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CONGRATULATIONS!

Congratulations on you purchase of a TL Audio M1 Tubetracker Mixer, the perfect solution for the modern day studio. Anyone who has used or heard our outboard equipment, will be familiar with its ability to create a warm, smooth, and clean sound that can inject wonderful 'feel' into your recordings.

Our acclaimed VTC and M4 consoles have also now been accepted as a great sounding must have in the professional recording environment, with many high profile users. The M1 offers you the TL Audio quality, in a desk that is modular, yet more compact and affordable than the VTC and M4 – and still offers you the same great sound.

The M1 Tubetracker is ideal for the home and project studio as well as broadcast and recording applications where space is limited.

The M1 offers you quality valve mic preamps on each channel, 3 band musical EQ with sweepable mid on each channel, 2 auxiliary sends per channel as well as effects returns and alternate monitoring options!

It has been designed to maximise today's recording techniques, with the addition of an optional ADAT interface for easy digital connection to your recording device and software. It makes the perfect front end to any digital recording set up, taking away that 'clinical, sterile' sound whilst adding warmth and depth to your recordings – giving you the sound that you have been longing for.

The M1 Tubetracker is compact in size and features all its connectors on the back of the top panel for easy connectivity and quick cable changes!

Available in 8 and 12 channel options the M1 combines that classic valve sound with modern functionality.

INTRODUCTION

The T L Audio M1 Tube Console combines classic valve techniques with low noise solid state circuitry to construct a mixing console offering high specification signal paths with the unique valve sound quality. 8 or 12 input channels are provided, all with mic and line inputs, switchable 48V phantom power, three band EQ, balanced insertion send and return, 2 Aux sends, Pan and 100mm Fader. Each channel also features a balanced Direct Output, which is pre EQ and independent of the channel fader.

Mixing is via balanced busses to valve mix amplifiers. The stereo output is controlled by a stereo 100mm fader, and monitored by illuminated moving coil VU meters. A stereo effects return, two 2-track returns and independent headphone output complete the monitoring facilities.

The M1 Tube Console is an ideal compliment to a digital audio workstation, but may also be used for direct to stereo recording, as outboard analogue EQ during mixdown, and as a complete studio, stage or PA mixer. The console is table-top mounting and supplied with a free standing power supply. There are no fans in either the power supply or console.

The block diagram of the M1 Tube Console is shown in fig.1. A discrete solid state, electronically balanced input amplifier is used, to achieve state of the art performance with very low noise, low distortion and wide bandwidth. The mic input is via an XLR socket and is suitable for low impedance (150-600 ohm) microphones, with a gain control range of +16 to +60dB plus a 30dB pad. A front panel switch selects 48V phantom power. The line input has an effective overall gain range of -20 to +20dB, allowing the valve stages to be fully driven from line level signals in the range -20 to +20dBu. The input stage is directly coupled to a second stage triode valve amplifier. Increasing the gain at the input stage allows the unique overdriven valve sound to be gradually introduced, as indicated by the "Drive" and "Peak" LEDs. Phase reverse and 90Hz high pass filter switches complete the input section.

A balanced insertion point is provided enabling the connection of external equipment (a compressor, for example) into the channel. The insert send socket may also be used as an independent output. The Direct Output from the channel is pre-fade and balanced. It is taken after the valve stage, insertion point and EQ, but is unaffected by the channel fader.

The equaliser section provides shelving LF and HF filters and swept frequency mid peaking filter. The EQ stages may be bypassed for easy A-B comparison.

Mixing to the stereo bus is via 100mm linear faders with rotary pan control. PFL and channel MUTE switches are also provided. Two Aux sends are available; Aux 1 is switchable pre or post fade, with Aux 2 permanently post fade. The stereo return can also send to Aux 1. The busses are balanced, with hybrid solid state / triode valve mix amplifiers on the stereo buss and solid state mix amplifiers on the Aux and PFL busses.

The monitoring outputs normally follow the stereo output, but may be switched to the 2 track return inputs, for example to check workstation outputs. Any PFL signal automatically over-rides the monitor output. The PFL level relative to the stereo level may be adjusted, and the overall monitor level is controlled by a high quality matched pot with 29mm diameter knob.

