

iK42 User Guide



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Important safety instructions

- Please read these instructions and keep them for future reference.
- Heed all warnings and follow all instructions.
- Keep this unit away from rain or moisture.
- Keep this unit away from heat sources.
- Keep this unit away from candles and flames.
- Don't block the ventilation openings.
- Install in accordance with Martin Audio instructions.
- Don't remove the protective earth from the power cable plug. This unit must be earthed.
- Protect the power cable from being walked on or damaged, particularly near the plugs.
- Unplug this unit during lightning storms or when unused for a long period of time.
- Don't open this unit. There is a risk of electric shock and there are no user-serviceable parts.
- If servicing is required, contact qualified service personnel.
- At the end of this unit's life, please use a recycling centre.

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Introduction

The Martin Audio iK42 advanced system amplifier represents current state-of-the-art technology in several areas. This product is the result of several years of research, from which many advances in switched mode power technologies and ever finer detail in signal processing have stemmed. Taking advantage of the latest advances in analogue to digital conversion technologies, the unit achieves performance levels among the very best that engineering permits.

Key features

- Four channels of sonically pure Class D amplification
 - Very high power density in 2U rack space
 - Up to 20,000 watts total power output, depending on load impedance
 - Packed with robust protection and monitoring systems to keep the show going
 - External Breaker Protection (EBP) limits the current draw to prevent breakers opening
 - Martin Audio minimal signal path design
 - Class leading sonic performance achieved by the use of state-of-the-art amplifier technologies and highly advanced DSP algorithms using Linea Micro Detail (LMD)
 - High performance 96 kHz DSP on all inputs and outputs
 - Up to 1,000 FIR filter taps at 48 kHz on each output channel (reserved for optimised systems)
 - High speed Ethernet communications supporting DHCP, static IP and auto IP and direct connection to a computer without the need for a router or a switch
 - Powerful Drive Module concept, abstraction from device centric to speaker based control
 - Innovative Component Presets to allow individual outputs to be used for selected drivers of a loudspeaker system
 - Unique VX limiter providing dynamic control for passive two-way enclosures
 - Unique LIR crossover shapes giving FIR-like performance without the drawbacks
 - Linear phase HF system EQ profiling which provides perfect integration between enclosures
- Innovative excursion control limiter with sliding High Pass Filter; limits only the damaging low frequencies
 - Transducer thermal modelling provides regulation limiters, addressing long term overload
 - Overshoot limiter governs amplitude of transient signals retaining average power whilst constraining peak energy
 - Analogue, AES3 and [Dante \(page 6\)](#) inputs
 - Automatic fallover from digital inputs

Other iKONS

There are three amplifiers in the iKON range. The only difference between these amplifiers is the total power output and the number of channels:

iKON	Channels	Power
iK42	4	20,000 watts
iK81	8	10,000 watts
iK41	4	6,000 watts

The power shown here is the total power output, depending on load impedance.

For further details, go to martin-audio.com.

Drive modules

The iK42 processor uses Drive Modules to order and group channels using a speaker-based approach to controlling, designing and recalling speaker configurations. A Drive Module is the processing provided by one input DSP block and a number of output DSP blocks. These blocks are associated by routing. For example, if input DSP block B is routed to outputs 3 and 4, then this is a two-way Drive Module with input DSP block B forming the 'Master' control, and output DSP blocks 3 and 4 providing the driver-related control. Overall, this forms the processing typically for one loudspeaker sub-system. You can use [Vu-Net \(page 33\)](#) to control and monitor the associated speaker.

The presets in the device are Drive-Module centric and are used to configure individual Drive Modules rather than the whole device. Importantly, Drive Modules move the focus away from the processing device and onto the loudspeaker systems. A Drive Module preset can be broken into components, allowing any output to be used

for any component within a Drive Module preset (that is, any driver in a loudspeaker sub-system).

For further details, see [Modules, preset components and snapshots \(page 13\)](#).

LIR crossover filtering

This device includes Linear Impulse Response (LIR) crossover filtering, which results in a linear phase crossover that has a constant delay regardless of frequency. This is unlike other types of crossover that delay different frequencies to different extents, thus smearing the arrival times. The LIR crossover can thus be described as having a flat group delay response and is thus entirely free of group delay distortion. The shape of the LIR crossover filter is quite similar to a 4th order or 24 dB/Oct Linkwitz-Riley filter and maintains zero phase difference between the adjacent bands across the crossover region to keep the polar response rock steady.

FIR linear phase equalisation

The input high shelf equalisers use finite impulse response (FIR) filtering to produce linear phase equalisation; that is, all frequencies are delayed by the same amount, perfectly preserving the transient response. This can also be important in applications where different amounts of EQ are applied to different parts of a speaker cluster, such as to add 'Throw' EQ boost so that parts of cluster that are throwing further can have HF absorption correction added. If this EQ is not linear phase, then the zones where the speakers combine may suffer frequency response anomalies. Linear phase equalisation solves this problem.

Non-Dante versions

Currently, all iKON amplifiers that we produce include Dante support. However, we have previously produced Dante and non-Dante versions of these amplifiers. If you have a non-Dante model, please ignore the Dante sections of this user guide.

If you are not sure whether your amplifier has Dante, check the Dante ports on the [back panel \(page 10\)](#) and if they are blanked off, you have a non-Dante model.

To upgrade a non-Dante model to Dante, you need to return your amplifier to your supplier and pay a fee. For further details, contact your supplier.

Quick start

This user guide tells you how to set up and configure your amplifier using the front panel. Note that most people connect the amplifier to a PC using Ethernet and set up and control the amplifier using [Vu-Net \(page 33\)](#) at the PC. In this case, you can use the front panel as a fall back when you don't have a PC available.

The only important setting where you have to use the front panel is IP mode (auto or static). You can't change this using Vu-Net.

Auto and static IP

If you only have a few iKON amplifiers, it is probably easiest and best to use auto IP.

If you have several iKON amplifiers, we recommend that you use static IP. This will allow you to create offline project files in [Vu-Net \(page 33\)](#) that target the static IP address. It will also make troubleshooting much easier.

- To choose the IP mode, see [IP mode \(page 31\)](#).
- To update the static IP address, see [Static IP address \(page 30\)](#),

For further details, see [Ethernet \(page 32\)](#).

Connecting the amplifier and PC

If you have several iKON amplifiers that you wish to control from a PC, you need to connect the PC and amplifiers using an Ethernet network.

If you only have one iKON amplifier, you can connect to a PC using an Ethernet network or alternatively using a direct connection (with no need for a network). With the direct connection you just run a network cable between the Ethernet socket on the amplifier and the Ethernet socket on the PC. With a small number of iKONs and no network, you could connect and set up the amplifiers one at a time using a direct connection.

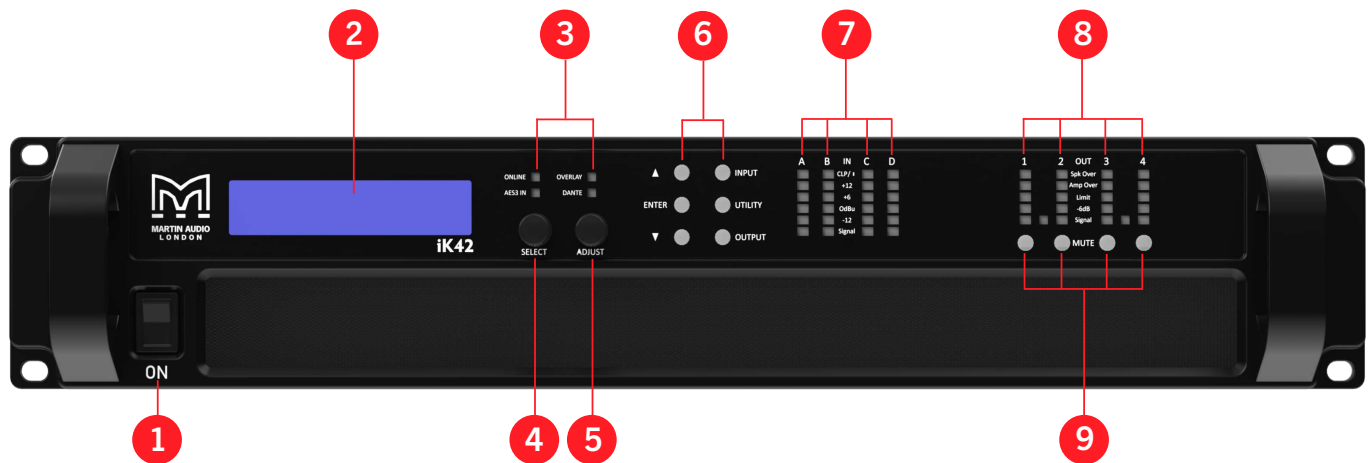
Installation

For details, see [Installation \(page 34\)](#).

Audio connections

For details, see [Audio connections \(page 36\)](#).

Front panel



- 1. Power switch.** The mains power on-off switch is at bottom front left. If the device is in **sleep mode** (page 28), you can wake it up using **Vu-Net** (page 33) or by switching this power switch off, pausing for a second or two and then switching it on again.
- 2. Graphical display.** The home screen shows an overview of the channel allocation. When you view the **input** (page 16), **output** (page 22) or **utility** (page 28) menus, the menu item names appear on the top line and the parameters to edit appear on the bottom line. If **SECURE** appears bottom left, this means that the amplifier is in **secure mode** (page 15) and you won't be able to edit any settings. To change the contrast, see **Screen contrast** (page 30).
- 3. Status Indicators.** The **ONLINE** indicator has three states:
 - **Off** – The unit isn't connected to **Vu-Net** (page 33).
 - **Flashing** – The unit is searching for an IP address. If the unit fails to find one, it will assign itself an IP address automatically and the indicator will stop flashing.
 - **On** – The unit is connected to **Vu-Net** (page 33).

The **AES3 IN** indicator is on when one or more of the inputs is configured to an AES3 source.

The **DANTE** indicator is on when one or more of the inputs is configured to Dante.

- 4. SELECT** (the left-hand dial) allows you to switch between parameters.
- 5. ADJUST** (the right-hand dial) allows you to edit parameters.



When you edit a parameter, the parameter changes straight away. There's no separate step to save changes.

- 6. Menu buttons.** When you press **INPUT**, **OUTPUT** or **UTILITY**, the button illuminates to show that you've selected this menu. The up ▲ and down ▼ buttons illuminate to show that you can use these buttons to page through the menu. The **ENTER** button illuminates whenever you need to press the **ENTER** button to confirm a change.

The **OUTPUT** button displays a menu of parameters associated with each **output channel** (page 22). The **INPUT** button displays a menu of parameters associated with **each input socket or each input DSP channel** (page 16). If you press **INPUT** or **OUTPUT** several times, you scroll through the input or output channels. After the last channel, you exit the menu and return to the home screen. The **UTILITY** button displays **miscellaneous parameters** (page 28). The

menus are mutually exclusive – if you press one of these buttons you switch to that menu. To exit from a menu, press **UTILITY**.

7. **Input Signal Indicators.** A set of five indicators show **Signal**, **-12, 0dBu**, **+6** and **+12** and **CLP/(mute)** for each of the DSP inputs A, B, C and D. The "signal present" indicators operate at approximately -40 dBu. The **CLP/(mute)** indicators warn you of input overload and operate at 1 dB before clip. This indicator also shows you when the [input is muted \(page 16\)](#).
8. **Limiter Indicators.** The output indicators show the status of the limiter and the output level. The level shown is the level before the limiter, referenced to the limiter threshold.

The **Signal** indicator shows when a signal is present on the output.

The **-6dB** indicator shows that the signal has reached 6 dB below the limiter threshold.

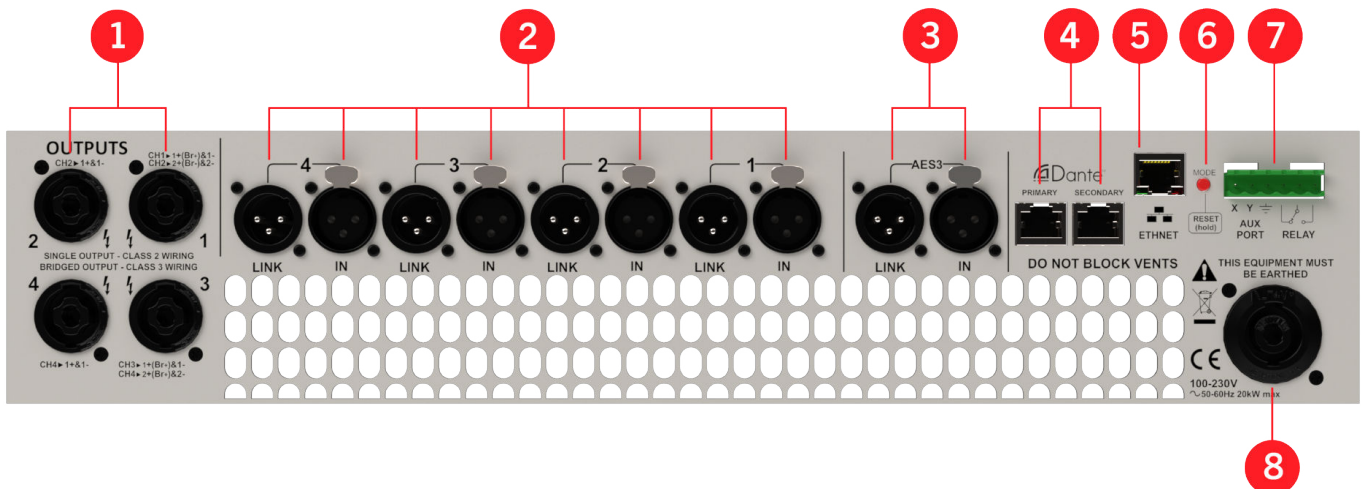
The **Limit** indicator shows that the threshold of that output channel has been reached.

Bridge Indicator. This illuminates when the channel pair is in [Bridge mode \(page 24\)](#). The amplifier uses the settings for the left channel of the pair.

Amplifier Indicator. This indicates when the amplifier protection systems are reducing the gain to keep the parameters of the amplifier within specification, or when the channel is clipping.

Driver Indicator. This indicates a signal 6 dB higher than the limiter threshold, or that the threshold of the excursion limiter has been exceeded, or that the thermal limiter is active. Please note that because of the long release time of the thermal limiter, this indicator may remain illuminated for several seconds after signal on that channel is reduced.
9. **Mute Buttons.** DSP output mute status is indicated and controlled by an illuminated button for each channel. If the amplifier protection systems mute a channel, the mute button flashes.

Back panel



- Loudspeaker connectors.** The amplifier has four speakON outputs labelled 1, 2, 3 and 4. For details, see [Output connections \(page 36\)](#).
- Analogue audio input connectors.** All audio connections are fully balanced and wired: pin 1 to ground (as required by the AES48 standard), pin 2 hot and pin 3 cold. For details, see [Audio connections \(page 36\)](#).
Analogue audio link connectors. Carries a duplicate (parallel) connection (to another amplifier for example).
- AES3 audio input connectors.** For inputting digital audio signals. The Input is fully balanced and wired: pin 1 to ground, pin 2 data+ and pin 3 data. The link connector allows a buffered AES3 signal to be passed on to another device.
- Dante ports.** Primary and Secondary [Dante ports \(page 6\)](#).
- Ethernet port.** Use this port to connect your amplifier and PC so that you can control the amplifier using [Vu-Net \(page 33\)](#). This software also allows you to update the amplifier firmware and to integrate with other Martin Audio products such as MLA and CDD-Live.
- Mode.** If you press and hold this button for about six seconds, the unit changes the IP mode to auto and searches for an IP address. If the IP mode is already auto, it just searches for an IP address. In both cases, if the amplifier fails to find an IP address, it allocates one. You can also [change IP mode using the utility menu \(page 31\)](#).
- Six-pin Phoenix connector.** Also known as a Euroblock connector. The first three pins are the [AUX port \(page 40\)](#) and the last three pins are the [Fault relay \(page 42\)](#).
- Power Inlet.** Connect to a suitable mains electricity supply using an earthed 32 amp powerCON connection power lead. The device has a switch mode power supply that is capable of operating with a nominal mains voltage of 100 V to 230 V, 50/60 Hz without reconfiguration. For details, see [AC power connection \(page 35\)](#).



The device must be earthed to a suitable power earth. Failure to do so will invalidate the warranty and could be hazardous. It could also affect performance and operation.

Example system setup

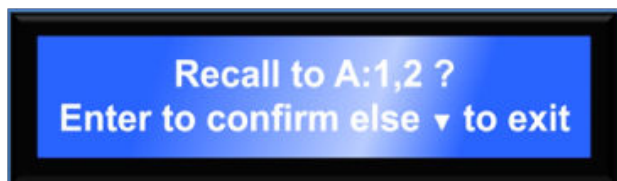
The iK42 is not only a powerful amplifier but it has extremely comprehensive processing abilities allowing complex systems to be designed to suit any application. However, often all you need to do is recall the appropriate preset, connect your speakers and source and away you go. As examples, this section shows you how to set up a system using an iK42 and XE500 or XE300 stage monitors.

To load XE500 presets

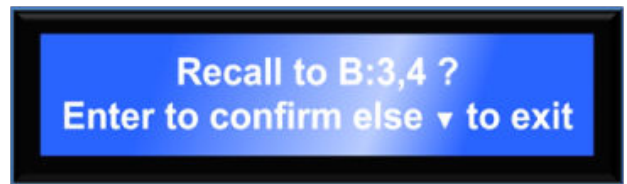
1. Connect the amplifier to the mains and connect the input feed and the monitors. The input feed can be analogue, AES3 or [Dante \(page 6\)](#).

