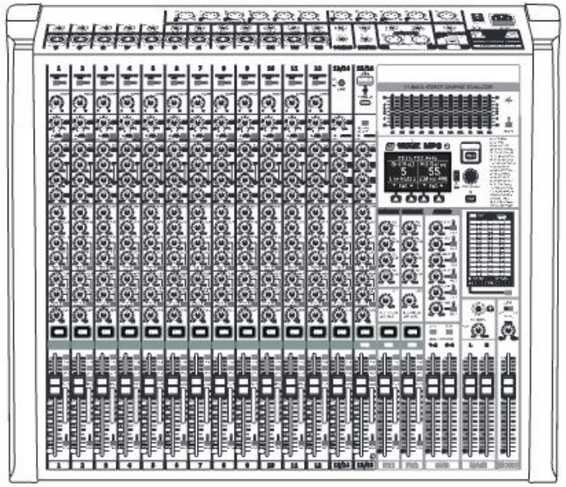


DynaTech

DynaTech

PROFESSIONAL MIXER

PRX12 & PRX16 & PRX24



Instruction Manual

Catalogue

Balanced/Unbalanced—What's the Difference?.....1

Signal Levels and the Decibel.....1

EQ or not to EQ.....2

Ambience.....3

The Modulation Effects.....3

Compression.....3

Setup.....4

Functional specification.....5-8

Effector and USB function operation.....9-12

Cabinet installation.....12

List of sockets and plugs.....12

Technical parameter.....13-14

Matters needing attention.....15

Troubleshooting.....16

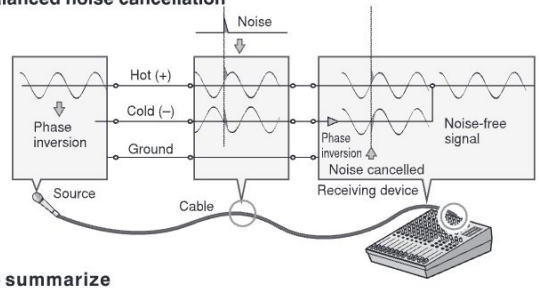
The operating theory of the sound mixer

If you are using the mixer for the first time, you may need to read this manual and select the basic operations according to your needs to help you better play the function and performance of the mixer.

Balanced/Unbalanced—What's the Difference?

In a word: “noise." The whole point of balanced lines is noise rejection, and it's something they're very good at. Any length of wire will act as an antenna to pick up the random electromagnetic radiation we're constantly surrounded by: radio and TV signals as well as spurious electromagnetic noise generated by power lines, motors, electric appliances, computer monitors, and a variety of other sources. The longer the wire, the more noise it is likely to pick up. That's why balanced lines are the best choice for long cable runs. If your “studio" is basically confined to your desktop and all connections are no more than a meter or two in length, then unbalanced lines are fine - unless you're surrounded by extremely high levels of electromagnetic noise. Another place balanced lines are almost always used is in microphone cables. The reason for this is that the output signal from most microphones is very small, so even a tiny.amount of noise will be relatively large, and will be amplified to an alarming degree in the mixer' s high-gain head amplifier.

Balanced noise cancellation

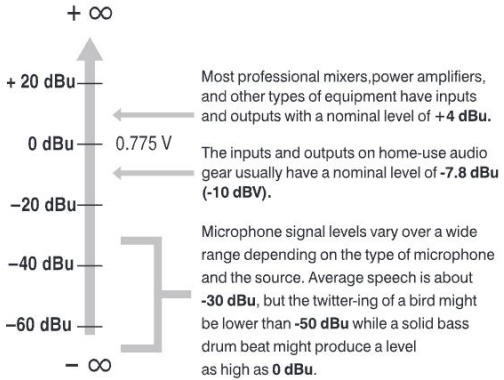


To summarize

Microphones	Use balanced lines.
Short line-level runs	Unbalanced lines are fine ifyou're in a relatively noise-free environment.
Long line-Level runs	The ambient electromag-netic noise level will be the ultimate deciding factor, but balanced is best.

Signal Levels and the Decibel

Let's take a look at one of the most commonly used units in audio: the decibel (dB). If the smallest sound that can be heard by the human ear is given an arbitrary value of 1,then the loudest sound that can be heard is approximately 1,000,000 (one million) times louder. That's too many digits to deal with for practical calculations, and so the more appropriate "decibel” (dB) unit was created for sound-related measurements. In this system, the difference between the softest and loudest sounds that can be heard is 120 dB. This is a non-linear scale, and a difference of 3 dB actually results in a doubling or halving of the loudness. You might encounter a number of different varieties of the dB: dBu, dBV, dBM and others, but the dBu is the basic decibel unit. In the case of dBu, “0 dBu" is specified as a signal level of 0.775 volts. For example, if a microphone's output level is -40 dBu (0.00775 V), then to.raise that level to 0 dBu (0.775 V) in the mixer's preamp stage requires that the signal be amplified by 100 times. A mixer may be required to handle signals at a wide range of levels, and it is necessary to match input and output levels as closely as possible. In most cases the “nominal" level for a mixer's input and outputs is marked on the panel or listed in the owner's manual.



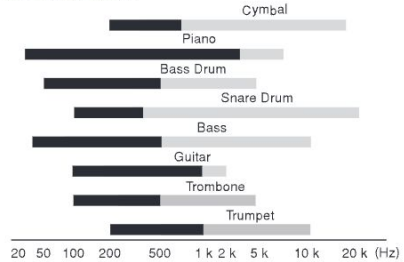
EQ or not to EQ

In general: less is better. There are many situations in which you'll need to cut certain frequency ranges, but use boost sparingly, and with caution. Proper use of EQ can eliminate interference between instruments in a mix and give the overall sound better definition. Bad EQ - and most commonly bad boost -just sounds terrible.

