CONTENTS

IMPORTANT SAFETY INSTRUCTIONS

Please read all of these instructions and save them for future reference. Follow all warnings and instructions marked on the unit.

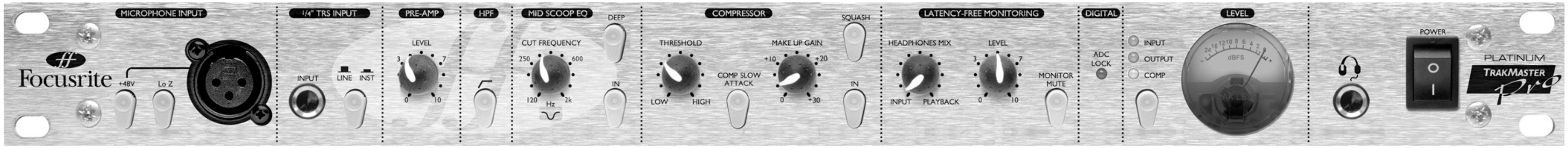
- Do not obstruct air vents in the rear panel. Do not insert objects through any apertures.
- Do not use a damaged or frayed power cord.
- Unplug the unit before cleaning. Clean with a damp cloth only. Do not spill liquid on the unit.
- Ensure adequate airflow around the unit to prevent overheating. We recommend leaving a blank 1U panel above the unit to aid ventilation.
- Unplug the unit and refer servicing to qualified service personnel under the following conditions: If the power cord or plug is damaged; if liquid has entered the unit; if the unit has been dropped or the case damaged; if the unit does not operate normally or exhibits a distinct change in performance. Adjust only those controls that are covered by the operating instructions.
- Do not defeat the safety purpose of the polarised or grounding-type plug. A polarised plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wider blade or the third prong is provided for your safety. When the plug provided does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

WARNING: THIS UNIT MUST BE EARTHED BY THE POWER CORD. UNDER NO CIRCUMSTANCES SHOULD THE MAINS EARTH BE DISCONNECTED FROM THE MAINS LEAD.

This unit is supplied with an external power supply dependant on the region in which the TrakMaster Pro is purchased. To avoid the risk of fire, replace the mains fuse only with the correct value fuse, as marked on the plug. The external power supply unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Focusrite dealer.

RACK VENTILATION: PLEASE ENSURE THE TRAKMASTER PRO IS PLACED TOWARDS THE BOTTOM OF YOUR EQUIPMENT RACK, WITH SUFFICIENT SPACE ABOVE AND BELOW FOR VENTILATION.

INTRODUCTION



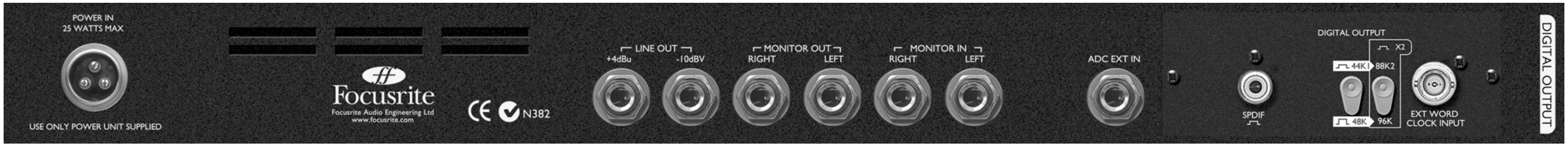
The Trakmaster Pro is the ideal choice for the recording or mixing of a mono signal, whether it's a vocal track or a bass guitar plugged directly into the front panel input. The TM Pro utilises classic Focusrite preamplification, vintage dynamics and EQ, also making use of the fantastic LATENCY-FREE MONITORING system much loved by other Pro Platinum users. The optional digital converter card means that the TrakMaster Pro can also output audio data in SPDIF format, for direct digital interfacing with the DAW ensuring maximum signal quality.

The mic pre featured in the TrakMaster Pro has a circuit design based around the popular Focusrite Green range and features an extremely transparent sonic character with very low distortion. It utilises the same full bandwidth philosophy of all Focusrite products, ensuring detail and clarity without obvious colouration.

Both XLR and TRS jack inputs are located on the front panel for quick and easy access, the latter of which can be used for either line level sources or DI-free plug-in of guitars and basses. The XLR connector also allows a microphone signal the option of a lower impedance input, should the microphone in use have a low output impedance (around 50Ω for example).

The TM Pro truly sets itself a class apart with the inclusion of a vintage optical COMPRESSOR circuit, capable of really heavy compression effects, and a MID SCOOP EQ section for further control over the presence and balance of the processed signal. With classic Focusrite analogue circuit design and a comprehensive digital output option, TM Pro is the perfect partner for a digital audio workstation, making high quality recording and signal processing easy.

REAR PANEL CONNECTIONS



The TrakMaster Pro features two line level outputs on 1/4" jack connectors, one balanced (TRS, +4dBu) and one unbalanced (mono plug, -10dBV); these outputs both transmit the same signal (that which is being processed by the TM Pro) but with either balanced or unbalanced interconnects. In addition to this, there are monitor inputs and outputs with 1/4" jack connectors for making use of the TrakMaster Pro's LATENCY-FREE MONITORING system.

