PROFESSIONAL MIXER

Professional

MIXER

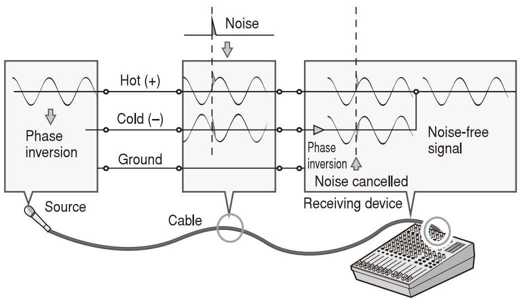
Instruction Manual

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

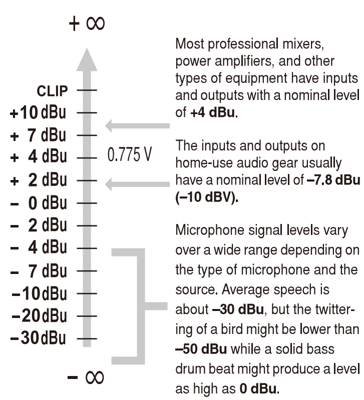
Microphones	Use balanced lines.
Short line-level runs	Unbalanced lines are fine if you're in a relatively noise-free environment.
Long line-level runs	The ambient electromagnetic noise level will be the ultimate deciding factor, but balanced is best.

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.

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

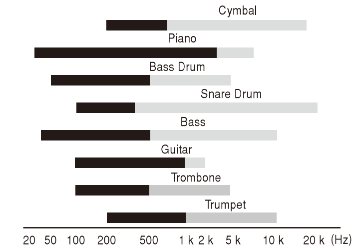
In general: less is better. There are many situations in which you'll need to cut certain frequency ranges, but use boost sparingly, and with caution. Proper use of EQ can eliminate interference between instruments in a mix and give the overall sound better definition. Bad EQ—and most commonly bad boost—just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

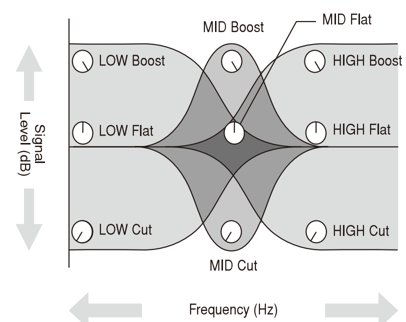
□ Harmonics: Multiples of the fundamental frequency that play a role in determining the timbre of the instrument.

> ***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.

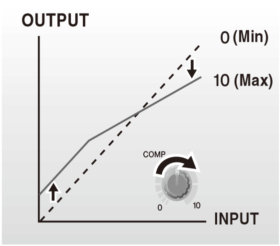
For chorus and flanging the signal is delayed by several milliseconds, with the delay time modulated by an LFO, and recombined with the direct signal. In addition to the phasing effect described above, the delay modulation causes a perceived pitch shift which, when mixed with the direct signal, results 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 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 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.

