PROFESSIONAL MIXER

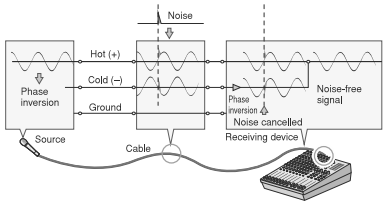
Instruction Manual

For get better performance and the utmost satisfaction from your new unit, please read this manual thoroughly before using, and retain it for future reference.

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



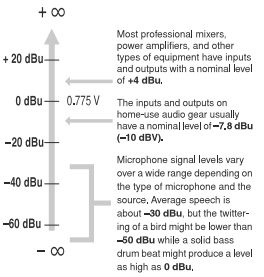
To summarize

Microphones	Use balanced lines.
Short line-level runs	Unbalanced lines are fine if you're in a relatively noise-free environment.
Long line-level runs	The ambient electromagnetic 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.



Most professional mixers, power amplifiers, and other types of equipment have inputs and outputs with a nominal level of **+4 dBu**.

The inputs and outputs on home-use audio gear usually have a nominal level of **–7.8 dBu (–10 dBV)**.

Microphone signal levels vary over a wide range depending on the type of microphone and the source. Average speech is about **–30 dBu**, but the twittering of a bird might be lower than **–50 dBu** while a solid bass drum beat might produce a level as high as **0 dBu**.

A mixer may be required to handle signals at a wide range of levels, and it is necessary 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.

To EQ or Not to EQ

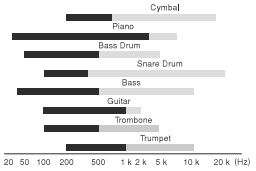
In general: less is better. There are many situations in which you'll need to cut certain frequency ranges, but use boost sparingly, and with caution. Proper use of EQ can eliminate interference between instruments in a mix and give the overall sound better definition. Bad EQ—and most commonly bad boost—just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

Cut for a 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.

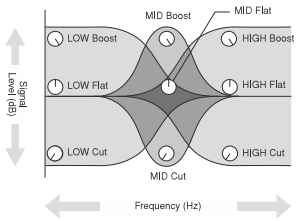
Some Frequency Facts

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 3,000 Hz. 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 ambience effects such as reverb or delay. The internal effects can be used to add reverb or delay to individual channels in the same way as external effects processors. (Refer to page 15).

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 perspective 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:

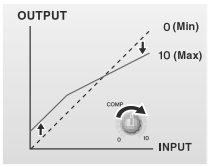
Phasing, Chorus, and Flanging

All of these effects work on basically the same principle: 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 reinforces the signal at others and this causes the shimmering sound we hear.

For chorus and flanging the signal is delayed by several 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 otherworldly 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 common 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 compression you'll be able to clearly hear whispered passages 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 MG compressor makes achieving great sound much easier: all you need to do is set a single "compression" control and all of the pertinent parameters are automatically adjusted for you.



Caution!

To prevent fire or shock hazard, do not expose the unit to rain or moisture.

Do not open the top cover (or the rear section), high voltage exist inside the unit dangerously, and no user serviceable parts inside.

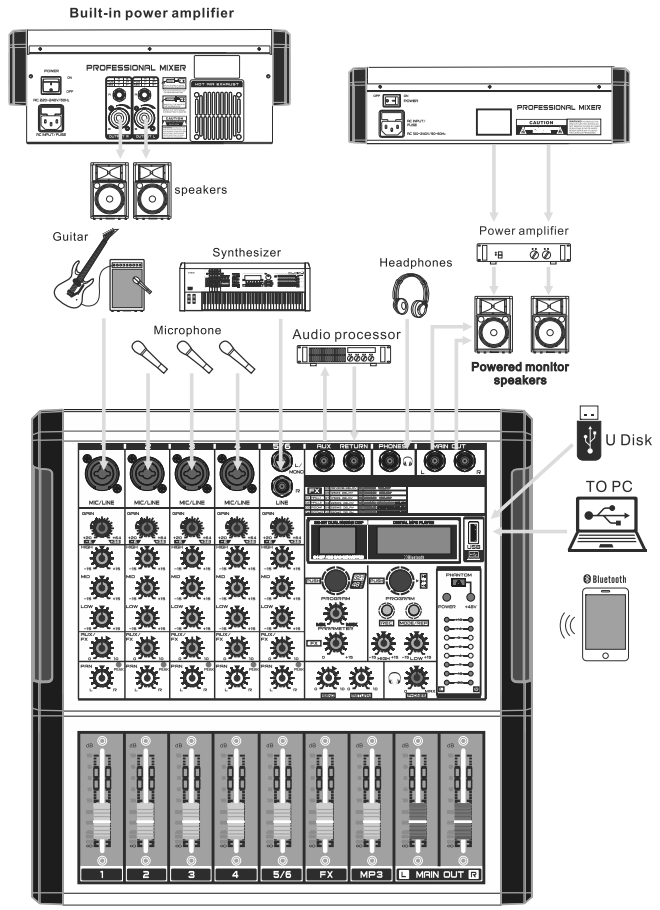
Refer servicing to qualified personnel.

