ARTHUR

· USER ·
MANUAL

SWISS MADE +

FORMAT 48 - ANALOG MODULAR MIXER

SCHERTLER

V1

WWW.SCHERTLER.COM

Some words from the electronic designer....

I am delighted that you have purchased SCHERTLER's flagship ARTHUR FORMAT 48 mixer. It is our pleasure to welcome you to a growing family of musicians and technicians, including many of the world's leading soloists and sound engineers, who have chosen to work with SCHERTLER® products.

ARTHUR represents a new concept that enables users to customize and configure a mixer according to their professional needs. The different units can be combined in virtually any order or quantity, within minutes, offering all the standard features you would expect to find on regular mixer channel strips, together with more innovative functions for enhancing workflow and ensuring best possible sound quality. But this modular approach is only part of the story: ARTHUR's advanced electronics also break new ground with a complete absence of negative feedback (NFB) in the mixer circuitry from input to output. This results in ultra-fast response and natural attack. Circuits are built using single, discrete Class-A electronic components and pure high-voltage DC amps (with no capacitors in the signal path), offering 30 dB headroom and low noise, as well as unparalleled stability, warmth and transparency.

While ARTHUR may not offer all the features of a large studio console - indeed, that is not the intention - its sound is reputedly outperforming some of the more established models. This, together with its unparalleled flexibility, makes ARTHUR ideal for a whole range of recording, performing and live sound applications.

We hope that working with your ARTHUR mixer will prove an enjoyable and rewarding experience.

Stephan SCHERTLER
President, electronic designer

and the SCHERTLER Team

INTRODUCTION

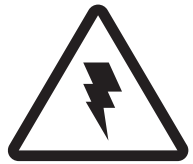
Thank you for choosing the ARTHUR FORMAT 48 mixer. This manual provides information on the available mixer modules, together with technical specifications and essential instructions for assembly. Whatever your current configuration, it is always possible to add additional modules or re-configure your setup to suit specific applications – see the table below for an overview of the various modules and their functionality.

To make the most of your ARTHUR mixer and to ensure trouble-free operation, please **read this manual carefully** before using the mixer for the first time. We also advise keeping the manual for future reference.

INPUTS	
MIC IN ULN	Ultra Low Noise microphone input for professional studio recording. Also ideal for high-quality recording/mixing during film or video production.
YELLOW	High-voltage Class-A instrument preamplifier, designed for direct connection without an external preamp, pedal or guitar amp. Accommodates virtually any musical instrument pickup, or a passive electronic instrument.
STEREO IN	High-quality Class-A line input module with balanced L/R inputs. Intended for use with audio devices such as effects units, recorders and players, preamps and electronic instruments.
MULTIPLE IN	4-channel Class-A line input module intended for use with audio devices such as effects units, recorders, preamps and electronic instruments.
MIC LINE X	Advanced 4-channel Class-A input module. Each channel features sophisticated switching that enables either a microphone or a line level sound source to be connected as required. Three extra AUX SENDS are included per channel.
OUTPUTS	
SUBGROUP	Class-A summing amp that functions as an additional unit for configuring a group of input modules that are to be controlled by a single pair of faders.
L/R MASTER	High-quality Class-A main output module that offers all the basic features required.
AUX MASTER	Additional high-quality Class-A output module designed to complement the L/R MASTER. Enables a larger choice of outputs to be offered.
MASTER X	Superior Class-A output module offering features required in a professional mixer configuration. Includes six independently controllable AUX OUTPUTS and dedicated VU metering. Audio transformers are separately available for all outputs.
FX	
SPRING	Sophisticated spring reverb unit with delay and echo. Designed to function both as an ARTHUR FORMAT 48 mixer module and a stand-alone reverb.

IMPORTANT SAFETY INFORMATION

SAFETY FIRST!
Safety is of major importance when operating any electrical equipment, so please note the following:
On a product, a lightning flash within a triangle indicates the presence of uninsulated "dangerous voltage" within the product enclosure. This may be of sufficient magnitude to cause risk of electric shock.



ELECTRICAL SAFETY
This information applies to all modules and power supplies that form your ARTHUR mixer:

* Before connecting your mixer to the mains, make sure that the mains voltage does not exceed the voltage specified on the mixer/power supply.
* Do not use your mixer if its power supply, cable or plug are not in perfect condition. Replace these as necessary, using the exact models/types specified. If any fixed cables need replacing, this should be done by a suitably qualified professional.
* Your mixer should only be connected to a mains socket with a ground protection system.
* When setting up or installing your mixer, make sure that the mains socket and the power supply's mains cable and plug are easily accessible.
* Do not, under any circumstances, 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 and third prong are provided for your safety. If the supplied plug does not fit your mains socket, consult an electrician for replacement of the obsolete socket.
* Do not expose your mixer to rain or any other water (even in small amounts). Do not use the mixer near water.
* Avoid spilling drinks or any other liquids on the mixer.
* Do not operate your mixer in excessively humid conditions. Avoid excessive heat from sunshine, fire or similar. If your mixer is being used in a dusty environment, make sure it is adequately protected.
* Avoid installing your mixer near any heat sources such as radiators, heat registers, stoves, or other heat-producing apparatus (including amplifiers).
* Do not put any sources of open flame (e.g. candles or pyrotechnics) on or near your mixer.
* Do not cover your mixer during use, or obstruct the ventilation flow in any way.
* Unplug your mixer during lightning storms, or if it is not going to be used for a while. (Remove the plug from the mains socket to completely disconnect the mixer.)
* Your mixer does not contain any "user serviceable" parts. Servicing and/or repairs should only be carried out by qualified personnel. See MAINTENANCE AND REPAIR (below).

OPERATIONAL SAFETY

* During installation or live performances, make sure that your mixer's power supply cable cannot be walked on, tripped over or "pinched" – particularly at sockets, around waste bins etc. Also make sure that the power supply cable is not "stressed" at its point of connection to the mixer.
* To avoid interference, do not install your mixer near power transformers, TV sets, RF transmitters, electric motors, or any other sources of electrical energy.
* To avoid potential accidents, only use attachments, accessories and other equipment such as carts, stands, tripods, brackets or cases that are specified or recommended by the manufacturer, or sold with your mixer.
* Loud volume levels can cause irreparable damage to hearing, so avoid the following while using your mixer:
  - acoustic feedback (never point microphones directly at a loudspeaker)
  - high levels of distortion
  - impulse noises (loud "pops") that can occur when a device is switched on/off, connected to or disconnected from a system.

MAINTENANCE AND REPAIR

* Your mixer can be carefully cleaned, as necessary, using a dry cloth. No water must be used.
* When cleaning, do not use any solvents (such as acetone or alcohol). These could damage the mixer's finish and its labeling.
* Visually check your mixer on a regular basis for any signs of wear and tear or damage, but do not attempt any kind of servicing or repair.
* If your mixer malfunctions, or has been damaged, e.g. if the power supply/cable or plug is damaged, liquid has been spilled or objects have fallen inside, the mixer has been exposed to rain or moisture, does not operate normally, or has been dropped, please call your nearest SCHERTLER technical assistance centre. (For more information, contact us at the address on the last page of this manual.)

CONNECTING THE UNITS

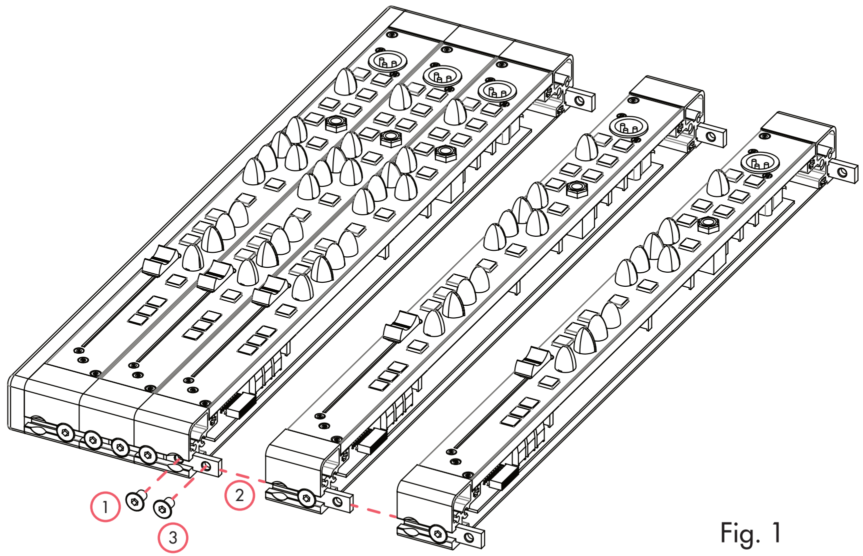


Fig. 1

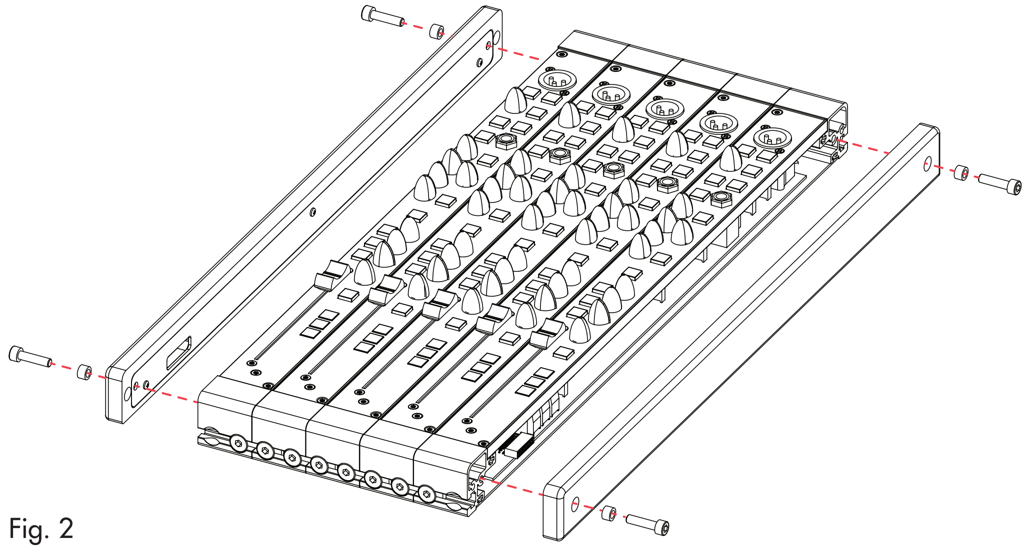


Fig. 2

Input and output modules can be connected in virtually any sequence according to your requirements. However, before assembly we recommend deciding on their basic order from left to right.

In most cases, inputs will be on the left following through to outputs on the right. If you are using an AUX MASTER and a L/R MASTER unit, it makes sense to put the AUX MASTER to the right of the L/R MASTER. This way the AUX faders will go from left to right numerically, i.e. AUX 1 on the L/R MASTER unit, followed by AUX 2 and AUX 3 on the AUX MASTER. However, you have total freedom to choose your own personal sequence - there are no electrical or mechanical restrictions or contraindications.

Once your basic order is determined, connect your first two units as follows (see Fig 1):

- Take the first unit (e.g. a MIC IN ULN) and start by fixing the supplied top and bottom connecting rods to this unit. To do this, loosen the screws (1 and 3) attached to each rod and insert the rod into the right side of the appropriate slot (2) on both the top and bottom of the unit.

- Using the supplied hex key, fix the first screw (1) in place on each rod, but do not tighten it yet.
- Take the next unit, (e.g. another MIC IN ULN) and connect it to right side of the first unit using the connecting rods.

- Gently press both units together, so that all the connectors slot into one another without being squashed.

- Tighten all the screws (3) on each connecting rod so that both units are firmly joined together.

- Connect further units in the same way. (The two connecting rods belonging to the last unit will be left over. Keep these in case any replacements are needed.)

- Now fit the side panels (see Fig.2). Fix these panels using the supplied screws. (The self-cutting screws need some force in order to be turned. Make sure you have inserted the screws straight into the hole during the first few turns.) Different side panels are available - see www.schertler.com for more details.

Note: If buying additional units, rather than removing one of the side panels it may be easier to split the mixer between two existing units and insert the new one in between. However, if your new unit needs to be added to the far left or right of the mixer, as with the SPRING reverb for example, then you will have to remove the relevant panel in order to accommodate the new unit.

POWER SUPPLIES

The ARTHUR modular mixer receives its electrical energy through an external power supply. Although you may have already purchased the necessary power supply together with your other units, here is a general overview of the different power supplies that are available:

ART48-PS12: Compact switching power supply for up to 12 units (max 1 amp)
ART48-PS25: Compact switching power supply for up to 25 units (max 2 amps)
ART48-PS36: High-end linear power supply for up to 15 units

ART48-PSPRO: High-end linear power supply catering for up to 35 units with 3 power outputs.

GROUND LIFT
In most cases (whether in live situations or in a recording studio) it is usual to consider the mixing console as the central ground point in the entire audio system. This consideration is important in order to avoid ground loops between the various pieces of audio equipment. The ART48-MASTER X module has a built-in Ground Lift switch to disconnect the mixer from the ground.

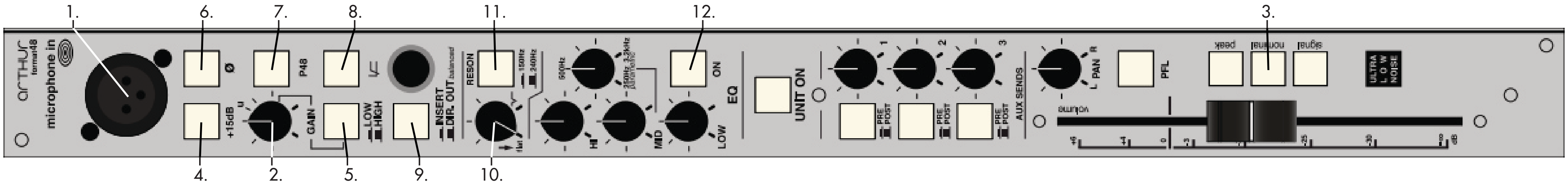
STARTING THE MIXER

Plug the power supply's connection cable into the L/R MASTER or MASTER X DC IN connector. ARTHUR has no on/off switch: when your mixer is not in use, its power supply should be disconnected. Note: The linear power supplies can be switched on or off using the on/off switch on the power supply itself.

Before switching on the mixer, make sure that the master faders are down, (or better still, that the L/R MASTER or MASTER X on/off buttons are in their "off" positions). This will avoid power-on "pops" from any loudspeakers that may have been left on by accident.
Note: A mixer should always be turned on before any of the subsequent devices in the audio chain.

IMPORTANT: Once the mixer is powered up, the audio electronics' DC servos need about 15 minutes to get steady. You can start using the mixer after just 10 seconds, but the slightly unbalanced DC levels will reduce the dynamics and you might hear a "click" noise when pressing some of the buttons, or a slight "crackle" when operating the rotary controls! These effects will disappear after the 15 minutes.

At all times, and particularly before important recordings, the mixer should be warmed up and run for about half an hour before any serious work begins.



1. XLR INPUT: This input receives balanced signals from -60 dBu to +9 dBu, which therefore permits you to connect any audio signal to the unit.

2. GAIN: Adjusting the GAIN affects the amplification rate of the input amplifier. A weaker signal is amplified to a nominal level of 0 dBV and a stronger signal is attenuated so that a nominal signal of 0 dB is always present at the output of the mic input amp.

3. VU METER: This lets you "read" the amount of gain set. Turn up the GAIN to a point where the red peak light occasionally shows. But don't worry too much about this. Thanks to the amount of headroom from input to output on ARTHUR, even strong overloads can be absorbed by the mixer's electronics without resulting in distortion.

4. +15 DB and 5. LOW/HIGH: These buttons help you control the GAIN. The following table shows the functionality of the GAIN plus additional +15 dB and LOW-HIGH buttons:

GAIN	+15 dB button	LOW - HIGH	Gain (dB)
max setting	off (white light)	low (white light)	45 dB
min setting	off (white light)	low (white light)	18 dB
max setting	on (green light)	low (white light)	60 dB
max setting		high (green light)	20 dB
60% setting U		high (green light)	0 dB unity gain
min setting		high (green light)	- 18 dB

The +15dB button is automatically excluded when the LOW-HIGH button is depressed, as it doesn't make sense to amplify and attenuate a signal at the same time. For most stage and recording situations involving dynamic and condenser mics, you may not need the extra 15 dB, but it could be helpful for ribbon mics and mics positioned at a distance, particularly during recordings.

Setting the LOW-HIGH button to HIGH (green light) and setting the GAIN on U means that the mic preamp will neither amplify nor attenuate the signal. This mode is called "unity gain" and is the correct setting for nominal 0 dB signals, or line signals from professional devices such as limiters, effect units or recorders.

6. PHASE (ø): This swaps the "hot" and "cold" aspects of the input signal, inverting its phase 180°. It can be helpful if, for example, two microphones are positioned at a distance, or reverse-faced (as with bottom and top snare drum miking).
Note: The button use is subjective and the results really need to be evaluated through listening.