The balanced (+4dBu) TRS jack MONITOR INPUT connectors allow the output of a stereo mixer or sound card to be connected to the TM Pro. There are also balanced (+4dBu) TRS MONITOR OUTPUT connectors, allowing the TrakMaster Pro to be connected directly to a pair of active monitors. See page 6 for details.

An ADC EXT IN connection, on balanced (+4dBu) TRS jack, allows an external mono signal to be fed into the TM Pro for digital conversion. This signal becomes the right channel of the stereo signal sent to the optional digital card (with the signal processed by the TM Pro as the left) and also the right channel of the (now stereo) 'input' signal whilst utilising LATENCY-FREE MONITORING. See the ADC EXT IN section on page 7 for further details.

GETTING STARTED – A QUICK GUIDE

1. Ensure that nothing other than the mains supply is connected to your TrakMaster Pro, then switch it on via the POWER switch on the right hand side of the unit. If your unit is permanently connected to a patchbay, ensure audio is not being fed to any connected speakers to avoid any turn-on speaker pops.

2. Connect the analogue line outputs of the TrakMaster Pro to your recorder or audio interface. If using the digital converter option, connect the TrakMaster Pro's SPDIF output to the SPDIF input of your recorder or audio interface. See page 7 for more information on the TrakMaster Pro digital options.

3. Ensure that the LEVEL dial in the preamp section is set fully anti-clockwise and that the compressor and EQ are switched out.

4. Connect your input source to the relevant input as required. A microphone should be plugged into the XLR MIC INPUT on the front panel. If you wish to connect a line-level source, connect this to the 1/4" TRS INPUT on the front panel; this will automatically switch off the microphone input. You may connect an electric guitar or bass to the 1/4" TRS INPUT.

5. If using the INSTRUMENT INPUT, ensure that the INST switch is engaged. If using the MICROPHONE (XLR) INPUT, ensure that nothing is connected to the 1/4" TRS INPUT.

6. If using a condenser microphone that requires phantom power, activate the +48V switch, next to the microphone input on the front panel. This will provide phantom power to the microphone connected to the MIC INPUT. If you are unsure as to whether your microphone requires phantom power, refer to its user guide, as phantom power will damage some microphones, most notably ribbon microphones or radio mic receivers. NB Dynamic microphones will be unaffected by phantom power.

7. Increase the LEVEL control in the PREAMP section until the signal can be heard. To visually monitor the incoming signal, press the switch beside the dBFS meter in the LEVEL section of the front panel until the input LED is lit (See LEVEL section). It is advisable to set the incoming signal to be an average of around -10dBFS, peaking at around -6dBFS to allow some headroom for occasional sudden loud bursts.

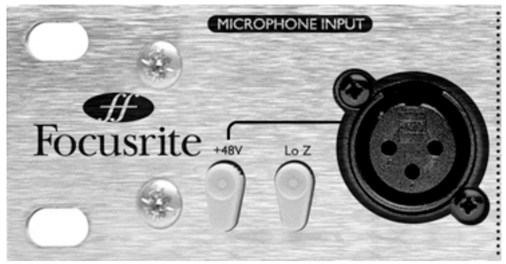
8. If using a microphone, ensure that the microphone placement is at its best. Before you start recording, move the microphone until you get as close as possible to the sound you want. Note that moving the microphone may have an effect on the level of the signal entering the TrakMaster Pro, requiring an alteration to the LEVEL setting.

FACILITIES AND CONTROLS

POWER

This switch turns the unit on. We recommend that the unit be powered up before connecting to any equipment that it is feeding, to avoid clicks or thumps, which may harm output devices. It is also a good idea to allow the unit to stabilise for a couple of minutes before use to ensure that the internal circuitry is properly acclimatised.

MICROPHONE INPUT



The TrakMaster Pro has a front panel XLR input to connect microphone sources to the preamplifier.

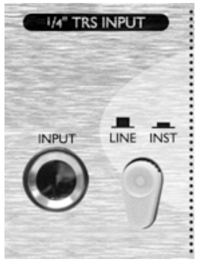
+48V (Switch)

This switch activates a phantom power circuit for the XLR input on the front panel, so that 48V is supplied to the connected microphone. If you are using a condenser microphone then this switch will need to be engaged. If you are unsure whether your microphone requires phantom power, refer to its user guide as phantom power will damage some microphones, most notably ribbon microphones. NB Dynamic microphones will be unaffected by phantom power.

Lo Z (Switch)

This switch changes the input impedance from 2.5kΩ to the lower value of 150Ω (when engaged). This should be utilised if the output impedance of the microphone is low, around 50Ω for example, or if a ribbon microphone is in use. Alternatively, this switch can be activated with a higher impedance microphone to experiment with different sounds.

1/4" TRS INPUT

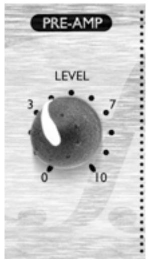


A line (and instrument) input is provided on the front panel via a balanced 1/4" TRS jack socket. An unbalanced signal will be accepted when in instrument mode. Connecting a jack plug into the front panel line input automatically cuts the mic XLR input and selects an appropriate gain calibration for the LEVEL dial in the Preamp section.

LINE/INST (Switch)

When engaged (in), the 1/4" TRS INPUT will accept an unbalanced instrument signal and the LEVEL dial in the Preamp section will be calibrated to an appropriate level.

