SWAMP
INDUSTRIES

S18 Manual

LIVE MIXER WITH DSP EFFECTS + USB

1. Safety Instructions

1. Read these instructions.

2. Keep these instructions.

3. Heed all warnings.

4. Follow all instructions.

5. Do not use this item near water.

6. Clean only with dry cloth.

7. Do not block any of the 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 items (including amplifiers) that produce heat.

9. Do not defeat the safety purpose of the polarized plug. The wide blade is provided for your safety. If the provided plug does not fit into the item or the mains socket, consult an electrician for replacement.

10. Protect the power cord from being walked on or pinched particularly at plug, convenience receptacles, and point where they exit from the item.

11. Only use attachments/accessories specified by the manufacturer.

12. Use only with a cart, stand, tripod, bracket or table specified by the manufacturer, or sold with the item. When a cart is used, use caution when moving the cart/item c ombination to avoid injury.

13. Unplug this item during lighting storms or when unused for long periods of time.

14. Refer all servicing to qualified service personnel. Servicing is required when the item 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 item, the item has been exposed to rain or moisture, does not operate normally, or has been dropped (note: accidental or cosmetic damage is not covered by the items 12 month warranty)

15. Please keep the unit in a safe environment.

16. Do not store anything on top of the item.

CAUTION
RISK OF ELECTRIC SHOCK

To reduce the risk of electric shock, do not remove any cover. No user-serviceable parts inside. Refer servicing to qualified personnel only.

The lightning flash with arrowhead symbol within the equilateral triangle is intended to alert the user to the presence of un-insulated “dangerous voltage” within the product’s enclosure that may be of sufficient magnitude to constitute a risk of electric shock.

The exclamation mark within the equilateral triangle is intended to alert the user to the presence of important operation and maintenance (servicing) instructions in the literature accompanying this appliance.

2. Control Elements and Connectors

This chapter describes the various control elements of your mixing console. All controls, switches and connectors will be discussed in detail.

2.1 Mono channels

2.1.1 Microphone and line inputs

Fig. 2.1: Connectors and controls of mic/line inputs

MIC

Each mono input channel offers a balanced microphone input via the XLR connector and also features switchable +48 V phantom power supply for condenser microphones. The studio grade preamps provide undistorted and noise-free gain as is typically known only from costly outboard preamps.

◊ **Please mute your monitor system before you switch on phantom power. Otherwise potentially damaging thumps will be sent to your speakers. Please also note the instructions in chapter 5.5 "Voltage supply, phantom power and fuse".**

LINE IN

Each mono input also has a balanced line input on a ¼" jack. You can also connect unbalanced devices using mono jacks to these inputs.

◊ **Please remember that you can use either the microphone input or the line input of a channel, but not both at the same time!**

INSERT

◊ **Insert points enable the processing of a signal with dynamic processors or equalizers. They are sourced pre-fader, pre-EQ and pre-aux send. Detailed information on using insert points can be found in chapter 5.3.**

◊ **Insert points are located on the rear of the console**

GAIN

Use the **GAIN** control to adjust the input gain. This control should always be turned fully counter-clockwise whenever you connect or disconnect a signal source to one of the inputs.

While the GAIN control is turned all the way down, connect your equipment. Set the GAIN control to the external devices' standard output level. If that unit has an output signal level display, it should show 0 dB during signal peaks.

Fine-tuning of a signal being fed in is done using the level meter. To route the channel signal to the level meter, you have to press the SOLO switch and set the SOLO MODE switch in the main section to PFL (LEVEL SET).

Using the GAIN control, drive the signal to the 0-dB mark. This way you have a vast amount of drive headroom for use with very dynamic signals. The CLIP display should light up only rarely, preferably never. While fine-tuning, the equalizer should be set to neutral.

LOW CUT

Additionally, the mono channels of the mixing consoles have a high-slope **LOW CUT** filter for eliminating unwanted, low-frequency signal components (75 Hz, 18 dB/octave).

COMPRESSOR

Each mono channel features a built-in compressor which lowers the dynamic range of the signal and increases its perceived loudness. The loud peaks are squashed down and the quiet sections are boosted.

