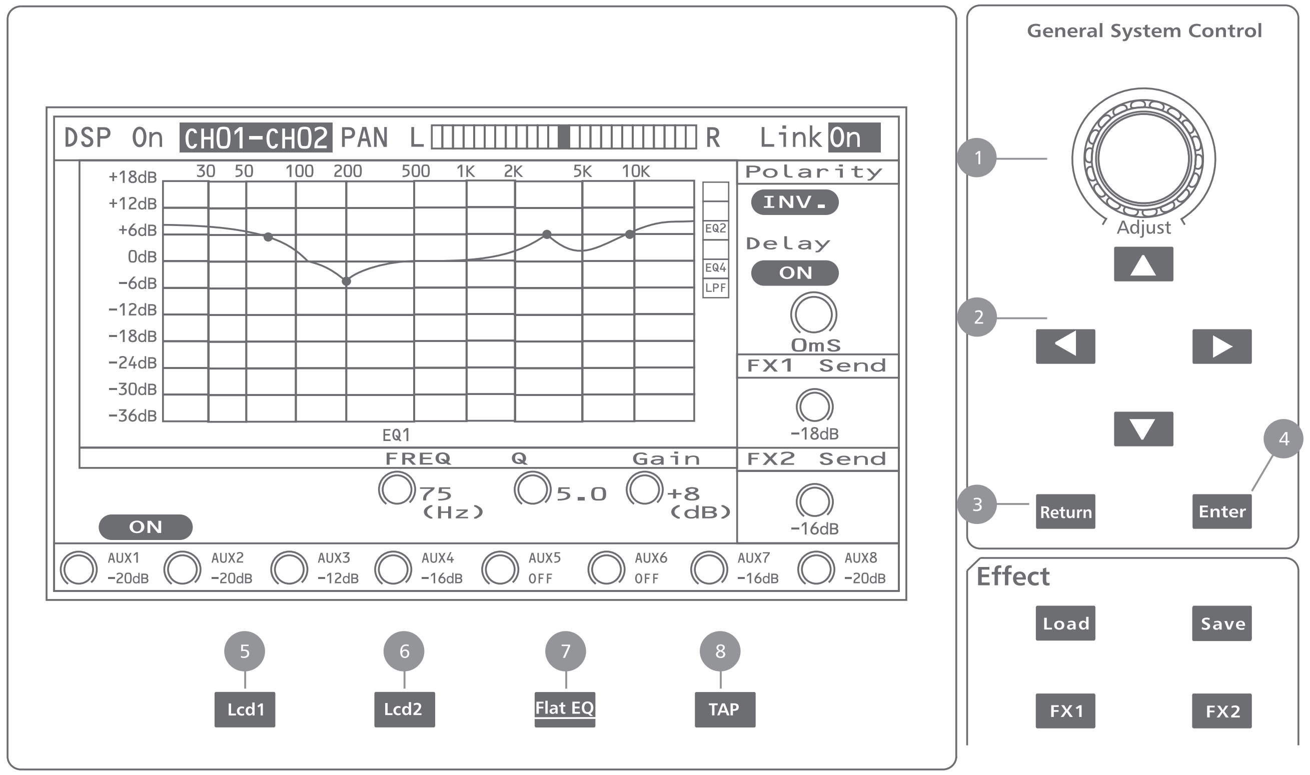
Purchase Date

Serial Number:
Date of Purchase:

2

Controls

Function Buttons



1) Adjust Encoder

This encoder adjusts the parameter values that are shown on the display. Turning it clockwise increases the value and counterclockwise decreases the value. This encoder also enables you to scroll through a displayed list and select a character for entry. Since the parameters controlled by this encoder vary from function to function, please note the on-screen instructions.

2) Left, Right, Up, Down

These buttons move the cursor around the display page, to select and delete parameters and options. Holding down a button moves the cursor continuously in the corresponding direction. Since the parameters controlled by these buttons vary from function to function, please note the on-screen instructions.

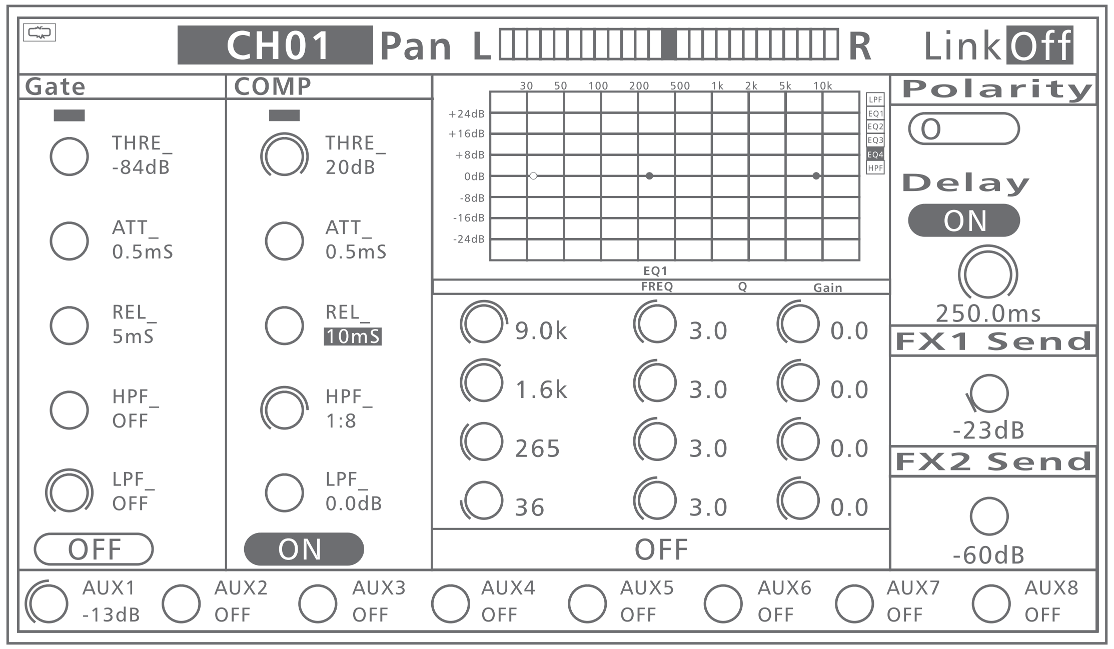
3) Return

Press this button to return to the previous page or exit.

4) Enter

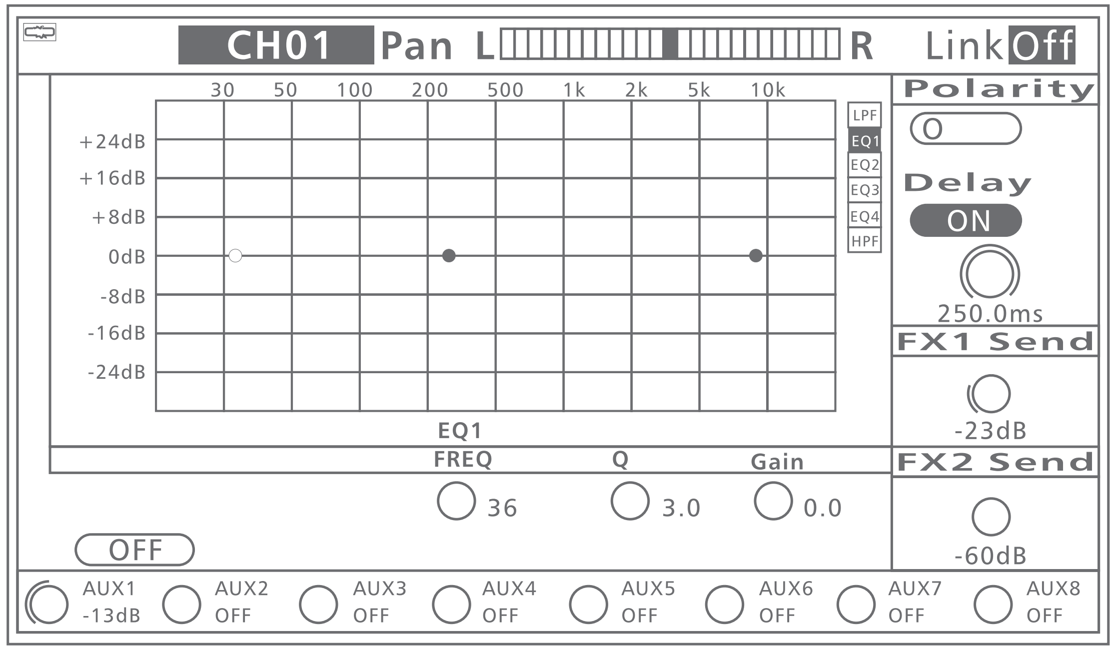
- Activate a selected function.
- Confirm the edited parameter values.

Controls



5) LCD1

- This button engages the primary LCD display which shows all of the corresponding input or output channel's DSP functions. It will illuminate to indicate that the LCD1 has been enabled.
- It also can be used as a selecting button.
- Since the parameters controlled by these buttons vary from function to function, please note the on-screen instructions.



6) LCD2

- This button engages the secondary LCD display showing the EQ function in greater detail for that corresponding input or output. It will illuminate to indicate that it has been enabled.

- It also can be used as a selecting button.
- Since the parameters controlled by these buttons vary from function to function, please note the on-screen instructions.

7) Flat EQ

- Set the selected channel's EQ to flat when the FLAT EQ function is enabled. EQ must be enabled.
- Delete preset. Press the **LOAD** button to display the Effect, GEQ, Scene or DSP channel preset list and choose the preset which you want to delete, then press the **FLAT EQ** button and follow the instructions that are shown on the LCD display.
- When saving Effect, GEQ, Scene or DSP channel settings, it can be used as the Caps Lock key, it functions the same as the Caps Lock key on a keyboard.

8) TAP

- Tempos can be tapped in to adjust the delay time on Delay and St Delay (Stereo Delay) functions in the FX1 and FX2 effects engines. The value will be shown in the TIME fields of the effects parameters
- TAP can also be used to assign GEQ to the various output buses.

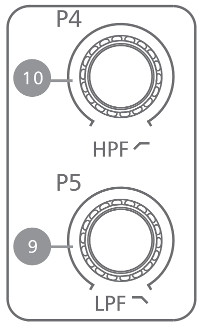
DSP Functions

The DSP functions on each input channel and output bus are the most important aspect of the MP24M. Here you can enable and control Gate, Compressor, EQ, Polarity, BBE Sonic Maximizer®, Panning, and Delay functions, as well as linking and routing for the selected channel or output bus.

BUS	Gate	Compressor	EQ	Polarity	Pan	Delay	BBE Sonic Maximizer®	Link	Output Assignment
Inputs (CH1 -24)	V	V	V	V	V	V	V	V	MAIN, Subgroups 1-4, AUX Sends 1-8, Internal AUX Sends 1-2
AUX Sends (1 - 8)	V	V	V	V	V		V	V	
Internal FX Sends 1 & 2			V						MAIN, Subgroups 1-4, AUX Sends 1-8
Subgroups 1 - 4	V	V	V	V	V	V	V	V	MAIN
MAIN Out	V	V	V	V		V	V		
AUX Ins 1 & 2			V				V		MAIN, Subgroups 1-4, AUX Sends 1-8
Tape In/ 2.4G Wire-			V						MAIN, Subgroups 1-4, AUX Sends 1-8
Talk Back			V						MAIN, Subgroups 1-4, AUX Sends 1-8

2 Controls

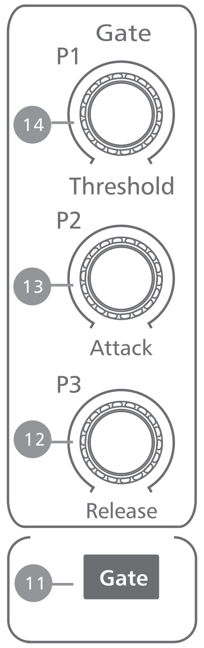
Low/High Pass Filter



9) Low Pass Filter
A low-pass filter passes low frequencies while attenuating higher frequencies. The low-pass filter's threshold can be set from 21Hz to 19.2 kHz. When set to its highest position, the filter is off. The slope of the Low Pass Filter is -12dB/octave.

10) High Pass Filter
A high-pass filter is a filter that passes higher frequencies but attenuate lower frequencies. The high-pass filter's threshold can be set from 21Hz to 19.2KHz. When set to its lowest position, the filter is off. The slope of the High Pass Filter is -12dB/octave.

Noise Gate



A noise gate attenuates signals below the set threshold and allows signals to pass through only when they are above the set threshold.

11) Gate
This button engages the Noise Gate for the selected channel or output bus. It will illuminate to indicate that the Gate has been enabled. The LCD display shows the Gate's settings. Its parameters can be adjusted by rotating P1~P3 knobs directly or by using the up & down & left & right keys to choose the function that you want to modify and using the Adjust Encoder wheel to set the value. Please notice that only if the Gate has been enabled can its parameters be adjusted.

- This also engages the HPF (High Pass Filter) and LPF (Low Pass Filter)

12) Release
The Release sets the amount of time it takes for the gate to close. It can be set from 0.5 to 200 ms. A fast release abruptly cuts off the sound once it has fallen below the threshold, a slower Release smoothly changes from open to closed, much like a slow fade out.

13) Attack
The Attack control sets the time for the gate to change from closed to open, much like a fade-in. It can be set from 0.5 to 200ms.

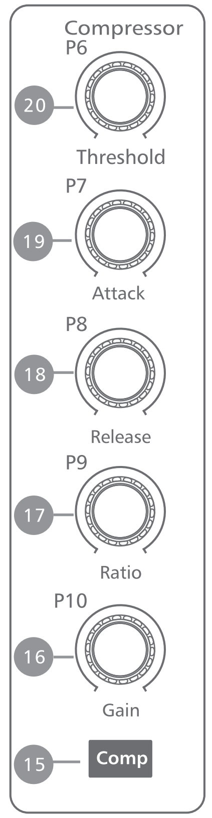
14) Threshold
The threshold control sets the level at which the gate will close. It can be set from 0 to-84dB.

BBE MP24M

Controls 2

Compressor

A compressor reduces the level of an audio signal if its amplitude exceeds a certain threshold.



15) Comp
This button engages or disengages the Compressor for the selected channel or output bus. It will illuminate to indicate that the compressor has been enabled. The LCD display shows the compressor setting in real time. Its parameters can adjusted by rotating P6~P10 knobs directly or by using the up & left & down & right key to choose the function that you want to modify and using the Adjust Encoder to set the value. Please notice that only if the Comp button has been enabled can its parameters be adjusted.

16) Gain
This encoder sets the output gain of the compressor for the selected channel or bus. The gain can be set from 0dB (no gain adjusted) to +24dB.

17) Ratio
This control sets the compression ratio for the selected channel or output bus. The ratio determines the amount of gain reduction. For example, a ratio of 4:1 means that if input level is 4 dB over the threshold, the output signal level will be 1 dB over the threshold. The ratio can be set from 1:1 to 10:1.

18) Release
This control sets the release time of the compressor for the selected channel or output bus. Release sets the length of time the compressor takes to return to its normal gain once the signal level drops below the threshold. Release can be set from 10 ms to 1 second.

19) Attack
This control sets the compressor's attack time for the selected channel or output bus. The attack time is the period when the compressor is decreasing gain to reach the level that is determined by the ratio. You can set the attack from 0.5 to 200 milliseconds.

20) Threshold
This control sets the compressor threshold for the selected channel or output bus. When the amplitude of the audio signal exceeds the threshold, the compressor will reduce the level of this signal by the set ratio. The threshold can be set from -30 to 20 dB.

2

Controls

High Definition Equalizer

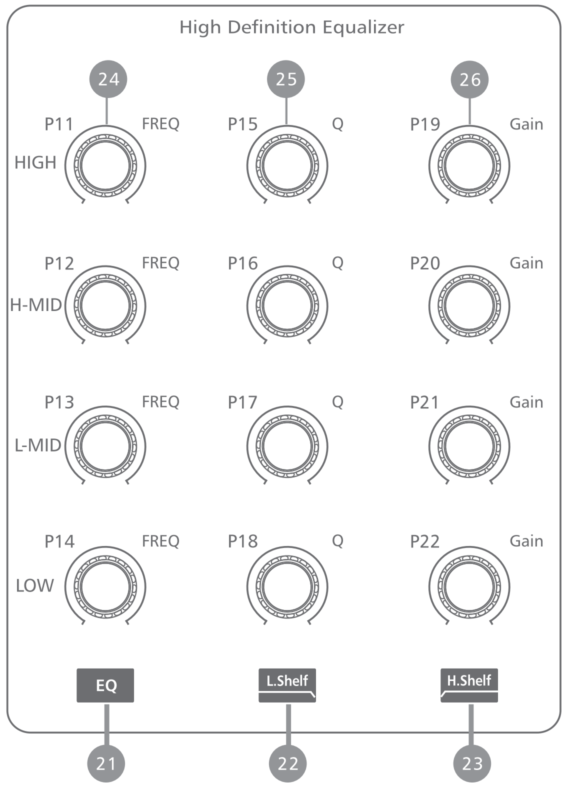
An equalizer is a filter that allows you to adjust the level of a frequency, or range of frequencies, in an audio signal. Make sure SELECT has been pressed on the channel or output you wish to EQ.

