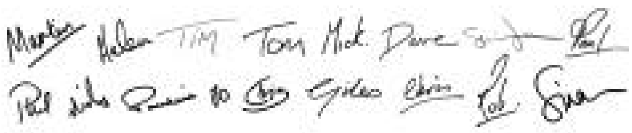
Introduction

Thank you for purchasing the ISA 428 Pre Pack brought to you by the Focusrite team – Martin, Crispin, Helen, Tim, Tom, Mick, Dave, Simon, Paul, Phil, Micky, Pauline, Jo, Chris, Giles, Chris, Rob and Simon.

The chaps at Focusrite are a jolly hard working bunch and take a great deal of pride in designing, building and delivering the best audio units around. We hope your new Focusrite unit lives up to that reputation, and that you enjoy many years of productive recording. If you would like to tell us about your recording experiences then please email us at: sales@focusrite.com



The Focusrite Team

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.
- 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 are 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 capable of operating over a range of mains voltages as marked on the rear panel. Ensure correct mains voltage setting and correct fuse before connecting mains supply. Do not change mains voltage settings while mains supply is connected. To avoid the risk of fire, replace the mains fuse only with the correct value fuse, as marked on the rear panel. The internal power supply unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Focusrite dealer.

Power Connections

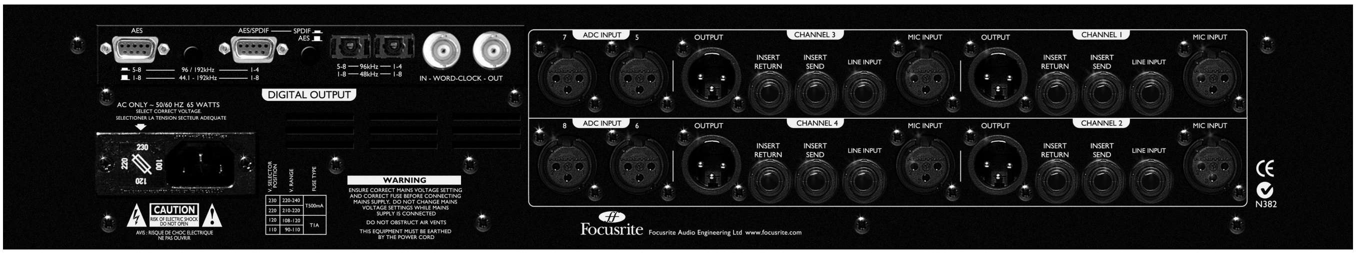
There is an IEC mains lead supplied with the unit, which should have the correct moulded plug for your country. The wiring colour code used is:

For units shipped to the USA, Canada, Taiwan and Japan:
Live - Black **Neutral** - White **Earth** - Green

For units shipped to any other country:
Live - Brown **Neutral** - Blue **Earth** - Green and Yellow

Signal Connections

(Optional ADC shown fitted)



XLR (Audio) Inputs and Outputs

All 3-pin XLR balanced audio connectors (Output, Mic IP and EXT A/D IP) are wired as follows:

Pin 1	Screen/Chassis
Pin 2	Audio 0°
Pin 3	Audio 180°

Line IP and Insert Sends and Returns

1/4" balanced jack wired as follows:

Tip	Audio 0°
Ring	Audio 180°
Sleeve	Screen/Chassis

Inst. Hi Z IP

1/4" unbalanced jack wired as follows:

Tip	Audio 0°
Sleeve	Screen/Chassis

Mic IP/Line IP/Inst. Hi Z IP (front panel)

Any one of these inputs may be used as the input to the ISA 428 channels 1-4. Signals routed to these inputs are referred to as the 'internal' channels or signal paths.

Outputs - Channels 1-4

These outputs are used as the main analogue signal outputs, and are fed by whatever is connected to the Mic Inputs, Line Inputs, or Instrument Inputs. These outputs also connect to the ADC 'internal' channels 1-4 via the Soft Limiter circuit.

Insert Send and Return

Allows an external unit, such as a Red 3 Compressor or Red 2 EQ, to be placed within the signal chain, prior to the output and post the high pass filter.

ADC Inputs 5-8

The ADC inputs 5-8 act as line level inputs and are used to route 'external' signals to the optional ADC channels 5-8 via the Soft Limiter. Using these inputs in conjunction with a single optional ADC card, up to eight analogue inputs can be routed to the eight digital outputs. Using these inputs in conjunction with a second ISA 428 unit and a single optional ADC card, eight pre amps can be routed to all eight ADC channels. (See 'Signal Connections' on page 8.)

Retrofitting the Optional ADC

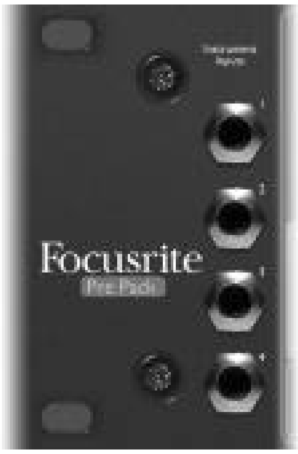
The optional ADC can be retrofitted to a standard ISA 428 at any time. The card can be fitted easily by the user – no engineering experience is required. Full fitting instructions for this option are included along with the ADC.

Getting to know the ISA 428

Power

Applies power to the unit. Turn on the ISA 428 before powering up devices to which the outputs are connected**.**

Instrument Inputs



Instrument sources may only be connected via the front panel. Four unbalanced Instrument input connectors are located to the far left of the front panel and are numbered 1-4, correlating to each of the four channels. These connectors are used primarily for connecting low level unbalanced signals such as those from passive guitars and basses, or from active instruments such as keyboards and electro-acoustic guitars.

Metering

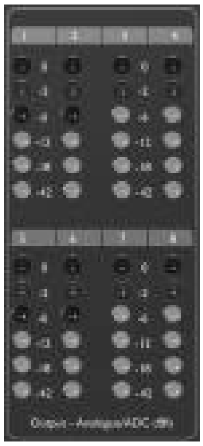
Peak Meter and O/L LED

This moving coil meter has been designed to have a very fast reaction time and a slow decay time, and so is perfect for displaying and holding the peak level of the channel signal during recording. The meter is fed with a signal taken from the point in the channel path after the high pass filter and before the insert send connector. The moving coil peak meter reading therefore informs the user of the signal level in the channel after gain has been applied by the pre-amp, and also shows the level of the signal being sent to any external equipment connected to the insert send output.