7. P48: This button, when depressed (red light), delivers 48 VDC of phantom power to the microphone, which will usually be a condenser or active ribbon type. A dynamic mic cannot normally "see" phantom power (as the name suggests), but passive ribbon mics could be permanently damaged by it. Only use this button with mics that you definitely know require phantom power in order to work. Note: The internal circuitry raises the 48 VDC slowly to avoid "pops" and to protect the microphone. Therefore, allow a few seconds for the mic to be working fully.

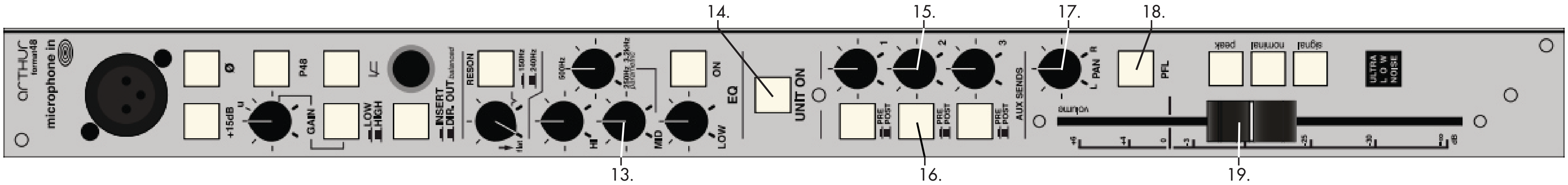
8. LOW CUT: This filter limits low frequencies at 100 Hz/second order, cutting out unwanted frequencies from "boomy" signals. It can also be helpful in shaping signals from smaller instruments (violin, mandolin etc).

9. INSERT/DIRECT OUT: The INSERT works in a similar way to inserts on other mixers. However, a related "bypass" button offers additional possibilities. When this is depressed, the signal in the mixer is interrupted and the INSERT will work in the usual way. By connecting a mono jack, you get the output line signal on the "tip". By connecting a standard stereo jack, you get the (output) signal from the "tip" (send) and the return signal will be connected to the mixer through the "ring" (return) of the jack. When the button is not depressed, the signal will not be interrupted by the insertion of a jack into the INSERT, working as an unbalanced DIRECT OUT . Here, the insert connection works as a sleeve out or "dry line out post input amp". You can connect a mono or stereo jack to the insert. Red light = depressed (interrupted.) White light = not depressed (bypassed).

10. RESON: The resonance filter is a kind of notch filter, but one that is gradually adjustable over its attenuation level. When set to the "flat" position it will be totally bypassed, thus having no effect on the incoming signal. This filter is designed to avoid (or at least attenuate) feedback on acoustic instruments that are miked up in live situations using pickups, e.g. the SCHERTLER DYN Series. A double bass or 'cello might get in resonance at ca. 150 Hz, whereas guitars, violins and similar musical instruments will do so at ca. 240 Hz. The Q is very high, cutting out a very narrow band at the respective frequency.
When the control is turned slowly clockwise, the filter will gradually attenuate at the chosen frequency (see 11 below).

11. 150Hz-240Hz: This lets you select the frequency. When the button is not depressed (red light), the filter will attenuate at 150 Hz. When the button is depressed (blue light), the filter will attenuate at 240 Hz.

12. EQ ON: This button activates or bypasses the EQ section (green light when activated/depressed, white light when deactivated). It can be useful for comparing an EQ configuration with the unfiltered sound. Note: The EQ ON button will not bypass the RESONANCE filter.



13. EQ – HI/MID/LOW: The HI control lets you tune the high range of the audio spectrum (from 4.5 kHz) by +/- 12 dB with a slope of 18 dB / octave. The 3rd order shelving filter “keeps” the circle of influence within the filter’s audio band so as not to overlap with the MID filters. This makes adjustment of the higher frequencies more accurate. The MID control, together with the MID FREQ control, acts on frequencies within a wide mid range of 250 Hz to 3 kHz, with amplification or attenuation of +/- 15 dB. The LOW control lets you adjust the signal from –20 dB/+15 dB up to 100 Hz with a slope of 12 dB / octave. The higher shelving filter prevents the low frequencies from overlapping with the parametric mid, making adjustment of the lower frequencies more accurate.

The HI, MID and LOW controls all have a detent at their mid positions. This indicates the filter’s “flat” position.

14. UNIT ON: This button connects/disconnects the output routing for all outputs (AUX1, AUX2, AUX3 and L/R) except the PFL routing. Its function is similar to the Mute button on other mixers, but here the functionality is reversed. When the button is depressed, all outputs get connected - whereas a Mute button disconnects the output when pressed. Also, whereas a Mute button normally only disconnects the L/R routing i.e. the channel’s fader, this button affects all outputs. Being able to switch off a channel strip makes sense in order to prevent the signal from still going through to stage monitors, or to the input of the reverb unit for example.
Note: Even if the button is in the switched off position (not depressed, white light), the PFL and INSERT will still be ready to function (as shown by the lights on their respective buttons) even though all other button lights are off.

15. AUX SENDS: Three rotary controls enable the level of each send (AUX1, AUX2 and AUX3) to be independently regulated.

16. PRE/POST buttons: Each AUX SEND can “read” the signal pre or post fade. These buttons let you select the pre or post fade mode. If a button is not depressed (red light), the signal arrives “post fade”. Here, the relevant auxiliary level is regulated by the level control, but the final outcome also depends on the channel fader’s position. (This is very useful when driving a reverb unit for example, where you can set the level control to create the desired proportion of reverb to dry signal, then maintain this proportion while any overall level changes are made using the channel fader.) If a button is depressed (blue light), the signal arrives “pre fade” i.e. it will be sent to the AUX MASTER without being influenced by the position or movement of the channel fader. (This configuration is useful when driving stage monitors or similar devices through the respective auxiliary master, where influence from the main channel fader is not required.
Note: As well as the L/R MASTER unit, you will also need the AUX MASTER unit to benefit from all the AUX SENDS and other additional options (see page 50). The L/R MASTER can only receive AUX 1.

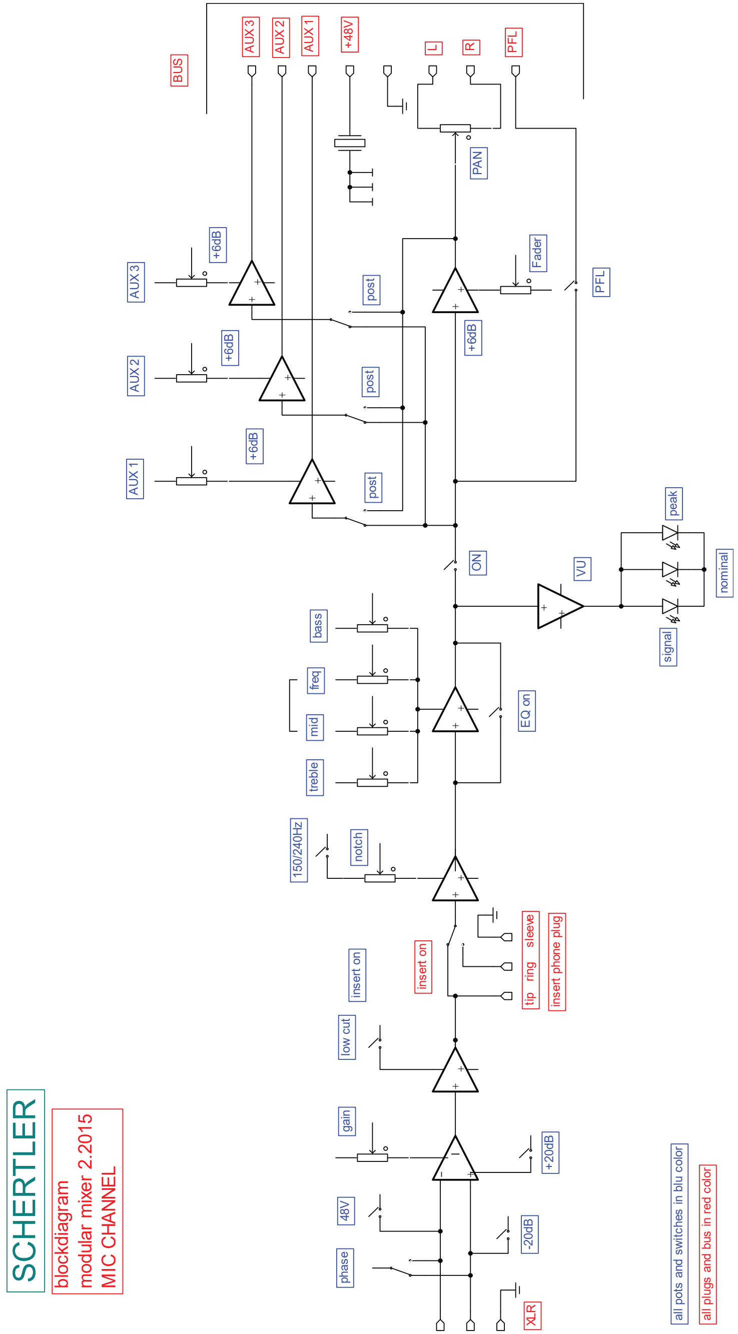
17. PAN: The PAN control lets you send the signal to the left or right channel. Its configuration is designed to guarantee minimum noise and maximum dynamics in the central position.

18. PFL: As well as being a traditional PFL (pre fade listen), this button also serves as a fourth AUX SEND, albeit without the possibility to set any send levels. Note: You can only operate the PFL through the AUX MASTER unit. All channels that have their PFL activated (red light) will be mixed in the AUX MASTER unit according to the GAIN levels set on each channel and sent to the PFL output, controlled by the AUX MASTER’s PFL fader. This function can be useful if you have an additional monitor and only need one signal, e.g. for a singer who only requires the “voice channel”. The PFL section still runs if the UNIT ON is deactivated (white light).

19. Channel Fader: The channel fader controls the total amount of signal going to the master. To exclude this signal from the L/R MASTER or MASTER X without changing the fader position, simply switch the UNIT ON button (14) to its OFF position (white light). The button will then act as a mute.

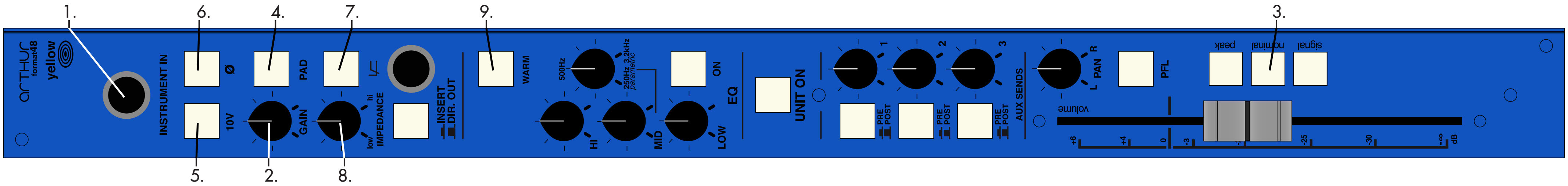
Note: This unit is also available in 4- and 8-channel configurations (MIC IN ULN x4 and MIC IN ULN x8). Equipped with 4 and 8 mic inputs respectively, each channel has identical features while the unit’s structure is sturdier (and the price is lower than 4 or 8 single modules!)

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	72 dBu
Pad (attenuation)	22 dB
Max input level	+4 dBu (low mode), +26 dBu (high mode)
Mic In connector	XLR
Mic In sensitivity	-60 to -18 dBu (low mode), -20 to +9 dBu (high mode)
Mic In impedance	4.7 kΩ
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	n.a.
Line In sensitivity	n.a.
Line In impedance	n.a.
Main Out connector	through L/R Master module
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	16 Hz to 125 kHz
Insert/Direct Out	1/4" stereo jack unbalanced
Insert/Direct Out level	26 dBu
Phantom power (nominal)	48 VDC
Stat power (10 VDC)	No
EQ	
Low	Shelving, +15 dB / -20 dB (@100 Hz)
Mid	±15 dB (250 to 3 kHz)
High	Shelving, ±12 dB (@45 kHz)
Filter	Reson (notch): -10 dB (@150 Hz - @240 Hz) Low Cut: shelving, 2nd order (cut freq. 100 Hz)
EIN	-128.7 dB
Distortion (THD+N @1kHz / 0 dBu output)	0.13% (through L/R Master) 0.12% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	3.8 W (80 mA)
Supply	Through L/R Master module
Modular	Yes



1. INSTRUMENT IN: This unbalanced input can receive signals from -42 dbu to +9 dBu. The 51 dB range means that you can connect any musical instrument pickup to the unit.

2. GAIN: Adjusting the GAIN control affects the amplification rate of the input amplifier. A weaker signal is amplified to a nominal level of 0 dBV and a stronger signal is attenuated so that a nominal signal level is always present at the output of the mic input amp.

3. VU METER: This lets you "read" the amount of gain set. Turn up the GAIN to a point where the red peak light occasionally shows. But don't worry too much about this. Thanks to the amount of headroom from input to output on ARTHUR, even strong overloads can be absorbed by the mixer's electronics without resulting in distortion.

4. PAD: Depressing the PAD button (green light) attenuates the signal by 15 dB. This is useful for active instruments that have a very strong output.

5. 10V: This delivers the necessary 10 VDC supply for all SCHERTLER electrostatic pickups (STAT- and BASiK Series), so they can be directly connected to this instrument input unit without using their original preamp. The 10V button can also be used for connecting unbalanced electret microphones to the unit.

6. PHASE: Combining two different sources can create phase cancellation, so the PHASE button (labelled ø) swaps the "hot" and "cold" aspects of the input signal, inverting the signal phase 180°. This can be helpful, for example, when a pickup is combined with a microphone (going through its own mic unit). Note: The ø button use is subjective and the results really need to be evaluated by listening.

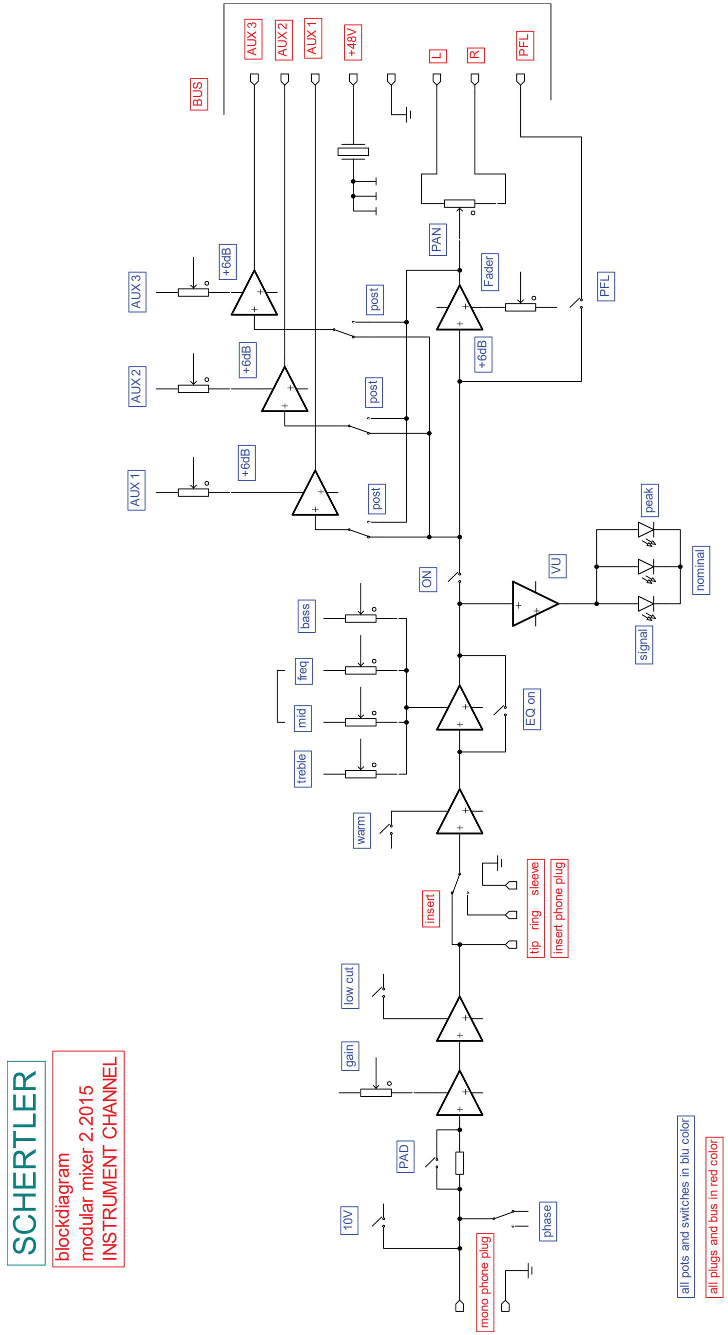
7. LOW CUT: This button limits low frequencies at 100 Hz/second order, cutting out unwanted frequencies from "boomy" signals as well as letting you shape the sound of an acoustic instrument.

8. IMPEDANCE: This changes the input impedance from 20 kΩ to 800 kΩ as the control is turned from low to hi. For active instrument pickups it should be set to low. For magnetic pickups it should be set between the extremes. For piezo pickups it should be set to hi. Magnetic pickups in particular can benefit from correctly adjusted input impedance.

