iK41 User Guide

MARTIN AUDIO
LONDON

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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 iK41 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 – packs four output channels and 6,000 kW into just 2U of rack space
- 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)
- 96kHz sampling frequency provides for a nominally flat response beyond 40kHz
- Rotary encoders, illuminated buttons and graphical display provide a rapid, intuitive and user-friendly control interface
- 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
- Dante audio networking with automatic fallover to Analogue or AES3
- AES3 inputs

Drive modules

The iK41 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 31) 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 subsystem).

For further details, see Modules, preset components and snapshots (page 15).

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 equilisation solves this problem.

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 a rear rack mounting kit (part number RACKKITC). 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.

AC Power Connection

The amplifier uses a 32A Neutrik powerCON locking AC power connector. Use only an AC power cable with a correctly terminated powerCON connector to make the connection to the mains power supply.

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 voltages, for details see Technical specifications (page 42). 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 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.

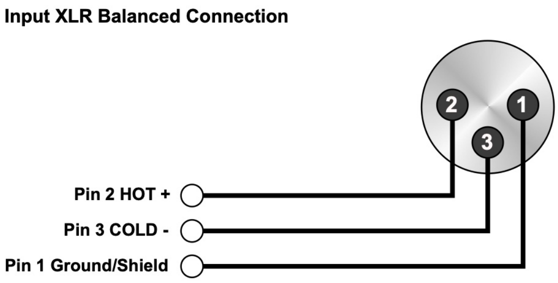
Audio connections

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.

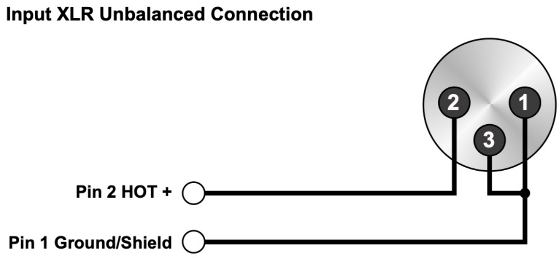


Unbalanced input connections

We don't recommend unbalanced connections.

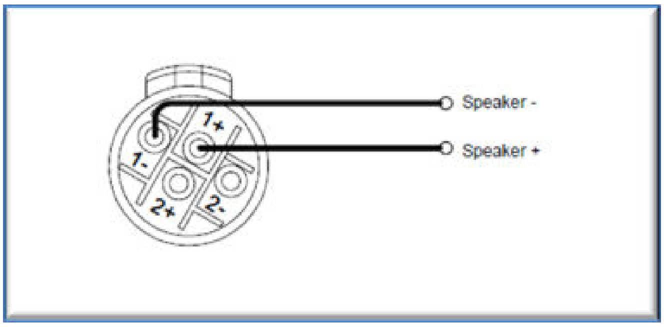
For an unbalanced audio source, use a cable connected as follows:

- 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.



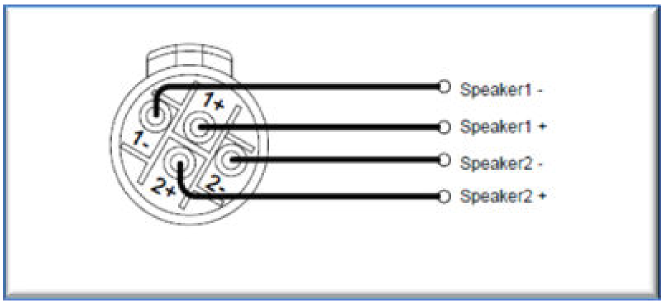
Output connections

The iK41 and iK42 amplifiers are fitted with one speakON connector per amplifier channel. The appropriate conductor terminations are shown below and on the rear panel of the unit.

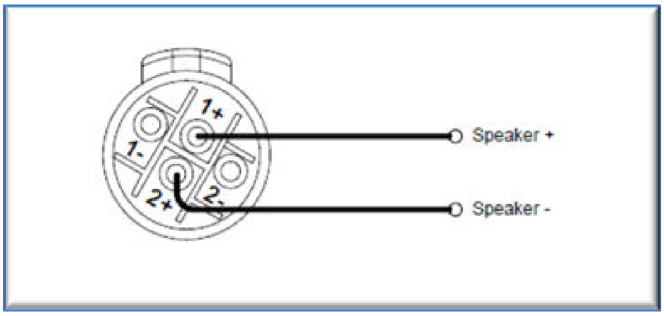


Additionally, the channel 2 output is duplicated on the speakON connector for amplifier channel 1 for bi-amp wiring. Similarly, the channel 4 output is duplicated on the speakON connector for amplifier channel 3. This can be useful for making a connection to two loudspeakers with one 4-core cable (i.e. bi-amp).

On the iK81 amplifier, all outputs are bi-amp; each speakON connector carries two amplifier outputs – Channels 1 and 2, Channels 3 and 4, Channels 5 and 6 and Channels 7 and 8.



In addition, the channel 1 or channel 3 connector (all the speakON connectors on the iK81 model) can also be used if the pair of amplifier channels is being operated in bridged mode.



More than one speaker can be connected 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.

Load matching

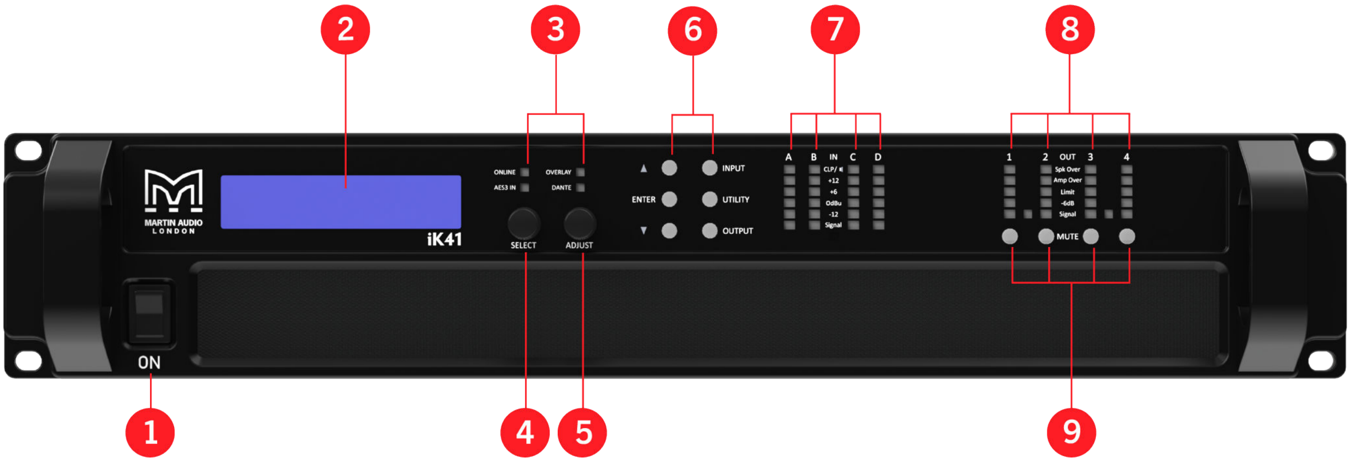
Each output of the device can be optimised to drive either a low impedance load (2, 3, 4, 6 or 8 ohms) or a constant voltage (25V, 70V or 100V line).