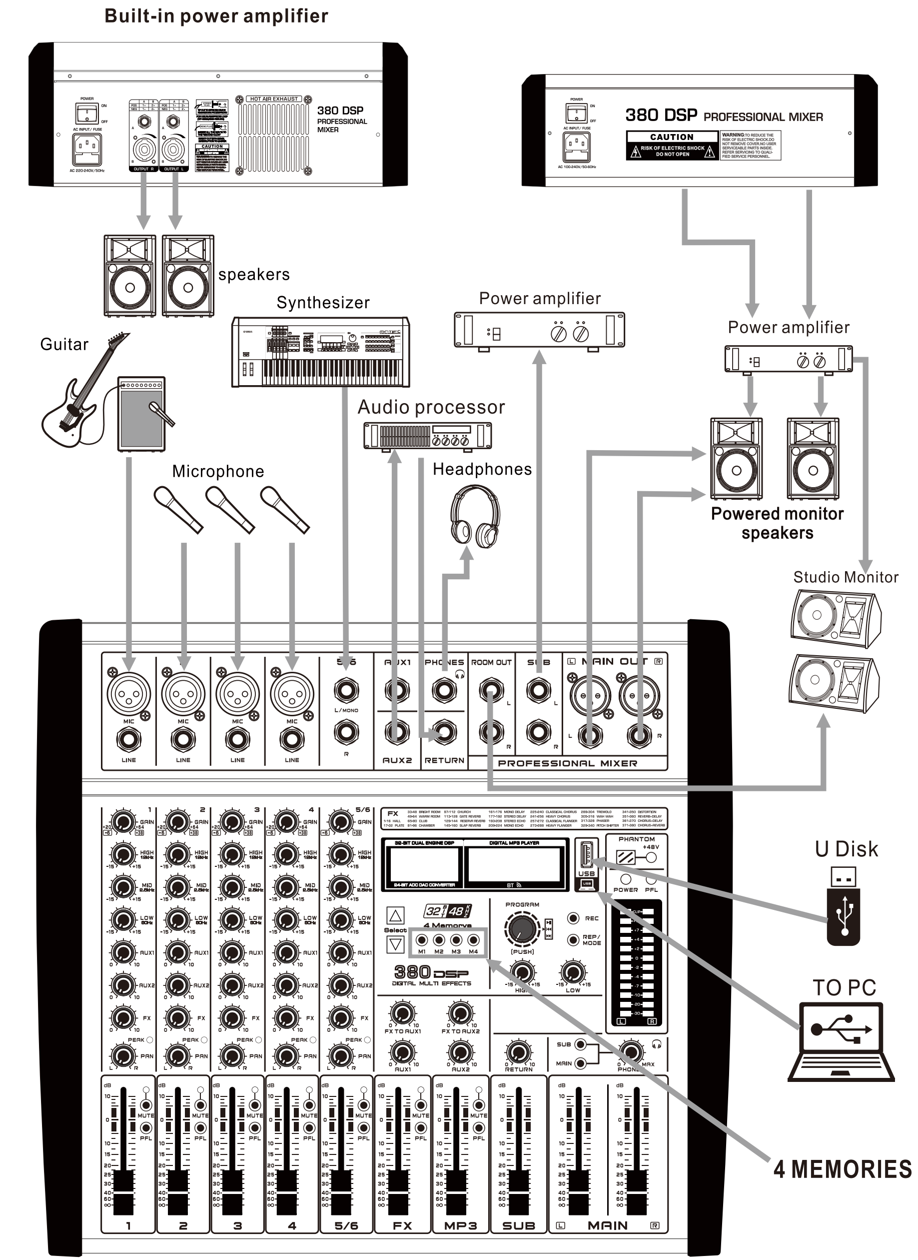
Refer servicing to qualified personnel.

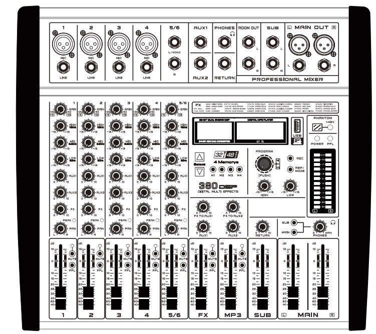
Precautions!

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

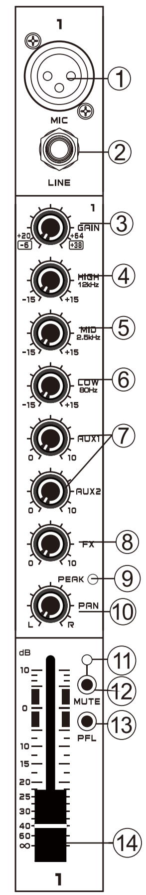
Connection diagram



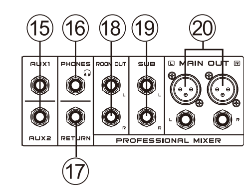


6/8/12/16 input channel mixer, new multi-voltage power supply for worldwide use
6/8/12/16 input channel,powered mixer

※ 6/8/12/16 channels mixer
※ Built-in Bluetooth,connect with mobilephone to other Bluetooth player
※ Built-in Mp3 player to play music in multiple formats and record
※ Connect USB to PC to play/record the stereo audio
※ Built-in 380 digital effects
※ Rugged steel chassis and exquisite panel on two sides
※ Built-in power amplifier,with complete protecting function,strong power