9. WARM: This button acts as a low pass filter. When depressed, it dampens higher frequencies to produce a warmer sound i.e. when using bridge-mounted pickups such as the SCHERTLER STAT Series for violin, cello and double bass.

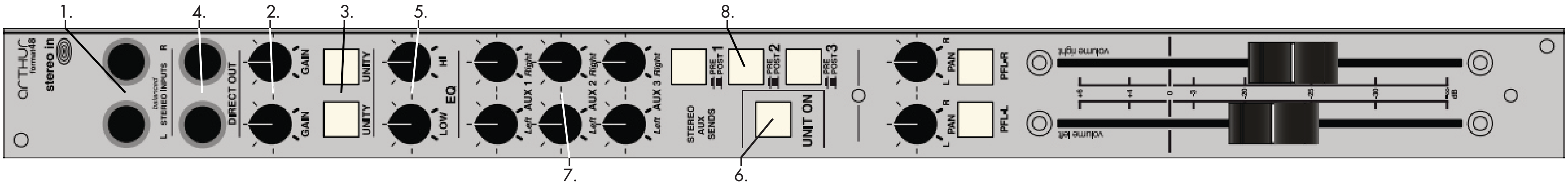
Note: Aside from all the above features, YELLOW is similar to the MIC IN ULN unit. Signal flow from INSERT to fader (INSERT/DIRECT OUT, EQ, UNIT ON, AUX SENDS, PAN and PFL) is identical to that of the MIC IN ULN. For more information please read pages 10-13.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	54 dBu
Pad (attenuation)	15 dB
Max input level	+24 dBu
Mic In connector	n.a.
Mic In sensitivity	n.a.
Mic In impedance	n.a.
Instrument In connector	1/4" jack unbalanced
Instrument In sensitivity	-42 to +9 dBu
Instrument In impedance	20 to 800 kΩ
Line In connector	n.a.
Line In sensitivity	n.a.
Line In impedance	n.a.
Main Out connector	through L/R Master module
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	50 Hz to 50 kHz
Insert/Direct Out	1/4" stereo jack unbalanced
Insert/Direct Out level	30 dBu
Phantom power (nominal)	No
Stat power (10 VDC)	Yes
EQ	
Low	Shelving, +15 dB / -20 dB (@100 Hz)
Mid	Parametric, ±15 dB (250 to 3 kHz)
High	Shelving, ±12 dB (@4.5 kHz)
Filter	Warm: Low pass, 2nd order (cut freq. 2 kHz) Low Cut: shelving, 2nd order (cut freq. 100 Hz)
EIN	n.a.
Distortion (THD+N @1kHz / 0 dBu output)	0.11% (through L/R Master) 0.11% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	3.7 W (78 mA)
Supply	Through L/R Master module
Modular	Yes



1. L/R STEREO INPUTS: STEREO IN has two fully balanced input connectors that use ¼" jacks. The "tip" connects as usual to the hot signal, the "ring" to the cold and the "sleeve" to the ground. You can also connect an unbalanced signal, for example from a keyboard.

2. GAIN: Although this unit is intended to accommodate nominal line level signals, the input amp's sensitivity can be adjusted from -24 dBu to +15 dBu using the GAIN control. Left and right input channels can be separately adjusted.

3. UNITY: Depressing the UNITY button sets the gain to 1 allowing the signal to flow - without amplification or attenuation - independently from the GAIN control setting. This could be particularly useful in a recording situation for example, where several "line level" devices might be connected to multiple STEREO IN units. Here, all inputs can be evenly set to an accurate nominal level by depressing the UNITY buttons.

4. DIRECT OUT: The DIRECT OUT connections work as a sleeve out or "dry line out post input amp". You can connect a 1/4" mono or stereo jack. On the V2 version of this module the unbalanced line signal will be transmitted through the "tip" of the jacks, while the V3 version features balanced DIRECT OUT connections.

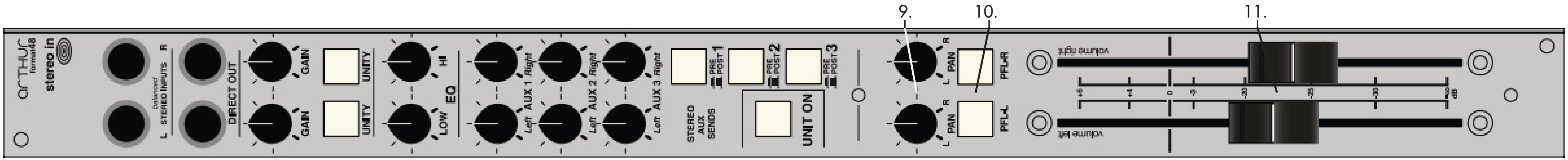
5. LOW/HI EQ: Line signals do not usually need strong shaping. However, a little correction of the higher and lower frequencies can be useful. The stereo LOW and HI filter controls affect both the left and right inputs simultaneously.

6. UNIT ON: This button connects or disconnects the output routing for all outputs (AUX 1, AUX 2, AUX 3 and L/R), with the exception of the PFL routing. Its function is similar to the Mute button found on other mixers, except that its functionality is reversed. When the UNIT ON button is depressed, all outputs are connected (whereas a Mute button in its depressed position disconnects the output). Also, a Mute normally only disconnects the L/R routing (the channel's fader), whereas this module's UNIT ON button affects all outputs. Being able to switch off a channel strip makes sense: for example, it prevents a signal from still going through to stage monitors, or to the input of a reverb unit.

7. AUX SENDS: There are three AUX SENDS, each with a separate left and right level control. This allows the AUX SENDS for each channel to be controlled independently when the STEREO IN module is used for two different mono line signals instead of a coupled left and right.

8. PRE/POST buttons: Each AUX SEND can "read" the signal either pre or post fade. When the button is not depressed (orange light), the signal arrives "post fade". Here, the relevant auxiliary level will be regulated by its level control, but the final outcome also depends the channel fader's position. When the button is depressed (blue light), the signal arrives "pre fade", i.e. it will be sent to the AUX MASTER without being influenced by the position or movement of the channel fader.

Note: As well as the L/R MASTER unit, you will also need the AUX MASTER unit to benefit from all the AUX SENDS and other additional options (see page 50). The L/R MASTER can only receive AUX 1.



9. PAN: Each mono input channel has a PAN control. While somewhat unusual for a stereo input, it makes this particular unit more versatile. The STEREO IN effectively consists of two independent mono input channels, each with their own controls - with the exception of the LOW/HI Filter that affects both inputs simultaneously.

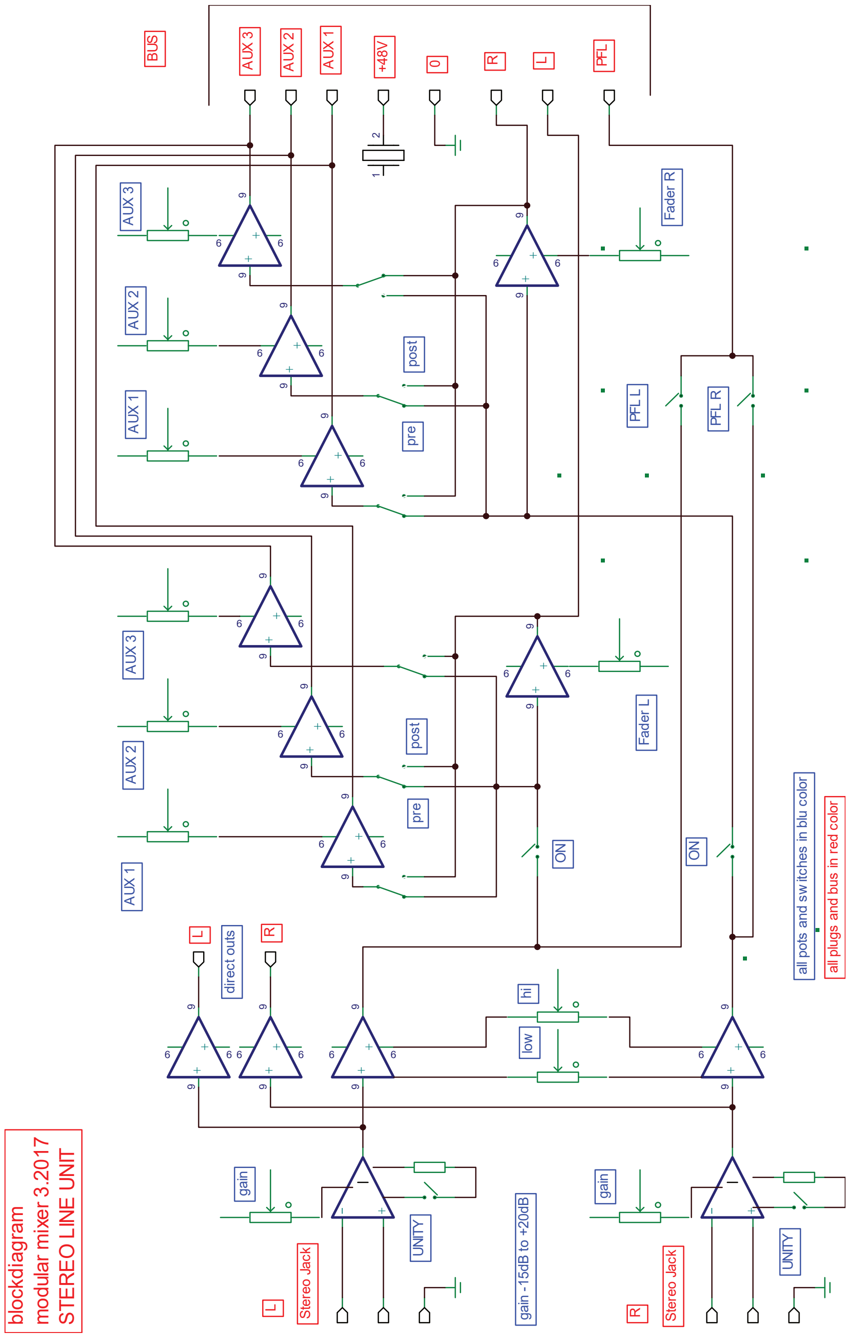
The two PAN controls therefore play an important role:
• When these are set hard left and right respectively, the unit gives you a stereo input (as its name suggests)
• When these are set to a central position, you have a stereo input "switched to mono".
• When plugging two independent sources into the STEREO IN, such as a guitar and a mono effect, you can use the PAN controls to route each source left and right as required.

10. PFL L/R: Each channel (left and right) has a PFL button. When depressed (red light), the PFL will send the respective signals to the phones or PFL output on the AUX MASTER unit. As well as being a traditional PFL (pre fade listen), this one also serves as a fourth AUX SEND, albeit without any possibility to set the level after the GAIN control. Any channel that has the PFL button activated (red light) will be mixed in the AUX MASTER unit and sent to the PFL output, controlled by the PFL fader on the AUX MASTER. This function can be useful when an additional monitor is needed, for example for a keyboardist, where only one signal might be required (in this case the signal from the "keyboard channel"). Note: The PFL section still runs if the UNIT ON button (6) is deactivated (white light).

11. L/R Faders: The channel fader and its associated functionality is probably the most important part of the output process. In most cases, the signal going through here will be mixed in the L/R MASTER and will appear on the main L/R outputs that drive the recording device or the front-of-house PA speakers. This is the signal that is usually heard by the public. Operation of the STEREO IN's fader section is identical to that on other mixing consoles. The faders let you control the total amount of signal going to the master. If you want to exclude the channel signals from the L/R MASTER without changing the fader position, simply deactivate the UNIT ON button (off = white light). This button will then be acting as a MUTE.

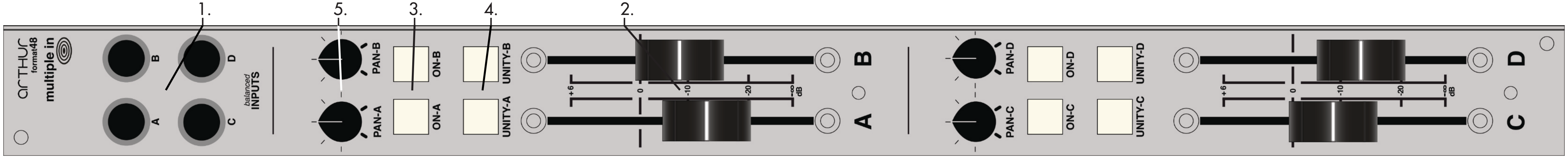
Note: This unit is also available in a STEREO IN x4 configuration, consisting of four STEREO IN units in a single module.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	36 dB
Pad (attenuation)	No
Max input level	+28 dBu
Mic In connector	n.a.
Mic In sensitivity	n.a.
Mic In impedance	n.a.
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	2x 1/4" jack balanced
Line In sensitivity	-24 to +15 dBu
Line In impedance	20 kΩ
Main Out connector	through L/R Master module
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	8 Hz to 90 kHz
Insert/Direct Out	2x 1/4" stereo jack unbalanced
Insert/Direct Out level	n.a.
Phantom power (nominal)	No
Stat power (10 VDC)	No
EQ	
Low	Shelving, ±16 dB (@500 Hz)
Mid	n.a.
High	Shelving, -15 dB / +14 dB (@1.2 kHz)
Filter	No
EIN	-109 dB
Distortion (THD+N @1kHz / 0 dBu output)	0.1% (through L/R Master) 0.09% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	5 W (105 mA)
Supply	Through L/R Master module
Modular	Yes



1. INPUTS (balanced): MULTIPLE IN has four balanced ¼″ jack inputs enabling it to accommodate four line signals (or two stereo inputs). The unit is mainly intended for use with audio devices such as recorders, effects units, preamps or electronic instruments that generate line signals (0 dBV) from their outputs. All four inputs will receive nominal line signals (0 dB), therefore no gain adjustment is previewed and the input amp's sensitivity is always 0 dB unity gain.

2. VOLUME: Each channel has its own volume control. This allows four independent mono signals to be individually controlled and mixed.

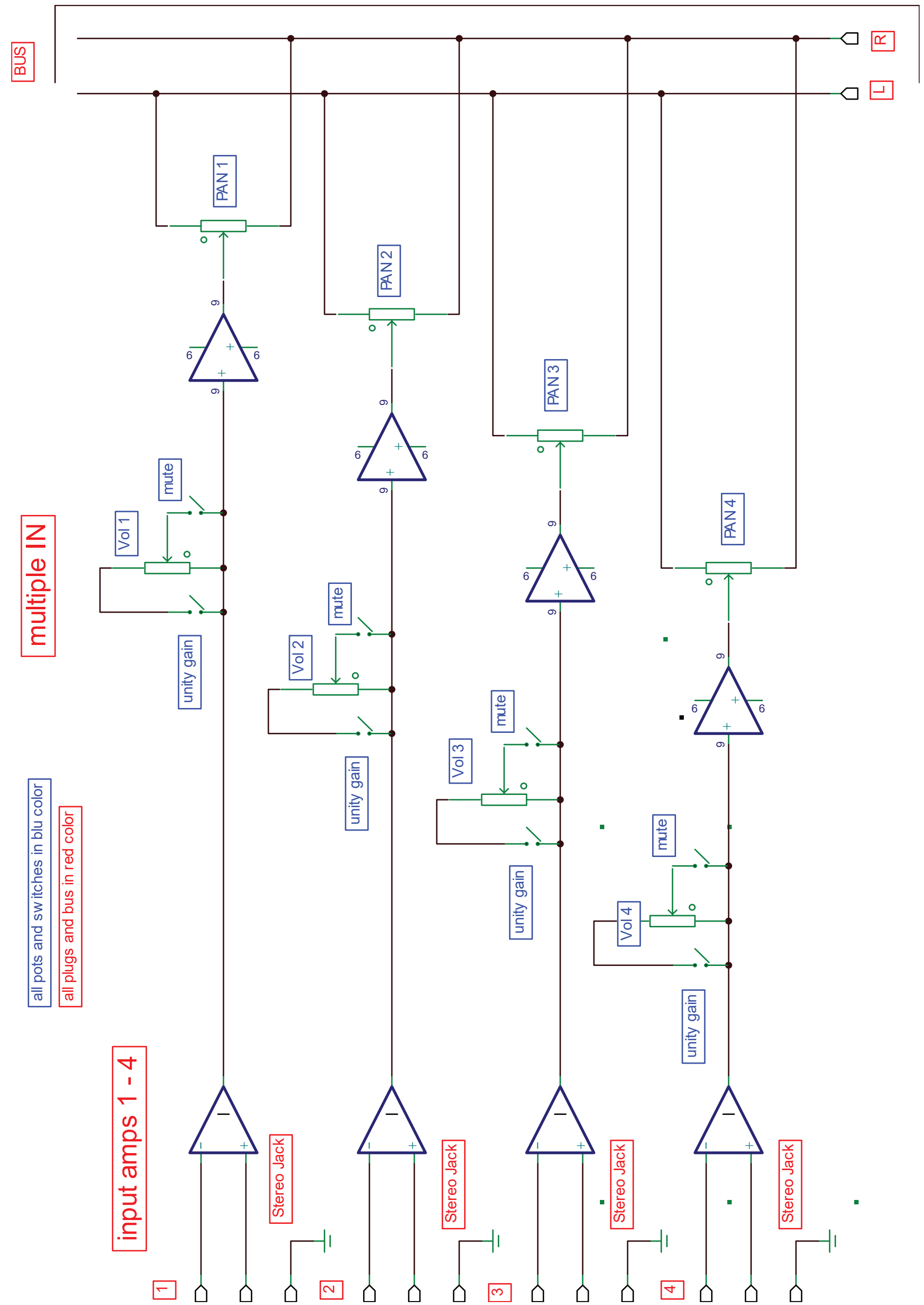
3. ON: Each channel has a separate ON button that acts as a mute. This lets you mute the respective signal without altering the position of the fader. (Note that the functionality is reversed when compared with a standard Mute button: when the ON button is depressed, the channel is connected)

4. UNITY: Depressing the UNITY button lets you set an amplification level of 0 dB independently from the position of the volume fader. This can be very useful when you have several similar sources connected to the unit, which you then want to send to the L/R BUS at the same nominal level, for example via one or two SUBGROUP units.