Connect the XE300 to speakON outputs 1 and 3 as pins 2+/- connect channels 2 and 4 specifically for bi-amp operations. For further details, see [Audio connections \(page 36\)](#).

2. Power up the amplifier.
3. Press **INPUT** so that **A** appears top left. This shows that you are editing input channel A. Press the down arrow button **▼** once so that **RECL** (for Recall Preset) appears top left.
4. Rotate **ADJUST** (the right-hand dial) so that **XE500** appears bottom right.
5. Press **ENTER**. The following message will appear and the **ENTER** button will start flashing:



6. Press **ENTER** to confirm. Input A now feeds output 1 with XE500 LF parameters and output 3 with XE500 HF parameters.
7. Press **INPUT** so that **B** appears top left. Note that when you switch from one input to the next, the amplifier shows the same menu item, in this case **RECL**.
8. Rotate **ADJUST** (the right-hand dial) to select the **XE500** preset and press **ENTER**.



9. Press **ENTER** to confirm. Preset loading is now complete and you can press **UTILITY** to escape.

The home screen shows that DSP A and B are loaded with the XE500 preset. Input 1 is routed to output 1 and 2, input 2 is routed to outputs 3 and 4. Inputs 3 and 4 aren't used and therefore aren't routed anywhere.

The amplifier is ready for use.

To load XE300 bi-amp presets

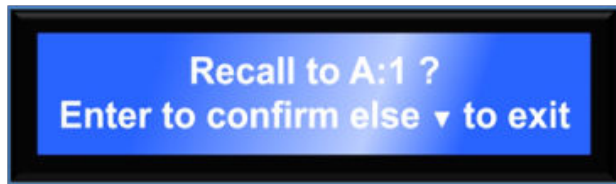
Follow the instructions in the previous section, but select preset **XE300 BI** instead of preset **XE500**.



Connect speakON outputs 1 and 3 in exactly the same way as with the XE500 monitors.

To load XE300 passive presets

1. Connect the amplifier to the mains and connect the input feed and the monitors. The input feed can be analogue, AES3 or [Dante \(page 6\)](#).
With the XE300 in passive mode, you can connect directly to speakON outputs 1, 2, 3 and 4.
2. Power up the amplifier.
3. Press **INPUT** so that **A** appears top left. This shows that you are editing input channel A. Press the down arrow button **▼** once so that **RECL** (for Recall Preset) appears top left.
4. Rotate **ADJUST** (the dial on the right) until the XE300 passive preset appears.
5. Press **ENTER** and the following message appears:



6. Press **ENTER** to confirm.
7. Press **INPUT** again to select input channel B.
8. Rotate **ADJUST** (the dial on the right) until the XE300 passive preset appears.

Press **ENTER** twice to confirm. When the preset is loaded press **INPUT** again to load the same preset into input C. Then press **INPUT** again to enter the preset into input D. Press **UTILITY** to exit.

The home screen now shows that DSP A, B, C and D are loaded with the XE300 passive preset. Input 1 is routed to output 1, input 2 to output 2 and so on.

The amplifier is ready for use.

Operation

Starting up the unit

The power switch is on the front of the unit at the lower left. When you switch on the amplifier, it goes through a start-up cycle and checks all the sub-systems. The screen shows you progress and when complete, the home screen shows the drive module configuration.

Modules, preset components and snapshots

A Drive Module represents a loudspeaker sub-system (for example, a sub and full range) and comprises one input channel routed to a number of output channels. The size of the Drive Module is determined by the number of outputs it includes. The amplifier can contain up to four Drive Modules.

DSP input A	Output 1
	Output 2

A Module Preset is a collection of settings (parameters) for a Drive Module of a particular size. The Preset contains a set of parameters for one input and a set of parameters for each of the outputs in the module. The most obvious example would be a bi-amp system, either individual cabinets such as a BlacklineX sub used with one of the full range cabinets, or a bi-amp speaker such as the XE500 monitor.

When you recall a Module Preset, it will automatically change the routing between Input DSPs and Outputs, consuming a number of outputs according to the size of the Drive Module. Recalling a Module Preset thus always creates a Drive Module with consecutive outputs.

Input component parameters	Output component parameters
	Output component parameters

A Component is a collection of settings (parameters) for one channel (input or output). Any one of the output components in a Module Preset may be recalled to any individual output.

A Snapshot is a device-wide representation of most of the settings in the device. This is represented as four Input Component numbers, four Output Component numbers, plus a number of machine-centric settings such as routing and input/output analogue/digital selection and so on.

Device settings	Input A component number	Output 1 component number
	Input B component number	Output 2 component number
	Input C component number	Output 3 component number
	Input D component number	Output 4 component number

Home screen

The home screen shows an overview of the configuration of the device. The device name is at the top with the four Drive modules (page 5) below. For each drive module, the top line shows the input DSP channel (A, B, C and D) and the name you've given each channel. The bottom line shows the physical input number and a list of the outputs that are routed from the drive module.

To update the device name and the channel names, you need to use [Vu-Net](#) (page 33). You can't update these at the [front panel](#) (page 8).

UNIT 001			
A Subs	B Left	C Right	D Spare
1+1:2	3:2	4:3	1:4

Drive modules

The iKON amplifiers use drive modules to represent loudspeaker sub-systems. Drive modules result in a less processor-centric and more speaker-orientated system design. A drive module is defined as the processing provided by one Input DSP and a number of outputs, which are associated with one-another by means of routing. For example, if Input DSP B is routed to outputs 3 and 4, then this is a two-way drive module; input DSP B forming the 'master' control, and output DSP 3 and 4 providing the driver-related control. The input DSP parameters control the drive module (and thus the speaker sub-system). You can control and monitor sub-systems using [Vu-Net](#) (page 33).

Drive module presets

Presets do not change the settings device-wide. Rather, recalling a Module Preset creates a Drive Module by 'consuming' a number of consecutive outputs and setting

up routing between the Input the preset was recalled on and those outputs. The parameters in that Drive Module are then set according to the parameters in the components in the Module Preset.

You can if you wish create modules with non-consecutive outputs by manually manipulating the routing and then recalling Component Presets to the individual outputs. You could then store the resulting system in a Snapshot, though you wouldn't be able to store the Module in a Module Preset.



When a Module Preset Recall consumes outputs to construct a module, it treats a pair of Bridged outputs as a single channel, so recalling a two-way Module Preset will consume three output channels if a Bridged pair is encountered. For further details, see [Bridge mode \(page 24\)](#).



DSP inputs are not the same as physical inputs. The iK42 has four audio inputs and four DSP inputs. This is a matrix mixing system where any physical inputs, be they analogue, AES3 or Dante, can drive any number of DSP inputs.

Component presets

A Component Preset represents the processing for just one output. Any part of a Module Preset can be recalled to any one output. A Drive Module comprised of parameters which have been recalled to its outputs using Component Preset Recalls can then be saved into another Module Preset provided the outputs remain consecutive (i.e. you have not changed the routing manually). If you have changed the routing manually, you can save the whole arrangement into a Snapshot. For further details, see [Modules, preset components and snapshots \(page 13\)](#).

Factory module presets

The amplifier contains a small library of factory presets designed to suit a range of enclosures. However, [Vu-Net \(page 33\)](#) has an extensive library of presets, so it's best to install presets using [Vu-Net \(page 33\)](#).

Factory presets may contain some parameters that are fixed and hidden from view and other parameters that you can edit. The number and type of hidden parameters is dependent on the factory preset. Typically the settings that are a function of the loudspeaker cabinet design are hidden. Factory presets are locked (as indicated by a 'box' symbol after the preset name), which means you can't overwrite them. However, you can create and update your own presets.

Storing module presets

For details of how to store drive modules, see [Store preset \(page 17\)](#).



You can only store drive module presets where the outputs are consecutive.



If you store a drive module preset at the [front panel \(page 8\)](#), you won't be able to edit the component names. To change component names, you need to save the module preset using [Vu-Net \(page 33\)](#).

Recalling module presets

For details, see [Recall preset \(page 16\)](#).

Snapshots

You can recall snapshots using:

- [Vu-Net software \(page 33\)](#)
- [The amplifier front panel \(page 30\)](#)
- Signals sent to the [AUX port \(page 40\)](#) on the rear of the amplifier.

When you recall a snapshot it recalls components to each input and output and it may change device-wide settings, effectively recalling a processor-wide preset.

The parameters inside the drive modules are not individually stored in snapshots. Recalling a snapshot triggers the recall of the appropriate input and output components, rather than restoring the parameters that were active when the snapshot was stored. This has the distinct advantage that the library of OEM presets can be updated without having to be concerned about what parameters might have been saved in snapshots. It does however require that any existing edits to the parameters

in drive modules are stored into drive module presets before you store the snapshot.

For further details, see [Modules, preset components and snapshots \(page 13\)](#).

Secure mode

Secure mode stops people from updating settings at the front panel. When the amplifier is connected to [Vu-Net \(page 33\)](#), the amplifier automatically switches to secure mode. If you're not using [Vu-Net \(page 33\)](#), you can switch secure mode on and off manually.

Note that in secure mode you can still:

- Mute and unmute output channels.
- View but not edit the [input \(page 16\)](#), [output \(page 22\)](#) and [utility settings \(page 28\)](#).

In secure mode, all indicators operate normally and the Ethernet port is unaffected.

To switch secure mode on or off

- Press and hold the **UTILITY** button for about 8 seconds.



If you're connected to [Vu-Net \(page 33\)](#), you can't switch off secure mode.

Revert to factory settings

If you wish to restart with the amplifier in a clean state, you can erase all settings and revert to the factory defaults. Note that you can't do this if the amplifier is in [secure mode \(page 15\)](#).

To revert to factory settings



This deletes all settings, so only do this if you're sure it's the right thing to do.

1. Simultaneously press and hold the up arrow button ▲ and the mute 1 button for about 8 seconds.
2. Press **ENTER** to confirm or press the down arrow button ▼ to cancel.

Updating inputs

The input menu allows you to choose:

- **GAIN** – see [Input gain, polarity and muting \(page 16\)](#)
- **RECL** – see [Recall preset \(page 16\)](#)
- **STOR** – see [Store preset \(page 17\)](#)
- **IN Input** – see [Routing \(page 17\)](#)
- **IN Style** – see [Routing \(page 17\)](#)
- **TYPE** – see [Input and fallover type \(page 17\)](#)
- **TRIM** – see [Trim digital input gain \(page 18\)](#)
- **Latency** – see [Drive module latency \(page 19\)](#)
- **FIR** – see [FIR high shelf EQ \(page 19\)](#)
- **EQ1 to EQ6** – see [Input parametric EQ filters \(page 20\)](#)
- **EQa and EQb** – see [FIR low shelf EQ \(page 20\)](#)
- **HPF** – see [High pass filter \(page 20\)](#)
- **DEL** – see [Input delay \(page 20\)](#)

Selecting the input channel

Press **INPUT** to select the input menu. The letter at top left shows the channel that you're editing and this could be any of the input channels: A, B, C or D. For example:

1. Press **INPUT** several times until channel D appears top left.
2. Press **UTILITY** to exit the menu.
3. Press **INPUT** and you go back to channel D.

If you use **INPUT** to scroll through the channels, the first channel is always A:

1. Press **INPUT** several times until channel D appears top left.
2. Press **INPUT** rather than **UTILITY** to exit the menu.
3. Press **INPUT** and you go back to channel A.

Note that when you select an input menu item and change to another channel, you remain on the same menu item. For example:

1. Press **INPUT** so that channel A appears top left.
2. Press the down arrow button ▼ until **RECL** appears top left (normally one press).

3. Press **INPUT**. You are now on menu item **RECL** for channel B.
4. Press **INPUT** three times to exit the menu.
5. Press **INPUT** again and you go back to channel A and menu item **RECL**.

Input gain, polarity and muting

Use the **GAIN** menu item to update the input channel gain, polarity and muting.

To update gain, polarity and muting

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. **GAIN** should appear top left. If not, press the down arrow button ▼ several times until it does.
3. Rotate **ADJUST** (the right-hand dial) to choose the gain (–40 to 20 dB in 0.2 increments).
4. Rotate **SELECT** (the left-hand dial) to choose **Pol** for polarity and rotate **ADJUST** (the right-hand dial) to choose **Norm** to use normal polarity or **Rev** to use reverse polarity.
5. Rotate **SELECT** (the left-hand dial) to choose **Mute** and rotate **ADJUST** (the right-hand dial) to select **Norm** to unmute or **Mute** to mute this input channel.
6. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Recall preset

Use the **RECL** menu item to recall presets. The presets included with the firmware have a square next to the number (with firmware 1.678 there are six presets). You can also create your own presets using [Store preset \(page 17\)](#). These presets won't have a square next to them.

To recall a preset

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the down arrow button ▼ several times until **RECL** appears top left (usually one press).

3. Rotate **ADJUST** (the right-hand dial) to choose the preset.
4. Press **ENTER** to recall this preset.
5. Press **ENTER** to confirm or press the down arrow button **▼** to cancel.
6. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Store preset

Use the **STOR** menu item to store presets. To recall presets, see [Recall preset \(page 16\)](#).

To store a preset

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the down arrow button **▼** several times until **STOR** appears top left (usually two presses).
3. Rotate **ADJUST** (the right-hand dial) to choose preset number (1 to 50).
4. Press **ENTER** to choose this preset number.
5. You can now enter a name for the preset, by rotating **ADJUST** (the right-hand dial) to select each character and rotating **SELECT** (the left-hand dial) to select the position in the name.
6. Press **ENTER** to choose this name.
7. Press **ENTER** to confirm or press the down arrow button **▼** to cancel.
8. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Routing

Use the **IN** menu item to set up routing.

Routing allows you to route any physical analogue or digital signal channel to any DSP input. This is effectively a matrix mixing system where all DSPs can be driven from any one input or from any two, three or four inputs. Summed inputs have 6dB of attenuation so that a sum of largely similar programme material remains at the correct calibrated level.

To set up routing

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).

2. Press the down arrow button **▼** several times until **IN** appears top left (usually three presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the routing for this channel. This can be 1, 2, 1+2, 3, 4, 3+4, 1+3, 1+4, 2+3, 2+4, 123, 1234, 124, 134, 234.
4. If you are on channel A, press the down arrow button **▼** so that **STYLE** appears top right and rotate **ADJUST** (the right-hand dial) to choose **Combo** or **Matrix**. This menu item isn't available for channels B, C and D and isn't support by [Vu-Net \(page 33\)](#).
5. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Input and fallover type

The **TYPE** screens allow you to select which input type to use for the channel (analogue, AES3 or Dante) and to specify what the amplifier should do if digital input fails or is inactive. There are two **TYPE** screens (one added with firmware 1.678).

Fallover

The second **TYPE** screen allows you specify the input and fallover type. You do this under the heading **Fallover** on the left of the screen. This can be:

- **Manual** – No automatic fallover. The amplifier uses the input type you specify on the right of the screen under the heading **Type** (see the next section).
- **Dante>AES3** – Use Dante and if this fails automatically switch to AES3.

This option is only available for channels A and B as channels C and D don't support AES3.

- **Dante>Analog** – Use Dante and if this fails automatically switch to analogue.

This option is only available for channels A and B. To select the same option for channels C and D, use **Auto**.

- **AES3>Analog** – Use AES3 and if this fails automatically switch to analogue.

This option is only available for channels A and B as channels C and D don't support AES3.

- **Auto** – For channels A and B, use Dante and if this fails, use AES3 and if this fails, use analogue. For channels C and D, use Dante and if this fails use analogue. This setting means that you can just plug

in Dante, AES3 or analogue and the amplifier will adapt to the signal.

Type

If you select a **Fallover** type of **Manual**, you need to specify the input type under the heading **Type** on the right of the screen:

- For channels A and B, you can set **Type** to Dante, AES3 or analogue.
- For channels C and D, you can only set **Type** to Dante or analogue, as these channels don't support AES3.

For **Fallover** types other than **Manual**, the amplifier shows the input type it's using on the right of the screen. For example, if **Fallover** is **Dante>AES3** and Dante fails, the type shown on the right will automatically change from Dante to AES3 at the same time that the input changes. For **Fallover** types other than **Manual**, you can't update the **Type**. The only exception is when **Speed** is set to **Latch** (see below) and digital input has failed. In this case, you can switch back manually using the **Type** setting.

Detect and speed

The first **TYPE** screen (added in firmware 1.678) allows you to specify how fallover works. This has two settings, **Detect** on the left and **Speed** on the right.

You can set **Detect** to:

- **Audio** – Fallover as soon as Dante audio stops even if the Dante network is still responding.
- **Dante** – Fallover only if Dante fails (Dante stops responding or has synchronisation problems).

The **Detect** setting doesn't apply to AES3.

You can set **Speed** to:

- **Auto** – Fallover as soon as digital input fails and fallback to it as soon as it returns.
- **Delayed** – Fallover as soon as digital input fails and if it returns, switch back to it after five minutes.
- **Latch** – Fallover as soon as a digital input fails but don't switch back to it. If you do want to switch back, you can do so manually using the **Type** setting (see above).

For Dante, what counts as failure depends on the **Detect** setting (see above).