Dual moving coil VU meters follow the monitor outputs. The meters are factory set for 0dB indication = +4dBu, but may be recalibrated using presets accessible from the front panel.

The Aux outputs are controlled by master level controls, and the stereo effects return provides a line level input, with gain and balance controls. All are equipped with PFL switches.

Optional digital modules are available, providing multiples of 8 inputs and outputs for the channel sections, plus the stereo main output.

Please read this manual fully before installing or operating the M1 Tube Console.

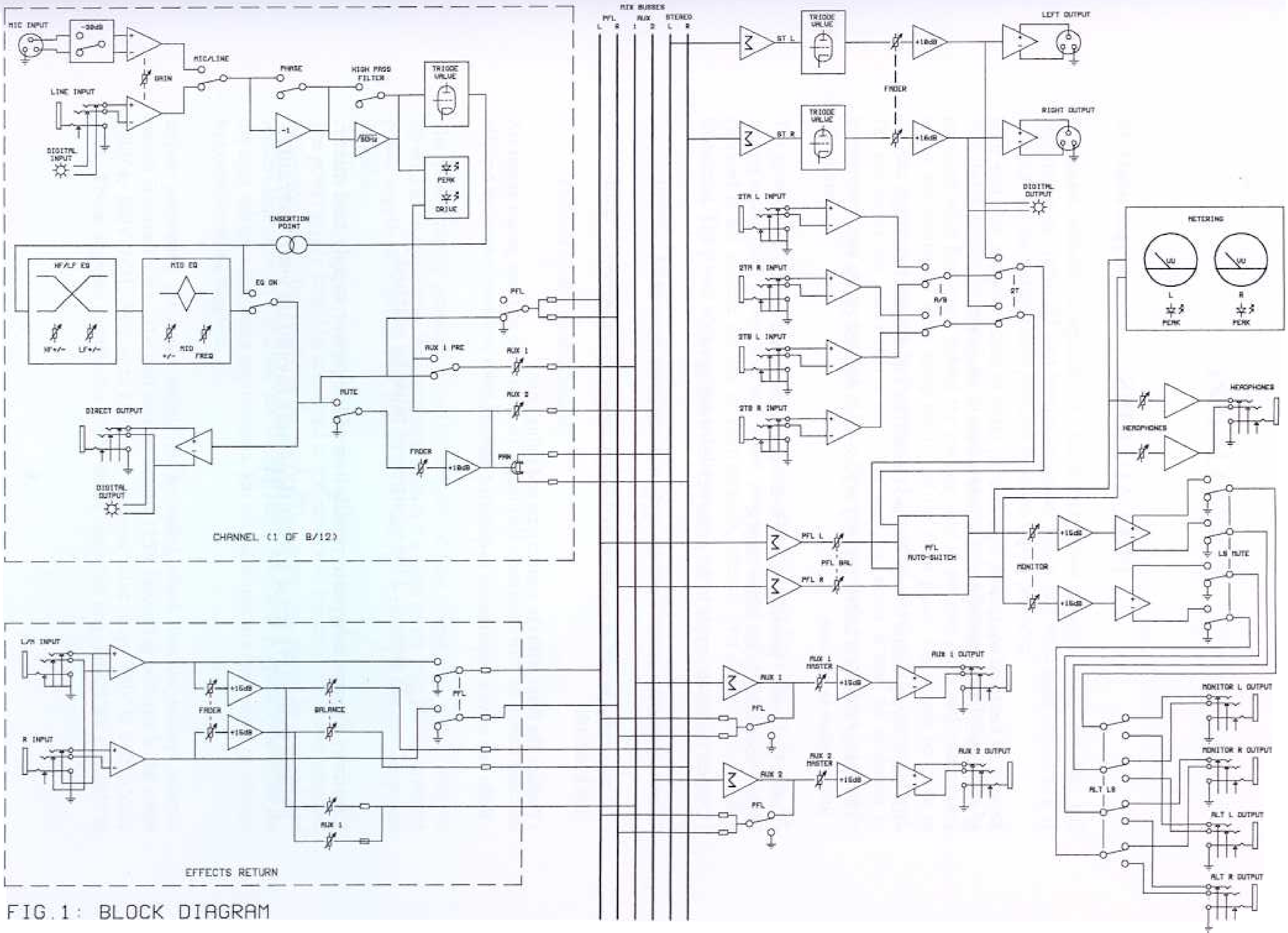

FIG.1: BLOCK DIAGRAM

PRECAUTIONS

As with all electrical equipment, care must be taken to ensure reliable, safe operation of your mixer. The following points should always be observed:

- All mains wiring should be installed and checked by a qualified electrician,
- The mixing console is heavy, and may require more than one person to lift and place in position,
- Ensure the correct operating voltage is selected on the power supply before connecting to the mains supply,
- Connect the power supply to the mixer before switching on,
- Never operate the mixer or power supply with any cover removed,
- Do not expose to rain or moisture, as this may present an electric shock hazard,
- Replace the fuses with the correct type and rating only,
- Refer any service requirement to qualified personnel.

Warning: This equipment must be earthed.

Caution: Ensure adequate ventilation of the power supply and mixer.

Failure to observe these precautions may invalidate your warranty.

INSTALLATION

AC Mains Supply.

The mixer includes a separate TL Audio M1 power supply, which provides DC outputs for the valve HT and heaters, phantom power and solid state circuitry. It is essential that the mixer is only powered from the supply provided.

The mixer should be placed on a solid, flat surface. Adequate ventilation must be provided, with free space behind the cooling slots in the rear panel and immediately above the ventilation slots along the top of the front panel. Care must be taken to provide appropriate support for the mixer and the cables which will be connected to it. Do not locate the mixer near a source of heat, or where it may be subjected to interference from other electrical or radio frequency generating equipment (e.g. mobile telephones).