21) EQ
This button engages the equalizer for the selected channel or output bus. It will illuminate to indicate that the equalizer has been enabled. The LCD display shows the EQ setting in real time. Its parameters can be adjusted by rotating P11~P22 knobs directly or by using the up & left & down & right key to choose the function that you want to modify and using the Adjust Encoder to set the value. Please notice that only if the EQ button has been enabled can its parameters be adjusted.

22) Low Shelf
Engaging Low Shelf changes the low band into a low shelving EQ which means a band of low frequencies at and below a user-set shelving frequency. The shelving frequency can be set by rotating the Frequency control.

23) High Shelf
Engaging High Shelf changes the high band into a high-shelving EQ which means a band of frequencies at and above a user set shelving frequency. The shelving frequency can be set by rotating the Frequency control.

24) Freq
These set the center frequency of the equalizer's Low/Low-mid/High-mid/High band separately. The center frequency is the middle of the pass-band between the lower and upper cutoff frequencies which define the limits of the band. The center frequency can be set from 21Hz to 19.2K Hz.



25) Q
These set the Q for the Low/Low-mid/High-mid/High band separately. The Q is the ratio of the center frequency to the bandwidth. If the center frequency is constant, the bandwidth is inversely proportional to the Q, which means that if you raise the Q, the bandwidth will be narrowed. It can be adjusted from 0.4 to 24.

26) Gain
These set the cut or boost gain at the center frequency for the Low/Low-mid/High-mid/High bands separately. It can be set from -24 to +24 dB.

BBE MP24M

Controls

2

Polarity & Delay & Pan & Link

27) INV
Press this button to invert the phase of the selected channel's signal (to alter the phase by 180°). If the phase reverse is active the button will illuminate. The LCD display shows the phase reverse setting in real time

28) Delay
This button engages and disengages the delay for the selected channel. It will illuminate to indicate that the delay has been enabled.

29) Delay Time
The control adjusts the delay time of the selected channel.

The LCD display shows the delay time in real time. It can be set from 0.0ms to 500ms. Please notice that only if the Delay button has been enabled can its parameter be adjusted.

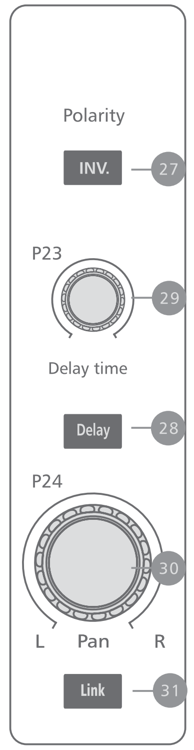
30) Pan
Controls panning for the selected input or output. The LCD display shows the setting in real time. If two channels have been linked as stereo pair, the LCD display will automatically change to stereo pan.

31) Stereo Link
Input channels, AUX buses, and Subgroups can be linked as a stereo pair. The stereo pairs are predefined and cannot be changed.

They are as follows:

Channels 1 & 2	Channels 19 & 0
Channels 3 & 4	Channels 21 & 2
Channels 5 & 6	Channels 23 & 4
Channels 7 & 8	AUX1 and AUX 2
Channels 9 & 10	AUX 3 & AUX 4
Channels 11 & 12	AUX 5 & AUX 6
Channels 13 & 14	AUX 7 & AUX 8
Channels 15 & 16	Subgroups 1 & 2
Channels 17 & 18	Subgroups 3 & 4

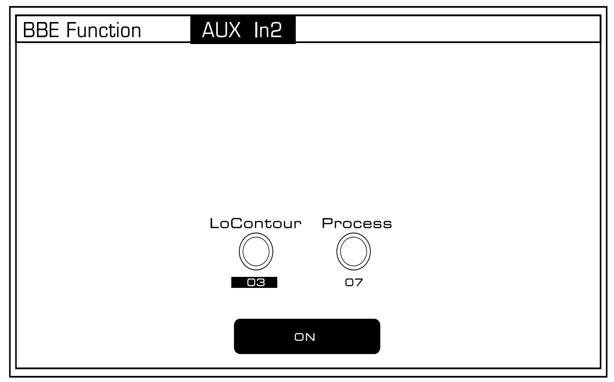
A stereo link can be enabled when either channel in the pair is selected by pressing the Link button. When the Stereo Link function is enabled, all DSP settings, Subgroup assignments, solo status and main assignments are passed to the other channel in the pair.



Please note that this is a nondestructive function, the other channel's previous setting will be restored after the Link button is disengaged. For example, if Channel 6 has been selected when the Stereo Link button is engaged, all of Channel 6's setting will be copied onto Channel 5. Channel 5's own setting will restore after the Link button has been disengaged.

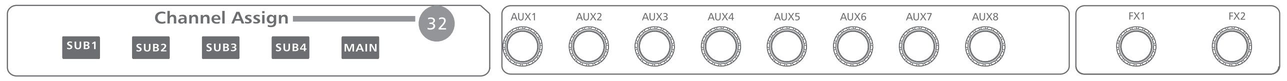
BBE Sonic Maximizer®

To engage the BBE Sonic Maximizer®, select the desired channel or output and then press the TAP button to access the control panel. Using the left and right arrow keys will toggle the between Lo Contour and Process functions; then use the Adjust Encoder to change the value. The Flat EQ and LCD2 buttons will turn the Process on and off for comparison.



Controls

Output Assignment



32) Channel Assign

A selected channel can be assigned to Sub group outputs 1-4 and Main outputs by pressing the corresponding button. They can also can be assigned to AUX sends 1-8 and FX Sends 1-2 by rotating the corresponding knob as well as adjusting the output level of the AUX or FX sends individually.

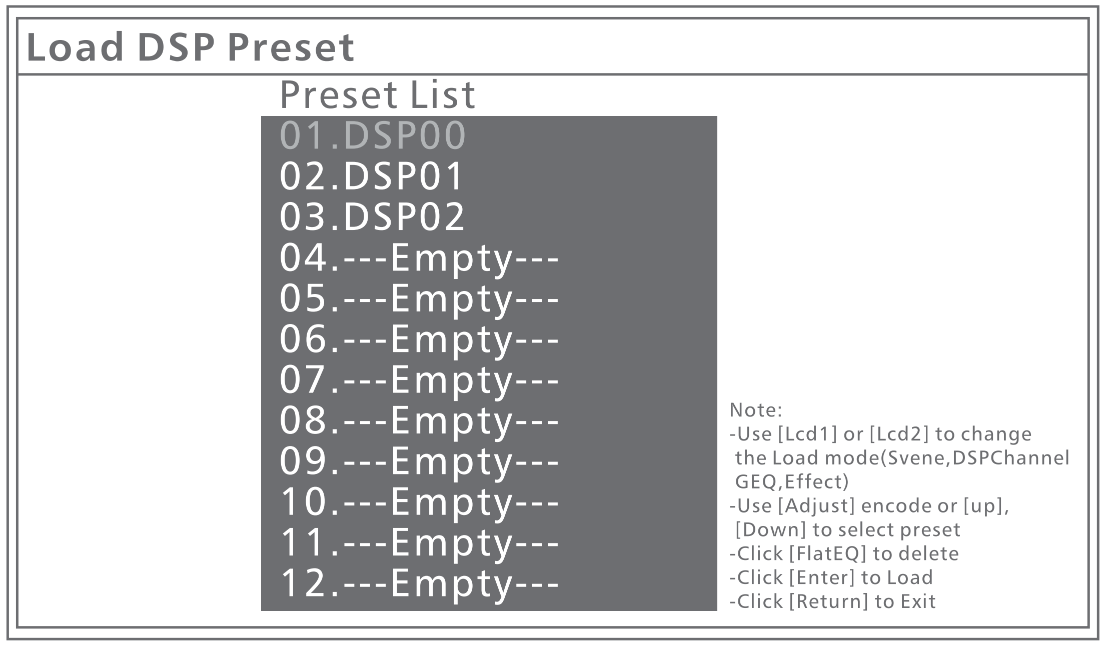
The twenty four Mic/Line inputs, the two AUXiliary inputs, Tape In, Talkback, 2.4G Wireless and FX 1 and 2 returns can be assigned to any or all of the Subgroup outputs, AUX Sends and the MAIN outputs.

Subgroups can only be assigned to the Main outs. The eight AUX sends cannot be assigned to a Subgroup or to the Main outputs. Only the twenty four channel inputs can be assigned to the FX sends.

DSP Load, Save,Copy

33) DSP Channel Load, Save, Copy

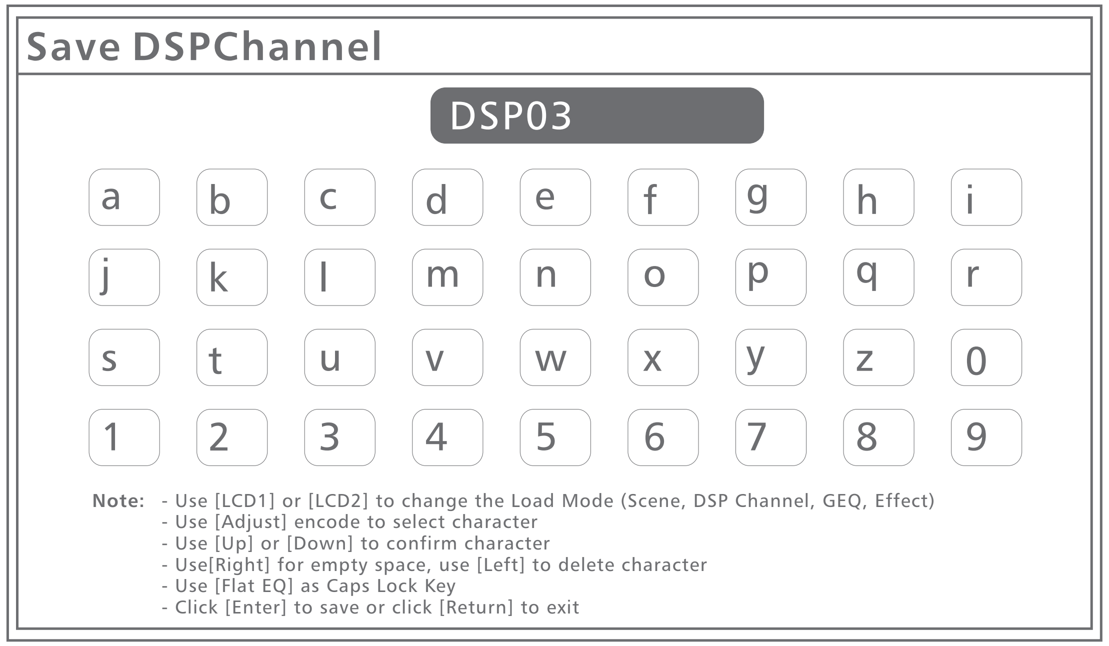
Using these buttons you can save the settings of a selected channel or output for future use, load a saved DSP user-defined preset or copy one channel's setting to another channel, or channels.



Load

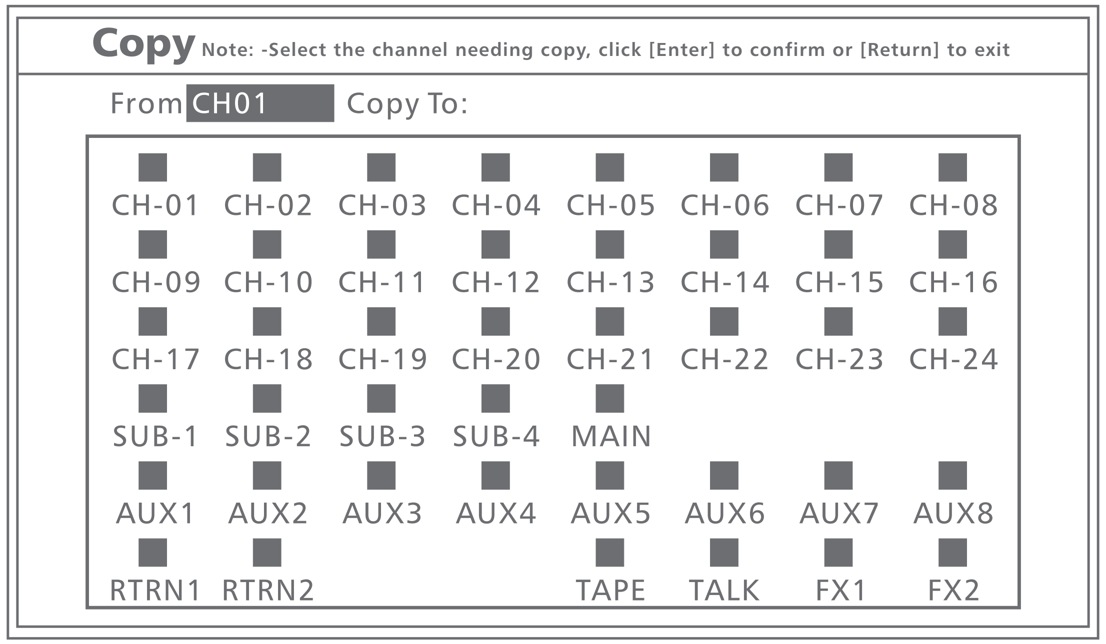
Press this button to select a DSP preset, then press **ENTER** to **LOAD**. By using the **LCD1** and **LCD2** buttons you can also select Scene, GEQ or FX user-defined preset libraries. The preset can be deleted by pressing the **FLAT EQ** after it has been chosen. Please note the instructions that are shown on the LCD display.

Controls



Save

Press this button to save the selected channel's or output bus's DSP setting as user-defined DSP preset for future use. Please note the instructions that are shown on the LCD display.



Copy

The selected channel's or output bus's setting can be copied onto other channels or buses by pressing the Copy button. The selected channel or bus will flash after the Copy button is pressed, you can then **SELECT** as many channels as you want to copy to. Pressing the **ENTER** button will confirm or press the Return button to exit. Please note the instructions that are shown on the LCD display.

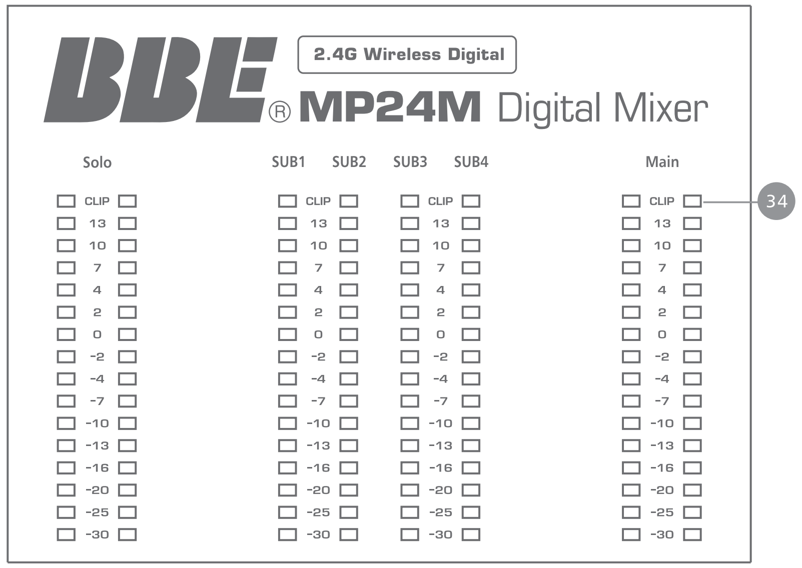
2 Controls

Metering

34) LED Meters
The LED meters show the signal levels.