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 24).

Front panel



1. **Power switch**. The mains power on-off switch is at bottom front left. If the device is in sleep mode (page 26), you can wake it up using Vu-Net (page 31) or by switching this power switch off and on again.
2. **Graphical display**. The home screen shows an overview of the channel allocation. When you view the input (page 19), output (page 22)or utility (page 27) menus, the menu 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 18) and you won't be able to edit any settings. To change the contrast, see Screen contrast (page 27).
3. **Status Indicators**. The **ONLINE** indicator has three states:
   - **Off**. The unit isn't connected to Vu-Net (page 31).
   - **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 31).

   The **AES3 IN** indicator illuminates when one or more of the inputs is using an AES3 source.

   The **DANTE** indicator illuminates when there is a Dante network feed.
4. **SELECT** (the left-hand dial) allows you to switch between parameters.
5. **ADJUST** (the right-hand dial) allows you to edit parameters.
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. For changes where you need to press the **ENTER** button to confirm, the **ENTER** button illuminates.

   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 19). 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 26). The menus are mutually exclusive – if you press one of these buttons you switch to that menu. To escape 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 19).

8. **Limiter Indicators.** The output indicators shows the status of the limiter and output level. The level indicated is that before the limiter, referenced to the limiter threshold. The `Signal` indicator shows when a signal is present on the output. The second indicator **–6dB** shows that the signal has reached 6 dB below the limiter threshold. The third indicator `Limit` indicates that the threshold of that output channel has been reached.

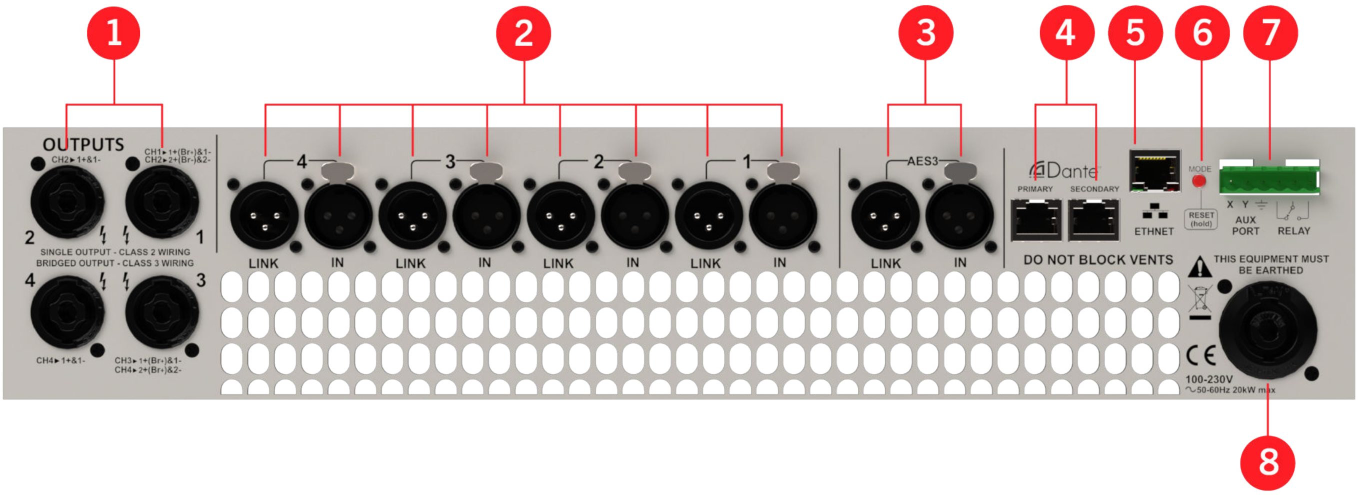
   Bridge Indicator. This illuminates when the channel pair is in Bridge mode (page 24). The controls for the left channel of the pair will determine the settings.

   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 6dB 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



1. **Loudspeaker Connectors**. The amplifier speakON outputs. Connect a loudspeaker to terminals 1+ and 1–. CH1 also carries the (duplicated) loudspeaker output for channel 2 on terminals 2+ and 2–. Similarly, CH3 carries the loudspeaker output for channel 4 on terminals 2+ and 2–. For Bridged mode (page 24), use terminals 1+ and 2+ of CH1 and/or CH3.
2. **Analogue Audio Link Connectors**. Carries a duplicate (parallel) connection (to another amplifier for example).
3. **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.
4. **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.
5. **Dante Ports**. Connection ports for Dante™ with the Primary and Secondary port convention available.
6. **Ethernet Communications Port**. Use this port to connect your amplifier and PC. This allows you to control the amplifier using Martin Audio Vu-Net (page 31). This software also allows you to update firmware and to integrate with other Martin Audio products such as MLA Series and CDD-Live.
7. **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 28).
8. **Auxiliary Port**. You can control the amplifier via this port by recalling snapshots or by muting or sleeping the amplifier. For details, see AUX port (page 32).
9. **Power Inlet**. The unit should be connected 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.

> ⚠ 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.

10. **Relay output**. This isolated relay output can be used to indicate abnormal conditions to external monitoring apparatus. For details, see Fault Relay (page 34).

Example system setup

The iK41 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 an example, this section shows you how to set up a system using an iK41 and XE300 stage monitors.

To load XE300 bi-amp presets

1. Connect the amplifier to the mains, to an input feed and to the monitors.

   The input feed can be balanced analogue to the input XLRs, AES3 to the dedicated AES XLR or Dante to the Dante RJ45 port.

   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 8).

2. Power up the amplifier. When this is complete, press **INPUT** so that **A** appears top left. This shows that you are editing input channel A. Press down arrow ▼ once so that **RECL** (for Recall Preset) appears top left.
3. Rotate **ADJUST** (the right-hand dial) so that **XE300 BI** appears bottom right.
4. Press **ENTER**. The following message will appear and the **ENTER** button will start flashing:

5. Press **ENTER** to confirm. Input A now feeds output 1 with XE300 LF parameters and output 3 with XE300 HF parameters.
6. 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**:
7. Rotate **ADJUST** (the right-hand dial) to select the **XE300 BI** preset and press **ENTER**

8. 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 XE300 BI 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 passive presets

1. Connect the amplifier to the mains, to an input feed and to the monitors.

   The input feed can be balanced analogue to the input XLRs, AES3 to the dedicated AES XLR or Dante to the Dante RJ45 port.

   With the XE300 in passive mode, you can connect directly to speakON outputs 1, 2, 3 and 4.

2. Power up the amplifier. When this is complete, press **INPUT** and press down arrow ▼ once to show the Preset recall for Channel A.
3. Rotate **ADJUST** (the dial on the right) until the XE300 passive preset appears.
4. Press **ENTER** and the following message appears:

5. Press **ENTER** to confirm.
6. Press **INPUT** again to select input channel B.
7. 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 subsystem (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

Navigation

Parameter navigation and adjustment on the iK41 is very straightforward. There is no concept of drilling down deeper into hidden menus; every parameter is accessible by simply scrolling across a ‘map’ of parameter pages which can be thought of as placed on a two-dimensional grid. Horizontally across the width of the grid are the various channels, and vertically up and down the grid are the parameter pages for each section of processing.