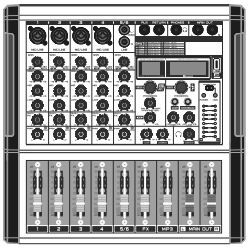
Precautions!

1.Do not use this apparatus near water, if any liquid or water fall into the cabinet, unplug the unit and have it checked by qualified personnel before operating it any further.
2.Clean only with dry cloth.
3.Do not block any ventilation openings.
4.Be sure that there is enough space around the unit for cooling purposes, do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
5.Operate only on designated power supply which is printing on the unit.
6.Unplug the unit from the wall outlet or set the Master switch to OFF if it is not to be used for several days.
7.To disconnect the cord, pull it out by the plug. Never pull the cord itself.
8.Please noted that all units is properly grounded, for your safety, you should never remove any gound connectors from electronic devices, or render them inoperative.

Contents

Connection diagram

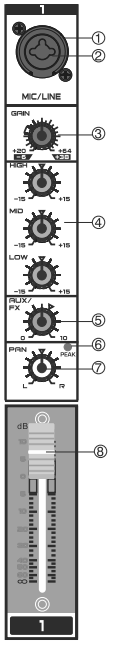




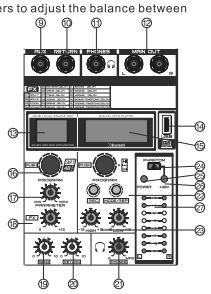
4/6/8/12/16 input channel mixer, new multi-voltage power supply for worldwide use

4/6/8 input channel,powered mixer

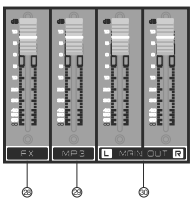
- ※*Built-in Bluetooth connect the mobile phone or other Bluetooth player*
- ※*Built-in MP3 player support a variety of formats music*
- ※*Connect the computer to record and play music*
- ※*Digital DSP,16 Multi-FX effects*
- ※*Ultra-musical 3-band EQ on all channels*
- ※*Peak LED all Channels*
- ※*High accurate level indicator*
- ※*Phantom power switch (+48V)*
- ※*Sealed rotary controls to resist dust and grime*
- ※*Rugged steel chassis*



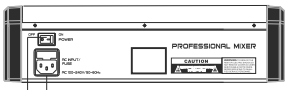
1.MIC Input jacks
These are balanced XLR-type microphone input jacks. (1 :Ground; 2: Hot; 3: Cold)
2.LINE Input Jacks (monaural channels)
These are balanced TRS phone-jack line inputs. (T: Hot; R:Cold; S: Ground). You can connect either balanced or unbalanced phone plugs to these jacks.
3.GAIN Control
Adjusts the input signal level. To get the best balance between the S/N ratio and the dynamic range, adjust the gain so that the PEAK indicator only occasionally and briefly on the highest input transients. The -60 to +10 scale is the MIC input adjustment range. The 40 to +10 scale is the LINE input adjustment range.
4.Equalizer 3(HIGH,MIDand LOW)
This three-band equalizer adjusts the channel's high,mid and low frequency bands. Setting the knob to the "0" position produces a flat response in the corresponding band. Turning the knob to the right boosts the corresponding frequency band, while turning to the left attenuates the band.
5.AUX/FX Control
The aux send marked FX offers a direct route to the built-in effects processor and is therefore post-fader and post-mute.
6.PEAK LED
The PEAK-LED lights up when the input signal is driven too high. If this happens, back off the TRIM control and, if necessary, check the setting of the channel EQ.
7.PAN Control
The PAN control determines the position of the channel signal within the stereo image. When working with subgroups, you can use the PAN control to assign the signal to just one output, which gives you additional flexibility in recording situations.
8.CHANNEL FADER
Adjusts the level If the channel signal.Use these faders to adjust the balance between the various channels.
9. AUX/SEND Jacks
The AUX SEND jack carries the master aux mix (from the channel's FX controls).
10.AUX/RETURN Jacks
These are unbalanced phone-jack type line inputs. These jacks are typically used to receive the signal returned from an external effect device(reverb, delay, etc.).
11.PHONES Jacks
Connect a pair of headphones to this TRS phone-type output jack.
12.MAIN OUT (L, R) Jacks
These jacks deliver the mixer's stereo output. You use these jacks, for example, to connect to power amplifier driving your main speakers.
13.EFFECTOR Display Show the kind of effector.
14.USB Jack
Used by U-Disk or computer sofeware for playing and recording.
15.MP3 Play Window
Show the Mp3 playing.time.song name and other play instruction.