Cut for cleaner mix

For example: cymbals have a lot of energy in the mid and low frequency ranges that you don't really perceive as musical sound, but which can interfere with the clarity of other instruments in these ranges. You can basically turn the low EQ on cymbal channels all the way down without changing the way they sound in the mix. You'll hear the difference, however, in the way the mix sounds more"spacious," and instruments in the lower ranges will have better definition. Surprisingly enough, piano also has an incredibly powerful low end that can benefit from a bit of low-frequency roll-off to let other instruments - notably drums and bass - do their jobs more effectively. Naturally you won't want to do this if the piano is playing solo. The reverse applies to kick drums and bass guitars: you can often roll off the high end to create more space in the mix without compromising the character of the instruments.You'll have to use your ears, though, because each instrument is different and sometimes you'll want the"snap" of a bass guitar, for example, to come through.

The fundamental ■ and harmonic □ frequency ranges of some musical instruments.



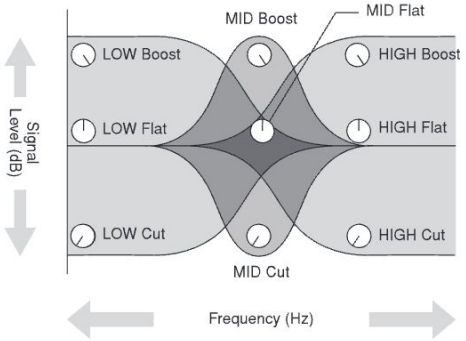
■ Fundamental: The frequency that determines the basic musical pitch.

□ Harmonics: Multiples of the fundamental frequency that play a role in determining the timbre of the instrument.

Some Frequency Facts
The lowest and highest frequencies than can be heard by the human ear are generally considered to be around 20 Hz and 20,000 Hz, respectively. Average conversation occurs in the range from about 300 Hz to about 3,000Hz. The frequency of a standard pitchfork used to tune guitars and other instruments is 440 Hz (this.corresponds to the"A3"key on a piano tuned to concert pitch). Double this frequency to 880 Hz and you have a pitch one octave higher (i.e."A4"on the piano keyboard). In the same way you can halve the frequency to 220 Hz to produce"A2"an octave lower.

Boost with Caution

If you're trying to create special or unusual effects, go ahead and boost away as much as you like. But if you're just trying to achieve a good-sounding mix, boost only in very small increments. A tiny boost in the midrange can give vocals more presence, or a touch of high boost can give certain instruments more"air".Listen, and if things don't sound clear and clean try using cut to remove frequencies that are cluttering up the mix rather than trying to boost the mix into clarity. One of the biggest problems with too much boost is that it adds gain to the signal, increasing noise and potentially overloading the subsequent circuitry.



Ambience

Your mixes can be further refined by adding ambi-ence effects such as reverb or delay. internal effects can be used to add reverb or delay to individual channels in the same way as external effects processors.

Reverb and Delay Time

The optimum reverb time for a piece of music will depend on the music's tempo and density, but as a general rule longer reverb times are good for ballads,while shorter reverb times are more suited to up-tempo tunes. Delay times can be adjusted to create a wide variety of“grooves". When adding delay to a vocal, for example, try setting the delay time to dotted eighth notes corresponding to the tune's tempo.

Reverb Tone

Different reverb programs will have different“reverb tone”due to differences in the reverb time of the high or low frequencies.' Too much reverb, particularly in the high frequencies, can result in unnatural sound and interfere with the high frequencies in other parts of the mix. It's always a good idea to choose a reverb program that gives you the depth you want without detracting from the clarity of the mix.

Reverb Level

It's amazing how quickly your ears can lose perspec-tive and fool you into believing that a totally washed-out mix sounds perfectly fine. To avoid falling into this trap start with reverb level all the way down, then gradually bring the reverb into the mix until you can just hear the difference. Any more than this normally becomes a“special effect."

The Modulation Effects:

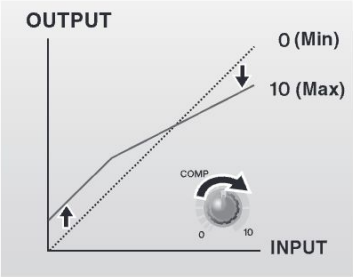
Phasing, Chorus, and Flanging

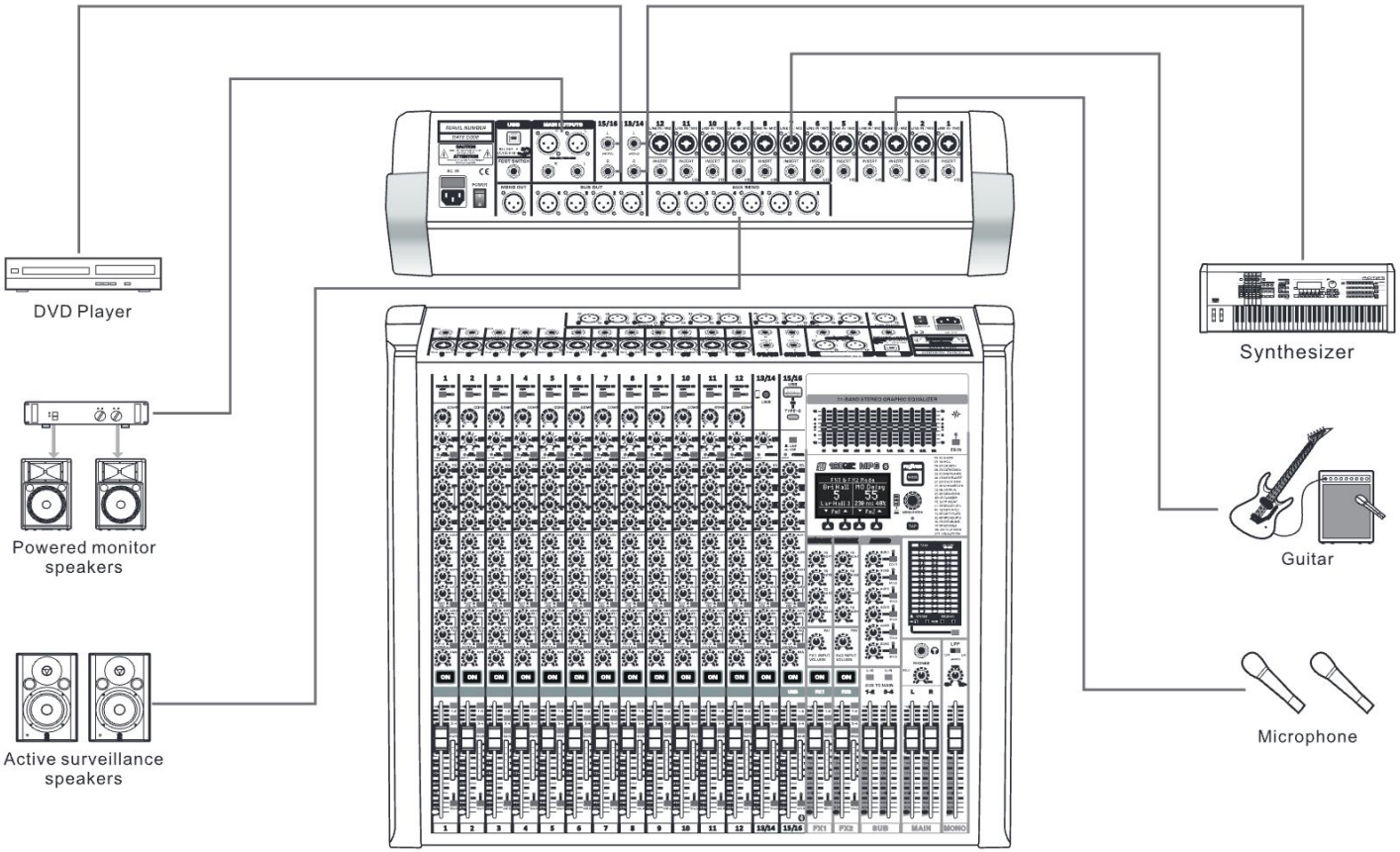
All of these effects work on basically the same princi-ple: a portion of the audio signal is “time-shifted" and then mixed back with the direct signal. The amount of time shift is controlled, or“modulated", by an LFO (Low-frequency Oscillator).For phasing effects the shift is very small. The phase difference between the modulated and direct signals causes cancellation at some frequencies and rein-forces the signal at others and this causes the shim-mering sound we hear.

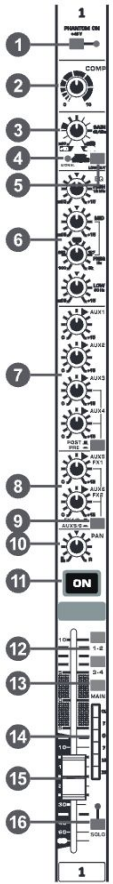
For chorus and flanging the signal is delayed by sev-eral milliseconds, with the delay time modulated by an LFO, and recombined with the direct signal. In addition to the phasing effect described above, the delay modulation causes a perceived pitch shift which,when mixed with the direct signal, results in a harmonically rich swirling or swishing sound. The difference between chorus and flanging effects is primarily in the amount of delay time and feedback used-flanging uses longer delay times than chorus,whereas chorus generally uses a more complex delay structure. Chorus is most often used to thicken the sound of an instrument, while flanging is usually used as an outright“special effect" to produce other-worldly sonic swoops.

Compression

One form of compression known as “limiting” can,when properly used, produce a smooth, unified sound with no excessive peaks or distortion. A com-mon example of the use of compression is to “tame” a vocal that has a wide dynamic range in order to tighten up the mix. With the right amount of compres-sion you'll be able to clearly hear whispered pas-sages while passionate shouts are still well balanced in the mix.Compression can also be valuable on bass guitar.Too much compression can be a cause of feedback, however, so use it sparingly. Most compressors require several critical parameters to be set properly to achieve the desired sound.The compressor makes achieving great sound much easier: all you need to do is set a single“compres-sion" control and all of the pertinent parameters are automatically adjusted for you.




DVD Player
Powered monitor speakers
Active surveillance speakers
Synthesizer
Guitar
Microphone



1. +48V phantom power switch

Phantom power to the input channel XLR jacks can be activated via the PHANTOM switch, Phantom power +48V is required to use a condenser microphone, the LED glows to indicate that phantom power is on. Dynamic microphones can usually still be used, as long as they are balanced.

2. COMP controlling knob

Adjust the amount of compression effect applied to the channel. When you turn the [COMP] knob to the right, the threshold, ratio, and output gain are adjusted simultaneously.
Threshold: +22dBu to-8dBu
Ratio: 1:1 to 4:1
Output gain: 0dB to +7dB
Starting time: about 25ms
Interpretation time: about 300ms

3. GAIN control

Adjusts the input signal level. For the best balance between signal-to-noise ratio and dynamic range, adjust the GAIN so that the CLIP indicator lights occasionally and briefly on the highest input transient.MIC Input Adjustment Range +10 to +60dB. LINE Input Sensitivity Adjustment Range -10 to +40dB.

4. SIGNAL led

Check the input signal.

5. LOW CUT switch

The console's mono channel is also equipped with a steep-edge low-cut filter, which you can use to eliminate unwanted portions of the low-frequency signal, attenuating frequencies below 75 Hz at a slope of 18 dB/octave.