Turn the COMP knob clockwise to add more compression effect. The adjacent LED with light when the effect is engaged.

2.1.2 Equalizer

All mono input channels have a 3-band equalizer with semi-parametric mid bands. All bands provide boost or cut of up to15 dB. In the central position, the equalizer is off (flat).

The circuitry of the EQs is based on the technology used in the best-known top-of-the-line consoles and providing a warm sound without any unwanted side effects. The result are extremely musical equalizers which, unlike simple equalizers, cause no side effects such as phase shifting or bandwidth limitation, even with extreme gain settings of ±15 dB.

Fig. 2.2: Equalizer of the input channels

The upper (HIGH) and the lower (LOW) bands are shelving filters that increase or decrease all frequencies above or below their cut-off frequency. The cut-off frequencies of the upper and lower bands are 12 kHz and 80 Hz respectively. For the mid range, the console features a semi-parametric equalizer with a filter quality (Q) of 1 octave, tunable from 100 Hz to 8 kHz. Use the MID control to set the amount of boost or cut, and the FREQ control to determine the central frequency.

2.1.3 Monitor and effects busses (Aux sends)

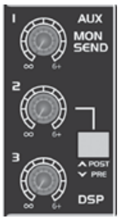


Fig. 2.3: Aux Send control MON and FX in the channel strips

Monitor and effects busses (AUX sends) source their signals via a control from one or more channels and sum these signals to a 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.

Pre-fader/post-fader

When using effects on a channel signal, it is usual to have the aux send post fader so that the balance between effect and dry signal stays constant even when the channel fader is altered. If this were not the case, the effects signal of the channel would remain audible even when the channel fader is turned all the way down. For monitoring, the aux sends are generally pre-fader, i.e. they operate independently of the position of the channel fader.

PRE

When the **PRE** switch is pressed down, the associated aux send is taken pre-fader.

DSP

The aux send marked **DSP** offers a direct route to the built-in effects processor and is therefore post-fader and post-mute. Please refer to chapter 4 "DIGITAL EFFECTS PROCESSOR" for detailed information.

◊ **If you are using the built-in effects processor, make sure that STEREO AUX RETURN 3 has nothing plugged into it otherwise the internal effects return will be muted. This is not relevant if you use the FX OUT jack to drive an external effects device.**

2.1.4 Routing switch, PAN, SOLO and channel fader

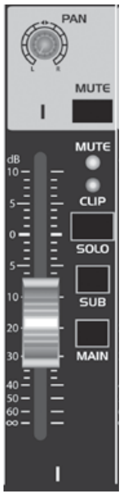


Fig. 2.4: The panorama and routing controls and the channel fader

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.

MUTE

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.

MUTE LED

The **MUTE LED** indicates a muted channel.

CLIP-LED

The **CLIP-LED** lights up when the input signal is driven too high. If this happens, back off the GAIN control and, if necessary, check the setting of the channel EQ.

SOLO

The **SOLO** 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 (Solo, stereo) (cf. chap. 2.3.10 "Level meters and monitoring").

SUB

The **SUB** switch routes the signal to the corresponding subgroups.

MAIN

The **MAIN** switch routes the signal to the main mix bus.

The channel fader determines the channel's volume in the main mix (or submix).

2.2 Stereo channels

2.2.1 Channel inputs

Fig. 2.5: The various stereo channel inputs

Each stereo channel has two balanced line level inputs on jacks for left and right channels. If only the left jack (marked "L") is used, the channel operates in mono. The stereo channels are designed to handle typical line level signals, and, depending on model, have a level switch (+4 dBu or -10 dBV) and/or a line GAIN control. Both jack inputs will also accept unbalanced connectors.

LEVEL

For level matching, the stereo inputs on the M16DSP have a **LEVEL** switch to select between +4 dBu and -10 dBV. At -10 dBV (homerecording level), the input is more sensitive than at +4 dBu (studio level).

2.2.2 Equalizer stereo channels

The stereo channels contain a stereo EQ section. The cut-off frequencies of the high and low bands are 12 kHz and 80 Hz respectively, while the center frequencies of the high-mid and low-mid bands are 3 kHz and 500 Hz respectively. The HIGH and LOW controls have the same characteristics as the EQ in the mono channels. Both mid range bands are of the peak filter type. A stereo EQ is superior to two mono EQs on a stereo signal as two separate EQs will usually result in a discrepancy between left and right channels.