1. **[MIC]** Input , use this socket to connect the microphone of BALANCED XLR PLUG
2. **[LINE]** Input,connect the microphone,musical instrument or audio device (CD player and so on)to the device. The socket is supporting UNBALANCED 1/4"PLUG
3. **[GAIN]** Controller, determines the basic volume of each channel, adjust them for the maximum volume performance, when the volume gets too loud, the [PEAK]LED indicator light will keep flashing.
4. **[HIGH]** Treble tone controller, adjust the controller to enhance or attenuate for 15dB at 12KHz.
5. **[MID]** Alto tone controller, adjust the controller to enhance or attenuate for 15dB at 2.5KHz.
6. **[LOW]** Bass tone controller, adjust the controller to enhance or attenuate for 15dB at 80Hz.
7. **[AUX CONTROL]** Used to adjust the output to AUX pin signal level.
8. **[FX]** Adjusts the level of the signal sent from the channel to the FX SEND buses.
9. **[PEAK]** 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
10. **[PAN]** 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.
11. **[INDICATOR]** the leds are on,switch means the channel is MUTE.
12. **[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.
13. **[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 before the pan control (PFL,mono).
14. **[CHANNEL FADER]** Adjusts the level of the channel signal. Use these faders to adjust the balance between the various channels.
15. **[AUX/SEND JACKS]** The AUX SEND jack carries the master aux mix (from the channel's FX controls).
16. **[PHONES JACKS]** Connect a pair of headphones to this TRS
17. **[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.).
18. **[ROOM OUT]** This ROOM OUT can connect audio devices in the minitor room to minitor the audio effect
19. **[SUB]** Group output: "L" is hybrid binaural output, "R" is mono out put, when connecting to "L" and "R" together, it is stereo output
20. **[MAIN OUT]** Output, connect the active speaker or amplifier.



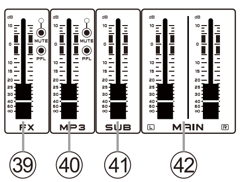
21. **[EFFECTOR DISPLAY]** The 380 effectors are shown.

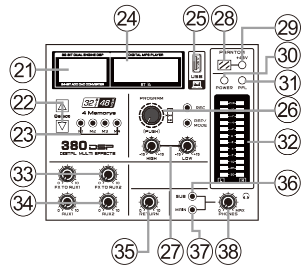
22. **[SELECT]** Press UP or DOWN button to select from the 380 DSP.

23. **[4 MEMORIES]** Press this button M1 , M2, M3, M4 to memory the DSP you selected last time.

24. **[MP3 PLAY WINDOW]** Show the Mp3 playing time song name and other play instruction.

25. **[USB INTERFACE]** Use for MP3 input or computer connection (can be used for U disk or computer software recording)





26. **[PROGRAM]** Control Knob --- Push down and turn this control knob to select one of the sixteen built-in sound effect modes.

27. **[HIGH•LOW]** The two-band equalizer adjusts the level of the two bands Mp3 player.

28. **[+48V 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.

29. **[+48v INDICATOR]** This indicator lights up when the +48V power is ON.

30. **[POWER INDICATOR]** This indicator lights up when the mixer's power is ON.

31. **[PFL]** After pressing the PFL button, the PFL indicator light will be up.

32. **[LEVEL METER]** Show the level signal's strong.

33. **[FX TO AUX]** Adjust the effect value to AUX 1 and AUX 2 output.

34. **[AUX1&AUX2]** Adjust the volume of AUX 1 and AUX 2 outputs.

35. **[RETURN]** Level controller, adjust the return level.

36. **[SUB SWITCH]** Press SUB to send the signal to the headphone jack output.

37. **[MAIN SWITCH]** Press MAIN to send the signal to the headphone jack output.

38. **[PHONES CONTROL]** Controls the level of the signal output to the PHONES jack OUT jacks.

39. **[FX SEND FADER]** Control effect input signal level.

