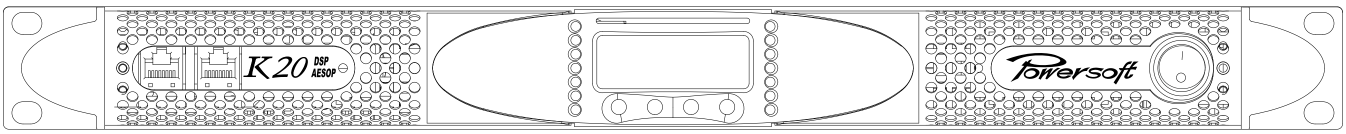
K Series amplifiers

K2 / K2 DSP + AESOP
K3 / K3 DSP + AESOP
K6 / K6 DSP + AESOP
K8 / K8 DSP + AESOP
K10 / K10 DSP + AESOP
K20 / K20 DSP + AESOP



User Guide v 2.3

November 2012

DO000044 REV 02
Powersoft S.r.l. • Via Enrico Conti, 5 • 50018 Scandicci (FI) • Italy
+39 055 735 0230 • sales@powersoft.it • www.powersoft-audio.com

Intentionally left blank

K Series

User Guide

Powersoft S.r.l. • Via Enrico Conti, 5 • 50018 Scandicci (FI) • Italy
+39 055 735 0230 • sales@powersoft.it • www.powersoft-audio.com

1 Warnings

1.1 Important Safety Instructions

CAUTION: IN ORDER TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT ATTEMPT TO OPEN ANY PART OF THE UNIT. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

"WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS APPARATUS TO RAIN OR MOISTURE. OBJECTS FILLED WITH LIQUIDS, SUCH AS VASES, SHOULD NOT BE PLACED ON THIS APPARATUS"

"TO COMPLETELY DISCONNECT THIS APPARATUS FROM THE AC MAINS, DISCONNECT THE POWER SUPPLY CORD PLUG FROM THE AC RECEPTACLE"

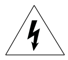
"THE MAINS PLUG OF THE POWER SUPPLY CORD MUST REMAIN READILY ACCESSIBLE"

SAFEGUARDS: Electrical energy can perform many useful functions. This unit has been engineered and manufactured to assure your personal safety. Improper use can result in potential electrical shock or fire hazards. In order not to defeat the safeguards, observe the following instructions for its installation, use and servicing.

- Read these instructions.
- Keep these instructions.
- Heed all warnings.
- Follow all instructions.
- Do not use this amplifier near water.
- Clean only with a dry cloth.
- Do not block any ventilation openings.
- Install in accordance with the manufacturer's instructions.
- Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
- Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet (for K2 and K3 only. K6, K8, K10 and K20 come with a special mains cable without plugs).
- Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
- Only use attachments/accessories specified by the manufacturer.
- Unplug this amplifier during lightning storms or when unused for long periods of time. Refer all servicing to qualified service personnel. Servicing is required when the amplifier has been damaged in any way. For example if the power-supply cord or plug have been damaged, if liquid has been spilled or objects have fallen into the amplifier, if the amplifier has been exposed to rain or moisture, if it has been dropped or if it does not operate normally.

CAUTION: To prevent fire hazard, Class 2 (for K2 and K3) and Class 3 (for K6, K8, K10 and K20) wiring cable should be used for connection with speakers. Cabling should be routed away from potential hazards to avoid damage to the insulation of the cable itself.

EXPLANATIONS OF GRAPHICAL SYMBOLS:



"The Lightning Flash with arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product enclosure that may be of sufficient magnitude to constitute a risk of shock to persons".

"The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product".

1.2 Approvals

The K series is installed according to the Canadian Electrical Code or National Electrical Code, as applicable.

Install this product in accordance with Canadian Electrical Code or National Electrical Code and other local electrical or building codes as applicable. Mount in rack only. The flexible mains cable must not pass through walls.

This equipment has been tested and found to compliant by Notified Body 2047 (Directive 2004/108/EC-EMC) pursuant to the product family standard for audio professional use: EN 55103-1 and EN 55103-2 standard; EN61000- 3 - 2 , EN 61000 - 3 - 3. Electromagnetic Ambients E4, E5.

This equipment has been tested and found to compliant by Notified Body 2047 (Directive 2004/108/EC-EMC) pursuant to the product family standard for audio professional use: Radiated emissions FCC standard section 15.109, IEC CISPR standard Pub. 22 ed 6.0 (2008-09) CLASS A chapter 7.1.1, Conducted emission FCC standard section 15.107, IEC CISPR standard Pub. 22 ed 6.0 (2008-09) CLASS B.

This equipment has been tested and found to comply by Notified Body 2047(Directive 2006/95/EC L.V.) pursuant to the audio apparatus safety requirements: Standard EN 60065

In a domestic environment this product may cause radio interferences in which case the user may be required to take adequate measures.

Average half-cycle r.m.s. inrush current on initial switch-on

K2, K2DSP, K3, K3DSP: 10 A

K6, K6DSP, K8, K8DSP, K10, K10DSP: 50 A

K20, K20DSP: 50 A

Average half-cycle r.m.s. inrush current after a supply interruption of 5s.

K2, K2DSP, K3, K3DSP: 10 A

K6, K6DSP, K8, K8DSP, K10, K10DSP: 10 A

K20, K20DSP: 10 A

1.3 Warning Notices

Note: This equipment has been tested and found to comply with the limits for Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Information to User

Alterations or modifications carried out without appropriate authorization may invalidate the user's right to operate the equipment.

1.3.1 Location

Install the amplifier in a well-ventilated location where it will not be exposed to high temperature or humidity. Do not install the amplifier in a location that is exposed to direct sun rays, or near hot appliances or radiators. Excessive heat can adversely affect the cabinet and internal components. Installation of the amplifier in a damp or dusty environment may result in malfunction or accident.

1.3.2 Precautions Regarding Installation

Placing and using the amplifier for long periods of time on heat generating sources will affect its performance. Avoid placing the amplifier on heat generating sources. Install this amplifier as far as possible from tuners and TV sets. An amplifier installed in close proximity of such equipment may experience noise or generic performance degradation.

WARNING: To prevent fire or electric shock:

- The ventilation openings must not be impeded by any item such as newspapers, tablecloths, curtains etc; keep a distance of at least 50 cm from the front and rear ventilation openings of the amplifier.
- Do not expose this amplifier to rain or moisture.
- This equipment must not be exposed to dripping or splashing liquids: objects filled with liquids, such as vases, must not be placed on the amplifier.

1.4 Safety Rules

- This device must be powered exclusively by earth connected mains sockets in electrical networks compliant to the IEC 364 or similar rules.
- It is absolutely necessary to verify this fundamental requirement of safety and, in case of doubt, require an accurate check by qualified personnel.
- The manufacturer cannot be held responsible for damages caused to persons, things or data due to an improper or missing ground connection.
- Before powering this amplifier, verify that the correct voltage rating is being used.
- Verify that your mains connection is capable of satisfying the power ratings of the device.
- Do not spill water or other liquids into or on the amplifier.
- Do not use this amplifier if the electrical power cord is frayed or broken.
- Do not remove the cover. Failing to do so will expose you to potentially dangerous voltage.
- No naked flame sources such as lighted candles should be placed on the amplifier.
- Provide a sectioning breaker between the mains connections and the amplifier. Suggested device is 32A/250VAC, C or D curve, 10KA (K6-K8-K10-K20) or 16A/250VAC, C or D curve, 10KA (K2-K3)
- Contact the authorized service center for ordinary and extraordinary maintenance.
- The power cord type is LAPP CABLE OLFLEX191 3G6 / SJT 3XAWG10 SALCAVI (Bahoing SJT 3x16AWG or I-sheng SGIS 3G1,5mmq for K3 - K2)

1.5 Speaker Damage

Powersoft Class D amplifiers are among the most powerful professional amplifiers available and are capable of producing much more power than many loudspeakers can handle. It is the user's responsibility to use speakers suitable to the amplifier and to use them in a sensible way that will not cause damage.

Powersoft will not be held responsible for damaged speakers. Consult the speaker manufacturer for power handling recommendations.

Even if you reduce the gain using the amplifier's front panel attenuation controls, it is still possible to reach full output power if the input signal level is high enough.

A single high-power tone can damage high frequency drivers almost instantaneously, while low frequency drivers can usually withstand very high, continuous power levels for a few seconds before they fail. Reduce power immediately if you hear any speaker "bottoming out" - harsh pops or cracking distortion that indicate that the speaker voice coil or diaphragm is striking the magnet assembly.

Powersoft recommends that you use amplifiers of this power range for more headroom (cleaner sound) rather than for increased volume.

1.6 Speaker Output Shock Hazard

A Class D amplifier is capable of producing hazardous output voltages. To avoid electrical shock, do not touch any exposed speaker wiring while the amplifier is operating.

This manual contains important information on operating your Powersoft amplifier correctly and safely. Please read it carefully before operating your amplifier. If you have any questions, contact your Powersoft dealer.