6. Equalizer

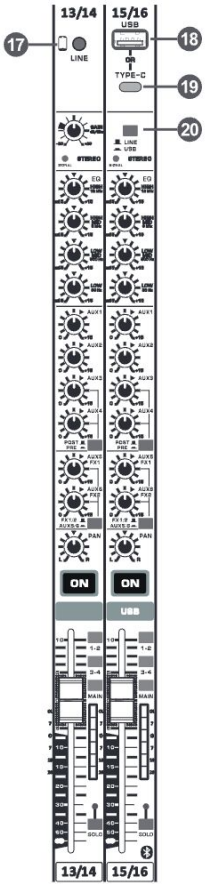
All single input channels are equipped with a 4-band sound quality adjustment knob with a semi-parametric mid range. Each band can be raised/lowered by up to 15 dB, and the equalizer has no effect in the center position. The high-frequency and low-frequency ranges are 12 kHz and 80 Hz. For the mid-frequency range the console has a semi-parametric tone adjustment between 100 Hz and 8 dry Hz. The tone is raised/lowered with the MID knob and the frequency is adjusted with the FREQ knob.

7. AUX control

The knobs of AUX1/2 adjust the level of the fader signals (before the faders are adjusted). The PRE switch on AUX4 can be used to select AUX3/4. When the PRE switch is pressed, it will send the pre-fader signals (signals before being adjusted by the fader), and when the POST switch is popped up, it will send the post-fader signals (signals after being adjusted by the fader) to the AUX bus.

8. FX control

These knobs adjust the signal level of Input FX Buses 1 and 2. Since the signal fed into the FX buses is post-fader (i.e., after being adjusted by the channel fader), its level is always determined by the combination of the FX control and the channel attenuator.
For stereo channels, the L (odd) and R (even) input signals are mixed before being routed to the FX buses.
Note: These knobs function as FX controls **only when the AUX5/AUX6 switch is in the OFF (▟) position**. If the switch is in the ON position, these knobs will adjust the output signal to AUX5 and AUX6 buses instead.



9. FX1, FX2 / AUX5, AUX6 switch
This switch selects whether the channel signal is sent to AUX Buses 5 and 6 or to FX Buses 1 and 2. When the switch is in the ON (▂) position, the signal is routed to AUX5 and AUX6 buses; when it is in the OFF (▃) position, the signal is sent to the FX buses. Please note that the signal sent to these buses is post-fader (i.e., after being adjusted by the channel fader).

10. PAN control
The PAN (Positioning Acoustic Pan) knob adjustment is used to determine the position of the channel's signal in the stereo field.
If grouping is used, you can assign the signal to either SUB1/3 (with the PAN knob turned to the far left) or SUB2/4 (with the knob turned to the far right).

11. ON switch
When this switch is on, signals from the corresponding channel can be sent to the bus, which is illuminated when it is on. When this switch is off, the corresponding signal input will not be sent to the AUX or SUB bus. Even if the [ON] switch is off, the SOLO signal for each channel can still be heard through the [PHONES] jack.

12. 1-2/3-4 switch
When this switch is turned on, the signal of the corresponding channel can be sent to the SUB bus.

13. MAIN switch
When this switch is turned on, the signal from the corresponding channel is sent to the stereo (MAIN MIX) main mix bus.

14. Level indicator light
Channel level display.

15. Fader channel
Adjusts the level If the channel signal. Use these Control to adjust the balance between the various channels.

16. SOLO switch
When this switch is turned on, the signal before the fader of the corresponding channel is monitored, and the input level of the corresponding channel before the fader is displayed on the level meter.
When the SOLO switch is turned on, the corresponding LED lights up.

17. LINE input jacks(stereo channel)
You can connect 3.5mm phone plugs to these jacks.

18. USB input jacks
The USB port allows you to insert a USB flash drive to play music/recordings.

19. TYPE-C input jacks
To connect a mobile phone or iPad, an OTG adapter or a dual-ended USB-C cable is required. add the playback signal to the USB channel. (Note: There may be system version compatibility issues when connecting to mobile phones.)

Caution: The computer USB input and TYPE-C input **cannot be used simultaneously**.

20. LINE input / USB input signal transfer switch
USB IN and LINE IN cannot be used simultaneously. Switch between USB IN and LINE IN via the button.

21. 11-band stereo graphic equalizeri

Equipped with an 11-band graphic equalizer. The equalizer is turned on by the EQ IN button, and you can use it to adjust the tone to the specific room space. At position "0", no frequency response processing is performed. To increase the frequency range, push the corresponding fader up or down.

22. FX TO AUX control

AUX control adjusts the effect signal of the aux setting effect processor.

23. FX1/2 control

The FX control is sent to the built-in effect processor.

24. AUX control

AUX control adjusts AUX output volume.

25. SUB TO MAIN switch

When this switch is turned on, the LEFT or RIGHT signal of the SUB is assigned to the MAIN MIX.

26. LEVEL indicator

Level meter LED displays the main output and monitoring signal levels.
When the output signal reaches the clipping time, the CLIP indicator lights up.

27. SUB switch

Listen to the output by selecting the MAIN output or SUB output through the switch, and by selecting the SUB1/2/3/4 output through the switch.

28. The phones earphone monitoring socket

Can be connected with a pair of stereo earphones to monitor the output signal.

29. Monitor control

The PHONES control adjusts the monitor output volume.

30. FX fader

FX Fader adjustment (MAIN MIX) the main mix sets the effects signal of the effects processor.

31. SUB fader

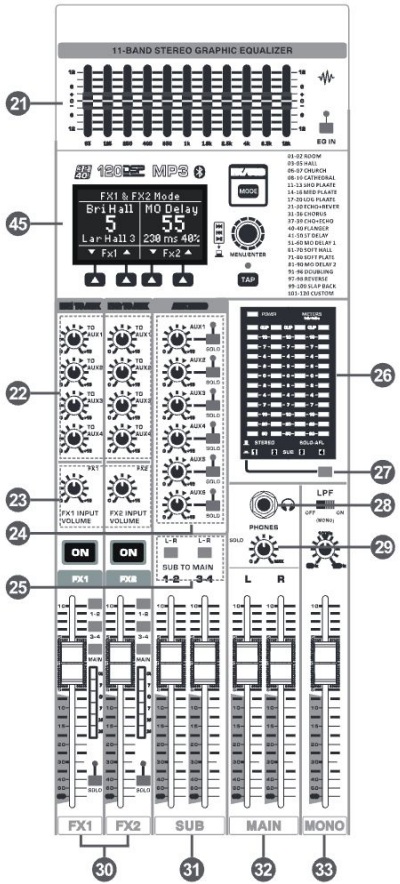
SUB fader adjusts the SUB output volume.