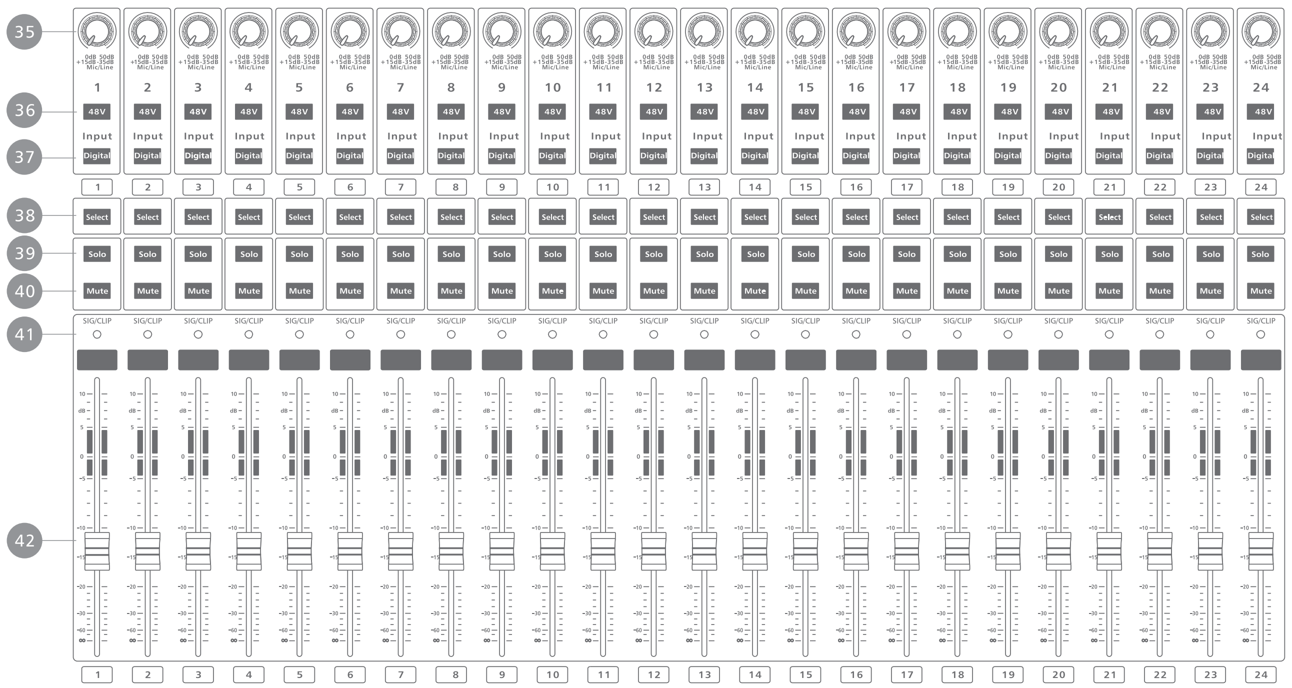
- Solo meters indicate the level of the Solo bus.

- SUB1~4 & Main meters indicate the output level of SUB1~4 & Main bus.



Input Channels

The BBE MP24M Digital Console has a total of thirty-one inputs: twenty-four Mic/Line input channels, two Stereo AUX inputs, Stereo Tape Return and a dedicated Talkback mic input.



Controls 2

35) Input Gain
The knob controls the gain of the channel's input.

It is very important to properly set the level of the input gain to minimize noise and avoid overload distortion.

36) 48V
Every microphone input is equipped with phantom power controlled by the 48V phantom power button. The 48V button will illuminate when phantom power is activated.

⚠ ***Please do not supply phantom power to any device which does not need phantom power, otherwise the device may be damaged.***

37) Digital
This button engages and disengages the channel from receiving it's corresponding digital input signal from the optional recording module.

38) Select
Press this button to enable DSP processing and output assignments to any or all of Main output, AUX sends 1-8 and Subgroups 1-4.

39) Solo
This assigns the channel to the solo bus.

40) Mute
Pressing this button mutes the channel to SUB 1-4 and Mains. It will only mute the AUX outputs assigned from that channel if the AUX send is in the post-fader mode. It will illuminate red when the channel is muted.

41) Sig/Clip LED
The Sig/Clip LED will be green when the level is above -30 dB and red when the level is greater than +15 dB.

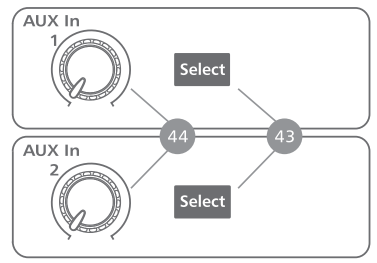
42) Channel Fader
Each channel features a 100 mm long-throw fader for accurate level adjustment.

AUX Inputs

The MP24M has two stereo AUX inputs that can be used for any line level signal returning to the console.

43) Select
Press the **SELECT** button to add EQ and assign its output to any or all of the Main output, AUX Sends 1-8 and Subgroup outputs 1-4.

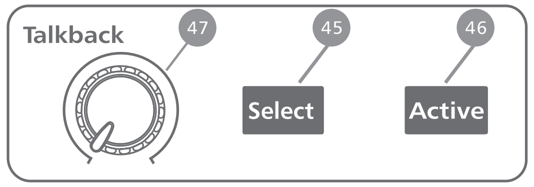
44) Level
This knob controls the overall volume of each of the stereo AUX In returns.



2 Controls

Talkback System

The MP24M features a Talkback microphone input with a 48V phantom power switch on the back panel.



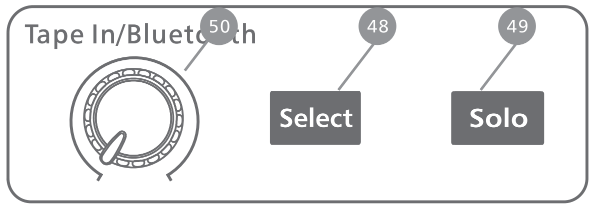
45) Select
Press this button to enable DSP processing and output assignments to any or all of Main output, AUX sends 1-8 and Subgroups 1-4.

46) Active
Press this button to turn the Talkback mic on and off. It will illuminate to indicate that the Talkback mic is active.

47) Level Control
This knob controls the overall volume of the Talkback mic.

Tape Input

The Tape Input can be used to connect a playback source without having to use main input channels on the MP24M.



48) Select
Pressing this button will enable you to add EQ and assign its output to any or all of the Main output, Subgroups 1-4 and AUX sends 1-8.

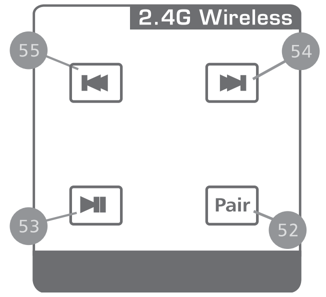
49) Solo
Pressing this button will assign the Tape In/ 2.4G Wireless to the Solo bus.

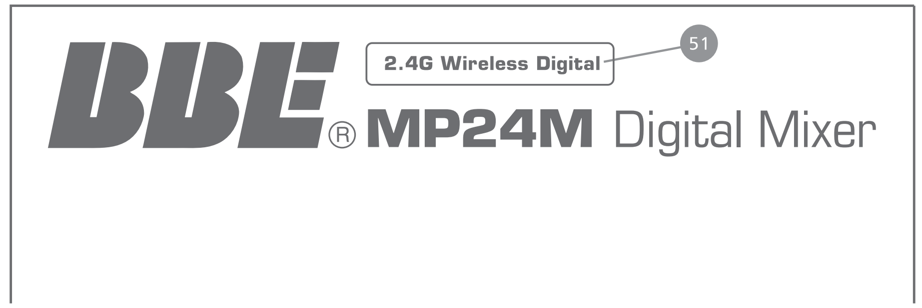
50) Level Control

This knob controls the level for both the Tape In/ 2.4G Wireless.

Controls 2

2.4G Wireless





The 2.4G Wireless input signal will be routed to the Tape In automatically once it is paired..

51) 2.4G Wireless LEDs
The two LEDs are used to display the different modes of the 2.4G Wireless function. The 2.4G Wireless will activate automatically in the standby mode after the MP24M is powered on:

- The right LED flashes twice every 2 seconds when not paired.
- The two LEDs flash quickly and alternately in the pairing mode.
- The right LED will stay lit after it has been paired with a device.

52) Pair
Press this button and hold for 2-3 seconds, it will change to pairing mode. In this mode, the two LEDs flash alternately and quickly, and you can use your mobile phone, tablet or PC 2.4G Wireless adapter to find the BT-2.1B device. Only if your device's 2.4G Wireless version is lower than 2.0, should you enter the password "0000", otherwise no password needed.

53) Play/Pause
Press the Play/Pause button to pause in play mode and play in pause mode.

54) Next
Press this button to go to the next selection.

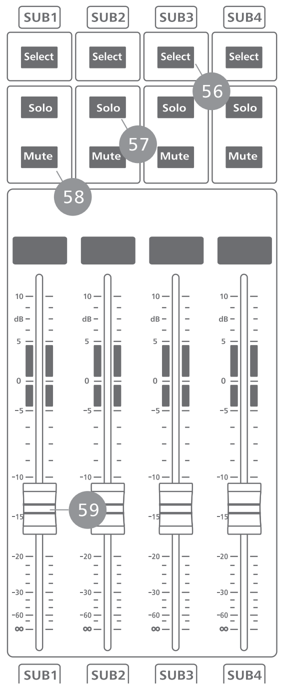
55) Back
Press this button to go to the previous selection.

2

Controls

Output Channels

There are four kinds output channels on the MP24M: Subgroup outputs 1-4; Main output; Monitor output and AUX sends.



Subgroup Outputs

56) Select
Press the **SELECT** button to access DSP processing and assign the SUB output to Main output. It will illuminate when enabled.

57) Solo
Pressing this button will send its signal to the solo bus.

58) Mute
Pressing this button will mute its output. It will illuminate red when it is enabled.

59) Fader
The 100mm long fader is for accurate level adjustment.

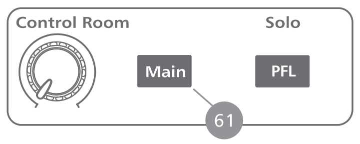
Main Output

60) Select Button
Press this button to enable Main output's Compressor, Gate, EQ, Polarity, Delay.

61) Main Solo
Pressing this will send the MAIN stereo output to the solo bus.

62) Mute
Pressing this button will mute its output.

63) Fader
The 100 mm long fader is for accurate level adjustment.



Controls

2

Monitor Output

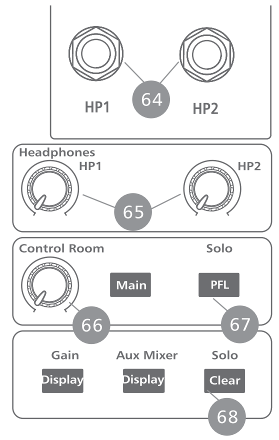
64) HP1 and HP2 Connectors
These are headphone ports for monitoring.

65) HP1 and HP2 Level
These knobs control the level of the two headphone outputs separately.

66) Control Room Level
This knob adjusts the level of the Control Room output.

67) PFL
Press the PFL button to change from After-Fader Listen (AFL), which is the default setting for the Solo bus, to Pre-Fader Listen. The Subgroups have no PFL soloing function. The AUX bus soloing is always PFL no matter the position of the PFL button.

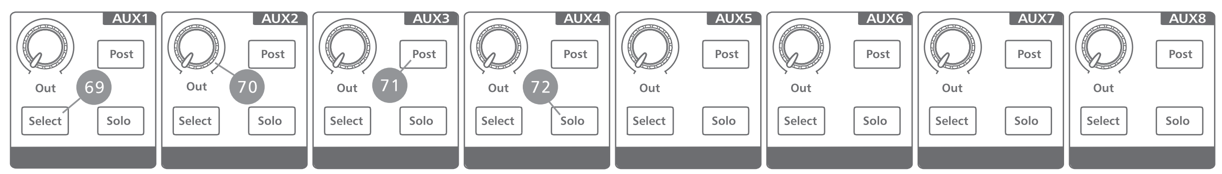
68) Solo Clear
Press this button to clear all console Solos currently engaged.



AUX and FX Send/Return

The BBE MP24M features eight AUX sends and two internal FX sends. The AUX buses are mono, however two AUX buses can be linked as a stereo bus; the FX sends cannot be linked.

AUX Sends



69) Select
Press this button to access all DSP functions.

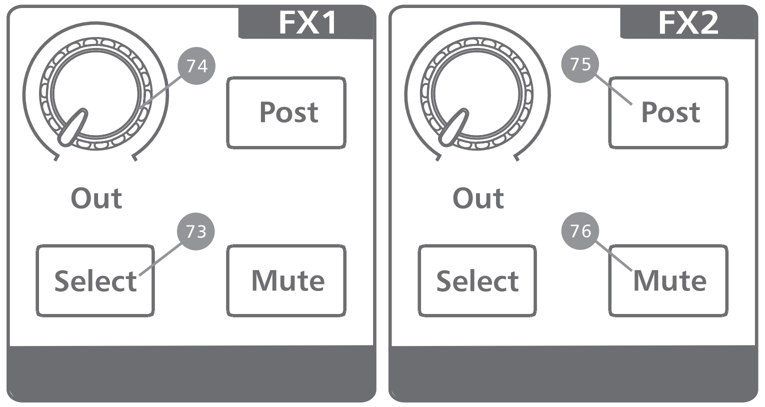
70) Output Level
This knob controls the overall output level of the AUX send.

71) Post-Fader
Press this button and that AUX send will derive its signals from all channels post-fader. If the button has not been pressed, by default, the AUX bus will derive its signals from all channels pre-fader and be unaffected by the fader position.

72) Solo
Pressing this button will send its output to the solo bus.

2 Controls

FX Send/Return



73) Select
Press this button to enable EQ and to assign it's stereo Return to any or all of the Main, Subgroups 1-4 or AUX sends 1-8 outputs.

74) Output Level
This knob controls the output level of the FX send.

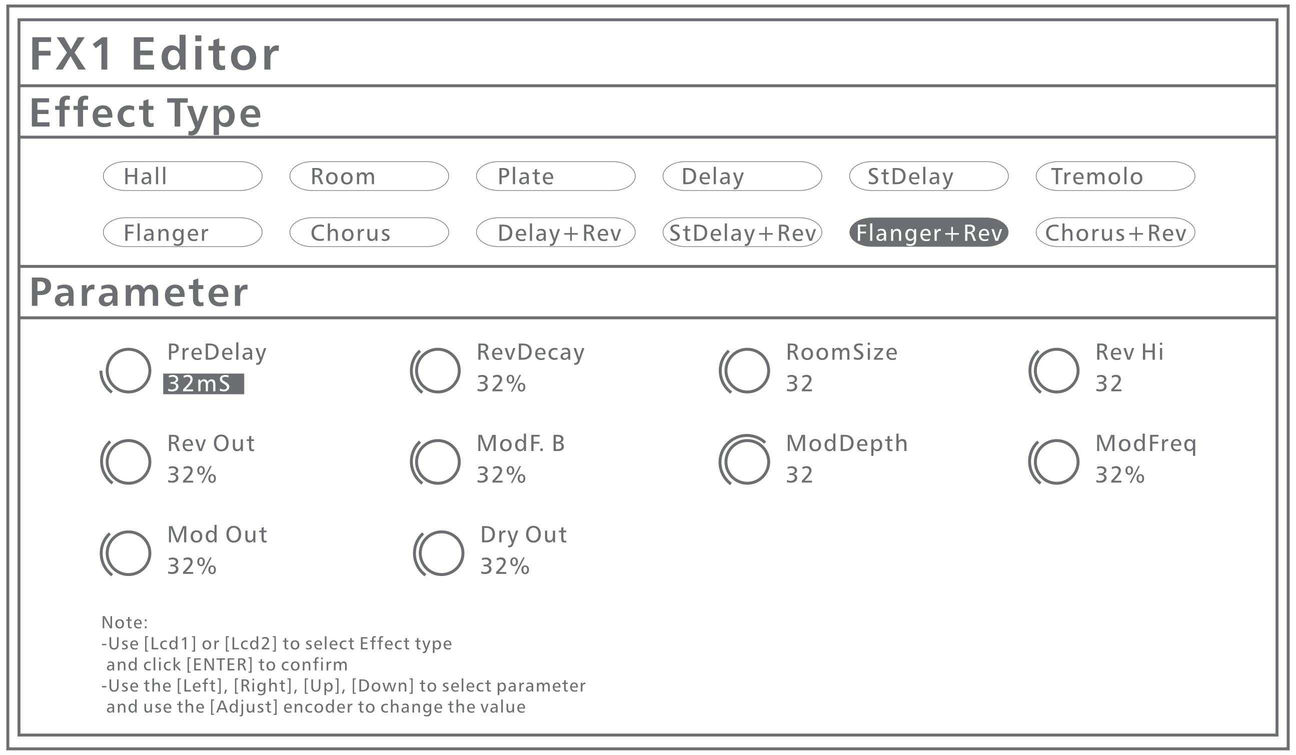
75) Post
Press this button and the FX send will derive its signals from all channels post-fader. If the button has not been pressed, by default, the FX Send will derive it's signal from all channels pre-fader and unaffected by the sending channel's fader position.

76) Mute
Press this button to mute the FX bus and all of its assigned outputs. It will illuminate red to show it is muted.

Controls 2

Digital Effects

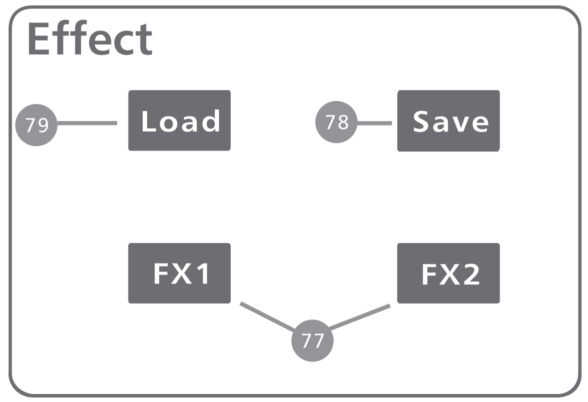
The MP24M features BBE's exclusive Digital FX Library.