The meter is calibrated relative to the overload point of the optional ADC card, (0dBfs on the meter refers to the maximum level that can be converted by the ADC), and relative to the analogue overload point, (where then 0dBfs on the meter refers to an analogue level of +22dBu, 6dB below the maximum analogue level available from the unit).

Within the meter is a hidden overload indicator marked O/L that lights (red) when the signal in the channel exceeds 0dBfs (+22dBu), and acts as an additional safety feature for monitoring peak signal levels.

Analogue/ADC dBfs Output meters



These vertical columns of LEDs indicate the peak signal levels of channels 1-8 in one of two modes, the modes being defined by the fitting of the optional ADC card as follows:-

Mode 1. Analogue only unit (no optional ADC card fitted).
Meters 1 to 4 indicate the analogue level at the ISA 428 XLR output connectors for channels 1 to 4. 0dBfs (reached when the red LED is lit) indicates that a signal level of +22dbu is present at the output (–18dBfs) therefore

indicating that there is a signal level of +4dBu at the output. Meters 5 to 8 have no function in this mode.

Mode 2. Digital (optional ADC card fitted).
Meters 1 to 8 indicate the signal level that exists in the signal path after the Soft Limiter and just before the point of conversion on the optional ADC card. 0dBfs (reached when the red LED is lit) indicates the maximum signal level that can be converted by the optional ADC card and should only be lit for very short durations to ensure a good quality recording with no digital overload. The meter signal is taken from a point after the Soft Limiter, so if the Soft Limiter has been selected to protect the ADC from overload, the effect of the Soft Limiter on the peak signal levels will be indicated by a reduction in peak levels on the LED meter.

Input Stage

Three input options are provided to give compatibility with microphone, line or instrument level sources.

Input

Pressing INPUT steps through each of the three inputs, as indicated by the corresponding LEDs. When the Mic LED is lit, the microphone input is active etc. Hence a mixture of microphone, line and instrument inputs may be selected across the four channels simultaneously.

Mic Input Gain

With the Mic input selected, the user has access to the full gain range in 10dB steps from 0dB to +60dB (yellow legend). The gain range is split between two gain modes depending upon the status of the 30-60 switch.

Mode 1 Mic Gain Range 0-30
With the 30-60 switch off, the rotary gain knob operates over a gain range of 0dB to +30dB, the level of gain chosen being indicated on the front panel by the outer arc of yellow numbers around the gain knob.

Mode 2 Mic Gain Range 30-60
With the 30-60 switch on (illuminated), the rotary gain knob operates over a gain range of 30dB to 60dB, the level of gain chosen being indicated on the front panel by the outer arc of yellow numbers around the gain knob.

An additional 20dB of gain can be applied to the signal after the Mic/line gain knob using the Trim knob. See 'Trim' control text below for full explanation.

Line Input Gain

With the line input selected, the user has access to gain settings ranging from –20dB to +10dB, indicated on the front panel by the arc of white numbers around the gain knob. The 30-60 switch is inactive when the line input is selected, as the gain range for Line level inputs is restricted to –20dB to +10dB in 10dB steps.

An additional 20dB of gain can be applied to the signal after the Mic/line gain knob using the Trim knob. See 'Trim' control text below for full explanation.

Instrument Input Gain

With the instrument input selected, gain is applied to the input signal by using the trim control only, which allows +10dB to +40dB of gain range. The level of gain chosen is indicated on the front panel by the outer arc of yellow numbers around the gain knob. This input is suitable for high impedance sources such as guitar or bass pickups (which may be connected directly without the need for an external DI box) or vintage synthesizers with high impedance outputs.

Trim

The Trim control provides additional variable gain of 0dB to +20dB when Mic or line inputs are selected. The level of gain chosen is indicated on the front panel by the inner arc of white numbers around the gain knob.

The additional 20dB of gain that can be applied to the Mic or Line signal is very useful for two reasons:

When high gain is required
The trim used in conjunction with the Mic gain of 60dB will give a total of up to 80dB of pre-amp gain, making it very useful for getting good digital recording levels from very low output dynamic and ribbon microphones.

Gain adjustment during recording
When small amounts of gain adjustment are needed to correct for performance level variations during recording, use the trim knob rather than the stepped Mic/Line gain knob, as switching the 10dB gain steps would be much too intrusive. It is therefore good practice to apply some Trim gain **before** using the 10dB stepped gain knob to find the optimum recording level so that the Trim control can be used to gently add or take away gain later, if so required.

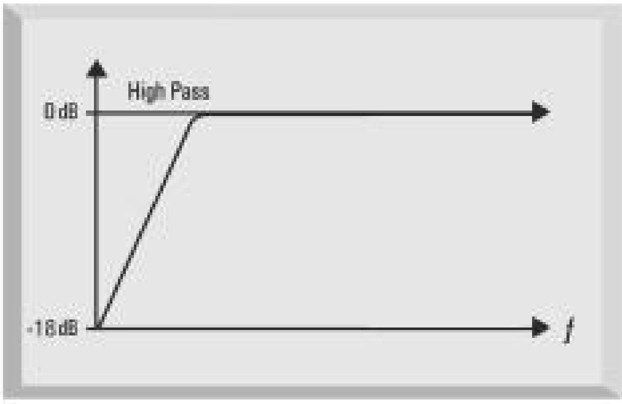
+48V

Pressing the +48V switch provides +48V phantom power, suitable for condenser microphones, to the rear panel XLR microphone connector. This switch does not affect the other inputs. If you are unsure whether your microphone requires phantom power, refer to its handbook, as it is possible to damage some microphones (most notably Ribbon Microphones) by providing phantom power.

Phase
Pressing PHASE inverts the phase of the selected input, to correct phase problems when using multiple microphones, or when incorrect wiring polarity has occurred.

Insert In
Pressing the INSERT switch (illuminated) breaks the signal path of the channel, so that the channel input signal is sent out of the unit from the rear panel Insert Send connector and returned to the same point in the signal chain via the rear panel Insert Return connector.

Filter In
Pressing the FILTER IN switch makes the Hi Pass Filters active in the audio path. The filter provides 18dB/octave roll-off. A variable control allows the roll-off frequency to be set within the range of 16Hz to 420Hz.



Impedance
Pressing IMPEDANCE steps through each of the four transformer pre-amp input impedance values, as indicated by the corresponding LEDs. By selecting different values for the impedance of the ISA 428 transformer input, the performance of both the ISA 428 pre-amp and the microphone connected can be tailored to set the desired level and frequency response.

Analogue to Digital converter (ADC) Soft Limiter

Pressing ADC SOFT LIMITER activates the ADC Soft Limiter - giving total protection of all 8 channels on the ADC.

