For more questions, please contact us at support@soundtown.com.

SOUND TOWN

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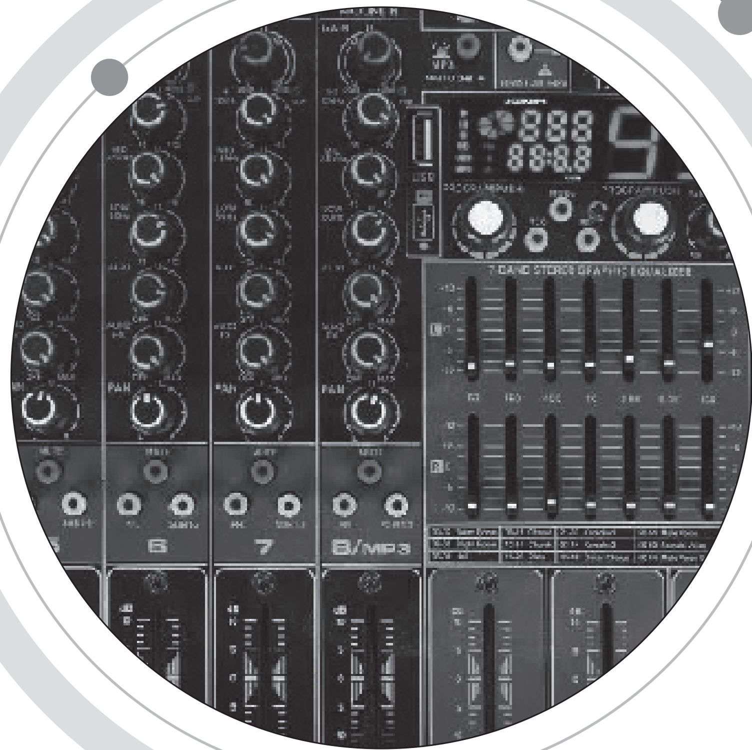


PRODUCT MANUAL

TRITON-TX802 / TRITON-TX1202

Professional Mixer with DSP

Important safety instructions

Caution !
To reduce the risk of electric shock, do not remove the top cover (or the rear section). No user serviceable parts inside. Refer servicing to qualified personnel

Caution !
To reduce the risk of fire or electric shock, do not expose this appliance to rain and moisture. The apparatus shall not be exposed to dripping or splashing liquids and no objects filled with liquids, such as vases, shall be placed on the apparatus.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Please read the manual.

This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure voltage that may be sufficient to constitute a risk of shock.

1. Keep these instructions.
2. Heed all warnings.
3. Follow all instructions.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Place the power cord so that it is protected from being walked on and sharp edges. Be sure that the power cord is protected particularly at plugs, convenience receptacles and the point where it exits from the apparatus.
11. The apparatus shall be connected to a MAINS socket outlet with a protective earthing connection.
12. Where the MAINS plug or an appliance coupler is used as the disconnect device, the disconnect device shall remain readily operable.

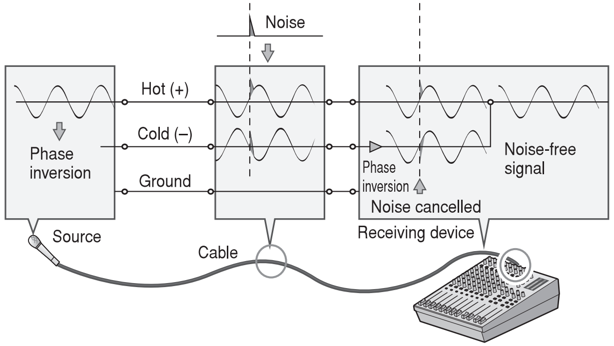
13. Only use attachments/accessories specified by the manufacturer.
14. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart / apparatus combination to avoid injury from tip-over.
15. Unplug this apparatus during lightning storms or when unused for long periods of time.
16. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

You've got yourself a mixer and now you're ready to use it.
Just plug everything in, twiddle the controls, and away you go ... right?
Well, if you've done this before you won't have any problems, but if this is the first time you've ever used a mixer you might want to read through this little tutorial and pick up a few basics that will help you get better performance and make better mixes.