2 Front and Rear Panel Reference Figures

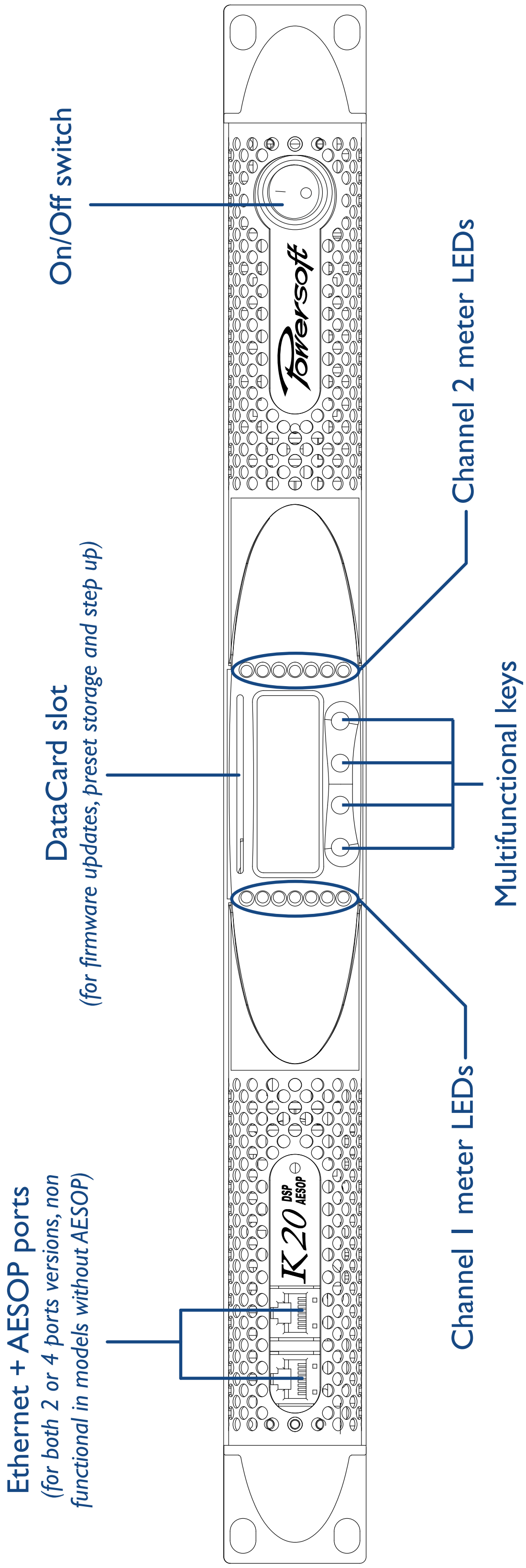


FIGURE 1: K Series front panel

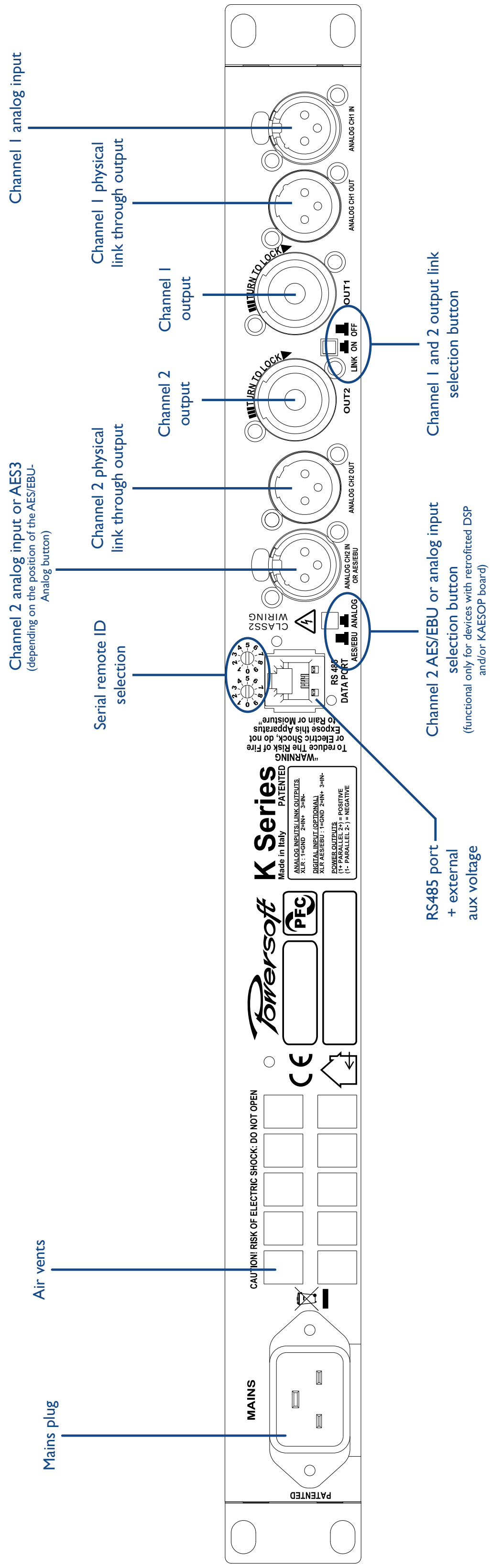


FIGURE 2: K2/K3 2-port version rear panel

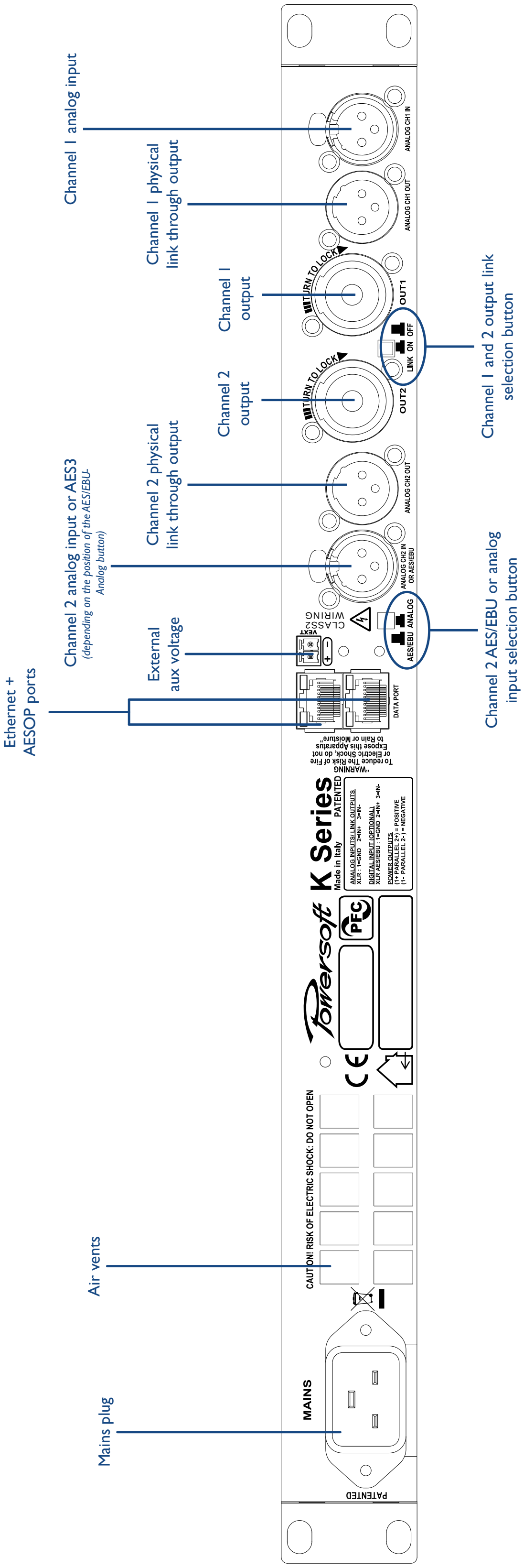


FIGURE 3: K2/K3 4-port version rear panel

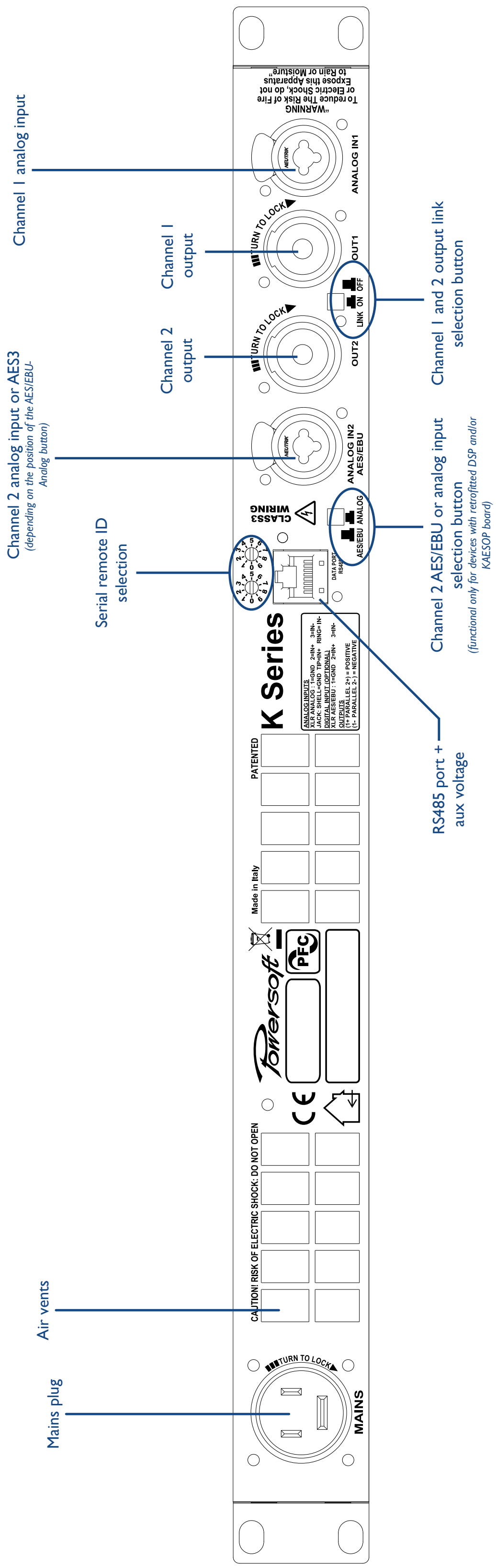


FIGURE 4: K6/K8/K10/K20 2-port version rear panel

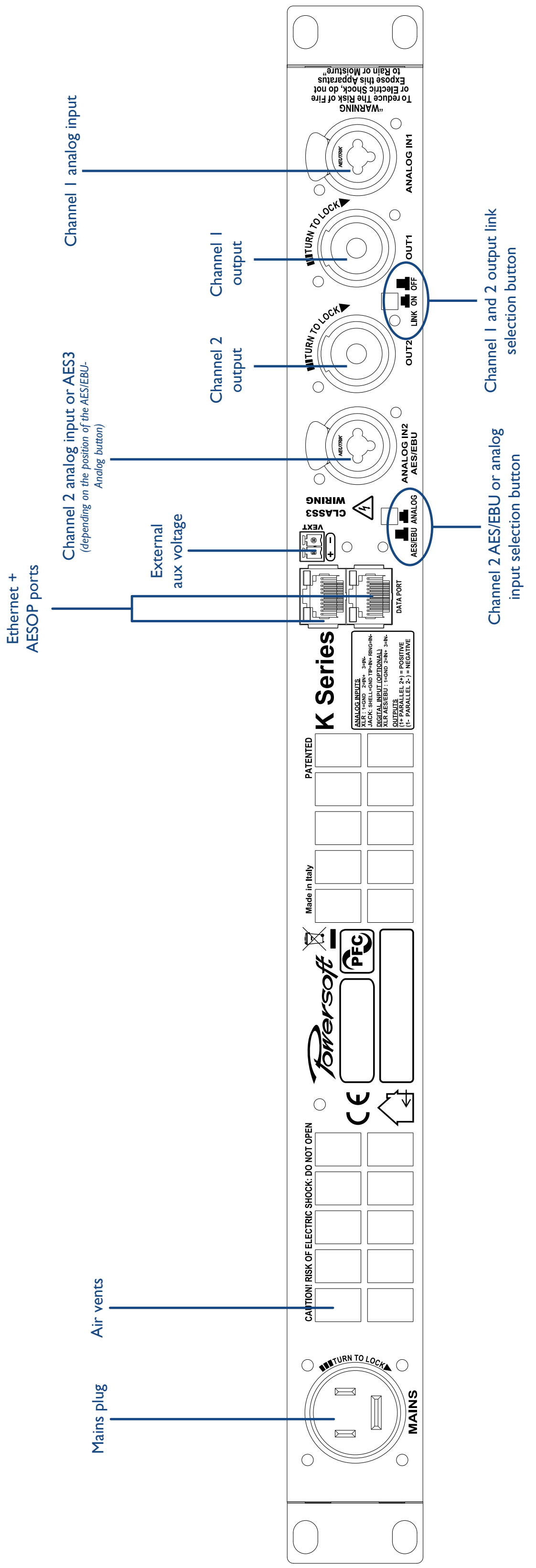
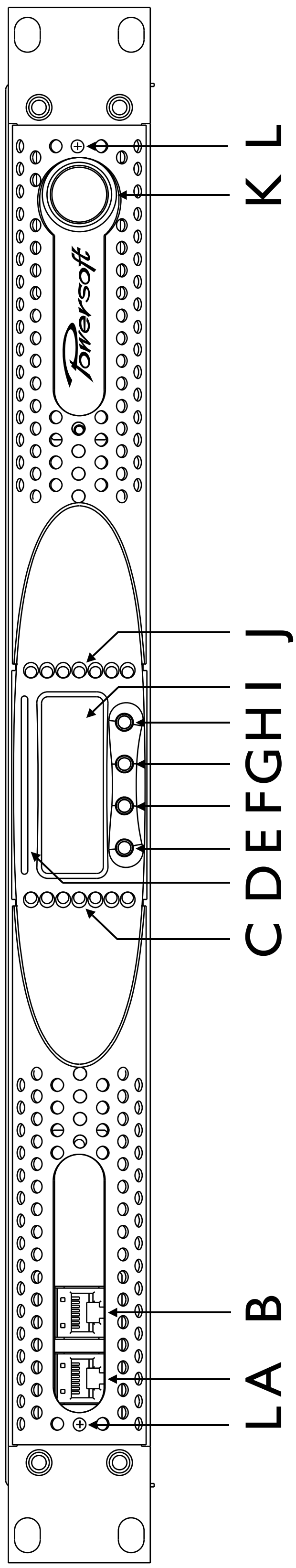