PRE-AMP



LEVEL (Knob)

This sets the level of the incoming signal. Connect an input signal to the unit, ensuring that the LEVEL control is set fully anti-clockwise, and increase the LEVEL control until a signal is displayed on the front panel meter (ensure that input is selected as the source for the meter - see LEVEL section) or at the destination (recording medium). If the meter displays above 0dBFS, you should reduce the LEVEL. It is advisable to set the incoming signal to be an average of around -10dBFS, peaking at around -6dBFS to allow some headroom for occasional sudden loud bursts.

With a MIC INPUT connected to the XLR connector (ensure nothing is connected to the 1/4" TRS input), the LEVEL control provides +13dB (fully anti-clockwise) to +60dB (fully clockwise) of gain.

With a LINE INPUT connected to the 1/4" TRS connector, the gain is adjustable from -10dB (fully anti-clockwise) to +36dB (fully clockwise).

With the INST switch activated and an instrument connected to the 1/4" TRS input, the LEVEL control provides +13dB to +60dB of gain.

HPF

(Switch)

This is a high-pass filter, which removes unwanted low frequencies such as stage rumble via microphone stands or 'proximity effect' (where low frequencies are over-emphasised when using certain types of microphone at close range). Engaging the switch inserts a 2-pole HP filter (-3dB at 120Hz) into the signal path.

MID SCOOP EQ

The MID SCOOP EQ section allows you to cut a band of selectable frequency by up to 10dB. This can be useful for removing troublesome frequencies when recording acoustic instruments and bass guitars, for example.

IN (switch)

The IN switch activates the MID SCOOP EQ into the signal path. When engaged, the red LED in the switch cap is lit.

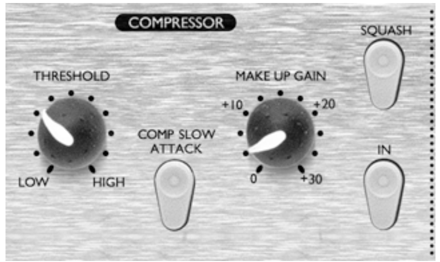
CUT FREQUENCY (knob)

Adjusting the CUT FREQUENCY knob allows the centre frequency of the reduced band to be adjusted. The frequency range is 120Hz to 2kHz.

DEEP (switch)

Engaging the DEEP switch increases the depth of cut from -5 dB to -10 dB.

COMPRESSOR



The COMPRESSOR acts like an automatic volume control, turning down the volume of a signal if it gets too loud. This reduces variation between loud and quiet passages, as it automatically reduces the gain when the signal exceeds a given volume, defined as the threshold. Using the COMPRESSOR helps to 'even out' a performance, stopping a signal from clipping and/or disappearing in the mix, and can also give it a whole new sonic character.

IN (switch)

This switch activates the COMPRESSOR into the signal path. When engaged, the red LED in the switch cap is lit.

THRESHOLD (knob)

Turning this knob clockwise decreases the amount of compression by raising the threshold and vice versa in the other direction. Note that the signal is only compressed when it exceeds the threshold, so quieter passages maintain their natural dynamic range, whilst loud passages (that exceed the threshold) are compressed.

MAKEUP GAIN (knob)

This knob sets the output volume of the compressed signal. Since compressing a signal makes it quieter, use the MAKEUP GAIN control to restore the signal to its original volume. Compare the volume of the original and the compressed signal by using the IN switch to switch the COMPRESSOR on and off. The gain can be modified by up to 30dB.

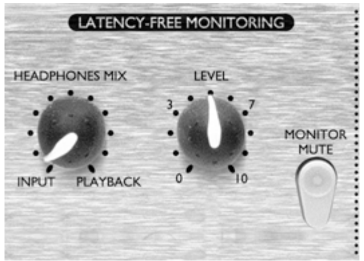
SQUASH (switch)

This switch changes the mode of the COMPRESSOR to a much more aggressive one, similar to that of a limiter. Engaging the switch raises the threshold and increases the ratio, allowing the user to create a 'squashed' effect on the processed signal, popular with designers of vintage compressors.

COMP SLOW ATTACK (switch)

Engaging this switch selects a slower attack time, which allows more of the transient peaks of the signal through the compressor. This can help retain a sense of the original signal's dynamics when compressing heavily. For example, this can be useful to allow compression of a snare drum without losing the initial 'crack' of the drum stick striking the snare skin. The COMP SLOW ATTACK switch will not function in SQUASH mode as the compression is too heavy to warrant a slow attack.

LATENCY-FREE MONITORING



Latency can be a major problem when recording to a computer-based digital audio workstation (DAW) via a sound card. If the signal being recorded has to pass through the DAW before being monitored, significant delays may occur as a result of the digital conversion and processing that takes place, making it difficult or impossible to sing, speak or play in time with any pre-recorded tracks being played back.

The TrakMaster Pro's LATENCY-FREE MONITORING section allows the user to monitor a mix of either the mono signal being recorded (fed directly from the unit before it passes through the digital recording system) or a stereo signal (if using the ADC EXT IN) and a stereo mix of pre-recorded tracks. The unit acts as a mini mixer and so latency is eliminated and the recording artist can speak, sing or play along to the pre-recorded tracks in perfect time.

Simply connect the audio outputs of the DAW to the 1/4" jack MONITOR INPUTS on the rear panel and this will become the PLAYBACK signal, ready for mixing with the INPUT signal. The input signal will be the mono recorded signal, sent to both L and R channels on the headphones, unless a second signal is connected to the ADC EXT IN.