ADC Soft Limiter Operation
The Soft Limiter circuit acts instantaneously, clamping down the audio so that the signal can never go above the maximum level that the ADC can accurately convert (0dBfs). Consequently, it is impossible to overload the connected optional ADC card. This function is a part of the ADC signal path, and so only operates when an ADC card is fitted. If no card is fitted then the switch is inactive.

8 Channel Digital Output Option

The ISA 428 can be used as a high quality 8 channel ADC for analogue transfer to digital with the addition of the optional ISA 428 digital output board. The 4 external ADC inputs and the main channel inputs can all be fed to the ADC, via the Soft Limiter, ensuring eight super-clean, protected, high-quality paths to digital. A single ISA 428 unit can act as an 8-channel digital input expansion unit to any DAW. Channels 1-4 always route respectively to ADC output channels 1-4.

Alternatively, two ISA 428 units with a single A/D option can be used to create an 8-channel mic pre to A/D system (see 'Signal Connections' on page 8). Digital formats available on the ADC are AES/EBU, SPDIF and ADAT™ optical format (the ADAT™ outputs can also operate in high speed SMUX mode for 96kHz transfer speeds, but are muted during 192kHz operation).

24-bit/96kHz ADAT™ interface operation
The card provides digital outputs for all eight ISA 428 channels, which operate over the sample frequency ranges 44.1, 48, 88.2 and 96kHz, and can be dithered to 16-, 20-, or 24-bit depths depending upon the destination.

The card features two ADAT™-type 'lightpipe' output connectors. For speeds up to 48kHz both connectors transmit all 8 channels simultaneously. However, ADAT™-type connectors are bandwidth-limited at sample rates above 48kHz - each audio channel uses two ADAT™ digital channels to accommodate the increased quantity of data, hence the need for two ADAT™ connectors to allow 8 channels of conversion at high speed.

The ADAT™ output connectors operate as follows:

44.1/48Khz sample rates:
Connector 1 = channels 1 to 8 in parallel.
Connector 2 = channels 1 to 8 in parallel (identical to connector 1)

88.2/96Khz sample rates:
Connector 1 = channels 1 to 4.
Connector 2 = channels 5 to 8.

ADAT™ lightpipe cables are available from your local dealer, or in the UK from Studiospares (tel +44 (0)20 7482 1692): stock number 585-510.

24-bit/192kHz AES/SPDIF operation
The card also provides AES and SPDIF format outputs via two 9-pin D-type connector on the rear panel. The full range of sample rates (up to 192kHz) and bit depths are available.

To access the digital signals from the 9-pin D-type output connectors, the A/D card must be purchased with either an AES or SPDIF D-Type conversion cable as follows:

AES cable: 9-pin D-type to 4 male XLR connectors.

SPDIF cable: 9-pin D-type to 4 male RCA (phono) connectors.

Note: cables need to be purchased separately. Since there are two different cable options - XLR for AES and RCA/phono for SPDIF - these are not included with the A/D converter options. Focusrite cables may be purchased from your local dealer. If you experience difficulty in obtaining these cables, contact your local distributor as listed in the back of this manual.

The AES/SPDIF Connector Configuration

There are two AES connectors which can be configured as follows:

AES/SPDIF 9-pin D-type
The connector labelled AES/SPDIF can be configured either as an AES or an SPDIF dedicated output using the AES/SPDIF switch next to it. When operating the connector in AES mode an AES cable is required. When operating in SPDIF mode the SPDIF RCA cable should be used, which automatically sets the output stream to consumer mode.

AES 9-pin D-type
The second AES connector labelled AES always transmits in AES-only mode regardless of the position of the AES/SPDIF switch. The switch between the two 9-pin D-type connectors selects the AES output between ‘1 wire’ and ‘2 wire’ mode as follows:

1 Wire Mode
Selected with the switch in the ‘out’ position. Both AES connectors transmit 8 channels of AES data simultaneously for all sample frequencies from 44.1 to 192kHz.

2 Wire mode
Selected with the switch in the ‘in’ position. Each AES connector transmits 4 channels of AES data separately for sample frequencies from 96kHz to 192kHz.

The reason for the two modes is that older equipment with 96kHz and 192kHz AES inputs can only receive speeds up to 192kHz by using both digital channels of a single AES connection (known as ‘2 wire’). Therefore one AES channel can only send a single channel of digital data, and thus the 9-pin D-type is switched from transmitting 8 channels of data to sending 4 channels of data. Therefore to send all 8 channels from the ISA 428 in this mode requires two AES connectors: one sending channels 1 to 4 (AES/SPDIF connector) and the other sending channels 5 to 8 (AES-only connector). Having this switch makes the ISA 428 useable with both old and new equipment.

Word Clock In and Out

The internal ADC can be synchronised to an external word clock. By pressing the front panel Ext switch, the synchronisation mode can be switched between standard external word clock and 256X external word clock. Both types of external word clock should be connected to the ISA 428 ADC card at the Word Clock In BNC connector. The Word Clock Out BNC connector either regenerates the external word clock connected at the Word Clock In BNC connector, or transmits the internal sample frequency of the ADC card. Where the ISA 428 is being used as a slave device within a larger digital system, the Word Clock Out BNC connector can be used to pass on the external word clock signal to the next device. When the unit is not slaved to another device and is in internal clock mode, the Word Clock Out BNC connector outputs the sample frequency selected on the ISA 428 front panel.

Digital Output Front Panel Controls

Clock Select

Pressing this switch allows the user to select between sample frequencies of 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, and 192kHz

Bit Depth Select

Selectable between 24, 20 and 16 bits.

Ext Select

Pressing EXT allows the ISA 428 to be slaved to an external word clock source. Selecting 256X allows the ISA 428 to be slaved to an external clock running at 256 times faster than the sample frequency and enables connection to systems such as the Digidesign 'Superclock' or other 256 slave clock devices.

Lock LED

When lit, LOCK indicates that the unit is synchronised to an external clock.

Applications

Mic Pre-amp Input Impedance

A major element of the sound of a mic pre is related to the interaction between the specific microphone being used and the type of mic pre-amp interface technology it is connected to. The main area in which this interaction has an effect is the level and frequency response of the microphone, as follows:

- **Level**
  Professional microphones tend to have low output impedances and so more level can be achieved by selecting the higher impedance positions of the ISA 428 mic pre-amp.

- **Frequency response**
  Microphones with defined presence peaks and tailored frequency responses can be further enhanced by choosing lower impedance settings. Choosing higher input impedance values will tend to emphasise the high frequency response of the microphone connected, allowing you to get improved ambient information and high end clarity, even from average-performance microphones.