5. PAN: If you want to connect one or two stereo sources such as recorders or keyboards, the PAN control allows the MULTIPLE IN to route the stereo signal to the L/R MASTER. Connect the left signal to Channel A setting the PAN hard left, and the right signal to Channel B setting the PAN hard right. The same can be done with a second set of stereo signals, connected to Channels C and D and panned hard left and right as above.

Note: A number of MULTIPLE IN units combined with one L/R MASTER unit will create a modular summing amp of outstanding quality. For example, four MULTIPLE IN units will configure into a 16-channel summing amp.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	6 dB
Pad (attenuation)	No
Max input level	+20 dBu
Mic In connector	n.a.
Mic In sensitivity	n.a.
Mic In impedance	n.a.
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	4x 1/4" jack balanced
Line In sensitivity	-61 to +6 dBu
Line In impedance	41.4 kΩ
Main Out connector	through L/R Master module
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	<10 Hz to 90 kHz
Insert/Direct Out	No
Insert/Direct Out level	n.a.
Phantom power (nominal)	No
Stat power (10 VDC)	No
EQ	
Low	No
Mid	No
High	No
Filter	No
EIN	-111.7 dB
Distortion (THD+N @1kHz / 0 dBu output)	0.15% 0.12% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	3.8 W (80 mA)
Supply	Through L/R Master module
Modular	Yes

SUBGROUP POSITIONING

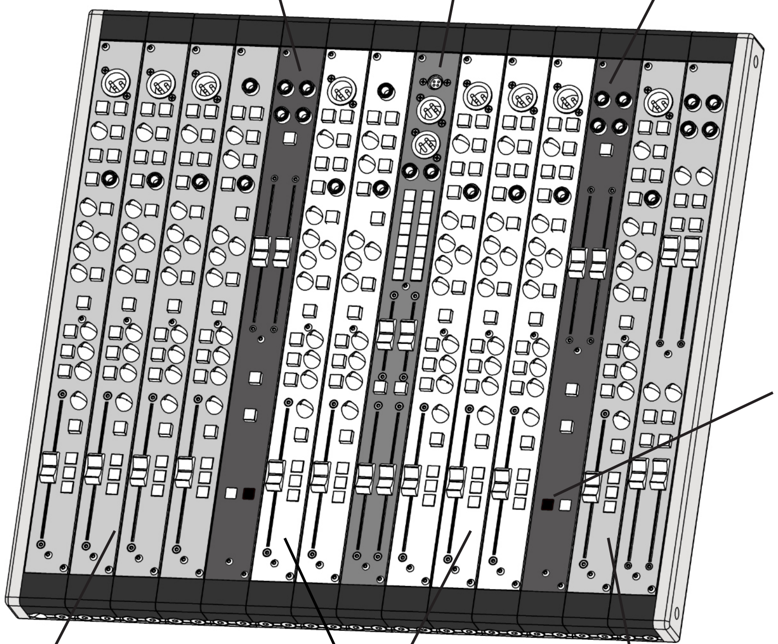


Fig. 1

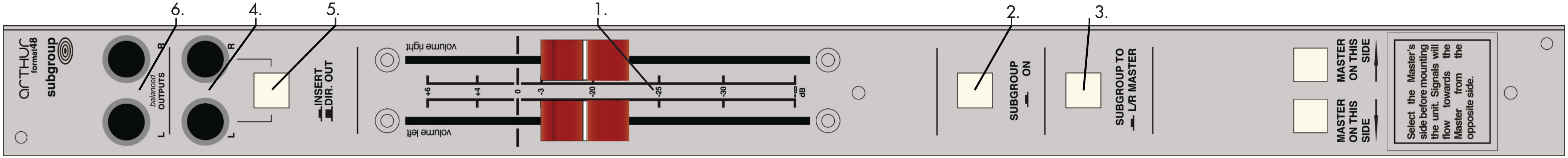
SUBGROUP is a high-quality, Class-A summing amp that functions as an additional unit for configuring a group of input modules that are to be controlled via a single pair of faders. Two SUBGROUP units can be directly mounted in the ARTHUR mixer - along with a theoretically infinite number of other SUBGROUP units that must be cabled to either a STEREO IN or MULTIPLE IN unit.

Fig. 1 shows an example of how to mount the SUBGROUP unit(s). If you are only using two units, which in most of cases should be sufficient, one SUBGROUP unit should be placed to the left of the L/R MASTER and the other should be placed to the right.

Before connecting the SUBGROUP to any other units, you might first need to set the MASTER SIDE SWITCH. This is inside the module on the circuit board near the BUS connector. If the L/R MASTER is to the right of your SUBGROUP, do not depress the switch. If it is to the left, depress it. Once the mixer is put together, one of two LEDs will light up showing which side has been selected.

The group of units being controlled via the SUBGROUP should be placed “before” the unit, i.e. on the opposite side from the L/R MASTER unit. Depress the SUBGROUP TO L/R MASTER button to let the summed signal flow back to the L/R BUS. Any other input units that are next to the L/R MASTER, on its left or right side, will be summed by the L/R MASTER and will not be affected by the SUBGROUP unit(s).

The mixer might therefore be configured as follows, working left to right: Input units controlled by the first SUBGROUP; the SUBGROUP itself (Master Side Switch not depressed); the independent inputs not affected by the SUBGROUP; the L/R MASTER unit; another set of independent inputs (if necessary); the second SUBGROUP (Master Side Switch depressed); the input units controlled by that second SUBGROUP.



1. L/R VOLUME Faders: These control the level of all connected input units.

2. SUBGROUP ON: Depressing this button let you mute the entire subgroup without using the faders.

3. SUBGROUP TO L/R MASTER: If using more than two SUBGROUP units, these additional units should be mounted towards the outer edges of the mixer. Their outputs must be connected to either a STEREO IN or a MULTIPLE IN unit using stereo jack cables. The STEREO IN / MULTIPLE IN unit(s) should also be positioned near the L/R MASTER, together with any independent "non influenced" input channels, so that their signal can flow directly to the L/R MASTER. Setting this button to OFF (not depressed) enables the signal to go through cables to the STEREO IN or MULTIPLE IN unit and from there to the L/R BUS.

Set the MASTER SIDE SWITCH on these units in the same way as for the first two SUBGROUP units (see page 31).

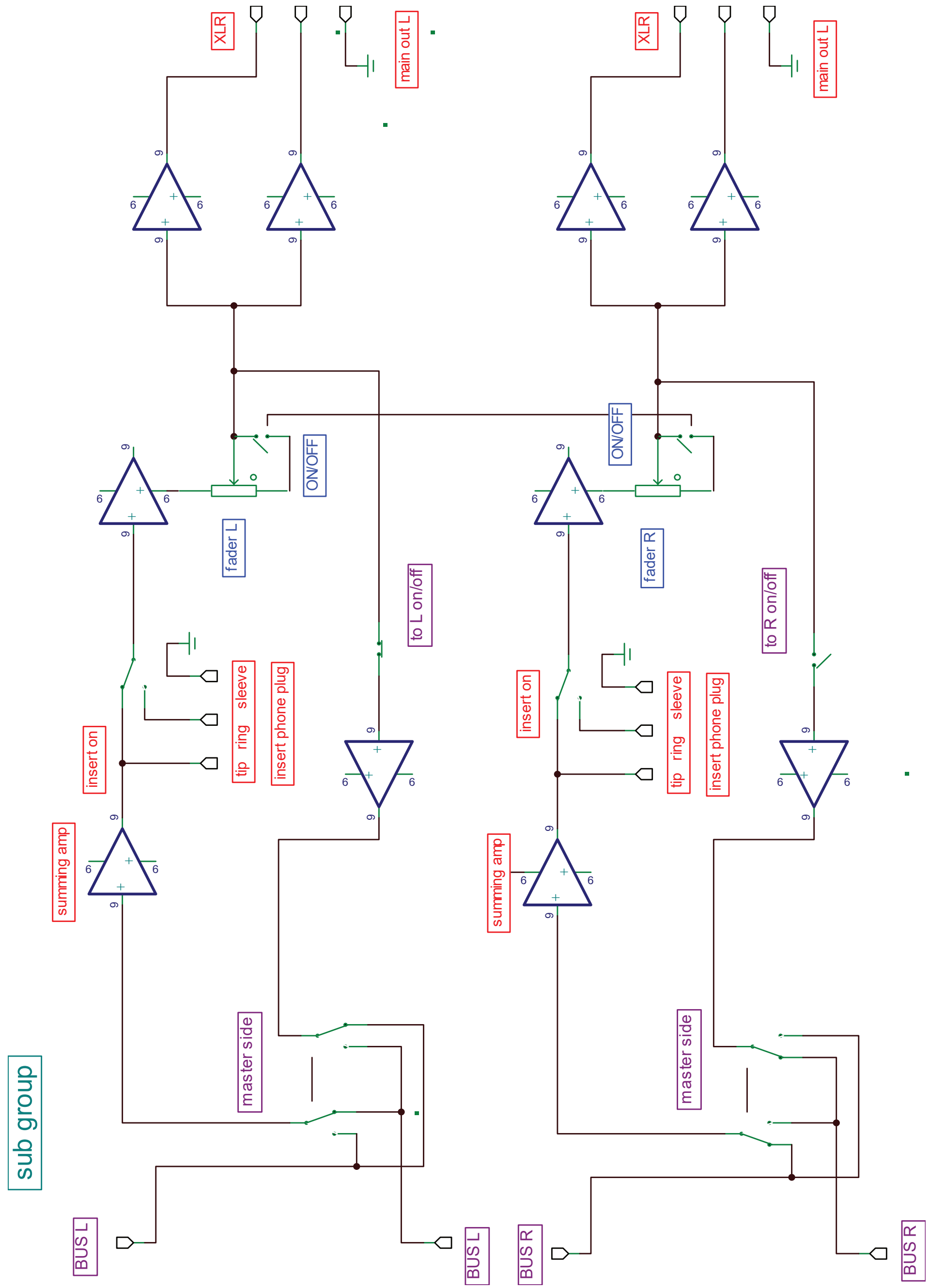
4. L/R INSERT: The INSERT follows the summing amps of the SUBGROUP unit. It works both when the signal is routed to the L/R BUS (first two SUBGROUPS) and when the signal is routed though cables to a separate STEREO IN / MULTIPLE IN unit (when there are more than two SUBGROUPS).

5. INSERT/DIRECT OUT: If this button is not depressed (white light = off) the signal will not be interrupted by the insertion of a jack into the INSERT. If the button is depressed (red light = on) the signal in the mixer will be interrupted and the INSERT will work in the usual way. You can connect a mono or stereo jack to the INSERT. By connecting a mono jack, you will get the output line signal from the "tip". By connecting a standard stereo jack, you get the output signal from the "tip" (send) and the return signal through the "ring" (return).

6. OUTPUTS (balanced): These outputs are designed to send balanced signals via ¼" jacks. The "tip" connects as usual to the hot signal, the "ring" to the cold signal, and the "sleeve" to the ground. The outputs can be used for connecting an additional SUBGROUP to a STEREO IN / MULTIPLE IN unit. They are also active in the "SUBGROUP to L/R MASTER" mode (see 3 above) and can therefore be used as a second independent subgroup output. Not depressing the SUBGROUP to L/R MASTER button lets the unit act as a simple "L/R Master", creating a "second mixer within the mixer".

Note: Another interesting application for these outputs is as a balanced insert. Connect a stereo equalizer to them, for example, and connect that unit's outputs to a STEREO IN or MULTIPLE IN unit. This lets you profit from the effect unit's balanced inputs and outputs. The application could be useful with any SUBGROUP unit, routing it from its outputs, via cable, to one of the aforementioned STEREO IN or MULTIPLE IN units (simply by not depressing and therefore not activating the SUBGROUP TO L/R MASTER button).

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	6 dB
Pad (attenuation)	No
Max input level	+24 dBu
Mic In connector	n.a.
Mic In sensitivity	n.a.
Mic In impedance	n.a.
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	From input modules
Line In sensitivity	-80 to +5 dB
Line In impedance	n.a.
Main Out connector	2x 1/4" jack balanced
Maximum output level	32 dBu
Main Out impedance	120 Ω
Main Out freq. response	<10 Hz to 40 kHz
Insert/Direct Out	2x 1/4" stereo jack unbalanced
Insert/Direct Out level	n.a.
Phantom power (nominal)	No
Stat power (10 VDC)	No
EQ	
Low	No
Mid	No
High	No
Filter	No
EIN	n.a.
Distortion (THD+N @1kHz / 0 dBu output)	0.01%
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	no
Power consumption	3.8 W (80 mA)
Supply	Through L/R Master module
Modular	Yes

SPRING POSITIONING

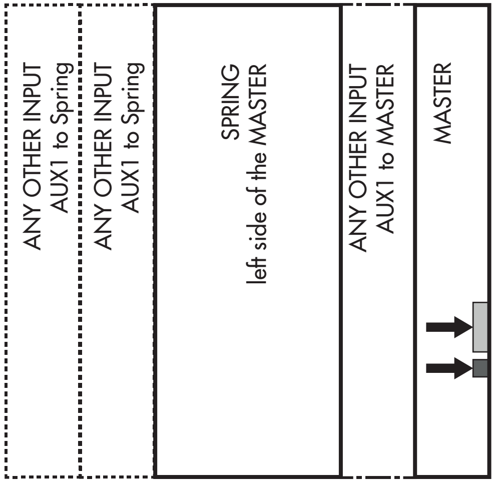


Fig. 1

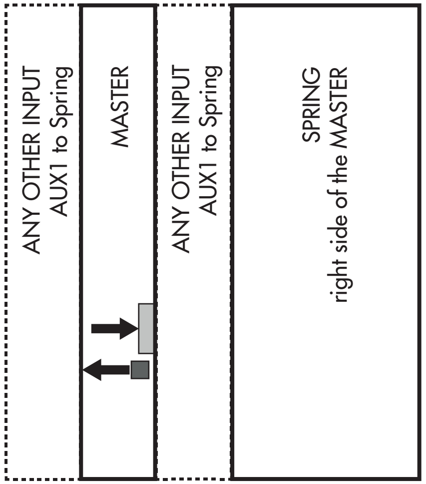


Fig. 2

released

depressed

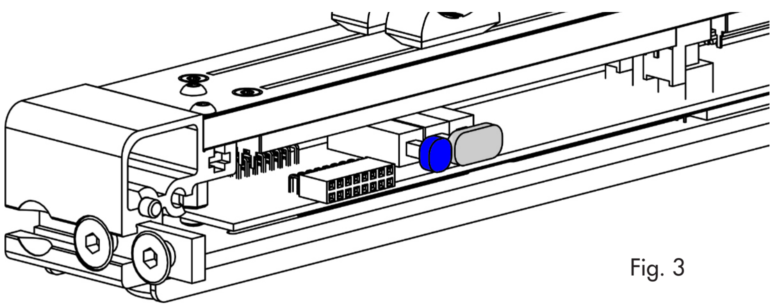


Fig. 3

WARNING: Remove the protection foam from the springs before first use. This foam is added to prevent the springs from breaking during shipping, but it also stops the sound of the unit.

The architecture of the SPRING reverb unit offers the best of both worlds: digital delay and analog decay. Digital delay provides a simple, efficient solution for delay lines that retard or echo the signal before being sent to the spring. However, when it comes to the more complex matter of reverberation, springs still provide a more "natural" and musical solution. The SPRING unit includes six springs to give a rich reverb effect that also has a time-adjustable decay.