FIGURE 5: K6/K8/K10/K20 4-port version rear panel



A - Ethernet port number 1
B - Ethernet port number 2

C - V meter for channel 1
D - SmartCard slot
E - Function button number 1
F - Function button number 2
G - Function button number 3
H - Function button number 4
I - LCD display
J - V meter for channel 2

K - On/Off switch
L - Grill filter screws

FIGURE 6: K Series detailed front panel view

3 Welcome

3.1 Introduction

Congratulations on buying a Powersoft K Series amplifier! Powersoft is a leading company in the field of high efficiency audio power management. The Powersoft Class D technology has changed the way the world looks at professional audio amplification: no other amplifier's performance comes close for applications demanding high power and long term reliability. Thanks to amazing reductions in heat output and weight, without sacrificing output powers, Powersoft amplifiers can be used in an unlimited range of PA applications such as opera houses, theaters, churches, cinema, and theme parks.

3.2 The K Series

K Series has many advanced features, digital control of many parameters, adjustable maximum mains consumption, selectable digital presets and a graphic display that shows detailed information of the status of the amplifier. All K Series amplifiers come with built in Power Factor Correction. This unique feature ensures that a predominantly resistive load is presented to mains thus minimizing current distortion and voltage/current displacement. This leads to improved performance of the amplifier at high levels of output and avoids mains-voltage collapses, typical of standard and switching power supplies. Another great advantage of this technology is that its performance is, to a large extent, independent of mains voltage. The rated output power does not vary with load/line conditions.

3.3 More sound and less weight

Class D technology based amplifiers are highly efficient, delivering greater power to speakers with reduced heat dissipation: typical running efficiency of output stages is 95%, with only 5% of input energy dissipated as heat. This allows for smaller dimensions, weight and power consumptions.

Contrary to conventional amplifiers which achieve highest efficiency only at full rated power output, Class D efficiency is almost independent of output level. Music has an average power density of 40% of its peak value; this means that other (non-class D) amplifiers can easily generate 10 times more heat than Powersoft products for the same sound pressure level.

Powersoft amplifiers deliver crystal-clear highs, and a tight, well-defined low end: the most accurate reproduction of an audio signal. Solid time proven design features ensure extremely high performance in terms of super low total harmonic distortion, optimal frequency response, high power bandwidth and damping factor across a vast number of application scenarios. Powersoft's multi patented application of Pulse Width Modulation (PWM) high frequency sampling techniques is just one of the many factors contributing to the K Series' high performance ratings across the audio bandwidth.

3.4 The Show Always Goes On

The K series offers complete protection against any possible operation error. Every amplifier in this series is designed to work under a large range of possible conditions, delivering maximum power with maximum safety and an outstanding long term reliability. Anticipating potential problems at the design stage means your show always goes on!

4 Installation

4.1 Unpacking

Carefully open the shipping carton and check for any noticeable damage; the figure below (FIGURE 7) shows the packing view. Every Powersoft amplifier is completely tested and inspected before leaving the factory and should arrive in pristine condition. In the unlikely event that you should encounter any damage, please notify the shipping company immediately. Be sure to save all packing materials for the carrier's inspection.

The K Series box contains the following:

- 1 K Series amplifier
- 1 x AC Mains cord
- 1 x User Guide

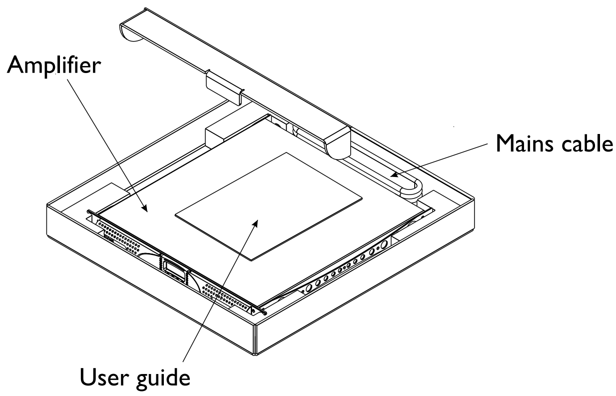


FIGURE 7: K Series packaging box

4.2 Mounting

All Powersoft amplifiers are designed for standard 19" rack mounting; there are four front panel holes and two rear-lateral holes. In order to limit the risk of mechanical damages, amplifiers must be fixed to the rack using both frontal as well as rear mounting holes.

4.3 Cooling

All Powersoft amplifiers implement a forced-air cooling system to maintain low and constant operating temperatures. Drawn by an internal fan, air enters through the slots in the front panel and is forced over all components, exiting at the back of the amplifier.

The amplifier's cooling system features an "intelligent" variable-speed DC fan which is controlled by heat sink temperature sensing circuits: the fan speed will increase only when the temperature recorded by the sensors rises over carefully predetermined values. This ensures that fan noise and internal dust accumulation are kept to a strict minimum. Should however the amplifier be subject to an extreme thermal load, the fan will force a very large volume of air through the heat sink. In the extremely rare event that the amplifier should dangerously overheat, sensing circuits shut down all channels until the amplifier cools down to a safe operating temperature. Normal operation is resumed automatically without the need for user intervention.

Caution regarding heat escape should be exercised when mounting K Series amplifiers. Exhaust cooling air is forced out through the rear of the chassis (see FIGURE 8); make sure there is enough space around the back of the amplifier for this air to escape. K Series amplifiers can be stacked one on top of the other due to the efficient cooling system they are equipped with. There is however a safety limit to be observed: in case a rack with closed back panels is used, leave one rack unit empty every four K Series amplifiers installed to guarantee adequate air flow.

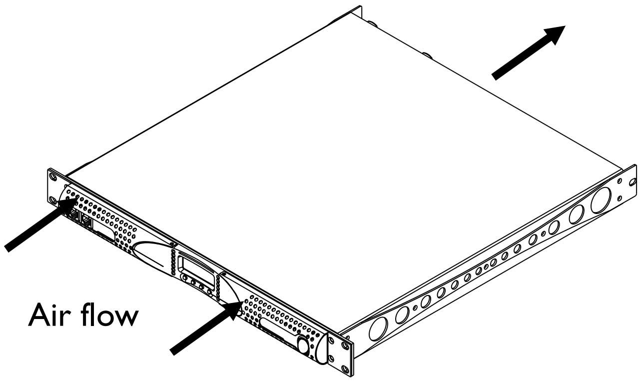


FIGURE 8: Forced air cooling: front to back airflow

4.4 Operating Precautions

Make sure the power switch is off before attempting to make any input or output connections.

Make sure the AC mains voltage used is within the acceptable operating voltage range specified in the K Series documentation (100V-240V ±10%). Damage caused by connecting the amplifier to an improper AC mains voltage is not covered by the warranty.

By using good quality input and speaker cables, the likelihood of erratic signal behavior is reduced to a minimum. Whether you make them or buy them, look for good quality wires, connectors and soldering techniques.

4.5 Grounding

There is no ground switch or terminal on the K Series amplifiers. All shield terminals of input connections are directly connected to the chassis. This means that the unit's signal grounding system is automatic. In order to limit hum and/or interference entering the signal path, use balanced input connections.

In the interests of safety, the unit MUST always operate with electrical safety earth connected to the chassis via the dedicated wire in the 3-wire cable. Never disconnect the ground pin on the AC mains power cord.

4.6 AC Mains connection

The AC Main connection is made via the CPC type connector (IEC20A for K3 and K2) on the rear side of the panel. The figure below shows how to connect the mains power cable to the amplifier. Make sure the AC mains voltage used is within the acceptable operating voltage range specified in the K Series documentation (100V-240V ±10%). It is important to connect the ground for safety, do not use adapters that disable the ground connection. All K Series amplifiers have an automatic power factor correction system for a perfect mains network interface. The amplifier is a resistive load for the mains network, minimizing the reactive power and the harmonic distortion on the current. The system allows performance to be maintained even in case of varying mains voltage.

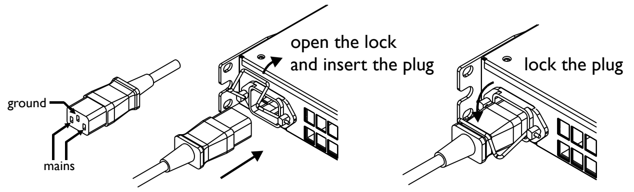


FIGURE 9: K2 and K3 only mains connection

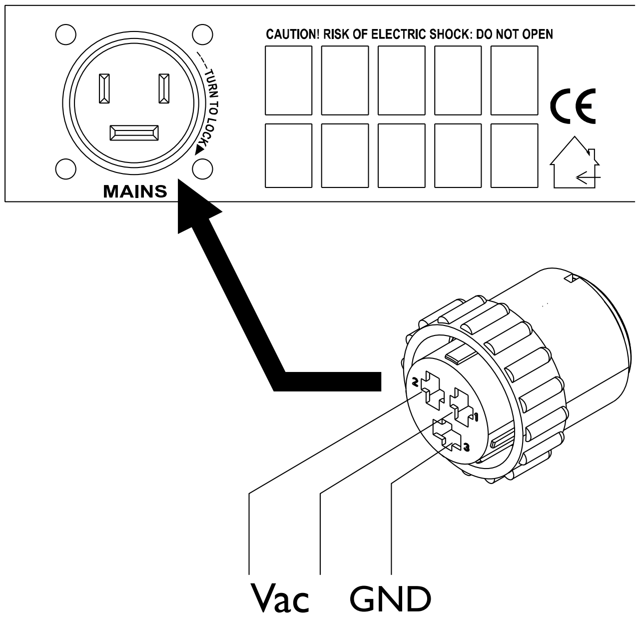


FIGURE 10: K6/K8/K10/K20 mains connection

5 Connections and Operation

This chapter provides information on amplifier connection and operation. For optimal amplifier performance, it is important to understand the meaning of the information that the K Series amplifier can provide regarding its status and configuration. This information is available to the user both via front panel indicators or via the Armonía client software when this is used. This chapter will break down all the front panel operations and monitoring functions the K Series amplifier is capable of. The remaining part of the chapter will explain how to correctly connect the amplifier's inputs and outputs.