The power supply is fitted with an internationally approved 3 pin IEC connector. A mating socket with power cord and mains plug is provided with the unit. If the plug supplied is not suitable for use in your location, consult your dealer or a qualified electrician. The power cord is wired as follows:

Brown: Live.

Blue: Neutral.

Green/Yellow: Earth (Ground).

All mains wiring **must** be performed by a qualified electrician with all power switched off, and the earth connection **must** be used.

The power supply generates a certain amount of heat during use, which must be allowed to dissipate from the ventilation slots in the sides of the unit. Never cover the power supply or locate it near a source of heat, for example a radiator or power amplifier.

The power supply should be connected to the mixer with the multi core cable supplied, before connecting the supply to the mains. The connectors should be inserted and secured into place by tightening the locking rings. The cable is specially selected for the high voltage and current requirements. Do not substitute any other cable or attempt to extend the cable supplied.

Before connecting the power supply to the mains, check that the voltage selector switch is correctly set. The unit may be set for 115V (accepting voltages in the range 110V to 120V, 60Hz AC), or to 230V (for voltages in the range 220V to 240V, 50Hz AC). Power supplies suitable for 100V mains operation (e.g. Japan) are available to special order. The fuse required is 20mm anti-surge, 1AT for 230V rated voltage or 2AT for 100/115V rated voltage.

Warning: attempted operation on the wrong voltage setting, or with an incorrect fuse, will invalidate the warranty.

Audio Connections.

Note: The mixer uses high quality standard 0.25" audio metal jack sockets. Good quality compatible 3 pin plugs should be used for all external connections. The use of non-standard plugs may stress the sockets and result in noisy or intermittent contacts. The use of balanced connections where possible is always recommended in an audio studio.

Mic Inputs.

Each channel has a female, 3 pin XLR connector for mic connection. It is compatible with either balanced or unbalanced signals, when the mating connector is appropriately wired:

Balanced inputs:

- Pin 1 = Ground (screen).
- Pin 2 = Signal Phase ("+" or "hot").
- Pin 3 = Signal Non-Phase ("-" or "cold").