To change the input and fallover type

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the down arrow button ▼ several times until the second **TYPE** screen appears (usually six presses).
3. Rotate **ADJUST** (the right-hand dial) to choose **Fallover** (see above).
4. If the fallover type is **Manual**, rotate **SELECT** (the left-hand dial) to choose **Type** and rotate **ADJUST** (the right-hand dial) to choose the input type (see above).
5. If the fallover type is anything other than **Manual**, press the up arrow button ▲ once to go to the first **TYPE** screen.

Rotate **ADJUST** (the right-hand dial) to choose **Detect** (see above).

Rotate **SELECT** (the left-hand dial) to choose **Speed** and rotate **ADJUST** (the right-hand dial) to select **Auto**, **Delayed** or **Latch** (see above).

6. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Trim digital input gain

This amplifier can use analogue inputs, AES3 digital inputs or [Dante digital inputs \(page 6\)](#). The **TRIM** menu item allows you update digital input gain.

AES inputs

When you [assign a DSP input channel to AES3 \(page 17\)](#), the **AES3** indicator illuminates.

There is no standard for the relative gains between analogue and AES3 inputs so depending on the levels delivered by the audio sources, you may need to adjust the digital input gain trims to normalise them. For example:

- To achieve 0dBFS = +18dBu, set the AES3 trim to –2dB.
- To achieve 0dBFS = +24dBu, set the AES3 trim to +4dB.

This device will automatically lock onto any sample rate within the range 28kHz and 108kHz.

Dante Inputs

Using [Dante input \(page 6\)](#), you can select any channel or channels as being sourced from this network. To do this,

connect the audio network connection to the connection on the rear of the Device and set the relevant [Input channels to Dante](#) (page 17).

When you select Dante as part of the input routing, the **Dante** indicator illuminates. This indicator will come on even if there are no cables plugged into the Dante ports on the Device.

As with the AES3 inputs, you can set the relative gain between analogue input and audio sourced from Dante.

The amplifier will automatically select the correct sample rate from the incoming stream.

For other details on the operation of Dante, please refer to the relevant manufacturer's documentation.

To trim digital input gain

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel](#) (page 16).
2. Press the down arrow button ▼ several times until **TRIM** appears top left (usually seven presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the AES3 gain (–40 to 20 dB in 0.2 increments).
4. Rotate **SELECT** (the left-hand dial) to choose **Dante** and rotate **ADJUST** (the right-hand dial) to choose the Dante gain (–40 to 20 dB in 0.2 increments).
5. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Drive module latency

Use the [Latency](#) menu item to see the current [Latency delay](#) (page 41).

To see the latency

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel](#) (page 16).
2. Press the down arrow button ▼ several times until **Latency** appears top left (usually eight presses).



The latency shown on the front panel doesn't include the [input and output latencies](#) (page 41).

3. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Input FIR

For input FIR, there are nine stages of equalisation available for each channel:

- Three shelving filters. These are the [high shelf filter](#) (page 19) and the [two low shelf filters](#) (page 20).
- Six parametric filters (page 20).

You can also apply a [high pass filter](#) (page 20).

FIR high shelf EQ

Use the **FIR** menu item to update frequency and gain for the FIR high shelf EQ. For details of FIR, see [input FIR](#) (page 19).

The input high shelf EQ is implemented using a Finite Impulse Response (FIR) filter and exhibits a linear phase response; that is all frequencies are delayed by the same amount. This can be important in applications where different amounts of EQ are applied to different parts of a speaker cluster, such as to add 'Throw' EQ boost so that parts of cluster that are throwing further can have HF absorption correction added. If this EQ isn't linear phase, then the zones where the speakers combine may suffer frequency response anomalies. Being a linear phase FIR equaliser, this necessarily introduces some [latency delay](#) (page 41), which is constant regardless of the settings.

If you switch off the FIR high shelf EQ (by setting the **Enable** parameter to **Off**) it's removed from the signal path entirely and no longer adds any latency.

To update the FIR high shelf EQ

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel](#) (page 16).
2. Press the down arrow button ▼ several times until **FIR** appears top left (usually nine presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the frequency.
4. Rotate **SELECT** (the left-hand dial) to choose **Enable** and rotate **ADJUST** (the right-hand dial) to select **On** or **Off**.
5. Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to change the cut and boost in 0.2 dB increments.
6. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Input parametric EQ filters

For each of the six input parametric EQ filters you can specify the frequency, bandwidth and gain. For details of FIR, see [input FIR \(page 19\)](#).

To update the six parametric filters

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the up arrow button ▲ several times until **EQ1** appears top left (usually five presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the frequency.
4. Rotate **SELECT** (the left-hand dial) to choose **Width** and rotate **ADJUST** (the right-hand dial) to update the bandwidth in **Q or octaves (page 30)**.
5. Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to update the gain (0.2 dB increments).
6. Press the up arrow button ▲ again so that **EQ2** appears top left. Update the second parametric filter and repeat for all six parametric filters.
7. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

FIR low shelf EQ

Use the **EQa** and **EQb** menu items to update frequency, slope and gain for the two low shelf EQ filters. For details of FIR, see [input FIR \(page 19\)](#).

Note that in [Vu-Net \(page 33\)](#):

- **EQa** is **LS1** (low shelf one).
- **EQb** is **LS2** (low shelf two).

To update the FIR low shelf EQ

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the up arrow button ▲ several times until **EQa** appears top left (usually three presses). This is low shelf one.
3. Rotate **ADJUST** (the right-hand dial) to update the frequency.
4. Rotate **SELECT** (the left-hand dial) to choose **Slope** and rotate **ADJUST** (the right-hand dial) to update the slope in dB. This is shown as Q in [Vu-Net \(page 33\)](#).

5. Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to change the cut and boost in 0.2 dB increments.
6. Press the up arrow button ▲ until **EQb** appears top left. This is low shelf two. Edit this in the same way as low shelf one.
7. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

High pass filter

Use the **HPF** menu item to edit the high pass filter for the input signal. You can update the frequency and select the following filter types:

1st	Not supported in Vu-Net (page 33)
Bes12	Bessel 12 dB
But12	Butterworth 12 dB
LR12	Linwitz-Riley 12 dB
Bes18	Bessel 18 dB
But18	Butterworth 18 dB
Bes24	Bessel 18 dB
But24	Butterworth 24 dB
LR24	Linwitz-Riley 24 dB
Har4th	Hardman 4th order – Not supported in Vu-Net (page 33)

To select the high pass filter

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the up arrow button ▲ several times until **HPF** appears top left (usually two presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the frequency. If you want to switch off the high pass filter, rotate to the left to set the frequency to **<<Out**.
4. Rotate **SELECT** (the left-hand dial) to choose **Shape** and rotate **ADJUST** (the right-hand dial) to choose the high pass filter type (see above).
5. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Input delay

Use the **DEL** menu item to add a delay to an input channel. See also [Latency delay \(page 41\)](#).

To update input delay

1. Press **INPUT** to select the channel. For details, see [Selecting the input channel \(page 16\)](#).
2. Press the up arrow button ▲ several times until **DEL** appears top left (usually one press).
3. Rotate **ADJUST** (the right-hand dial) to change the delay in ms. To remove the delay, set the delay to 0 ms.
4. To step through the channels press **INPUT** several times, or press **UTILITY** to exit the menu.

Updating outputs

The output menu allows you to choose:

- **GAIN** – see [Output gain and polarity \(page 22\)](#)
- **RECL** – see [Recall preset \(page 22\)](#)
- **DRIV** – see [Drive load \(page 22\)](#)
- **ROUT** – see [Output routing \(page 23\)](#)
- **LIM** – see [Output VX limiter \(page 23\)](#)
- **Tmax** – see [Output thermal limiter \(page 23\)](#)
- **Xmax** – see [Output excursion limiter \(page 24\)](#)
- **AMP** – see [Bridge mode \(page 24\)](#)
- **EQ<** – see [Output high shelf filter \(page 25\)](#)
- **EQ1 to EQ8** – see [Output parametric EQ filters \(page 25\)](#)
- **EQ>** – see [Output low shelf filter \(page 26\)](#)
- **LPF** – see [Output low pass filter \(page 26\)](#)
- **HPF** – see [Output high pass filter \(page 26\)](#)
- **DEL** – see [Output delay \(page 27\)](#)

Selecting the output channel

Press **OUTPUT** to select the output menu. The number at top left shows the channel that you are editing and this could be any of the output channels: 1, 2, 3 or 4.

Selecting output channels works in the same way as selecting input channels; for details, see [Selecting the input channel \(page 16\)](#). The only difference is that input channels have letters (A, B, C and D) and output channels have numbers.

Output gain and polarity

Use the **GAIN** menu item to adjust the output channel gain and polarity.

To update gain and polarity

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. **GAIN** usually appears top left. If not, press the down arrow button ▼ several times until it does.
3. Rotate **ADJUST** (the right-hand dial) to choose the gain (–40 to 20 dB in 0.2 increments).

4. Rotate **SELECT** (the left-hand dial) to choose **Pol** and rotate **ADJUST** (the right-hand dial) to choose **Rev** for reverse polarity or **Norm** for normal polarity.
5. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Recall preset

Use the **RECL** menu item to recall a preset to an output channel. The **Comp** parameter shows a component number with the following format:

ModulePreset.ComponentNumber

For example, 3.2 is module preset 3 and component number 2.

The preset name is shown to the right. With firmware 1.678, the amplifier has 11 output presets.

To recall a preset

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **RECL** appears top left (usually one press).
3. Rotate **ADJUST** (the right-hand dial) to select a preset.
4. Press **ENTER** to recall this preset.
5. Press **ENTER** to confirm or press the down arrow button ▼ to cancel.
6. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Drive load

Use the **DRIV** menu item to adjust the optimal drive level for a given driver impedance. The screen also shows the live measured impedance value. If it displays a question mark "?", this means that there's insufficient signal level to measure the impedance.

To update the drive load

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **DRIV** appears top left (usually two presses).

3. Rotate **ADJUST** (the right-hand dial) to choose the load. The possible values are: 25V Line, 70V Line, 100V Line, 2 ohm, 3 ohm or Auto.
4. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output routing

Outputs can be driven from any DSP input. This routing is the fundamental means by which Drive Modules are created. By default, routing always uses consecutive channels. However, you can create non-consecutive Drive Modules. For further details, see [Drive modules \(page 5\)](#).

To update routing

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **ROUT** appears top left (usually three presses).
3. Rotate **ADJUST** (the right-hand dial) to choose None, DSP A, DSP B, DSP C or DSP D.
4. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Limiters

The iKON amplifiers include three limiters in the output signal path:

- [VX limiter \(page 23\)](#)
- [Thermal limiter \(page 23\)](#)
- [Excursion limiter \(page 24\)](#)



Although these limiters offer protection for amplifiers and drivers, they can never protect from all possible scenarios. Martin Audio is not responsible for any damage that might occur.

Output VX limiter

Use the **LIM** menu item to configure the VX limiter. This is one of [three output limiters \(page 23\)](#) in the amplifier.

The VX limiter is a peak-detecting signal limiter. The VX mode parameter determines the style of limiter. When Virtual Crossover (VX) mode is off, the limiter is

controlled in a conventional manner; the only controls being Threshold and Overshoot.

The Overshoot limiter prevents the signal from exceeding the threshold during the attack phase of the main limiter by more than a predetermined amount. The optimal Overshoot setting is usually about 8 dB. Overshoot settings lower than this will sound progressively 'harder'.

When VX mode is on, you can choose the crossover point of a 'virtual crossover', which incorporates two limiters per output so you can individually limit the drivers in a passive two-way enclosure using individual thresholds and optimised attack and release characteristics for each. The Threshold of the second 'Hi' limiter is set relative to the threshold of the first 'Lo' limiter.

This Limiter introduces some delay. With VX mode off, this delay will depend on the lowest high-pass frequency used in the crossover filters in a given [Drive Module \(page 5\)](#). With VX mode on, the delay is related to the Split frequency. This delay will be applied to all of the outputs in a given Drive Module to keep them in phase. For further details, see [Latency delay \(page 41\)](#).

To update limiter settings

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **LIM** appears top left (usually three presses).
3. Rotate **ADJUST** (the right-hand dial) to change **Thresh** the threshold value.
4. Rotate **SELECT** (the left-hand dial) to choose **Over** and rotate **ADJUST** (the right-hand dial) to change the overshoot value (see above).
5. Rotate **SELECT** (the left-hand dial) to choose **VxMde** and rotate **ADJUST** (the right-hand dial) to switch VX mode on or off (see above).
6. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output thermal limiter

The thermal limiter is designed to protect the driver against damage due to over-heating. This is one of [three output limiters \(page 23\)](#) in the amplifier. You configure this using the **Tmax** menu item.

The thermal limiter models the temperature of the driver and constrains the output signal level in order to keep the average output power below a predetermined limit.

It applies attack and release characteristics to go some way towards modelling the complex thermal circuit of a driver's voice coil and magnet assembly.

There are three parameters:

- **Threshold** – The continuous RMS voltage which the driver should be able to withstand. This is calibrated at the output of the amplifier. You can switch off the thermal limiter by setting the threshold to off.
- **Attack** – The time-constant of the speed at which the driver heats up (in seconds).
- **Release** – The time-constant of the speed at which the driver cools down (expressed as a multiple of the attack time).

To update thermal limiter settings

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **Tmax** appears top left (usually five presses).
3. Rotate **ADJUST** (the right-hand dial) to change **Thresh** the threshold value. To switch off the thermal limiter, rotate the dial to the right until **Off** appears.
4. Rotate **SELECT** (the left-hand dial) to choose **Attack** and rotate **ADJUST** (the right-hand dial) to change the attack value (0.01 to 34.4 s).
5. Rotate **SELECT** (the left-hand dial) to choose **Rel** and rotate **ADJUST** (the right-hand dial) to change the release value (1.0X to 16.4X).
6. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output excursion limiter

Use the **Xmax** menu item to update the excursion limiter settings. This is one of [three output limiters \(page 23\)](#) in the amplifier.

The excursion limiter protects the driver against excessive linear movement of the cone and voice-coil which could otherwise cause mechanical damage. Since this movement (excursion) is largely related to the inverse of the signal frequency, drivers are prone to being damaged by very low frequencies. This limiter is progressively more sensitive at lower frequencies and, rather than varying the gain to provide the limiting action, it uses a sliding high-pass filter to progressively curtail the low-frequency response, effectively limiting the linear excursion to below the Xmax specification of the driver.

To set the limiter up, it is necessary to know the shape of the family of Excursion versus Frequency curves of the driver for various drive voltage levels. A curve should then be chosen where the slope is high, where it passes through the specified Xmax value for the driver. The peak voltage and frequency of this point should then be noted.

The Xmax limiter is usually then set up using just two parameters:

- **Threshold** – the peak voltage of the point arrived at above. This is calibrated at the output of the amplifier. The Excursion Limiter can be defeated by setting Threshold to Off.
- **Frequency** – The frequency at which the above threshold voltage is appropriate.

A further parameter **Min** may also be available for more advanced applications. This allows the increasing limiting action at lower frequencies to level-off below a certain frequency. In most application, this would be left set to its default value of 5Hz.

To update excursion limiter settings

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the down arrow button ▼ several times until **Xmax** appears top left (usually six presses).
3. Rotate **ADJUST** (the right-hand dial) to change **Thresh** the threshold value (1 to 201V or **Off**).
4. Rotate **SELECT** (the left-hand dial) to choose **Freq** and rotate **ADJUST** (the right-hand dial) to change the frequency (10 to 1,000 Hz).
5. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Bridge mode

If you select Bridge Mode, two amplifier channels drive one loudspeaker with greater power. In this mode, only one set of output controls is active per pair of amplifier channels since both of the amplifiers in the pair are driven with the same signals, as determined by the left-hand (lower numbered) channel of the pair. Bridge settings should be set up before recalling Drive Module presets. Module Preset Recall will see a bridged pair of outputs as a single channel. When Bridged, the partner (even numbered) channel doesn't show any signal on the limiter meter, though the amplifier protection indicator does still operate. For this channel, the mute button doesn't

operate and you can't update any of the output channel parameters.

Note that the gain of a bridged pair of channels is 6 dB higher than a single channel. As the gain for a single channel (with all DSP level controls set to 0 dB) is 32 dB, the gain for a pair of bridged channels is 38 dB. This may impact limiter settings and the balance of levels in a system where some amplifier channels are bridged and others are in standard mode.

To bridge or unbridge two channels

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).

To bridge channels, you must select an odd-numbered channel. For example, to bridge channels 3 and 4, select channel 3.

2. Press the down arrow button ▼ several times so that **AMP** appears top left (usually seven presses).



This menu item doesn't exist for even-numbered channels.

3. Rotate **ADJUST** (the right-hand dial) to set **Bridge** to **On** or **Off**.

If you bridge channels, an orange bridge indicator lights up between the two channels on the right-hand side of the [front panel \(page 8\)](#).

4. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output FIR

For output FIR, there are ten stages of equalisation available for each channel:

- Two shelving filters: the [high shelf filter \(page 25\)](#) and the [low shelf filter \(page 26\)](#).
- [Eight parametric filters \(page 25\)](#).

You can also apply a [high pass filter \(page 26\)](#) and a [low pass filter \(page 26\)](#).

Note that the amplifier also has 1,000 FIR filter taps at 48 kHz on each output channel. This is reserved for optimised systems and Martin Audio's **Display** software. You can't upload your own file.

Output high shelf filter

Use the **EQ<** menu item to edit the high shelf filter for the output channel. You can update the frequency, slope and gain. For details of FIR, see [output FIR \(page 25\)](#). The high shelf filter is shown as **HS** in [Vu-Net \(page 33\)](#).