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

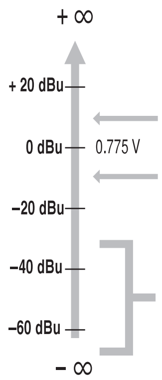


To summarize

Microphones	Use balanced cables
Short line-level runs	Unbalanced cables are fine if you're in 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.



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**.

To EQ or Not to EQ

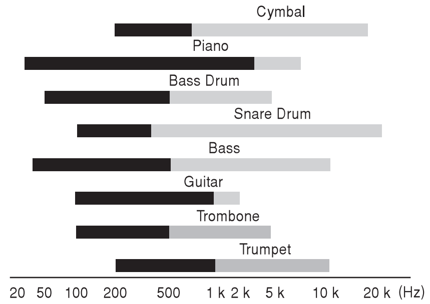
In general: less is better. There are many situations in which you'll need to cut certain frequency ranges, but use boost sparingly, and with caution. Proper use of EQ can eliminate interference between instruments in a mix and give the overall sound better definition. Bad EQ-and most commonly bad boost-just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

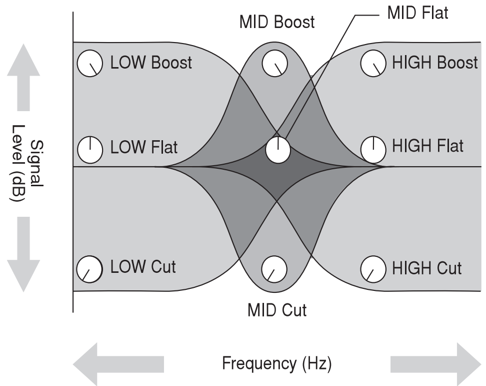
The lowest and highest frequencies than can be heard by the human ear are generally considered to be around 20 Hz and 20,000 Hz, respectively. Average conversation occurs in the range from about 300 Hz ta about 3,000 Hz. The frequency of a standard pitchfork used to tune guitars and other instruments 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 mixer's 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.

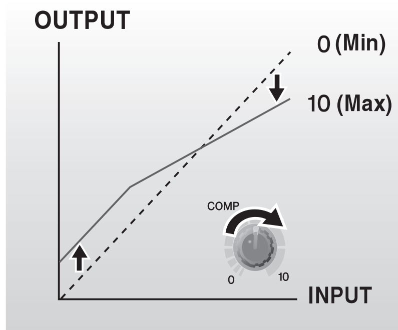
For chorus and flanging the signal is delayed by several milliseconds, with the delay time modulated by an LFO, and recombined with the direct signal. In addition to the phasing effect described above, the delay modulation causes a perceived pitch shift which, when mixed with the direct signal, results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