Unbalanced inputs:

- Pin 1 = Ground (screen)
- Pin 2 = Signal Phase ("+" or "hot").
- Pin 3 = Signal Ground

When using unbalanced signals, the signal ground may be obtained by linking pins 1 and 3 in the mating XLR connector. Good quality screened cable should be used, particularly for microphone or low level sources, to prevent hum or noise pickup.

Line Input.

A 3 pin, 0.25” jack plug is required, which is also compatible with balanced or unbalanced sources, when wired as follows:

Balanced inputs:

- Tip = Signal Phase (“+” or “hot”),
- Ring = Signal Non-Phase (“-” or “cold”),
- Screen = Ground.

Unbalanced inputs:

- Tip = Signal Phase (“+” or “hot”),
- Ring = Ground,
- Screen = Ground.

Insertion Points.

The insertion points are also interfaced via 3 pin, 0.25” switched jack. The pin connections are:

- Tip = Signal Phase (“+” or “hot”),
- Ring = Signal Non-Phase (“-” or “cold”),
- Sleeve = Ground,

The insertion point is balanced, with send and return on separate jacks. A cable inserted into the send socket only may be used as an additional output without breaking the channel signal path. A cable inserted into the return socket will break the insertion loop. The insertion sends and returns automatically adjust for balanced or unbalanced connection.

Direct Output.

The Track Output connector is a 3 pin, 0.25” jack socket. The output automatically adjusts for balanced or unbalanced connection.

The mating connector should be wired as follows:

Balanced outputs:

Signal Phase (“+” or “hot”),

- Ring = Signal Non-Phase (“-” or “cold”),

- Screen = Ground.

Unbalanced outputs:

- Tip = Signal Phase (“+” or “hot”),

- Ring = Ground,

- Screen = Ground.

Main Stereo Output.

The main outputs are via balanced, 3 pin male XLR connectors. The mating connectors should be wired as follows:

- Pin 1 = Ground (screen),

- Pin 2 = Signal Phase (“+” or “hot”),

- Pin 3 = Signal Non-Phase (“-” or “cold”).

If an unbalanced output is required, pins 1 and 3 should both be connected to ground.

Monitor Outputs.

The monitor outputs are via balanced, 3 pin jack connectors. The mating connectors should be wired as follows:

- Sleeve = Ground (screen),

= Signal Phase (“+” or “hot”),

- Ring = Signal Non-Phase (“-” or “cold”).

If an unbalanced output is required, the sleeve and ring should both be connected to ground.

Aux Outputs.

The Aux outputs are also via balanced, 3 pin jack connectors. The mating connectors should be wired as follows:

- Sleeve = Ground (screen),
- Tip = Signal Phase ("+" or "hot"),
- Ring = Signal Non-Phase ("-" or "cold").

If an unbalanced output is required, the sleeve and ring should both be connected to ground.

Stereo 2T Returns.

Two stereo inputs are provided to the monitor section, to allow off-tape monitoring, etc. The inputs are via balanced, 3 pin jack connectors. The mating connectors should be wired as follows:

- Sleeve = Ground (screen),
- Tip = Signal Phase ("+" or "hot"),
- Ring = Signal Non-Phase ("-" or "cold").

If an unbalanced signal is used, the sleeve and ring should both be connected to ground.

Effects Return.

A stereo input is provided, for effects returns or other line level sources. The connectors are 3 pin jack sockets, and the mating connectors should be wired as follows:

- Sleeve = Ground (screen),
- Tip = Signal Phase ("+" or "hot"),
- Ring = Signal Non-Phase ("-" or "cold").

If unbalanced signals are used, the sleeve and ring should both be connected to ground. The right input of the return is normalled to the left input, allowing the left input to be used a mono return.

Headphone Outputs.

The front panel jack socket is suitable for driving headphones of 32 ohms or higher impedance. A standard 0.25" jack plug should be used.

Meter Calibration.

The meter "0VU" trims may be used to re-adjust the calibration of the VU meters if a setting other than 0VU = +4dBu is required. A steady signal of the desired line up level should be applied, and a small screwdriver inserted through the front panel to adjust the meter to the line up point on the scale.

Digital Modules.