To edit the high shelf filter

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the up arrow button ▲ several times until **EQ<** appears top left (usually thirteen presses).
3. Rotate **ADJUST** (the right-hand dial) to change the frequency.
4. Rotate **SELECT** (the left-hand dial) to choose **Slope** and rotate **ADJUST** (the right-hand dial) to update the slope.
5. Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to update the gain.
6. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output parametric EQ filters

For each of the eight output parametric filters you can specify the frequency, bandwidth and gain. For details of FIR, see [output FIR \(page 25\)](#).

To update the eight parametric filters

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the up arrow button ▲ several times until **EQ1** appears top left (usually five presses).
3. Rotate **ADJUST** (the right-hand dial) to choose the frequency.
4. Rotate **SELECT** (the left-hand dial) to choose **Width** and rotate **ADJUST** (the right-hand dial) to update the bandwidth in [Q or octaves \(page 30\)](#).
5. Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to update the gain (0.2 dB increments).
6. Press the up arrow button ▲ again so that **EQ2** appears top left. Update the second parametric filter and repeat for all eight parametric filters.

- To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output low shelf filter

Use the **EQ>** menu item to edit the low shelf filter for the output channel. You can update the frequency, slope and gain. For details of FIR, see [output FIR \(page 25\)](#). The low shelf filter is shown as **LS** in [Vu-Net \(page 33\)](#).

To edit the low shelf filter

- Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
- Press the up arrow button ▲ several times until **EQ>** appears top left (usually four presses).
- Rotate **ADJUST** (the right-hand dial) to change the frequency.
- Rotate **SELECT** (the left-hand dial) to choose **Slope** and rotate **ADJUST** (the right-hand dial) to update the slope.
- Rotate **SELECT** (the left-hand dial) to choose **Gain** and rotate **ADJUST** (the right-hand dial) to update the gain.
- To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output low pass filter

Use the **LPF** menu item to edit the low pass filter for the output channel. You can update the frequency and the filter type. The filter types available are the same as for the [high pass filter \(page 26\)](#).

To edit the low pass filter

- Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
- Press the up arrow button ▲ several times until **LPF** appears top left (usually three presses).
- Rotate **ADJUST** (the right-hand dial) to change the frequency. To remove the low pass filter, turn the dial to the right until **Out>>** appears.
- Rotate **SELECT** (the left-hand dial) to choose **Shape** and rotate **ADJUST** (the right-hand dial) to select the low pass filter type (see above).
- To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output high pass filter

Use the **HPF** menu item to edit the high pass filter for the output channel. You can update the frequency and the filter type:

1st	Not supported in Vu-Net (page 33)
Bes12	Bessel 12 dB
But12	Butterworth 12 dB
LR12	Linwitz-Riley 12 dB
Bes18	Bessel 18 dB
But18	Butterworth 18 dB
Bes24	Bessel 18 dB
But24	Butterworth 24 dB
LR24	Linwitz-Riley 24 dB
Har4th	Hardman 4th order – Not supported in Vu-Net (page 33)
But48	Butterworth 48 dB
LR48	Linwitz-Riley 48 dB
Har8th	Hardman 8th order – Not supported in Vu-Net (page 33)
LIR	Linear impulse response – Not supported in Vu-Net (page 33)

Hardman filter

The Hardman type filter is always described by its order. This is because the filter becomes progressively steeper rather than following a linear slope and so a dB/octave description is not accurate.

Linear impulse response filter

Unique to Martin Audio, linear impulse response (LIR) crossover filtering gives a linear phase crossover that has a constant delay regardless of frequency. This is unlike other types of crossover that delay different frequencies to a different extent, thus smearing the arrival time. The LIR crossover can therefore be described as having a flat group delay response and is thus entirely free of group delay distortion. Common FIR filtering can provide this too, but FIR filtering does have complications and disadvantages when compared with LIR filtering.

The shape of the LIR crossover filter is similar to a 4th order Linkwitz-Riley filter, and maintains zero phase difference between the adjacent bands across the crossover region to keep the polar response rock steady.

Note that very narrow bandwidths are not possible with this crossover type. If the low pass frequency is too close to the high pass frequency, then the filter will ‘mute’.

Linear phase filtering necessarily introduces delay; the laws of physics demand it. To keep this delay to a minimum, we recommend that more conventional

crossover shapes (such as Linkwitz-Riley) are used for the very lowest frequency highpass edge, particularly if this is less than perhaps 100Hz, which is well below the frequency thought to cause audible 'Group Delay Distortion'.

This constant delay will depend on the lowest high-pass frequency used in the crossover filters in a given [Drive Module \(page 5\)](#). For further details, see [Latency delay \(page 41\)](#).

To edit the high pass filter

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the up arrow button ▲ several times until **HPF** appears top left (usually two presses).
3. Rotate **ADJUST** (the right-hand dial) to change the frequency. To remove the high pass filter, turn the dial to the left until <<**Out** appears.
4. Rotate **SELECT** (the left-hand dial) to choose **Shape** and rotate **ADJUST** (the right-hand dial) to select the high pass filter type (see above).
5. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Output delay

Use the **DEL** menu item to add a delay to an output channel. See also [Latency delay \(page 41\)](#).

To update output delay

1. Press **OUTPUT** to select the channel. For details, see [Selecting the output channel \(page 22\)](#).
2. Press the up arrow button ▲ several times until **DEL** appears top left (usually one press).
3. Rotate **ADJUST** (the right-hand dial) to change the delay in ms. To remove the delay, set the delay to 0 ms.
4. To step through the channels press **OUTPUT** several times, or press **UTILITY** to exit the menu.

Updating utilities

From the utility menu you can choose:

- **STER** – see [Stereo linking \(page 28\)](#)
- **DANT** – see [Dante name \(page 28\)](#)
- **VER** – see [Firmware version \(page 28\)](#)
- **ECO** – see [Power saving \(page 28\)](#)
- **PWR** – see [External breaker protection \(page 29\)](#)
- **ALM** – see [Alarm \(page 29\)](#)
- **Aux** – see [AUX style \(page 29\)](#)
- **ParaEQ** – see [Parametric EQ bandwidth \(page 30\)](#)
- **Screen** – see [Screen contrast \(page 30\)](#)
- **RECL** – see [Recall snapshot \(page 30\)](#)
- **STOR** – see [Store snapshot \(page 30\)](#)
- **IP Static** – see [Static IP address \(page 30\)](#)
- **IP Mode** – see [IP mode \(page 31\)](#)
- **IP Curr** – see [Current IP address \(page 31\)](#)

Stereo linking

You can stereo link inputs A and B and/or inputs C and D. If you link inputs and then edit one of them, the amplifier will automatically edit both of the linked inputs.



You can only link inputs that are of equal size.

The amplifier stores the stereo linking state in [snapshots \(page 14\)](#) but not in [presets \(page 13\)](#).

To stereo link channels

1. Press **UTILITY**.
STER appears top left as this is the first item in the utility menu.
2. Rotate **SELECT** (the left-hand dial) to highlight the text below **Link A** or **Link C**.
3. Rotate **ADJUST** (the right-hand dial).
To link inputs A and B, select **Link A** on the top line and **To B** below.

To link inputs C and D, select **Link C** on the top line and **To D** below.

To unlink inputs, select **None**.

4. Press **UTILITY** to exit the menu.

Dante name

The **DANT** menu item displays the unique amplifier name used with [Dante \(page 6\)](#). You can only change the Dante name using the **Dante Controller** software. You can't change the Dante name at the front panel or in [Vu-Net \(page 33\)](#).

To view the Dante name

1. Press **UTILITY**.
2. Press the down arrow button ▼ once so that **DANT** appears top left. The Dante name for this amplifier is bottom right.
3. Press **UTILITY** to exit the menu.

Firmware version

The **VER** menu item displays the firmware and [Dante \(page 6\)](#) version numbers.

- To update the firmware, use [Vu-Net \(page 33\)](#).
- To update Dante, use the **Dante Controller** software.

To view the firmware version

1. Press **UTILITY**.
2. Press the down arrow button ▼ twice so that **VER** appears top left. The screen shows the amplifier model, the firmware version (under the heading **Ver**) and the Dante version.
3. Press **UTILITY** to exit the menu.

Power saving

Use the **ECO** menu item to select standby or sleep mode.

- **Standby mode** – You specify the number of minutes with no audio detected before the amplifier goes into standby mode. Alternatively, you can put the amplifier into standby manually using the **S'dby** parameter.

In standby, as soon as the amplifier detects an audio signal it automatically comes out of standby. To bring the amplifier out of standby manually, press the up arrow ▲. Note that this button flashes every five or six seconds to remind you that this is the button to press. Alternatively, you can bring the amplifier out of standby using [Vu-Net \(page 33\)](#).



In standby mode, the fast fan will switch on to discharge the power. It may switch on for a few minutes or sometimes longer, depending on the amount of power that needs to be discharged.

- **Sleep mode** – You specify the number of minutes with no audio detected before the amplifier switches off.

To start using the amplifier again, switch the power switch on the [front panel \(page 8\)](#) off, pause for a second or two and then switch it on again. If you switch the amplifier off and on too quickly the amplifier will remain switched off. Alternatively, you can bring the amplifier out of sleep using [Vu-Net \(page 33\)](#). Note that you will have to wait while the amplifier goes through all the usual start up checks.

To change power saving modes

1. Press **UTILITY**.
2. Press the down arrow button ▼ three times so that **ECO** appears top left.
3. Rotate **SELECT** (the left-hand dial) to highlight the text below **Save**.
4. To switch off standby mode (see above), rotate **ADJUST** (the right-hand dial) to the right until **Man** (for manual) appears.
To switch on standby mode, rotate **ADJUST** (the right-hand dial) to select a number of minutes (3 to 89).
5. Rotate **SELECT** (the left-hand dial) to highlight the text below **Sleep**.
6. To switch off sleep mode (see above), rotate **ADJUST** (the right-hand dial) to the right until **Man** (for manual) appears.
To switch on sleep mode, rotate **ADJUST** (the right-hand dial) to select a number of minutes (3 to 89).
7. To switch on standby mode straight away, rotate **SELECT** (the left-hand dial) to highlight the text below

S'dby and rotate **ADJUST** (the right-hand dial) to the left. The text on the bottom line changes from **Now<<** to **Bye** and the amplifier goes straight into standby mode (see above).

8. Press **UTILITY** to exit the menu.

External breaker protection

If you use the amplifier on a mains supply with a restricted capacity, use external breaker protection (EBP) to limit the output power. This will avoid the nuisance of tripping mains circuit breakers or fuses. Set the EBP value to be the same value as the circuit breaker or fuse.

To set External Breaker Protection

1. Press **UTILITY**.
2. Press the down arrow button ▼ four times so that **PWR** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to select the amp value (between 9 and 50 A). The screen displays the draw (which you can't edit).
4. Press **UTILITY** to exit the menu.

Alarm

The **ALM** screen allows you to configure the relay action when there is an alarm from the amplifier. This alarm can be received by a monitoring system such as Q-Sys or Creston. The relay port uses pins 4, 5 and 6 of the Phoenix connector (also known as a Euroblock) on the [back panel \(page 10\)](#). Pins 1, 2 and 3 of this connector are for the [AUX port \(page 40\)](#).

To edit the relay action for alarms

1. Press **UTILITY**.
2. Press the down arrow button ▼ five times so that **ALM** appears top left.
3. Rotate **SELECT** (the right-hand dial) to choose **Fault only** or **Fault or Check**.
4. Press **UTILITY** to exit the menu.

AUX style

This allows you to view or update the way that the amplifier responds to signals sent to the AUX port. For further details, see [AUX port \(page 40\)](#).

To update the AUX style

1. Press **UTILITY**.
2. Press the down arrow button ▼ six times so that **Aux** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to select the AUX style. For details, see [AUX port \(page 40\)](#).
4. Press **UTILITY** to exit the menu.

Parametric EQ bandwidth

This allows you to choose whether to edit the bandwidth of the parametric equalisers in octaves or Q. This applies to the [Input parametric EQ filters \(page 20\)](#) and the [Output parametric EQ filters \(page 25\)](#).

To set the parameter EQ bandwidth

1. Press **UTILITY**.
2. Press the down arrow button ▼ seven times so that **ParaEQ** appears top left.
3. For octaves, rotate **ADJUST** (the right-hand dial) to select **BW=oct**.
For Q, rotate **ADJUST** (the right-hand dial) to select **BW=Q**.
4. Press **UTILITY** to exit the menu.

Screen contrast

This allows you to change the screen contrast.

To update the screen contrast

1. Press **UTILITY**.
2. Press the down arrow button ▼ eight times so that **Screen** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to select a screen contrast between 0 and 100%. You can't edit **Check1** and **Check2**.
4. Press **UTILITY** to exit the menu.

Recall snapshot

This allows you to recall a [saved set of amplifier settings \(page 30\)](#). For further details, see [Snapshots \(page 14\)](#).

To recall a snapshot

1. Press **UTILITY**.

2. Press the down arrow button ▼ nine times so that **RECL** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to select the snapshot number (between 1 and 20).
4. Press **ENTER**.
5. Press **ENTER** again to confirm or press the down arrow button ▼ to cancel.
6. Press **UTILITY** to exit the menu.

Store snapshot

This allows you to save a set of amplifier settings. To recall the settings, use [Recall snapshot \(page 30\)](#).



You need to store presets before you can store snapshots.

For further details, see [Snapshots \(page 14\)](#).

To store a snapshot

1. Press **UTILITY**.
2. Press the down arrow button ▼ ten times so that **STOR** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to select the snapshot number to use (between 1 and 20).
4. Press **ENTER**.
5. You can now enter a name for the snapshot (up to 12 characters).

Rotate **ADJUST** (the right-hand dial) to select each character. Scrolling right gives A to Z followed by six symbols followed by a to z in lower case. Scrolling left gives the numbers 0 to 9 and some symbols.

Rotate **SELECT** (the left-hand dial) to move forward or backward in the name.

6. Press **ENTER**.
7. Press **ENTER** again to confirm or press the down arrow button ▼ to cancel.
8. Press **UTILITY** to exit the menu.

Static IP address

The **IP Static** screen allows you to specify the IP address of the amplifier.



The **IP Static** screen only appears in the utility menu if you've selected static IP using **IP mode** (page 31). If you're using automatic IP, this menu item won't appear in the utility menu.

To edit the IP address

1. Press **UTILITY**.
2. Press the up arrow button ▲ twice and check the **IP mode** (page 31). If it's auto, change it to static.
3. Press the up arrow button ▲ once so that **IP Static** appears top left.
4. Rotate **ADJUST** (the left-hand dial) to select the part of the address to edit.
5. Rotate **SELECT** (the right-hand dial) to update the address.
6. Press **UTILITY** to exit the menu.

IP mode

The **IP Mode** screen allows you to choose static or automatic IP addressing.

If you're using static IP, the home screen shows the IP address on the top line as flashing text.



If you change IP mode from static to automatic, you will need to wait for the amplifier to pick up an address. This could take a few minutes.

To edit the IP mode

1. Press **UTILITY**.
2. Press the up arrow button ▲ twice so that **IP Mode** appears top left.
3. Rotate **SELECT** (the right-hand dial) to choose **Static** or **Auto**.
4. To adjust the subnet mask, rotate **ADJUST** (the left-hand dial) to the right to highlight the subnet mask. Then rotate **SELECT** (the right-hand dial) to choose a mask value between 1 and 29.
5. If you've chosen static IP, press the up arrow ▲ once so that **IP Static** appears top left. Use that screen to specify the **Static IP address** (page 30).

6. Press **UTILITY** to exit the menu.

Current IP address

The **IP Curr** screen displays the current Ethernet IP address. You can't edit the IP address using this screen.

If you're using static IP, press up arrow ▲ twice from the **IP Curr** screen to go to the **Static IP address screen** (page 30).

To view the IP address

1. Press **UTILITY**.
2. Press the up arrow button ▲ once so that **IP Curr** appears top left. The IP address is divided into three sections. For example, IP address 169.254.241.120 appears like this:

Curr		
169.254	241	120

3. Press **UTILITY** to exit the menu.

Ethernet

Ethernet configurations

IP addressing by the iK42 can be completely automatic, with no setup required. To select auto IP, see [IP mode \(page 31\)](#).

When first installing and launching [Vu-Net \(page 33\)](#), the computer Firewall may ask you to allow Vu-Net to access the network.



Allow Vu-Net to have access to both private and public networks.

DHCP

There are two primary IP address ranges – one used when there is a DHCP server, and another ('Link Local') where there is no DHCP server (so the Device and the Computer will instead use 'Auto IP' to allocate themselves an IP address). Both the device and the computer must be in the same IP address range. In a local network environment such as an office where there is a DHCP server, both the computer and the Device will be in the DHCP IP address range, and so will connect immediately.

Auto IP

When switched on, the device will initially search for a DHCP server (during this time the Online Indicator flashes). As it can take up to one minute to establish that there is no DHCP server available, this is the time it may take before Auto IP is entered. Please be aware that it can also take some time from a computer being switched on in an isolated network (without a DHCP server), or unplugged from a network with DHCP to time out of DHCP searching, so it will not connect immediately to amplifiers that are already using Auto IP. The time it takes before it decides to revert to Auto IP depends on the operating system but it can take several minutes to acquire an Auto IP address.

Static IP

To select static IP, see [IP mode \(page 31\)](#). To select the IP address, see [Static IP address \(page 30\)](#).

If the amplifier or computer has a static IP address and a different IP address range (i.e. a different subnet), [Vu-Net \(page 33\)](#) may not be able to 'see' the device.