77) FX 1 & 2
Press this button to show and edit the parameters of the BBE Digital FX. The modified effect can be saved as a user-defined preset for future use by simply pressing the Save button and following the instructions that are shown on the LCD display.

78) Save
Press this button to save the modified effect as a preset. Please note the instructions that are shown on the LCD display. Please read DSP Save in section *DSP Load, Save, Copy* on page 9 as reference.

79) Load
Press this button to Load a user-defined effect preset. The preset can be deleted by pressing the **FLAT EQ** button after it has been chosen. Please note the instructions that are shown on the LCD display. Please read DSP Load in *DSP Load, Save, Copy* section on page 9 as reference.

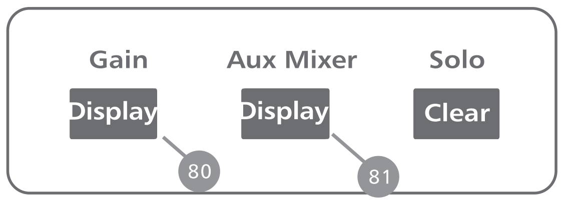


2

Controls

Preset	Description	Parameter
Hall	Simulates the reverb of a large room or hall	PRE Delay; Decay; Room Size; Hi Damp; EFX Out; Dry out
Plate	Simulates the reverb of a classic bright vocal plate	PRE Delay; Decay; Room Size; Hi Damp; EFX Out; Dry out
St delay	Stereo repeating, decaying echo effect	L Time; R Time; L Decay; R Decay; Hi Damp; EFX Out; Dry Out
Flanger	Comb-filter effect	Feed Back; Depth; Mod Freq; EFX Out; Dry Out
Delay+Rev	Repeat echoes combined with large room and hall reverb	PRE Delay; Rev Decay; Room Size; Rev Hi; Rev Out; Echo Time; Echo Hi; Echo F.B.; Echo out; Dry Out
Flanger+Rev	Comb-filter effects combined with large room and hall reverb	PRE Delay; Rev Decay; Room Size; Rev Hi; Rev Out; Mod F.B.; Mod Depth; Mod Freq; Mod Out; Dry Out
Room	Simulates the reverb of a small or medium sized room.	PRE Delay; Decay; Room Size; Hi Damp; EFX Out; Dry Out
Delay	Mono repeating, decaying echo effect	Time; Decay; Hi Damp; EFX Out; Dry Out
Tremolo	Amplitude modulation effect	Feed Back; Depth; Mod Freq; EFX Out; Dry Out
Chorus	Timbre and pitch modulation effect	Feed Back; Depth; Mod Freq; EFX Out; Dry Out
St Delay+Rev	Stereo repeat echoes combined with large room and hall reverb	PRE Delay; Rev Decay; Room Size; Rev Hi; Rev Out; L Time; R Time; L Decay; R Decay; Echo Hi; Echo Out; Dry Out
Chorus+Rev	Timbre and pitch modulation effects combined with large room and hall reverb	PRE Delay; Rev Decay; Room Size; Rev Hi; Rev Out; Mod F. B; Mod Depth; Mod Freq; Mod Out; Dry Out

Gain Display



80) Gain Display

In these screens there are three bars per channel that represent the following:

- The first bar represents the current fader position of that channel.

- The second bar represents the fader position as it was the last time the MP24M was powered down. If a user-defined Scene preset is loaded, the second bar will represent the position of the fader when the Scene was saved.

By matching the position of the first bar to the second bar (by moving the channel fader up or down), fader positions saved in the Scene can be totally recalled.

- The third bar represents the real-time post-fader output level of that channel.

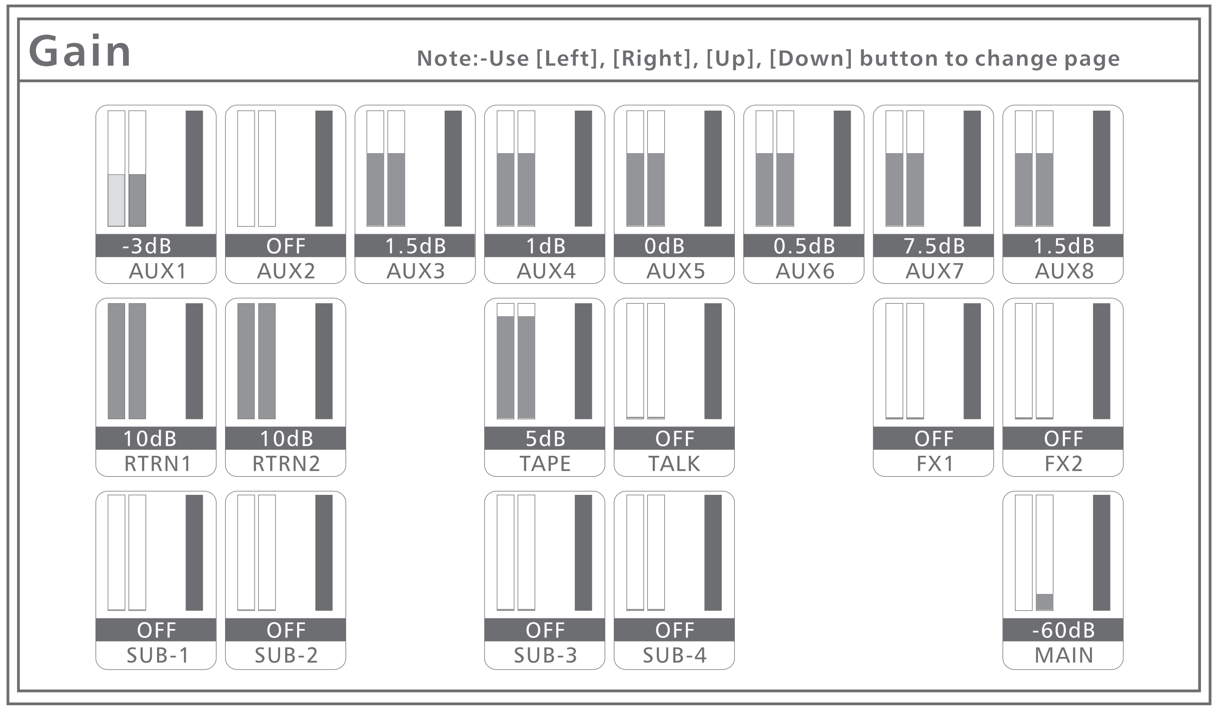
The arrow buttons will toggle between Gain Display and Fader Volume.

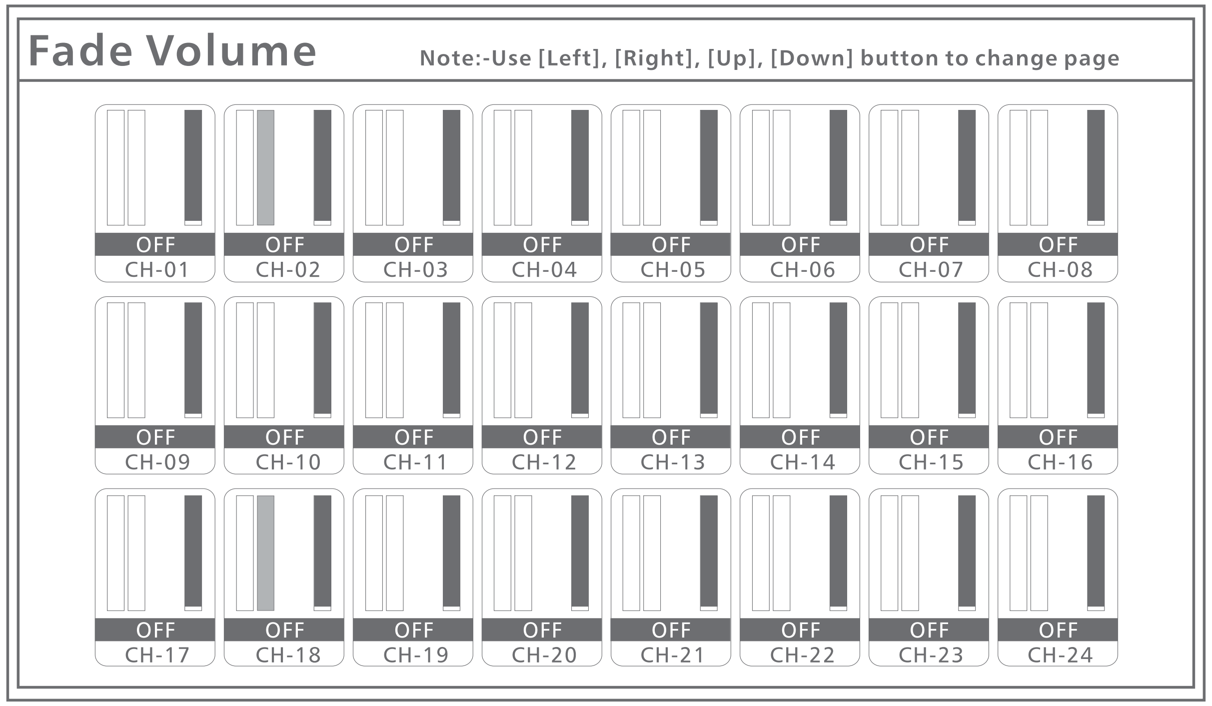
81) AUX Mixer Display

Press this button to display and adjust each channel’s level to each of the AUX and FX sends. The LCD1 and LCD2 buttons will scroll through each of the AUX and FX Send buses.