To view a parameter, repeatedly press the desired `INPUT` or `OUTPUT` channel button until the desired channel is reached. Then repeatedly press the up and down buttons to scroll through the processing parameters for the selected input/output. For example, to adjust the first band of parametric EQ on input C, press 'Input' three times, 'C' will be displayed on the left of the LCD window. Then press 'Up' five times. The display will scroll from Gain to Delay, High pass filter, two low shelf filters before arriving at the first parametric which may now be edited using the select and adjust controls. Two encoders allow you to select and adjust a parameter. Often, several parameters will be shown in various zones on the display. To select a parameter for adjustment, turn the left-most encoder such that the parameter you wish to adjust is highlighted. Then turn the right-most encoder to adjust the value of that parameter. Turning this encoder clockwise will increase the value of a parameter, or anticlockwise will decrease it. The encoders are velocity-sensitive so turning an encoder rapidly will cause the action to ‘accelerate’, so the value changes more rapidly.

Home screen

The home screen shows an overview of the configuration of the device. The user-defined 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 to this channel. The bottom line shows the physical input number and a list of the outputs which are routed from the Drive Module.

Drive modules

The iK41 uses 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 2-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 the Drive Module control panel in Vu-Net (page 31).

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 iK41 has four audio inputs and four DSP inputs. This is a matrix mixing system where any physical inputs, be they analogue, AES3 or networked audio feeds, 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 may 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 the routing has been changed manually, then the whole arrangement may be saved into a Snapshot. For further details, see Modules, preset components and snapshots (page 15).

Factory module presets

The device will contain a library of Factory Presets designed to suit a range of enclosures.

Factory Presets may contain some parameters that are fixed and hidden from view; the remainder of the parameters are available for user manipulation. The number and type of hidden parameters is dependent on the Factory Preset, typically Device Name DSP Input Input Name Physical Input Outputs assigned crossover frequencies, output delay and some EQs are hidden; those settings that are a function of the loudspeaker cabinet design and should not require adjustment for different applications. Factory Presets are locked (as indicated by a 'box' symbol after the Preset name) so they cannot be overwritten. The user can, however, store an edited version of a Factory Preset in any free preset location.

In addition to the Factory Presets the device may have further 'Skeleton Presets' which will help to create new presets. They can be used to develop settings for any loudspeaker combination and are recalled in the same way as the Factory Presets described above. These Presets are also usually locked but the user can name and store their own edited versions in any free preset location.

Storing module presets

Once a drive module has been created it can be stored by pressing the **INPUT** button until the edited channel is reached, then pressing the down button until store page is reached. Using **ADJUST** encoder will change the preset number. When the destination preset is reached, pressing the **ENTER** button will enable the name associated with that preset to be changed. Once the name changing is active, the character to be changed will be highlighted and the **ADJUST** encoder will edit the character. Using the **SELECT** encoder will move through the character positions. Once the new preset name has been assembled, the operation can be confirmed by pressing the **ENTER** button, then a message will be displayed, **Enter to confirm or ▼ to exit**; pressing **ENTER** will store the preset.

Storing a Drive Module preset for a module which is not configured with consecutive outputs is not permitted.

When storing a Drive Module preset on the device, Component Names cannot be edited. To change Component names, you need to save the Module Preset using Vu-Net (page 31).

Recalling module presets

To recall a Drive Module preset, press the **INPUT** button, then use the down ▼ button navigate to the RECL Preset page. Using the **ADJUST** encoder will scroll through the presets available. When the desired preset is reached, pressing **ENTER** will display the message **Enter to confirm or ▼ to exit**, pressing **ENTER** will recall the preset.

Recalling components

To recall a Component Preset (to a single output), press the **OUTPUT** button, then use the down button navigate to the RECL Preset page. Using the encoder, A will scroll through the component presets available (as indicated by the ModulePreset.Component number and ModulePreset.Component name). When the desired component is reached, pressing **ENTER** will flash the Enter button. Pressing **ENTER** again will then recall the component preset.

Snapshots

You can recall snapshots using:

- Vu-Net software (page 31)
- The amplifier front panel (page 10)
- Signals sent to the AUX port (page 32) on the rear of the amplifier.

The snapshot menu (page 28) is accessed via the utility pages. Recalling a Snapshot (page 28) triggers the recalling of a Component to each input and output, and may change other device-wide settings, effectively recalling a processor-wide preset.

The parameters inside the Drive Modules are not individually stored in Snapshots. Recalling a Snapshot will merely trigger 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 may 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 15).

Secure mode

Secure mode stops people from updating settings at the front panel. When the amplifier is connected to Vu-Net (page 31), the amplifier automatically switches to secure mode. If you're not using Vu-Net, you can switch on secure mode manually.

Note that in secure mode you can still:

- Mute and unmute output channels.
- View but not edit the input, output and utility settings.

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, you can't switch off secure mode.

Revert to factory settings

If you wish to start with a clean state, you can erase all settings and revert to the factory defaults.

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 ▲ button and the mute 1 button for about 8 seconds.
2. To confirm, press **ENTER**.

   To cancel, press down arrow ▼.

Input menu

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**. The channel letter appears top left to show which channel you're editing. To edit another channel, press **INPUT** again.
2. **GAIN** should appear top left. If not, press down arrow ▼ several times until it does.
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose the gain (–40 to 20 dB in 0.2 increments).
4. Rotate **SELECT** (the left-hand dial) to select **Pol** for polarity.
5. Rotate **ADJUST** (the right-hand dial) to select **Norm** for normal or **Rev** for reverse polarity.
6. Rotate **SELECT** (the left-hand dial) to the right to select **Mute**.
7. Rotate **ADJUST** (the right-hand dial) to select **Norm** for normal or **Mute**.
8. To edit the next channel, press **INPUT**.
9. To exit the menu, press **UTILITY** or press **INPUT** several times to step through all the input channels.

Fallover type

The Fallover type allows you to select what type of input to use by default and which type of input to use if the default input fails.

When Fallover AES3>Analogue is selected (on an input channel which supports AES3), then the AES3 source will be automatically selected if it has a valid audio stream on it. If the AES3 stream should fail, then Analogue is automatically selected instead. When Fallover Dante>Analogue is selected (when the Dante option is fitted), then the Dante source will be automatically selected if it has a valid audio stream on it. If the Dante stream should fail, then Analogue is automatically selected instead.

Similarly, it is possible to select Fallover Dante>AES3 on a channel that supports AES3 and Dante is present.

The 'Auto' setting allows the highest priority input source that is active to be automatically selected, so the user could just plug a source into any input and it will be automatically selected. The priorities are: Dante first, AES3 second, Analogue third.

Note that any automatic selection will take precedence over manual selection, so if you try to manually select Dante when there is no valid Dante stream, then it will revert to the Fallover input source.