IP Troubleshooting

If [Vu-Net \(page 33\)](#) can't connect to the amplifier:

- A Router acting as a DHCP server is highly recommended as this provides the most trouble-free way of administering IP addresses. Always switch on any DHCP server before connecting either the computer or amplifier to the network.
- If not using a DHCP server, check that the Current IP address in the device is compatible with the IP address of the computer. Generally, the leftmost two sets of 3 digits should be the same.
- If there is no Router in the system acting as a DHCP server, wait 10 minutes (for the computer to acquire the correct IP address) and try again.
- Check that the Firewall in the computer will allow Vu-Net access to the network for both private and public networks.

Vu-Net software

Martin Audio's Vu-Net software allows you to control and monitor iKON amplifiers from a PC. To do this, connect the amplifiers and PC using an Ethernet network and run Vu-Net on the PC. Vu-Net also allows you to control and monitor many Martin Audio loudspeaker systems.

Vu-Net is a Windows app available as a free download from our website martin-audio.com.



We don't support macOS, but some of our users run this app successfully on virtual platforms such as Parallels Desktop, VMware Fusion and Apple Boot Camp Assistant.

- Output current: Driver current (A) – For each output, the average current being delivered to the driver.
- Load impedance: Driver impedance (ohm) – For each output, the average impedance of the driver.
- Global protection systems: Protection limiting (dB) – For each output, the amount that the amplifier is 'dimming' itself in order to protect itself from potential damage.

To download Vu-Net

1. Go to the [Martin Audio website](http://martin-audio.com) and select **Support > Software**.
2. Scroll to the **Vu-Net** section and click **DOWNLOAD VU-NET**.

Using Vu-Net

For full details, see the [Vu-Net User Guide](#).

To download the Vu-Net User Guide

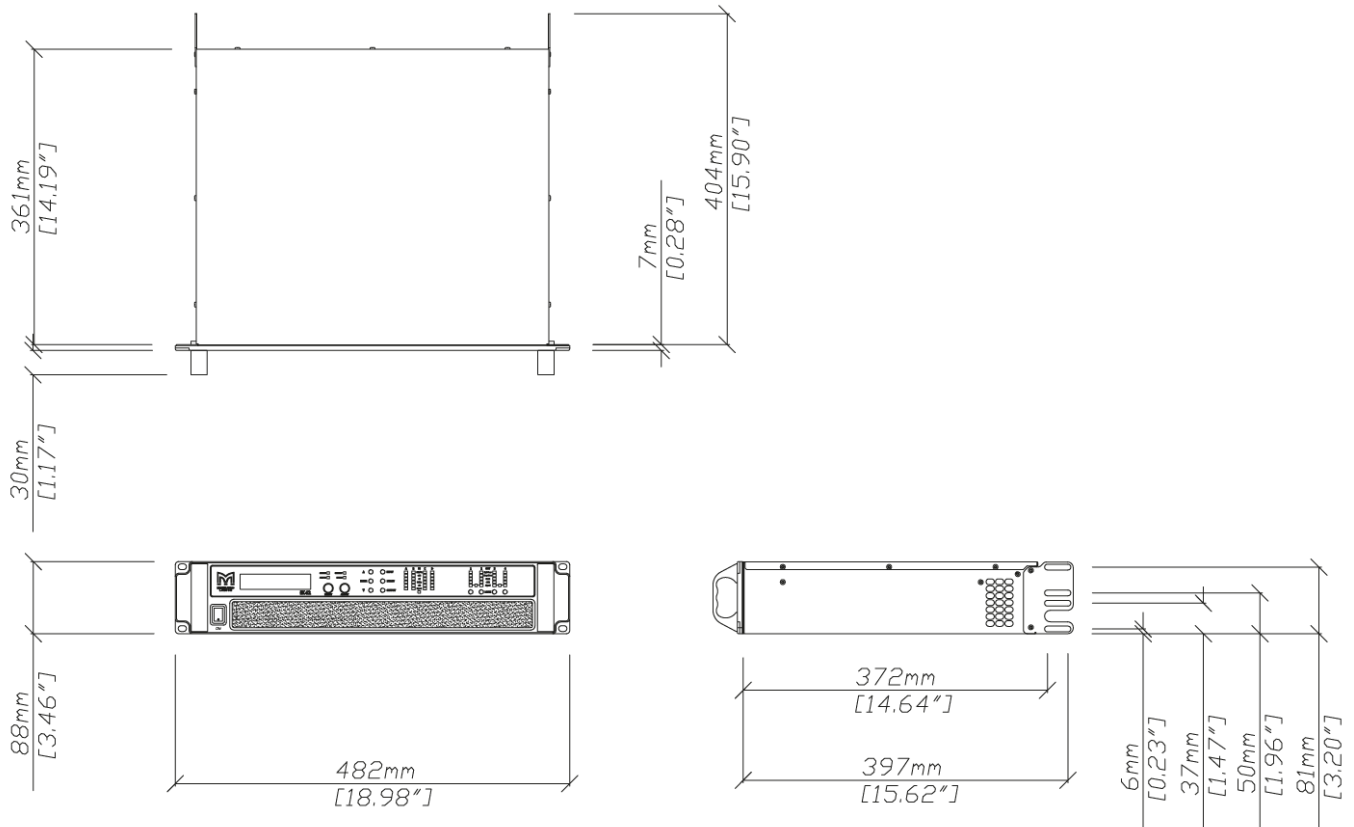
1. Go to the [Martin Audio website](http://martin-audio.com) and select **Support > Software**.
2. Scroll to the **Vu-Net** section and click **USER GUIDE**.

Performance logging

[Vu-Net \(page 33\)](#) allows you to see graphs of performance against time, recording events over a period of up to three days. Breaks in the log due to power-downs are shown in blue. The graphs show:

- Supply current (A) – How much current is being supplied to the device via the mains inlet.
- Power systems: DC link voltage (V) – The voltage which is derived from the mains supply voltage.
- Thermal management: Thermal capacity (%) – The percentage of the available thermal capacity that has been used.

Installation



iKON amplifiers are designed to be mounted in standard 19" rack enclosures.

If you have a fixed installation, with the bottom unit supported and no gaps between units, you can just use the front panel 19" rack holes.

If you have a mobile rack, you must support the rear of the units with [rack ears](#) (page 38). Any damage caused by insufficient rack support is not covered by the warranty.

To prevent damage to the front panel, we recommend that you fit plastic cups or washers underneath the rack-mounting bolt heads.

You can mount multiple iKON amplifiers without any ventilation gaps between them. However, you must make sure that there is an unobstructed flow of clean air from the front to the rear of the units. You must not cover the air intakes on the front of the units or the exhaust vents on the rear of the units. You must make sure that you aren't

just continually circulating hot air through the amplifiers from the back of the rack to the front.

The amplifier should never be exposed to rain or moisture during operation or storage. If the unit does come into contact with moisture, remove the AC power cable immediately and leave the unit in a warm location to dry out.

Note that when equipment is taken from a cold location to a hot humid one, condensation can form inside the device. Before you connect the AC power cable, always allow time for equipment to reach the same temperature as the surrounding environment



Keep your amplifier away from dirt, liquids and vapour from theatrical smoke and fog machines.

Damage from these sources isn't covered by the warranty.

mains and will automatically resume normal operation if the mains is stable and has returned to the initial range selected when the unit was first powered up.

AC power connection

The amplifier uses a 32A Neutrik powerCON locking AC power connector. You must use an AC power cable with the correct powerCON connector.

The amplifiers are designed to operate on 50/60 Hz AC power. The power supply sections automatically configure themselves for either 115V or 230V nominal voltage at start up. The amplifiers will operate over an extended range of supply voltage. The thresholds of these ranges are as follows:

- 115V range: 75 – 138Vrms
- 230V range: 138 – 275Vrms

Note that whilst the amplifier will operate correctly at voltages indicated, the specified output power will only be achieved when operating with the stated nominal voltages.

Once an amplifier has configured itself for a particular range it will not change range, even if the mains voltage varies wildly, unless power is completely interrupted for several seconds or is cycled by the user.

During start up the amplifier checks the mains environment and waits for the mains to settle within either the 115V or 230V range. If the mains takes a long time to settle, the amplifier will display one of the following messages on the front panel:

Message	Problem
Mains too low	Voltage below the 115V range
Mains too high	Voltage above the 230V range
Mains unstable	Voltage within the 115V or 230V ranges but is unstable

When an amplifier is running normally, it continuously monitors the mains to ensure that it remains within the initially selected range. If the mains voltage is outside that range the amplifier will enter protection mode and briefly display the message **Mains out of range** followed by one of the messages above. In protection mode, audio will be muted but communication with the amplifier will still be possible. The amplifier will continue to monitor the

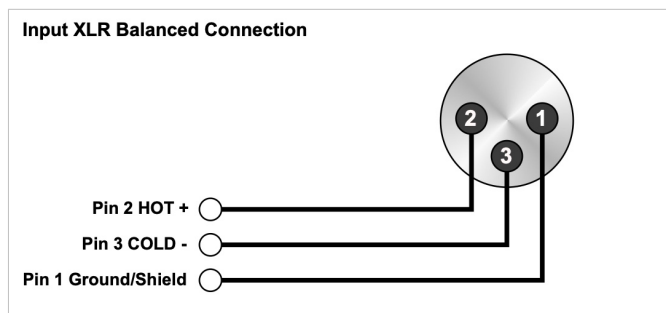
Audio connections

Input connections

For each input channel there is a female XLR connector for analogue inputs. There is also one female XLR for one stream (two channels) of AES3 digital audio. Note that only two channels of AES3 digital audio are available. The Dante option permits more channels of Digital Audio inputs.

Use a cable connected as follows:

- HOT, + or 'in phase' connected to pin 2 of the XLR connector.
- COLD, – or 'out of phase' connected to pin 3 of the XLR connector.
- The chassis and cable shield connected to pin 1 of the XLR connector. This connection is required for EMC performance and regulations.



Unbalanced input connections

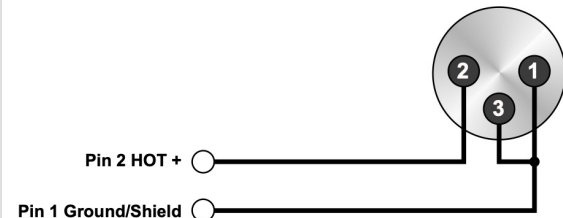


We don't recommend unbalanced connections.

For an unbalanced audio source, use a cable connected as follows:

- Signal conductor connected to pin 2 of the XLR connector.
- COLD or cable screen connected to pin 1 of the XLR connector with a short connection between pin 1 and pin 3.

Input XLR Unbalanced Connection



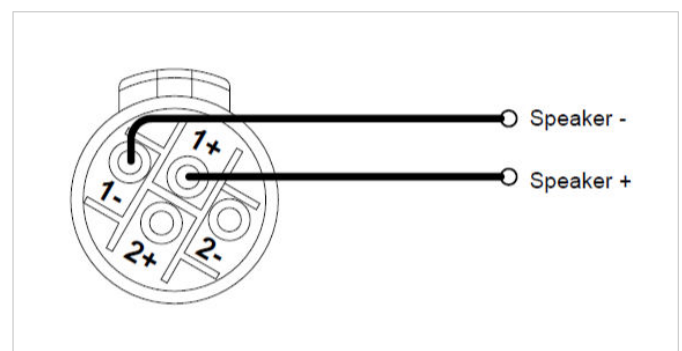
Output connections

The iK41, iK42 and iK81 amplifiers have four speakON connectors for audio output.

- For the iK41 and iK42 these are labelled 1, 2, 3 and 4 on the [back panel \(page 10\)](#).
- For the iK81 these are labelled 1&2, 3&4, 5&6 and 7&8 on the [back panel \(page 10\)](#). Connection 1&2 is for channels 1 and 2, connection 3&4 is for channels 3 and 4 and so on.

Note that you can connect more than one speaker to each channel provided the total impedance per channel is 2 ohms or more. In [bridged mode \(page 24\)](#) the minimum total impedance is 4 ohms.

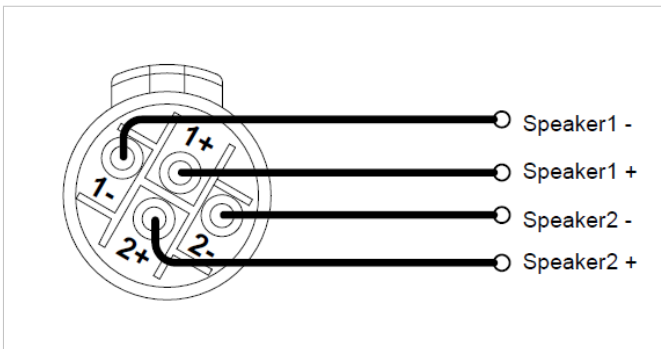
The iK41 and iK42 have one connector for each amplifier channel. You connect the loudspeaker to terminals 1+ and 1- as shown here:



Alternatively, for the iK41 and iK42, you can use bi-amp wiring with connectors 1 and 3. This is because:

- Channel 2 output is duplicated on terminals 2+ and 2- of connection 1.
- Channel 4 output is duplicated on terminals 2+ and 2- of connection 3.

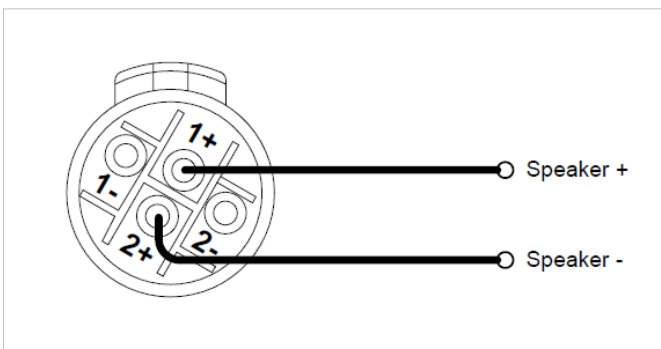
With bi-amp wiring you connect two loudspeakers with one 4-core cable as follows:



The iK81 has four bi-amp outputs with each speakON connector carrying two amplifier outputs. You wire each connection as shown above.

Bridge mode

If you [bridge \(page 24\)](#) a pair of amplifier channels, you connect to terminals 1+ and 2+ as follows:



For the iK41 and iK42, this applies to connectors 1 and 3.

For the iK81, this applies to all four connectors.

Load matching

Each output of the device can be optimised to drive either a low impedance load (2, 3, 4, 6 or 8 ohms) or a constant voltage (25V, 70V or 100V line).

With the low impedance settings, it's not critical that the setting matches the impedance of the connected load, but if it does, this will maximise the power that is available for the load.

The constant voltage settings specify the maximum RMS voltage that the amplifier will produce.

To choose the load matching, use the [Drive load on the Output menu \(page 22\)](#).

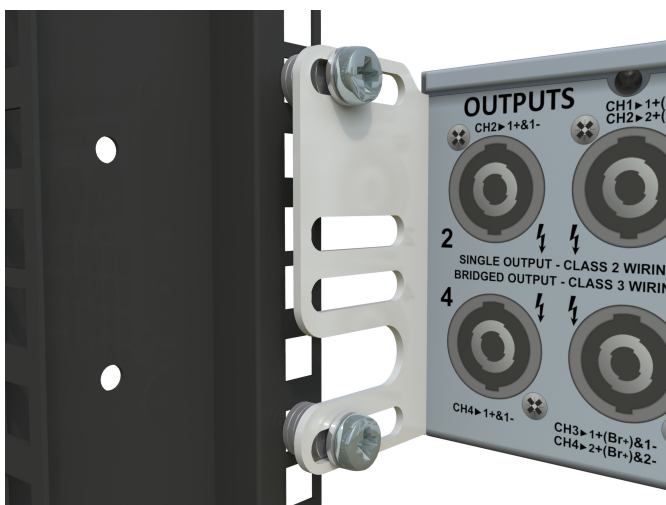
Rack ears

Rack ears (part number RACKKITC) provide rear support for iKON amplifiers installed in 19" racks. We recommend that you use rack ears for all rack mounted amplifiers, particularly for touring racks, where the vibration from movement can place unwanted stresses on the amplifier chassis if it is only supported at the front.

You can install the rack ears in two ways: removable or locked. Removable installation allows easy removal of the amplifier from the rack with no need for access to the rear:



With locked installation, you attach the amplifier to the rack at the rear, preventing removal from the front:



For removable installation, you attach amplifier brackets, attach rack brackets and then slot the two brackets together.

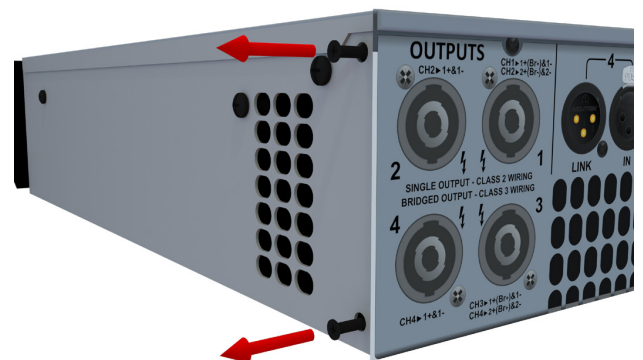
For locked installation, you attach amplifier brackets and then attach these brackets to the rear rack strip. With this method, you don't use the rack brackets.