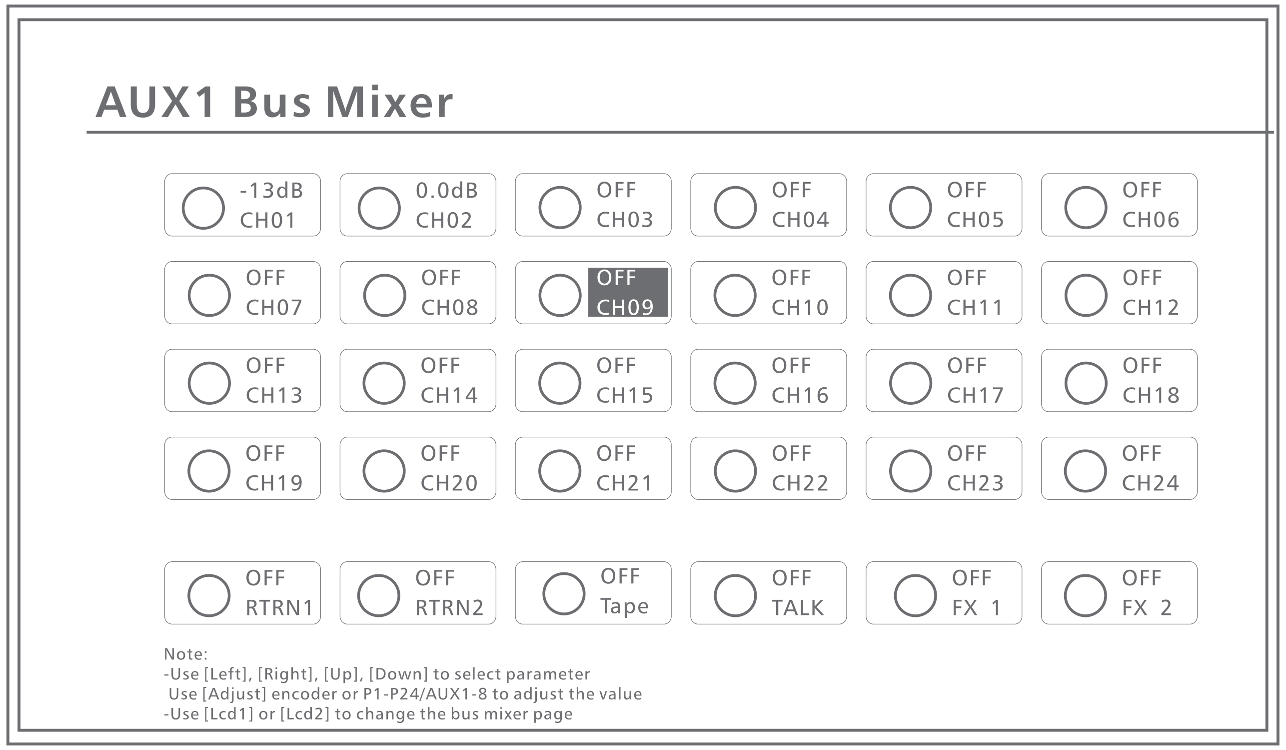
Controls

2

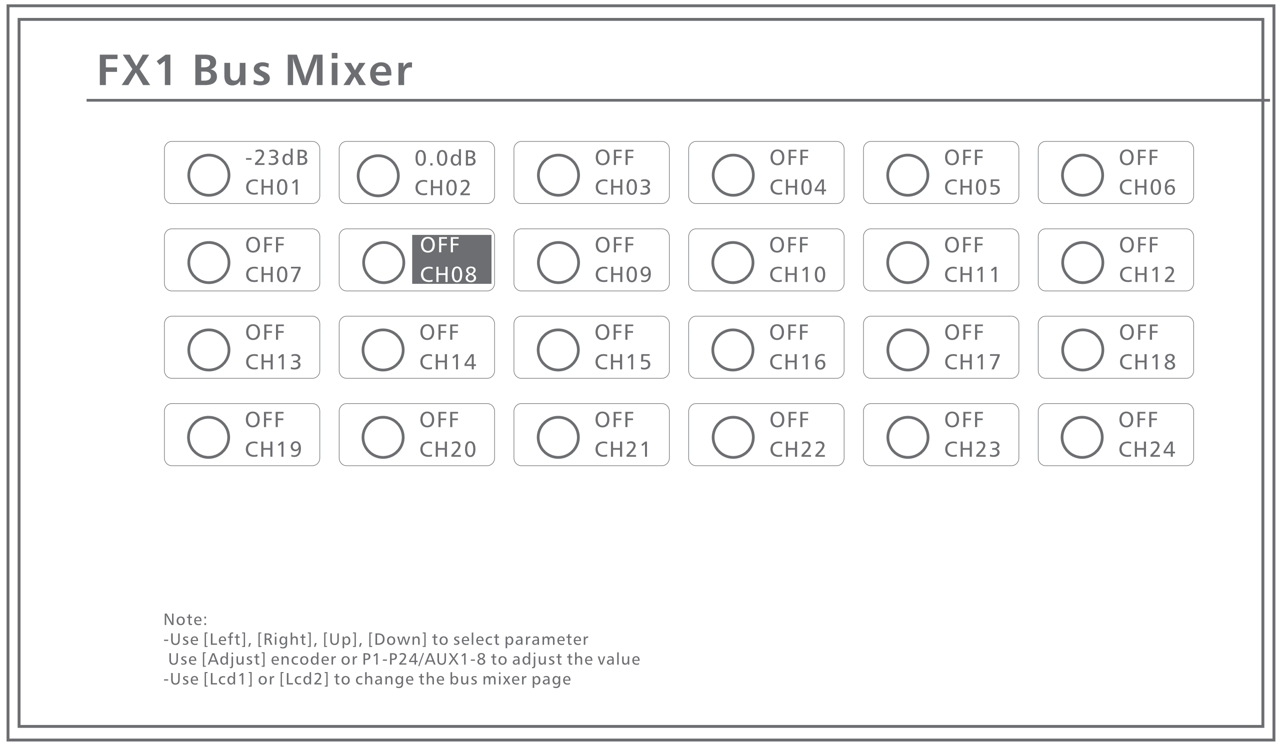




AUX Mixer Display

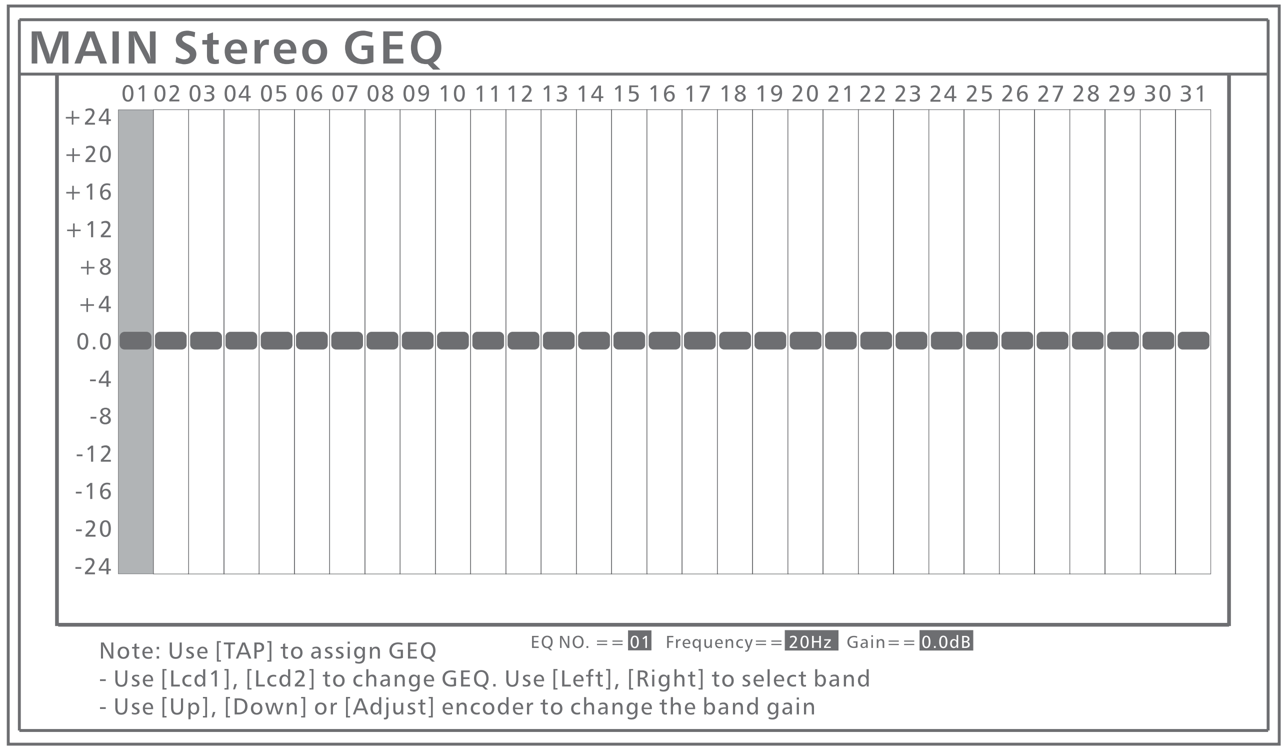


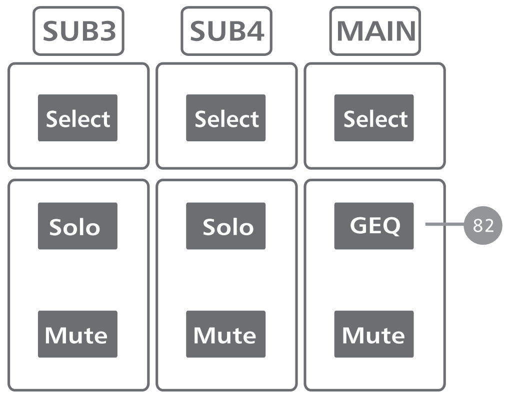
AUX Sends 1-8



Sends 1-2

GEQ





82) GEQ

Press this button to set the 31-band EQ. It will illuminate when engaged. The MP24M features Main, Subgroup and AUX, 31-band, 1/3 octave graphic EQs. The 31 bands range from 20Hz to 20 kHz. There is one MAIN Stereo GEQ, four Subgroup GEQs and eight AUX GEQs.

The EQ band number, Frequency and Gain value which you are adjusting will be shown on the LCD below the graphic curve. Please follow the instructions that are shown on the LCD display to adjust the value. The **FLAT EQ** can help you reset the whole 31 bands to flat.

The GEQ settings can be saved as a user-defined preset for future use by pressing the Save button and following the instructions that are shown on the LCD display. The preset can be recalled by pressing the **Load** button and deleted by pressing the **FLAT EQ** button after it has been chosen.

Please read section *DSP Load, Save, Copy* on page 9 as reference.

2

Controls

Scene

System, Load & Save

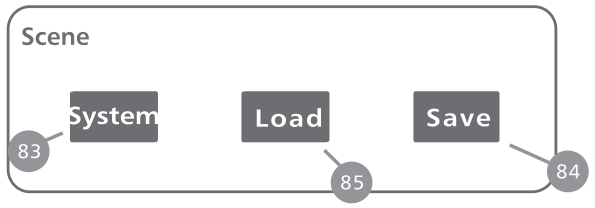
Every control, parameter and fader position, everything on the console, can be saved as a Scene. Saving Scenes allows the user to quickly recall set-ups days, weeks, or even months, later.

83) System

Press this button to show and edit system wide parameters. Use the **UP** & **LEFT** & **DOWN** & **RIGHT** key to choose the parameter that you want to modify. The LCD Backlight and AES Output Source can be adjusted using the Adjust Encoder. Reset needs to have the Enter button pressed first, then follow the instruction that is shown on the LCD display.

84) Save

Press this button to save the current Scene as a user-defined preset for future use. Please do as the instructions specify that are shown on the LCD display.



85) Load

Press this button to Load a user-defined Scene preset. Scene presets can be selected from the preset library and loaded by pressing Enter. The chosen preset can be deleted by pressing the **FLAT EQ.**

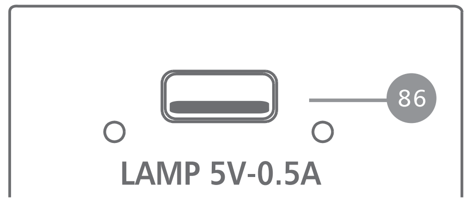
Controls

2

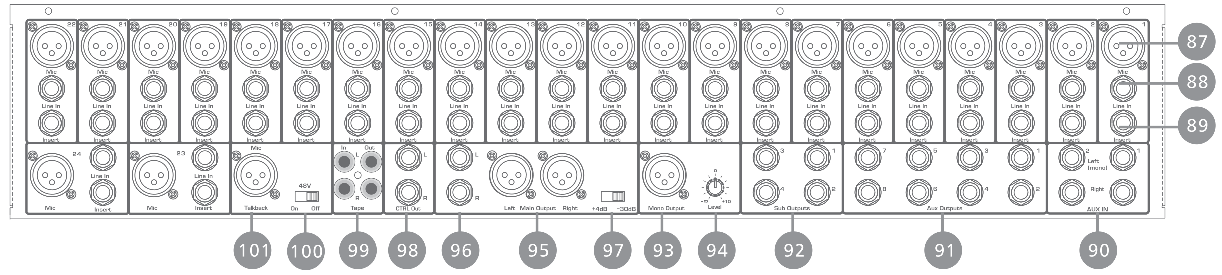
Lamp

86) LAMP 5V-0.5A

This can connect with a 5V-0.5A lamp which can help you use the BBE MP24M in low-light situations. It can also be used as a USB charge port.



Rear Panel



87) Microphone Inputs

The MP24M features twenty-four of BBE's proprietary MASS® microphone pre-amplifiers. The MASS® mic preamp topology features a Class A input buffer followed by a dual-servo gain stage. This arrangement results in ultra-low noise and wide gain control making the MASS® preamp one of the most accurate mic preamps currently available.

88) Line-level Input

The line-level input is a 1/4", unbalanced connector. Each channel of the MP24M has a line-level input. The microphone-preamp circuit will be bypassed if the Line-level Input has been engaged.

⚠ ***Please notice that there will be a momentary spike in the output when plugging in a microphone or a line-level input device, or turning phantom power on or off. Mute or turn down the channel fader before changing connections or turning phantom power on or off.***

89) Insert

The insert point utilizes a 1/4" TRS connector which can be used to connect external processors. The insert's send is after the channel's gain control and before the digital bus. These are configured tip=send, ring=return

90) AUX Inputs

The two stereo AUX inputs are normally used as effects returns but can be used for any line level signal returning to the mixer. The input is 1/4" TRS balanced. If a mono signal has to be returned to the mix, connect it to the left input, then the right as well as the left side will get the signal.

2

Controls

91) AUX Outputs
The output from AUXs 1-8 are routed to these eight outputs. The AUX mixes can be created for monitoring and outboard effects processing. These are 1/4” TRS balanced connections.

92) SUB Outputs
These are 1/4” TRS balanced outputs for each Subgroups 1-4.

93) Mono Output
This XLR balanced output carries a mono, summed version of the stereo signal from the Main bus.

94) Mono Output Level
This knob controls the level of the Mono Output signal. The signal can be attenuated to -∞ and boosted up to + 10 dB.

95) Main Output XLR
These are XLR balanced outputs.

96) Main Output Phone Jack
These are 1/4” unbalanced outputs.

97) Main Output Level
This switch controls the output level of the XLR and 1/4” balanced Main outputs. It can be switched between -30 dB or +4 dB.

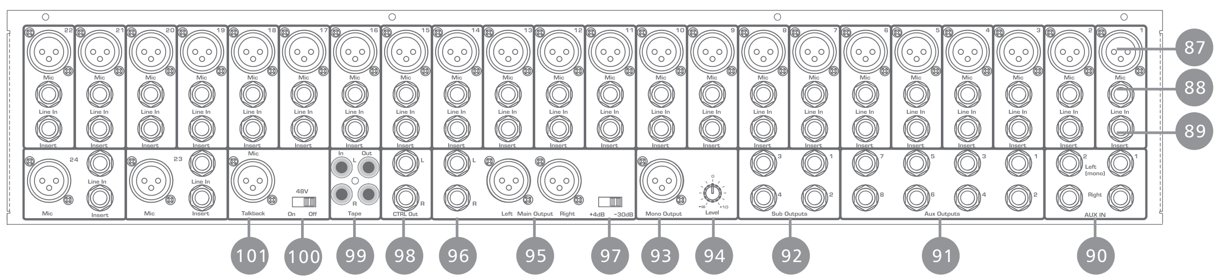
98) CTRL Output
These are 1/4” TRS balanced outputs. The output volume is controlled by the knob marked Control Room on the top panel.

99) Tape In/Out
These RCA inputs and outputs can be used to connect a CD player, MP3 player or other device. The Tape In level is controlled by the Tape In knob on the top panel. The Main output bus is routed post-fader to the tape output. The Tape In is superseded by a paired 2.4G Wireles device. The 2.4G Wireless device will take precedence.

100) 48V Phantom Power
This will supply 48V Phantom Power to the Talkback mic if required.

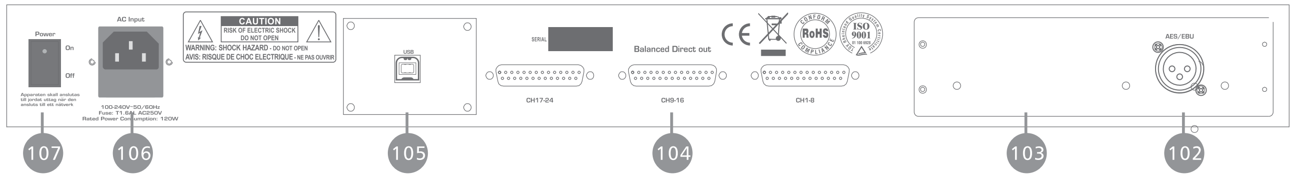
⚠ ***Please do not supply phantom power to any device which do not need phantom power otherwise the device may be damaged.***

101) Talkback Mic Input
Mics connected to the Talkback Mic input can be used for communicating with the stage. It is controlled by the Talkback section on the front of the top panel.



Controls

2



102) AES/EBU Digital Output Port
One Stereo Output pair can be assigned to the AES-EBU digital output. You can choose from the System menu the following output options:

MAIN

SUB 1/2

SUB 3/4

AUX 1/2

AUX 3/4

AUX 5/6

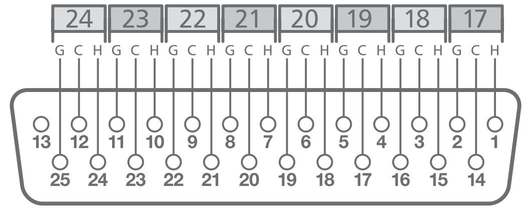
AUX 7/8

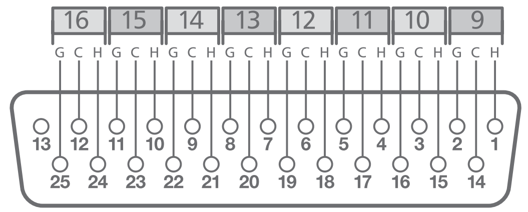
103) Option Module
This port will accommodate future expansion modules.

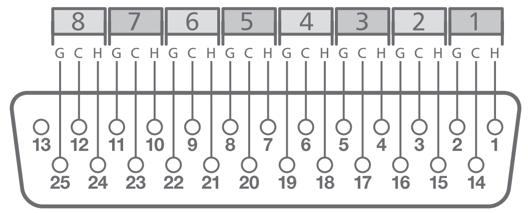
104) Balanced Direct Output
Channels 1-24 balanced, direct analog outputs. The DB25 connectors divide the channels into three groups of eight. These outputs are post-gain, pre-insert, and pre-A/D converter. Only the microphone inputs and line-level inputs can be sent through the direct outputs.

The MP24M DB25 connectors utilize the Tascam DA-88 pinout standard.

H = Hot C = Cold G = Ground







105) USB Connect Port
This port is for USB connection to a computer.

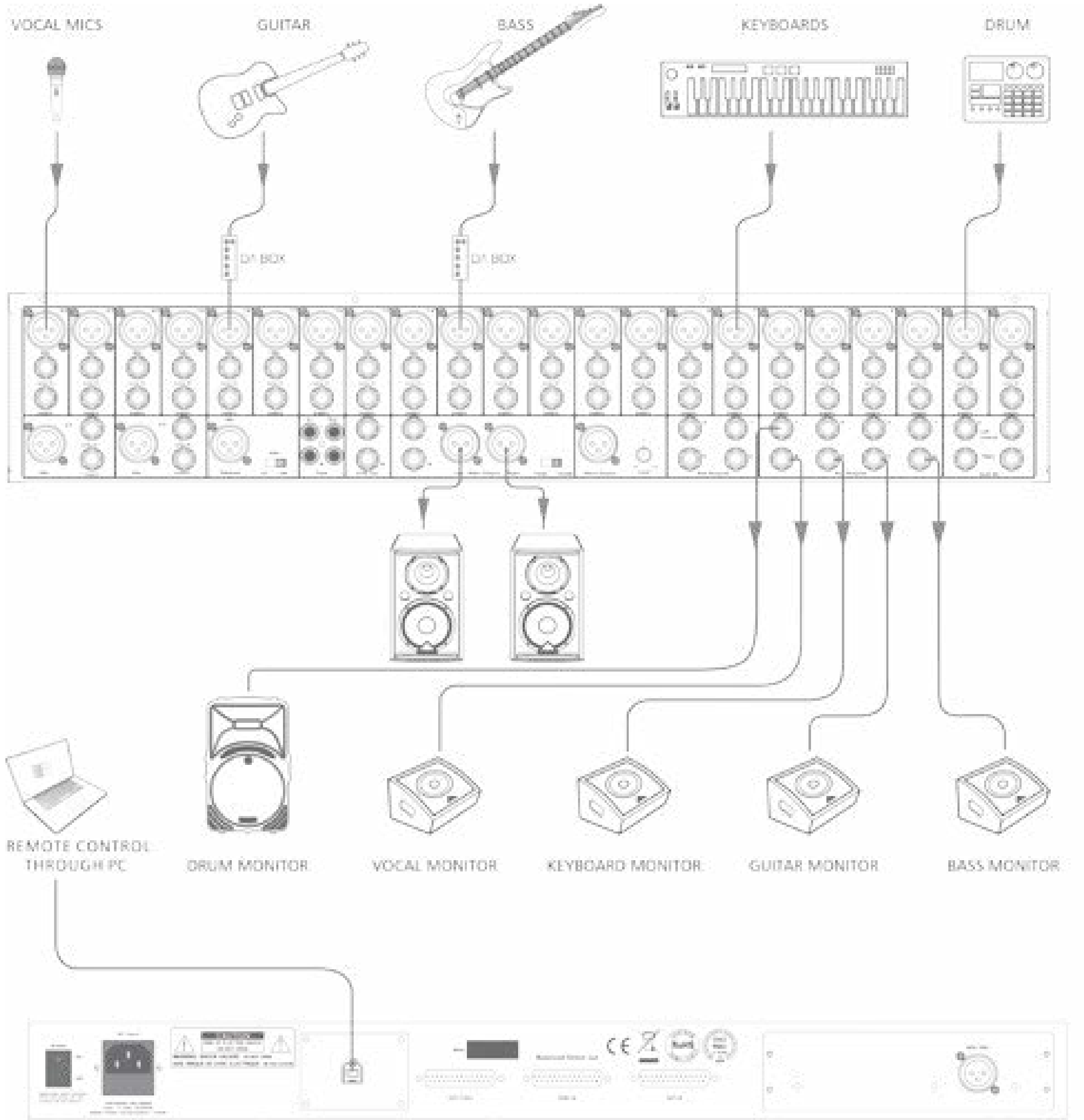
106) A/C Power Connection
The provided power cable can be plugged in here.

107) Power Switch
Push the top part of the switch to turn on and the bottom part to turn off.

2

Controls

Hookup



3

Connecting to a Computer

The MP24M is not only a mixer but also a very powerful computer interface which allows it to be controlled by a computer.

System Requirements

The minimum computer system requirements for your MP24M are:

Operating Systems:

- Windows XP (SP2 or greater) 32-bit and 64-bit
- Windows Vista 32-bit and 64-bit (not recommended)
- Windows 7 32-bit and 64-bit
- Windows 8 64-bit

Minimum Hardware Requirements:

- Intel Celeron (R) CPU 2.20G Hz
- 1 GB RAM
- Graphics card video memory 764MB

Installation the Driver

Please download the PL2303_Prolific_DriverInstaller driver and Digital Mixer software to your computer from the BBE Sound website: www.bbesound.com.