2.2.3 Aux sends stereo channels

In principle, the aux sends of the stereo channels function the same way as those of the mono channels. As the aux sends are mono, the send from a stereo channel is first summed to mono before it reaches the aux bus.

2.2.4 Routing switch, solo and channel fader

Fig. 2.6: Balance control and mute switch

BAL

The **BAL** (BALANCE) control has a similar function to the PAN control in the mono channels.

The balance control determines the levels of the left and right input signals relative to each other before both signals are routed to the left/right main mix bus (or odd/even subgroup).

The remaining control elements in the stereo channels perform the same functions as their counterparts in the mono channels (MUTE switch, MUTE and CLIP LEDs, SOLO switch, SUB and MAIN switches and channel fader).

2.3 Interface panel and main section

Where it was useful to trace the signal flow from top to bottom in order to gain an understanding of the channel strips, we now look at the mixing console from left to right. The signals are, so to speak, collected from the same point on each of the channel strips and then routed to the main section all together.

2.3.1 MON control, aux sends 1, 2 and 3 (FX)

Turning up the AUX 1 control in a channel routes the signal to the aux send bus 1.

AUX SEND 1 and 2

The **AUX SEND 1** control governs the master send level of the mix created by the individual channel AUX 1 sends.

Likewise, the **AUX SEND 2** contol is the master control for the aux 2 bus.

Fig. 2.7: The AUX SEND controls of the main section.

AUX SEND 3 (DSP)

The **DSP** control determines the signal level for effects processing, i.e. regulates the level to an external (or the internal) effects device.

SOLO

You can use the SOLO switch to separately monitor the aux sends via the CONTROL ROOM/PHONES outputs and check these with the level meters.

◊ **If you want to monitor the signal of just one AUX bus, none of the other SOLO SWITCHES should be pressed and the MODE switch should be in the SOLO position (not depressed).**

2.3.2 Aux send jacks

Fig. 2.8: Aux send jacks

AUX SEND 1 & 2

The **AUX SEND** jack should be used when hooking up a monitor power amp or active monitor speaker system. The relevant aux path should be set pre-fader.

◊ **AUX SEND 1 is hard wired as pre-fader and hense called MON.**

As already mentioned, the aux sends in the channels— if set post-fader— can be used to connect to external effects devices.

AUX SEND 3

The **AUX SEND 3** jack carries the master aux mix (from the channel's FX controls). You can connect this to an external effects device to process the FX bus. The processed signal can then be brought from the effects device back into the STEREO AUX RETURN jacks.

STEREO AUX RETURN 1 & 2

The **STEREO AUX RETURN 1 & 2** jacks 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.

◊ **You can also use these jacks as additional line inputs.**

All stereo aux returns are balanced, but can of course also be used with unbalanced connectors. If you use an aux send for monitoring, the associated unused stereo aux returns are available for other line level signals (e.g. keyboards).

◊ **A signal fed into the stereo return jacks can be output via an aux send jack. More information on this can be found in chapter 2.3.5 "STEREO AUX RETURN 1/2 (TO AUX SEND)".**

STEREO AUX RETURN 3

The **STEREO AUX RETURN 3** jacks accept the effects mix return (created using the channel FX sends). If these jacks are already in use as additional inputs, you can route the effects signal back into the console via a different channel. The advantage of this is that you can now use that channel's EQ on the effects return signal.

◊ **In this instance, the FX control of the channel being used as an effects return should be turned fully counterclockwise, otherwise feedback problems could occur!**

◊ **If you wish to use the internal effects processor, do not plug any connectors into the STEREO AUX RETURN FX jacks, unless you want to tap the processed signal via the FX OUT**

MUTE

Press the **MUTE** switch to mute the monitor send.

SOLO

The SOLO switch routes the monitor send to the solo bus (post-fader and post-mute) or to the PFL bus (pre-fader and pre-mute). The position of the SOLO MODE switch in the main section determines which of the buses is selected.