To change the fallover type

1. Press **INPUT**. The channel letter appears top left to show which channel you're editing. To edit another channel, press **INPUT** again.
2. Press down arrow ▼ several times until **TYPE** appears top left (usually five presses).
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose fallover type. This can be Manual, Dante>AES3, Dante>Analog, AES3>Analog, Auto.
4. Rotate **SELECT** (the left-hand dial) to select **Type**.
5. Rotate **ADJUST** (the right-hand dial) to the left or right to select the type.
6. To edit the next channel, press **INPUT**.
7. To exit the menu, press **UTILITY** or press **INPUT** several times to step through all the input channels.

Trim digital input gain

In addition to the usual analogue inputs, the Device can also accept AES3 digital inputs or Dante digital input.

AES inputs

When a DSP input channel is assigned to an AES3 channel, the "AES3" indicator illuminates.

There is no 'standard' for the relative gains between Analogue and AES3 so depending on the levels delivered by the audio source, 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, it is possible to select any channel(s) 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 Type menu Type parameter to “Dante”.

When Dante has been selected as part of the input routing, the “Dante” indicator above the dials illuminates. This indicator will come on even if there are no cables plugged into the networked audio port on the Device.

As for the AES3 inputs, you can set the relative gain between an Analogue input and the signals 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**. The channel letter appears top left to show which channel you're editing. To edit another channel, press **INPUT** again.
2. Press down arrow ▼ several times until **TRIM** appears top left (usually six presses).
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose the AES3 gain (–40 to 20 dB in 0.2 increments).
4. Rotate **SELECT** (the left-hand dial) to select **Dante**.
5. Rotate **ADJUST** (the right-hand dial) to the left or right to choose the Dante gain (–40 to 20 dB in 0.2 increments).
6. To edit the next channel, press **INPUT**.
7. To exit the menu, press **UTILITY** or press **INPUT** several times to step through all the input channels.

Delay

The delay page which controls the amount of delay associated with the input channel selected and is adjustable from 0 to 998ms. The delay parameter is adjustable in fine steps at low values; the adjustment becomes progressively coarser as the value increases.

High Pass Filter

System high pass filtering is provided for the input signal. Filter type is selectable from 1st order, Butterworth, Bessel, Linkwitz- Riley and Hardman. Filter slopes of up to 4th order or 24dB / octave are provided. Not all filter types are available in all slopes. For example 18dB / octave Linkwitz-Riley filters do not exist.

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

Parametric Equalisation

There are nine stages of equalisation available for each input channel, three shelving filters and six parametric filters.

FIR Shelving EQ

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 which 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. Being a linear phase FIR equaliser, this necessarily introduces some latency delay, which is constant regardless of the settings. However, when the ‘Enable’ parameter is set to “Off”, it is removed from the signal path entirely, so it does not add any latency. In this page you can change the frequency parameter from 2kHz to 20kHz, enable/disable the filter, and change the cut or boost in 0.2dB increments. The filter (and its associated latency) can be completely removed by setting the enable parameter to the “Off” position. Note that this EQ can only be used in Module Groups if set to ‘On’.

See also Latency delay (page 33).

Parametric filters

Parametric filters are defined by frequency, bandwidth and gain. The frequency is adjustable over the range from 10Hz to 25.6kHz. The bandwidth shown as Width on the screen, ranges from 0.10 octaves to 5.2 octaves. Bandwidth can be shown and adjusted as Q or Octaves (Oct). Gain is adjusted in 0.2dB increments.

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 pairs of inputs “1+2”, “3+4”, “1+3”, “1+4”, “2+3” or “2+4”. Summed inputs have 6dB of attenuation so that a sum of largely similar programme material remains at the correct calibrated level.

Output menu

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**. The channel number appears top left to show which channel you're editing. To select another channel, press **OUTPUT** again.
2. **GAIN** usually appears top left. If not, press down arrow ▼ several times until it does.
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose the gain (–40 to 20 dB in 0.2 increments).
4. Rotate **SELECT** (the left-hand dial) to select **Pol** for polarity.
5. Rotate **ADJUST** (the right-hand dial) to select **Norm** for normal or **Rev** for reverse polarity.
6. To edit the next channel, press **OUTPUT**.
7. To exit the menu, press **UTILITY** or press **OUTPUT** several times to step through all the output channels.

Delay

The delay page controls the amount of delay associated with the output channel selected and is adjustable from 0 to 998ms. The delay parameter is adjustable in fine steps at low values; the adjustment becomes progressively coarser as the value increases.

High and low pass filters

High pass and low pass crossover filtering is provided for the output signal. Filter type is selectable from 1st order, Butterworth, Bessel, Linkwitz-Riley, Hardman and LIR. Filter slopes of up to 8th order or 48dB / octave are provided. Not all filter types are available in all slopes. For example 18dB / octave Linkwitz-Riley filters cannot be selected because they do not exist.

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

LIR crossover filtering

Unique to Martin Audio, Linear Impulse Response (LIR) crossover filtering gives a Linear Phase crossover that has a constant delay regardless of frequency (unlike other types of crossover which 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 has 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.

For further details, see Latency delay (page 33).

Parametric Equalisation

There are ten different EQ filters; two shelving filters and eight parametric filters. Parametric filters are defined by frequency, bandwidth and gain. The frequency is controlled over the range from 10Hz to 25.6 kHz. The bandwidth, shown as Width on the screen, ranges from 0.10 octaves to 5.2 octaves. Bandwidth can be shown and adjusted as Quality Factor or (Q), or Octaves (Oct). Gain is adjustable in 0.2dB increments.

For further details, see Parametric EQ bandwidth (page 27).

Limiters

The iKON amplifiers include three limiters in the output signal path. Please note that whilst 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.

VX Limiter

This 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 threshold during the attack phase of the main limiter by more than a predetermined amount. The optimal Overshoot setting is usually about 8dB. Lower Overshoot settings will sound progressively 'harder'.

When VX mode is engaged, 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 2-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. In non-VX mode, this delay will depend on the lowest high-pass frequency used in the crossover filters in a given Drive Module. In VX mode, 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 33).

Output thermal limiter

The Thermal Limiter is designed to protect the driver against damage due to over-heating. It 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.

Three parameters are available for adjustment:

- Threshold – the continuous RMS voltage which the driver should be able to withstand. This is calibrated at the output of the amplifier. The Thermal Limiter can be defeated by setting the Threshold to the maximum "Off" value.
- 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**. The channel number appears top left to show which channel you're editing. To select another channel, press **OUTPUT** again.
2. Press down arrow ▼ several times (usually five times) until **Tmax** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to the left or right to change **Thresh** the threshold value (1 to 201V or **Off**).
4. Rotate **SELECT** (the left-hand dial) to the right to select **Attack**.
5. Rotate **ADJUST** (the right-hand dial) to the left or right to change the attack value (0.01 to 34.4 s).
6. Rotate **SELECT** (the left-hand dial) to the right to select **Rel** for release.
7. Rotate **ADJUST** (the right-hand dial) to the left or right to change the release value (1.0X to 16.4X).
8. To edit the next channel, press **OUTPUT**.
9. To exit the menu, press **UTILITY** or press **OUTPUT** several times to step through all the output channels.