Various microphone/ISA 428 pre-amp impedance combinations can be tried to achieve the desired amount of colouration for the instrument or voice being recorded.

To understand how to use the impedance selection creatively it may be useful to read the following section on how the microphone output impedance and the mic pre-amp input impedance interact.

Switchable Impedance: In Depth Explanation

Dynamic moving coil and condenser microphones
Almost all professional dynamic and condenser microphones are designed to have a relatively low nominal output impedance of between 150Ω and 300Ω when measured at 1kHz. Microphones are designed to have such low output impedance because the following advantages result:

- They are less susceptible to noise pickup.
- They can drive long cables without high frequency roll-off due to cable capacitance.

The side-effect of having such low output impedance is that the mic pre-amp input impedance has a major effect on the output level of the microphone. Low pre-amp impedance loads down the microphone output voltage, and emphasises any frequency-related variation in microphone output impedance. Matching the mic pre-amp resistance to the microphone output impedance, (e.g. making a pre-amp input impedance 200Ω to match a 200Ω microphone) still reduces the microphone output and signal to noise ratio by 6dB, which is undesirable.

To minimise microphone loading, and to maximise signal to noise ratio, pre-amps have traditionally been designed to have an input impedance about ten times greater than the average microphone, around 1.2kΩ to 2kΩ. (The original ISA 110 pre-amp design followed this convention and has an input impedance of 1.4kΩ at 1kHz.)

Input impedance settings greater than 2kΩ tend to make the frequency-related variations of microphone output less significant than at low impedance settings. Therefore high input impedance settings yield a microphone performance that is more flat in the low and mid frequency areas and boosted in the high frequency area when compared to low impedance settings.

Ribbon microphones
The impedance of a ribbon microphone is worthy of special mention, as this type of microphone is affected enormously by pre-amp impedance. The ribbon impedance within this type of microphone is incredibly low, around 0.2Ω, and requires an output transformer to convert the extremely low voltage it can generate into a signal capable of being amplified by a pre-amp. The ribbon microphone output transformer requires a ratio of around 1:30 (primary: secondary) to increase the ribbon voltage to a useful level, and this transformer ratio also has the effect of increasing the output impedance of the mic to around 200Ω at 1kHz.

This transformer impedance, however, is very dependent upon frequency - it can almost double at some frequencies (known as the resonance point) and tends to roll off to very small values at low and high frequencies. Therefore, as with the dynamic and condenser microphones, the mic pre-amp input impedance has a massive effect on the signal levels and frequency response of the ribbon microphone output transformer, and thus the 'sound quality' of the microphone. It is recommended that a mic pre-amp connected to ribbon microphone should have an input impedance of at least 5 times the nominal microphone impedance.

For a ribbon microphone impedance of 30Ω to 120Ω the input impedance of 600Ω (Low) will work fine and for 120Ω to 200Ω ribbon microphones the input impedance setting of 1.4kΩ (ISA 110) is recommended.

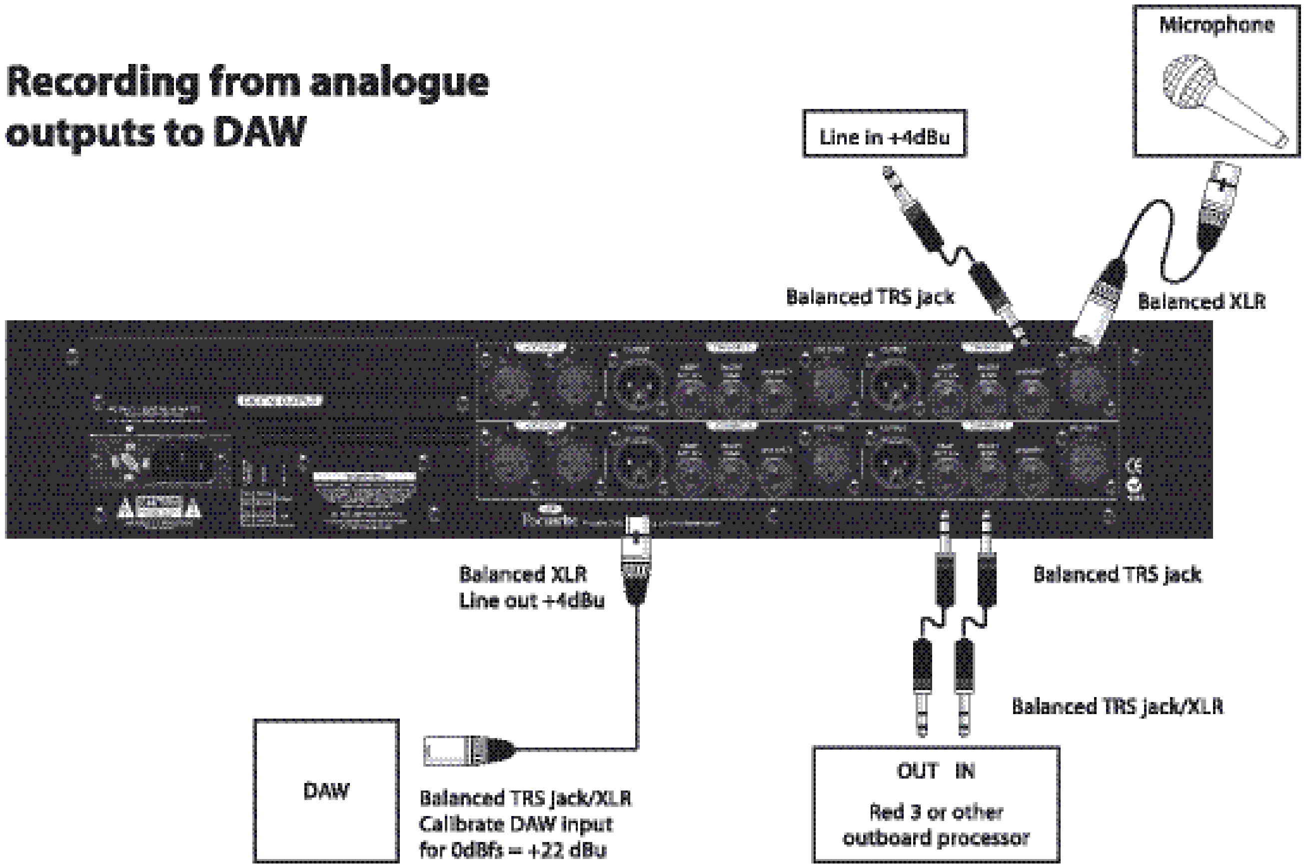
Impedance Setting Quick Guide
In general the following selections will yield the following results:

High mic pre-amp impedance settings
- Will generate more overall level
- Will tend to make low- and mid-frequency response of the microphone flatter
- Will improve high-frequency response of the microphone.