2.3.5 Stereo aux return control

STEREO AUX RETURN 1

The **STEREO AUX RETURN 1** control determines the level of this signal in the main mix. If STEREO AUX RETURN 1 is used as effects return, this will determine the level of the effects when mixed with any "dry" channel signal.

◊ **When used in this way, the effects device should be set at 100% effect.**

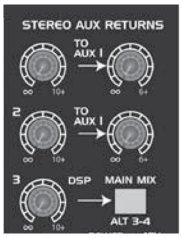


Fig. 2.12: Stereo aux return and stereo aux return (to aux send) controls

STEREO AUX RETURN 1/2 (TO AUX SEND)

The two right-hand **STEREO AUX RETURN** controls have a special function: they can be used to add an effect to a monitor mix. An example follows

Monitor mix with effect

In this instance, your effects device should be set up as follows: the AUX SEND 2 jack should be connected to the L/Mono input of your effects device, with its outputs coming back into the STEREO AUX RETURN 1 jacks.

Connect the AUX SEND 1 jack output to the amplifier of your monitor system. The AUX SEND 1 master control determines the overall volume of the monitor mix.

Using the STEREO AUX RETURN (TO AUX SEND) control, the effect signal can now be blended into the monitor mix.

STEREO AUX RETURN 3

Use the **STEREO AUX RETURN 3** control to determine the level of the signal routed from the AUX RETURN FX jacks to the main mix. If nothing is connected to these jacks, the output of the built-in effects module will appear.

MAIN MIX / TO SUBS

This switch routes the signal fed in via the STEREO AUX RETURN FX jacks either to the main mix (not pressed) or to the submix (pressed).

PHONES/CTRL ROOM ONLY

Use this switch to route the signal appearing at the AUX RETURN 4 jacks to the control room and headphones outputs.

2.3.8 2TK/USB input, 2TK/USB output

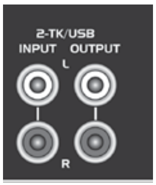


Fig. 2.15: 2-track connectors and lamp socket

2TK/USB INPUT

The **2TK/USB INPUT** jacks (RCA) are designed to accept a 2-track recorder (e.g. DAT recorder), or they can be used as stereo line input. The output signal of a second mixer can also be connected here. If you connect the output of a hi-fi amplifier (with a source selection switch) to the 2TK INPUT, you can easily listen to additional sources (eg. cassette recorder, MD player, sound card, etc.).

2TK/USB OUTPUT

These connectors are wired in parallel to the MAIN OUT and carry the main mix signal (unbalanced). Connect this to the inputs of your recording device. The final output level can be adjusted via the high-precision MAIN MIX fader.

◊ **If you connect a compressor or a noise gate post 2-track output, the main mix fader will probably not be able to create a satisfactory fade-out effect.**

2.3.10 Level meter and monitoring

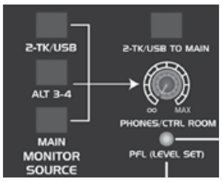


Fig. 2.16: Control room and phones sections of the V18

2TK/USB

The **2TK/USB** switch routes the signal from the 2TK/USB INPUT jacks to the level meter, the CONTROL ROOM OUT outputs and the PHONES jack—this is a simple way to check recorded signals via monitor speakers or headphones.

SUB 1-2

The **SUB 1-2** switch routes subgroup 1-2 to the level meter, CONTROL ROOM OUT and phones.

MAIN MIX

The **MAIN MIX** switch sends the main mix to the CONTROL ROOM OUT and the PHONES output as well as to the level meter.

PHONES/CTRL ROOM

Use this control to adjust the control room output level and the headphones volume.

2 TRACK TO MAIN

When the **2 TRACK TO MAIN** switch is depressed, the 2-track input is routed to the main mix and thus serves as an additional input for tape machines. You can also connect MIDI instruments or other signals here that do not require any further processing. At the same time, this switch disables the main mix to tape output link.

POWER

The blue **POWER LED** indicates that the device is switched on.

+48 V

The red "+48 V" LED lights up when phantom power is switched on. Phantom power is required to operate condenser microphones.