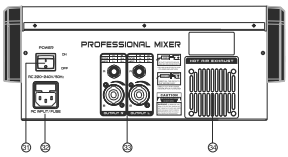


16.PROGRAM Control
You can select the effect preset by turning the PROGRAM control. The display flashes with the number of the current preset. To recall the selected preset, press on the button; the flashing stops. You can also recall the selected preset with the foot switch.
17.PARAMETER Control
Used to adjust the depth of the selected effect, speed, etc
18.FX Control
Used to adjust size effect.
19.SEND Control
Use this fader can control the MONO output jack.
20.RETURNS Control
Adjusts the level at which the signal received at the RETURN jacks (L(MONO)and R) is sent to the STEREO.
21.PHONES Control
Controls the level of the signal output to the PHONES jack OUT jacks.
22. MP3 control
a. Selected songs/Play/Pause: When playing music, rotate to change up/down the song, press to pause/play.
b. Recording: When playing music, long press to record, short press to finish recording and enter playing of the recording music.
When playing the recording music, short press to switch to play the USB music, play from the first USB music.
When playing the USB music, short press to switch to play the recording music, play from first recording music.
c. Mode/Repeat: Short press to switch the model of USB and BLUETOOTH, long press to repeat the playing song.
When playing the repeated song, long press to return to normal play.
23.EQ of MP3 player
The two-band equalizer adjusts the level of the two bands Mp3 player.
24. +48 V PHANTOM Power
This switch toggles phantom power on and off. When the switch is on the mixer supplies +48V phantom power to all channels that have XLR mic input jacks. Turn this switch on when using one or more phantom-powered condenser microphones.
25.POWER Indicator
This indicator lights up when the mixer's power is ON.
26.+48V Indicator
This indicator lights up when the +48V power is ON.
27. Level Meter
Show the level signal's strong
28.FX SEND Fader
Control effect input signal level.
29.MP3 VOL Fader
Change VOL button can be control the VOL of Mp3.
30.MAIN MIX Fader
You use the high-precision quality faders to control the output level of the main mix.

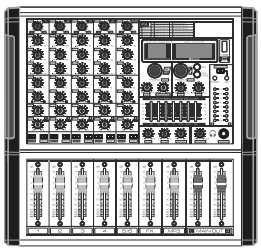


Built-in power amplifier





31: 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.
32: 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.
33. AMPLIFIER OUTPUT
Connect with two 4ohm speakers.
34: COOLING FAN
Cooling the amplifier to avoid the amplifier too hot to be broken.



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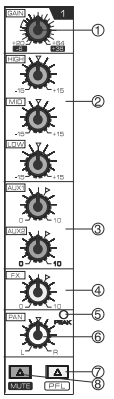

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3.AUX1/AUX2 Control
[AUX1] Control --- This control adjusts the level of the input sent to the AUX1 output (refer to numeral 36 in section MAIN TERMINALS on page 10).
[AUX2] Control --- This control adjusts the level of the input sent to the AUX2 output (refer to numeral 35 in section MAIN TERMINALS on page 10).
4.FX Control
Adjusts the level of the signal sent from the channel to the FX SEND buses.
5.PEAK LED
The PEAK-LED lights up when the input signal is driven too high. If this happens, back off the TRIM control and, if necessary, check the setting of the channel EQ.
6.PAN Control
The PAN control determines the position of the channel signal within the stereo image. When working with subgroups, you can use the PAN control to assign the signal to just one output, which gives you additional flexibility in recording situations.
7.PFL SWITCH
The PFL switch is used to route the channel signal to the PFL bus (Pre Fader Listen). This enables you to listen to a channel signal without affecting the main output signal. The signal you hear is taken either be fore the pan control (PFL, mono)
8.MUTE SWITCH
The MUTE switch breaks the signal path pre-channel fader, hence muting that channel in the main mix. The aux sends which are set to post-fader are likewise muted for that channel, while the pre-fader monitor paths remain active irrespective of whether the channel is muted or not.
9.CHANNEL FADER
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