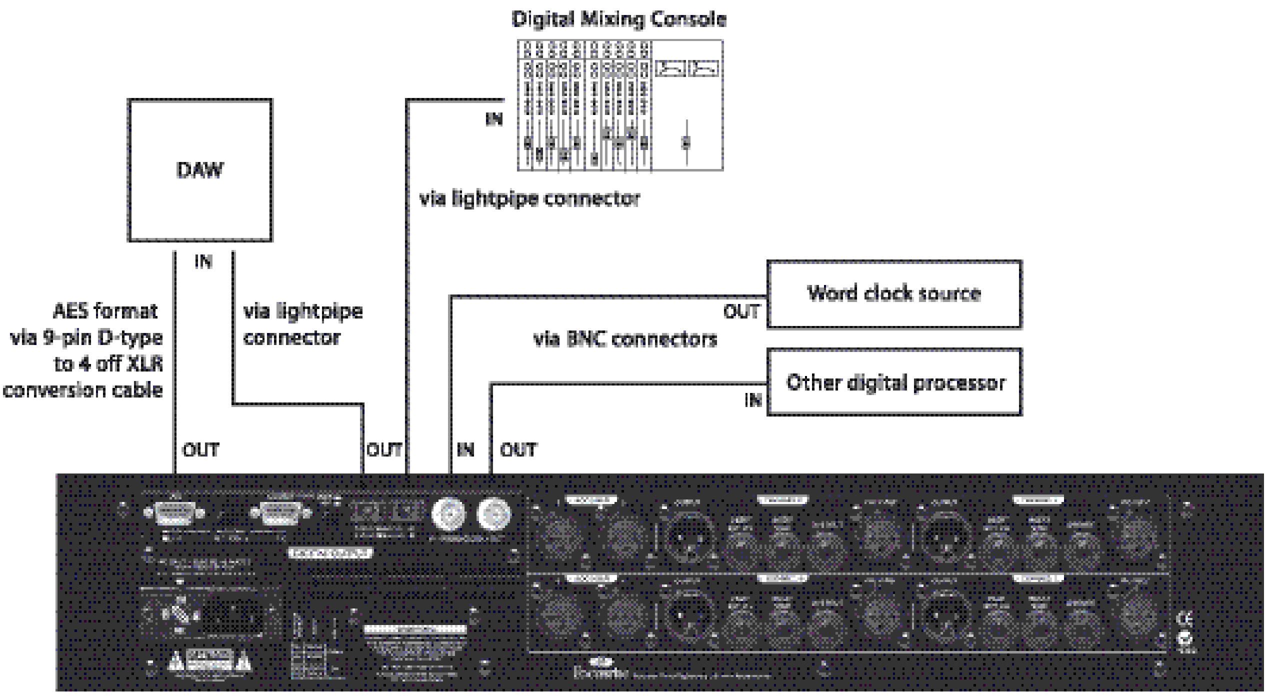
Low pre-amp impedance settings
- Will reduce the microphone output level
- Will tend to emphasise the low- and mid-frequency presence peaks and resonant points of the microphone

Signal Connections

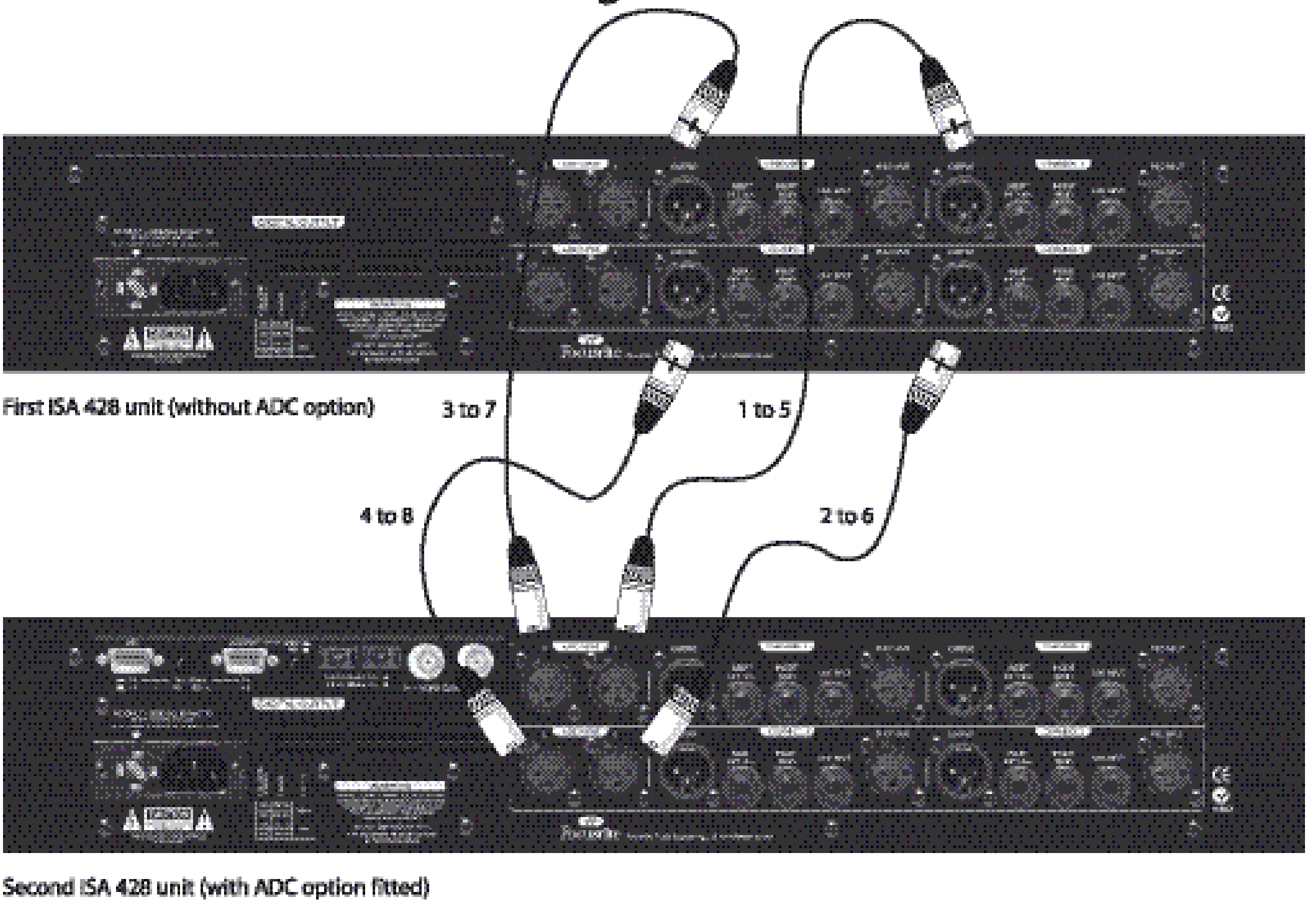
Recording from analogue outputs to DAW



Digital connections



Two ISA 428 units used as single 8-channel ADC



Using the ISA 428 with a Digidesign 192 HD™ interface

The ISA 428 can be used with the Digidesign 192 HD™ interface in one of two ways as follows:

Analogue mode
The XLR analogue outputs of the ISA 428 can be connected to the +4dBu balanced 25-way DB25 connector, labelled 'Analog Input', of the HD interface using a DB25 to 8 off female XLR cable. These cables are available directly from Digidesign or from Hosa cable (part no. DTF 805).

N.B. Digidesign cables are also available as follows:

Description	Part no
DB25-XLR M+F AES/EBU Snake, 12'	MH080
DB25-XLRM Snake, 12'	MH081
DB25-XLRF Snake, 12'	MH082
DB25-TRS Snake, 12'	MH083
DB25-XLR M+F AES/EBU Snake, 4'	MH097
DB25-XLRM Snake, 4'	MH098
DB25-XLRF Snake, 4'	MH099
DB25-TRS Snake, 4'	MH100

The ISA 428 ADC level meters are calibrated to show 0dBfs at +22dBu and the HD interface unit can be calibrated such that it too indicates 0dBfs at +22dBu which will make interfacing and level checking between the two units easier. To recalibrate the HD unit follow the instruction described in the Digidesign 192 HD™ user guide under '192 I/O Calibration Mode Instructions'.

Digital Mode
Connecting the ISA 428 AES digital output 9-way connector to the HD unit 25-way 'AES/EBU I/O' connector requires two cables. The Focusrite ISA 428 9-way cable to 4 off XLR is required to retrieve the AES signals from the ISA 428 ADC card, and this should be connected to the AES input connectors on the Digidesign 25-way to AES I/O XLR cable. This cable is available directly from Digidesign and is called a 'DB25-XLR M+F AES/EBU DigiSnake™'.

Note on HD AES channel restriction
The AES/EBU DB25 way connector on the rear of the HD unit can accept 8 channels of AES/EBU digital data over four AES cables for speeds up to 96kHz. However, for speeds over 96kHz (up to 192kHz) the four AES inputs of the HD interface can only accept 4 channels of audio data.

When the ISA 428 is operating at speeds up to 96kHz a single Focusrite 9-way to AES XLR cable can route all 8 channels of its digital audio into the HD interface. When the ISA 428 is operating at 176.4 to 192kHz a single Focusrite 9-way to AES XLR cable can only route 4 channels of its digital audio into the HD interface. Depending upon which 9-way connector the Focusrite cable is attached to on the ADC card, either channels 1 to 4 or 5 to 8 can be routed into the HD unit.

Other Compatible Focusrite Products

Red 2
A dual equaliser which features Focusrite’s much sought-after warmth and smoothness. Features include 4 band EQ - HF and LF shelving, true parametric High- and Low- Mids, and High-pass and Low-pass filters.

Red 3
A Class A VCA-based dual mono/stereo compressor/ limiter. The compressor/limiter features a single Class A VCA to achieve truly high class compression and limiting free of the usual compromises.

Red 7
A powerful and versatile direct recording processor, featuring a compressor, de-esser and exciter. It features a single Red Range mic pre, plus a mono channel of the dynamics from the Red 3, with the addition of a de-esser/exciter.

ISA 430 Producer Pack,
The only Focusrite product to include a wide range of different classic Focusrite modules in a single frame. In addition to retaining the classic look and sound of the original ISA 110 EQ and some of the original ISA 130 dynamics, it adds new processing technology, state-of-the-art routing flexibility, and digital connectivity.

ISA 220 Session Pack
The ISA 220 Session Pack provides all the legendary audio processing tools required to infuse your session with Focusrite's renowned sonic performance. It features many of the original circuits of the flagship ISA 430 Producer Pack along with some new features of its own.

Platinum Compounder
A high performance dual-channel dynamics processor designed for the quality conscious professional and project studio owner. The combination of high quality compression with the powerful Bass Expander makes this unit a must have for any dance music engineer or musician.

Platinum MixMaster
An analogue stereo audio processor designed primarily for project studio mastering. However, with so many useful features in one box, anyone involved in the business of making music will quickly find it indispensable at other stages of the recording process too.

Platinum Penta
Every dynamics processor you'll ever need, squeezed into one 2U rack-mountable unit, the Penta features a Focusrite Class A pre-amp on the front end with both mic and instrument inputs accessible directly from the front fascia. Following this is a Stereo Preset Compressor, which is entirely editable and features Focusrite’s exclusive Tube Sound Technology.

Platinum Trak Master
Never before has there been a more affordable tracking device, which still manages to encompass the design philosophy and integrity that have ensured that Focusrite be held in such high esteem over so many years. A high quality mic pre, intuitive compression, a three band flexible EQ, and 'tube sound' control come together in this unit to ensure you have all you require to get a quality signal tracked.

Platinum VoiceMaster Pro.
VoiceMaster Pro represents a new generation of voice recording equipment. The award-winning Class A preamp captures every nuance from any source, whilst latency-free monitoring ensures direct and delay-free mix control. Tools such as the voice-optimised EQ, Vintage Harmonics and Tube Sound allow you to get creative with a touch of class, putting your own stamp on every recording.

Check out www.focusrite.com for further information on any of these products and for any future products which may be compatible with the ISA 428.

FAQs

Q: What are the basic features of the ISA 428?
A: Four Focusrite mic pre's, eight line inputs, four instrument inputs, optional 8-channel 192kHz A/D conversion.

Q: Which applications is the ISA 428 suitable for?
A: The ISA 428 can be used as a multi-channel high quality front end for Digital Audio Workstations, allowing multi-channel recording to HD. Equally it can be used simply as the perfect interface/ A/D converter for synths/other line level devices. It also provides additional channels for anyone who has run out of mic pre's on their analogue console (either live or recording,) and is especially useful as a source of additional mic pre's for digital consoles.

Q; Which Focusrite pre is featured on the ISA 428?
A: It's the original transformer-balanced mic pre that featured in the classic analogue Focusrite consoles in the 1980s. This is also the classic pre that features in the ISA 430 Producer Pack.