5.1 Connecting Audio Inputs

5.1.1 Analog Connection

Input connections are made via the 3-pin XLR-female type or 1/4" phone Jack connectors on the rear side of the amplifier. The polarity is shown in the following figures:

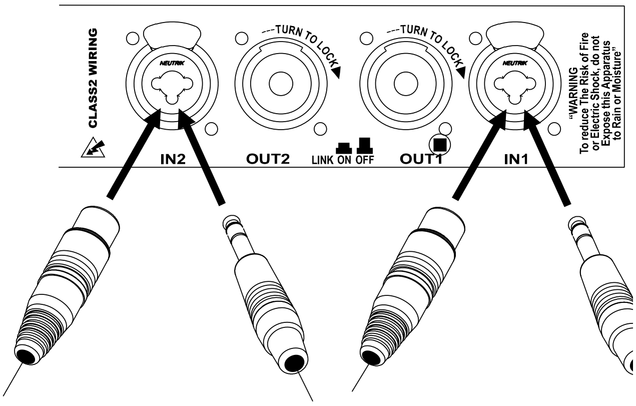


FIGURE 11: *Audio input connection for K6/K8/K10/K20 models*

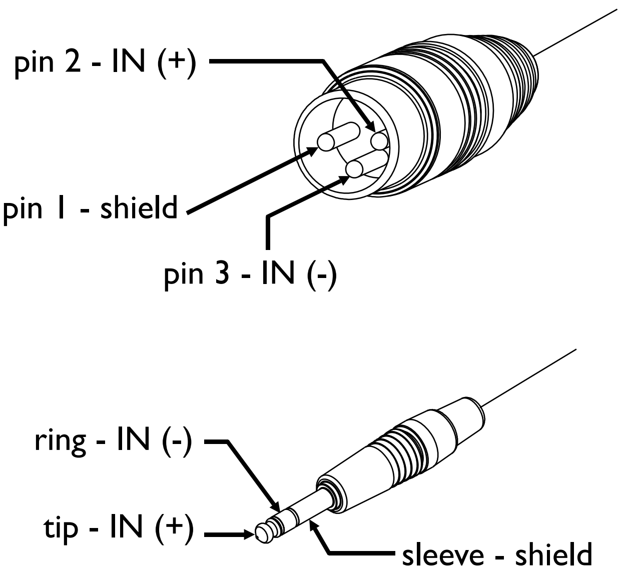


FIGURE 12: *Audio input connections polarity*

The figure below shows the connection of analog inputs for balanced or unbalanced line. You can use both configurations, but you must consider that unbalanced and long lines can introduce noise in the audio system. The "Link On/Off" switch located in the rear panel is for direct paralleling of the rear input connectors. The remaining input connectors can be used to carry signal to other amps.

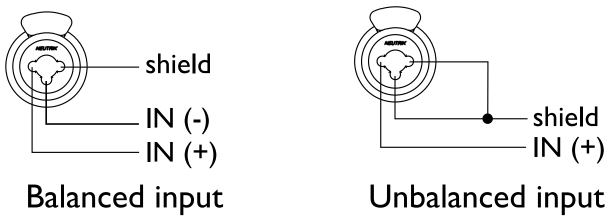


FIGURE 13: *Balanced and unbalanced input connections*

XLR pinout chart:

XLR Pin number	Assigned to
1	shield
2	hot (+)
3	cold (-)

Audio jack pin out summary:

Connector element	Assigned to
sleeve	shield
tip	hot (+)
ring	cold (-)

For K3 and K2 models, input connections are shown in the figure below; analog inputs for balanced and unbalanced lines are also available for these models.

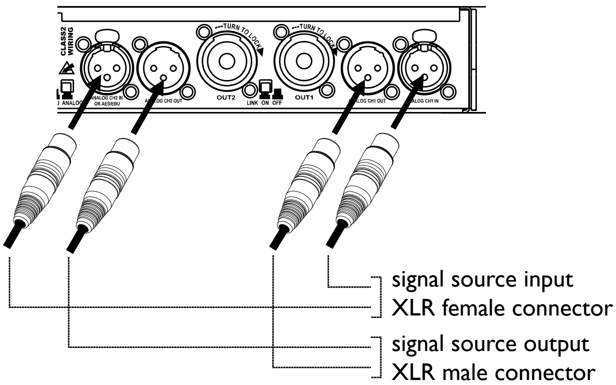


FIGURE 14: *K2 and K3 models audio input connections*

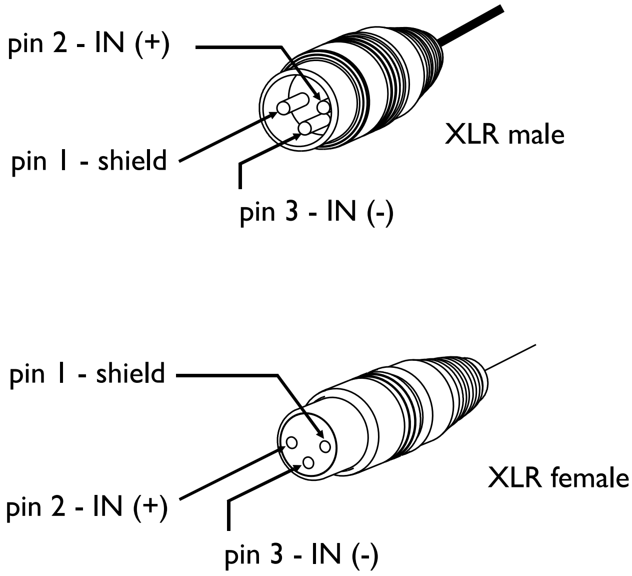


FIGURE 15: *K2 and K3 models audio input connections polarity*

5.1.2 AES/EBU Connection

On DSP equipped amplifiers, CH2 becomes the AES/EBU input when the AES/EBU pushbutton is released (see FIGURE 16); in this mode, if an analog input in CH2 is applied, the ANALOG CH2 OUT is off. If CH2 is to be used as an analog input, the AES/EBU pushbutton must be pressed.

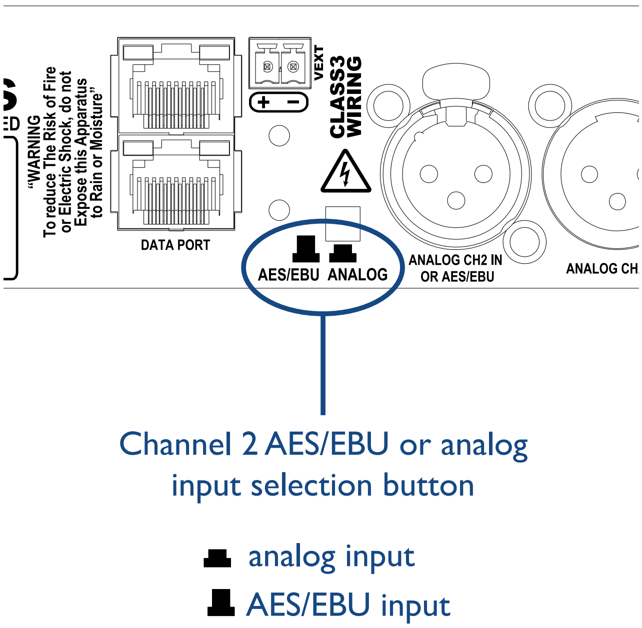


FIGURE 16: AES/EBU or analog input selection for channel 2

5.2 Connecting Audio Outputs

Audio output connections are made via Neutrik® speakon connectors.

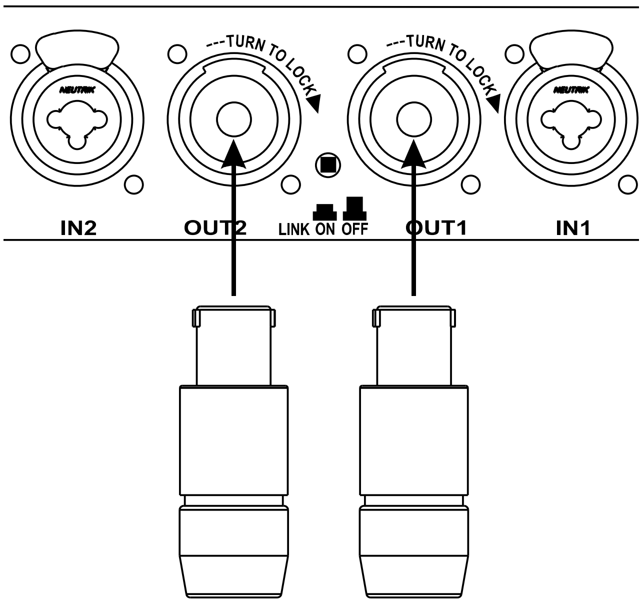


FIGURE 17: Audio output connector

Use suitable wire gauges to minimize power and damping factor losses in speaker cables. All K Series amplifier outputs can also be configured to work in bridge mode. For each device, the 1+ and 2+ pins of speakon connectors are internally physically bridged together. They are the positive pole of the channel output. Pins 1- and 2- are also bridged together. They form the negative pole of the channel output. Please note that in order to remain within safe operating conditions, when using loads of 4 Ω or less (8 Ω or less in bridge mode),connections must be made with a four wire cable. Use one cable for each SpeakOn contact for either bridge or stereo connections as shown in the following figures.