40. **[MP3 VOL FADER]** Change VOL button can be control the VOL of MP3.

41. **[SUB FADER]** Control sub volume

42. **[MAIN MIX FADER]** You use the high-precision quality faders to control the output level of the main mix.

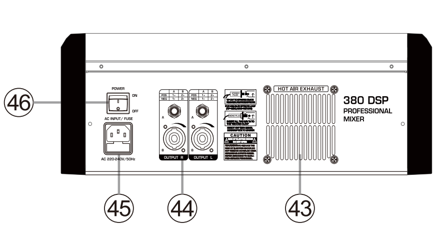
Built-in power amplifier

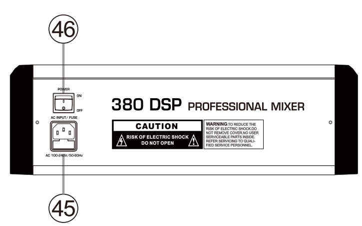
43. **[COOLING FAN]** Cooling the amplifier to avoid the amplifier too hot to be broken.

44. **[AMPLIFIER OUPUT]** Connect with two 4ohm speakers.

45. **[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 connecting is made via a cable with IEC mains connector.

46. **POWER SWITCH**





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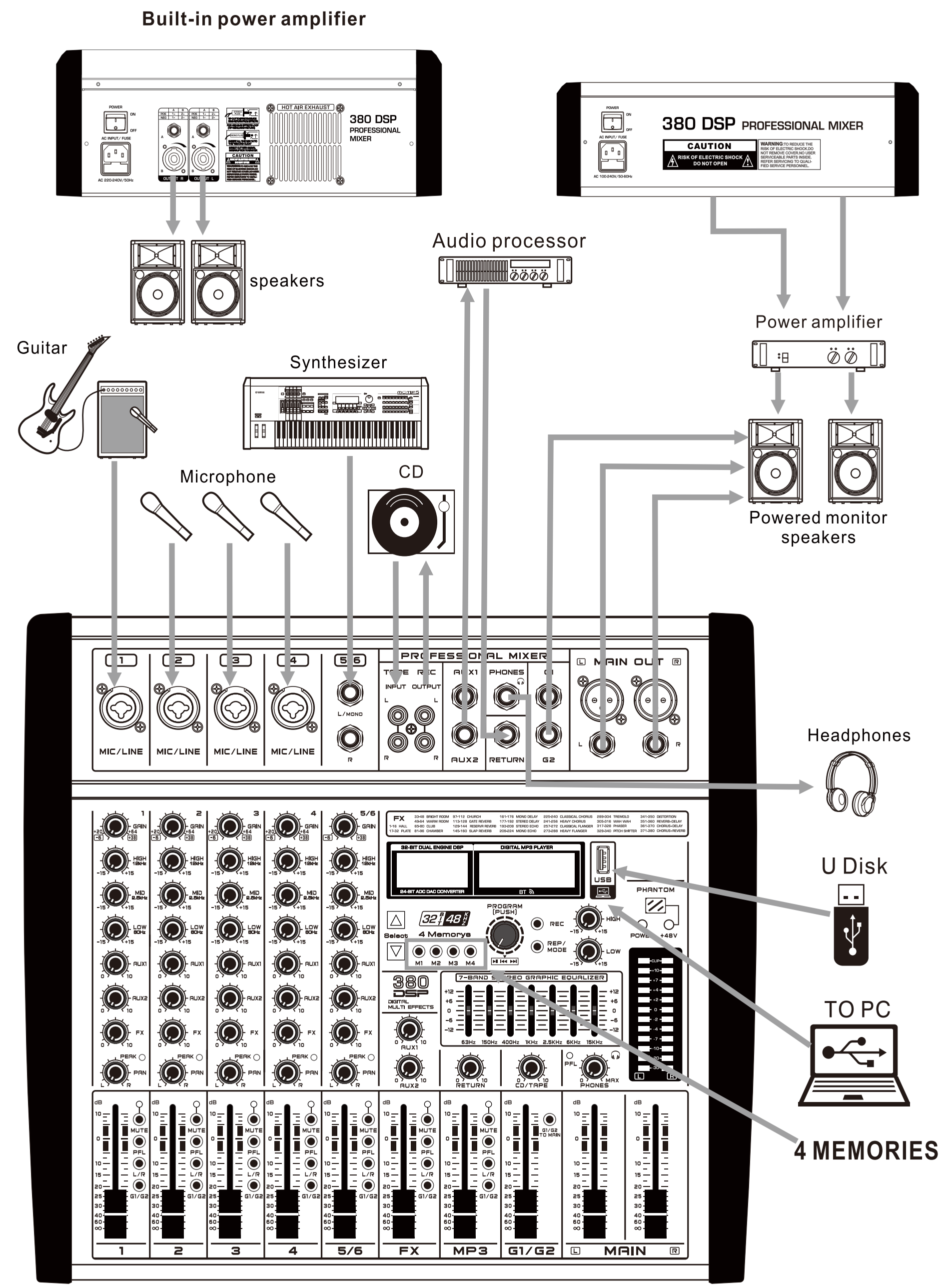