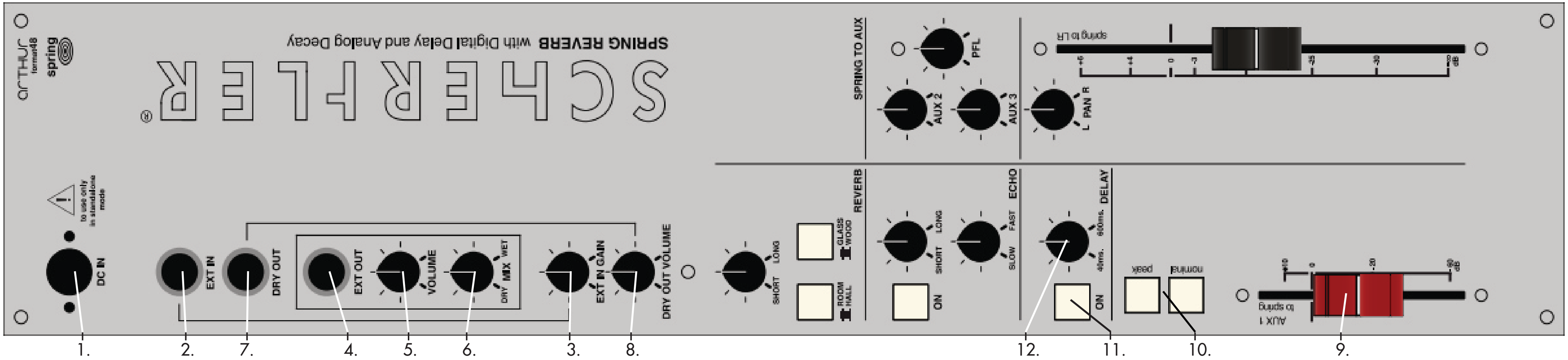
Positioning the SPRING unit

The SPRING unit has been configured to sum the signal from the AUX 1 bus. All input units mounted to the left of the SPRING will therefore flow to it. On its right side, the SPRING unit interrupts the AUX 1 bus, so the AUX 1 signal from any subsequent input units on this side will be summed by a second

SPRING reverb, or by the original AUX 1 master on the L/R MASTER unit. This enables you to create several independent AUX 1 lines.

As an example, working from left to right on the mixer, you could start with a MIC IN ULN unit, then add a SPRING unit followed by all the other input and output units. In this particular case, only the first mic unit (e.g. for a lead vocal) will benefit from the SPRING reverb. All other input units mounted to the right of the SPRING will be summed through the L/R MASTER (see Fig. 1) and will receive their reverb from another (external) effects unit. If you want to use the SPRING reverb on all the input units, you must place it right next to the L/R MASTER, on the left hand side.

Important: If you want your SPRING unit to be the final unit on the right side of the mixer - as many users prefer - you must first deactivate the summing amp on the L/R MASTER (see Fig. 2). This is done by depressing the blue button (see Fig. 3) on the L/R MASTER circuit board, which is positioned just under the fader next to the bus connector.



1. DC IN: SPRING is designed to function both as an ARTHUR mixer module and as a stand-alone reverb unit. Stand-alone use requires a PS-12 power supply to be connected to the DC IN. (When used as an ARTHUR mixer module, the SPRING unit is powered by the mixer's supply, via the bus.)

2. EXT IN: An external signal source such as a guitar pedal or computer can be connected here. The signal will then appear at the AUX 1 to SPRING fader for processing.

3. EXT IN GAIN: Adjustable from -5 dB to +20 dB, this gain control enables both line signals and weaker signals from pedals and other devices to be accommodated by the EXT IN (above).

4. EXT OUT: When used as a stand-alone unit, the SPRING's routing section (L/R fader, AUX 1/2 and PFL) will be out of action. So the unit's output is routed through the EXT OUT instead.

5. VOLUME and 6. MIX: The VOLUME control (5) sets the output level while the proportions of original sound (dry) and reverberated sound (wet) are set using the MIX control (6). Note: If the SPRING unit is connected to an external mixer, you might set the MIX knob to the right (wet) in order to add the original sound through your mixer's loop. If using the unit with guitar pedals, where no parallel loop is provided, the MIX control will then prove extremely helpful.

7. DRY OUT: SPRING includes a separate "AUX 1" output, referred to as the DRY OUT. Directly connected to the summed signal of the AUX 1 bus, it enables the AUX 1 signal to be picked up parallel to the SPRING unit, in order to drive an additional outboard reverb or other effects unit. These additional effects devices will return via the RETURN input on the L/R MASTER unit (mono), or via a separate STEREO IN or MULTIPLE IN unit.

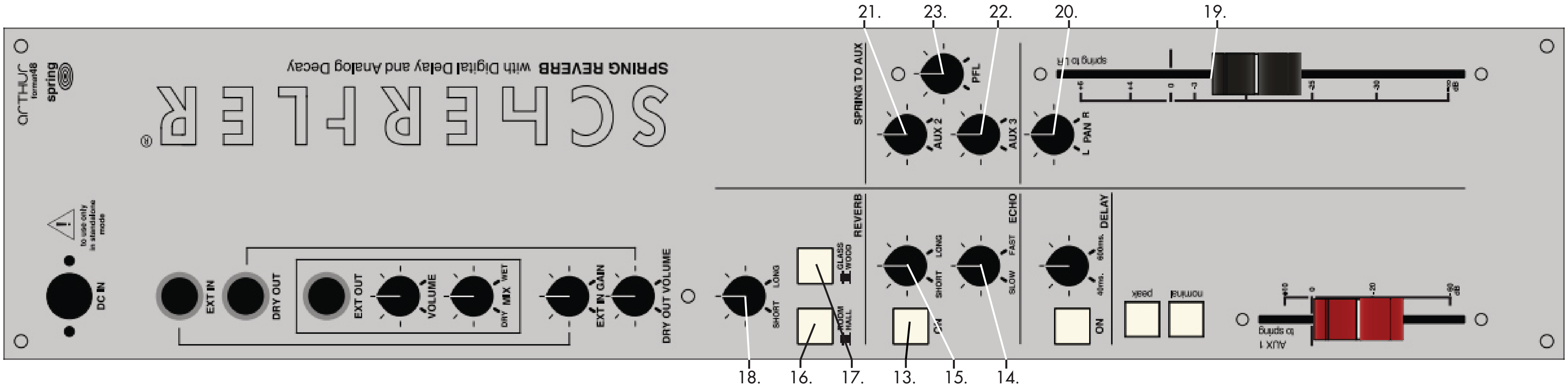
8. DRY OUT VOLUME: This controls the level of the DRY OUT signal (see 7 above).

9. AUX 1 to SPRING: SPRING receives and sums the AUX 1 from the bus. Setting the AUX 1 control on each individual input channel (e.g. MIC IN, YELLOW, STEREO IN) lets you send the desired amount of signal to the SPRING unit. This fader lets you control the amount of signal sent to the actual springs.

10. NOMINAL/PEAK VU: The two "nominal and peak" LED VU controls will help you determine the best position for the AUX 1 to SPRING fader. Note: Keeping the signal down on the fader creates a smoother reverberation effect with slightly longer delay. Pushing the signal on the fader makes the reverberation sound richer and shorter, but also slightly more metallic.

11. DELAY ON: Depressing the DELAY ON button (green light) lets you add a retard or delay before the signal is sent to the springs. This effect imitates the room acoustic of a concert hall or church, where an instant may pass before you hear reflections from the ceiling and walls (100 ms for example – see 12 below).

12. 20ms/200ms: This control reacts on the delay time, which is adjustable from 20 ms to 200 ms.



13. ECHO ON: Depressing this button (green light) activates the ECHO section. This puts a soft echo on the springs, imitating room acoustics with added echo effects such as a large bathroom. The section has four selectable drivers that influence the springs in different ways:

14. SLOW/FAST: Use this control to set the speed of the echo repetitions.

15. SHORT/LONG: Use this control to set the length of the echo before it fades away.

The REVERB section lets you choose between four different spring characteristics:

16. ROOM/HALL: These settings result in richer or more transparent reflections.

17. WOOD/GLASS: These settings result in warmer or harsher reflections.

18. SHORT/LONG: This makes the reverberation time longer as it is turned from left to right i.e. short to long. Aside from determining the amount of reverberated signal applied to the original, it also forms an important "artistic" parameter. In a ballad or spacey piece of music, a long decay will fit well giving depth to the overall sound. However, in a fast piece, a short decay will keep the result drier and less "confused".

19. SPRING to LR Fader: This sends the reverberated signal to the Master L and R outputs.

20. PAN: This alters the proportion of reverberated signal sent to the left or right channel.

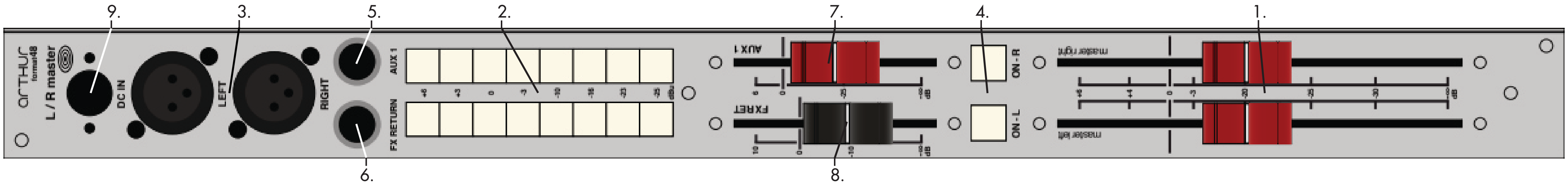
21. AUX 2 and 22. AUX 3: These controls regulate the amount of reverb signal sent to the respective auxiliary masters.

23. PFL: Here, the PFL has a slightly different function, being used more as a "volume" control. For example, if the PFL master output is feeding an additional stage monitor, this controllable PFL allows you to add a gradual amount of reverb to the monitor line.

SIGNAL FLOW

TECHNICAL INFORMATION

Weight	0.9 kg
Dimensions (LxDxH)	9.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	27 dB
Pad (attenuation)	No
Max input level	+22 dBu
Mic In connector	n.a.
Mic In sensitivity	n.a.
Mic In impedance	n.a.
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	1/4” jack balanced (EXT. IN)
Line In sensitivity	+8 to +24 dB
Line In impedance	48 kΩ
Main Out connector	1/4” jack balanced (EXT. OUT)
Maximum output level	22 dBu
Main Out impedance	270 Ω
Main Out freq. response	50 Hz to 50 kHz (EXT. OUT)
Aux Out freq. response	5 Hz to 50 kHz
Insert/Direct Out	No
Phantom power (nominal)	No
Stat power (10 VDC)	No
EQ	No
Filter	No
EIN	-102.2 dB
Distortion (THD+N @1kHz / 0 dBu output)	0.2%
Effect	Analog spring reverb with integrated digital delay (20 ms - 200 ms) + echo (40 ms “short” - 690 ms “long”); Room/Hall - Glass/Wood
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	5.8 W (120 mA)
Supply	Through L/R Master module and 50 VDC input (stand alone mode, w/ power supply)
Modular	Yes



1. L/R Master Faders: Depending on their position, these faders will attenuate or amplify the left and right signals from all the input units. The resulting signal levels can be seen on the VU meters (see 2 below).

2. VU meters: These meters show the amount of signal being sent to the LEFT and RIGHT Outputs (see 3 below). Note: ARTHUR features 30 dB of headroom, so its electronics and output amps are rarely likely to clip. The purpose of the VU meters is therefore to monitor the level of signal flow to successive audio devices.

3. LEFT and RIGHT Outputs (XLR): These send the audio signal to a PA system or recorder etc.

4. L/R ON: The Left and Right ON buttons connect the L/R faders with the output amps. They are similar to the Mute buttons used on other mixers, but their functionality is reversed. When the ON buttons are depressed, the outputs are connected (whereas a Mute button disconnects the output in its depressed position). These buttons can be useful in a live situation, to instantly disconnect “the stage from the PA” for example, without changing the position of the master faders.

5. AUX 1 OUT and 6. FX RETURN: The L/R MASTER unit has a single AUX SEND and FX RETURN enabling you to connect a reverb unit. This is very useful in a small mixer configuration combined with a small PA where, in most cases, reverb is required but is sufficient in a single loop mono setup. The reverb unit’s input will be connected to the AUX 1 OUT and its output to the FX RETURN. This way you can control the amount of reverb for every input unit (for example your mic units) using the channel’s AUX 1 control. This must be set to “post fade” (orange light on the post/pre button), making the reverb amount dependent on the channel fader position.

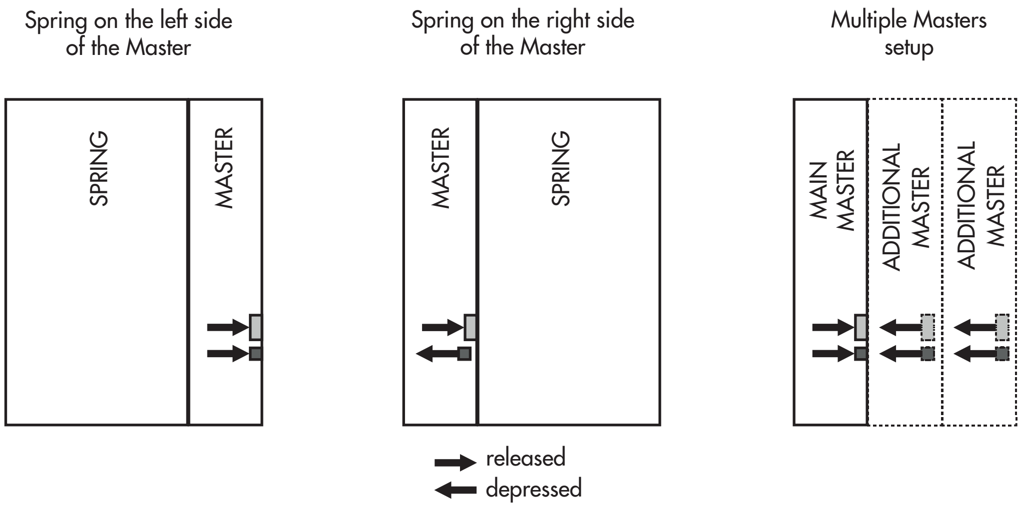
7. AUX 1 Fader: This controls the total amount of signal sent out to the external effect.

8. FX RET Fader: This controls the amount of reverb you will mix (in the same proportion) into to the left and right signals on the L/R MASTER outputs.

Note: If you are intending to use the SPRING reverb, the L/R MASTER’s AUX 1 section must be deactivated. This is because the reverb unit will sum all the signals coming from the AUX 1 bus. The L/R MASTER unit has a switch on the circuit board that deactivates the summing function of its AUX 1 section. Change this switch from its default setting (AUX 1 on) to AUX 1 off. In this position, AUX 1 on the L/R MASTER unit will stop running and the reverb unit will “take over” the AUX 1 signals from all channels. (You can still pick up the summed AUX 1 signal from the reverb unit for external use.)

9. DC IN: The mixer’s power supply is connected here. From this 48 VDC input, all other units attached to the L/R MASTER will also be served via the power supply. Please see p.8 for more information on power supplies.