If using two TrakMaster Pros and only one digital card, the signal from the second unit can be connected to the ADC EXT IN on the rear panel; this incoming signal is then treated as the right hand channel of a stereo pair (with the TrakMaster it's connected to as the left hand channel). The stereo pair then becomes the INPUT signal, sent to the L and R channels of the headphones accordingly and the optional digital card for conversion to SPDIF format.

The signal fed to the MONITOR OUTPUTS is the same signal as the one sent to the headphones; a blend of the INPUT and PLAYBACK signals (described above), controlled by the HEADPHONE MIX knob (see below).

HEADPHONE MIX (knob)

This knob allows the user to blend between the INPUT signal (whatever is being recorded - see above) and the PLAYBACK (the signal connected to the Monitor inputs on the rear panel), to create a mix in the headphones. This mix is also sent to the MONITOR OUTPUTS on the rear panel.

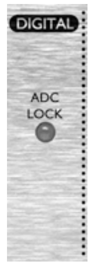
LEVEL (knob)

This knob sets the gain level of the signal sent to the headphones and the (PLAYBACK) signal sent to the MONITOR OUTPUTS on the rear panel. Rotating the dial clockwise increases the level of both the headphones and the speakers connected.

MONITOR MUTE (switch)

Engaging this switch will mute the signal sent to the MONITOR OUTPUTS on the rear panel, for use when recording (with a microphone) and listening to speakers in the same room for example.

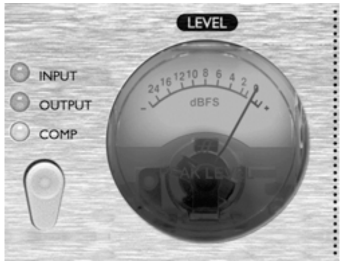
DIGITAL



ADC LOCK (LED)

The ADC LOCK LED indicates, if lit, that the optional ADC, if installed, is correctly synchronised to the external word clock, if one is in use.

LEVEL



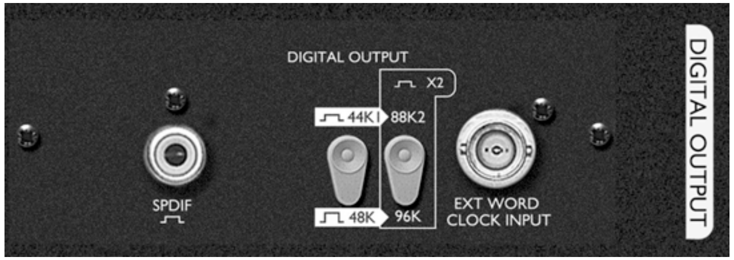
The meter on the front panel displays the level of the selected signal in dBFS, where 0dBFS (digital clipping point) relates to +22dBu of analogue signal level or 0dB of compressor reduction (if viewing the COMPRESSOR). If the INPUT or OUTPUT signals show over 0dBFS, they should be reduced using either the LEVEL knob in the PREAMP section (for Input and Output without dynamics) or MAKEUP GAIN knob in the COMPRESSOR section (for Output with dynamics). It is advisable to set the incoming signal to be an average of around -10dBFS, peaking at around -6dBFS to allow some headroom for occasional sudden loud bursts. To change the source signal fed to the meter, simply use the switch as described below.

Pressing the switch beneath the INPUT, OUTPUT and COMP LEDs selects the source signal that the meter on the front panel displays. Pressing the switch repeatedly will cycle through the three options in sequence. If COMP is selected then the meter displays the gain reduction of the compressor, where 0dB indicates that the processed signal is unaffected.

ADC EXT IN

This balanced (+4dBu) TRS jack connector on the rear panel allows an external mono source, from another Focusrite channel strip for example, to be fed into the TM Pro for digital conversion and subsequently monitoring purposes. The connected signal becomes the right channel of the stereo signal (with the signal processed by the TM Pro as the left channel) sent to the optional digital card for conversion. With nothing connected to ADC EXT IN, the stereo signal sent to the ADC for digital conversion will be the same mono internal signal processed by the TrakMaster Pro on both the L and R channels. This stereo signal also becomes the 'input' signal in the LATENCY-FREE MONITORING section. See the LATENCY-FREE MONITORING section on page 6 for further details.

OPTIONAL DIGITAL OUTPUT



The TrakMaster Pro can be fitted with a digital output card. This allows quality digital conversion of the recorded signal within the unit, plus an additional signal if making use of the ADC EXT IN connector (see ADC EXT IN section for further details), before transmitting directly to the DAW (interface or soundcard) or other recording medium.

The optional digital converter card features one SPDIF output. Digital audio can be transmitted at either 44.1, 48, 88.2 or 96kHz sample rates (selected using the switch(es) on the rear panel) and with 24-bit resolution.

SPDIF OUTPUT

This connector transmits SPDIF format digital audio using an RCA phono connector. If 16-bit resolution is required, the receiver should dither the incoming signal to convert to 16-bit.

SAMPLE FREQUENCY

Two switches give a choice of four sample frequencies as marked on the rear panel. The left hand switch selects between 44.1kHz (switch in) and 48kHz (switch out), and the right hand switch doubles the selected frequency when switched in.