◊ **While phantom power is switched on, do not connect or disconnect microphones on the mixer (or the stagebox/wallbox). Connect any microphones before switching on phantom power. Additionally, monitor/PA speakers should be muted before you activate the phantom power supply. After switching on, wait approx. one minute before adjusting the input gain so that the system has time to stabilize.**

2.3.11 Level Meter

The high-precision level meters always give you an accurate display of signal level.

LEVEL SETTING:

When recording to digital recorders, the recorder's meter should not go into overload. This is because, unlike analog recordings, it takes only slightly excessive levels to create unpleasant digital distortion.

When recording to analog, the VU meters of the recording machine should reach approx. +3 dB with low-frequency signals (e.g. kick drum). Due to their inertia, VU meters tend to display too low a signal level at frequencies above 1 kHz. You should only drive instruments such as a Hi-Hat as far as -10 dB. Snare drums should be driven to approx. 0 dB.

◊ **The peak meters of your M series display level almost independent of frequency. A recording level of 0 dB is recommended for all types of signal.**

SOLO MODE

The **SOLO MODE** switch determines whether the channels' SOLO switch operates as PFL (Pre Fader Listen) or as solo (Solo In Place).

PFL (LEVEL SET)

To activate the PFL function, press the MODE switch. The PFL function should, as a rule, be used for level setting (GAIN). The signal is sourced pre-fader and assigned to the mono PFL bus. In "PFL" mode, only the left side of the peak meter is in operation. A PFL'd channel should be driven to the 0 dB mark of the VU meter.

SOLO (NORMAL)

When the MODE switch is not depressed, the stereo solo bus is active. Solo is actually short for "Solo In Place". This is the customary method for listening to an individual signal or to a group of signals. As soon as a solo switch is pressed, all channels not solo selected are muted in th e monitor path (control room and phones). A channel's position in the stereo image is maintained. The solo bus carries the output signals of the channel pan controls, the aux sends and the stereo line inputs.
All aux returns can be routed to the solo bus.
The solo bus is taken post-fader.

◊ **The PAN control in the channel strip offers a constant power characteristic. This means that the signal is always at a constant level, irrespective of position in the stereo panorama. If the PAN control is moved fully left or right, the level in that channel increases by 4 dB. This ensures that, when set at the center of the stereo image, the audio signal does not appear louder. For this reason, with the solo function activated (Solo in Place), audio signals from channels with PAN controls that have not been moved fully left or right are displayed at a lower volume than in the PFL function.**

As a rule, solo signals are monitored via the control room outputs and headphones jack and are displayed by the level meters. If a solo switch is pressed, the signals from the 2TK input, the subgroups and the main mix are cut from these outputs and the level meter.

MAIN SOLO

The **MAIN SOLO** LED lights up as soon as a channel or aux send solo switch is pressed. The SOLO MODE switch must be set to "Solo".

PFL (LEVEL SET)

The **PFL** (LEVEL SET) LED indicates that the peak meter is set to PFL mode.

Fig. 2.17: PHONES jack

PHONES jack

You can connect headphones to this 6.3mm stereo jack
The signal routed to the **PHONES** connection is the same as that routed to the control room output.

2.3.12 Subgroups and main mix fader

SUB & MAIN faders control the output level of the subgroups and the main mix. Left and right buttons above each sub fader feed the sub channel to left and/or right side of the main mix.

4. Digital Effects Processor

24-BIT MULTI-EFFECTS PROCESSOR

Here you can find a list of all presets stored in the multi-effects processor. This built-in effects module produces high-grade standard effects such as reverb, chorus, flanger, delay and various combination effects. Use the Aux Send **FX** on the channels and the Aux Send **FX** master control to determine the input signal of the effects processor.

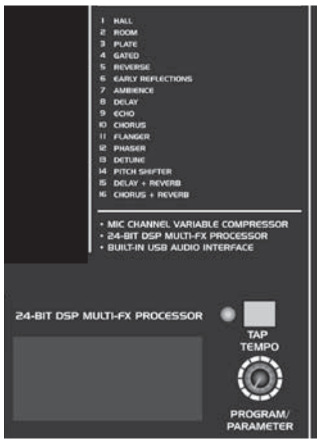


Fig. 4.1: Digital effects module

The built-in stereo effects processor has the advantage that it does not need to be wired up. This excludes the danger of humming or level mismatch right from the start and thus considerably facilitates use.