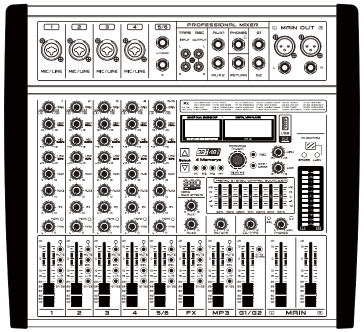

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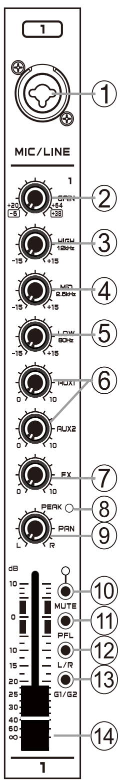

1. **[MI/ LINE]** Mono input, connect the microphone, musical instrument or audio device (CD player and so on) to the device. The sockets are supporting both XLR and UNBALANCED 1/4" plug.
2. **[GAIN]** Controller, determines the basic volume of each channel, adjust them for the maximum volume performance, when the volume gets too loud, the [PEAK]LED indicator light will keep flashing.
3. **[HIGH]** Treble tone controller, adjust the controller to enhance or attenuate for 15dB at 12KHz.
4. **[MID]** Alto tone controller, adjust the controller to enhance or attenuate for 15dB at 2.5KHz.
5. **[LOW]** Bass tone controller, adjust the controller to enhance or attenuate for 15dB at 80Hz.
6. **[AUX CONTROL]** Used to adjust the output to AUX pin signal level.
7. **[FX]** Adjusts the level of the signal sent from the channel to the FX SEND buses.
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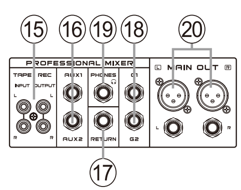
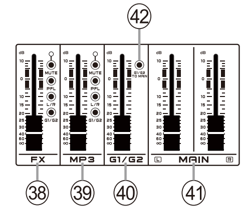
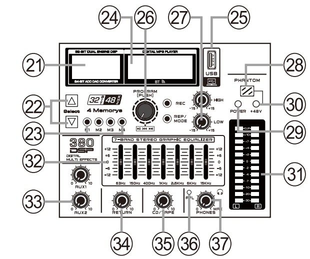
10. **[MUTE SWITCH]** The MUTE switch breaks the signal path pre-channel fader, hence muting that channel in the main mix. The aux sends which are set to post-fader are likewise muted for that channel, while the pre-fader monitor paths remain active irrespective of whether the channel is muted or not.
11. **[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).
12. **[L/R SWITCH]** Press the button to add the subchannel signal to the main output.
13. **[G1/G2 SWITCH]** Press this button to group this channel out to G1G2.
14. **[CHANNEL FADER]** Adjusts the level of the channel sıgnal. Use these faders to adjust the balance between the various channels.
15. **[CD / TAPE INPUT/OUTPUT SOCKET]** The TAPE IN jacks(on stereo RCA)allow for the connection of play-back devices such as CD players,etc. Use the TAPE OUT jacks to connect,for example,a tape deck for recording applications.
16. **[AUX/SEND JACKS]** The AUX SEND jack carries the master aux mix (from the channel's FX controls).
17. **[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.).
18. **[G1/G2SOCKER]** After marshalling each subchannel, signal output is performed from this point.
19. **[PHONES]** Socket, connect an earphone, the socket is supporting stereo phone plug.
20. **[MAIN OUT]** Output, connect the active speaker or amplifier.