Then open the PL2303_Prolific_DriverInstaller driver and complete the installation following the on screen instruction.

4

Tutorials

Microphone Types

The MP24M works with all types of microphones, including dynamic, ribbon, and condenser microphones.

Condenser Microphones

Condenser microphones generally capture sound with excellent fidelity and are one of the most popular microphone choices for studio recording and, increasingly, for live performance as well. Condenser microphones require a power source, which can be provided by a small battery, an external power supply, or phantom power, which is usually provided by a mixer, preamplifier, or direct (DI) box. Phantom power is sent over the mic cable that carries the audio signal; the term derives from the fact that there is no visible power cord, and the voltage is not perceptible in the audio path. The MP24M sends 48 VDC phantom power from the XLR inputs only.

Dynamic Microphones

Dynamic microphones are possibly the most widely used microphone type, especially in live sound applications. They are relatively inexpensive, resistant to physical damage, and typically handle high sound-pressure levels (SPL) very well. Unlike condenser microphones, dynamic microphones typically do not require a power source.

Dynamic microphones tend to generate low output voltages, so they typically need more preamp gain than condenser microphones.

Ribbon Microphones

Ribbon microphones are a special type of dynamic microphone and get their name from the thin metal ribbon used in their design. Ribbon microphones capture sound with very high fidelity. However, they are often very fragile (many newer models are less so) and typically cannot handle high sound-pressure levels.

Ribbon microphones do not require phantom power. In fact sending phantom power to a ribbon microphone can severely damage it usually beyond repair.

Microphone Placement

Grand Piano

Generally use condenser mics. Place one microphone above the high strings and one microphone above the low strings. Experiment with distance (the farther back the more room you will capture). This technique can be used for live and studio applications.

Electric Guitar

Place a dynamic microphone an inch or two away from the speaker of the guitar amplifier. Experiment with exact location. In an amp with multiple speakers, experiment with each one to see if one sounds better than the others.

Acoustic Guitar

Point a small-diaphragm condenser microphone at the 12th fret, approximately 8 inches away. Point a large-diaphragm condenser microphone at the bridge of the guitar, approximately 12 inches from the guitar. Experiment with distances and microphone placement.

Tutorials

4

Bass Guitar (Direct and Speaker)

Plug the electric bass guitar into a direct box. Connect the instrument output from the direct box to a bass amplifier. Place a dynamic microphone an inch or two away from the speaker and connect it to a MP24M microphone input. Connect the balanced output from the direct box to a microphone input on a different channel of the MP24M.

Drum Overheads (XY example)

Place two small-diaphragm condenser microphones on an XY stereo-microphone holder (bar). Position the microphones so that each one is at a 45-degree angle, pointed down at the drum kit, approximately 7 or 8 feet above the floor or drum riser. Experiment with height.

Snare Drum (top and bottom)

Point a dynamic microphone at the center of the snare, making sure it is placed so that the drummer will not hit it. Place a small-diaphragm condenser microphone under the drum, pointed at the snares since the bottom mic will capture the bottom head moving outward, it’s necessary to invert the (INV) on the bottom mic. Experiment with the placement of both microphones.

A Brief Tutorial on Dynamics Processing

Common Questions Regarding Dynamics Processing

What is dynamic range?

Dynamic range can be defined as the ratio between the loudest possible audio level and the lowest possible level.

For example, if a processor states that the maximum input level before Distortion is +24 dB, and the output noise floor is -92 dB, then the processor has a total dynamic range of 24 + 92 = 116 dB.

The average dynamic range of an orchestral performance can produce signal levels from -50 dB to + 10 dB, on average. This equates to a 60 dB dynamic range. Although 60 dB may not appear to be a large dynamic range, do the math, and you'll discover that +10 dB is 1,000 times louder than -50 dB!

Rock music, on the other hand, has a much smaller dynamic range: typically -10 dB to +10 dB, or 20 dB total dynamic range. This makes mixing the various signals of a rock performance together a much more tedious task.

Why do we need compression?

Consider the previous discussion: You are mixing a rock performance with an average dynamic range of 20 dB. You wish to add an uncompressed vocal to the mix. The average dynamic range of an uncompressed vocal is around 40 dB.

In other words, a vocal performance can go from -30 dB to +10 dB. The passages that are +10 dB and higher will be heard over the mix. However, the passages that are at -30 dB and below will never be heard over the roar of the rest of the mix. A compressor can be used in this situation to reduce (compress) the dynamic range of the vocal to around 10 dB. The vocal can now be placed at around +5 dB. At this level, the dynamic range of the vocal is from 0 dB to +10 dB. The lower level phrases will now be well above the lower level of the mix, and louder phrases will not overpower the mix.

The same points can be made about any instrument in the mix. Each instrument has its place, and a good compressor can assist the engineer in the overall blend.

Does every instrument need compression?

This question may lead many folks to say "absolutely not, over-compression is horrible." That statement can be qualified by defining over-compression. The term itself must have been derived from the fact that you can hear the compressor working. A well-designed and properly adjusted compressor should not be audible! Therefore, the over-compressed sound is likely to be an improper adjustment on a particular instrument – unless, of course, it is done intentionally for effect.

Why do we need noise gates?

Consider the compressed vocal example discussed earlier; you now have 10 dB of dynamic range for the vocal channel. Problems arise when noise or instruments in the background of the vocal mic become more audible after the lower end of the dynamic range is raised. You might attempt to mute the vocal between phrases in an attempt to remove the unwanted sounds; however, this would probably end disastrously. A better method is to use a noise gate. The noise-gate threshold could be set at the bottom of the dynamic range of the vocal, say -10 dBu, such that the gate would shut out the unwanted signals between the phrases.

If you have ever mixed live sound, you know the problems cymbals can create by bleeding through the tom mics. As soon as you add some highs to get some snap out of the tom, the cymbals come crashing through, placing the horn drivers into a small orbit. Gating those tom mics so that the cymbals no longer ring through them will give you an enormous boost in cleaning up the overall mix.

Dynamics processing is the process of altering the dynamic range of a signal, thereby enhancing the ability of a live sound system to handle the signal without distortion or noise and aiding in placing the signal in the overall mix.

Types of Dynamics Processing

Compression

"Punch", "Apparent Loudness", "Presence"; these are just three of the many terms used to describe the effects of compression.

Compression is a form of dynamic-range (gain) control. Audio signals have very wide peak-to-average signal-level ratios (sometimes referred to as dynamic range, which is the difference between the loudest level and the softest level). The peak signal can cause overload in the sound-reinforcement chain, resulting in signal distortion.

A compressor is a type of amplifier in which gain is dependent on the signal level passing through it. You can set the maximum level a compressor allows to pass through, thereby causing automatic gain reduction above some predetermined signal level, or threshold. Compression refers, basically, to the ability to reduce, by a fixed ratio, the amount by which a signal's output level can increase relative to the input level. It is useful for lowering the dynamic range of an instrument or vocal. It also assists in the mixing process by reducing the amount of level changes needed for a particular instrument.

For example, a vocalist who moves around in front of the microphone while performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this problem by reducing the louder passages enough to be compatible with the overall performance.

How severely the compressor reduces the signal is determined by the compression ratio and compression threshold. A ratio of 2:1 or less is considered mild compression, reducing the output by a factor of two for signals that exceed the compression threshold. Ratios of 10:1 or more are considered hard limiting.

As the compression threshold is lowered, more of the input signal is compressed (assuming a nominal input-signal level). Care must be taken to not over-compress a signal, as too much compression destroys the acoustic dynamic response of a performance. (That said, over compression is used by some engineers as an effect, with killer results!)

Compressors are commonly used in a variety of audio applications. For example:

A kick drum can get lost in a wall of electric guitars. No matter how much the level is increased, the kick drum stays lost in the "Mud." A touch of compression can tighten up that kick-drum sound, allowing it to punch through without having to crank the level up.

A vocal performance usually has a wide dynamic range. Transients (normally the loudest portions of the signal) can be far outside the average level of the vocal signal. Because the level can change continuously and dramatically, it is extremely difficult to ride the level with a console fader. A compressor automatically controls gain without altering the subtleties of the performance.

4 Tutorials

A solo guitar can seem to be masked by the rhythm guitars. Compression can make your lead soar above the track without shoving the fader through the roof.

Bass guitar can be difficult. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor give your bass the punch it needs to drive the bottom of the mix.

Compressors - Terminology

Threshold. The compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes eligible for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed (assuming you have a ratio setting greater than 1:1).

Ratio. The ratio is the relationship between the output level and the input level. In other words, the ratio sets the compression slope. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed such that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB. This produces a compression gain reduction of 0.5 dB. As you increase the ratio, the compressor gradually becomes a limiter.

Attack. Attack sets the speed at which the compressor acts on the input signal. A slow attack time allows the beginning envelope of a signal (commonly referred to as the initial transient) to pass through the compressor unprocessed, whereas a fast attack time immediately subjects the signal to the ratio and threshold setting of the compressor.

Release. Release sets the length of time the compressor takes to return to its normal gain once the signal level drops below the threshold. The value expressed as the duration required for the level to change by 6 dB. Very short release times can produce a very choppy or"jittery" sound, especially in low-frequency instruments such as bass guitar. Very long release times can result in an over-compressed sound

Makeup Gain. When compressing a signal, gain reduction usually results in an overall reduction of level. The gain control allows you to restore the loss in level due to compression (like readjusting the volume) whilst mainlining dynamic control.

Noise Gates

Threshold. The gate threshold sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set fully counterclockwise, the gate is turned off (always open), allowing all signals to pass through unaffected.

Gate Attack. The gate attack time sets how soon the gate opens when the signal exceeds the threshold level on the selected channel. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack. Too fast of an attack can, on these slow-rising signals, cause an artifact in the signal, which is heard as a click. All gates have the ability to click when opening but a properly set gate will never click.

Release. The gate-release time determines how fast the gate for the selected channel closes. The value is expressed as the duration required for the level to change by 6 dB. Release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause "chattering" in percussive instruments. Longer release times usually eliminate "chattering" and should be set by listening carefully for the most natural release of the signal.

Equalizers

The MP24M is equipped with a 4-band parametric equalizer on every input and output bus. Here's a brief explanation of how an EQ functions, as well as some charts to help you navigate the frequency ranges of various instruments so you can quickly choose the best EQ setting.

What is an EQ?

An equalizer is a filter that allows you to adjust the level of a frequency, or range of frequencies, of an audio signal. In its simplest form, an EQ will let you turn the treble and bass up or down, allowing you to adjust the coloration of, let's say, your car stereo or iPod. Good equalization is critical to a good mix.

When used correctly, an equalizer can provide the impression of nearness or distance, "fatten" or "thin" a sound, and help blend or provide separation between similar sounds in a mix allowing them to both shine through the mix.

Parametric EQ

The parametric EQ and semi-parametric EQ are the most common equalizers found in recording and live situations because they offer continuous control over all parameters. A parametric EQ offers continuous control over the audio signal's frequency content.

Q

Q is the ratio of center frequency to bandwidth, and if the center frequency is fixed, then bandwidth is inversely proportional to Q – meaning that as you raise the Q, you narrow the bandwidth. In fully parametric EQs, you have continuous bandwidth control and/or continuous Q control, which allows you to attenuate or boost a very narrow or wide range of frequencies.

A narrow bandwidth (higher Q) has obvious benefits for removing unpleasant tones. Let's say the snare drum in your mix has an annoying ring to it. With a very narrow bandwidth, you can isolate this one frequency (usually around 1 kHz) and remove, or reject it. This type of narrow band-reject filter is also known as a notch filter. By notching out the offending frequency, you can remove the problem without removing the instrument from the mix.

A narrow bandwidth is also useful in boosting pleasant tones of an instrument such as the attack. Take for instance, a kick drum. A kick drum resonates somewhere between 60 to 125 Hz, but the attack of the kick drum is much higher at 2 to 5 kHz. By setting a narrow bandwidth and boosting the attack a bit, you can achieve a punchier kick drum without overpowering the rest of the mix.

A broad bandwidth accentuates or attenuates a larger band of frequencies. The broad and narrow bandwidths (high and low Q) are usually used in conjunction with one another to achieve the desired effect.

Let's look at our kick drum again. We have a kick drum that has a great, big, low-end sound centered around 100 Hz and an attack hitting almost dead-on at 4 kHz. In this example, you would use a broad bandwidth in the low frequency band, centered at 100 Hz, and a narrow bandwidth boosted at 4 kHz. In

4 Tutorials

this way you are accentuating the best and downplaying everything else this particular kick drum has to offer.

Shelving EQ

A shelving EQ attenuates or boost frequencies above or below a specified cutoff point. Shelving equalizers come in two different varieties: high-pass and low-pass.

Low-pass shelving filters pass all frequencies below the specified cutoff frequency while attenuating all the frequencies above it. A high-pass filter does the opposite: passing all frequencies above the specified cut-off frequency while attenuating everything below.

Graphic EQ

The MP24M features four stereo, 31-band, graphic EQs that can be inserted on a variety of buses. A graphic EQ is a multiband equalizer that uses sliders to adjust the amplitude for each frequency band. It gets its name from the positions of the sliders, which graphically display the resulting frequency-response curve. The encoders in the EQ part of the MP24M are used to make amplitude adjustments, and the LCD display shows the value. The center frequency and bandwidth are fixed; the level (amplitude) for each band is the only adjustable parameter.

Graphic EQs are generally used to fine-tune the overall mix for a particular room. For instance, if you are mixing in a "dead" room, you may want to boost high frequencies and roll off some of the lows. If you are mixing in a "live" room, you might need to lower the high-midrange and highest frequencies. In general, you should not make drastic amplitude adjustments to any particular frequency bands. Instead, make smaller, incremental adjustments over a wider spectrum to round out your final mix. To assist you with these adjustments, here is an overview of which frequencies affect different sound characteristics:

Sub-Bass (16 Hz to 60 Hz). The lowest of these bass frequencies are felt, rather than heard, as with freeway rumbling or an earthquake. These frequencies give your mix a sense of power even when they only occur occasionally. However, overemphasizing frequencies in this range will result in a muddy mix.

Bass (60 Hz to 250 Hz). Because this range contains the fundamental notes of the rhythm section, any EQ changes will affect the balance of your mix, making it fat or thin. Too much emphasis will make for a boomy mix.

Low Mids (250 Hz to 2 kHz). In general, you will want to emphasize the lower portion of this range and deemphasize the upper portion. Cutting the range between 250 Hz and 500 Hz will help improve clarity in almost any live venue. The range between 500 Hz and 2 kHz can make midrange instruments (guitar, snare, saxophone, etc.) "honky" and too much boost between 1 kHz and 2 kHz can make your mix sound thin or "tinny".

High Mids (2 kHz to 4 kHz). The attack portion of percussion and rhythm instruments occurs in this range. High mids are also responsible for the projection of midrange instruments.

Presence (4 kHz to 6 kHz). This frequency range is partly responsible for the clarity of a mix and provides a measure of control over the perception of distance. If you boost this frequency range, the mix will be perceived as closer to the listener. Attenuating around 5 kHz will make the mix sound further away but also more transparent.

Brilliance (6kHz to 16kHz). While this range controls the brilliance and clarity of your mix, boosting it too much can cause some clipping so keep an eye on your main meter.

Tutorials 4

Equalization Setting: How to Find the Best & Leave the Rest

How do you find the best and worst each instrument has to offer and adjust their frequency content accordingly? Here's a quick guide:

First, solo just the instrument with which you are working. Most engineers start building their mix with the drums and work from the bottom up (kick, snare, toms, hi-hat. overheads). Each instrument resonates primarily in a specific frequency band, so if you are working on your kick-drum mic, start with the lowest band of the EQ. Tune in the best-sounding low end and move on to the attack. It is not uncommon to hear an annoying ringing or a "twang" mixed in with your amazing-sounding low end and perfect attack, so your next task will be to find that offending frequency and notch it out. Once you are satisfied with your kick drum, mute it, and move on to the next instrument.

Taking your time with equalization is well worth the effort. Your mix will have better separation and more clarity.

Additional advice:

- You can only do so much. Not every instrument can or should have a full, rich low end and a sharp attack.

If every instrument is EQ'd to have the same effect, it will lose its identity in the mix. Your goal is not individual perfection, it is perfection in unity.

- Step away from the mix. Your ears get fatigued, just like the rest of you. If you are working particularly hard on one instrument, your ears will be quite literally numbed to that frequency range.

- Your memory is not what you think it is. Comparing a flat EQ and the curve that you've created allows you to see and hear exactly what you've done. So be honest with yourself. Sometimes that EQ setting you've been working on for 15 minutes is not the right choice, so move on.