These effect presets are classical "mixing effects". If you move the STEREO AUX RETURN FX control, you mix the channel signal (dry) and the effect signal. You can control the balance between the two signals with the channel fader and the STEREO AUX RETURN FX control.

FX OUT

Th e M16DSP has a seperate output for the effects device, which is unbalanced and stereo (tip = left signal; ring = right signal; sleeve = ground/shielding). Thus, you can record, for example, a vocal track enhanced with reverb in parallel to a "dry" vocal track; when doing the mix-down later on, you can freely determine the amount of reverb added.

DSP MUTE FOOTSWITCH

Connect a standard foot switch to the foot switch jack and use this to switch the effects processor on and off. A light at the bottom of the display indicates wheater the effects processor has been muted by the foot switch.

The LED level meter on the effects module should display a sufficiently high level. Take care to ensure that the clip LED only lights up at peak levels. If it is lit constantly, you are overloading the effects processor and this could cause unpleasant distortion.

PROGRAM

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.

5. Rear Panel Connectors

5.1 Main mix outputs and control room outputs

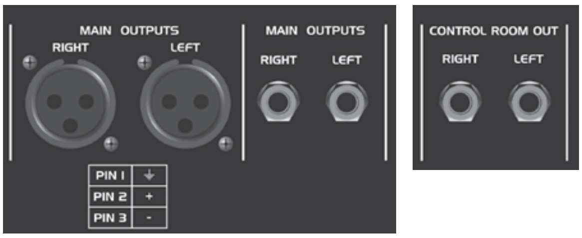


Fig. 5.1: Main Mix outputs, main mix insert points and control room outputs

MAIN OUTPUTS

The **MAIN** outputs carry the MAIN MIX signal and are on balanced XLR jacks with a nominal level of +4 dBu. In parallel with this, 6.3mm phone jacks carry the main mix signal in a balanced format

CONTROL ROOM OUTPUTS (CTRL OUT)

The control room output is normally connected to the monitoring system in the control room and carries the stereo mix or, when selected, the solo signals.

5.2 Subgroup outputs

Fig. 5.2: Subgroup outputs

SUB OUTPUTS

The subgroup outputs are unbalanced and provide the mix of those channels assigned to each subgroup with the SUB switch next to the channel faders. Thus, you can, for example, route a subgroup to a second console or use the output as a recording output in parallel to the main outputs. Panning left feeds to sub1, panning right feeds to sub2.

5.3 Inserts

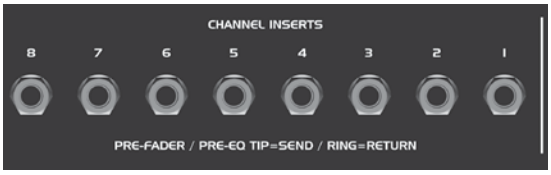


Fig. 5.3: Insert points

Insert points are very useful to process channel signals with dynamic processors or equalizers. Unlike reverb or other effects devices, whose signals are usually added to the dry signal, dynamic processors are most effective on the complete signal. In this case, aux send paths are a less-than-perfect solution. It is better to interrupt the signal path and insert a dynamic processor and/or equalizer. After processing, the signal is routed back to the console at precisely the same point it left. However, the channel signal path is interrupted only if a plug is inserted into the corresponding jack (stereo phone plug: tip = signal output; ring = return input). All mono input channels are equipped with inserts. They are pre-fader, pre-EQ and pre-aux send. Inserts can also be used as pre-EQ direct outputs, without interrupting the signal path. To this end, you will need a cable fitted with mono phone plugs on the tape machine or effect device end, and a bridged stereo phone plug on the console side (tip and ring connected).

5.5 USB input/output

Fig. 5.5: USB input/output

The M16DSP mixer line has built-in USB connectivity, allowing stereo signals to be sent to and from the mixer and a computer. The audio sent from the mixer to a computer is identical to the MAIN MIX. Audio being sent to the mixer from a computer can be routed to the main mix with the 2-TK/USB TO MAIN button.