Optional DO-8 digital modules are available for the M1. An 8 channel input/output module may be fitted, providing a digital input which is normalled to the channel line inputs (i.e. inserting an analogue input cable into the line socket will override the digital input), plus digital outputs that follow the channel Direct outputs. When fitted to the 12 channel version of the M1, the digital input and output connections are to channels 1 to 8 only. There is a separate DO-2 module available for the master section, which provides the stereo output in digital format.

The specifications and user instructions for the digital modules are supplied separately.

OPERATION

Input Selection

You must ensure that the correct input connector, mic or line, is being used from the required source. You can select the type of input by pressing the 'LINE' switch at the top right of each channel strip in or out. In the "up' position the mic input is selected; depressing the 'LINE' switch selects the line input as the source. Note that you can have mic and line inputs connected at the same time but will only be able to select a single input at a time.

Input Gain Control

The gain control should be set to obtain the best signal to noise ratio, whilst preserving adequate headroom. Any changes in gain level should be gradual to avoid sudden overload or severe distortion. Extra care should be taken with higher level inputs, and high gain settings should be used sparingly to avoid an increase in the noise floor and the introduction of distortion.

Line Input

A typical line input will require less gain than a microphone signal as it is a 'hotter' source. You should have the gain set to a minimum when connecting a line input, then gradually increase it to achieve the required level.

Microphone Input

When connecting a microphone to the M1, start with the gain set to a minimum value and gradually increase until the sound source is just tickling 0dB on the meters when the source sound is at average output level. You can than choose how much you would like to drive the valves to produce the required sound. If Phantom power is needed, activate it before setting the gain by depressing the button marked '48V' on the channel strip. Although the M1 provides 48V of phantom power, microphones requiring less than 48V can still be connected without any difficulties.

CAUTION: Operation of the phantom power switch, or plugging a microphone in with the phantom power applied, may cause an audible click or thump in your loudspeakers. To avoid this happening, ensure that the channel fader and stereo master faders are set to a minimum before operating the switch or plugging in a microphone. Switching between mic and line with high input gain settings may also cause an audible thump if the level control is not turned down.

Phase Reverse

The phase reverse switch is used to invert the phase of the input signal. It is active on both mic and line inputs. This function could be required when processing a signal that is out of phase with other signals in a mix, in which case the resultant phase error typically appears as a loss of low frequency content, due to cancellation of out of phase components. Phase reverse is commonly used when recording the bottom of a snare drum (if also using a mic on top of the snare), the back of a guitar cab (if also recording signal from the front of the cab), and when performing the stereo recording technique known as an MS pair.

High Pass Filter

This low cut filter provides 12dB per octave of gain reduction with the -3dB point being at 90Hz. Like the phase reverse switch, the high pass filter is active on both mic and line inputs, and is ideal for removing low frequency rumble. The filter can be useful in restricting 'popping' on vocals or even low frequencies caused by contact with microphone stands or microphone cables. Popping is an undesirable thump that is caused by close-miking certain spoken or sung letters, namely 'P' or 'B'. These particular letters cause a sudden expulsion of air that can result in an audible thump. As this thump has a lot of low frequency content the high pass filter can help to reduce the problem, as can using a pop filter (a device usually made out of nylon material similar to stockings) suspended in front of the microphone. The low cut filter is easily bypassed for quick A/B comparison.

30dB Pad

Occasionally when using sensitive condenser microphones the source signal may be too loud for the input preamp. In this situation, to avoid any overloading or distortion of the mic preamp stage, the 30dB pad can be used to reduce the input gain to a more manageable level. The 30dB pad only applies to the microphone input.