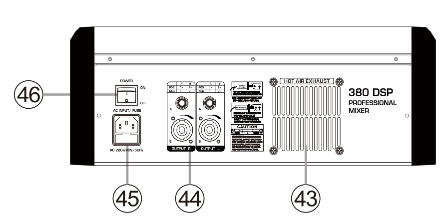


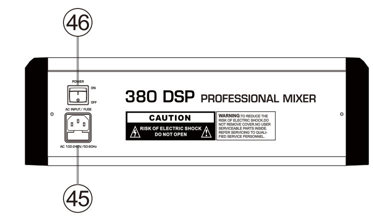
21. **[EFFECTOR DISPLAY]** The 380 effectors are shown.
22. **[SELECT]** Press UP or DOWN button to select from the 380 DSP.
23. **[4 MEMORIES]** Press this button M1 , M2, M3, M4 to memory the DSP you selected last time.





24. **[MP3 PLAY WINDOW]** Show the Mp3 playing time song name and other play instruction.
25. **[USB INTERFACE]** use for MP3 input or computer connection (can be used for U disk or computersoftware recording)
26. **[PROGRAM]** Control Knob --- Push down and turn this control knob to select one of the sixteen built-in sound effect modes.
27. **EQ OF MP3 PLAY** The two-band equalizer adjusts the level of the two bands Mp3 player.
28. **[+48V 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.
29. **[POWER INDICATOR]** This indicator lights up when the mixer's power is ON.
30. **[+48v INDICATOR]** This indicator lights up when the +48V power is ON.
31. **[LEVEL METER]** Show the level signal's strong.
32. **ST GRAPHIC EQUALIZER** This 7-band equalizer adjusts the sound of the signal send to The MAIN OUT jacks.
33. **[AUX1&AUX2]** Adjust the volume of AUX 1 and AUX 2 outputs.
34. **[RETURN]** Level controller, adjust the return level.
35. **[CD TAPE]** CD/TAPE signal input socket - This socket is used to adjust the CD/TAPE volume of effect.
36. **[PFL]** After pressing the PFL button, the PFL indicator light will be up.
37. **[PHONES CONTROL]** Controls the level of the signal output to the PHONES jack OUT jacks.
38. **[FX SEND FADER]** Control effect input signal level.
39. **[MP3 VOL FADER]** Change VOL button can be control the VOL of MP3.
40. **[G1/G2 FADER]** Group signal output volume control.
41. **[MAIN MIX FADER]** You use the high-precision quality faders to control the output level of the main mix.
42. **[G1/G2 TO MAIN]** Press this button to output the marshalling signal group to the main channel output





43. **[COOLING FAN]** Cooling the amplifier to avoid the amplifier too hot to be broken.
44. **[AMPLIFIER OUPUT]** Connect with two 4ohm speakers.
45. **[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 connecting is made via a cable with IEC mains connector.
46. **POWER SWITCH**