Q: Do the pre's have the usual phantom power and High Pass Filter controls?
A: Yes, and more… each pre has switchable mic impedance so that you can match to your chosen mic's impedance, or 'mismatch' for creative 'input response colours'. Also there are inserts per channel, plus sweepable HPF, phase reverse, and phantom power.

Q: What's significant about the impedance switching for each pre?
A: Each pre can either be matched perfectly to any microphone, (vintage or modern,) or offset to offer a variety of 'response colours' by interacting with any particular microphone. The impedance of each pre amp is switchable (via a single switch labelled 'Impedance') between 4 settings: original ISA 110 (Zobal network influence for the classic Vintage Focusrite sound,) Low (600 Ohms, tends towards flat and tight sounding,) High (2.4k, relatively open,) and Higher (6.8K, relatively lively, great for room ambience.)

Q: Are insert points featured?
A: Yes, switchable in- or out- of circuit on each of channels 1-4.

Q: What are the four extra inputs on the left hand side of the front fascia for?
A: They are unbalanced inputs which enable you to easily connect unbalanced sources like guitars/basses to the unit without the need for an external DI box.

Q: Why are there eight line inputs but only 4 mic pre's?
A: For two reasons. Firstly it means you can route 8 line level sources to the 8-channel 192k converter with a single unit. Secondly, you can convert the 4 pre system to an 8-pre system with 2 units, see below.

Q: Why is it called "428"?
A: Because it's easy to convert a 4-pre-to-digital system to an 8-pre-to-digital system.

Q; Can I use 2x ISA 428s with 1x A/D card? If so, how does the routing work?
A: Yes. Mic pre's 1 to 4 on the first unit route directly (via the limiter) to digital outputs 1, 2, 3, 4. Mic pre's 5 to 8 on the second unit then route out of their respective analogue outputs, and loop back in to the line inputs 5, 6, 7, 8 on the first unit. These then route directly (via the limiter) to digital outputs 5-8.

Q: But if I run an 8-channel system how can I monitor levels?
A: Easy. Each 428 includes 8 output meters, as well as 4 channel peak-reading input meters.

Q: What's the specification of the A/D option?
A: AES, (both single and dual wire specs.) SPDIF, ADAT™ formats, sample rates selectable between 44.1, 48, 88.2, 96, 176.4 and 192kHz, (ADAT is max 96kHz of course, via 2 ports,) 16, 20, and 24 bit with adaptive dithering, internal or external word clock, and 256X clock, S/N Ratio better than 120dBfs 'A-weighted'. Connections are via 2 x 9-pin D-type connectors and standard lightpipes, word clock is via BNC in and out.

Q: Does the ISA 428's A/D option feature word clock as standard?
A: Yes, word clock may be fed in via a BNC connector on each A/D to allow syncronisation to any word clock master source.

Q: Why do I need word clock anyway?
A: When using multiple pieces of digital equipment it is necessary to make sure that their bit-streams are all in sync. In order to do this all equipment need to be synchronised to a common word clock system. Somewhere in this system a word clock 'master' must be dictating the word clock for the rest of the equipment ('word clock slaves') to follow. Failure to sync all pieces of digital equipment to a common word clock source will result in audible clicks and glitches in programme material. Note that the 428 regenerates word clock at its BNC output, further boosting word clock stability.

Q: What if my system only operates at 16 bit, 44.1kHz?
A: No problem, the ISA 428 features adaptive dithering and supports most sample rates, (simply select the rate you need from the ISA 428's front panel,) so you can use an ISA 428 with a 16, 20 or 24 bit system operating at 44.1, 48, 88.2, 96, 176.4 or 192kHz.

Q: How many rack spaces does ISA 428 take up?
A: The ISA 428 is a 2U device.

Q: What rear panel connections are featured?
A: The ISA 428 has 4 XLRs for microphone input, and 8 XLR line level inputs. There are 4 balanced XLR analogue outputs, plus separate balanced quarter inch send and return insert sockets per channel. Plus digital connections if the optional A/D is fitted of course, see above. Finally there's a voltage-switching power socket to connect to the internal power supply.

Q: Is the ISA 428 a Class A device, and why is that important?
A: Yes, the ISA 428 is a Class A device. Why? Class A is a type of amplifier design in which you have a standing DC current running through your amplifier circuits all the time. As the signal comes along you vary what you're taking from that, rather than switching between supplying a positive current for one half of the waveform and a negative current for the other half. This results in the ability to represent audio in a more linear (distortion free) manner all the way through the circuit. Cheaper processors use IC amplifiers which run close to Class B and don't have the same standing DC current, which means the transistors inside the chips switching off and on, inevitably resulting in a less linear performance.

Q: Should balanced connectors be used with the ISA 428?
A: Yes, where possible. Alternatively, if using an unbalanced instrument source, you can connect to the four unbalanced 1/4" inputs.

Q: What's the ISA 428's bandwidth? Does it 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 bandwidth is the same as the classic Focusrite units of old: 10Hz-200kHz!

Q: Is there an optional digital input card?
A: No, because the ISA 428 is primarily a 'front end' product. In other words, the only devices which are likely to be connected to the 428's inputs are analogue sound sources such as microphones, guitars etc.

Q: Why is the 24-bit 192kHz specification important?
A: An A/D converter works by sampling the audio waveform at regular points in time, and then quantizing those values into a binary number, which relates to the number of bits specified. The quantized signal must then be passed through a D/A converter before it becomes audible. In simple terms, the D/A essentially joins the dots plotted by the A/D converter when the signal was first converted to digital. The number of dots to join, combined with how little those dots have been moved, determines how accurate the final signal will be compared to the original. The greater the sample rate and bit rate, the more accurate the whole digital process is. So 24 bit/192kHz performance will ensure more accurate digital transfer of your audio information compared to the old 16-bit/44.1kHz standards. This is especially important if further digital signal processing is to be applied to the signal once converted to digital, as any mathematical operations taking place on the data, (for example as a result of a gain change, or dynamic effect process,) may result in quantization and rounding errors. The higher the resolution of the digital data, the smaller the audible effect of these errors.

Q: What is dithering? Why do I need it?
A: When dropping down from e.g. 24 bit to 16 bit, quantizing errors occur, (because 24 bit sampling involves more samples than 16 bit, so when you reduce the bit depth the extra samples have 'nowhere to go.') At high signal levels these errors are random and not audible, but at lower signal levels the errors correlate more closely to the audio and become audible as distortion. Dithering effectively 'randomises' the truncation errors at lower levels, causing the 'least significant bit' distortion to disappear.