Requirements

- The amplifier rack must have 19" rack strip at the front and rear. For removable installation, the depth between front and rear is relatively unimportant as the rack bracket allows for up to 180 mm of variation. For locked installation, the slots in the amplifier bracket allow up to 25 mm of variation.
- Four standard 19" rack cage nuts and fixing screws (most commonly M6 in Europe) to attach the brackets to the rear rack strip.
- A Pozidriv PZ1 screwdriver.
- A suitable screwdriver for the rack screws.
- A small pair of pliers or a 7 mm spanner.

To attach the amplifier brackets

1. Remove the two screws shown below. You can discard these screws.



2. Attach the support bracket to the side panel using the M3 hex head screws and M2.5 Allen key (hex wrench) supplied with the kit.

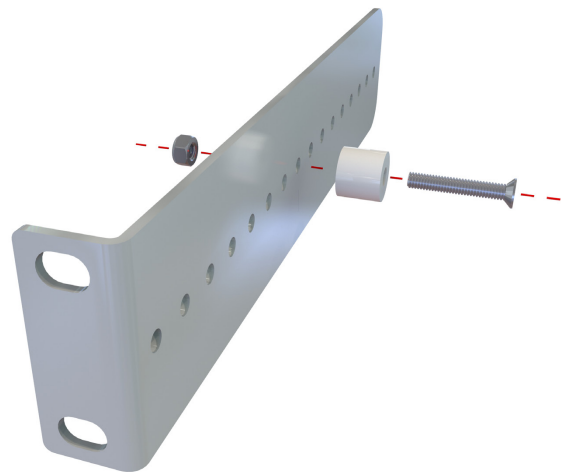


3. Attach the second amplifier bracket in the same way.

You now either attach rack brackets (for removable installation) or attach the amplifier brackets to the rack strip (for locked installation).

To attach the rack brackets

1. Fit the amplifier in the rack using the front panel rack screws.
2. Temporarily hold one of the rack brackets on the rear rack strip so that the holes in the bracket line up with the slot in the amplifier bracket.
3. Make a note of the hole that is closest to the centre of the slot in the amplifier bracket.
4. Remove the amplifier from the rack.
5. Fit the nylon bush to the hole you identified in step 3. Use the M4x25 countersunk screw and M4 Nylok nut supplied with the kit.



6. Fit the nylon bush to the same hole in the second bracket.
7. Fit two cage nuts onto the rear rack strip at the correct height to ensure that the rack brackets will engage with the amplifier brackets.
8. Attach each bracket with two rack screws.
9. Fit the amplifier in the rack and check that the amplifier bracket engages on the nylon bushes on both sides, supporting the weight of the amplifier at the rear.
10. Screw the amplifier to the rack using the four rack mount holes in the front panel.

To attach the amplifier brackets to the rack strip

1. Attach an amplifier bracket to the return on the rear rack strip, as shown in the picture near the start of this section. Use two standard rack cage nuts and screws. If there is a gap between the bracket and rack strip, use washers or spacers to close the gap; don't close the gap by tightening the rack screws so that the brackets bend.
2. Attach the second amplifier bracket in the same way.

AUX port

AUX X	AUX Y	2+Mute (event or state)	3 snaps (event or state)	4 snaps (state)	3+Mute (state)	2+Sleep (state)	Mute Slp (state)
Open	Open	No change	No change	Recall snap 1	Recall snap 1	No change	No change
Gnd	Open	Recall snap 1	Recall snap 1	Recall snap 2	Recall snap 2	Recall snap 1	Mute
Open	Gnd	Recall snap 2	Recall snap 2	Recall snap 3	Recall snap 3	Recall snap 2	Sleep
Gnd	Gnd	Mute	Recall snap 3	Recall snap 4	Mute	Sleep	Sleep

The AUX port allows you to send signals to the amplifier so that it recalls [snapshots \(page 14\)](#), mutes or sleeps, as shown in the table above. The AUX port consists of the first three pins of the six-pin Phoenix connector (also known as a Euroblock) on the [back panel \(page 10\)](#). The three pins are X, Y and ground, as labelled on the back panel. To trigger inputs X and Y you can:

- Connect the input to ground using a simple contact closure device (a relay or switch). There is no requirement for an external voltage.
- Send a logic signal to the input. The logic low must be less than +0.5V and the logic high must be less than +24V.

AUX styles

As summarised in the table above, you can configure the AUX port with the following styles:

- **None** – Ignore all signals.
- **2+Mute** (event or state) – Recall snapshot 1 or 2 by applying a momentary or static connection to AUX port input X or Y. Mute the device by applying a momentary or static connection to both AUX port inputs.
- **3 Snaps** (event or state) – Recall snapshot 1, 2 or 3 by applying a momentary or static connection to AUX port inputs X or Y.
- **4 Snaps** (state) – Recall snapshot 1, 2, 3 or 4 by applying a static connection to AUX port inputs X or Y.
- **3+Mute** (state) – Recall snapshot 1, 2 or 3 by applying a static connection to the AUX port inputs X or Y. Mute the device by applying a static connection to both AUX port inputs.
- **2+Sleep** (state) – Recall snapshot 1 or 2 by applying a static connection to AUX port inputs X or Y. Sleep

the device by applying a static connection to both AUX port inputs.

- **Mute Slp** (state) – Mute the device by applying a static connection to AUX input X. Sleep the device by applying a static connection to AUX input Y.

To select the style, use the [Aux style page of the Utility menu \(page 29\)](#).

Event or state

With the first two AUX styles (**2+Mute** and **3 snaps**) you can use events or states:

- With events, the signal is a momentary push-button or momentary relay contact closure.
- With states, the signal is something like a rotary switch that holds the closure.

With the last four AUX styles (**4 snaps**, **3+Mute**, **2+Sleep**, **Mute Slp**), you can only use states. With these AUX styles, the selected snapshot is only held while the connection pattern persists.



If the connection pattern is static, the snapshot recall overrides [Vu-Net \(page 33\)](#) and the front panel menu (the snapshot menu items won't be available). The home screen shows the snapshot number as a reminder that the snapshot has been loaded due to an AUX port signal.

Latency delay

Digital Signal Processing and conversion between different formats of signal (analogue, digital, network and so on) introduces some delay (latency) to the signal path. Of course, we strive to minimise these latencies. The following tables show the principal latencies introduced by the device.

Input latency

Analogue input	0.385 ms
Or digital input at 96 KHz sample rate	0.5 ms
Or digital input at 48 KHz sample rate	0.66 ms

Output latency

Analogue output	0.402 ms
Or AES3 output	0.1 ms
Or Dante output (excluding Dante network latency)	1.29 ms @ 44.1 kHz
	0.68 ms @ 48 kHz
	0.45 ms @ 96 kHz

Processing latency

Unless you're using LIR crossover filters or FIR high shelf EQs, the processing latency is always 1.57 ms. So to calculate total latency, you normally just add 1.57 ms to the input and output delays.

If you're using LIR or FIR, the processing latency calculations are shown in the table below. Note that the processing latency can be more than 1.57 ms. When high pass frequency is set below 40 Hz the filter will automatically revert to Linkwitz-Riley; this is set so that the latency will not exceed 30 ms.

Input HiShelf FIR	0.4 ms (0 ms if set to 'Off')
LIR crossover filter	1.19 ms / crossover high pass frequency in kHz, limited to 30 ms maximum
VxLim Limiter (VX mode off)	0.12 ms / crossover high pass frequency in kHz, limited to 1.57 ms maximum
Or VxLim Limiter (VX mode on)	0.358 ms / Vx split frequency in kHz, limited to 1.57 ms maximum

If you're using input FIR filters, LIR filters or VX Limiters, it's best to view the Drive Module latency on the front panel rather than calculating the value. For details, see [Drive module latency \(page 19\)](#).

Example latency calculation

Analogue input	0.385 ms
Analogue output	0.402 ms
Input high shelf FIR (off)	0 ms
LIR crossover (500 Hz)	2.380 ms
VxLim Lim (VX mode on, Vx split frequency 1 kHz)	0.358 ms
Total	3.525 ms

Note that the latencies within a Drive Module are equalised among outputs of that Drive Module. That is, the device automatically adds padding delay to some outputs so that the total latency is the same for each output of a Drive Module.

This latency equalisation does not extend outside a Drive Module, so Drive Modules are not guaranteed to have the same latency as one another.

Protection systems

Comprehensive protection features preserve the longevity of the loudspeaker and amplifier by continuously monitoring several critical parameters, and reducing the gain, or muting the amplifier either temporarily or permanently depending on the nature and seriousness of the fault or misuse. The amplifier will recover and restart if at all possible, but may remain in shut down if a serious fault persists.

Limiters deal with routine over-driving of the amplifier, making sure that the driver(s) are not pushed too hard. The limiter indicators will warn you when the driver is being driven into limit.

Minor faults are dealt with by 'dimming' the amplifier, reducing the level to a sufficient degree and for a sufficient time that the amplifier is able to recover gracefully without any user interaction. When the fault condition has passed, the amplifier will recover automatically.

When the protection systems are reducing the level, this is indicated by illuminating the left-hand amplifier channel protection indicator. This will also illuminate when the output voltage is 'clipping'. This indicator will remain permanently illuminated if the channel has been muted by the protection systems. The mute button will also flash and there will usually be a warning message on the display. Some types of protection affect all output channels, so you may see the amplifier indicator illuminated and the mute buttons flashing on every channel.

Summary of protection indication

The protection indicators illuminate for the following reasons:

Amplifier indicator	Driver indicator	Notes
	Limiter 6dB over	
	Thermal limiting	
	Excursion limiting	
Amplifier clipping		
Amplifier clip limiting		
Amplifier current limiting		
Amplifier VHF limiting		

Amplifier indicator	Driver indicator	Notes
PSU current limiting		Will show on all output channels
PSU power limiting		Will show on all output channels
Thermal limiting		Will show on all output channels

Fault relay

The fault relay port consists of pins 4, 5 and 6 of the six-pin Phoenix connector (also known as a Euroblock) on the [back of the amplifier \(page 10\)](#). The diagram printed on the back panel shows the idle state of the relay (when the amplifier is off). When the amplifier is switched on, the relay energises and when a fault occurs, the relay de-energises. This means that a fault and a loss of power are both seen as a fault.

Tipi control

Tipi is a simple but powerful ASCII text based protocol for controlling and monitoring iKON amplifiers using third-party control panels or software applications over Ethernet (10/100/1G Ethernet using TCP/IP protocol port 51456).

- Messages are text strings (8-bit ASCII) of up to 255 characters
- Messages start with a \$ character
- Messages end with a carriage return character (<CR>, \r, ASCII 13, hexadecimal 0x0D)
- Fields are separated by one or more space characters. No space is required after the start delimiter \$ or before the end delimiter <CR>
- Don't use spaces in name fields (you could use underscores)
- Numbers are expressed in decimal units
- You can use uppercase, lowercase or mixed case
- Responses use uppercase for commands (for example: NOTIFY) and camel case for method names (for example: Out1/Gain)

The message format is S C M V E

- S is the start character \$
- C is the command (GET, SET and so on).
- M is the method name. This could be Snapshot (for example \$SET Snapshot 8<CR>) or it could be fields separated by a forward slash (for example \$SET Out1/Eq3Freq 330Hz). The fields are usually:
 - The path (the input or output), for example InA, InD, Out1, Out4
 - The parameter name, for example Eq3Freq
- V is the value, for example -3.8dB. Note that you can omit the unit of measure, for example -3.8
- E is the end delimiter <CR>
- Any characters before the start delimiter or after the end delimiter are ignored.

This means that you can have an LF or NUL character after the <CR> if you wish.

Commands

- SET This tells the amplifier to set the specified parameter to a particular value.
Example: \$SET Out2/Gain 3.5dB<CR>
- GET This requests the value of the specified parameter. The amplifier will return a NOTIFY command appended with the value of the parameter.
Example: \$GET Out8/Eq2Freq<CR>
Example response: \$NOTIFY Out8/Eq2Freq 330Hz<CR>
Note that since parameter values are quantised in the device, the value returned in any subsequent GET response may not be exactly the same as the value in a SET command. For example \$SET Out1/Gain -22.415dB<CR> might set the parameter value to -22.42dB. Similarly, if a SET command attempts to set a parameter to a value outside the permitted range, a subsequent GET command will return the permitted value.
- VERSION This requests the Tipi version number. There are no further parameters for this command.
Example: \$VERSION<CR>
Example response: \$NOTIFY VERSION 1.28<CR>
- NOP Useful for TCP/IP “keep alive”
Example: \$NOP<CR>

For a list of example commands, see [Tipi examples \(page 45\)](#).

Responses

- NOTIFY The response to a GET or VERSION command.
Example: \$NOTIFY Out2/Eq3Gain 2.6dB<CR>
- ERROR This is returned when a command is wrongly formatted or not supported. Error responses return the original command along with a brief description of the error and the error code (see the table of error codes below). For example:
\$ERROR <command> BadCommand 06<CR>

For a list of example responses, see [Tipi examples \(page 45\)](#).

Parameter values

Specify parameter values using the units shown here:

Suffix	Parameter	Example
Hz	Frequency	650.3Hz
Oct	Bandwidth	0.32Oct
dB	Decibels	-3.4dB
ms	Milliseconds	120.3ms
X	Multiplier	3.2X
:1	Ratio	4:1
R	Resistance (ohm)	4.2R
Min	Minutes	122.5Min
V	Voltage	233.6V
A	Current	72.3A
%	Percent	52.5%
Blob	Binary large object	

You can omit units or include them if you wish. If you do include the units, put the unit straight after the value with no space in between.

Tipi doesn't support multipliers such as k for 1000. For example, to specify 15kHz use 15000Hz.

Use decimal values with any desired precision. The amplifier truncates the value to the precision of the amplifier.

To specify boolean values use `yes` or `no`. For example:
`$SET Out1/Mute yes`

To specify items such as high pass filter shape `Bes24` (24dB Bessel) use index numbers rather than text.

Error codes

01, 02, 03, 04	Bad command - incomplete
05	Bad command - unexpected length
06	Bad command
07	Bad command - param value
08, 12	Unknown method
09	Unknown method type
10	Bad command - expected yes or no
11	Bad command - float format
Others	Unknown error

Faults

14	Fan fault
15	PSU power fault

16	VHF fault
17	Driver current fault
18	Mains voltage fault
19	Temperature fault
20	HT fault
21	PSU shutdown fault
22	DC fault
23	Starting fault

Incident codes

-1	PSU current attention
-2	Driver impedance 1 attention
-3	Driver impedance 2 attention
-4	Driver impedance 3 attention
-5	Driver impedance 4 attention
-6	Driver impedance 5 attention
-7	Driver impedance 6 attention
-8	Driver impedance 7 attention
-9	Driver impedance 8 attention
-10	Slow fan attention
-11	PSU power attention
-12	Temperature attention
-13	Starting attention

Tipi examples

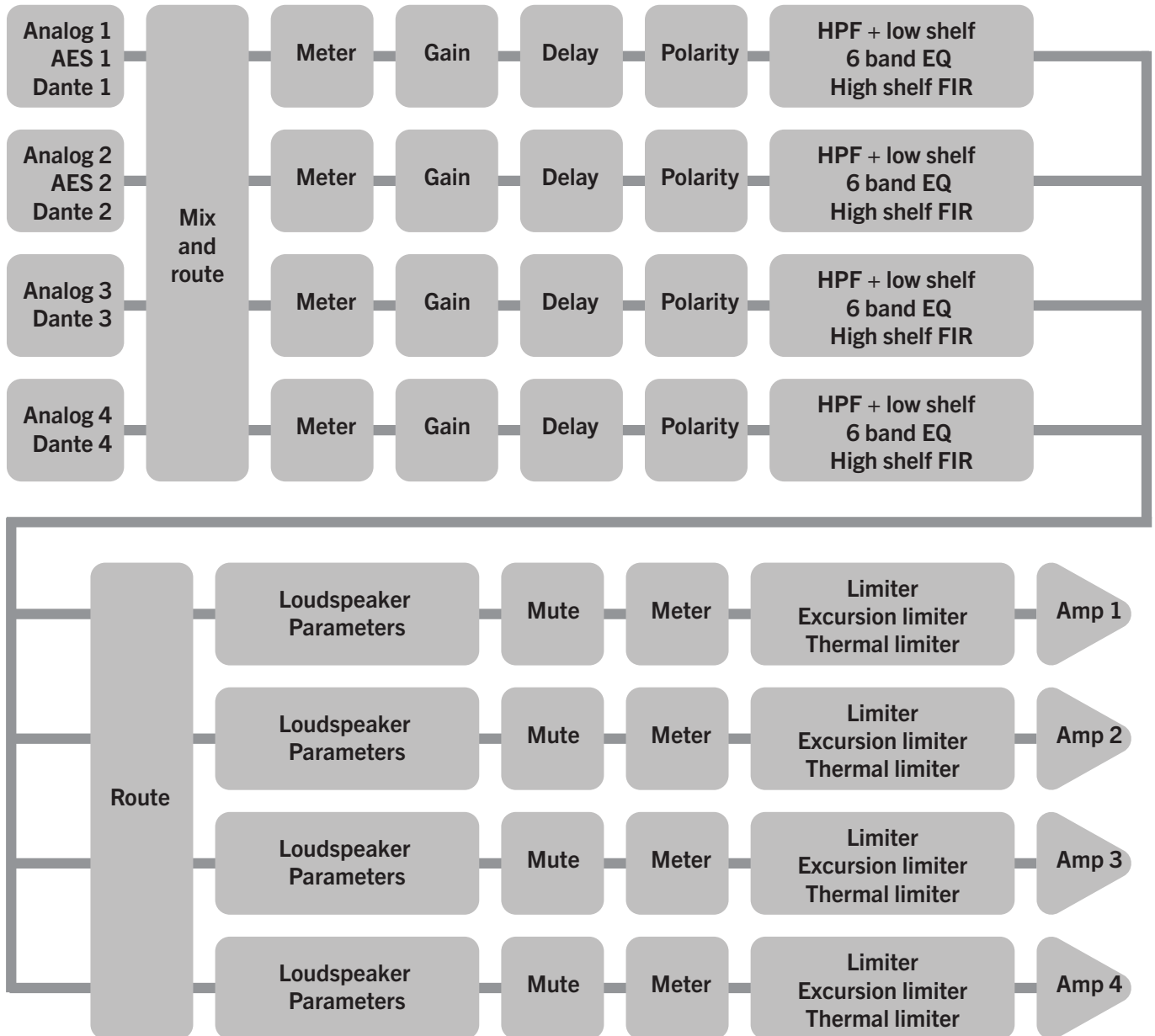
SET examples

Parameter	Example	Notes
Input mute (yes or no)	\$SET InB/Mute yes	Mute input B
Input gain (–40dB to 20dB)	\$SET InA/Gain -3.2dB	Set the gain of input A to –3.2dB
Input polarity (yes or no)	\$SET InC/Pol yes	Reverse polarity for input C
Output mute (yes or no)	\$SET Out3/Mute yes	Mute output 3
Output gain (–40dB to 20dB)	\$SET Out2/Gain –3.2dB	Set the gain of output 2 to –3.2dB
Snapshot (1 to 20)	\$SET Snapshot 5	Recall Snapshot 5
Sleep (yes or no)	\$SET Sleep yes	Sleep the amplifier
Standby (yes or no)	\$SET Standby yes	Set the amplifier to standby