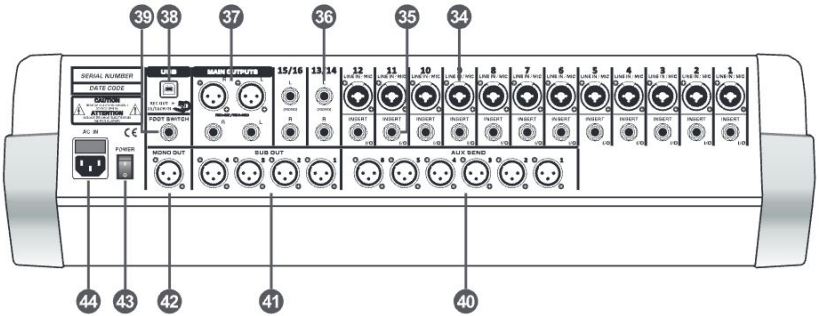
32. MAIN fader

MAIN fader adjusts the volume of the main output.

33. MONO channel

The MONO channel is equipped with an ultra-low output filter, and the frequency range is adjusted between 80Hz-120Hz through the potentiometer. To activate the filter, please turn the switch on the channel to the ON position.





34. MIC/LINE input jacks

The balanced XLR microphone input jack and the balanced TRS ear instrument input jack can be connected.

35. INSERT jacks

These jacks can be used to insert an external signal-processing device between the equalizer and fader of the corresponding monaural input channel. The INSERT jacks are ideal for connecting devices such as graphic equalizers,compressors,or noise filters into the corresponding channels.

36. LINE input jacks(stereo channel)

The unbalanced TRS ear instrument input jack can be connected.

37. MAIN OUT (L, R) jacks

These jacks deliver the mixer's stereo output. You use these jacks, for example, to connect to the power amplifier driving your main speakers.

38. Computer USB Jacks

Connect the computer sound card to play music/recording, and add the playback signal to the USB channel.

39. The foot switch socket

can be connected with a foot switch to control the output of the effector to turn off or on.

40. AUX 1/2/3/4/5/6 SEND jack

External effects devices or stage/studio monitoring systems can be connected.

41. SUB 1/2/3/4 marshalling output socket

can be connected to the input port of multi track recorder, external mixer or other equipment.

42. MONO output jack

The MONO output interface is ideal for connecting a subwoofer, and can also set the frequency range required by the subwoofer.

43. POWER switch

Use the POWER switch to turn on the mixing console. The POWER switch should always be in the "Off" position when you are about to connect your unit to the mains. To disconnect the unit from the mains, pull out the main cord plug. When installing the product, ensure that the plug is easily accessible.

44. Fuse holder/iec mains receptacle

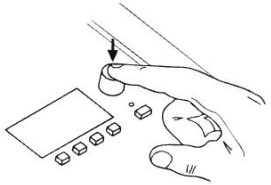
The console is connected to the mains via the cable supplied, which meets the required safety standards. Blown fuses must only be replaced by fuses of the same type and rating. The mains connection is made via a cable with IEC mains connector. An appropriate mains cable is supplied with the equipment.

45. Effector and USB function operation

Provides two independently controllable, identically configured 32-bit stereo effect units-FX1 and FX2. Each unit provides 100 program presets, which can be selected by using the up/down buttons. The preset parameters can be edited and stored as user presets (101-120), see Figure 5 for details.100 presets are divided into groups according to their different effect structures. The programs in each preset group are arranged in order, the higher the number, the stronger the type of FX provided. Presets 1-20 provide high-quality reverb operation effect programs, which are also suitable for live performances, recording studios or home recording applications. Program numbers 21-40 provide mixed effects of echo + reverb and chorus Type, while the numbers 41-60 provide different delay effects. The last group from 61-100 provides different doubling effect presets and special delay and reverberation programs. During the initialization period of the FX unit (when the power is turned on), the preset 05 (BRIGHT HALL, Large HALL 3) is selected as FX1, and the FX2 unit is set to the preset 55 (MONO DELAY, 230ms 40%). These two effects are equally applicable to live performances and recording applications. This is a factory preset configuration that can be changed at any time.

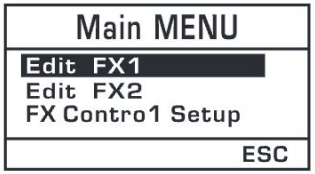
1. Menu mode

(1.1) Press the MENU/ENTER rotary encoder in the effect mode to enter the menu mode. When entering the menu mode, the main menu is displayed.



Picture(1.1):Press the MENU/ENTER rotary encoder to enter the menu mode

(1.2) In the menu mode, the MENU/ENTER rotary encoder and the four function keys at the bottom of the display screen are used for navigation and parameter selection and editing. Rotate MENU/ENTER to move the encoder to the left or right to move the cursor in the menu. Press the MENU/ENTER rotary encoder to select or execute the highlighted menu item. If the menu item is highlighted, the corresponding dialog box will open. If the parameter is highlighted, the value will be displayed in reverse font and can be edited by turning the MENU/ENTER rotary encoder. Press MENU/ENTER again to apply the edited parameter value to the rotary encoder. The bottom line of the display shows the functions of the four function keys. The following table describes some of the functions available in the menu dialog.and recording applications. This is a factory preset configuration that can be changed at any time.