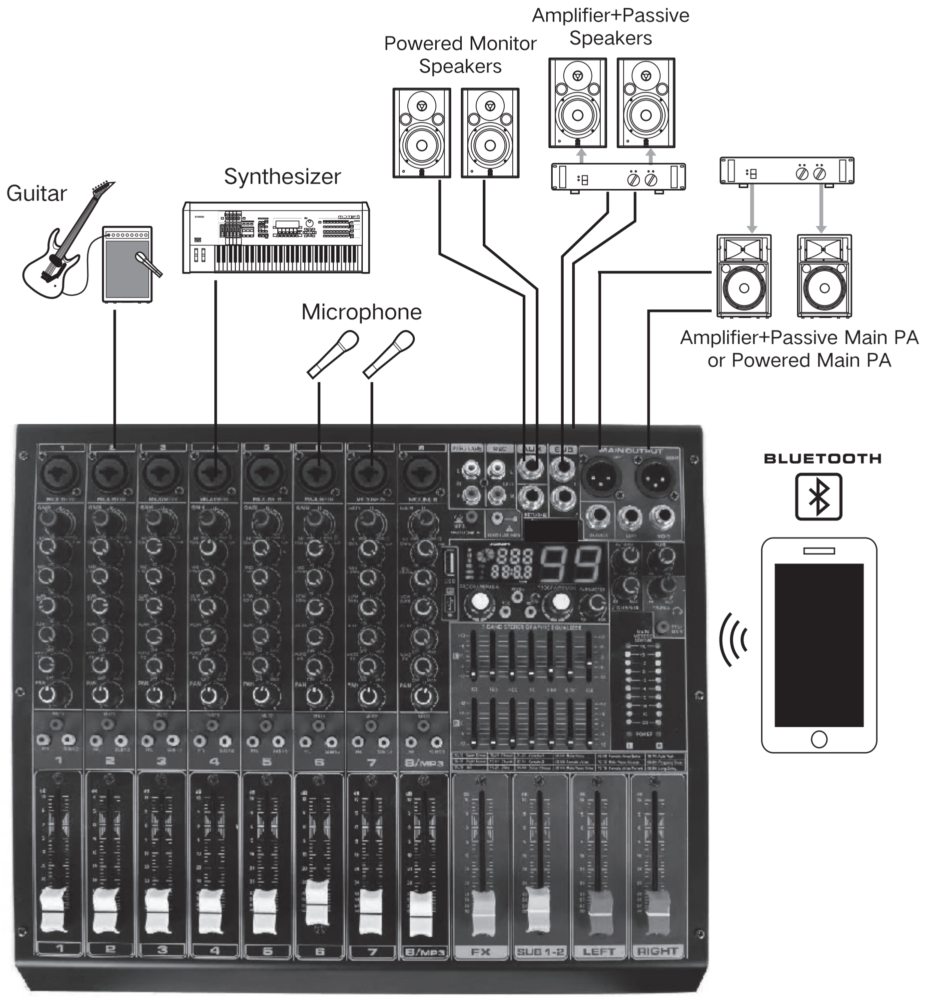
Compression

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.

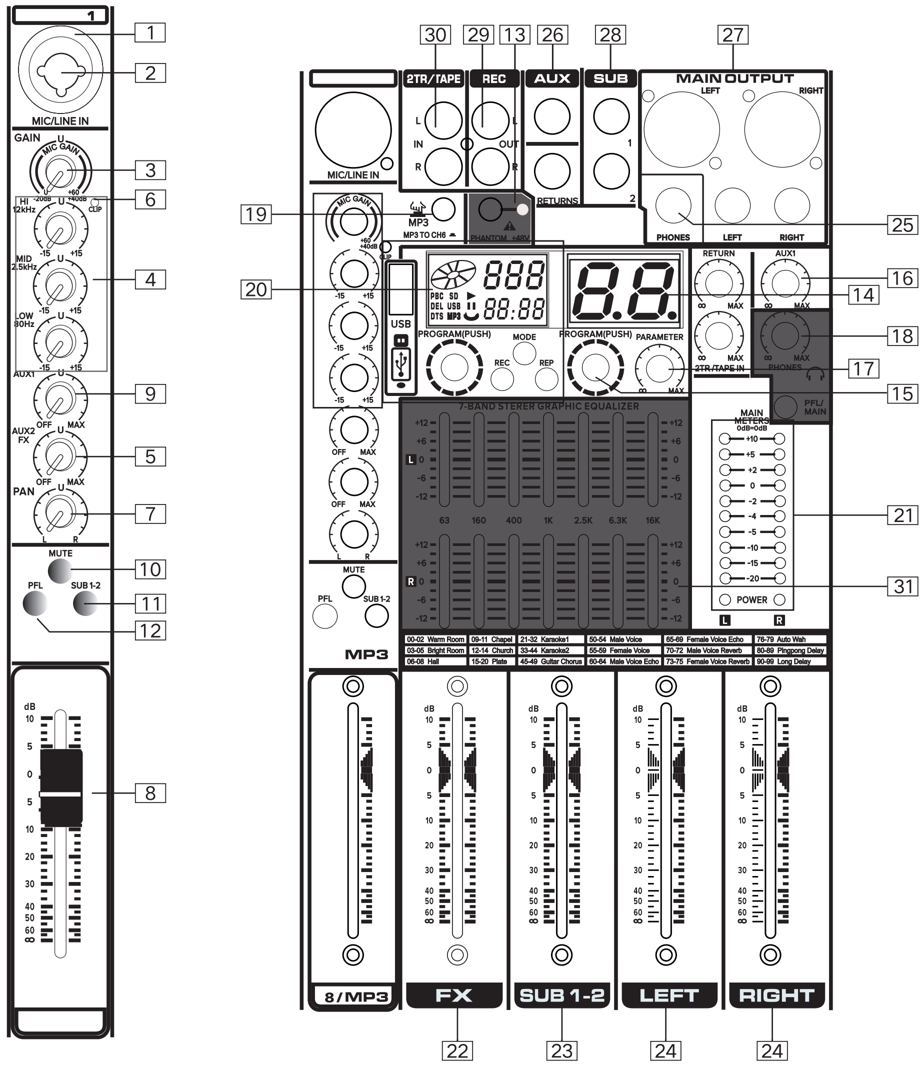
Most compressors require several critical parameters to be set properly to achieve the desired sound. The MG compressor makes achieving great sound much easier: all you need to do is set a single "compression" control and all of the pertinent parameters are automatically adjusted for you.