Equalisation

Before switching the EQ into circuit, it is advisable to set the cut/boost controls to their centre, or flat, position. The EQ is brought into circuit with the 'EQ' push switch, signalled by a green LED. Each channel has three bands of equalisation: shelving low frequency (LF) (i.e. it extends from the selected frequency to the extreme low frequency limit of the equaliser's response.), peaking sweepable mid (i.e. it boosts or cuts a section of the audio spectrum around its selected centre frequency only) and shelving high frequency (HF). The LF shelf operates at a frequency of 80Hz, while the HF shelf is set at 12kHz. The EQ slopes have a second order 12dB/octave response, and an associated gain control on both bands provides up to 15dB of cut or boost on each selected frequency, controlling the full range of frequencies below the LF corner frequency and above the HF corner frequency. The mid band has a fixed Q of 0.7

(which corresponds to a bandwidth of around 1.5 octaves) allowing +/- 15dB cut or boost over a moderately broad band of frequencies. The centre frequency is selected via the dedicated variable frequency control, and the required amount of cut or boost is applied using the associated gain control. The range of frequencies selectable is 150Hz to 7kHz.

You should frequently A/B the sound you are Equalising by pressing and depressing the EQ button, it is astonishing how quickly the ear can adjust to changes. By making frequent comparisons you can make sure you do not stray from the required sound. It is also worth spending some time getting used to the EQ on the M1 - it sounds great and you'll be amazed how versatile and musical it can be.

Insertion Points

The insert points are configured to be post the input amplifier with its valve stage, but pre EQ and fader. Typical applications include the connection of a compressor (for example the TL Audio 5021) into the channel, or perhaps a dedicated outboard equaliser (e.g. the 5013) for extensive EQ correction or effect.

Aux Sends

Each channel strip has 2 auxiliary sends, one of which can be set to pre fade send. The pre fade send would typically be used for creating a headphone cue, whilst the post fade send would be used as an effects send (i.e. reverb, etc). Aux 1 is switchable to pre, whilst Aux 2 is permanently fixed to post fade.

Direct Output

The Direct Output is configured to be post the input amplifier with its valve stage, the insertion point and EQ, but pre the mute switch and fader. The signal sent to the recorder (or PA system, etc) therefore has the valve warmth and optional EQ and external processing (e.g. compression).

The level of the Direct Output is controlled by the input Gain control, whilst the output to the mix bus is controlled by the fader.

Pan Control

The pan control positions the image within the stereo field, from fully left in the anticlockwise position, through centre at the dented position to fully right in the clockwise position. The gain law is -3dB at the centre.

Mute Switch

The channel may be muted or switched on without affecting the level set on the channel fader. Red LED indicators show which channels are currently muted.

PFL Switch

When PFL (Pre Fade Listen) is activated a yellow LED illuminates above the button on the channel strip and also in the master section. PFL allows you to monitor the incoming source before it reaches the channel fader and the signal is sent to the main output. It is normal to use PFL when setting the gain level for an incoming source.

Channel Fader

A 100mm fader is located at the bottom of the channel, and sets the level of the channel's signal being fed to the stereo mix bus. The fader provides up to 10dB of gain at its highest point.

Drive and Peak LEDs

The yellow Drive LED provides a visual indication of the signal level through the input valve stage, and therefore the extent of 'warming' or valve character being introduced. The drive LED will gradually illuminate as the input level or gain is increased, over the range +4dBu to +12dBu. The red Peak LED operates as a conventional warning that clipping is about to occur. The operating level of both input and post fader stages is monitored, and the LED illuminates at a threshold of +19dBu, when there is less than 7dB of headroom remaining to the direct output. Normal operation would be to set the input gain so that the Drive LED is illuminating regularly, with occasional illumination of the Peak LED occurring on loud transients.

Stereo Mix

A stereo 100mm fader controls the stereo mix output level.

Aux Masters

Each aux send features a rotary master level control, which governs the overall level from the aux output. An associated PFL switch allows each aux send to be auditioned in isolation, by placing the aux signal (pre the master level control) on the PFL buss.

The Aux master controls provide ample additional gain of up to +15dB.

Effects Return

The stereo effect return is equipped with a rotary control and a L/R balance control. The return is fed directly into the stereo mix. A PFL switch allows the return to be auditioned in isolation. A mono signal can be fed to the return by connecting it to the left hand return input only. This will automatically feed the mono signal equally to both left and right hand sides of the stereo buss simultaneously. The return also is equipped with a level control to send to Aux 1, which can be used, for example, to send reverb into the cue mix. The aux send is pre fade, so the effect - or other signal - can be sent into the cue mix without necessarily adding it to the stereo output. The effects return is typically used to feed the output of an FX device back into the stereo mix, adding 'wet' signal to the 'dry' stereo mix. However, the return can also be used as simple extra line inputs for signals that require no EQ or FX to be added to them via the mixer.