EXT WORD CLOCK INPUT

If an external word clock source is attached to this BNC connector, the TrakMaster Pro will attempt to synchronise to it. When the unit is correctly locked to the external clock source, the ADC LOCK LED in the DIGITAL section on the front panel will illuminate. The LED should remain permanently lit; any flashing or flickering indicates jitter or other timing discrepancies, which need investigating. The frequency being received at EXT WORD CLOCK INPUT should be selected on the ADC card to achieve lock.

DIGITAL OUTPUT INSTALLATION GUIDE

The kit should contain: -

Qty	Description
1	24/96 analogue to digital converter card
2	Crosshead screws
2	Support pillars
2	Nyloc nuts

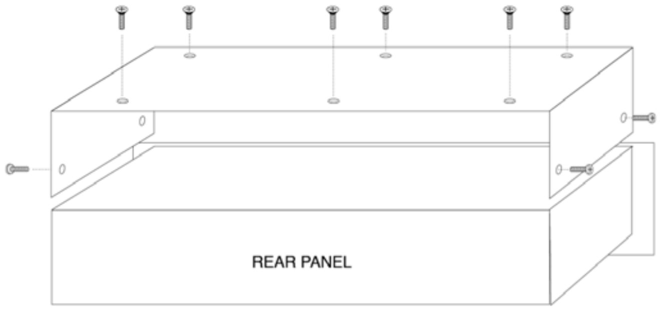
Tools required: -

No. 1 crosshead screwdriver, Pozihead preferred.
M3 nut driver/spanner.

WARNING!
THE MODULE SHOULD BE DISCONNECTED FROM THE AC POWER BEFORE ATTEMPTING TO CARRY OUT THE FOLLOWING INSTRUCTIONS.
ALLOW THE MODULE TO COOL BEFORE STARTING INSTALLATION OF THE DIGITAL OPTION.
ANTI-STATIC PRECAUTIONS SHOULD BE TAKEN WHEN HANDLING THE CARD OUTSIDE OF ITS ANTI-STATIC BAG; ONLY HANDLE THE CARD BY GRIPPING THE CARD BY ITS EDGES AND AVOID TOUCHING ANY OF THE COMPONENT PARTS OTHER THAN THE CABLE AND CONNECTORS. PLACE THE UNIT ON A CLEAN, FLAT SURFACE.

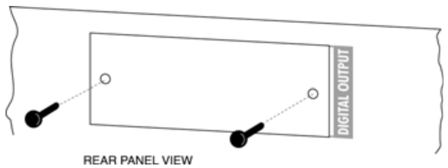
Top cover removal

Remove the 10 crosshead screws fixing the top cover to the top and sides of the module.



Digital option cover removal

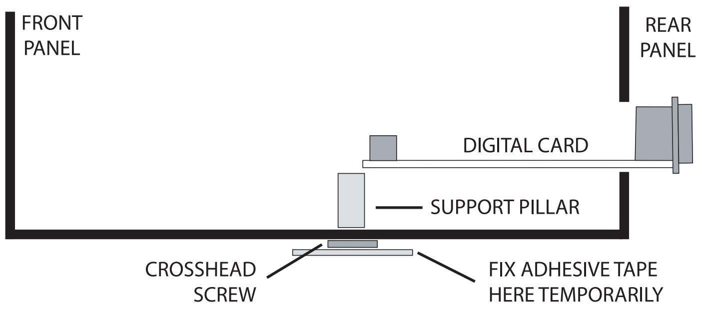
The rear panel digital connector area is accessed by removing the small rear cover plate next to DIGITAL OUTPUT. The plate is removed by removing the two crosshead screws. Retain these screws for securing the digital card in place later.



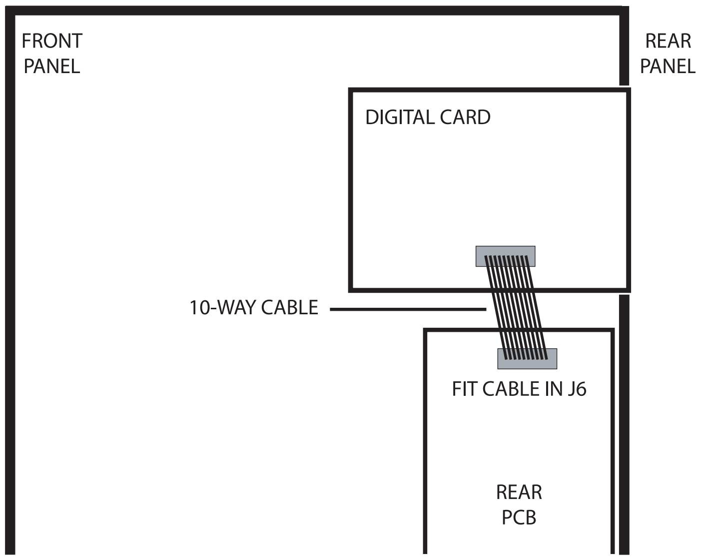
Installing digital card

The digital card is mounted in place using the digital connector fixings and the two plastic support pillars, which should be fitted to the module base using the two crosshead screws supplied.