Xmax Excursion Limiter

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 X-max specification of the driver.

To set the limiter up, it is necessary to know the shape of the family of Excursion vs. Frequency curves of the driver for various drive voltage levels. A curve should then be chosen where the slope is high where it passes though the specified X-Max 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 the maximum “Off” value.
- 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.

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. For this channel, the mute button doesn't operate and you can't update any of the output channel parameters. However, the Amplifier protection indicator does still operate.

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 on limiter settings and 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**. The channel number appears top left to show which channel you're editing. To select another channel, press **OUTPUT** again.

   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 button ▼ several times (usually seven presses) so that **AMP** appears top left. Note that 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.
4. To exit the menu, press **UTILITY** or press **OUTPUT** several times to step through all the output channels.

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 manually. For further details, see Drive modules (page 5).

To update routing

1. Press **OUTPUT**. The channel number appears top left to show which channel you're editing. To select another channel, press **OUTPUT** again.
2. Press down arrow ▼ several times (usually three times) until **ROUT** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose None, DSP A, DSP B, DSP C or DSP D.
4. To edit the next channel, press **OUTPUT**.
5. To exit the menu, press **UTILITY** or press **OUTPUT** several times to step through all the output channels.

Drive load

Use **Drive load** to adjust the optimal drive level for a given driver impedance. The screen also shows the live measured impedance value. If there's insufficient signal level to measure the impedance, the value is shown as **"?"**.

To update the drive load

1. Press **OUTPUT**. The channel number appears top left to show which channel you're editing. To select another channel, press **OUTPUT** again.

2. Press down arrow ▼ several times (usually two times) until **DRIV** appears top left.
3. Rotate **ADJUST** (the right-hand dial) to the left or right to choose the load. The possible values are: 25V Line, 70V Line, 100V Line, 2 ohm, 3 ohm, 4 ohm, 6 ohm, 8 ohm or Auto.
4. To edit the next channel, press **OUTPUT**.
5. To exit the menu, press **UTILITY** or press **OUTPUT** several times to step through all the output channels.

Utility menu

From the utility menu you can choose:

- **STER** – see Stereo linking (page 26)
- **DANT** – see Dante name (page 26)
- **VER** – see Firmware version (page 26)
- **ECO** – see Power saving (page 26)
- **PWR** – see External breaker protection (page 27)
- **ALM** – see Alarm (page 27)
- **Aux** – see AUX style (page 27)
- **ParaEQ** – see Parametric EQ bandwidth (page 27)
- **Screen** – see Screen contrast (page 27)
- **RECL** – see Recall snapshot (page 28)
- **STOR** – see Store snapshot (page 28)
- **IP Static** – see Static IP address (page 28)
- **IP Mode** – see IP mode (page 28)
- **IP Curr** – see Current IP address (page 29)

Stereo linking

You can stereo link inputs A and B or inputs C and D or both. When you edit a linked input, you 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 but not in Presets.

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 the left or right to highlight the text below **Link A** or **Link C**.
3. Rotate **ADJUST** (the right-hand dial) to the left or right.

   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. 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 31).

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

This displays the Firmware and Dante version numbers. To update the Firmware, use Vu-Net (page 31). 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 power saving mode, sleep mode and manual standby.

> With the current firmware (version 1.654), keep power saving and sleep switched off and avoid using manual standby.

To switch off power saving

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 the left or right to highlight the text below **Save**.
4. Rotate **ADJUST** (the right-hand dial) to the right until **Man** appears. This switches off power saving by switching it to manual.
5. Rotate **SELECT** (the left-hand dial) to the left or right to highlight the text below **Sleep**.
6. Rotate **ADJUST** (the right-hand dial) to the right until **Man** appears. This switches off sleep by switching it to manual.
7. Avoid rotating **SELECT** (the left-hand dial) to highlight the text below **S'dby** as this is where you select standby mode.
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 the left or right to select the amp value (between 9 and 50 A). The screen also displays the draw (you can't edit this).
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 style connector on the back panel. Pins 1, 2 and 3 of this connector are for the AUX Port.

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 select **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 32).

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 the left or right to select the AUX style. For details, see AUX port (page 32).
4. Press **UTILITY** to exit the menu.

Parametric EQ bandwidth

This allows you to view or adjust the bandwidth of parametric equalisers in either octaves or Q. You view and edit the bandwidth of parametric equalisers in the input (page 19) and output (page 22) menus.

To set the parameter EQ bandwidth

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

   To select **Q**, rotate **ADJUST** (the right-hand dial) to the left or right 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 28).

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 the left or right to select the snapshot number (between 1 and 20).
4. Press **ENTER**. The message **Recall snapshot?** appears.
5. To confirm, press **ENTER**.

   To cancel, press the down arrow button ▼.
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 28).

> You need to store presets before you can store snapshots.

For further details, see Snapshots (page 17).

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 the left or right 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**.

   To confirm, press **ENTER** again.

   To cancel, press down ▼ arrow.
7. 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 28). 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 28). 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 the left or right to select the part of the address to edit.
5. Rotate **SELECT** (the right-hand dial) to the left or right 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 select **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 select 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 28).
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 28).

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 iK41 can be completely automatic, with no setup required. To select auto IP, see IP mode (page 28).

When first installing and launching Vu-Net (page 31), 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 28). To select the IP address, see Static IP address (page 28).

If the amplifier or computer has a static IP address and a different IP address range (i.e. a different subnet), Vu-Net (page 31) may not be able to 'see' the device.

IP Troubleshooting

If Vu-Net (page 31) 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

If you connect iKON amplifiers and a PC via an Ethernet network, you can control and monitor the amplifiers from your PC. You do this using Martin Audio Vu-Net softare running on the PC. You can also use Vu-Net to control and monitor selected Martin Audio speaker systems.

Vu-Net is a Windows app available as a free download from the Martin Audio website.

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

To download Vu-Net

1. Go to the Martin Audio website and select **Support** > **Software**.
2. Scroll to the **Vu-Net** section and click **Download**.

Using Vu-Net

For full details of how to use Vu-Net, see the **Vu-Net User Guide**. To download this, go to the website location described in the previous section.

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 17), mutes or sleeps, as shown in the table above. The AUX port consists of the first three pins of the six-pin Phoenix style connector (or Euroblock) on the back panel (page 12). 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 27).

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 uses 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 31) 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 HiShelf 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.

To see the Drive Module latency

1. Press **INPUT**. The channel letter appears top left to show which channel you're editing. To edit another channel, press **INPUT** again.
2. Press the down button ▼ seven times so that **Latency** appears at the top of the screen.

> The latency shown doesn't include the input and output latencies.

3. Press **INPUT** again to see the latency for the next channel.
4. To exit the menu, press **UTILITY** or press **INPUT** several times to step through all the input channels.