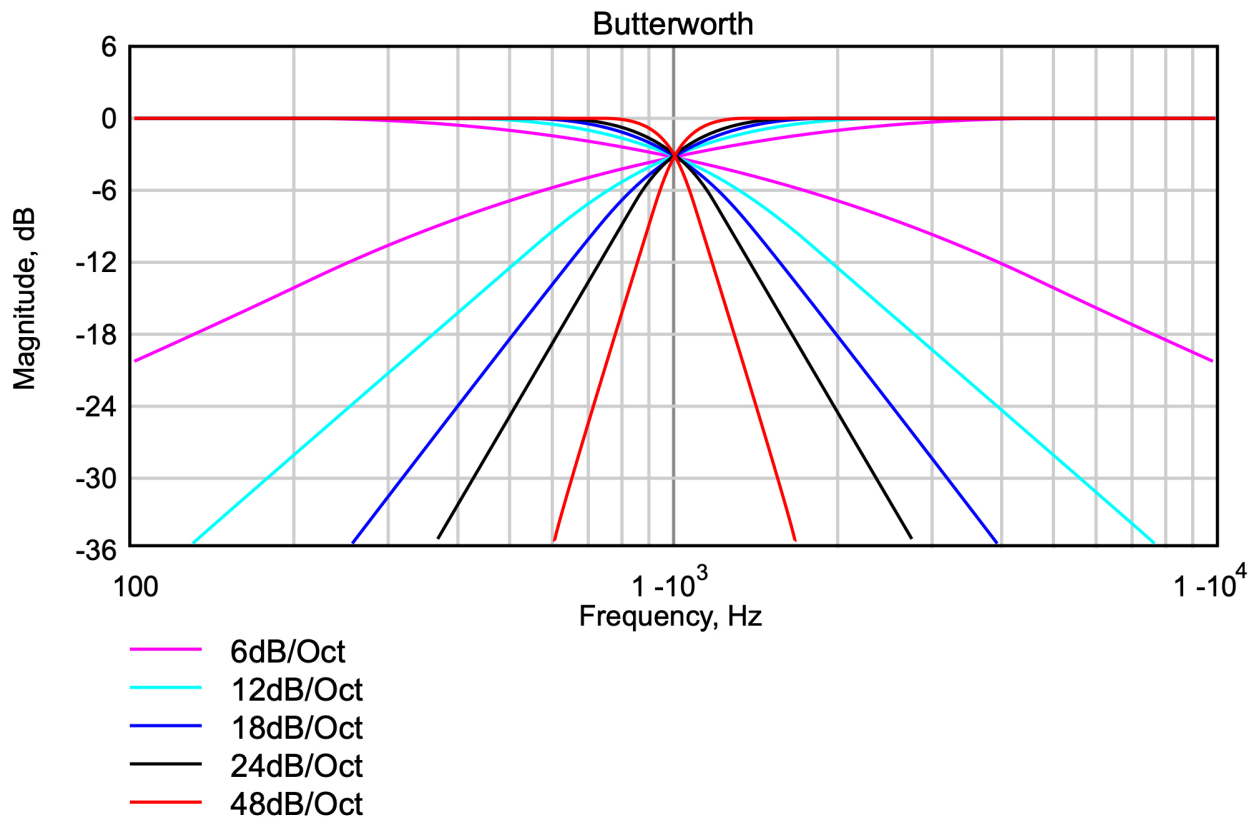
GET examples

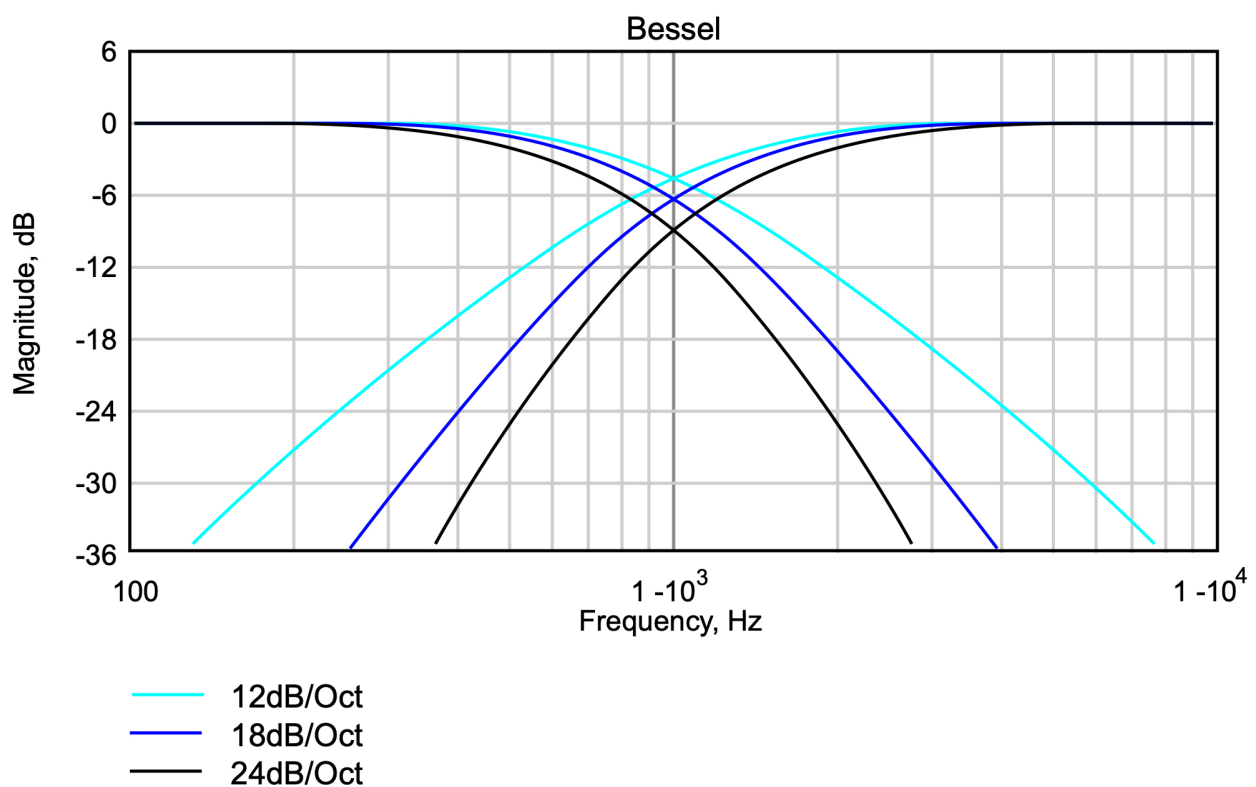
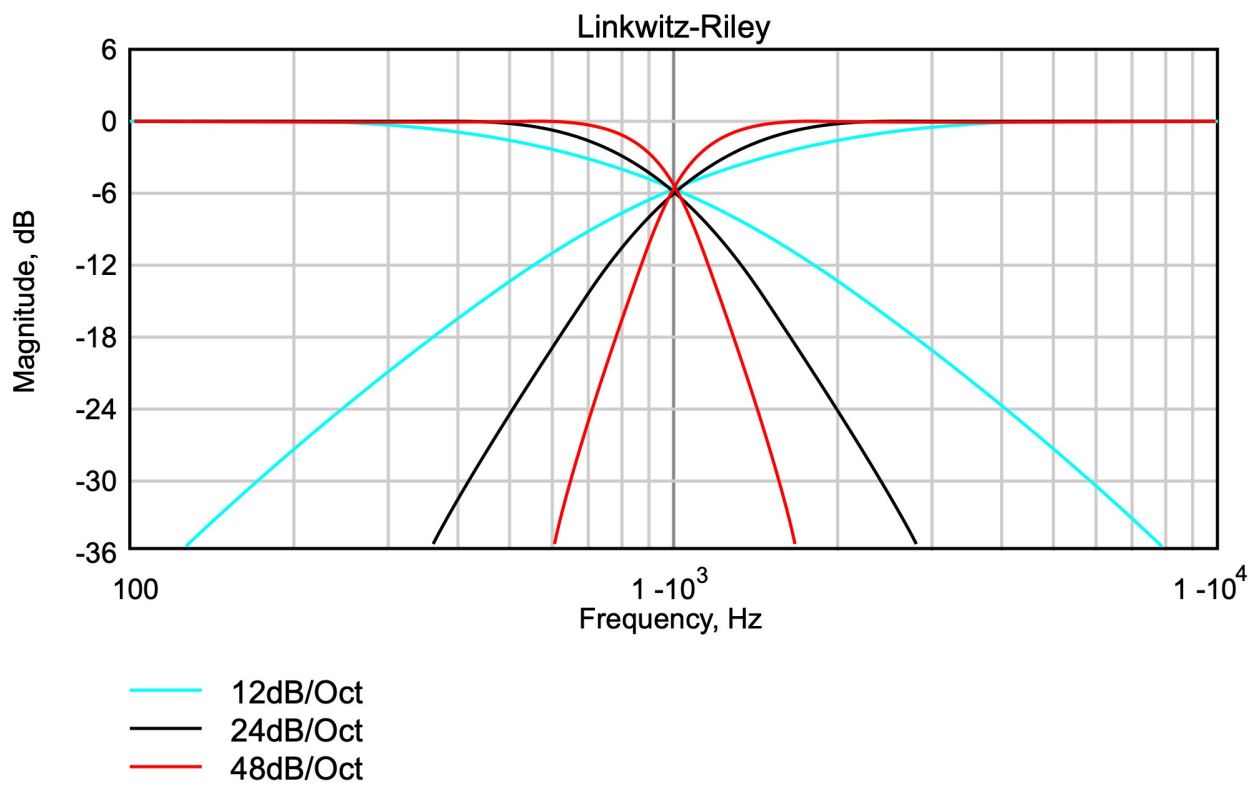
Parameter	Example	Example response
Get input meter	\$GET InD/Meter	\$NOTIFY InD/Meter 1.2dB
Get output meter	\$GET Out3/Meter	\$NOTIFY Out3/Meter 2.3dB
Input mute	\$GET InB/Mute	\$NOTIFY InB/Mute yes
Input gain	\$GET InA/Gain	\$NOTIFY InA/Gain –3.2dB
Input polarity	\$GET InC/Pol	\$NOTIFY InC/Pol yes
Output mute)	\$GET Out3/Mute	\$NOTIFY Out3/Mute no
Output gain	\$GET Out2/Gain	\$NOTIFY Out2/Gain –3.2dB
Drive impedance ("0.1R" when the value can't be determined)	\$GET Out4/Imp	\$NOTIFY Out4/Imp 4.2R
Sleep	\$GET Sleep	\$NOTIFY Sleep no
Standby	\$GET Standby	\$NOTIFY Standby no
Firmware type (not product name)	\$GET Type	\$NOTIFY Type 1234
Firmware version (multiplied by a thousand)	\$GET Version	\$NOTIFY Version 1234

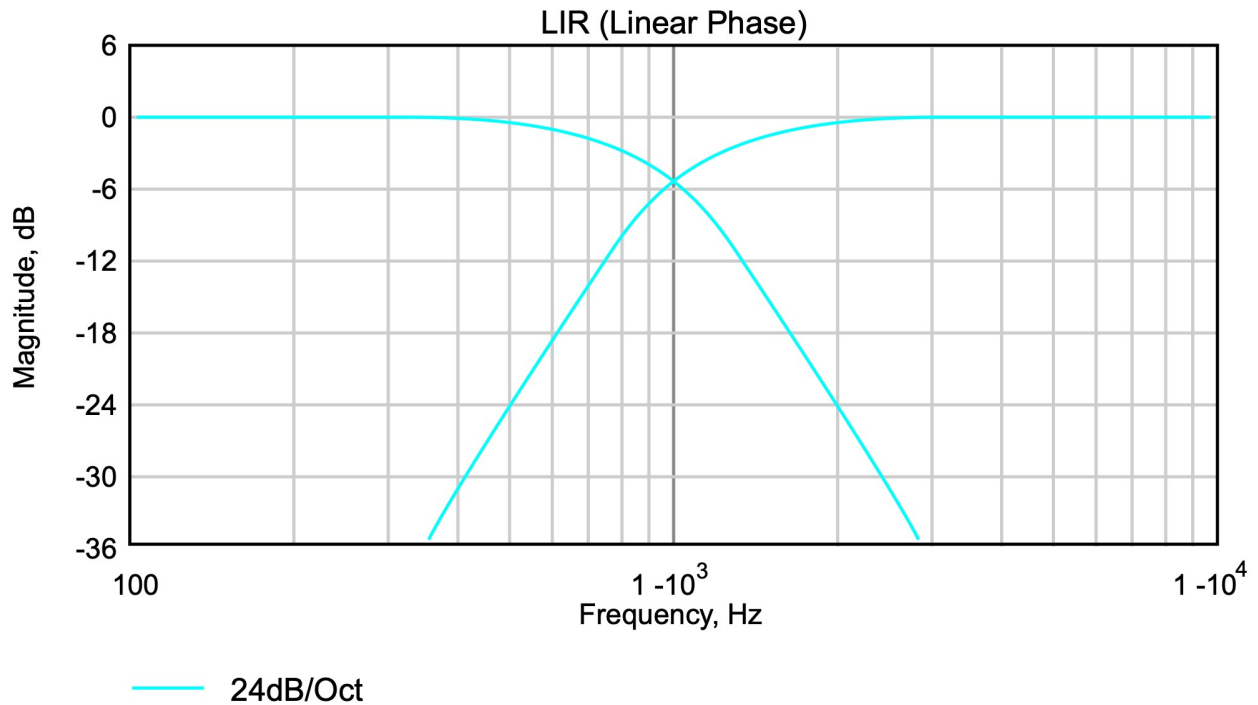
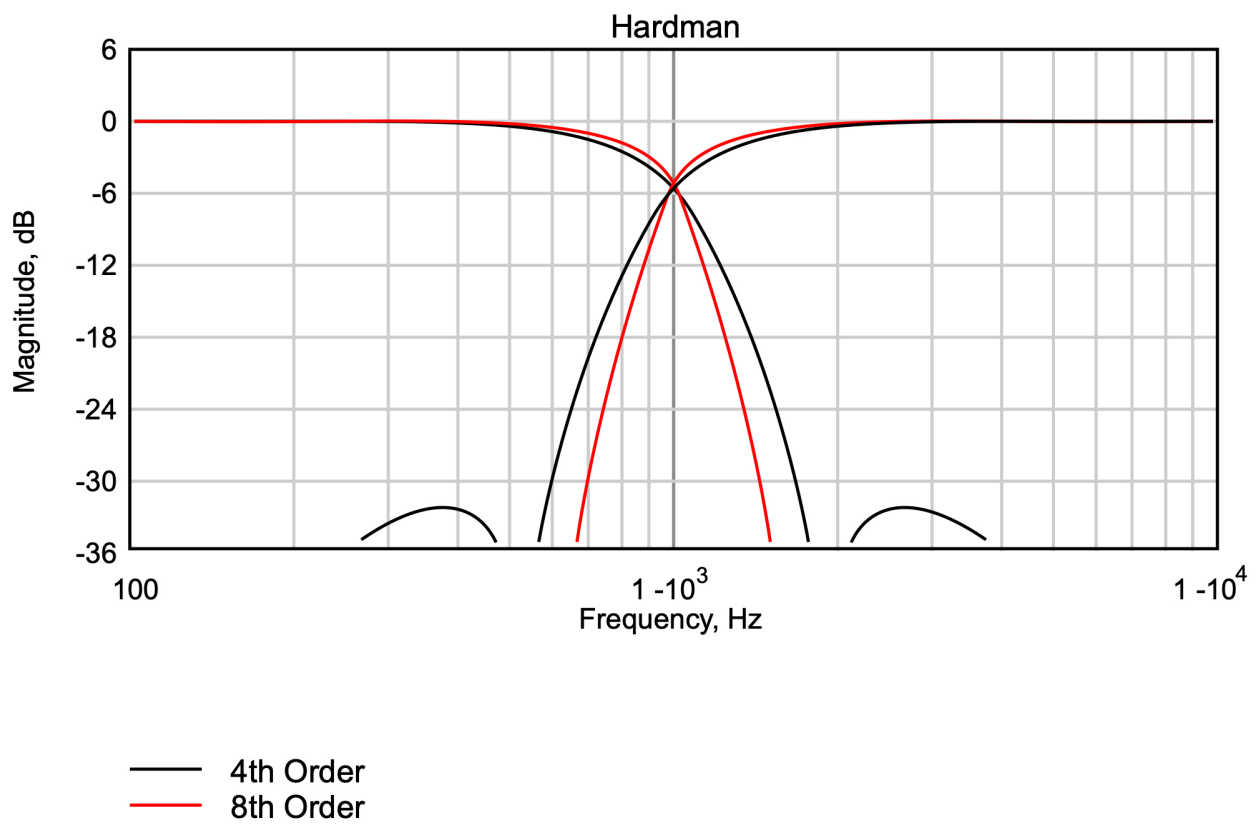
Signal path