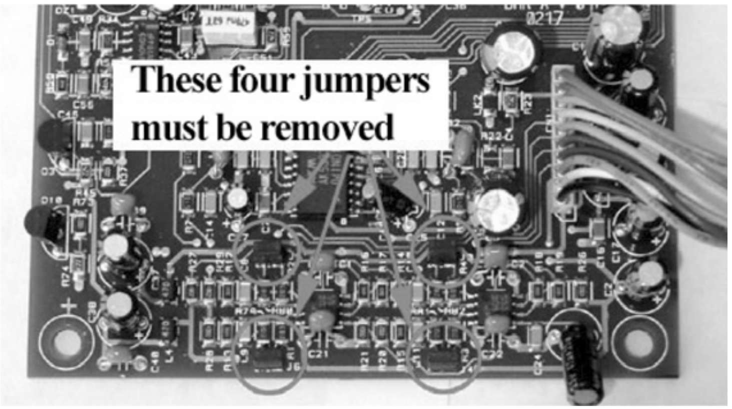
SIDE VIEW



TOP VIEW

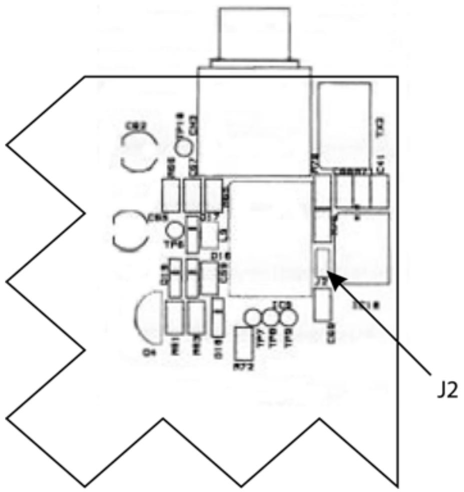


1. Place module on a flat clean surface with the bottom facing uppermost and offer the two crosshead screws supplied to the two countersunk holes in the base, using a small piece of adhesive tape to secure the head of the screw to the bottom surface of the module.
2. Place module top uppermost and from the inside of the module place the two plastic support pillars over each screw.
3. Now offer the digital card from the rear of the module, lining up the screws from the base with the two holes in the card, and secure it in place with the two Nyloc nuts supplied.
4. Using the two screws that secured the original blanking panel, secure the digital card to the rear panel.
5. Plug in the 10-way cable to J6 on the main circuit board, and remove the adhesive tape from the base of the module.
6. Remove links J3, J4, J5 and J6 as shown below:



Selecting for Professional or Consumer Use

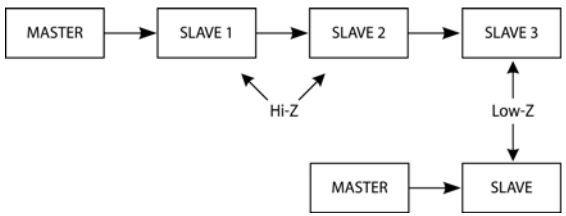
This link determines the SPDIF output level for use with various Professional and Consumer products. Please consult your recording product/soundcard manual to determine which setting is appropriate for your system.

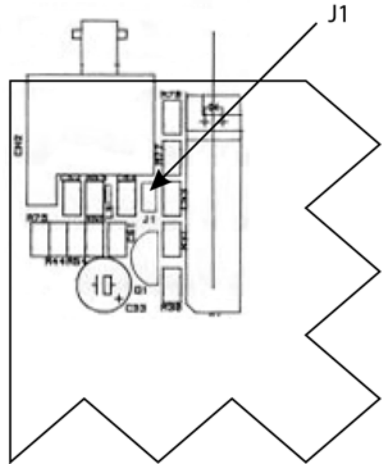


- For Professional use, J2 should be left fitted, as supplied from factory.
- For Consumer use, J2 should be removed.

Selecting Word Clock Impedance

When using the module as the only slave, or at the end of a chain of several slaves, 75 Ω Lo Z should be selected. If using the module with several other slave modules in a chain, Hi Z should be selected. See the following diagram:





- For 75 Ω Low-Z operation, J1 should be left fitted, as supplied from factory.
- For Hi-Z operation, J1 should be removed.

Replacing the Top Cover

The top cover should now be replaced using the 10 crosshead screws. The installation is now complete and the unit can be reconnected to the AC power.

Notes

- When syncing to an external clock the sampling frequency should be set to match between the two pieces of equipment.

FREQUENTLY ASKED QUESTIONS

Q: What is the difference between +4dBu and -10dBV?
A: These are different signal operating levels. +4dBu usually refers to professional equipment and -10dBV usually refers to semi-professional or consumer equipment. It is important to make sure that any two or more devices connected to each other are operating at the same signal level. The +4dBu/-10dBV line outputs on the rear of the TrakMaster Pro give the option of either analogue output level. If the +4dBu output of a device feeds the -10dBV input of another device, this may cause the second device to overload. Alternatively, if the -10dBV output of a device feeds the +4dBu input of another device, the second device may receive a signal level which is too low (i.e. too quiet). -10dBV devices are usually connected using a mono 1/4" jack, this is known as an 'unbalanced' connection. +4dBu devices are usually connected using a TRS (stereo) 1/4" jack, or XLRs. This is known as a 'balanced' connection.

Q: Should I use balanced connectors with my TrakMaster Pro?
A: Preferably yes; the Line level analogue input and +4dBu output are balanced, operating at +4dBu. If you wish to connect unbalanced equipment to the TrakMaster Pro's line input however, this can be done, but the signal level will be 6dB lower. The 1/4" TRS input will expect to receive an unbalanced signal if the INST switch is engaged.