Example latency calculation

Analogue input	0.385 ms
Analogue output	0.402 ms
Input HiShelf 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		
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

Three connections are available via the Phoenix connector for this relay. The schematic representation printed on the rear panel shows the ‘idle’ state of the relay (when the amplifier is not switched on). When the amplifier is switched on, the relay will normally energise. When a Fault incident occurs, the relay will be de-energised. This will allow either a Fault Incident or a loss of power to be 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>

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>

Parameter values

Specify parameter values using the units shown here:

Suffix	Parameter	Example
Hz	Frequency	650.3Hz
Oct	Bandwidth	0.32Oct

Suffix	Parameter	Example
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	BadCommandIncomplete
05	BadCommandUnexpectedLength
06	BadCommand
07	BadCommandParamValue
08, 12	UnknownMethod
09	UnknownMethodType
10	BadCommandExpectedYesNo
11	BadCommandFloatFormat
Others	UnknownError

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 Imedance1 Attention
–3	Driver Imedance2 Attention
–4	Driver Imedance3 Attention
–5	Driver Imedance4 Attention
–6	Driver Imedance5 Attention
–7	Driver Imedance6 Attention
–8	Driver Imedance7 Attention
–9	Driver Imedance8 Attention
–10	Slow Fan Attention
–11	PSU Power Attention
–12	Temperature Attention
–13	Starting Attention

Tipi examples

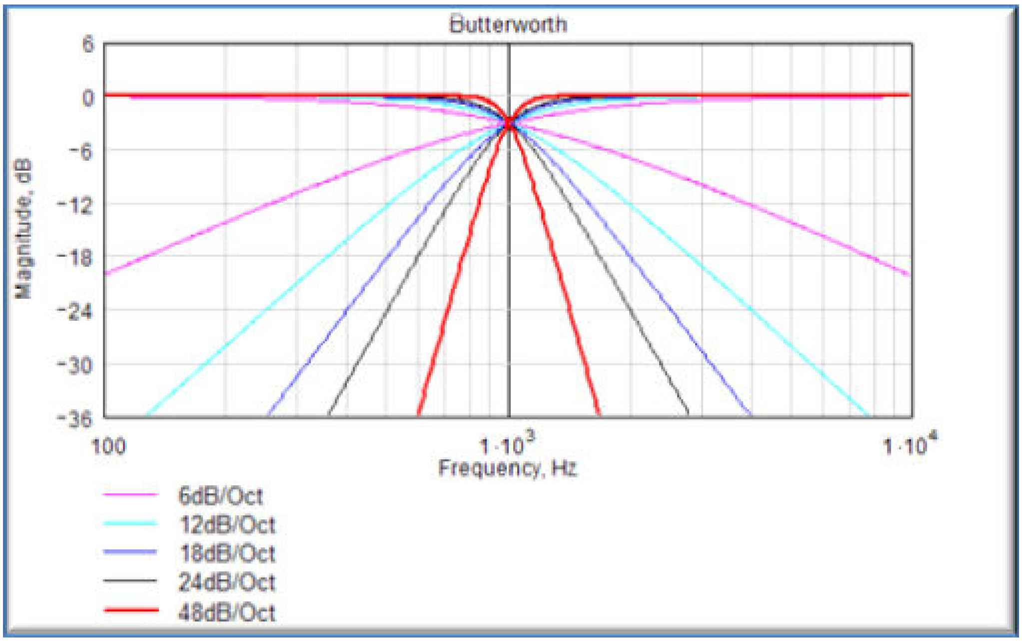
SET examples

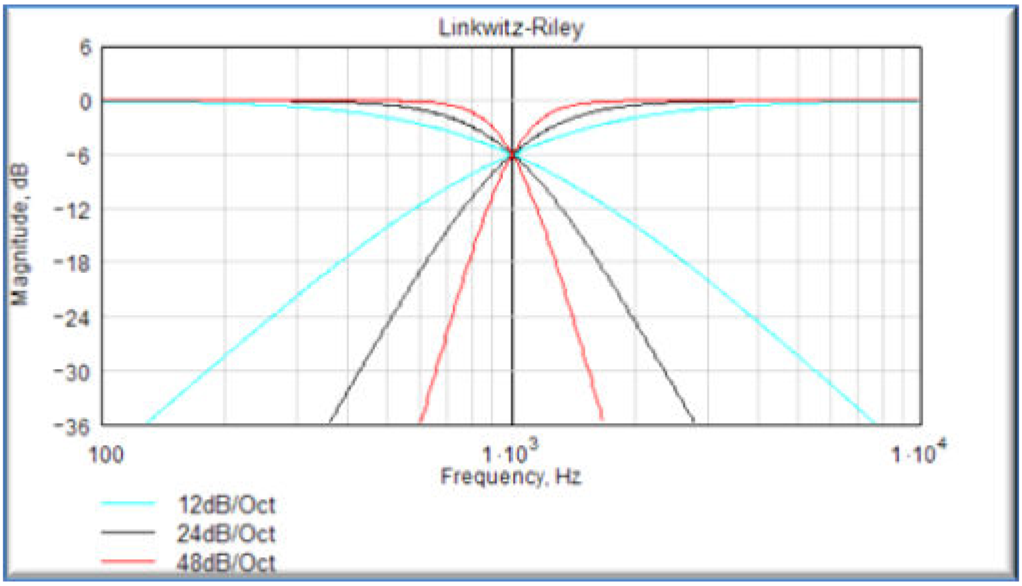
Parameter	Example	Notes
Input mute (yes or no)	$SET InB/Mute yes	Mute input B
Input gain (–40dB to 20dB)	$SET InA/Gain -3.2dB	Set the gain of input A to –3.2dB
Input polarity (yes or no)	$SET InC/Pol yes	Reverse polarity for input C
Output mute (yes or no)	$SET Out3/Mute yes	Mute output 3
Output gain (–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