Connect the USB type B plug into the USB jack on the mixer, and the other end into a free USB port on your computer.
This system uses generic ASIO drivers. Check for operating system updates if there are any driver issues.

5.6 Voltage supply, phantom power supply and fuse

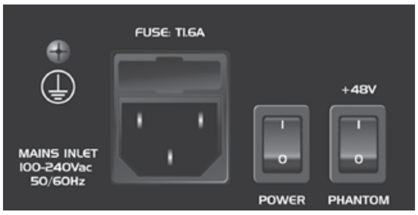


Fig. 5.6: Voltage supply and fuse

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.

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. If mounting in a rack, ensure that the mains can be easily disconnected by a plug pull or by an all-pole disconnect switch on or near the rack.

◊ **Attention: The POWER switch does not fully disconnect the unit from the mains. Unplug the power cord completely when the unit is not used for prolonged periods of time.**

PHANTOM switch

The **PHANTOM** switch activates the phantom power (necessary to operate condenser microphones) on the XLR sockets of the mono channels.
The red **+48 V** LED illuminates when phantom power is on. As a rule, dynamic microphones can still be used with phantom power, provided that they are wired in a balanced configuration. In case of doubt, contact the microphone manufacturer!

◊ **Connect microphones before you switch on the phantom power supply. Please do not connect microphones to the mixer (or the stagebox/wallbox) while the phantom power supply is switched on. In addition, the monitor/PA loud-speakers should be muted before you activate the phantom power supply. After switching on, wait approx. one minute to allow for system stabilization.**

6. Installation

6.1 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.

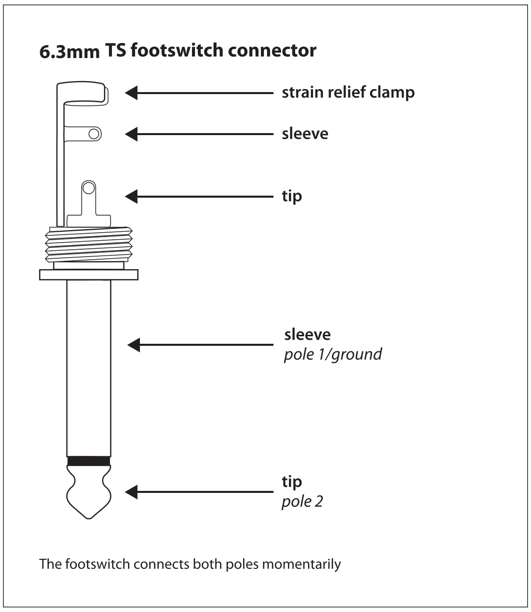


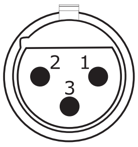
Fig. 6.1: Foot switch connector

6.1.1 Audio connections

Please use commercial RCA cables to wire the 2-track inputs and outputs.

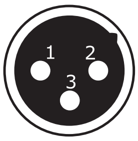
You can, of course, also connect unbalanced devices to the balanced input/outputs. Use either mono plugs, or use stereo plugs to link the ring and shaft (or pins 1 & 3 in the case of XLR connectors).

Balanced use with XLR connectors



input

1 = ground/shield
2 = hot (+ve)
3 = cold (-ve)



output

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

Fig. 6.2: 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.**

Unbalanced 6.3mm TS connector

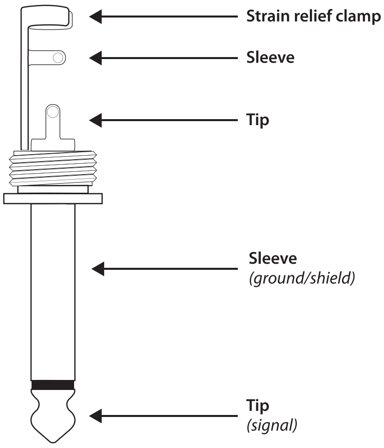
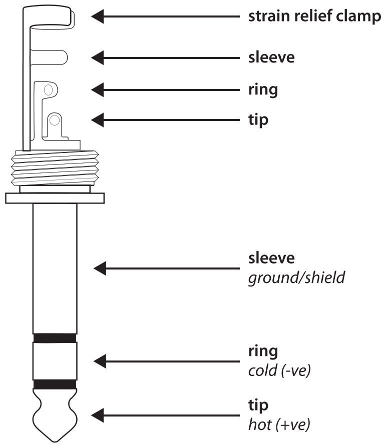