Q: Does the TrakMaster Pro have the same kind of spectacular bandwidth that has given the Red and ISA range units their reputation for 'open-ended' sound?
A: Yes. The audio bandwidth of the TrakMaster Pro is 10Hz to 200kHz!

Q: Can I take my TrakMaster Pro with me when I travel internationally?
A: It depends. There are three versions of the TrakMaster Pro mains transformer. One is suitable for use in North America and Japan, with mains voltages in the 100-120V range. The other two versions are designed for use in the UK and Europe (the only difference being the mains plug), with mains voltages in the 200-240V range. If you buy a TrakMaster Pro in a particular territory, it will be configured for ONLY that territory's mains voltage range. For example, if you're travelling from the USA to the UK, you CANNOT use your US model TrakMaster Pro with the supplied mains transformer. But if the mains voltage in the country you're visiting is in the same range, you can use the TrakMaster Pro with no problems - so taking a TrakMaster Pro from the USA to Japan, or from Germany to France, for example, would be fine. An external supply for different territories is available from your local Focusrite distributor.

Q: Can I retrofit a digital board to an analogue TrakMaster Pro at a later date?
A: Yes, and you can do it yourself - it can easily be retro-fitted by the customer without any soldering etc, just a few screws to undo and one clip-connector to join to the main PCB.

TROUBLESHOOTING

No LEDs illuminate
- Is the POWER switched on?
- Is the correct Power Supply Unit being used?
- Has the fuse in the mains plug blown?

No output when using the MIC INPUT
- Is the power switched on?
- Is the LEVEL set correctly? (See 'Facilities and Controls' section for details.)
- Is something connected to the LINE input on the front panel?
- For microphones that require phantom power, is the +48V switch switched in? (If you are unsure whether your microphone requires phantom power, check the user guide for your microphone.)

No output when using the LINE INPUT
- Is the power switched on?
- Is the LEVEL set correctly? (See 'Facilities and Controls' section for details.)
- Is the INST switch on the front panel switched out?

No output when using the INSTRUMENT INPUT
- Is the power switched on?
- Is the LEVEL set correctly? (See 'Facilities and Controls' section for details.)
- Is the INST switch on the front panel switched in?

The COMPRESSOR is not working
- Is the compressor section's IN switch engaged?
- Is the LEVEL set correctly? If set too low, the signal level may not be high enough to activate the compressor.
- Is the THRESHOLD control set correctly? If set too high, the input level may not reach the threshold at which compression starts.

The MID SCOOP EQ is not working
- Is the EQ section's IN switch engaged?
- Is the CUT FREQUENCY control set to a frequency that is present in the signal?

No ADC LOCK
- Is your external wordclock source transmitting wordclock?
- Is an external wordclock cable connected from the output of the source to the input of the TM Pro?
- Is the sample frequency set to match that of the TM Pro?

No output from the digital output option
- Is the sample frequency set correctly?
- Is the receiving device set to receive at 24-bit resolution?
- Is the receiving device set to external sync?

CONTACTING US

If you have any further questions about your TrakMaster Pro, or are continuing to have difficulty, you can visit the support section at www.focusrite.com or email us for help at tech@focusrite.com. Alternatively, telephone us on +44 (0)1494 462246, or contact your local distributor (see listing at the back of this manual).

SPECIFICATIONS

Mic Input Response

Gain = +13dB to +60dB
Input impedance = 2.5kΩ (150Ω in Lo Z mode)
EIN = 126dB @ 60dB gain with 150Ω termination & 22Hz/22kHz filter
THD+N @ min gain (+13dB) = 0.001% with 0dBu input & 22Hz/22kHz filter
THD+N @ max gain (+60dB) = 0.009% with -38dBu input & 22Hz/22kHz filter
THD+N @ max input (+8dBu) = 0.001% with 22Hz/22kHz filter
Frequency response @ min gain (+13dB) with -13dBu input = -3dB @ 10Hz & -2.3dB @ 200kHz
Frequency response @ max gain (+60dB) with -60dBu input = -3dB @ 13Hz & -3dB @ 78kHz
CMRR @ max gain (+60dB) = 80dB

Line Input Response

Gain = -10dB to +36dB
Input impedance = 24kΩ
Noise @ unity gain (0dB) = -88dBu with 22Hz/22kHz filter
S/N ratio relative to max headroom (+36dBu) = 124dB
S/N ratio relative to 0dBFSs (+22dBu) = 110dB
THD+N @ unity gain (0dB) = 0.001% with 0dBFS (+22dBu) input and 22Hz/22kHz filter
Frequency response @ unity gain (0dB) = -2.8dB @ 10Hz & -3dB @ 200kHz

Instrument Input Response

Gain = +13dB to +60dB
Input impedance = 1MΩ
Noise @ min gain (+13dB) = -87dBu with 22Hz/22kHz filter
Noise @ max gain (+60dB) = -45dBu with 22Hz/22kHz filter
THD+N @ min gain (+13dB) = 0.004% with 0dBu input & 22Hz/22kHz filter
Frequency response @ min gain (+13dB) with -13dBu input = -3dB @ 10Hz & -2.3dB @ 200kHz