Connection



Control Elements & Connectors



Control Elements & Connectors

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 **TRIM 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 9 lights 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 (HIGH, MID and 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 **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 **CLIP 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. For example, when routing to subgroups 3 and 4, panning hard left will route the signal to group output 3 only, and panning hard right will route to group output 4 only.

8 **CHANNEL FADER**
Adjusts the level of the channel signal. Use these faders to adjust the balance between the various channels.

9 **AUX Control**
Monitor and effects buses (AUX sends) source their signals via a control from one or more channels and sum these signals to a so-called bus. This bus signal is sent to an aux send connector (for monitoring applications: MON OUT) and then routed, for example, to an active monitor speaker or external effects device. In the latter case, the effects return can then be brought back into the console via the aux return connectors. All monitor and effects busses are mono, are tapped into post EQ and offer amplification of up to +15 dB.

10 **MUTE Switch**
To silences or deactivates the corresponding channel on the console. This means that no sound from the muted channel will be sent to the master mix or the sound system.

11 **SUB (1to2) Switch**
This switch aligns the channel to a subgroup.

12 **PFL (SOLO) Switch**
This switch is used to route the channel signal to the solo bus (Solo In Place) or 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) or after the pan and channel fader.

13 **PHANTOM +48V Switch**
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. The LED lights up when phantom power is switched on.

14 **EFFECTOR DISPLAY**
Show the type of effector.

15 **EFFECTOR PROGRAM Dial**
You can select the effect preset by turning the PROGRAM control. The display flashes with the number of the current preset. To confirm the selected preset, press on the button; the flashing stops.

16 **AUX / RETURN faders**
Use these two faders to control the AUX / RETURN output jacks. 26

17 **PARAMETER**
Adjusts the parameter (depth, speed, etc.) for the selected effect. The last value used with each effect type is saved.

18 **PHONES/CTRL ROOM ONLY Control**
Use this control to adjust the control room output level and the headphones volume.

19 **MP3 SWITCH**
Press to switch the MP3 input signal to channel 8.

20 **MP3 player**
The Mp3 player support USB flash drive and bluetooth connection.
(Please do not charge from the USB port!)

Mp3 LCD display
Indicate the Mp3 status, time, song name and other play instruction.

MODE switch
Press to interchange between USB or bluetooth mode.

PROGRAM Dial
You can select the Mp3 setting by turning and pressing the PROGRAM control. The display flashes with the status of the current setting, press to confirm the selected setting and the flashing stops.

REC
Press to start recording and press again to stop recording.

REP
Press to interchange between sequential playback or loop playback.

BLUETOOTH CONNECTION
Press the MODE button and swtich to bluetooth mode, the console will become discoverable as 'TRITON'. In the settings of your bluetooth enabled devices, set it to "discover". From the Bluetooth device list, select the 'TRITON' device then the bluetooth will be connected.

21 **Level Meter**
POWER Indicator: This indicator lights when the mixer power is ON.
Signal level LED: The "0dB" segment corresponds to the nominal output level. The "+10dB" indicator lights up when the output reaches the clipping level, and the fader should be adjusted to reduce output.

22 **FX Fader**
To adjust the 'effect' input signal level.

23 **SUB 1-2 Fader**
To adjust the level of subgroup channel.

24 **MAIN MIX FADER**
Use the high-precision faders to control the overall output level of the main mix.