Fig. 6.3: 6.3mm mono plug

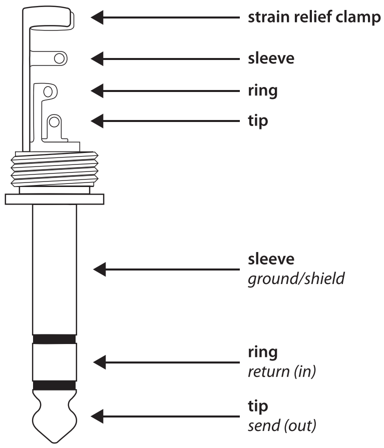
Balanced ¼" TRS connector



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

Fig. 6.4: ¼" stereo plug

Insert send return ¼" TRS connector



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

Fig. 6.5: Insert send/return stereo plug

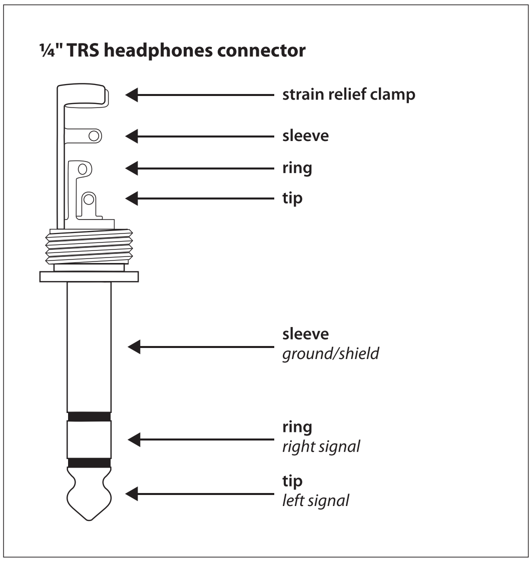


Fig. 6.6: Stereo plug for headphones connection

5. Features

- 16 effect DSP engine with parameter control
- Integral USB interface for PC/Mac
- Variable compressor per mono channel
- +48V phantom on XLRs (globally switched)
- 75Hz lo-cut filter per mono channel
- Multiple pre/post fade Aux feeds
- PFL/Solo monitoring with separate outputs
- Channel inserts/direct per mono channel
- Stereo return for each Aux feed
- Balanced/unbalanced outputs
- Internal power supply (mains IEC)

6. Specifications

Model no.	S18
Order ref.	170.824
Power consumption	40W
Power supply	110-240Vac 50/60Hz (IEC)
Phantom power	+48V (globally switched to XLR inputs)
Input level : Mic	+22dBu
Input level : Line	+20 dBu
Output level : Line	+28dBu (XLR); +22dBu (jack)
Frequency response	20Hz - 30kHz (±1dB)
Effects	16-programme 24-bit DSP engine
T.H.D.	<0.005% (+14dBu @ 1kHz)
Crosstalk	>89dB @1kHz
S/N ratio	-97dBu (channel fader down)
Low-cut filter	75Hz 18dB/oct (mono inputs)
USB port	Stereo in/out 16-bit, 48kHz
Inputs : Mic/Line	9 XLR/jack (bal/unbal)
Inputs : Line	4 mono/stereo jack (bal/unbal)
Inserts	Mono channel inserts/direct (TRS jack)
EQ: High	12kHz shelving ±15dB
EQ: Mid	100Hz - 8kHz swept ±15dB (mono inputs)
EQ : Lo-mid	500Hz ±3dB (stereo channels)
2-track	L+R RCA in/out
Outputs	Main (XLR/jack); Sub; Control Room
Dimensions	478 x 122 x 430mm
Weight	6.22kg

Note: Specifications and design are subject to change without notice for purpose of improvement.