Level Meter

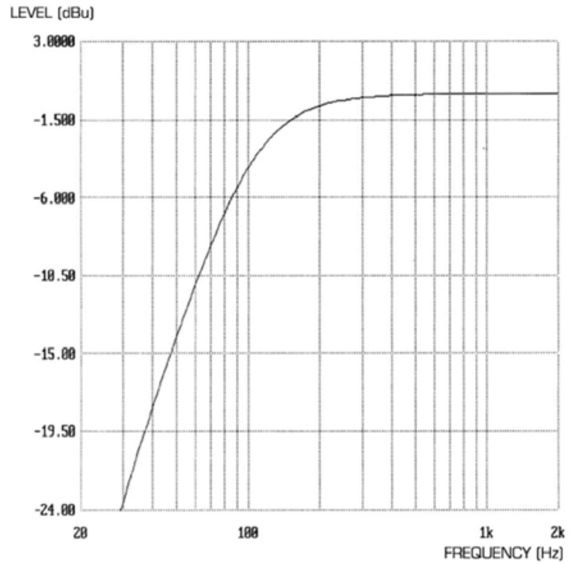
Peak Level Moving Coil Meter

Input & Output:
-24dBFS to +2dBFS (-2dBu to +24dBu)
+22dBu = 0dBFS

Gain Change:
0dB to 24dB

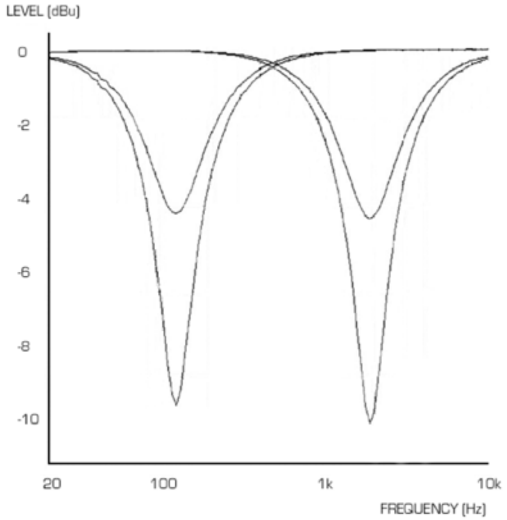
Hi Pass Filter

Roll-off = 12dB/octave 2-pole filter
Cutoff frequency = -3dB @ 120Hz, -6dB @ 85Hz, -12dB @ 56Hz
Frequency Range:



Mid Scoop EQ

EQ shape = peak
Centre frequency = variable between 120Hz and 2kHz
Cut (DEEP switch out) = -5dB
Cut (DEEP switch in) = -10dB
Q (DEEP switch out) = 0.7
Q (DEEP switch in) = 2



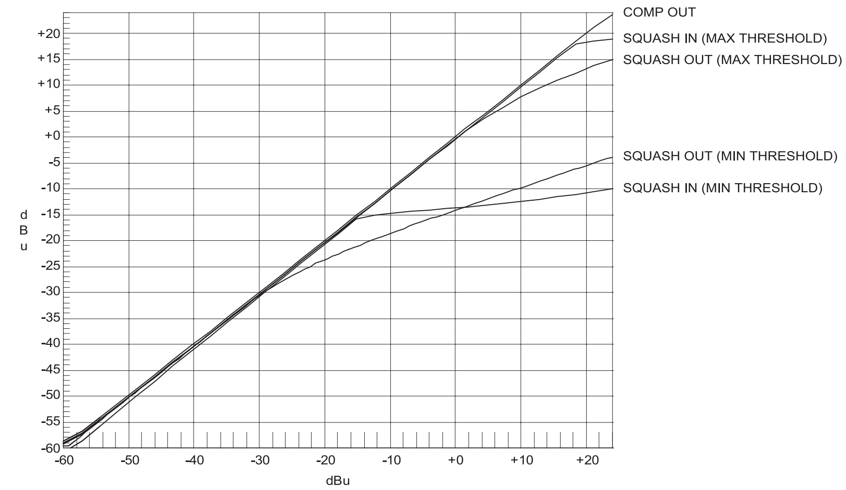
Compressor

Threshold range (SQUASH switch out) = -54dBFS (-32dBu) to -18dBFS (+4dBu)
Threshold range (SQUASH switch in) = -37dBFS (-15dBu) to -4dBFS (+18dBu)
Compressor ratio (SQUASH switch out) = 2.5:1
Compressor ratio (SQUASH switch in) = 35:1

Attack and Release preset values dependent on switch settings as follows:

	Attack	Release
COMP	35ms	126ms
COMP+SLOW ATTACK	120ms	350ms
SQUASH	8ms	1s

Noise = -90dBu measured with a 22Hz/22kHz bandpass filter.
Make-up gain = 0 to +30dB.



ADC Performance

Sample Frequency = 44.1kHz, 48kHz, 88.2kHz & 96kHz
Bit Depth = 24bit
Maximum analogue input level = +22dBu (0dBfs)
Dynamic Range = 109dB 'A' Weighted

EXT WORDCLOCK INPUT

BNC connector

ADC EXT IN

Balanced 1/4” TRS jack socket input (+4dBu)

MONITOR IN

2 x balanced 1/4” TRS jack socket inputs (+4dBu)

MONITOR OUT

2 x balanced 1/4” TRS jack socket outputs (+4dBu)

LINE OUT

Balanced 1/4” TRS jack socket output (+4dBu)
Unbalanced 1/4” TRS jack socket output (-10dBV)

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