25 **PHONES Jack**
Connect a pair of headphones to this TRS phone-type output jack.

26 **AUX / RETURN Jacks**
Two 1/4" jacks are provided generally serve as the return for the effects mix (created using the post-fader aux sends) by connecting the output of an external effects device. If only the left jack is connected, the AUX / RETURN is automatically switched to mono.

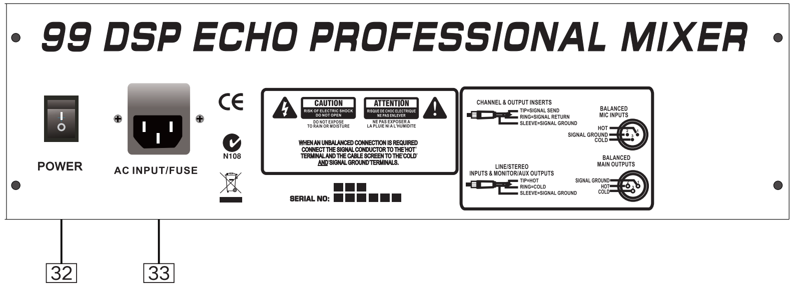
27 **MAIN OUTPUT**
Two balanced XLR sockets and two unbalanced 1/4" jacks are provided for main mix output (left & rihgt).

28 **SUB (1to2) Jacks**
Two unbalanced 1/4" jacks are provided for subgroup channels output.

29 **REC OUT (L, R) Jacks**
Two RCA jacks are provided to connect to an external recorder such as an MD recorder in order to record the same signal that is being output via the STEREO OUT jacks.

30 **2TR/TAPE**
Two RCA jacks are provided to input a stereo sound source. Use these jacks when you want to connect a CD player directly to the mixer.

31 **DUAL 7-BAND STEREO GRAPHIC EQUALIZER**
The graphic stereo equalizer allows you to adjust the volume of specific frequency ranges within an audio signal, which helps to shape the tonal balance of signal and correct for room acoustics.



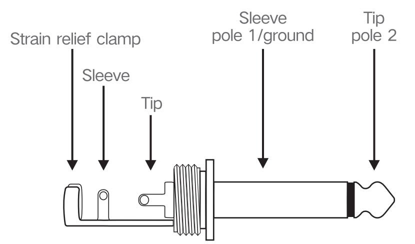
32 **POWER Switch**
Use the POWER switch to turn on the mixing console. The POWER switch should always be in the "Off" position when you are about to connect your unit to the mains. To disconnect the unit from the mains, pull out the main cord plug. When installing the product, ensure that the plug is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

33 **IEC MAINS RECEPTACLE / FUSE HOLDER (110V~230V, 50-60Hz)**
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.

WIRING

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



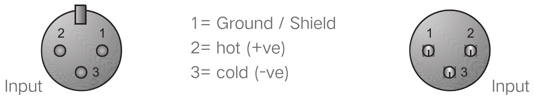
The footswitch connects both poles momentarily

WIRING

Audio connections

Please use commercial RCA cables to wire the 2-TR 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).

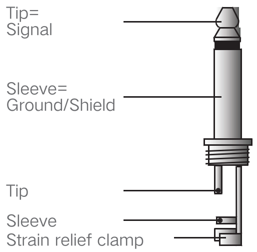
Balanced use with XLR connectors



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

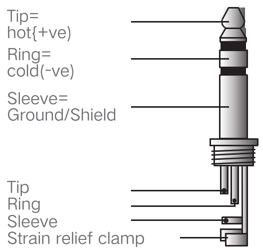
Caution! You must never use unbalanced XLR. connectors (PIN 1 and 3 connected) at the MIC input jacks if you want to use the phantom power supply.

1/4" mono plugs



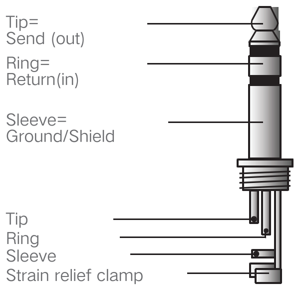
Unbalanced 1/4" TS connector

Balanced 1/4" TRS connectors



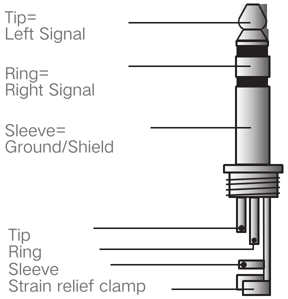
For connection of balanced and unbalanced plugs, ring and sleeve have to be bridged at the stereo plug.