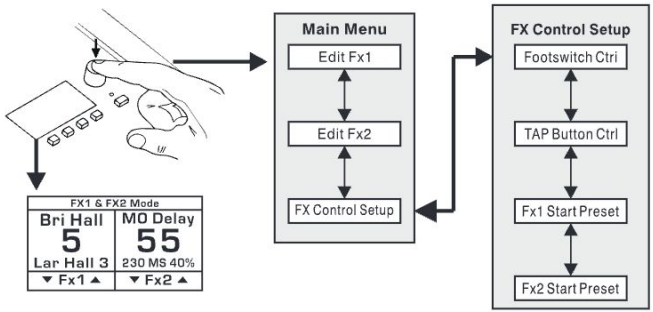


Picture(1.2):The main menu of the menu mode

(1.3) Function description of function keys

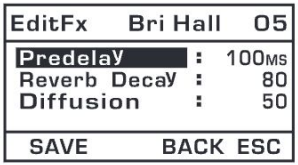
BACK	Return to the previous menu.
ESC	Exit all edits in the dialog box and return to the effect mode.
◄OR►	Move the cursor to the left or right.
SAVE	Open the FX user preset list and save the edited effect as a user preset.
OK	Confirm entry, for example when editing a user-preset name.

2. Menu structure



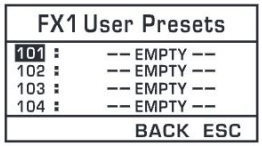
3. Edit FX1/FX2

(3.1) This dialog box allows you to edit the effect parameters of FX1 or FX2. The available parameters will depend on the type of effect, see Figure 4.4 for details. In addition, you can assign a custom name to the editing effect and store it as a user preset. Up to 20 user presets can be created.



Figure(3.1):Edit FX1 dialog

(3.2) Turn the MENU/ENTER rotary encoder to highlight an item in the left column of the Edit FX menu. Press and turn the encoder MENU/ENTER key to select the corresponding value in the right column. Now turn the MENU/ENTER to rotate the encoder to edit the value. Press MENU/ENTER again to rotate the encoder to return to the left column. Press the SAVE function key to save the edited effect as a user preset. The FX User Preset dialog box appears.



Figure(3.2):FX user preset dialog

(3.3) Turn the MENU/ENTER rotary encoder to select where the user preset editing effects should be stored. Press the MENU/ENTER rotary encoder to open the dialog box for setting the FX name.



Figure(3.3):Set FX name dialog

(3.4) This dialog box is used to change the description information of the user preset. Turn the MENU/ENTER rotary encoder to the left or right to edit the highlighted character. Press the MENU/ENTER rotary encoder to select the desired symbol and move the cursor to the next character. Press ◄ or ► function key to move the cursor to the left or right. Select the "↵" symbol or press the "OK" function key to exit name editing. Press the BACK function key to return to the menu.

4. FX control settings

(4.1) This dialog box is used to configure the effector FX1 or FX2. In addition, this dialog is used to select the default effect when the effect is turned on.

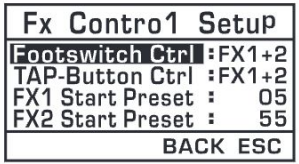


Figure (4.1): FX control setting dialog box

(4.2) FX control setting parameters

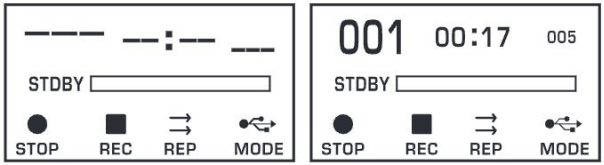
Parameter	Description instructions
Footswitch Ctrl	Select FX1, FX2 or FX1+2 to control one or two effects using the foot switch.
TAP-Button Ctrl	Select FX1, FX2 or FX1+2, and use the TAP button to control the delay time of one or two effects.
FX1 Start Preset	Effect FX1, select the default effect when the effect is turned on.
FX2 Start Preset	Effect FX2, select the default effect when the effect is turned on.

(4.3) Turn the MENU/ENTER rotary encoder to highlight the entry in the left column of the FX Control Setup dialog box. Press the MENU/ENTER key of the rotary encoder, select the corresponding value in the right column, then turn the MENU/ENTER rotary encoder to edit the value, and press the MENU/ENTER rotary encoder again to apply the new value. The entries in the FX Control Setup dialog box are shown in the table (Figure 4.4). Press the BACK function key to return to the menu. Press the ESC function key to return to the effect mode.

(4.4) Effect Parameter Setting

No.	Effect group	Description	Suitable for use
01-10	Bright Hall	Bright reverb, concert hall, church, cathedral	Vocals, Horn, Strings
11-20	Bright Plate	Bright plate, no audible reflections	Piano, Guitar, Drums, Vocals
21-30	Echo+Reverb	Bright echo/reverb mix	Specially for Live vocals, strings, horns
31-33	Chorus	Light chorus	Piano, Guitar, Bass, Rhodes, Strings
34-36		Deep chorus	Organ, Piano, Guitar, Bass, Rhodes, Strings
37-39	Chorus+Echo	Deep chorus with fading echo	Organ, Guitar , Strings
40	Flanger	Real late sixties jet flanger	Drums, Prcussion, Bass, Strings, Vocals
41-50	Stero Delay	L/R echoes	Combined with a reverb effect well suited for vocals, horns, strings
51-60	Mono Delay	Centered echoes, slowly fading (40% feedback)	Combined with a reverb effect well suited for vocals, horns, strings
61-70	Soft Hall	Extremely smooth reverb, concert hall, church, cathedral	Vocals, Horns, Strings, Home recording
71-80	Soft Plate	Smooth plate, no audible reflections	Piano, Guitar, Drums, Vocals, Home recording
81-90	Mono Delay	Centered echoes, vastly fading (20% feedback)	Fast fading slap back echoes for vocals, percussion. Combined with a reverb effect well suited for vocals, horns, strings
91-92	Doubling	Doubling effect without coloration	Vocals, Horns,Strings,Organ
93-96		Doubling effects	Snare drum, Kick drum
97-98	Reverse	Reverse reverb	Snare drum, Kick drum
99	Slap-Back	Fast slap back echo without repeats	Vocals, Kick drum, Snare drum
100		Slow slap back echo without repeats	Vocals, Kick drum, Snare drum

5. USB player operation



(5.1) Button description: (STOP) stop playback, (REC) recording mode, (REP) sequence and loop playback conversion, (MODE) U disk playback and Bluetooth switching.