MULTIPLE OUTPUTS



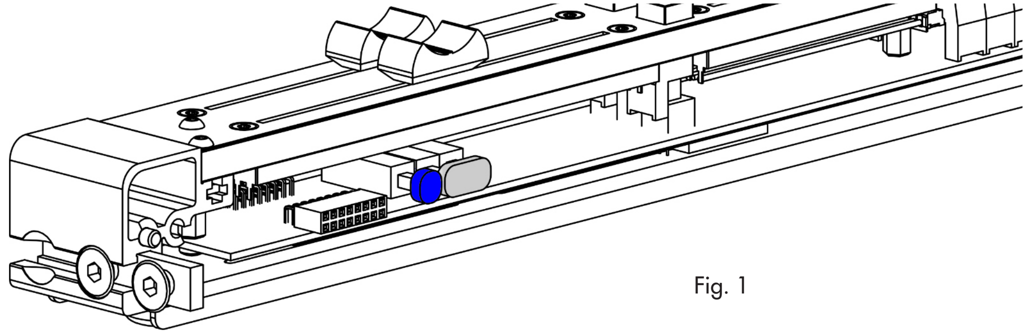


Fig. 1

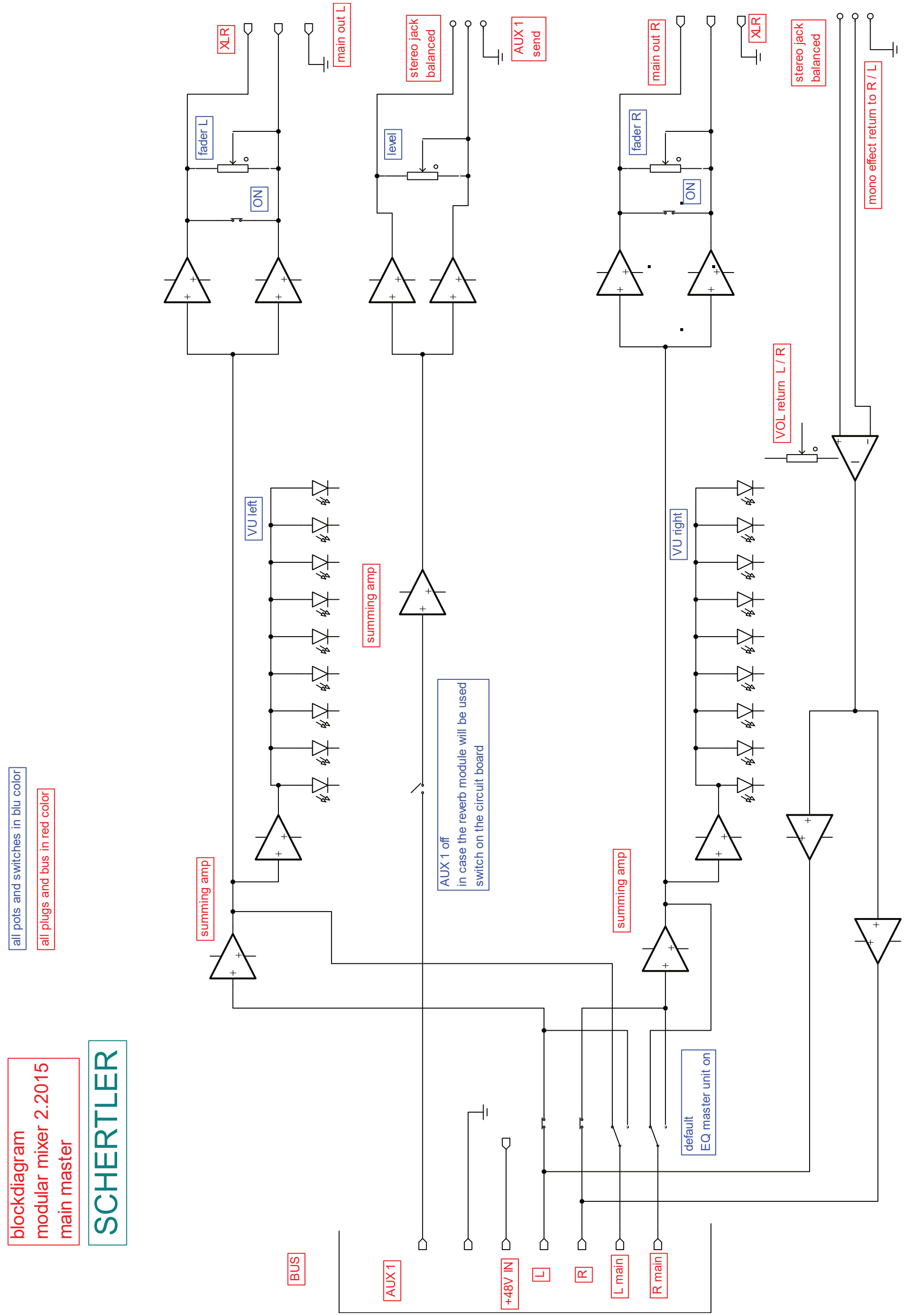
A standard mixer used for live and recording applications only needs one master L/R for connection to a PA system or recording device, for example. However, there are many situations in which more than one L/R output is not only useful but essential.

ARTHUR's flexible modular concept enables the combination of a quasi-infinite number of outputs. In a hotel audio installation for example, a microphone and a stereo input might be sufficient for inputting the sound sources, but more outputs would be needed for it to be independently controllable in different areas such as the bar, restaurant or lobby. Another example could be a "splitter mixer" for audio demonstration, with only one stereo input unit (for the audio source) and numerous L/R MASTER units for sending the audio signal to different loudspeakers for A/B comparison. Even on a standard mixer, a second master output could be helpful for controlling a subsidiary part of the PA system, like side fills for example.

The mixer's structure only allows for one summing amp, which will sum all left and right signals coming from the bus. The L/R MASTER unit features a white button - positioned on the circuit board (see Fig. 1) to deactivate that summing function. If you use more than one L/R MASTER unit, only one unit should be left with its default setting (i.e. with summing on); all the other L/R MASTER units must have both buttons depressed.

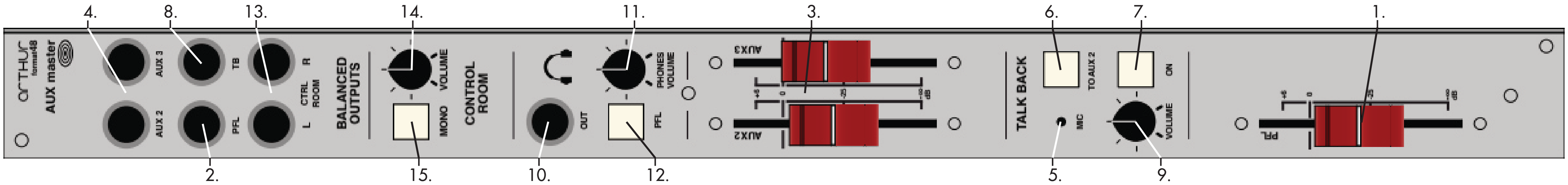
Note: If you want to add a SPRING reverb to the right of the Master, forming the final unit on the right of the mixer as many users prefer, you must first deactivate the summing amp on the L/R MASTER. Do this by depressing the blue switch on the L/R MASTER circuit board.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	6 dB
Pad (attenuation)	No
Line In connector	1/4" jack balanced (FX RETURN)
Line In sensitivity	+22 dBu
Line In impedance	20 kΩ
Main Out connector	2x XLR
Maximum output level	31 dBu
Main Out impedance	120 Ω
Main Out freq. response	16 Hz to 125 kHz
Control Room connector	n.a.
Control Room level	n.a.
Control Room impedance	n.a.
Control Room distortion (THD+N, @1 kHz)	n.a.
Control Room freq. response	n.a.
Aux Out connector	1/4" jack balanced
Aux Out level	22 dBu
Aux Out impedance	180 Ω
Aux Out distortion (THD+N, @1 kHz)	0.16%
Aux Out freq. response	16 Hz to 125 kHz
PFL connector	n.a.
PFL level	n.a.
PFL impedance	n.a.
PFL distortion (THD+N, @1 kHz)	n.a.
PFL freq. response	n.a.
Phones connector	n.a.
Phones level	n.a.
Phones impedance	n.a.
Phones distortion (THD+N, @1 kHz)	n.a.
Phones freq. response	n.a.
Insert/Direct Out	No
Insert/Direct Out level	n.a.
Distortion (THD+N @1kHz / 0 dBu output)	0.04% 0.13% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	no
Power consumption	6.2 W (130mA)
Supply	50 VDC input - w/ power supply
Modular	Yes



1. PFL Fader: All signals from any input units that have their PFL buttons depressed (red light), will be controlled via the PFL fader and sent to the output via the PFL OUT (see 2 below). The PFL function of the headphones is not affected by this fader.

2. PFL OUT: PFL signals going to the output are routed via the PFL OUT. This particular function can be useful when an additional monitor is needed, for example for a singer, where just the signal from the "voice channel" is needed.

3. AUX 2/AUX 3 Faders and 4. AUX 2/3 OUT: The AUX 2 and AUX 3 faders control the amount of signal on the AUX 2 OUT and AUX 3 OUT respectively.

5. TALKBACK MIC port: The AUX MASTER unit includes a TALKBACK section with an integrated condenser microphone. This is designed to ease the sound engineer's communication with the stage or recording room.

6. TO AUX 2: The preamplified signal from the TALKBACK microphone can be sent directly to the AUX 2 OUT (see 4 above) by depressing the TO AUX 2 button. It will then be added to the AUX 2 signal heard in the stage monitors for example. (Therefore, use AUX 2 for stage monitors.)

7. TB ON: Depressing this button activates the Talkback section (yellow light).

8. TB OUT: The preamplified signal from the microphone can be connected from the TB OUT to any (active) loudspeaker system in the recording room.

9. VOLUME (Talkback): This dedicated control lets you set the correct level of talkback. (Don't forget to deactivate the TB On button when you have finished speaking!)

10. HP OUT and 11. PHONES VOLUME: Connect your headphones to the HP OUT (10) and adjust the VOLUME control (11) to a comfortable sound level. You will hear the signal from the L/R MASTER independently from the signal level determined by the position of the master faders.

12. PFL to HP: When you depress the PFL to HP button (red light), the PFL signal will appear on the headphones, provided that the PFL button is depressed on at least one of the input channels. This function can be helpful when listening to a single input channel, or when listening to it before fading it to the main outputs (by having deactivated the channel UNIT ON button, or having pushed down the channel fader).

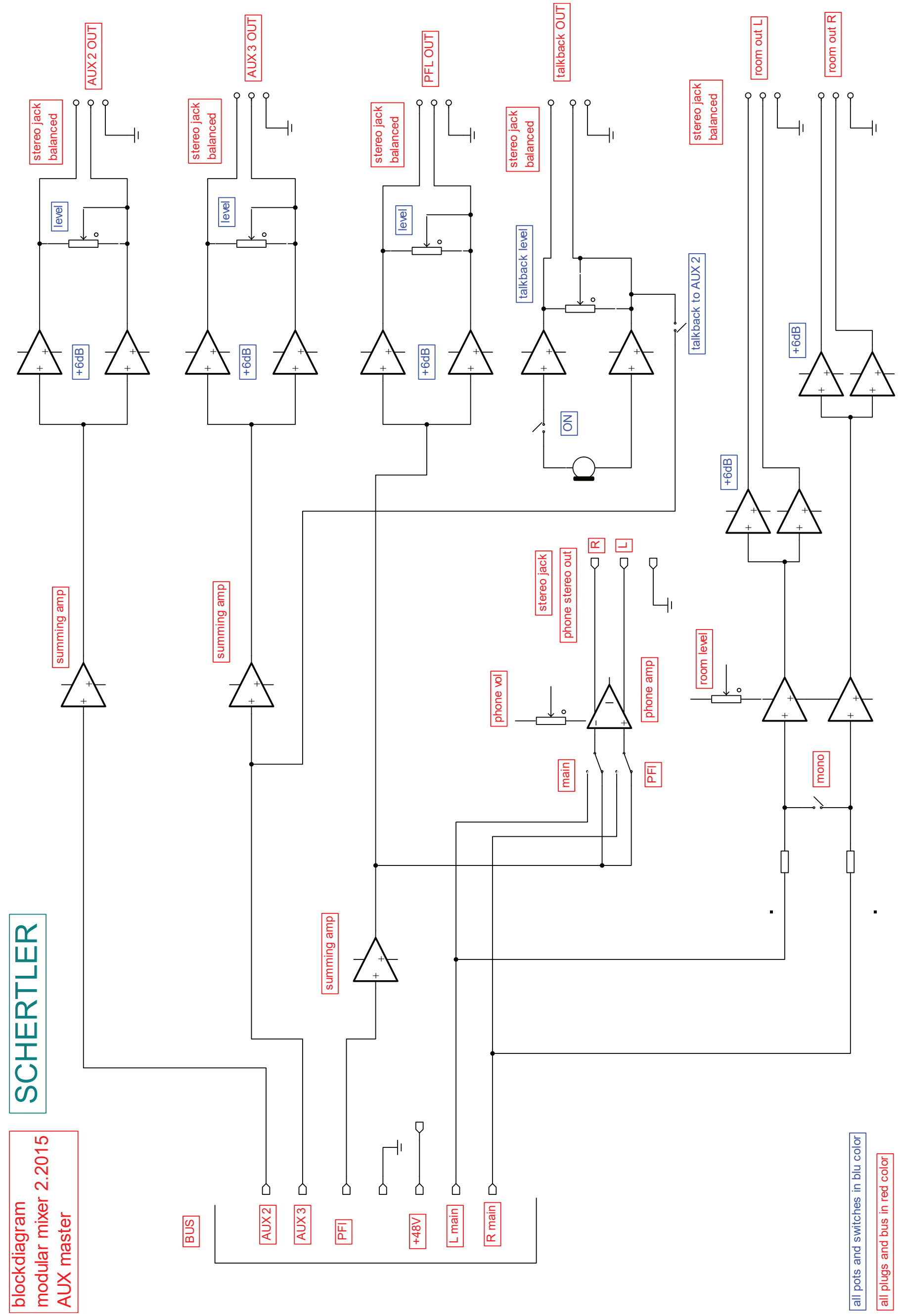
13. LEFT/RIGHT CONTROL ROOM: The control room section is an additional stereo L/R output that can be used to drive devices such as room monitors in a recording studio. Other devices can also receive a stereo signal from this output, e.g. a subsidiary PA, side fills, a second recording device etc. Plug the active studio monitors (or the amp for driving passive monitors) into the LEFT and RIGHT outputs. Note: these outputs (along with the AUX 2/3 OUT, PFL OUT and TB OUT) are designed to send balanced signals using ¼" jacks. The "tip" connects as usual to the hot signal, the "ring" to the cold signal and the "sleeve" to the ground. An unbalanced signal can also be connected to these outputs via a mono jack.

14. VOLUME (Control Room): This control is used to set the volume levels of the L/R CONTROL ROOM outputs (above).

15. MONO: This button enables you to listen to a stereo program that has been reproduced in mono, in order to evaluate its "mono compatibility".

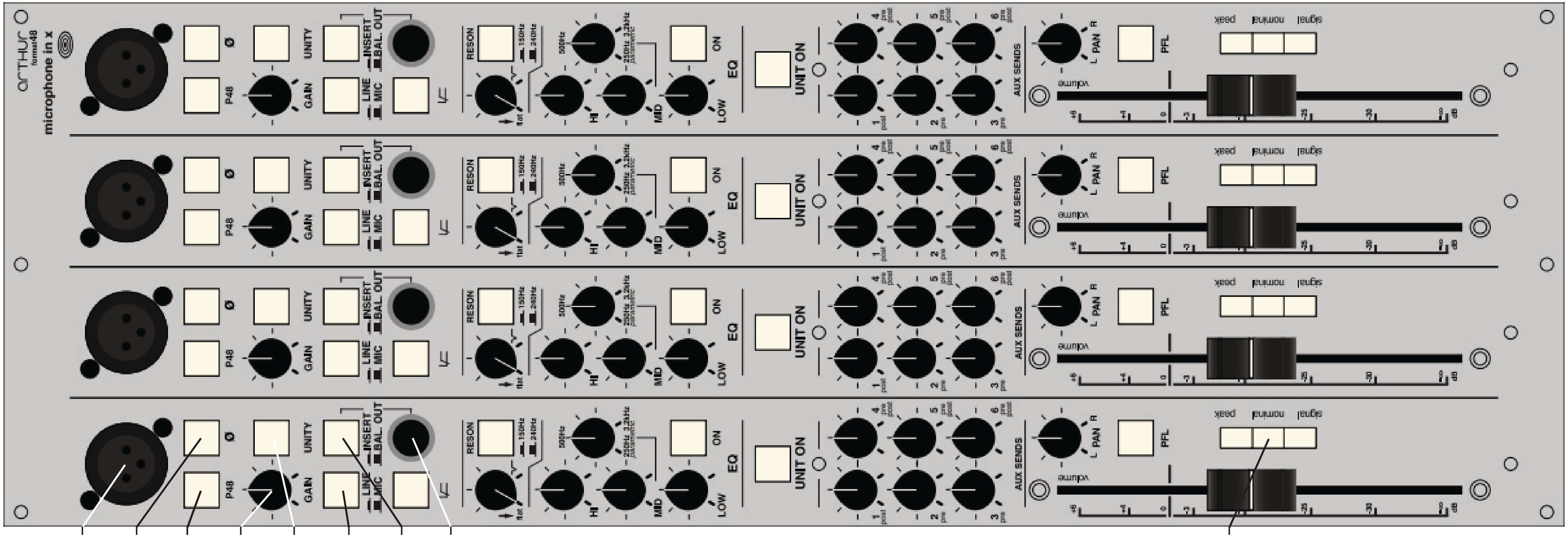
Practical tip: It makes sense to install the AUX MASTER unit to the right of the L/R MASTER unit. This way the AUX faders will go from left to right in numeric order (AUX 1 on the L/R MASTER, AUX 2 and AUX 3 on the AUX MASTER).

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	0.5 kg
Dimensions (LxDxH)	3.6 x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	6 dB
Pad (attenuation)	n.a.
Line In connector	n.a.
Line In sensitivity	n.a.
Line In impedance	n.a.
Main Out connector	n.a.
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	n.a.
Control Room connector	1/4" jack balanced
Control Room level	25 dBu
Control Room impedance	200 Ω
Control Room distortion (THD+N, @1 kHz)	0.12%
Control Room freq. response	20 Hz to 60 kHz
Aux Out connector	1/4" jack balanced
Aux Out level	26 dBu
Aux Out impedance	180 Ω
Aux Out distortion (THD+N, @1 kHz)	0.06%
Aux Out freq. response	16 Hz to 125 kHz
PFL connector	1/4" jack balanced
PFL level	25 dBu
PFL impedance	180 Ω
PFL distortion (THD+N, @1 kHz)	0.05%
PFL freq. response	16 Hz to 100 kHz
Phones connector	1/4" jack balanced
Phones level	25 dBu
Phones impedance	>32 Ω
Phones distortion (THD+N, @1 kHz)	0.12%
Phones freq. response	16 Hz to 50 kHz
Insert/Direct Out	No
Insert/Direct Out level	n.a.
Distortion (THD+N @1kHz / 0 dBu output)	n.a.
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	3.2 W (67 mA)
Supply	Through L/R Master module
Modular	Yes (only one possible in each ARTHUR)



The MIC/LINE X's four input channels are identically equipped.