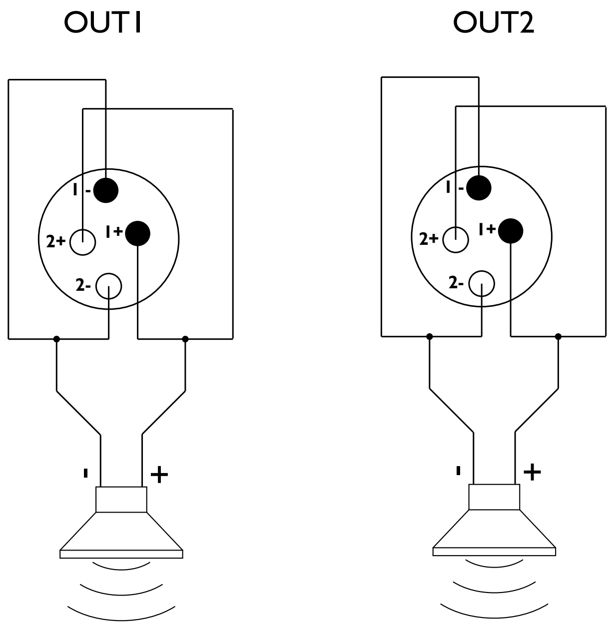


FIGURE 18: Audio output connection in stereo mode

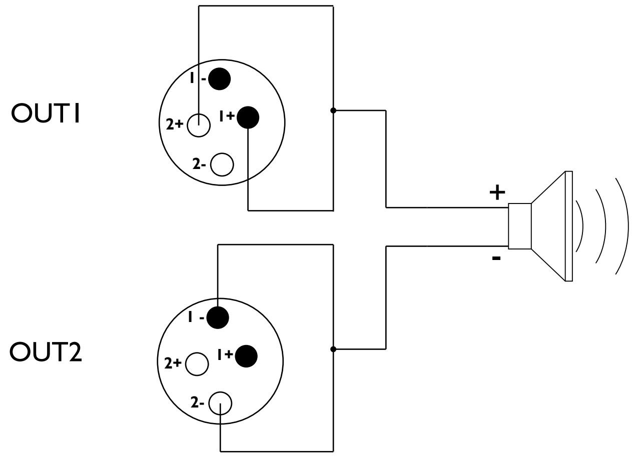


FIGURE 19: Audio output connection in bridge mode

5.3 Internal Signal Path Polarity

In order to increase the power's supply energy storage efficiency, signals coming from channels 1 and 2 are polarity reversed one with respect to the other when entering the amplifier. This ensures a symmetrical use of the voltage rails: if, for example, both channels' 1 and 2 input signals are going through a peak at the same time, channel 1's energy will come from the positive voltage rails while channel 2, whose polarity is reversed with respect to channel 1, will be fed energy from the negative voltage rails. In this manner, the power supply will work symmetrically, with one channel catered by the positive rails and the other by the symmetrical negative rails. Channel 2's signal will be polarity reversed once more to ensure that both channels output with the same polarity as their corresponding input signals. For this reason it is very important not to invert the polarity of either channels before feeding them to a K Series amplifier. A double polarity inversion (the first by the user inserting the input signal and the other by the amplifier's internal circuitry) results in no inversion at all. If this were the case, both channels would be weighing on only one side (positive or negative) of the power supply's voltage rails. This would result in an inefficient use of the power supply's energy.

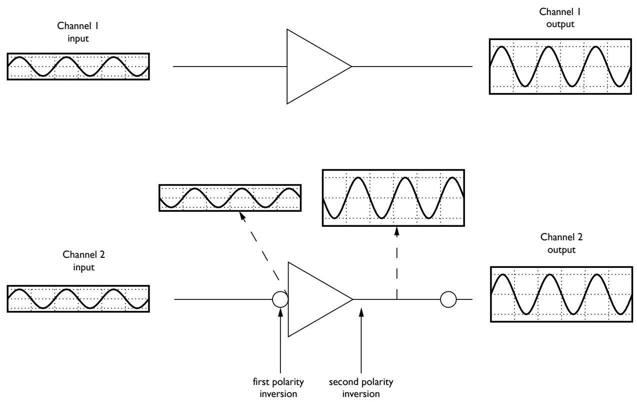


FIGURE 20: Internal signal path polarity with example input signals. Both channels 1 and 2 are fed the same sine signal

Please pay special attention in using balanced inputs on all measurement equipment (such as oscilloscope probes) when you are bench testing.

5.4 Remote Control Connection

5.4.1 V Ext

The "V Ext" terminal is used to remotely turn on, turn off or put in standy any K Series amplifier. The "V Ext" signal reaches the amplifier via pin 2 of the rear Ethernet connector for 2 port models. Four port models have a dedicated 2 pin Phoenix port located near the rear Ethernet ports. When the V ext port is powered by and external 12 VDC 1A power supply, an internal controller is enabled to listen for incoming device power-on/off/standby commands.

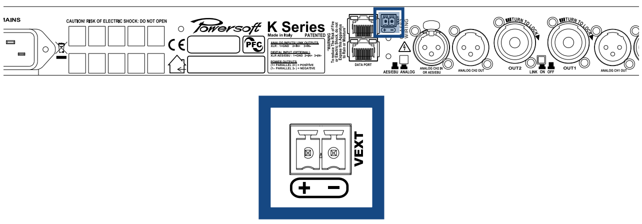


FIGURE 21: Vext Phoenix connector in 4 port K Series amplifiers

5.4.2 Serial Connection

K Series amplifiers without an optional KAESOP board can be remotely controlled via an RS485 connection. Remote connection data cables must have an 8 pin modular plug to be inserted in the RJ45 jack labelled "DATAPORT" on the rear of the amplifier. By plugging an 8 pin modular plug and selecting the unit's remote ID via the rotary trimmers, the amp is ready to be remotely controlled. Please note that ID numer 00 is not allowed.See FIGURE 22 for details.

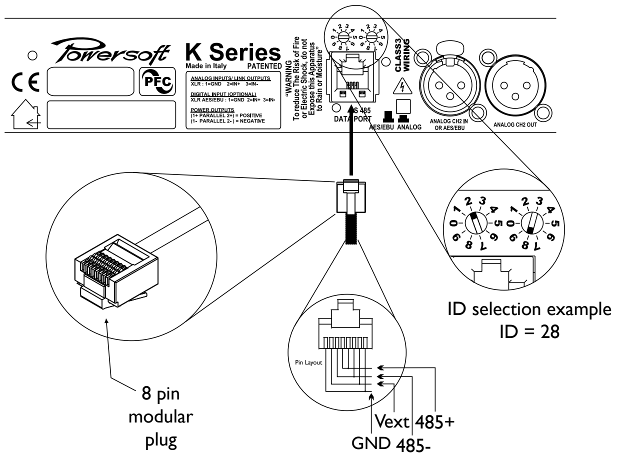


FIGURE 22: Remote connection jack, plug and ID selection

Remote connection jack pinout chart:

1	GND
2	Vext
3	485 -
4	485 +
5	485 +
6	485 -
7	Vext
8	GND

5.4.3 Ethernet Connection

K Series amplifiers can be remotely controlled via an Ethernet connection if provided with a KAESOP board. Two- or four-ports amplifiers allow Ethernet data connections with a variety of possible topographies. See "9 Network Operations" on page 29 for more details. If four plugs are present (two in the front and two in the back of the amp), the pair in the back are master ports, while the two in front are slave ports.

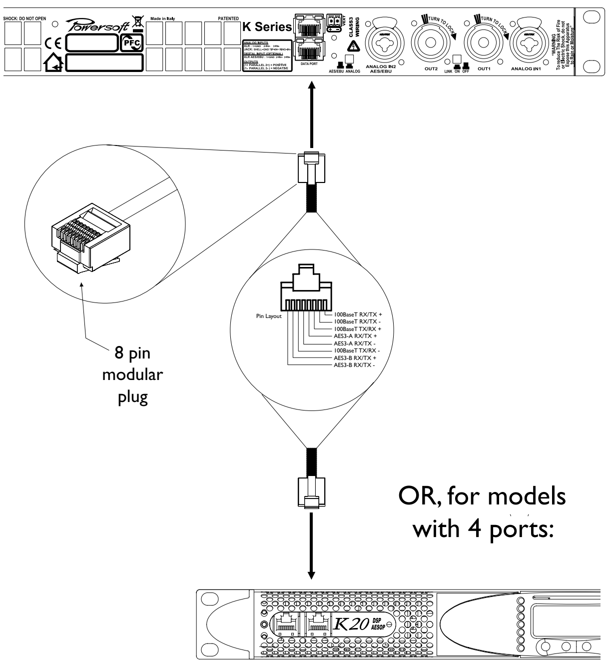


FIGURE 23: Ethernet connection ports for 2-port and 4-port amplifiers

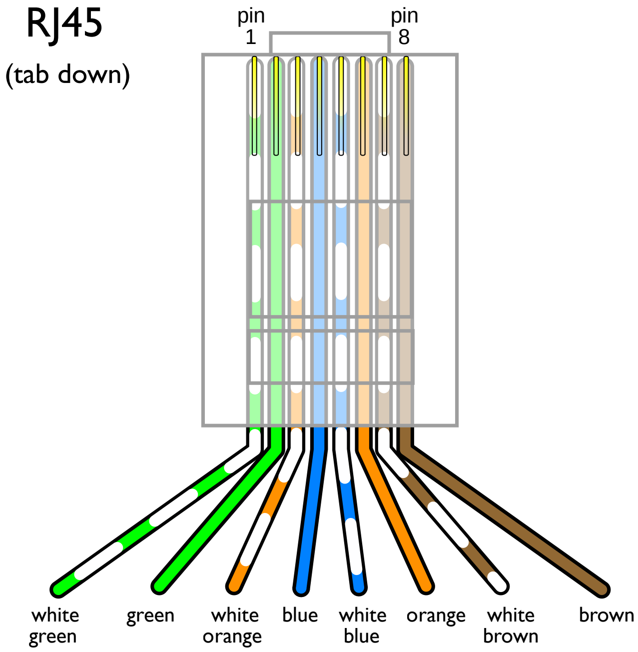


FIGURE 24: RJ45 jack pinout for KAESOP connections

The RJ45 LEDs are coded s follows:

green LED: indicates the passage of control data

yellow LED: indicates the passage of AES3 signals

Remote connection jack pinout chart:

pin	color	RJ45 KAESOP pin out
1	white/green	100BaseT AutoMDI RX/TX +
2	green	100BaseT AutoMDI RX/TX -
3	white/orange	100BaseT AutoMDI TX/RX +
4	blue	AES3-A RX/TX +
5	white/blue	AES3-A RX/TX -
6	orange	100BaseT AutoMDI TX/RX -
7	white/brown	AES3-B RX/TX +
8	brown	AES3-B RX/TX -

5.5 Amplifier Setup and Settings

5.5.1 Introduction

In all K Series amplifiers, the combination of the front panel buttons together with the LCD display allow the user access to detailed information and complete control over the amplifier's status. Each button has multiple functions and the display shows the current active function for each button. This chapter illustrates all the functions and settings accessible via the amplifier front panel. FIGURE 6 illustrates all K Series front panel elements.

Armonía Pro Audio Suite

All the setup and settings functions described in this section can be accessed through a comptuer by installing Powersoft's Armonía Pro Audio Suite software. Armonía is a software environment entirely developed in-house by Powersoft. Its two main features are full end user remote control of the amp and its signal processing capabilities. The intuitive interface provides reliable information and real time control of all DSP functions (see "18.4.1 Powersoft's Armonía Pro Audio Suite" on page 45). Refer to the Armonía manual for installation and configuration of the client software. Armonía is free. It can be downloaded after signing up for our user forum: see the "Armonía Support Forum" section at

http://www.powersoft-audio.com/

5.5.2 The main screen and the LED bars

When the amp is turned on, the main screen appears after a short presentation.

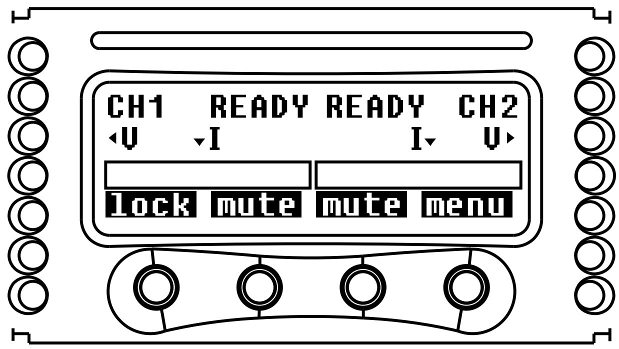


FIGURE 25: K Series main screen

The first line of the screen will read "WAIT" while the system undergoes an initial batch of internal tests to determine the status of the amp. If all parameters are normal, "READY" will replace "WAIT" on the display. System parameters are continuously monitored by the internal controller. If any parameter value should fall out of its correctly working range, a code error relative to that particular parameter will appear on the third line of the LCD meter at the corresponding channel number. Should the parameter be out of range for both adjacent channels, the error code will appear in between the two compromised channels.