Insert send/return 1/4" TRS connectors



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

1/4" TRS headphone connectors



SPECIFICATIONS

MICROPHONE INPUTS (XENYX MIC PREAMP)

Type	XLR, electronically balanced, discrete input circuit
Mic E.I.N. (20 Hz - 20 kHz)	
@ 0 Ω source resistance	-134 dB / 135.7 dB A-weighted
@ 50 Ω source resistance	-131 dB / 133.3 dB A-weighted
@ 150 Ω source resistance	-129 dB / 130.5 dB A-weighted
Frequency response	<10 Hz - 150 kHz (-1 dB), <10 Hz - 200 kHz (-3 dB)
Gain range	+10 to +60 dB
Max. input level	+12 dBu @ +10 dB Gain
Impedance	approx. 2.6 kΩ balanced
Signal-to-noise ratio	110 dB / 112 dB A-weighted (0 dBu In @ +22 dB gain)
Distortion (THD+N)	0.005% / 0.004% A-weighted
Line input	
Type	¼" TRS connector electronically balanced
Impedance	approx. 20 kΩ balanced
	10 kΩ unbalanced
Gain range	-10 to +40 dB
Max. input level	30 dBu

FADE-OUT ATTENUATION[1] CROSSTALK ATTENUATION)

Main fader closed	90 dB
Channel muted	89 dB
Channel fader closed	89 dB

FREQUENCY RESPONSE

Microphone input to main out	
<10 Hz - 90 kHz	+0 dB / -1 dB
<10 Hz - 160 kHz	+0 dB / -3 dB
Stereo inputs	
Type	¼" TRS connector, electronically balanced
Impedance	approx. 20 kΩ
Max. input level	+22 dBu
EQ mono channels	
Low	80 Hz / ±15 dB
Mid	100 Hz - 8 kHz/ ±15 dB
High	12 kHz/ ±15 dB
EQ stereo channels	
Low	80 Hz / ±15 dB
Low Mid	500 Hz / ±15 dB
High Mid	3 kHz / ±15 dB
High	12 kHz / ±15 dB
Aux sends	
Type	¼" TS connector, unbalanced
Impedance	approx. 120 Ω
Max. output level	+22 dBu
Stereo aux returns	
Type	¼" TRS connector,
	electronically balanced
Impedance	approx. 20 kΩ bal. / 10 kΩ unbal.
Max. input level	+22 dBu
Main outputs	
Type	XLR, electronically balanced and ¼" TRS balanced
1622FX only:	¼" TS connector unbalanced
Impedance	approx. 240 Ω symm. / 120 Ω unbalanced
Max. output level	+28 dBu
	+22 dBu
Control room outputs	
Type	¼" TS connector unbalanced
Impedance	approx. 120 Ω
Max. output level	+22 dBu
Headphones outputs	
Type	¼" TRS connector, unbalanced
Max. output level	+19 dBu / 150 Ω (+25 dBm)

DSP

Converter	24-bit Sigma-Delta, 64/128-times oversampling
Sampling rate	40 kHz

MAIN MIX SYSTEM DATA[2]

Noise	
Main mix @ -∞, Channel fader @ -∞	-101 dB -100 dB
Main mix @ 0 dB, Channel fader @ -∞	-93 dB -96 dB -87 dB
Main mix @ 0 dB, Channel fader @ 0 dB	-81 dB -83 dB -80 dB
Power supply	
Mains voltage	100 V~230 V , 50/60 Hz
Fuse	230 V :T 5 A H 250 V
Mains connection	Standard IEC receptacle

Measuring conditions:

1: 1 kHz rel. to 0 dBu; 20 Hz – 20 kHz; line input; main output; unity gain.

2: 20 Hz – 20kHz; measured at main output. Channels 1 – 4 unity gain; EQ flat; all channels on main mix; channels 1/3 as far left as possible, channels 2/4 as far right as possible. Reference = +6 dBu.