Specifications

		6-CH	8-CH	12-CH	16-CH
Frequency Response		20Hz-20KHz ±1dB+4dBµ@1KHz			
Total Harmonic Distortion		<0.5% @+4dBµ (22Hz-22KHz)			
Channel		6-CH	8-CH	12-CH	16-CH
Input Channels	MIC/LINE	4(XLR)+4(6.35)	6(XLR)+6(6.35)	10(XLR)+10(6.35)	14(XLR)+14(6.35)
	Stereo/LINE	1(5/6)	1(7/8)	1(11/12)	1(15/16)
	MP3	1	1	1	1
	USB Sound Card	2.0Type-A	2.0Type-A	2.0Type-A	2.0Type-A
	Bluetooth	1	1	1	1
Output Channels	Main/XLR	L/R	L/R	L/R	L/R
	Main/6.35Jack	L/R	L/R	L/R	L/R
	Phones/6.35Jack	1	1	1	1
	ROOM OUT	1(L/R)	1(L/R)	1(L/R)	1(L/R)
	SUB	L/R	L/R	L/R	L/R
Auxiliary	AUX 1(OUT)	1	1	1	1
	AUX 2(OUT)	1	1	1	1
	Return(IN)	1	1	1	1
Input Channel Function	MUTE/PFL	MUTE and PFL for each channel			
	EQ/3-band	HIGH : Gain/± 15dB、Frequency :12KHz shelving MID : Gain/± 15dB、Frequency :2.5KHz shelving LOW : Gain/± 15dB、Frequency : 80Hz shelving			
	Peak LED	LED turns on when post EQ signal reaches 3dB below clipping level			
Level Meter		2 × 12-segment LED meter , (CLIP,+10,+7,+4,+2,0,-2,-4,-7,-10,-20 ,-30dB)			
Effects/FX		380 Digital Effects (4 Memories)			
Phantom Power Voltage		+48V	+48V	+48V	+48V
Power Source		AC 100-240V (50Hz-60Hz)			
Power Consumption		20W	25W	30W	35W

Built-in power amplifier

Power Source	AC 100-120V or 220-240V (50Hz-60Hz)
Output Power	2×350W or 2×550W (4Ω)

		6-CH	8-CH	12-CH	16-CH
Frequency Response		20Hz-20KHz ±1dB+4dBµ@1KHz			
Total Harmonic Distortion		<0.5% @+4dBµ (22Hz-22KHz)			
Channel		6-CH	8-CH	12-CH	16-CH
Input Channels	MIC/LINE	4(XLR+6.35Jack)	6(XLR+6.35Jack)	10(XLR+6.35Jack)	14 (XLR+6.35Jack)
	Stereo/LINE	1(5/6)	1(7/8)	1(11/12)	1(15/16)
	MP3	1	1	1	1
	USB Sound Card	2.0Type-A	2.0Type-A	2.0Type-A	2.0Type-A
	Bluetooth	1	1	1	1
Output Channels	REC/RCA	1	1	1	1
	Main/XLR	L/R	L/R	L/R	L/R
	Main/6.35Jack	L/R	L/R	L/R	L/R
	Phones/6.35Jack	1	1	1	1
	G1/G2	2	2	2	2
Auxiliary	AUX 1(OUT)	1	1	1	1
	AUX 2(OUT)	1	1	1	1
	Return(IN)	1	1	1	1
Input Channel Function	MUTE/PFL L-R/G1G2	MUTE/PFL/L-R/G1G2 for each channel			
	EQ/3-band	HIGH : Gain/± 15dB、Frequency :12KHz shelving MID : Gain/± 15dB、Frequency :2.5KHz shelving LOW : Gain/± 15dB、Frequency : 80Hz shelving			
	Peak LED	LED turns on when post EQ signal reaches 3dB below clipping level			
Level Meter		2 × 12-segment LED meter , (CLIP,+10,+7,+4,+2,0,-2,-4,-7,-10,-20 ,-30dB)			
Main control equalizer		7-band stereo graphic equalizer			
Effects/FX		380 Digital Effects (4 Memories)			
Phantom Power Voltage		+48V	+48V	+48V	+48V
Power Source		AC 100-240V (50Hz-60Hz)			
Power Consumption		20W	25W	30W	35W

Built-in power amplifier

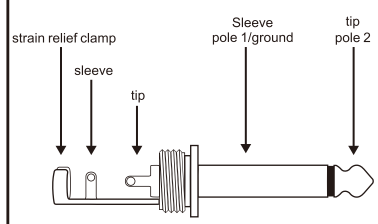
Power Source	AC 100-120V or 220-240V (50Hz-60Hz)
Output Power	2×350W or 2×550W (4Ω)

Installation

Cable connections

You will need a large number of cables for the various connections of the console. The illustrations below show the wiring of these cables. Be sure to use only high-grade cables.