Q: Can I retrofit a digital board to an analogue ISA 428 at a later date?
A: Yes, and you can do it yourself - it can easily be retro-fitted at any time without any soldering etc, just a few screws to undo, and one clip-connector to join to the main PCB.

Q: How can the ISA 428 operate with the ADAT lightpipe format at 96kHz? I thought the maximum sample rate for this format was 48kHz?
A: Not any more. The ISA 428 supports the new 96kHz ADAT specification, using two discrete optical ports.

Q: How many digital outputs can I use at the same time?
A: It depends on which sample rate you choose.
At 48kHz or lower: 8 AES + 8 S/PDIF (or 2 x 8 AES) + 2 x 8 ADAT = max. 32 outputs simultaneously.
At 96kHz: 8 AES* + 8 S/PDIF (or 2 x 8 AES) + 8 ADAT = max. 24 outputs simultaneously.
At 192kHz: 8 AES* + 8 S/PDIF (or 2 x 8 AES,) no ADAT** = max. 16 outputs simultaneously.
* NB Both AES single-wire and split-wire configurations are supported; split wire will obviously reduce the maximum number of simultaneous outputs.
**At 192kHz: the ADAT ports are muted.
The digital outputs can always be fed with any mix of mic, line, inst. inputs.

Q: When the A/D is used, are my analogue outputs available for use?
A: Yes. The 4 line outputs can run simultaneously with all of the ADAT, AES and S/PDIF outputs.

Q: Do I need to buy an option cable to use either of the A/D cards?
A: Yes; ADAT optical cables are available from many sources, and Focusrite offer their own 8-channel 9-pin to 4 phono (RCA) connectors S/PDIF cable and 9-pin to 4 XLR connectors AES-EBU cable.

Specifications

Mic Input Response

- Gain range = 0dB to 60dB in 10dB steps.
- Input Impedance, variable as follows:-

Switched Impedance setting	Equivalent Input Impedance at 1KHz
Low	600Ω
ISA110	1400Ω
Med (Medium)	2400Ω
High	6800Ω

- EIN (equivalent input noise) = -128dB measured at 60dB of gain with 150 Ohm terminating impedance and 20Hz/22kHz bandpass filter.
- Noise at main output with gain at unity (0dB) = -97dBu measured with a 20Hz/22kHz bandpass filter.
- Signal to noise ratio relative to max headroom (28dBu) = 125dB
- Signal to noise ratio relative to 0dBfs (+22dBu) = 119dB
- THD at medium gain (30dB) = 0.001% measured with a 1KHz -20dBu input signal and with a 20Hz/22kHz bandpass filter.
- Frequency response at minimum gain (0dB) = -0.25dB down at 20Hz and –3dB down at 120kHz.
- Frequency response at maximum gain (60dB) = –2.5dB down at 20Hz and –3dB down 120kHz.
- CMRR at full gain (60dB) = 80dB. (check 220 figure)

Line Input Response

- Gain range = -20dB to +10dB in 10dB steps.
- Input Impedance = 10kΩ from 10Hz to 200kHz.
- Noise at main output with gain at unity (0dB) = -91dBu measured with a 20Hz/22kHz bandpass filter.
- Signal to noise ratio relative to max headroom (28dBu) = 119dB
- Signal to noise ratio relative to 0dBfs (+22dBu) = 113dB
- THD at unity gain (0dB) = .002% measured with +4dBu input signal and with a 20Hz/22kHz bandpass filter.
- Frequency Response at unity gain (0dB) = 0.25dB down at 20Hz and –3dB down at 140kHz.

Instrument Input Response

- Gain range = 10dB to 40dB continuously variable
- Input Impedance = >1Meg Ohm.
- Noise at minimum gain (0dB) = -90dBu measured with a 20Hz/22kHz bandpass filter.
- Noise at maximum gain (40dB) = -78dBu measured with a 20Hz/22kHz bandpass filter.
- THD at minimum gain (0dB) = .006% measured with –10dBu input signal and with a 20Hz/22kHz bandpass filter.
- Frequency Response at 10dB gain = 0.2dB down at 26Hz and 0dB at 32kHz.
- Frequency Response at 40dB gain = -3dB down at 26Hz and –3dB down at 32kHz.

High Pass Filter

- Roll off = 18dB per octave 3 pole filter
- Frequency Range = continuously variable from 16Hz to 420Hz measured at the 3dB down point.

Input Meter

- Calibrated for 0dBfs = +22dBu and indicates the level after the High Pass Filter and before the Insert Send output.
- O/L LED = is lit when signal level reaches a level greater than 0dBfs.

Soft Limiter

- Threshold = -6dBfs (+16dBu)
- Limiter ratio is level dependent as follows:-

Signal Level	Input to Output Level Reduction Ratio
-6dBfs to –4dBfs	Infinite:1
-4dBfs to 0dBfs	2:1
0dBfs to +6dBfs	1.5:1

- Attack time = instant
- Release time = instant
- Noise = -95dBu measured with a 20Hz/22kHz bandpass filter.

ADC Meter

- 6 LED meter is calibrated relative to 0dBfs where 0dBfs = +22dBu (the maximum level which can be correctly converted by the optional internal A/D converter before overload occurs). The meter calibration points are as follows:

Meter panel calibration value in dBfs	Equivalent dBu value
0dBfs	+22dBu (the maximum level into the converter)
-2dBfs	+20dBu
-6dBfs	+16dBu
-12dBfs	+10dBu
-18dBfs	+4dBu (the average level to allow 20dB of headroom for EQ and dynamics processing).
-42dBfs	-20dBu

Warranty

All Focusrite products are covered by a warranty against manufacturing defects in material or craftsmanship for a period of one year from the date of purchase. Focusrite in the UK, or its authorised distributors worldwide, will do their best to ensure that any fault is remedied as quickly as possible. This warranty is in addition to your statutory rights.

This warranty does not cover any of the following:

- Carriage to and from the dealer or factory for inspection or repair.
- Labour charge if repaired other than by the distributor in the country of purchase or Focusrite in the UK.
- Consequential loss or damage, direct or indirect, of any kind, however caused.
- Any damage or faults caused by abuse, negligence, improper operation, storage or maintenance.

If a product is faulty, please first contact the dealer from which the product was purchased. If the product is to be shipped back, please ensure that it is packed correctly, preferably in the original packing materials. We will do our best to remedy the fault as quickly as possible.

Please help us to serve you better by completing and returning the Warranty Registration Card, or registering online at http://www.focusrite.com. Thank you.

Accuracy

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