1. XLR input: This input can receive balanced signals from -63 dBu to +6 dBu in MIC mode - a range of 69 dB, which therefore permits you to connect any audio signal to the mic unit. (Also see 6 and 7 below.)

2. GAIN: Adjusting the Gain affects the amplification rate of the input amplifier. A weaker signal is amplified to a nominal level of 0 dBV and a stronger signal is attenuated so that a nominal signal level of 0 dB is always present at the output of the mic input amp. (Also see 6 and 7 below.)

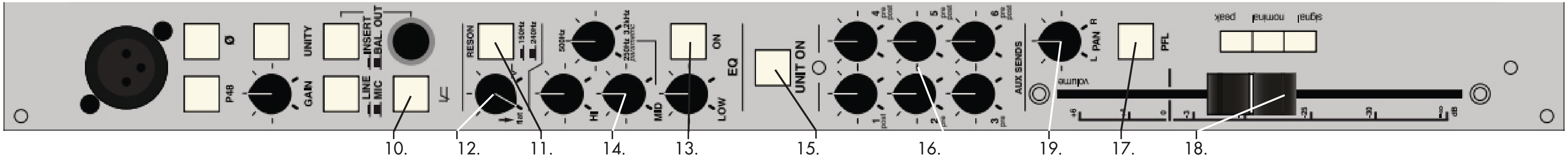
3. VU METER: This lets you "read" the amount of gain set. Turn up the GAIN to a point where the red peak light occasionally shows. But don't worry too much about this. Thanks to the amount of headroom from input to output on ARTHUR, even strong overloads can be absorbed by the mixer's electronics without resulting in distortion.

4. P48: In MIC mode, depressing the P48 button (red light), delivers 48 VDC of phantom power to the microphone, which, in most cases will be a condenser or active ribbon type. A dynamic mic cannot normally "see" phantom power (as the name suggests), but passive ribbon mics could be permanently damaged by it. Only use this button with mics that you definitely know require phantom power in order to work. Note: The internal circuitry raises the 48 VDC slowly to avoid "pops" and to protect the microphone. Therefore, allow a few seconds for the mic to be working fully.

5. PHASE (ø): This swaps the "hot" and "cold" aspects of the input signal, inverting its phase by 180°. It can be helpful if, for example, two microphones are positioned at a distance, or reverse- faced (i.e. as with bottom and top snare drum miking). Note: The button use is subjective and the results really need to be evaluated through listening.

6. UNITY and 7. LINE/MIC: The LINE/MIC (gain) button (7) adapts the signal processing to the nature of the input signal. When this button is set to LINE (green light), the impedance is automatically adapted for line signals and the 48 VDC phantom power is deactivated to prevent any damage to the "line" device connected. When the channel is switched to LINE mode, the sensitivity range adapts from -21 dBu to +8 dBu. You can automatically set the nominal level at 0 dBV by activating the UNITY button (6).

8. INSERT/BALANCED DIRECT OUT and 9. INSERT: The INSERT (9) works in a similar way to inserts on other mixers. However, a related "bypass" button (8) offers additional possibilities. When this is depressed, the signal in the mixer is interrupted and the INSERT works in the usual way. By connecting a



mono jack, you get the output line signal on the “tip”. By connecting a standard stereo jack, you get the (output) signal from the “tip” (send) and the return signal will be connected to the mixer through the “ring” (return) of the jack. When the button is not depressed, the signal will not be interrupted by the insertion of a jack into the INSERT. Here, the insert connection works as a sleeve out or “dry line out post input amp”. You can connect a mono or stereo jack to the INSERT. The unbalanced line signal will be transmitted through the “tip” of the jack. Red light = depressed (Insert functionality active.) White light = not depressed (Insert functionality bypassed).

10. LOW CUT: This filter limits low frequencies at 100 Hz/second order, cutting out unwanted frequencies from “boomy” signals. It can also be helpful in shaping signals from smaller instruments (violin, mandolin etc).

11. RESON: The resonance filter is a kind of notch filter, but one that is gradually adjustable over its attenuation level. It is designed to avoid (or at least attenuate) feedback on acoustic instruments that are miked up in live situations using pickups, e.g. the SCHERTLER DYN Series. A double bass or ‘cello might get in resonance at ca. 150 Hz, whereas guitars, violins and similar musical instruments will do so at ca. 240 Hz. The Q is very high, cutting out a very narrow band at the respective frequency.
If the RESON control is set to the FLAT (far left) position it will not be active, thus having no effect on the incoming signal. When the control is turned slowly clockwise, the filter will gradually attenuate at the chosen frequency (see 12 below).

12. 150Hz-240Hz: This lets you select the frequency. When the button is not depressed (red light), the filter will attenuate at 150 Hz. When the button is depressed (blue light), it will attenuate at 240 Hz.

13. EQ ON: This button activates or bypasses the EQ section (green light when activated/depressed, white light when deactivated). It can be useful for comparing an EQ configuration with the unfiltered sound. Note: The EQ ON button will not bypass the RESONANCE filter.

14. EQ – HI/MID/LOW: The HI control lets you tune the high range of the audio spectrum (from 4 kHz) by +/- 14 dB with a slope of 18 dB / octave. The 3rd order shelving filter “keeps” the circle of influence within the filter’s audio band so as not to overlap with the MID filters. This makes adjustment of the higher frequencies more accurate.

The MID control, together with the MID FREQ control, acts on frequencies within a wide mid range of 250 Hz to 3 kHz, with amplification or attenuation of +/- 12 dB. (The MID control affects the amplitude - amplification or attenuation - while the MID FREQ control affects the frequency.)

The LOW control lets you adjust the signal by +/-16 dB up to 110 Hz with a slope of 12 dB / octave.

The higher order of the shelving filter prevents the low frequencies from overlapping with the parametric mid, making adjustment of the lower frequencies more accurate.

The HI, MID and LOW controls all have a detent at their mid positions.

This indicates the filter’s “flat” position.

15. UNIT ON: This connects/disconnects the channel’s output routing for all outputs (AUX1 to AUX6 and L/R) except the PFL routing. Its function is similar to the Mute button used on other mixers, but here the functionality is reversed. When the button is depressed, all outputs get connected - whereas a Mute button disconnects the output when pressed. Also, whereas a Mute button normally only disconnects the L/R routing i.e. the channel’s fader, this button affects all outputs. Being able to switch off a channel strip makes sense in order to prevent the signal from still going through to stage monitors, or to the input of the reverb unit for example. Note: Even if the button is in the switched off position (not depressed, white light), the PFL and INSERT will still be ready to function, (as shown by the lights on their respective buttons) even though all other button lights are off.

16. AUX SENDS: There are six rotary controls – one for the level control of each send (AUX1 to AUX 6). This enables the individual AUX SENDS to be controlled independently from one another.

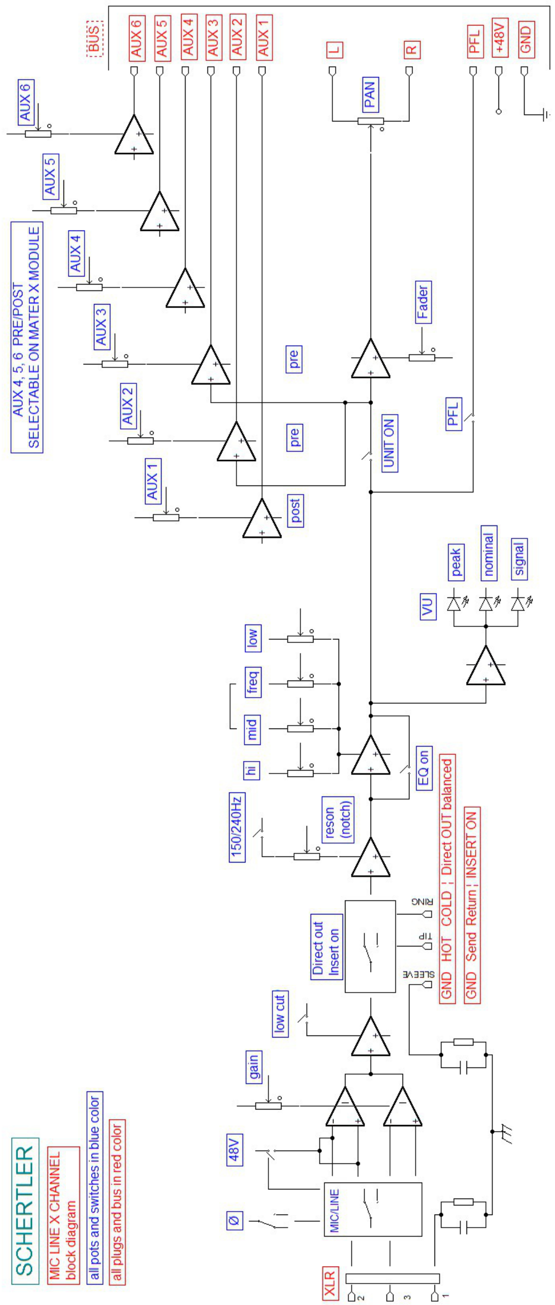
The signal to AUX1 is “taken” post fade because this auxiliary line can be used with the SPRING unit for reverb effect, while AUX2 and AUX3 are pre fade for use with stage monitors. The other three AUX can be independently assigned pre- or post fade on the MASTER X unit. This means you can have an AUX configuration of up to one post and five pre (1 effect + 5 monitors) or up to four post and two pre (4 effects + 2 monitors). The level of the AUX SENDS can be read on the MASTER X unit’s dedicated VU meter.

17. PFL: As well as being a traditional PFL (pre fade listen), the PFL button also serves as an additional AUX send, albeit without the possibility to set any levels. All channels that have their PFL activated (red light) will be mixed in the MASTER X unit and sent to the PFL output, controlled by the PFL fader and “visible” on the MASTER X’s VU meter. This function can be useful if you have an additional monitor and only need one signal, e.g. for a singer who only requires the “voice channel”. Note: The PFL section still runs if the UNIT ON is deactivated (white light).

18. Channel Fader: This controls the total amount of signal going to the master. To exclude this signal from the MASTER X without changing the fader position, simply switch the UNIT ON button to its “off” position (white light). The button will then act as a Mute.

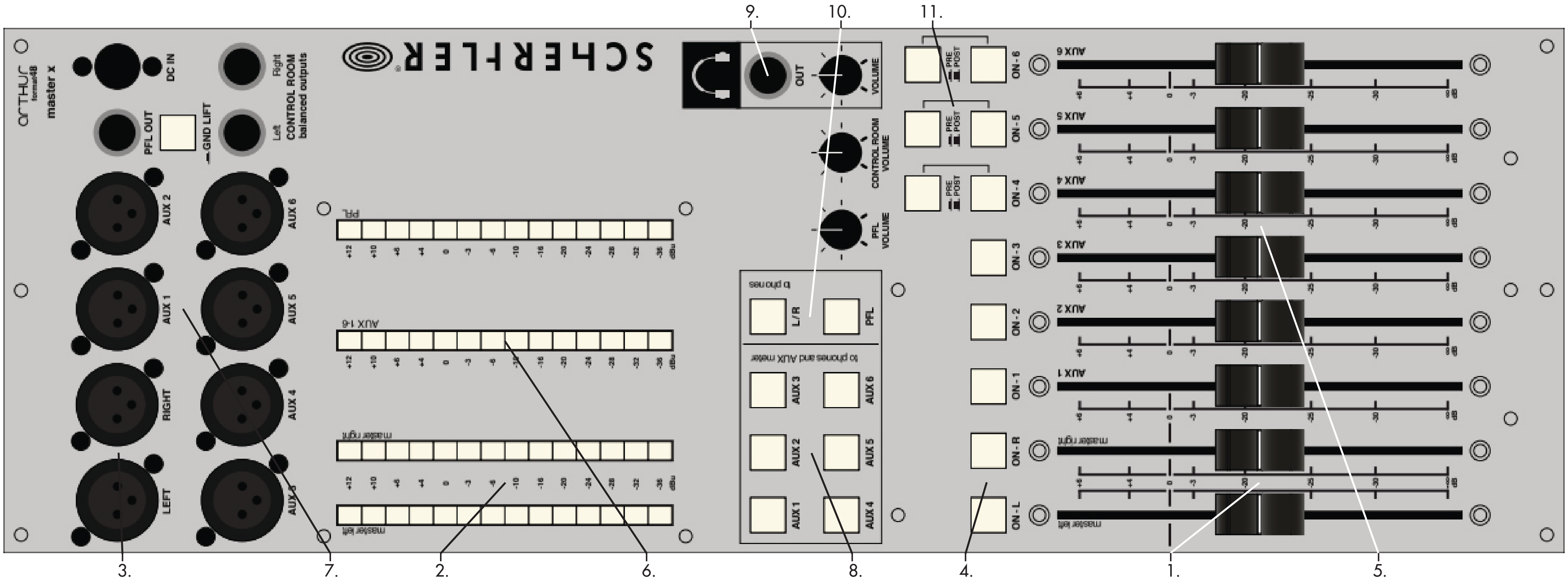
19. PAN: The PAN control lets you send the signal to the left or right channel. Its configuration is designed to guarantee minimum noise and maximum dynamics in the central position.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	1.75 kg
Dimensions (LxDxH)	14.3x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	72 dB
Pad (attenuation)	n.a.
Max input level	+24 dBu (Line mode)
Mic In connector	XLR
Mic In sensitivity	-63 to +6 dBu
Mic In impedance	4.6 kΩ
Instrument In connector	n.a.
Instrument In sensitivity	n.a.
Instrument In impedance	n.a.
Line In connector	XLR
Line In sensitivity	-21 to +8 dBu
Line In impedance	43 kΩ
Main Out connector	through MASTER X module
Maximum output level	31 dBu
Main Out impedance	n.a.
Main Out freq. response	<10 Hz to 55 kHz
Insert/Direct Out	1/4″ stereo jack / balanced
Insert/Direct Out level	26 dBu
Phantom power (nominal)	48 VDC
Stat power (10 VDC)	No
EQ	
Low	Shelving, ±16 dB (@110 Hz)
Mid	Parametric, ±12 dB (250 to 3 kHz)
High	Shelving, ±14 dB (@4 kHz)
Filter	Reson (notch): -10 dB (@150 Hz - @240 Hz) Low Cut: shelving, 2nd order (cut freq. 100 Hz)
EIN	-129.4 dB
Distortion (THD+N @1kHz / 0 dBu output)	0.041% 0.04% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	No
Power consumption	20.1 W (420 mA)
Supply	Through MASTER X module
Modular	Yes



1. LEFT/RIGHT Master Faders: Depending on their position, these faders attenuate or amplify the left and right signals from all the input units. The resulting signals can be seen on the VU meters (see 2 below).

2. VU METERS: These show the amount of signal being sent to the LEFT and RIGHT Outputs (see 3 below). Note: ARTHUR features 30 dB of headroom, so its electronics and output amps are rarely likely to clip. The purpose of the VU meters is therefore to monitor the level of signal flow to successive audio devices.

3. LEFT and RIGHT Outputs (XLR): These send the audio signal to a PA system or recorder etc.

4. L/R ON: These buttons connect the L/R and AUX faders with the output amps. They are similar to the Mute buttons used on other mixers, but their functionality is reversed. When the ON buttons are depressed, the outputs are connected (whereas a Mute button disconnects the output in its depressed position). These buttons can be useful in a live situation, to instantly disconnect "the stage from the PA" for example, without changing the position of the master faders.

5. AUX Master Faders: This upgrade to the standard L/R MASTER unit offers six AUX master faders that attenuate or amplify the AUX signals from all input units. The resulting signals can be seen on the AUX 1-6 METER (see 6 below).

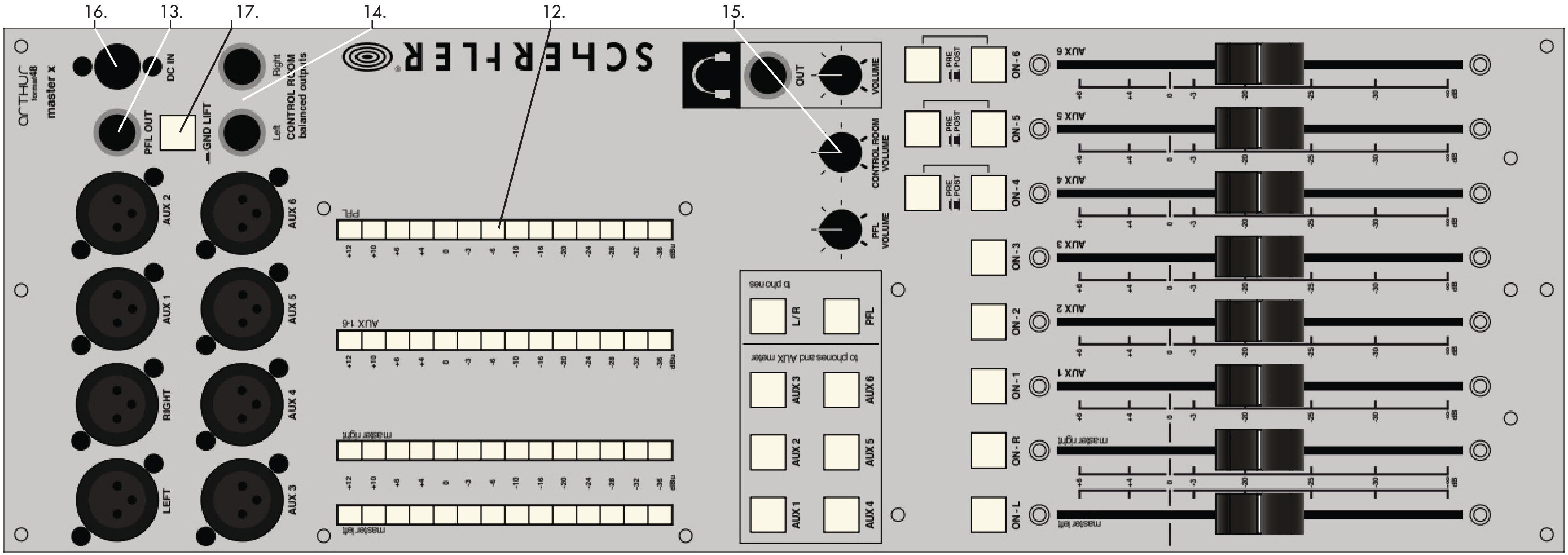
6. AUX 1-6 METER: This VU meter shows the amount of signal being sent to the AUX 1 - AUX 6 output connectors (see 7 below).