1/4" TS footswitch connector



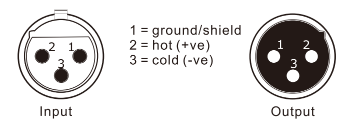
The footswitch connects both poles momentarily

Foot switch connector

Audio connections

Please use commercial RCA cables to wire the 2-track inputs and outputs.
You can, of course, also connect unbalanced devices to the balanced input/outputs. Use either mono plugs, or use stereo plugs to link the ring and shaft (or pins 1 & 3 in the case of XLR connectors).

Balanced use with XLR connectors

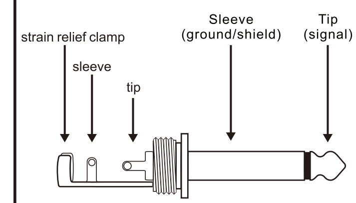


For unbalanced use, pin 1 and pin 3 have to be bridged

XLR connections

☞ Caution! You must never use unbalanced XLR . connectors (PIN 1 and 3 connected) at the MIC input jacks if you want to use the phantom power supply.

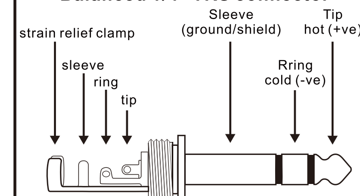
6.3: 1/4" mono plug



Unbalanced " TS connector

6.3: 1/4" mono plug

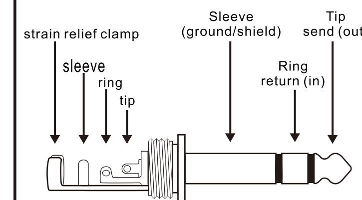
Balanced 1/4" TRS connector



For connection of balanced and unbalanced plugs, ring and sleeve have to be bridged at the stereo plug.

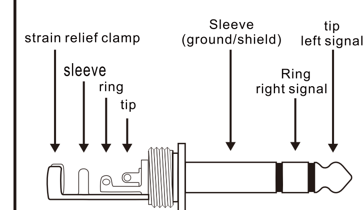
Balanced 1/4" TRS connector

Insert send return 1/4" TRS connector



Connect the insert send with the input and the insert return with the output of the effects device.

Insert send/return stereo plug



1/4" TRS headphones connector

Troubleshooting

Should problems occur, they are in many cases, due to simple operation mistakes or the like. On the basis of the following checks, you will be able to rectify a number of problems yourself without difficulty. If the problem cannot be remedied after the following checks, please consult with your dealer.

PROBLEM	POSSIBLE CAUSES	SOLUTIONS
Power can not be turned on	Power Supply cord was not connected or was not connected securely.	Securely connect the power supply cord to the mixer DC input and/or the AC power outlet.
	The power supply cord is defective.	Replace the power supply cord.
	The AC power outlet has no power.	Connect the power supply to an AC power outlet with proper power.
	The AC power source is from an AC power extension cord and the power switch of the extension cord is not turned on.	Turn on the power switch of the AC power extension cord.
No output sound	The power is off.	Turn on the power.
	The stereo level fader was turned to minimum.	Adjust the stereo level fader to have an optimal output level.
	The main output audio cable is missing or defective.	Connect, repair or replace the audio cables.
One channel no sound	The gain control knob to the channel was turned to minimum.	Adjust the gain control knob to that channel to have an optimal output level.
	The level control knob to the microphone channel was turned to minimum.	Adjust the level control knob to that channel to have an optimal output level.
Microphone no sound	No phantom power to the condenser microphone	Turn on the phantom power.
	The gain control knob to the microphone channel was turned to minimum.	Adjust the gain control knob to that microphone channel to have an optimal microphone output level.
	The level control knob to the microphone channel was turned to minimum.	Adjust the level control knob to that microphone channel to have an optimal microphone output level.
Distorted sound	The amplitude of the input signal is over the threshold.	Adjust the gain control knob to lower the input gain.
	The amplitude of the main output signal is over the threshold of the connected amplifiers or active speakers.	Adjust the stereo level fader to lower the main output level.