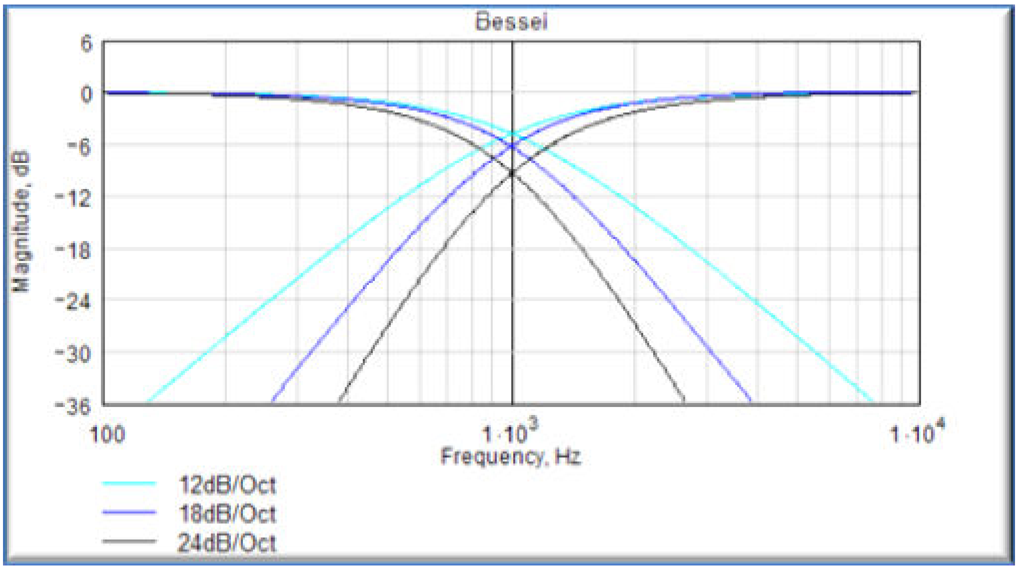
GET examples

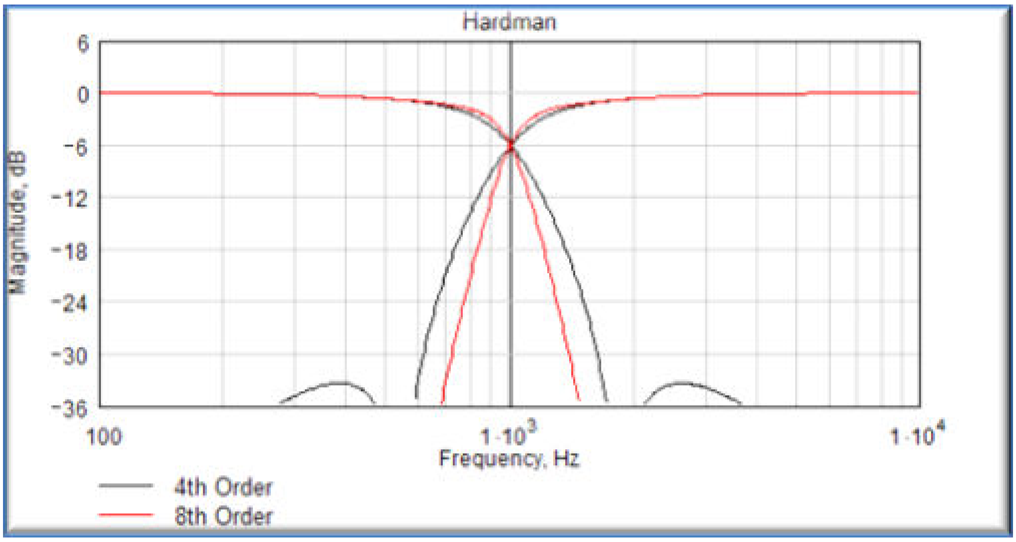
Parameter	Example	Example response
Get input meter	$GET InD/Meter	$NOTIFY InD/Meter 1.2dB
Get output meter	$GET Out3/Meter	$NOTIFY Out3/Meter 2.3dB
Input mute	$GET InB/Mute	$NOTIFY InB/Mute yes
Input gain	$GET InA/Gain	$NOTIFY InA/Gain –3.2dB
Input polarity	$GET InC/Pol	$NOTIFY InC/Pol yes
Output mute)	$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