Monitor and Headphone Level

The M1 provides a stereo monitor output, for connection to a power amplifier and monitor loudspeakers, plus a front panel headphone output. The Monitor level control governs the level of signal fed to both these outputs. The monitor signal normally follows the main stereo output, but will automatically switch over to the PFL buss whenever any PFL button is pressed, this condition being indicated by a yellow LED. Similarly, the monitor signal will follow the 2T return when the 2T return switch is activated in the master section.

The "ALT LS" switch mutes the main monitor outputs, and activates the alternate loudspeaker sends.

PFL Balance Level

The PFL Balance control allows adjustment of the PFL signal level relative to the main stereo monitor level, as the PFL signal could be considerably louder than the stereo output - depending on the mix and sound source.

2-Track Returns

When recording the stereo mix output onto DAT or CD-R, for example, it is sometimes useful to be able to monitor the output from the mastering machine, in place of the mixer stereo output, as confirmation that the signal is actually being recorded (this is particularly useful when using a four headed DAT machine or a three -headed analogue 2T machine, since the off-tape signal can be monitored). This is achieved by selecting the 2T Return switch near the monitor level control. This facility may also be used for playback after recording. The M1 has the facility to connect two 2T returns, with the 2TA or 2TB switch selecting which of the returns is active when the "2T" switch is engaged

Loudspeaker Mute

This switch mutes the monitor and alternate LS outputs. This is particularly useful when using headphones, enabling the main monitors to be muted without needing to turn off their amplifiers. It is also useful for temporarily muting the monitors without needing to alter any fader levels, for instance during a telephone call.

It is good practice to mute the outputs when switching the M1 on or off, to avoid any loud "thumps" in the loudspeakers.

Metering

Two illuminated VU meters on the M1 will normally monitor the main stereo output, but automatically switch to monitor the PFL buss when any PFL switch is engaged. Similarly, when a 2T return is selected as the monitor source the meters will automatically switch to monitor that 2T return. The meters are calibrated to indicate 0VU when a signal level of +4dBu is generated at the main outputs. A pair of red Peak LEDs operate on the stereo buss as a conventional warning that clipping is about to occur. The LEDs illuminate at a threshold of +19dBu, when there is less than 7dB of headroom remaining on the main outputs. Normal operation would be to set the stereo buss fader levels so that occasional illumination of the Peak LEDs occurs on loud transients.

Digital Output Option

The optional DO8 card can be fitted to the back of the M1, providing 8 channels over ADAT lightpipe. This professional protocol makes for easy connection when conducting computer based recording, and makes the M1 the perfect front-end solution for digital recording set ups.

There is also the optional DO2 card that fits into the back of the master section of the M1 and provides stereo digital output in SPDIF format (RCA Phono connectors). The DO2 has a fixed 24 bit word length, but the sample rate is switchable between 44.1, 48, 88.2 and 96 kHz. These are selectable from the back of the digital card.

Wordclock Input

A rear panel input (on the back of the DO2 & DO8 card) is provided for connecting the M1 to an external master clock source. The wordclock input connector is a BNC-type terminated in 75 ohms.

A FINAL WORD

We hope you will enjoy your new M1 console, we're confident that once you start using it you'll wonder how you managed without one!

For best results, switch your M1 console on shortly before the start of your session to allow the tubes to fully warm up. Enjoy!

SERVICE

Should the mixer or power supply require service, it must be taken or posted to an authorised dealer. Please retain the original packing for possible future use, and ensure the unit is suitably protected during transit. The manufacturer cannot accept responsibility for damage caused during transportation.