- Never be afraid of taking a risk. The best EQ tricks were found by mad scientists of sound. With every instrument, there are frequencies that can be attenuated or boosted to add clarity or fullness. Altering the wrong frequencies can make an instrument shrill, muddy, or just downright annoying. The following two charts suggest frequency ranges that should be accentuated or downplayed for the most common instruments. These are just suggestions; the frequencies may need to be adjusted up or down depending on the instrument, room, and microphone.

4 Tutorials

Table 1

Instrument	What to Cut	Why to Cut	What to Boost	Why to Boost
Human Voice	7 kHz	Sibilance	8k Hz	Big sound
	2 kHz	Shrill	3 kHz and above	Clarify
	1 kHz	Nasal	200-400 Hz	Bottom end
	80 Hz and below	Poppy		
Piano	1-2 kHz	Tinny	5 kHz	More essence
	300 Hz	Boomy	100 Hz	Bottom end
Electric Guitar	1-2 kHz	Shrill	3 kHz	Clarify
	80 Hz and below	Muddy	125 Hz	Bottom end
Acoustic Guitar	2-3 kHz	Tinny	5 kHz and above	Sparkle
	200 Hz	Boomy	125 Hz	Full
Electric Bass	1 kHz	Thin	600 Hz	Growl
	125 Hz	Boomy	80 Hz and below	Bottom end
String Bass	600 Hz	Hollow	2-S kHz	Sharp attack
	200 Hz	Boomy	125 Hz and below	Bottom end
Snare Drum	1 kHz	Annoying	2 kHz	Crisp
			150-200 Hz	Full
			80 Hz	Deep
Kick Drum	400 Hz	Muddy	2-5 kHz	Sharp attack
	80 Hz and below	Boomy	60 -125 Hz	Bottom end
Toms	300 Hz	Boomy	2-S kHz	Sharp attack
			80-200 Hz	Bottom end
Cymbals	1kHz	Annoying	15 kHz	Air
			8-12 kHz	Brilliance
			7-8 kHz	Sizzle
Horns	1 kHz	Honky	8-12 kHz	Big sound
	120 Hz and below	Muddy	2 kHz	Clarify
String section	3 kHz	Shrill	2 kHz	Clarify
	120 Hz and below	Muddy	400-600 Hz	Lush and full

Tutorials 4

Table 2

50Hz	*add fullness to low frequency instruments (kick, tom, bass} *decrease bass “boom” to increase overtones and clarity in the mix
100Hz	*add harder bass to low frequency instruments *add fullness to guitar & snare *add warmth to piano & horns *decrease “boom” in guitars to increase clarity
200Hz	*add fullness to vocals *harder hitting snare & guitar *decrease muddiness in vocals & mid-range instruments *decrease long bass overtones in cymbals
400Hz	*add clarity to bass lines (especially at softer playback volumes) *decrease dullness in kick and tom *decrease ambience in cymbals
800Hz	*add clarity & punch to bass lines *decrease thinness or overly bright overtones in guitars
1.5KHz	*add clarity & punch to bass lines *decrease dullness in guitars
3KHz	*more “pluck” to bass *more attack to acoustic & electric guitars *more attack on lower piano register *more clarity to vocals *increase breathiness in background vocals *disguise slightly out of tune vocals and guitars
5KHz	*more vocal presence *more attack on kick & tom *more finger sound on bass *more attack on piano and acoustic guitar *brighter rock guitar *make background instruments sound more distant *soften a “thin” guitar
7KHz	*more attack on kick & tom *more attack to percussion *reduce dullness on vocals *more finger sound on bass *add sharpness to synths, rock guitars, acoustic guitar & piano *reduce “S” sound, known as sibilance, on vocals
10KHz	*brighten vocals, acoustic guitar & piano *more hardness on cymbals *reduce sibilance sound on vocals
15KHz	*breathier vocals *brighten cymbals, string instruments & flutes *make samples & synths sound more real

4 Tutorials

General EQ Suggestions

The right EQ setting for any given instrument will depend upon the room and the tonality of the instrument.

Vocals

Pop Female Vocals

Low	Low	Low	Low	Low
On/off	Shelf	Freq (Hz)	Q	Gain
On	Off	130	0.6	-2

Low mid	Low mid	Low mid	Low mid
On/off	Freq (Hz)	Q	Gain
On	465	0.6	-2

High mid	High mid	High mid	High mid
On/off	Freq (KHz)	Q	Gain
On	2.4	0.4	+2

High	High	High	High	High
On/off	Shelf	Freq (KHz)	Q	Gain
On	Off	6.0	03	+B

Rock Female Vocals

Low	Low	Low	Low	Low
On/of	Shelf	Freq (Hz)	Q	Gain
On	On	155	NA	+4

Low mid	Low mid	Low mid	Low mid
On/off	Freq (Hz)	Q	Gain
On	465	0.4	+6

High mid	High mid	High mid	High mid
On/off	Freq (KHz)	Q	Gain
On	1.4	0.6	+6

High	High	High	High	High
On/off	Shelf	Freq 0<Hz	Q	Gain
On	Off	4.2	0.5	+2

Pop Male Vocals

Low	Low	Low	Low	Low
On/off	Shelf	Freq (Hz)	Q	Gain
On	Off	22.5	0.3	-2

Low mid	Low mid	Low mid	Low mid
On/off	Freq (Hz)	Q	Gain
On	960	0.3	0

High mid	High mid	High mid	High mid
On/off	Freq (KHz)	Q	Gain
On	2.0	0.6	+2

High	High	High	High	High
On/off	Shelf	Freq (KHz)	Q	Gain
On	Off	7.2	0.5	+4

Tutorials 4

Rock Male Vocals

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	155	0.5	+2

Low mid	Low mid	Low mid	Low mid
On/off	Freq (Hz)	Q	Gain
On	265	0.3	-6

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	2.4	0.6	-2

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	On	7.2	0.6	+4

Percussion Snare

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	130	0.6	-4

Low mid	Low mid	Low mid	Low mid
On/off	Freq (Hz)	Q	Gain
On	665	0.5	+4

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	1.6	0.3	+4

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	On	4.2	f\VA	+4

Left/Right (Stereo) Overheads

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	108	0.6	-2

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	385	0.6	-2

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	2.9	0.3	0

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	On	8.0	WA	+4

Kick Drum

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	108	0.4	+4

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	265	2.0	-4

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	1.6	0.6	0

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	Off	6.0	2.0	+4

4 Tutorials

Fretted Instruments

Electric Bass

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	On	36	NIA	-8

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	130	0.4	+4

High mid	High mid	High mid	High mid
On/off(KHz)	Q	Gain	On/off
On	2.0	0.6	+4

High	High	High	High	High
Shelf	(KHz)	Q	Gain	
On	On	4.2	NIA	+1

Acoustic Guitar

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	155	0.4	+4

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	665	2.0	+2

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	2.0	0.3	0

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	On	6.0	NIA	+4

Distorted Electric Guitar

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	Off	320	0.5	+6

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	960	0.4	0

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	3.5	1.0	+4

High	High	High	High	High
On/off	High Shelf	(KHz)	Q	Gain
On	On	12	NAI	0

Keyboards

Piano

Low	Low	Low	Low	Low
On/off	Shelf	(Hz)	Q	Gain
On	On	108	NIA	-2

Low mid	Low mid	Low mid	Low mid
On/off	(Hz)	Q	Gain
On	665	0.2	+2

High mid	High mid	High mid	High mid
On/off	(KHz)	Q	Gain
On	2.9	0.4	+2

High	High	High	High	High
On/off	Shelf	(KHz)	Q	Gain
On	Off	7.2	0.6	+4

Tutorials 4

Subgroup Mixing

A Subgroup allows you to combine multiple channels to a single bus so that the overall level for the entire group is controlled by a single fader. In addition to level control, the MP24M allows you to apply the noise gate, delay, compression, and EQ to the group as a whole, in addition to the processing available for each channel. Subgroups can also be solo'd and muted.

You will find many uses for subgroups that will make mixing more convenient and will provide better control of your mix. In this section, we explore two different ways in which subgroups can help you to create a more efficient mixing environment and a more successful live mix.

Instrument Groups

Grouping individual instruments that create a section in your mix has obvious advantages: The entire group can be muted or solo'd, brought up or down in a mix, and faded in or out for a more polished intro or outro. Some of the most common submix groups are drums, backing vocals, horn sections, and string sections. Drums are a classic application for subgroup mixing. We will be using a drum group in this particular example but these principles can be applied when grouping any type of instrument section in a live mix.

A drum group is especially useful when every piece in the drum kit has a microphone on it. In this example, our drums will be connected to the MP24M as follows:

Channel 1: Kick
Channel 2: Snare Top
Channel 3: Share Bottom
Channel 4: Floor Tom
Channel 5: Tom 1
Channel 6: Tom 2
Channel 7: Overhead Left
Channel 8: Overhead Right
Channel 9: Hi-Hat

We will create a stereo subgroup by linking Subgroups 1 and 2

1. The first step in creating a subgroup is to get a good mix of the instruments you are grouping, in this case, the drums. With the drummer's assistance, set the input trim, EQ, and dynamics for each drum separately. As you select and solo each channel, assign each channel to be routed to Subgroup 1.

2. After you have gone through the entire kit and are satisfied with each channel's level, EQ, and dynamics, unsolo your final channel. Have the drummer play the entire kit and set the relative volume and panning for each mic in the mix. Choose the Select button above Subgroup 1.

3. Enable Link and turn the Pan knob all the way clockwise to set the stereo pan to hard left and right. Now Subgroups 1 and 2 are linked, with Sub 1 panned hard left and Sub 2 panned hard right. The channel panning is preserved.

4. Now assign Subgroup 1 to the Main outputs; since Subgroup 2 is linked to Sub 1, it is automatically assigned to the Main outs as well. You can now add dynamics processing and EQ to the stereo drum group. Subgroup 1's fader now controls the level for the left side of your drum mix, and Subgroup 2's fader now controls the right side. The AUX bus enables you to create and send AUXiliary mixes that are separate from the main and subgroup mixes.

Effects Group

This is perhaps one of the most creative ways in which a subgroup can be used. By assigning an effects mix to a subgroup, the front-of-house engineer can become, in effect, a member of the band. This is especially useful when employing specialty or signature effects. For instance, a typical vocal mix for an electronic band dramatically trails off in a wash of reverb, whereas a reggae band usually has delay on vocals. The MP24M allows you to assign either or both of the onboard effects buses to a subgroup. So let's take the example of the reggae band.

1. In this example we will assign the delay on FX 1 (effects bus 1) to Subgroup 2. Press the FX1 button in the Effect section to access the Effects menu.

2. In the FX 1 parameters, use the LCD1/LCD2 button to find a suitable delay, and adjust its parameters following the instructions that are shown in the LCD display.

3. Next, decide which channels should be sent to the effects bus. In dub and reggae music, the vocals are most commonly sent to a delay, so let's send our two vocalists on channels 10 and 11 to that delay.

4. Press the AUX Mixer Display button and use the LCD 1/LCD 2 button to find the FX1 BUS Mixer, and follow the instructions that are shown in the LCD display to set the levels of channels 10 and 11 to be a little more than 50%.

5. Press the Select button for FX1 and assign this bus to Subgroup 2 and to the Main outputs. If you like you can also add some dynamics processing and EQ at this point.

6. Press the Select button for Subgroup 2 and assign the group to the Main output. (Because a delay can increase the signal's volume quite dramatically, you may want to experiment with the delay at its most intense setting, with FX1's output turned up, and use the compressor for Subgroup 2 to keep the level under control.)

The level of the vocal delay is now controlled by the Subgroup 2 fader, and you can use it to season the reggae band's performance. The Tap button allows you to go one step further and set the tempo of the delay to match the tempo of the song.

There are several advantages to assigning an effect like delay or reverb to a subgroup rather than simply leaving it on the effects AUX bus:

- You can quickly add or subtract the effect by grabbing a fader.
- The effect can be muted or solo'd.
- The performers on stage can have a different amount of the effect in their monitor mix than the audience hears in the main mix, enabling you to reduce the possibility of feedback while providing the performers with the tools they need for their best performance.

AUX Bus Mixing

The AUX bus provides outputs to create auxiliary mixes that are separate from the main and subgroup mixes. The MP24M is equipped with ten AUX buses: AUX 1 through 8, which have physical output jacks, and FX 1 and 2, which are the internal effects buses. AUX buses can be used for many applications, the two most common of which are creating monitor mixes and inserting external effects processors into the mix. As with the subgroup buses, the MP24M allows you to add global dynamics processing and EQ to these AUX buses, in addition to individual channel processing.

Monitor Mixing

Creating custom monitor mixes for your musicians is critical. If musicians can't hear themselves or other members, their performance will suffer. A monitor mix can be mono or stereo. Most often, an individual live monitor mix is mono and is sent to a floor-wedge, side-fill monitor, or in-ear monitor transmitter.

1. As an example, let's create a monitor mix on AUX 1. To begin, press the AUX Mixer Display button and use the LCD1/LCD2 button to select the AUX1 Bus Mixer. The LCD display will show the send-level of each channel sending to this AUX bus. Keep in mind that the AUX mix is completely independent of every other output (main bus, subgroups, direct out, etc.). Follow the instruction that is shown in the LCD display to adjust the channel level which sends to AUX1. Ask the musicians what they would like in their monitor mix and use their requests as a starting point.
2. By pressing the Select button for AUX 1, you can add dynamics processing and EQ to the overall monitor mix. These are especially useful for eliminating feedback in a monitor. Keep in mind that an equalizer can also be used to increase the presence of an instrument by boosting that particular frequency range without necessarily boosting the volume in the mix. This is great for getting the lead guitar to cut through in the guitarist's monitor mix and to provide that extra rumble in the bassist's mix. You can listen to the AUX mixes you are creating, using your headphones or your control-room monitor, by simply soloing the AUX send.

Effects Processing.

There are at least two advantages to using an AUX bus as an effects send rather than using a channel insert: several channels can be sent to a single processor, and you can vary the level sent from each channel to the processor, allowing you to create an effects mix.

The MP24M features two internal effects buses. These are used much in the same way the AUX buses are used to create monitor mixes, as described in the previous section. This section will detail how to use an external effects processor with your MP24M mixes.

1. In this example, we will use AUX 1 to feed an external effects processor. To begin, connect your external effects processor to your MP24M thusly:

   - Connect the balanced inputs of external effects processor to MP24M's AUX 1 output.

   - Connect the balanced outputs of external effects processor to MP24M's AUX Input 1.

2. Turn the AUX output level to 12 o'clock and press the Select buttons.
3. Press the AUX Mixer Display. The LCD display will show the output levels of each of the 24 channels. Use the LCD1/LCD2 buttons to scroll to the AUX1 Bus Mixer. You can use the P1-P24 encoders or other selecting buttons and the Adjust Encoder to control the output level of each channel into AUX 1.

Let's say that you are inserting an external reverb to liven up a relatively dead room. You might send a little bit of each input to the reverb, but you probably will not want much of the drums and bass to be processed, as too much reverb could reduce their impact and leave your mix without a sturdy foundation. So rather than turning the AUX-send level for the kick drum channel all the way up, turn it to the 7 or 8 o'clock position, so that only a small portion of the kick drum input will be affected by the reverb.

Once you have determined your effects mix, you can press the Select button for AUX 1 to add dynamics processing and EQ to the AUX mix before it is sent to the external effects processor. The effects processor's output is patched to AUX Input 1, now you can use the Select button for AUX Input 1 to add dynamics processing and EQ to the reverb-enhanced signal. The knob for AUX Input 1 controls the level of the AUX input relative to the level of your main mix.

Reverb

Reverberation - or reverb, as it is more commonly known, is perhaps the most widely used effect. Natural reverb is created by sound waves reflecting off of a surface or many surfaces. For example, when you walk across the wooden stage in a large hall, thousands of reflections are generated almost instantaneously as the sound waves bounce off the floor, walls, and ceilings. These are known as early reflections, and their pattern provides psycho-acoustic indications as to the nature of the space that you are in, even if you can't see it. As each reflection is then reflected off of more surfaces, the complexity of the sound increases, while the reverb slowly decays.

The reason for the widespread use of reverb is fairly self-evident: human beings don't live in a vacuum. Because our brains receive cues about the nature of the space around us based partially on audio reflections, a sense of space makes an audio recording sound more natural and, therefore, more pleasing.

The following parameters can usually be adjusted in a reverb effect:

- ***Decay.*** Decay is the time required for the reflections (reverberation) to die away. In most modern music production, reverb decay times between one and three seconds are prevalent. A reverb setting with strong early reflections and a quick decay are a great way to create a stereo effect from a mono source.
- ***Predelay***. Predelay is the time between the end of the initial sound and the moment when the first reflection becomes audible. Imagine you're back on that stage in a large music hall. This time you stand on the very edge of the stage and shout "Hello world!" toward the center of the hall. There will be a brief pause before you hear the first noticeable reflections of your voice, because the sound waves can travel much further before encountering a surface and bouncing back. (There are closer surfaces, of course notably the floor and the ceiling just in front of the stage but only a small part of the direct sound will go there, so those reflections will be much less noticeable.) Adjusting the predelay parameter on a reverb allows you to change the apparent size of the room without having to change the overall decay time. This will give your mix a little more transparency by leaving some space between the original sound and its reverb.
- ***HF and LF decay***. The types of surfaces in a space also affect the sound. Carpet and soft furnishings will absorb more high-frequency waves, thereby reducing the high-frequency decay time, while hard surfaces such as tile or stone reflect sound extremely well, resulting in a "brighter" ambience. Similarly, the high-frequency (HF) and low-frequency (LF) decay time allow you to adjust the "brightness" or "darkness" of the reverb, enabling you to better emulate these environmental factors.