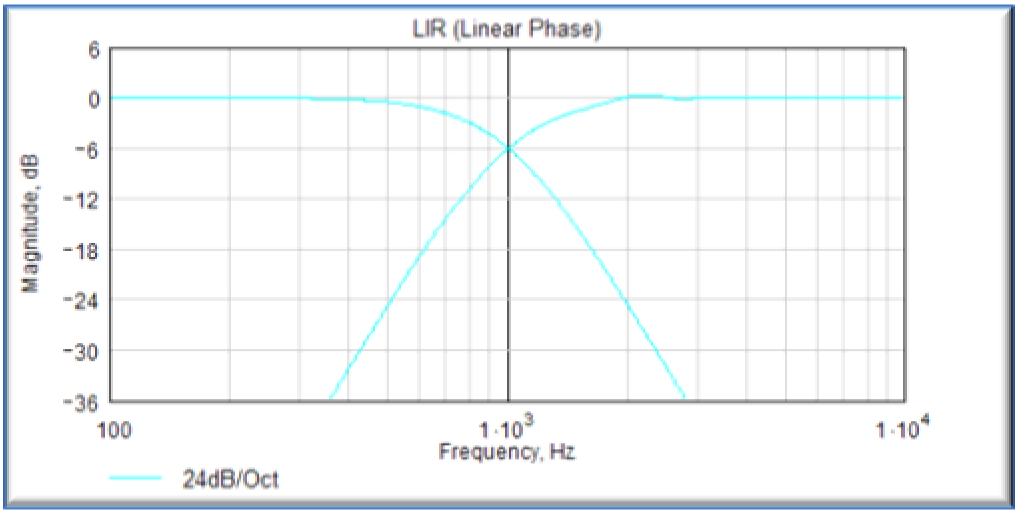
EQ and filter response graphs

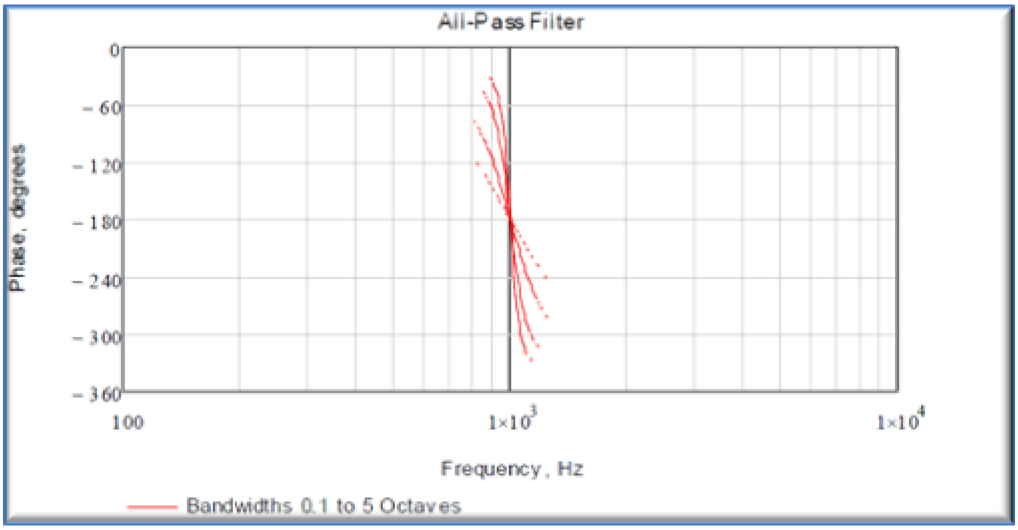


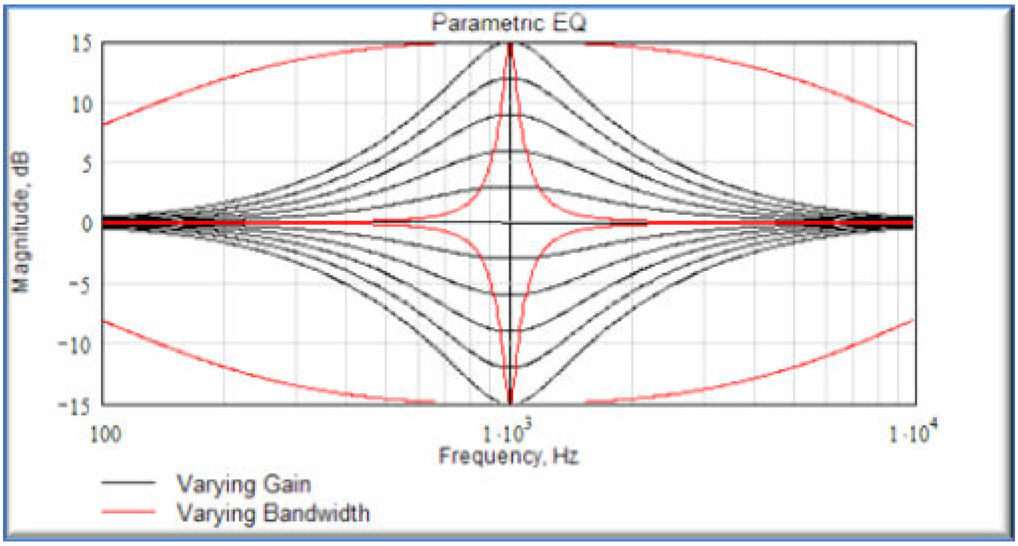



Bessel
Magnitude, dB
6
0
-6
-12
-18
-24
-30
-36
100
$1 \cdot 10^3$
$1 \cdot 10^4$
Frequency, Hz
12dB/Oct
18dB/Oct
24dB/Oct


Hardman
Magnitude, dB
6
0
-6
-12
-18
-24
-30
-36
100
$1 \cdot 10^3$
$1 \cdot 10^4$
Frequency, Hz
4th Order
8th Order


LIR (Linear Phase)
Magnitude, dB
6
0
-6
-12
-18
-24
-30
-36
100
$1 \cdot 10^3$
$1 \cdot 10^4$
Frequency, Hz
24dB/Oct


All-Pass Filter
Phase, degrees
0
− 60
− 120
− 180
− 240
− 300
− 360
100
1×10³
1×10⁴
Frequency , Hz
Bandwidths 0.1 to 5 Octaves


Parametric EQ
Magnitude, dB
15
10
5
0
-5
-10
-15
100
1·10³
1·10⁴
Frequency, Hz
Varying Gain
Varying Bandwidth

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
132	2.0	1.0	132	114	450

Running (all channels driven)								
Load mode	Load (ohms)	Signal duty and Crest Factor	Input power (watts)	Input current (amps)		Thermal dissipation		
				120Vac	230Vac	watts	kcal/hr	Btu/hr
2 ohm	2	1/8, cf = 4.0 (12dB)	1022	12.8	6.7	272	234	928
2 ohm	4	1/4, cf = 2.8 (9dB)	991	12.5	6.5	241	207	822
2 ohm	4	1/8, cf = 4.0 (12dB)	563	7.9	4.1	188	162	642
4 ohm	4	1/4, cf = 2.8 (9dB)	1780	21.1	11.0	280	241	955
4 ohm	4	1/8, cf = 4.0 (12dB)	970	11.5	6.0	220	189	751
4 ohm	8	1/4, cf = 2.8 (9dB)	963	11.5	6.0	213	183	727
4 ohm	8	1/8, cf = 4.0 (12dB)	552	7.3	3.8	177	152	604
8 ohm	8	1/4, cf = 2.8 (9dB)	914	10.8	5.7	164	141	558
8 ohm	8	1/8, cf = 4.0 (12dB)	536	7.1	3.7	161	138	549
100V	–	1/8, cf = 4.0 (12dB)	758	9.0	4.7	177	152	604
70V	–	1/8, cf = 4.0 (12dB)	970	11.5	6.0	220	189	751
25V	–	1/8, cf = 4.0 (12dB)	538	7.1	3.7	196	168	669

- 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
- 100V mode is limited to 77.5Vrms

Technical specifications

General

Number of Output channels	Four
Total power output	6,000 watts RMS, depending on load impedance
Input types	Analogue, AES3, Dante
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 (105°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 dBr at 1 kHz and –75 dBr 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 LMD (Linea-Micro-Detail) algorithms
Sample rate	96 kHz throughout
Special functionality:	Class leading limiter suite (see Protection systems (page 34))
	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)
	1,500 W into 2 ohm
	1,500 W into 4 ohm
	750 W into 8 ohm
	325 W into 16 ohm
	3,000 W bridged per channel pair, 4 ohm
	3,000 W bridged per channel pair, 8 ohm
CV line output	685 W, 25 V line
	1,500 W, 70 V line
	1,163 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

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.

System protection	Speaker protection
Excessive output current	Audio soft-clip limiter
Excessive power supply current	VxLim, Multiband peak limiter
Excessive amplifier section temperature	VxMax, Multiband overshoot limiter
Excessive power supply section temperature	Vx-Xmax, Driver excursion limiter
Excessive DSP section temperature	Vx-Tmax, Driver thermal limiter (long term power limiter)
Mains voltage out of range	DC offset protection
Fan speeds out of range	Excessive HF energy (VHF) limiter
Internal power rails out of range	

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

An inbuilt alarm and notification system can be configured to indicate problems to remote devices either via the network or the volt-free changeover relay contacts accessibly on the rear panel.

Physical	
Cooling	Variable speed fans
Airflow	Front to back
Air filtration	Washable media, changeable without the use of tools
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	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 (rear support for iKON amplifiers installed in 19" racks)	Part number RACKKITC

Loudspeaker compatibility

The following table shows the compatability of iK41 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	✓	✓	–1.5dB	N/A	N/A
	CDD15	N/A	–0.2dB	–0.2dB	x	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	✓	✓	x	N/A	N/A
	FP15	N/A	–1.2dB	–1.2dB	x	N/A	N/A
LE monitors	LE100	N/A	✓	✓	✓	N/A	N/A
	LE200	N/A	–0.2dB	–0.2dB	x	N/A	N/A
O-LINE	O-LINE	✓	✓	✓	✓	N/A	N/A
SX subwoofers	SX110	N/A	✓	✓	✓	x	x
	SX210	N/A	✓	✓	✓	x	x
	SX112	N/A	✓	✓	✓	x	x
	SX212	N/A	N/A	–0.3dB	x	N/A	x
	SX115	N/A	x	x	x	✓	–0.3dB
	SXC(F)115	N/A	x	x	x	✓	–1.2dB
	SX215	N/A	N/A	x	x	N/A	–0.3dB
	SX118	N/A	x	x	x	✓	–1.2dB
	SXC(F)118	N/A	x	x	x	✓	–1.2dB
	SX218	N/A	x	x	x	✓	–1.2dB
	SXH(F)218	N/A	x	x	x	N/A	x
TORUS	T8	N/A	✓	✓	✓	N/A	N/A
	T12	N/A	–0.2dB	–0.2dB	x	N/A	N/A
Wavefront Precision	WPM	✓	✓	✓	✓	N/A	N/A
	WPS	N/A	✓	✓	x	N/A	N/A
	WPC	N/A	x	x	x	N/A	N/A
	WPL	N/A	x	x	x	N/A	N/A
XE monitors	XE300	N/A	✓	✓	✓	N/A	N/A
	XE500	N/A	x	x	x	N/A	N/A

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