(5.2) Through the MODE button, DSP and USB switch between interface operations.

(5.3) After inserting the U disk in the USB display interface to read the track, press the rotary encoder MENU/ENTER button to play and pause, and the rotary encoder to switch the song up/down.

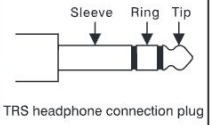
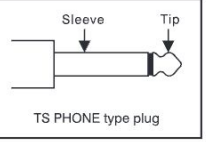
(5.4) For recording operation and playback, press the REC button to enter the start recording mode. During the recording, you can press the MENU/ENTER button of the rotary encoder to pause the recording. After the recording is completed, press the STOP button to stop the recording, and enter the recording folder to play back the current recording. File, you can also turn the MENU/ENTER rotary encoder to select the track playback of the recording file, long press the REC button to exit the recording and switch to USB playback.

Installation

1. Cabinet installation

This mixer needs at least a 12U cabinet, and the space is for the connection of cables. Cabinet size: 10U is about 445mm, 11U is about 489mm, 12U is about 533mm, 14U is about 623mm, in the mixer package, you can find two assembly cabinet side panels, they can be installed on both sides of the mixer. Before assembling the cabinet side panels and fixing them on the mixer, please remove the screws on the original left and right side panels. Then please install the two cabinet side panels with the screws provided with the cabinet side panels. Please note that the assembly side panel distinguishes the left and right. After installation, install the mixer on a universal 12U rack. Please always maintain sufficient air circulation to avoid overheating of the machine.

2. List of sockets and plugs

MIC/LINE, MIC STEREO OUT	Pin 1: Ground Pin 2: Hot (+) Pin 3: Cold (-)	
MIC/LINE, AUX SEND SUB OUT, MONO OUT STEREO OUT	Tip: hot (+) Ring: cold (-) Sleeve: Ground	
PHONES	Tip: L Ring: R Sleeve:Ground	
LINE (STEREO INPUT CHANNEL)	Tip: hot (+) Sleeve: Ground	
EFFECT MUTE FOOT SWITCH	Tip:trigger Sleeve: Ground	

Technical parameter

Microphone input (MIC preamplifier)

Type	XLR, Balanced, Separate input circuit

MIC E.I.N. (20Hz - 20KHz)

@0Ω Source impedance	-134dB/135.7dB A-Weighted
@50Ω Source impedance	-131dB/133.3dB A-Weighted
@150Ω Source impedance	-129dB/130.5dB A-Weighted
Frequency response	<10Hz - 150KHz(-1dB) <10Hz - 200KHz(-3dB)
Gain range	+10dB ~ +60dB
Maximum input level	+12dBu@ +10dB Gain
Impedance	About 2.6KΩ balanced
Signal-to-noise ratio	110dB / 112dB A-Weighted (0dBu In @ +22dB Gain)
Distortion (total harmonic distortion+noise)	0.005% / 0.004% A-Weighted

Line input

Type	6.3mm stereo channel plug , balanced
Impedance	About 20 KΩ balanced 10 KΩ unbalanced
Gain range	-10dB~+40dB
Maximum input level	+22dBu@ 0dB Gain

Signal attenuation (crosstalk attenuation)

Main output fader off	90dB
Channel mute	89dB
Channel fader off	89dB

Frequency response

Microphone input to main output	
<10Hz ~ 90KHz	+0dB / -1dB
<10Hz ~ 160KHz	+0dB / -3dB

Stereo input

Type	6.3mm stereo channel plug , unbalanced
Impedance	About 10KΩ
Maximum input level	+22dBu

EQ mono channel

Low	80Hz / ±15dB
Mid sweep	100Hz-8kHz / ±15dB
High	12kHz / ±15dB

EQ stereo channel

Low	80Hz / ±15dB
Low Mid	500Hz / ±15dB
High Mid	3kHz / ±15dB
High	12kHz / ±15dB

Aux auxiliary output

Type	6.3mm mono channel plug , unbalanced
Impedance	About 120Ω
Maximum output volume	+22dBu

Sub grouping output

Type	6.3mm mono channel plug , unbalanced
Impedance	About 120Ω
Maximum output volume	+22dBu

Main output

Type	XLR, balanced, 6.3mm stereo channel plug, balanced
Impedance	About 240Ω balanced, 120Ω unbalanced
Maximum output volume	+22dBu

Headphone output

Type	6.3mm stereo channel plug , unbalanced
Maximum output volume	+19dBu/150Ω(+25dBm)

Digital effector(DSP)

Converter	32-bit digital-to-analog converter 64/128-times (sampling rate)
Scan speed	40KHz

Main Mix system data

Noise	
Master fader-∞ Channel fader-∞	-99dB/-101dB A-Weighted
Master fader 0dB Channel fader-∞	-84dB/-87dB A-Weighted
Master fader 0dB Channel fader 0dB	-80dB/-82dB A-Weighted

Output power

4 0hm per channel

Power supply

Mains voltage	
USA/Canada	120 V~, 60 Hz
China/Korea	220 V~,50/60 Hz
Europe/Australia	230 V~, 50 Hz
Japan	100 V~,50-60 Hz
Fuse 100-120V~	
Fuse 220-240 V~	

Power consumption

Power consumption	
Mains connector	IEC standard receptacle

Dimensions/weight

Dimensions (H x W x D)
Weight

Please note these specifcations are preliminary and conceptual in nature,and as such are subject to change as product development progresses.This information is supplied for market research purposes only and is notto be made public in any manner.