The LED columns on the front of the amp can work as output voltage or current meters. When the LED bars are set to meter output voltage, for example, the meters on the LCD screen will indicate output current values. The vice versa is true: LED bars set as output current meters, LCD display bars become output voltage meters.

The single LEDs can have multiple functions:

LED	Color	Solid color	Blinking
●	Red	Channel output level has reached clipping limits OR channel has been muted for protection[1]	tone detection problem
●	Yellow	temperature of power circuits is above 85°C OR output level[2] -2dB	power circuits temperature critical (80° - 85°C)
●	Green	output level[2] -3dB	
●	Green	output level[2] -6dB	
●	Green	output level[2] -9dB	
●	Green	output level[2] -15dB	
●	Green	input signal is above -60dBV OR output level[2] -18dB	

[1]in case of a short circuit protection event, the LCD screen will read "PROT"
[2]with respect to the output clipping threshold

5.6 Front Panel Buttons

The fourth line of the front panel LCD screen shows the functions of the buttons immediately below. A beep confirms that a button has been pressed; please note that this sound is not mutable. Pressing the button directly below the "menu" label on the LCD screen gives access to the amplifier's main menu. If an Armonía client is connected to the amplifier, a yellow blinking LED will appear in the software workspace view.

6 The main menu

The K series main menu can be accessed by pressing the first button on the right, underneath the LCD label "menu". FIGURE 25 shows the new button setup adopted to allow users to navigate the amp's internal menu. The up and down arrows allow to scroll the menu items. To access further menu voices branching off a specific menu item, select it and press the "menu" button once. FIGURE 26 and FIGURE 27 show the various submenus accessible from the main menu. Each menu function will be described in the following chapters.

Some submenus in the K Series amps require the user to set a numerical value for specific parameters using the front panel buttons. In order to speed this process up, these submenus dedicate two of the four available buttons to switching to a fast or slow parameter increment mode. When in the "slow" mode, the up and down arrows increase or decrease the parameter by a the smallest amount possible. The "fast" mode will increase or decrease the parameter value by an amount equal to 10 times the amount increased in the "slow" mode. For example:

in "slow" mode: a single "+" button press will increase the Max mains current from 22 A to 23 A

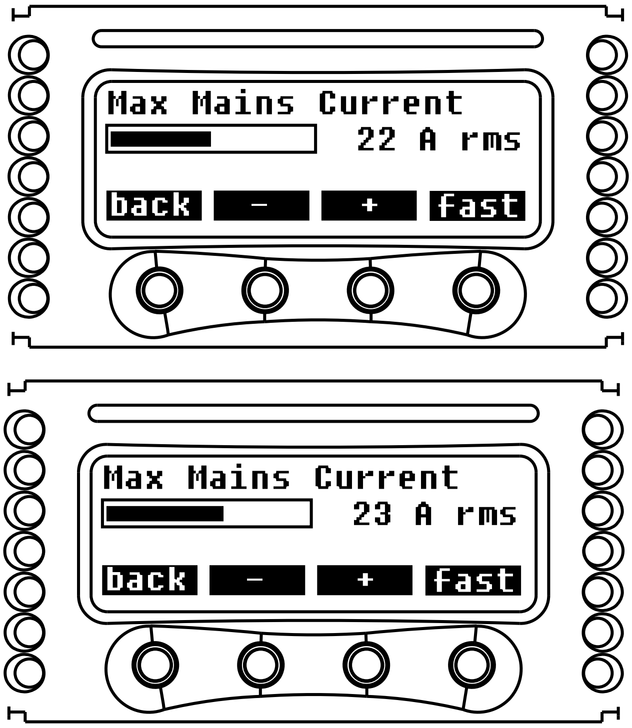


FIGURE 28: K Series slow parameter increase

in "fast" mode: a single "+" button press will increase the Max mains current from 22 A to 32 A

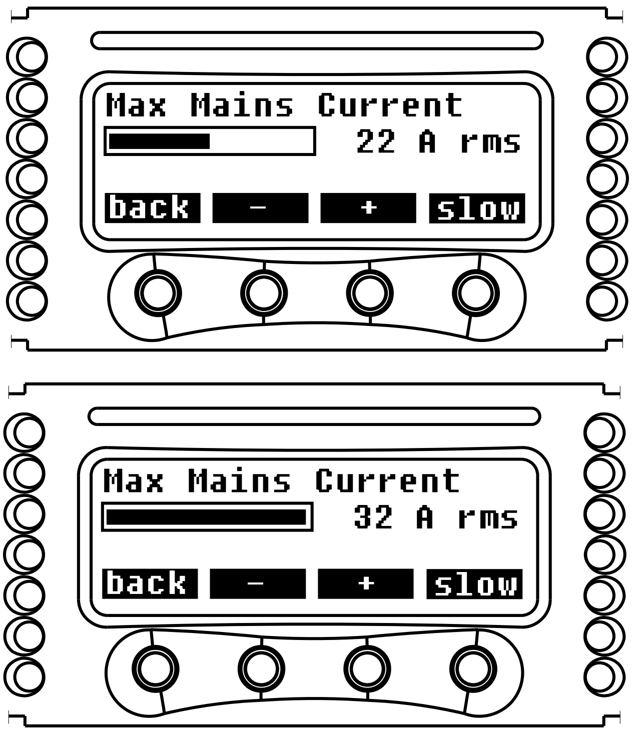
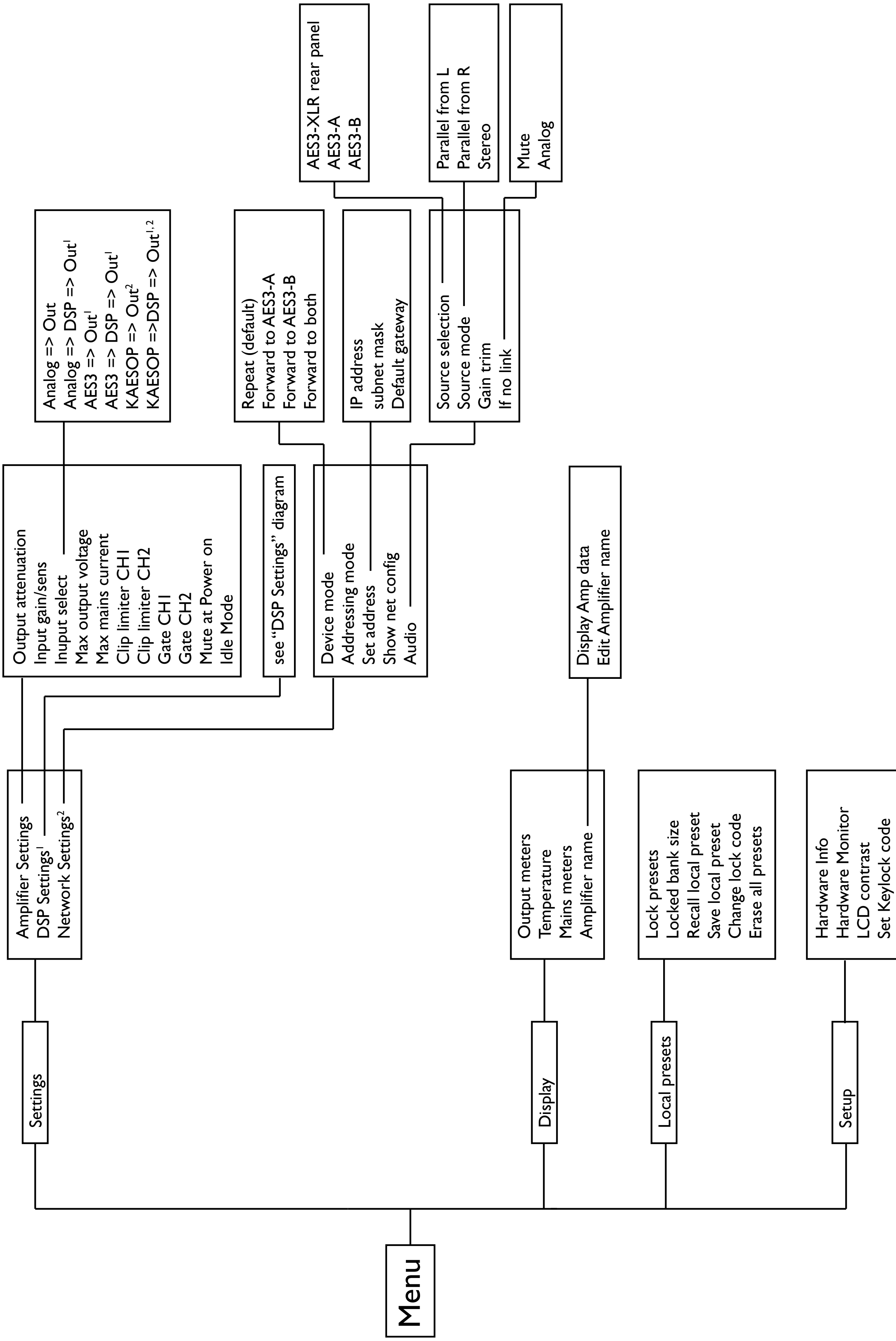


FIGURE 29: K Series fast parameter increase



[1] Available only with optional DSP board
[2] Available only with optional KAESOP board