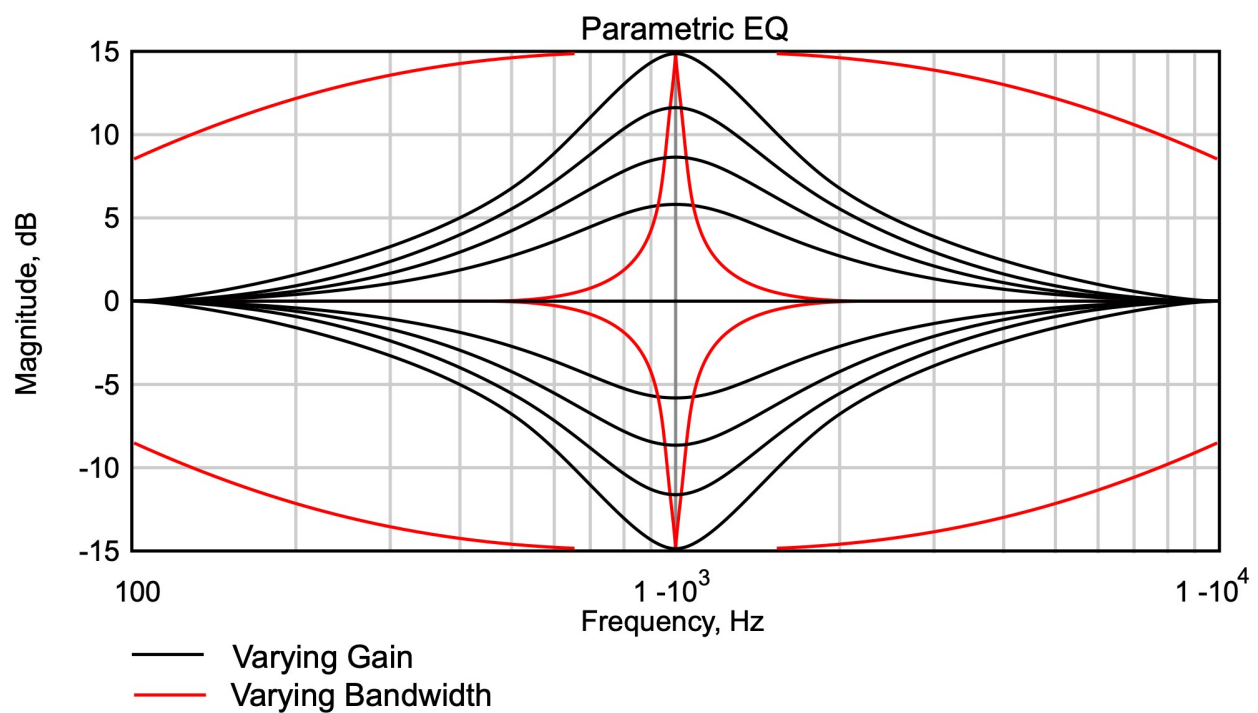
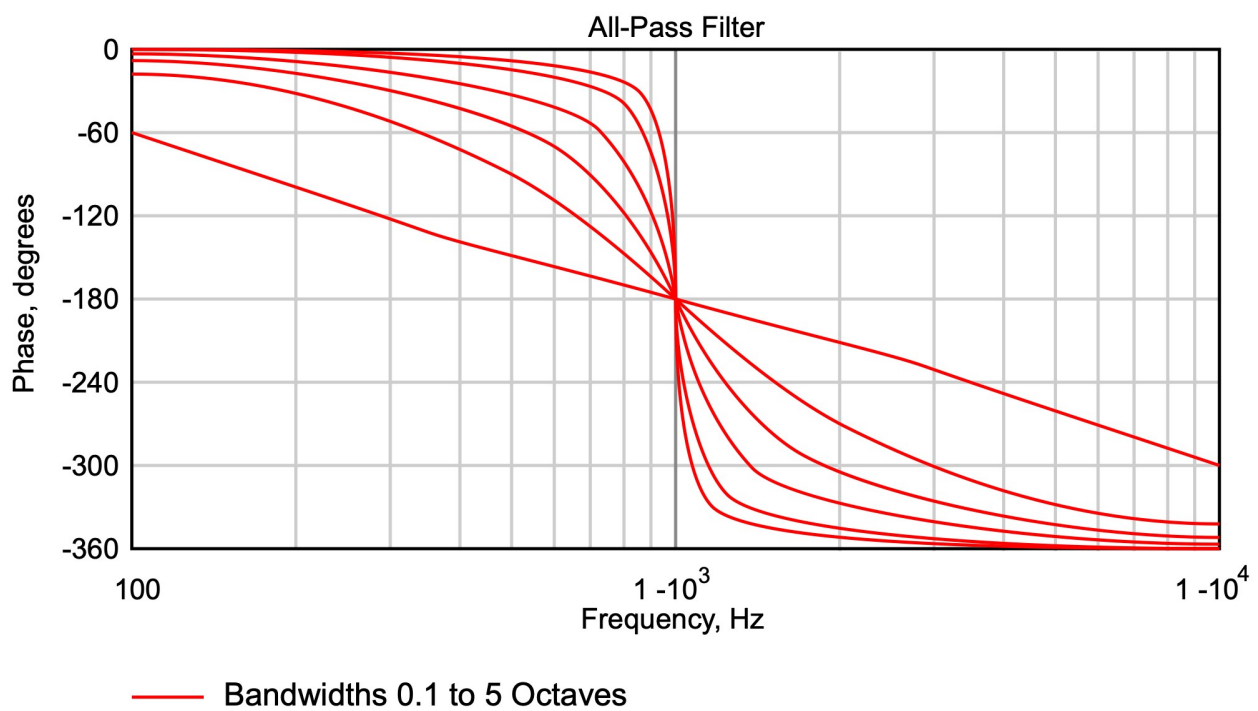


EQ and filter response graphs









Power draw and thermal dissipation

Sleep mode (slow wake up)					
AC Mains Power Draw (watts)	Current Draw (amps)		Thermal Dissipation		
	120Vac	230Vac	watts	kcal/hr	Btu/hr
4.5	0.4	0.2	4.5	4	15

Standby mode (fast wake up)					
AC Mains Power Draw (watts)	Current Draw (amps)		Thermal Dissipation		
	120Vac	230Vac	watts	kcal/hr	Btu/hr
60	1.0	0.5	60	52	205

Running with no audio signal					
AC Mains Power Draw (watts)	Current Draw (amps)		Thermal Dissipation		
	120Vac	230Vac	watts	kcal/hr	Btu/hr
195	2.9	1.5	195	168	665

Running in 2 ohm mode (all channels driven)								
Load mode	Load (ohm)	Signal duty and Crest Factor	Input power (watts)	Current Draw (amps)		Thermal Dissipation		
				120Vac	230Vac	watts	kcal/hr	Btu/hr
2 ohm ^a	2	1/8, cf = 4.0 (12dB)	3,000	33.5 ^b	17.5	500	430	1,706
2 ohm ^a	4	1/4, cf = 2.8 (9dB)	3,475	28.8 ^b	20.3	475	408	1,621
2 ohm ^a	4	1/8, cf = 4.0 (12dB)	1,780	19.7	10.3	280	241	955
2 ohm ^a	8	1/4, cf = 2.8 (9dB)	1,750	19.2	10.0	250	215	853
2 ohm ^a	8	1/8, cf = 4.0 (12dB)	975	11.0	5.8	225	193	767
100V	–	1/8, cf = 4.0 (12dB)	3,000	33.5 ^b	17.5	500	430	1,706
70V	–	1/8, cf = 4.0 (12dB)	2,159	23	12	409	351	1,394
25V	–	1/8, cf = 4.0 (12dB)	896	10.2	5.3	271	233	926

^aThe iK42 does not have 4 and 8 ohm low Z modes

^bThe EBP limiter should be set to 32A, but will not activate on any sensible program material

- Measurements done using a Hameg HM8115-2 power analyser
- Amplifier configured to have no audio processing
- All measurements done at 230Vac, 50Hz
- Current Draw values for 120Vac are calculated values

Technical specifications

General	
Number of output channels	Four
Total power output	20,000 watts RMS, depending on load impedance
Input types	Analogue, AES3, Dante (page 6)
Control, monitoring & alarm	Ethernet, configurable function Volt-free relay and contact closure port
Energy saving modes	Standby and deep sleep, both with auto-sleep timers
System sleep and wakeup	Front panel switch, network command, contact closure and audio detection
Max ambient temperature (full power, no limiting)	40°C (104°F)

Audio	
Amplifier topology	Proprietary 5th generation Martin Audio Class D
Amplifier modulation scheme	Low feedback, multiple loop, with feedforward error correction
Dynamic range (analogue input to speaker output)	>113 dBA typically
Dynamic range (AES3 or Dante input to output)	>114 dBA typically
Gain (with all DSP level controls set to 0 dB)	32 dB
Frequency response	<7 Hz to >30 kHz, 4 ohm, -2.5 dB
Total harmonic distortion, THD	<0.05% typ, 1 kHz, AES 17, 4 ohm load
Inter-channel crosstalk (worst combination of channels)	Better than -85 dB at 1 kHz and -75 dB at 10 kHz
Slew Rate	>60 V per microsecond typical
Damping factor (Ref 8 ohm)	>800 at amplifier output
Maximum analogue input level	+20 dBu
Analogue input sensitivity range for full output	0 dBu to +20 dBu, continuously adjustable
Analogue input	20,000 ohm, electronically balanced
Analogue link	Directly connected to the analogue input
Analogue ground scheme	AES48 standard compliant
AES3 input	Transformer isolated with active cable equalisation for extended range
AES3 link	Active signal regeneration with automatic direct bypass to the AES3 input if the unit is unpowered
AES3 supported sampling rates	24 kHz to 192 kHz (auto locking)

Digital processing	
Resolution	40 bit, using proprietary Linea Micro Detail (LMD) algorithms
Sample rate	96 kHz throughout
Special functionality:	Class leading limiter suite, see Protection systems (page 42)
	Hardman crossover filters (better out of band rejection than Linkwitz-Riley)
	LIR crossover filters (Linear Phase without the compromises of FIR filters)
	FIR Shelving EQ filters (for linear phase filtering)

Power output	
Power specification	RMS output power per channel, all channels driven with continuous program material and a nominal ambient temperature of 40°C (105°F)
	5,000 W into 2 ohm
	3,000 W into 4 ohm
	1,500 W into 8 ohm
	750 W into 16 ohm
	10,000 W bridged per channel pair, 4 ohm
	6,000 W bridged per channel pair, 8 ohm
CV line output	1,250 W, 25 V line
	3,500 W, 70 V line
	5,000 W, 100 V line

Power supply	
Topology (main power supply)	3rd generation Series Resonant
Topology (auxiliary and standby supplies)	Low quiescent Eco-Flyback
Internally stored energy	>600 joules
Mains input voltage range (automatically configured)	85 V to 240 V
Mains input frequency range	47 Hz to 63 Hz
Mains inrush current (max for <10 ms)	6 A at 115 V, 12 A at 230 V

Protection systems

System protection	
	The control and protection systems endeavour to deliver the maximum power possible for a given set of conditions, applying limiters only in extreme circumstances
	Muting will only occur when a dangerous situation is detected and normal operation will automatically resume when the condition clears
	Detect excessive output current

System protection

Detect excessive power supply current
Detect excessive amplifier section temperature
Detect excessive power supply section temperature
Detect excessive DSP section temperature
Detect mains voltage out of range
Detect fan speeds out of range
Detect internal power rails out of range

Speaker protection

Audio soft-clip limiter
VxLim, Multiband peak limiter
VxMax, Multiband overshoot limiter
Vx-Xmax, Driver excursion limiter
Vx-Tmax, Driver thermal limiter (long term power limiter)
DC offset protection
Excessive HF energy (VHF) limiter

Power distribution protection systems

Mains inrush current limiting (soft start and anti-surge)
Mains average current limiting (mains breaker / fuse trip protection)
Randomised initialisation when powered up to reduce the peak power demand in large systems

Monitoring and logging

You can configure the inbuilt alarm and notification system to indicate problems to remote devices either via the network or the [fault relay port \(page 42\)](#)

Supply current logged vs time
Supply voltage logged vs time
Thermal Capacity logged vs time
Each driver current logged vs time
Each driver impedance logged vs time
Protection limiting for each output logged vs time
Number of power cycles counted
Number of mains brownout events counted
Fan speeds continuously monitored
Fan under-speed events counted
Various protection mute events counted
Driver Impedance continuously monitored

Physical

Cooling	Variable speed fans
Airflow	Front to back
Air filtration	Washable media, changeable without the use of tools

Physical

Analogue IN and LINK connectors	Genuine Neutrik™ XLR
AES3 IN and LINK connectors	Genuine Neutrik™ XLR
Audio output connector	Genuine Neutrik™ speakON
Mains input connector	Genuine Neutrik™ 32A powerCON
Dante Primary and Secondary (page 6)	Shielded RJ45
Relay output and contact closure inputs	Phoenix pluggable terminal block.
Front panel display	Graphical, backlit, high contrast, daylight visible
Front panel encoders	Two, indented, velocity sensitive
Front panel push buttons	Large, tactile, illuminated
LED indicators	Bright, easily differentiated
Enclosure	Standard 19" 2U (88 mm) with handles and optional rear support system
Depth (behind rack ears)	357 mm (14")
Net Weight	12.5 kg (27.5 pounds)

Options

Rear rack ears (page 38) (rear support for iKON amplifiers installed in 19" racks)	Part number RACKKITC
--	----------------------

Loudspeaker compatibility

The following table shows the compatibility of the iK42 with core Martin Audio loudspeakers.

		16 Ω 325w	8 Ω 750w	4 Ω 1500w	2 Ω 1500w	8 Ω Bridged 3000w	4 Ω Bridged 3000w
CDD	CDD5	N/A	✓	✓	✓	N/A	N/A
	CDD6	N/A	✓	✓	✓	N/A	N/A
	CDD8	N/A	✓	✓	✓	N/A	N/A
	CDD10	N/A	✓	✓	✓	N/A	N/A
	CDD12	N/A	✓	✓	✓	N/A	N/A
	CDD15	N/A	✓	✓	✓	N/A	N/A
FlexPoint	FP4	✓	✓	✓	✓	N/A	N/A
	FP6	✓	✓	✓	✓	N/A	N/A
	FP8	N/A	✓	✓	✓	N/A	N/A
	FP12	N/A	✓	✓	✓	N/A	N/A
	FP15	N/A	✓	✓	✓	N/A	N/A
LE monitors	LE100	N/A	✓	✓	✓	N/A	N/A
	LE200	N/A	✓	✓	✓	N/A	N/A
O-LINE	O-LINE	✓	✓	✓	✓	N/A	N/A
SX subwoofers	SX110	N/A	✓	✓	✓	N/A	N/A
	SX210	N/A	✓	✓	✓	N/A	N/A
	SX112	N/A	✓	✓	✓	N/A	N/A
	SX212	N/A	N/A	✓	✓	N/A	N/A
	SX115	N/A	✓	✓	-1.1dB	✓	✓
	SXC(F)115	N/A	-1.2dB	-1.2dB	x	✓	✓
	SX215	N/A	N/A	-0.3dB	-1.1dB	N/A	✓
	SX118	N/A	-1.2dB	-1.2dB	x	✓	✓
	SXC(F)118	N/A	-1.2dB	-1.2dB	x	✓	✓
	SX218	N/A	-1.2dB	-1.2dB	x	✓	✓
TH series	SXH(F)218	N/A	N/A	-3.0dB	x	N/A	✓
	THS	N/A	-0.2dB	-0.3dB	-1.1dB	N/A	N/A
TORUS	THH, THV	N/A	✓	✓	-0.8dB	N/A	N/A
	T8	N/A	✓	✓	✓	N/A	N/A
Wavefront Precision	T12	N/A	✓	✓	✓	N/A	N/A
	WPM	✓	✓	✓	✓	N/A	N/A
	WPS	N/A	✓	✓	N/A	N/A	N/A
	WPC	N/A	✓	✓	N/A	N/A	N/A
XE monitors	WPL	N/A	✓	✓	N/A	N/A	N/A
	XE300	N/A	✓	✓	✓	N/A	N/A
	XE500	N/A	N/A	✓	✓	N/A	N/A

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