The mixer is supported by a limited warranty for a period of one year from the date of purchase. During this period, any faults due to defective materials or workmanship will be repaired free of charge. The warranty excludes damage caused by deliberate or accidental misuse, operation on the incorrect mains voltage, or without the correct type and value of fuse fitted. If claiming service under warranty, proof of purchase date must be included. It is the user's responsibility to ensure fitness for purpose in any particular application. The warranty is limited to the original purchase price of the equipment, and specifically excludes any consequential damage or loss.

Please complete and return the enclosed user registration card, and record the following details:

Serial Number............................

Date purchased..........................

Dealer..

M1 MIXER TECHNICAL SPECIFICATION.

Mic Input:	Balanced XLR socket with switchable 48V phantom power, Gain range +16dB to +60dB with 30dB pad, Frequency response +0, -1dB, 20Hz to 40KHz (at 40dB gain), Input noise (EIN) -128dBu (150 ohm source, 22Hz to 22KHz), Switchable +48V Phantom Power.
Line Input:	Balanced TRS jack socket, Input impedance 22Kohm, Gain range -20dB to +20dB, Maximum input level +26dBu.
Phase Rev	Applies to Mic and Line inputs.
High Pass Filter:	-3dB @ 90Hz, second order. Applies to Mic and Line inputs.
Frequency Response:	(Line Input to Direct Output) +0, -0.5dB, 10Hz to 40KHz.
Distortion:	0.4% typical (measured Line Input to Direct Output at +4dBu input and 0dB gain over a bandwidth of 20Hz to 20KHz. Distortion is predominately tube generated second harmonic. Increasing the "Drive" level by applying more input gain progressively increases the distortion).
Noise:	-89dBu, 22Hz to 22KHz, (Line Input to Direct Output, Input gain @ 0dB).
EQ:	3 Band, with shelving LF/HF and peaking Mid, EQ "On" switch and LED, HF +/-15dB @ 12KHz, Mid +/-15dB @ 150Hz to 7KHz, LF +/-15dB @ 80Hz.
Insert Point:	Balanced send and return on TRS jacks.
Direct Output:	Balanced output on TRS jack. Nominal level +4dBu. Pre fade, independent of mute switch and fader. Maximum level +26dBu.
Fader:	100mm "K" Series, Mute and PFL switches and LED's, Routing to L+R busses via Pan control, "Drive" LED with illumination increasing from +4dBu to +12dBu indicating valve signal level, "Peak" LED illuminating @ +19dBu, monitoring channel input amplifier and post fader signal.

L+R Outputs:	Balanced insertion points with +4dBu / -10dBu switch, Stereo Master fader, 100mm "K" Series, Output via balanced XLR connectors, Noise -76dBu (all channels Line Input @ 0dB gain) Maximum level +26dBu.
Aux Sends:	Aux 1 switchable "Pre" fader, Aux 2 post fader, Channel send level controls plus master level with PFL switch.
Stereo Return:	Balanced stereo returns, Left input normalled to Right input for mono compatibility, Aux 1 send level control, Level and balance controls and PFL switch with LED.
Monitoring:	Balanced outputs on TRS jacks, High quality level control with 29mm knob, PFL relative level trim control with +/-20dB range, Switch with LED to select one of two balanced 2 Track returns, LS mute switch with LED, Alt LS switch with LED, Headphone output with independent level control, Twin illuminated 45mm VU meters with "Peak" LED's, Front panel accessible 0VU trims.
Power Supply:	External free standing supply, 2 metre cable attached to desk with plug and socket on PSU, PSU switchable for 110-220V 60Hz or 220-240V 50Hz use, Typical power consumption 100/140VA (8/12 channels).
Digital I/O:	Optional DO-2 output modules for L+R outputs, Optional DO-8 module for channels: outputs follow "Direct" outputs, inputs normalled to analogue "Line" inputs. Please see separate sheets for digital specifications.
Dimensions:	675mm (29.5") deep, plus at least 100mm (4") at rear of console for PSU and audio connections, 190mm (7.5") high. Length of 8 channel console 475mm (18.7"), Length of 12 channel console 618mm (24.25"). (Dimensions include wood trim).

The above figures are representative of typical production consoles, but are not guaranteed limits for any particular console. These specifications are subject to change without notice.