FIGURE 26: K Series main menu map

DSP Settings

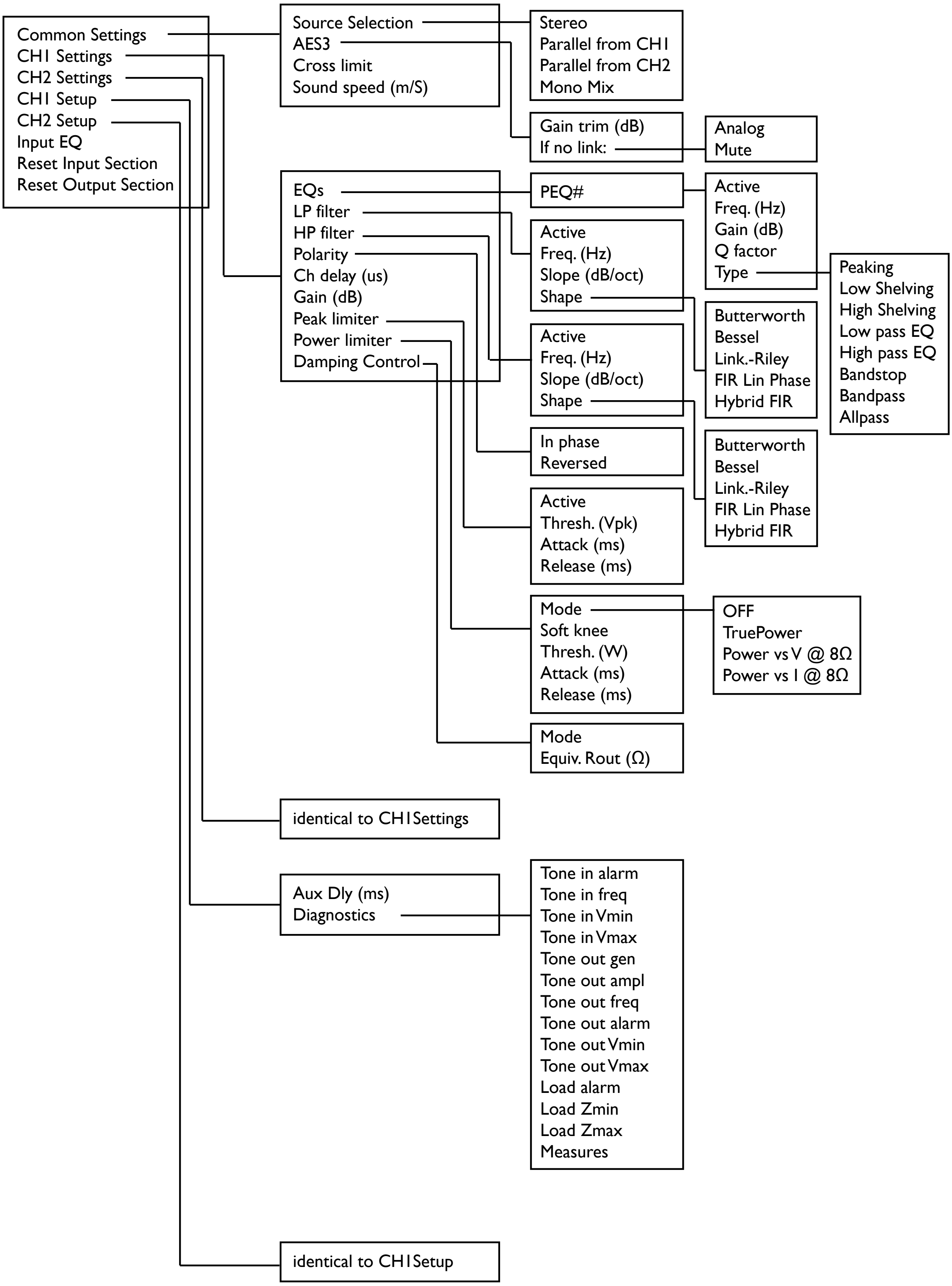


FIGURE 27: K Series DSP settings submenu. Available only for amps with the optiotnal DSP

7 Amplifier Settings

7.1 Output attenuation

The output attenuation screen sets the amplifier's output attenuation level. The user can choose whether to set output attenuation for channel 1, channel 2 or both by cycling through the right most button. The "+" and "-" buttons change the value of the output attenuation in the range from 0 to -30dB. A single "+" or "-" button press will increase or decrease the output attenuation by 1dB. **Note: for ideal sonic performance, select a 0dB output attenuation (meaning no attenuation), and select the proper gain/sensitivity level as explained in the next paragraph.**

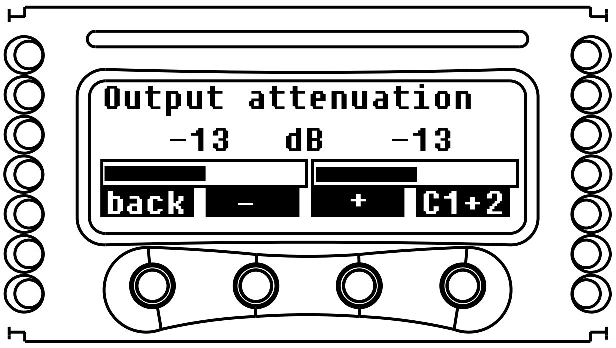


FIGURE 30: K Series output attenuation

7.2 Input Gain/Sensitivity

All K Series amplifiers allow selection of input sensitivity to allow correct sensitivity matching with other third party equipment. The user can choose whether to set the input gain/sensitivity for channel 1, channel 2 or both by cycling through the right most button. The "+" and "-" buttons change the value of the input gain and corresponding sensitivity. The allowed gain values are 26dB, 29dB, 32dB and 35dB. The table below shows the input sensitivity values for the K Series amplifiers. These are the maximum RMS voltage values of an input 1kHz sine wave before clipping occurs at the output stage. These values are reported with respect to the amplifier's gain.

K Series gain sensitivity. Input signal: 1kHz sine wave. Voltage values are RMS:

Gain (dB)	K2	K3	K6	K8	K10	K20
26	4.48	5.30	5.11	5.50	6.34	7.37
29	3.17	3.75	3.62	3.90	4.49	5.22
32	2.47	2.66	2.56	2.75	3.18	3.68
35	1.59	1.88	1.81	1.95	2.25	2.62

The maximum balanced input signal before saturation of the input stage of the amplifier occurs with respect to the amplifier's gain is presented in the chart below. Input signal: 1kHz sine wave. Voltage values are RMS:

Gain (dB)	dBV	dBu	V_{RMS}
26	25.0	27	18
29	21.6	24	12
32	19.0	21	9
35	15.6	18	6

7.3 Input select

K Series amplifiers allow the user to choose three different input modes (if available): Analog, AES3[1] and/or [2], and KAESOP[2]. Each of these inputs can either be processed by the internal DSP or not. The up and down buttons on the "Input select" screen toggle between the available input sources. The "sel" button locks the selected option.

The possible input/signal path configurations are:

- Analog ==> Out (analog input and direct output)
- Analog ==> DSP ==> Out[1] (analog input and internal DSP processing, output)
- AES3 ==> Out[1] and/or [2] (AES3 input, direct output)
- AES3 ==> DSP ==> Out[1] and/or [2] (AES3 input, internal DSP processing, output)
- KAESOP ==> Out[2] (AES3 input, direct output)
- KAESOP ==>DSP==>Out[1] and [2] (KAESOP input, internal DSP processing, output)

[1] Available only with optional DSP board
[2] Available only with optional KAESOP board

7.4 Max output voltage

The max output peak voltage of K series amplifiers can be set by the user. It is possible to set output peak voltage levels for channel 1, channel 2 or both by pressing the "C1+2" button. The "+" and "-" buttons change the value of the max output peak voltage.

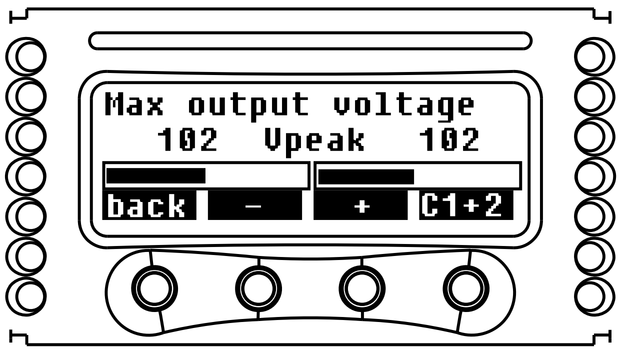


FIGURE 31: Max output voltage settings screen

The ranges available are shown in the table below:

Amplifier model	Peak output voltage (V)
K2	40 to 140
K3	40 to 165
K6	40 to 153
K8	40 to 169
K10	40 to 200
K20	40 to 225

7.5 Max mains current

The maximum current the amplifier can draw from the mains can be set by the user through the front panel of all K series amplifiers.

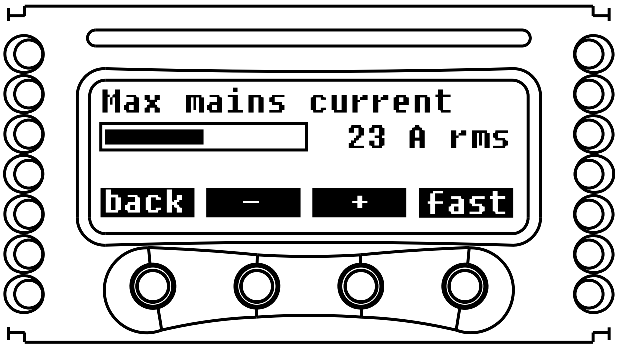


FIGURE 32: Max mains current set up screen

The "+" and "-" buttons allow setting of the value of the max rms mains current. Acceptable values are within the 8 A to 16 A for K2s and K3s and from 15A to 32A range for all other K amplifiers. Setting the maximum mains current determines the current threshold at which a C-Type current breaker will trip.

7.6 Clip Limiter CH1 - CH2

The clip function can be used to prevent distortion caused by clipping of the excessive output signal amplitude. This feature can be disabled or enabled by pressing the on/off button in the when the clip limiter voice is selected in the Amplifier settings menu:

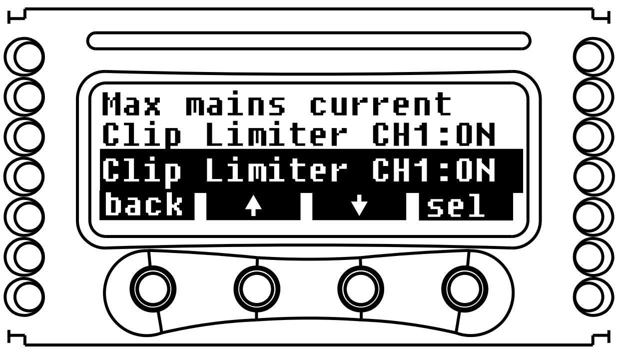


FIGURE 33: Clip limiter setting for channels 1 and 2 separately

Please note that clip limiters can be set independently for both channels.

CAUTION: disabling clip limiters can potentially damage loudspeakers. The amplifier's internal clip limiters should not be deactivated unless the limiting function is implemented by an external device such as digital system controllers. In this case, it is extremely important to correctly set limiting parameters in order to preserve loudspeakers from excessively powerful and potentially hazardous driving signals.

7.7 Gate CH1 - CH2

This function allows to mute the amplifier channels individually if the input signal amplitude falls below the values shown in the following table:

Gain (dB)	dBV	dBu
26	-54	-52
29	-57	-55
32	-60	-58
35	-63	-61

This function can be enabled and disabled by pressing the right most front panel button corresponding to the "on" or "off" label.

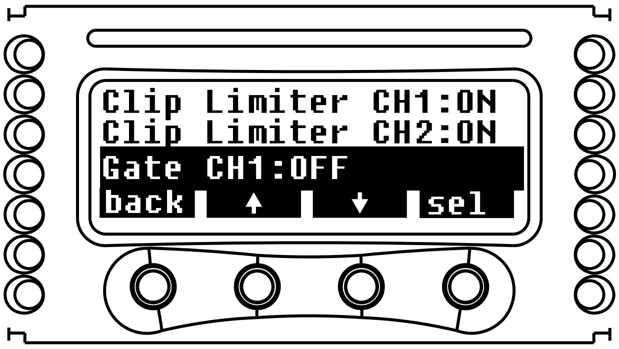


FIGURE 34: Single channel output gate selection screen

Gating the output is delayed by 5 seconds after the input signal falls below the threshold. If the channel is muted, the bottom green LED in the corresponding front panel LED column is off.