Important safety instructions



The terminal device with this mark has a strong current and presents a danger of electric shock. Use only high quality professional speaker cables with ¼ "TS or twist lock plugs. All installations or adjustments must be carried out by qualified professionals.

This sign reminds you that there is a dangerous voltage in the product that is not insulated and there is a risk of electric shock.

This sign reminds you to check the attached important usage and maintenance instructions. Please read the manual.

CAUTION

To avoid the risk of electric shock, do not open the top cover (or back baffle). There are no parts in the equipment that can be repaired by users. Please refer the maintenance to a qualified professional.

CAUTION

To avoid the risk of fire or electric shock, do not expose this equipment to rain or moisture. This equipment should not be splashed by liquid, and containers containing liquid should not be placed on it, such as vases, etc.

CAUTION

Maintenance instructions are for qualified professional maintenance personnel only. To avoid the risk of electric shock, do not perform any repairs other than those mentioned in the instruction manual. All repairs must be carried out by qualified professionals.

1. Please Read these instructions.
2. Please keep these instructions properly.
3. Please heed all warnings.
4. Please follow all instructions.
5. Do not use this product near water.
6. Please clean the product with a dry cloth.
7. Do not block the vent. Please follow the manufacturer's instructions when installing this product.
8. Do not install this product near heat source, such as radiator, stove or other heat generating equipment (including power amplifier).
9. Do not remove the safety device of the polar plug or the ground plug. The ground plug is composed of two plug contacts and a ground head. If the supplied plug does not fit your socket, please find an electrician to replace it with a suitable socket.
10. Properly protect the power cord from tramping or puncturing, especially paying attention to power plugs, multi-purpose sockets and equipment connections.
11. Please only use the ancillary equipment and accessories specified by the manufacturer.
12. Use only carts, shelves, tripods, stands and tables specified by the manufacturer or sold with the product.
    If using a trolley to carry equipment, please place the equipment safely to avoid injury caused by the trolley and equipment tipping over.
13. In case of lightning and thunder or if the equipment is not used for a long time, please pull out the power plug.
14. All repairs must be carried out by qualified maintenance personnel. Repairs are required when equipment is damaged, such as damaged power cord or plug, liquid flowing into or foreign object falling into the equipment, equipment being rained on or damp, equipment not functioning properly or broken.
15. Grounding protection must be provided when the device is connected to the power supply.
16. If a power plug or appliance coupler is used as a power outage device, it shall be ensured that they are in a readily operable state at all times.
17. This product is only applicable to areas below 2000 meters above sea level. This product is only applicable to non-tropical climate conditions.



Troubleshooting

Problem	The solution
The power won't turn on.	☐ Is the console connected to a separate power source (generator, etc.) or an electrical outlet with a switch? Check that the unit is switched on.
No sound output	☐ Are external instruments (including microphones) and speakers connected correctly? ☐ Are the connection cables shorted? ☐ Are the [GAIN] knobs, channel faders, master faders, and [SUB] faders of each channel set to the appropriate level? ☐ Are the BUS DISTRIBUTION switch and [CD/MP3] switch set correctly?
No sound output from STEREO OUT jacks	☐ Are the [ON] switch and [MAIN] switch of the channel used turned on?
No sound is output from the SEND (AUX1-4) jacks.	☐ Are the [SEND MASTER] knob and [AUX 1-2] settings for each channel correct? ☐ Is the [ON] switch on for the channel you are using?
No sound output from the [PHONES] jack.	☐ Is the [SOLO] switch for channels not in use turned on? ☐ Turn off the [SOLO] switch.
Low, distorted or noisy sound	☐ Is the microphone connected to the [MIC] jack or [MIC/LINE] jack? ☐ Is the [PHANTOM +48V] switch turned on when using a condenser microphone? ☐ Is the [PAD] switch on? Turn this switch off when using a low output level sound source such as a microphone. ☐ Is the output signal level of the instrument connected to the mixer set correctly? ☐ When the output level of the connected instrument is +0dBu, either turn on the [PAD] switch on the mono channel, or use the stereo input channel. ☐ If an input channel provides both XLR input jacks and phone type input jacks, or both phone type input jacks and RCA pin jacks, are both jacks connected? Please use only one type of jack. ☐ Are the [GAIN] knobs, channel faders, master faders, and [SUB] faders of each channel set to the proper level? ☐ Is the effect or compressor level too high? Use the [FX] knob, [FX] fader and [COMP] knob to lower the level.

Problem	The solution
Cannot use effects	☐ Are the [FX] knobs for each channel adjusted to the proper level? ☐ Are the [FX] faders adjusted to the proper level? ☐ Is the [FX] bus assignment switch set correctly? ☐ If an external effect unit is connected to the [SEND(AUX1-2)] socket, is the [AUX1-2] knob of [SEND MASTER] set correctly?
Audio sound is not clear	☐ Is the [HPF] switch on? ☐ Is the equaliser adjustment (HIGH/MID/LOW) correct?
No output from console signal	☐ Are active speakers connected to the [MONO OUTPUTS] jacks? Adjust the signal output from the [MONOOUTPUTS] jack with the [MONO] fader.
Different left and right levels of stereo input signals	☐ Is [PAN] set to the centre position? If the image is set to the centre, try reversing the left and right input connections. If the left and right signals are reversed and the lower volume side is also reversed, check the instrument or device that is the source of the signal. ☐ Are certain types of cables being used to connect both left and right input signals? Cables with built-in impedance attenuate the signal.
The sound level is unstable and inconsistent.	☐ Is the compression level set too high? Adjust the reduced level with the [COMP] knob.