Delay

A delay essentially creates an echo, although you can often use delays to create more complex time-based effects. The source signal is delayed so that it is heard later than it actually occurred.

Delay Time. Delay time is the time between the source signal and its echo. The simplest delay effect is a single repeat. A short delay between 30 and 100 ms can be used to create slap-back echo, while longer delay times produce a more distant echo. Delay times that are too short to hear as distinct echoes can be used to create thickening effects. Whether these echoes are timed with the tempo is a matter of stylistic choice.

Variable Feedback. Variable feedback, or regeneration, produces multiple decaying repeats. Increasing the feedback value increases the number of echoes as well as the resonance that is created as one echo disappears into another.

Note: Using the Tap button you can speed up or slow down these repeats or, more commonly, time the repeats to occur with the tempo of the music.

Level Setting Procedure in Detail

Setting the proper levels is an important part of getting the right sound. The following steps will assist you in quickly setting your levels.

1. Turn each of the 24 trims to around 11 o'clock.
2. Press the Select button
3. In the Solo bus section, select PFL
4. Press the Solo button and adjust the volume for your headphones or control-room monitors.
5. Solo your first channel and turn the trim to the desired level on the meter.

The Solo Bus

The Solo bus is extremely useful in setting levels for monitor mixes, dialing in dynamics processing on each channel, and fixing issues during a live show without interrupting the main mix.

The Solo bus has two different modes: AFL (the default setting) and, PFL.

AFL (After-Fade Listen). AFL sends the channel or subgroup signal to the Solo bus post-fader so that you can control the level of the solo'd signal with the fader. This is the MP24M's default setting.

PFL (Pre-Fade Listen). PFL sends the channel or subgroup signal to the Solo bus before it reaches the fader, so the fader does not affect the solo'd signal.

Using the Solo Bus for Monitoring

When mixing live it is often necessary to quickly listen to just one instrument or group. The Solo and Monitor buses can be used together for this purpose. It is important to note that if you wish to monitor with speakers, rather than headphones, it is necessary to connect the speakers to the Control Room outputs on the back of your MP24M rather than to one of the main output pairs.

First decide whether you want to listen to your solo'd channels before or after the fader setting. If you'd like to monitor before the fader level, press the PFL button in your Solo bus section. Next, press the Solo buttons on the channels and subgroups you want to monitor. Finally, dial in a comfortable listening volume for your headphones or monitors.

This feature can also be used to listen to a monitor mix that is being routed to an AUX send. Let's say your vocalist on stage is complaining that there is too much bass in his monitor, but you are confident that no bass is being sent to that particular AUX send.

You could be mistaken, but most likely an open microphone on stage is picking up the bass signal. To determine the cause, solo only the AUX send in question. You can now listen to exactly the same mix as your troubled vocalist and fix his monitor mix quickly.

This application is also useful in heading off a feedback problem.

5

Technical Information

Specifications

Microphone Inputs	
Frequency Response to Direct Output	20Hz~100KHz at 0dBu ± 1.5dB
Frequency Response to Main Output	20Hz~20KHz at 0dBu ± 1.5dB
Distortion (THD&N) to Direct Output	<0.005% at 0dBu 1KHz
Distortion (THD&N) to Main Output	<0.01% at 0dBu 1KHz
SNR (Signal to Noise Ratio)	107dB
Maximum Input Level	+21dBu
Phantom Power (+/-3V)	+48VDC
Line Inputs	
Frequency Response to Direct Output	20Hz~100KHz at 0dBu 1.5dB
Frequency Response to Main Output	20Hz~20KHz at 0dBu 1.5dB
Distortion (THD&N) to Direct Output	<0.005% at 0dBu 1KHz
Distortion (THD&N) to Main Output	<0.01% at 0dBu 1KHz
Gain	-15dBu~+35dBu
Maximum Input Level (Gainat0dBu)	+20dBu
AUX1~2 inputs	
Frequency Response to Main Output	20Hz~20KHz at +0dBu ± 1.5dB
Distortion (THD&N) to Main Output	<0.01% at 0dBu 1KHz
Gain	-∞ to+10dBu
Maximum Input Level	+22dBu
Tape L/R inputs	
Frequency Response to Main Output	20Hz~20KHz at +4dBu ± 1.5dB
Distortion (THD&N) to Main Output	<0.01% at 0dBu 1KHz
Gain	-∞ to+10dBu
Maximum Input Level	+21dBu
Talkback MIC	
Frequency Response to Main Output	20Hz~20KHzat+0dBu ± 1.5dB
Distortion (THD&N) to Main Output	<0.01% at 0dBu 1KHz
Gain	-∞ to+10dBu
Phantom Power (+/-3V)	+48VDC
Main Outputs	
Maximum Output Level	+20dBu

Technical Information

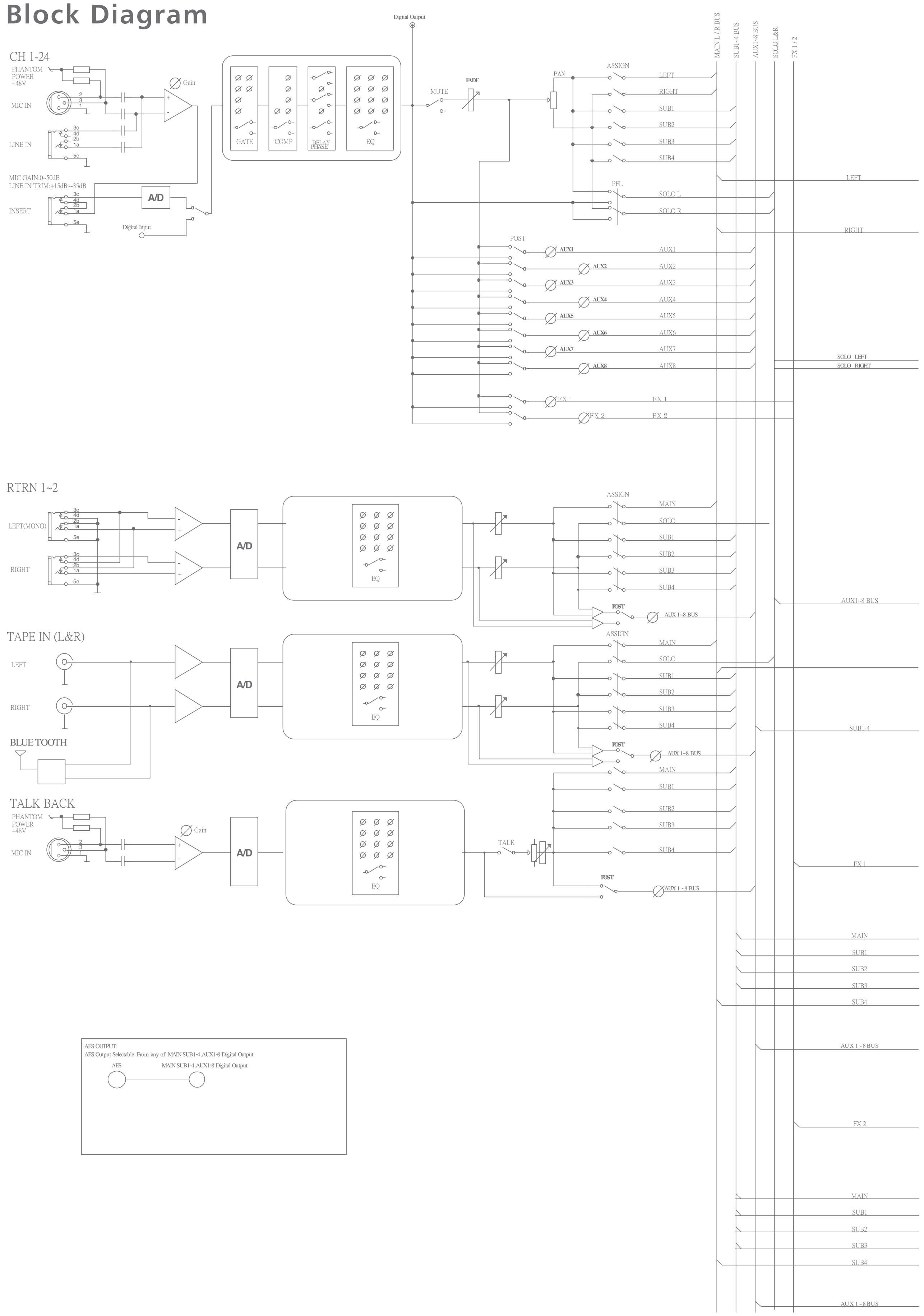
5

Mono Output	
Maximum Output Level	+22dBu
AUX1~8 Outputs	
Maximum Output Level	+18dBu
SUB1~4 Outputs	
Maximum Output Level	+18dBu
Tape Outputs	
Maximum Output Level	+14dBu
Control Room Outputs	
Maximum Output Level	+18dBu
System Crosstalk	
Input to Output (at+4dBu 1KHz)	-83dBu
Adjacent Channels (at+4dBu 1KHz)	-82dBu
Noise (Bus noise)	-85dBu
Noise Gate	
Threshold Range	-84dBu - 0dB
Attack time	0.5mS~200mS
Release time	5mS~1S
Compressor	
Threshold Range	-30dBu -+20dB
Attack time	0.5mS~200mS
Release time	10mS~1S
Ratio	1:1~1:10
Gain	0dBu - +24dB
EQ	
Low (Low Pass or Low Shelf)	21Hz~19.2KHz +/- 24dB
Low Mid	21Hz~19.2KHz +/- 24dB
High Mid	21Hz~19.2KHz +/- 24dB
High (High Pass or High Shelf)	21Hz~19.2KHz +/- 24dB
Digital Audio	
ADC Dynamic Range	114dB
DAC Dynamic Range	114dB
Internal Processor	SHARC 32-bit, floating point
ADC,DAC bit depth / Sampling Rate	24 bit / 48 kHz
Impedances	
Microphone inputs	1.4K Ω
Channel Insert return	2.5K Ω
All other inputs	10K Ω or greater
Tape out	1K Ω
All other outputs	120 Ω
Operating Free-air temperature range	**0-40C**
Storage temperature range	**-20~60 C**

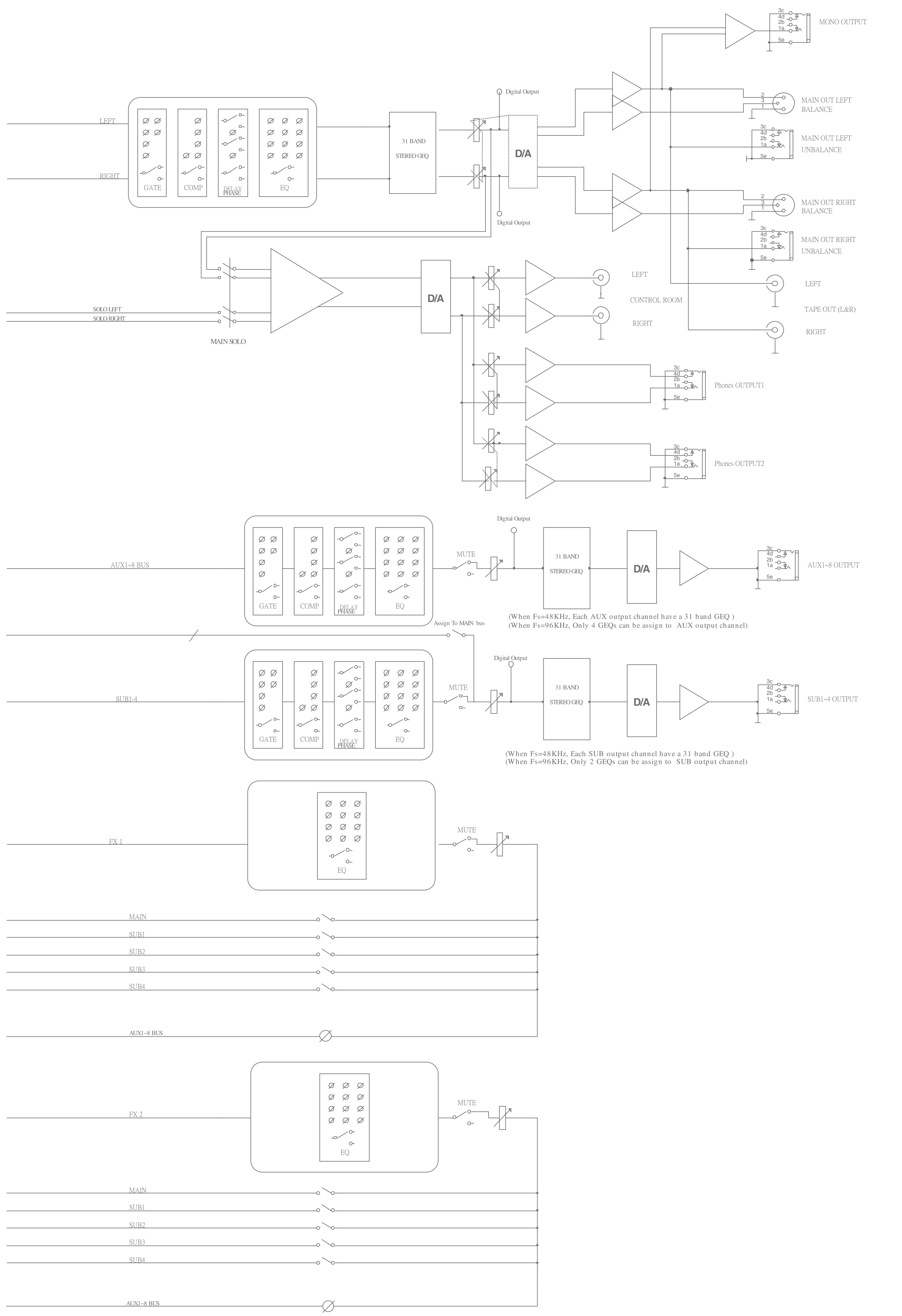
5

Technical Information

Block Diagram



Technical Information



6

Trouble Shooting

Problem	Possible Cause	Suggested Solution
Pops and clicks/distortion	Noise gate's release time is too short	Adjust the Release time to a suitable value
	The input's gain level is too high	Adjust the input gain to a suitable level
No Output on Channel	No signal input	Make sure there is signal in the input channel
	The input channel not assigned to an output bus	Assign the input to an output bus
Fader movements don't affect the audio	The fader has failed	Change the parts ask the qualified personnel
	The input channel not assigned to an output bus	Assign the input to an output bus
No output on the monitor bus	The volume of monitor bus is too low	Set a suitable volume for the monitor bus (headphone or Control Room)
	No channel has been solo'd	Make sure the Solo button of the bus which you want to monitor has been pressed and illuminated

7

Warranty

Warranty registration of the unit to BBE Sound, Inc. is not necessary. It is strongly recommended that you retain a copy of the bill of sale for future reference.

IT IS THE SOLE RESPONSIBILITY OF THE END USER TO PROVIDE THE BILL OF SALE OR OTHER MEANS OF PROOF OF PURCHASE TO VALIDATE THE WARRANTY IF WARRANTY SERVICE IS REQUESTED.

The BBE MP24M is warranted against defects in material and workmanship for a period of one (1) year from date of purchase from BBE Sound Inc. or from an authorized dealer. During this period, we will repair units free of charge providing that they are shipped prepaid to:

BBE Sound, Inc.
2548 E. Fender Avenue
Fullerton, California, 92831

We will pay return UPS shipping charges within the USA. All charges related to non-UPS shipping, including customs clearance, will be billed. The warranty will be honored for the longer of either 90 days from the date of any service or the remainder of the original 5 Year factory warranty.

This warranty will be consider null and void by BBE Sound, Inc. if any of the following is found:

1. The equipment has been physically damaged.
2. The equipment shows signs of abuse.
3. The equipment has been electrically damaged by improper connection or attempted repair by the customer or a third party.
4. The equipment has been modified without authorization.
5. The bill of sales indicates that the purchase date of the equipment is not within the warranty period.

All non-warranty repairs are warranted for a period of 90 days from the date of service.

BBE Sound, Inc. is NOT LIABLE FOR CONSEQUENTIAL DAMAGES. Should the unit fail to operate for any reason, our sole obligation is to repair it as described above.

DO NOT RETURN ANY PRODUCT TO THE ABOVE ADDRESS WITHOUT INSTRUCTIONS AND AUTHORIZATION ISSUED BY BBE, FULLERTON CALIFORNIA

BBE Sound Inc.

www.BBEsound.com