7. AUX Outputs (XLR) and 8. Selector buttons: From these AUX outputs, signal is sent to external effects or stage monitors. You can choose which AUX signal is seen on the VU meter by selecting the relevant AUX TO PHONES and AUX METER button. These buttons are also used to select which AUX signal can be sent to the PHONES output (see 9 below). This enables more than one AUX signal to be selected at the same time.

9. PHONES output: High-quality headphone preamp with 1/4" jack connector.

10. TO PHONES: The L/R and PFL TO PHONES buttons activate the relevant audio signal to be sent to the PHONES output.

11. AUX 4-6 PRE/POST buttons: These switch the pre/post fade positions of AUX 4-6. The pre/post fade status of AUX 1-3 depends on the input modules connected to the MASTER X:
- using the standard input modules (e.g. MIC IN ULN, YELLOW, STEREO IN), the pre/post position can be selected on the input module itself.



- using the MIC LINE X input module, AUX 1 is set post fade while AUX 2 and 3 are set pre fade. None of these can be switched.
Note: The MASTER X unit features a switch (positioned on the circuit board) that deactivates the summing function of its AUX 1 section. If you change this switch from its default setting (AUX 1 on) to AUX 1 off, AUX 1 on the MASTER X unit will stop running and the SPRING reverb unit will “take over” the AUX 1 signals from all channels. In this situation, you can however still pick up the summed AUX 1 signal from the reverb unit for external use.

The combination of these switches allows you to connect up to one SPRING (post fade) with five stage monitors (pre fade), or up to one SPRING with three other external effects (post fade) and two stage monitors (pre fade on AUX 2 and 3).

12. PFL VU meter: This shows the sum of the PFL signals activated on the input modules.

13. PFL OUT: 1/4” balanced jack output connector.

14. CONTROL ROOM Outputs: 1/4” jack balanced L/R output connectors.

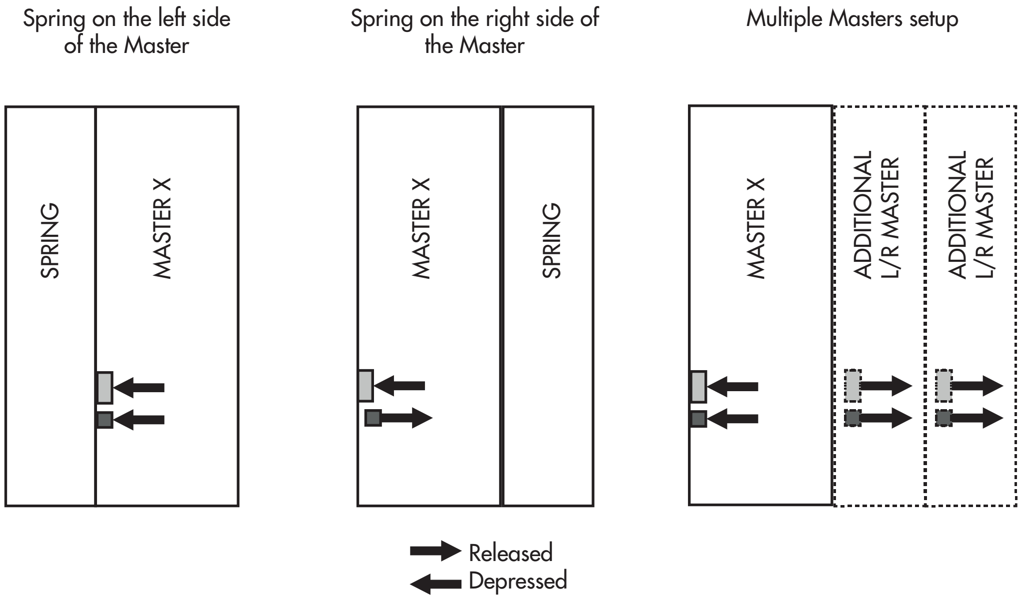
15. PFL, CONTROL ROOM and PHONES VOLUME: These controls regulate the output volume from the PFL OUT (13), CONTROL ROOM VOLUME (14) and PHONES output (9) respectively.

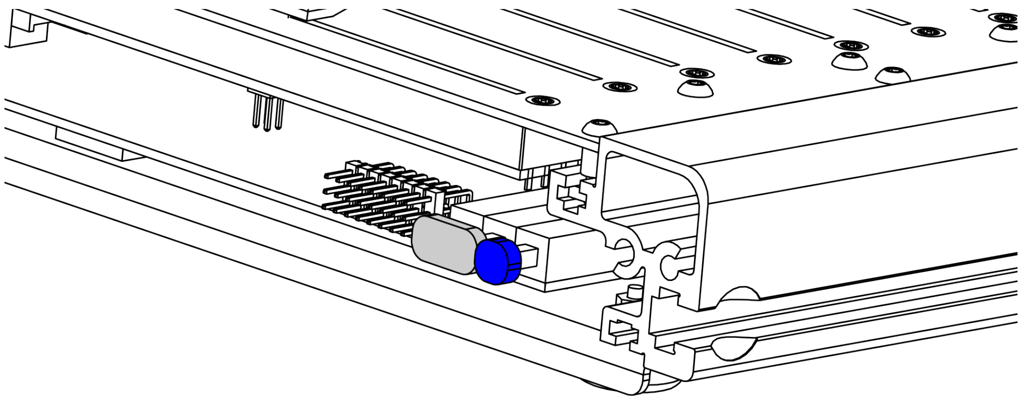
16. DC IN Connector: The mixer’s power supply is connected here. From this 48 VDC input, all other units attached to the MASTER X will also be served by the power supply. (For more information, please see the SCHERTLER website/Pro Audio/Modular mixer/Accessories).

17. GROUND LIFT: Sometimes the shielding of the audio cables can be connected to ground through the other devices connected to the mixer. However, to avoid any possible ground loop noises, the GROUND LIFT button disconnects the ground from all audio output ground connector contacts (XLR and jacks). If you have customized your MASTER X with transformers on the outputs (see the section on Customization) it is better to always keep the ground connection lifted in order to get the most “noiseless” audio signal.

CUSTOMIZATION
Thanks to its innovative ground connection system, MASTER X’s default configuration has electronically balanced outputs. However, as an optional extra it is also possible to customize the output sound by equipping each of the L/R MASTER and AUX 1-6 XLR outputs with a Lundahl transformer. If you have bought your MASTER X without any transformers, or perhaps with just a few, you can always send the module back to SCHERTLER and upgrade it with as many transformers as you need. For example, if you know that some of your stage monitors require the transformers and others don’t, you may just want to equip selected AUX lines with the transformers, as opposed to all six.

MULTIPLE OUTPUTS



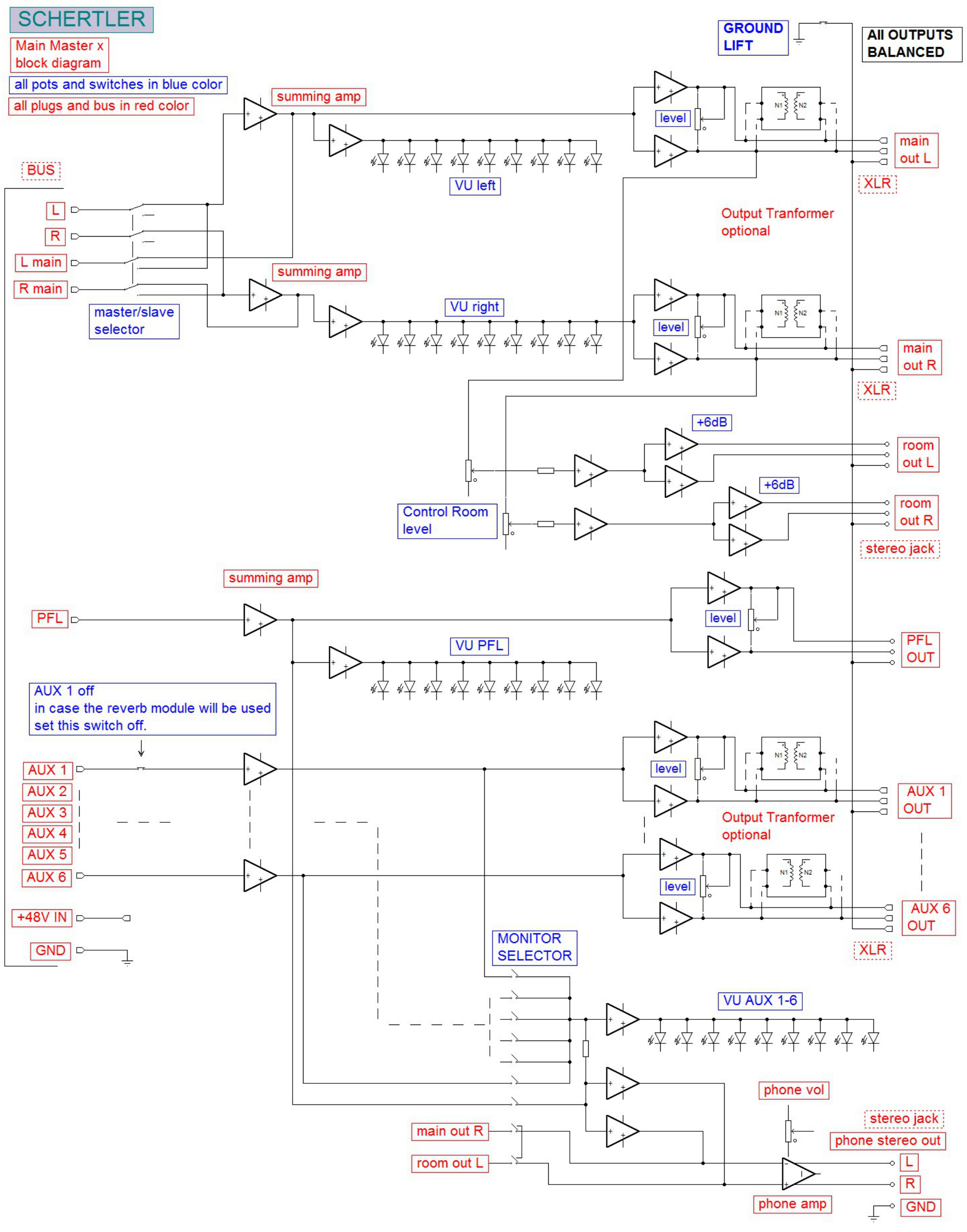


As shown in the L/R MASTER section, there are many situations in which more than one L/R output might be required. Even if only one MASTER X module can be mounted on a mixer, it is possible to add multiple L/R MASTER units to it.

The mixer's structure only allows for one summing amp, which will sum all left and right signals coming from the bus. The MASTER X unit has a white button positioned on the circuit board (see Fig. 1) for deactivating that summing function. If you use more than one master module, only one unit should be left with its default setting (i.e. with summing on); all the other master modules must have both buttons depressed.

Note: If you want to add a SPRING reverb to the right of the Master, forming the final unit on the right of the mixer as many users prefer, you must first deactivate the summing amp on the MASTER X. Do this by depressing the blue switch on the MASTER X circuit board.

SIGNAL FLOW



TECHNICAL INFORMATION

Weight	1.65 kg
Dimensions (LxDxH)	14.3x 5.8 x 47.5 cm
Construction	Anodized aluminum assembled box
Total gain	6 dB
Pad (attenuation)	n.a.
Line In connector	n.a.
Line In sensitivity	n.a.
Line In impedance	n.a.
Main Out connector	XLR
Maximum output level	31 dBu
Main Out impedance	120 Ω
Main Out freq. response	<10 Hz to 50 kHz
Control Room connector	1/4" jack balanced
Control Room level	30 dBu
Control Room impedance	200 Ω
Control Room distortion (THD+N, @1 kHz)	0.12%
Control Room freq. response	<10 Hz to 50 kHz
Aux Out connector	XLR
Aux Out level	30 dBu
Aux Out impedance	150 Ω
Aux Out distortion (THD+N, @1 kHz)	0.075%
Aux Out freq. response	<10 Hz to 50 kHz
PFL connector	1/4" jack balanced
PFL level	24 dBu
PFL impedance	180 Ω
PFL distortion (THD+N, @1 kHz)	0.05%
PFL freq. response	<10 Hz to 50 kHz
Phones connector	1/4" jack balanced
Phones level	14.5 dBu
Phones impedance	>32 Ω
Phones distortion (THD+N, @1 kHz)	0.05%
Phones freq. response	<10 Hz to 50 kHz
Insert/Direct Out	No
Insert/Direct Out level	n.a.
Distortion (THD+N @1kHz / 0 dBu output)	0.075% 0.08% (2nd harmonic)
Effect	No
Preamp	Class-A, no negative feedback, no integrated circuits
Audio transformer	optional
Power consumption	18.2 W (380 mA)
Supply	50 VDC input - w/ power supply
Modular	Yes (only one possible in each ARTHUR)

FAQ

Where can I buy a case for my ARTHUR mixer?

ARTHUR cases, along with other SCHERTLER product accessories, are available from the online web shop at www.schertler.com. You can buy the product itself and/or its accessories from the relevant product page. Three case models are available for mixers with up to 10, 20 and 30 modules respectively.

What is the purpose of secondary L/R MASTER modules?

A L/R MASTER module set as a slave allows you to multiply the Master outputs, the control room and FX RETURN input as well as providing power for a larger number of modules by connecting an extra power supply. An exception is made for the AUX 1 output, which is completely cancelled on the secondary L/R MASTER module (in order not to create conflict with the main L/R MASTER module). This can be useful for secondary listening, for example between various stage areas such as side- or front fields, or in the studio where one L/R MASTER is connected to the recorder and others are connected to different pairs of reference monitors. It can also be useful in environments such as hotels and restaurants for independent control of PA systems in various rooms.

How do I set a L/R MASTER as a slave?

To set a L/R MASTER module as a slave you need to press two buttons that are mounted on the electronics board under the faders. Because these buttons are not accessible from the outside, this operation must be performed before assembling the mixer modules, or when the unit is not connected to any others.

What is the maximum number of available AUX SENDS?

On standard input modules there are 3 AUX SENDS, each of which can be set pre or post fade directly on the module. X SERIES modules (MIC LINE X and MASTER X) offer 6 AUX SENDS, preserving complete compatibility with standard modules.

How do I choose the right power supply for my ARTHUR?

The choice of power supply largely depends on the number of modules in your ARTHUR mixer configuration. Two switching power supply models are available, which can supply up to 12 or 25 modules respectively. There are also two professional models with linear transformers, able to supply a greater number of modules, or with a lower noise level.

Do I need to use a DI with the YELLOW module?

This instrument input is designed to receive the signal directly from the instrument via a standard jack cable, so it is not necessary to use a DI. Also, because this input module is designed for small to medium-size situations, for example where the mixer is managed directly by the musician on stage (without the need for a long cable run), or where an instrument is recorded in a studio control room directly connected to the mixer, a DI would not be needed here.

What is the difference between the MIC IN ULN and MIC LINE X?

As well as a larger number of AUX SENDS, the MIC LINE X has two independent preamplifiers for mic and line signals. In addition to the different sensitivity and impedance, when the preamp is set to LINE the P48 function is also disabled to avoid possible damage to the connected device if the phantom power is left on.

APPENDIX

WARRANTY
All SCHERTLER products are covered by a limited two-year factory warranty in respect of manufacturer defects. Details can be obtained from your local dealer / representative.
SCHERTLER SA strongly believes in “common sense”. Therefore, misuse of our products is not covered under rights obtained through our warranty policy, or through internationally recognized terms and conditions. For more information on warranty, please visit the General Condition’s page at www.schertler.com

PRODUCT DISPOSAL
This product must not be disposed of in general household waste. It should be taken to a disposal center for electrical / electronic waste. Please note any local or national regulations that may be applicable here.

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