7.8 Mute At Power On

This functions allows the user to automatically mute all channels when the amplifier is turned on. Toggle the on or off status by pressing the front panel button below the "sel" label.

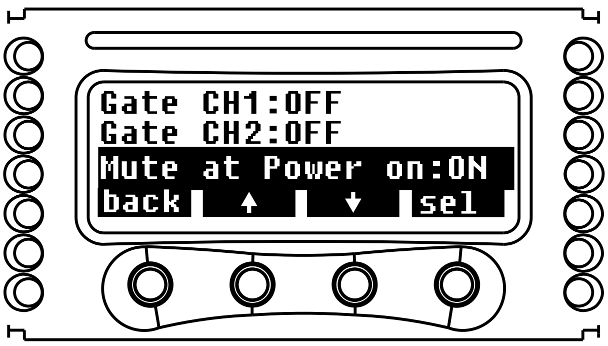


FIGURE 35: Mute at power on function enabled in the settings menu

If this function is enabled, a "Muted" label will appear at the main screen next to each channel at the next power on. Press the button underneath the "mute" label in the front screen to unmute the channel.

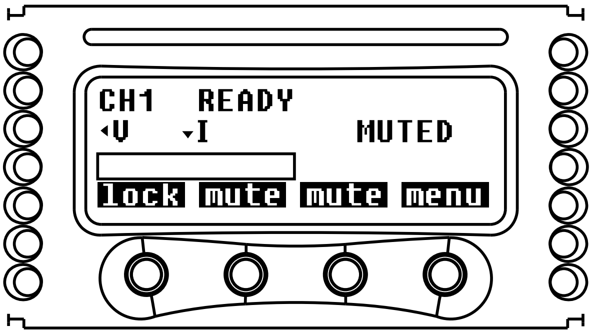


FIGURE 36: Right channel muted, left channel unmuted. Toggle mute status by pressing the "mute" button

7.9 Idle Mode

The idle mode function is a power saving feature. When this function is activated, the output stage is turned off after no input signal greater than -60dBV approximately is detected for a user selectable amount of time, saving about 40W of power per channel (see the table in "7.7 Gate CH1 - CH2" on page 21 for the exact wake up from idle voltage values) This results in reduced heating, longer amplifier and fans life, and, especially for fixed installations which are permanently turned on, a lower electricity bill. Exiting from idle mode is quasi-instantaneous.

In order to set the time after which the amplifier enters in idle mode, push the right most button labelled "sel" when the idle mode line is highlighted. This will open the "Idle state timeout" screen. Using the central buttons, select the desired time. In the "slow" mode, a single button press will increase or decrease the time by one minute. The "fast" mode will bring this up to 10 minute steps. The timeout range goes from 0 to 720 minutes.

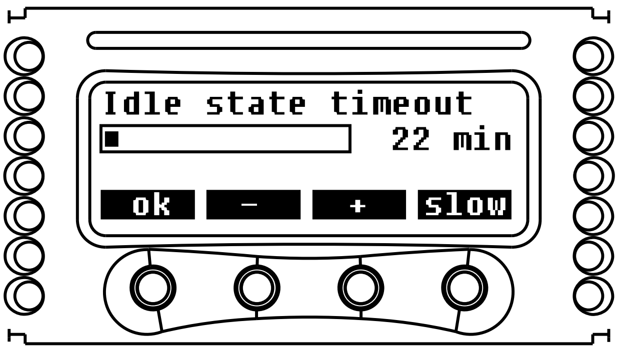


FIGURE 37: Idle timeout set to 22 minutes

8 DSP Settings

The KDSP is a digital signal processing (DSP) add-on board for Powersoft K Series amplifiers. KDSP can be used to optimize the performance of the audio system by means of fully customizable crossovers and equalizers. Exceptionally high reliability is guaranteed in all conditions by advanced limiters, and continuously monitored loudspeaker parameters. This chapter illustrates the features and operational modes of the KDSP board.

8.1 The DSP Processing Chain

The core of the KDSP board is an advanced digital sound processor based on a floating point SHARC® processor. FIGURE 38 shows a block diagram of the DSP processing chain.

8.2 DSP Settings Menu

The DSP settings menu is subdivided in single channel settings or "common" settings affecting both channels.

8.2.1 Common Settings

8.2.1.1 Source Selection

This menu allows to choose the input signal to be processed by the DSP. The possible options are:

- Stereo: the signal coming from channel 1 is processed and routed out to output channel 1. Similarly, the input signal coming from Channel 2 is processed and then routed out to output channel 2.
- Parallel from CH1: the input signal from channel 1 feeds two parallel, distinct and independent processing branches. The result of one branch is sent to output channel 1, while the result of the other branch is sent to output channel 2.
- Parallel from CH2: the input signal from channel 2 feeds two parallel, distinct and independent processing branches. The result of one branch is sent to output channel 1, while the result of the other branch is sent to output channel 2.
- Mono Mix: the input signals from channel 1 and 2 are summed together and divided by two in order to maintain a consistent output level. This mono mix signal is fed to both output channels.

8.2.1.2 AES3

This menu controls the AES3 input stream options. The AES3 source can enter the amplifier from the rear XLR or from the KAESOP board (if present) based on the type of input selection (see "7.3 Input select" on page 20).

8.2.1.3 Gain trim (dB)

This menu allows the user to set the gain to be applied to the signal coming from the AES3 digital input. Setting a 0dB gain makes the full-scale digital signal equivalent to an analog input signal of 20dBu.

8.2.1.4 If no link

This menu controls the amplifier's behavior should the AES3 signal connection fail or become unreliable. The AES3 connection is considered unreliable when transmission errors are greater than 1% of total data transmitted. The possible options are:

- Mute: when the AES3 connection fails, the amplifier mutes the output.
- Analog: when the AES3 connection fails, the amplifier will rely on the analog input as backup. This source signal switching is done in real time in order to avoid any glitches in the audio feed. If the input levels are correctly matched between analog input and AES3 input (use the AES3 Gain trim parameter), the switch between AES3 and analog will be inaudible.

When using the analog input to backup a failed AES3 feed, the analog input connection must be setup based on source type of input AES3 stream:

AES3 from rear XLR:

the primary audio signal for this amplifier configuration is an AES3 signal, fed via the rear panel IN2 with the rear signal type push button set to "AES/EBU". The backup analog cable, with an analog signal identical to that provided by AES3, should be plugged in the IN1 (analog) plug. The amplifier's source selection must be set to "Input from CH1". If the AES3 feed should fail, the amplifier will automatically fall back to the analog input on the CH1 plug. The signal levels of both primary AES3 and backup analog signals should be carefully matched so they are identical. This can be done using the Gain trim parameter or by adjusting the analog signal level.

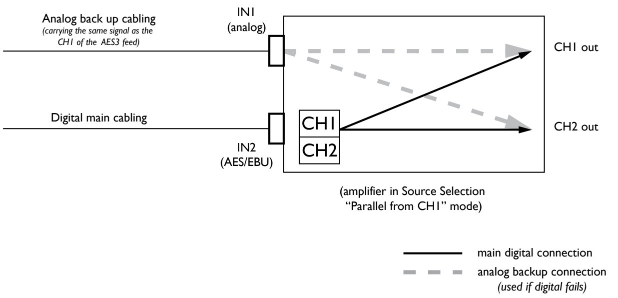


FIGURE 39: Analog back up mode connection: in this example, the amplifier is set to output the AES3 CH1

AES3 from KASEOP:

the primary audio signal for this amplifier configuration is an AES3 signal, fed via an Ethernet port. The backup analog cable, with an analog signal identical to that provided by AES3, should be plugged in the IN1 (analog) and IN2 (set to analog) plugs. The amplifier's source selection can be set to any possible input. If the AES3 feed should fail, the amplifier will automatically fall back to the analog input on the CH1 and CH2 plugs. The signal levels of both primary AES3 and backup analog signals should be carefully matched so they are identical. This can be done using the Gain trim parameter or by adjusting the analog signal level.

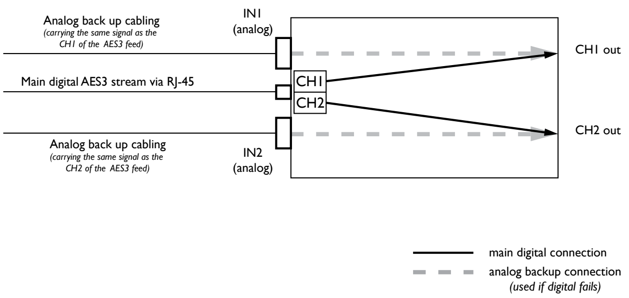


FIGURE 40: Analog back up mode connection: in this example, the amplifier is set to output the AES3 stream in stereo mode. Other configurations of the amplifier mode are possible.

When the AES3 stream is lost and the analog backup kicks in, a message on the front panel is displayed and an alarm is sent to the remote client if one is connected to the amplifier.

8.2.1.5 Cross Limit

In case of power limiting of only one channel, (see "Limiters" on page 25) the gain reduction on one channel is mirrored to the other channel in order to maintain consistent levels. This is useful in two ways speakers where the limitation of one channel alone leads to an unbalanced sound. This function can be turned on or off.

8.2.1.6 Sound speed (m/s)

This menu allow the user to set the sound velocity used for time to distance conversions throughout the local interface. It can be set from 320 m/s to 360 m/s.

8.2.2 Channel Settings

All of the following settings are available for both channel 1 and channel 2. In all the following menus and submenus, the channel number whose properties are being edited is shown in the top right hand corner of the menu. If a specific parameter affects both channels, the top right hand corner will report this as "1+2".

8.2.2.1 EQs

This menu gives access to the parametric output equalizer input interface. This menu lists the 16 parametric filters one by one. The current selected filter number is shown on the left of the first line. By pressing the up and down pointing arrows, it is possible to move from one filter to the next. The filter parameters are reported on the screen.

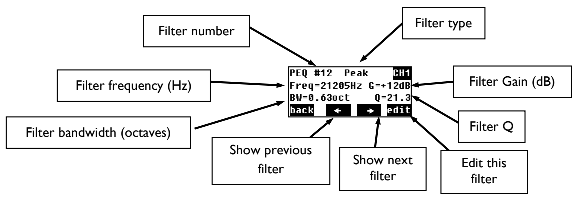


FIGURE 41: Parametric Equalizer (PEQ) information window

Specifically:

- Active: determines if the filter is enabled or not (flat response) Gain(dB): filter gain. Can be set only if the filter is a peaking or shelving filter. Acceptable values go from -15 to +15dBs in 0.1dB steps
- Q factor: quality factor of the filter. This can be user set for all filters except shelving filters. Acceptable values range from 0.1 to 30 with 0.1 steps.
- Bandwidth (oct): the bandwidth of the filter expressed in octaves around the central frequency. This value is the inverse of the Q factor; therefore, its value is determined by setting the Q factor.
- Type: allows the user to select the filter type:
    1. Peaking
    2. Low Shelving (3 to 15dB/oct)
    3. High Shelving (3 to 15dB/oct)
    4. Low pass EQ
    5. High pass EQ
    6. Bandstop

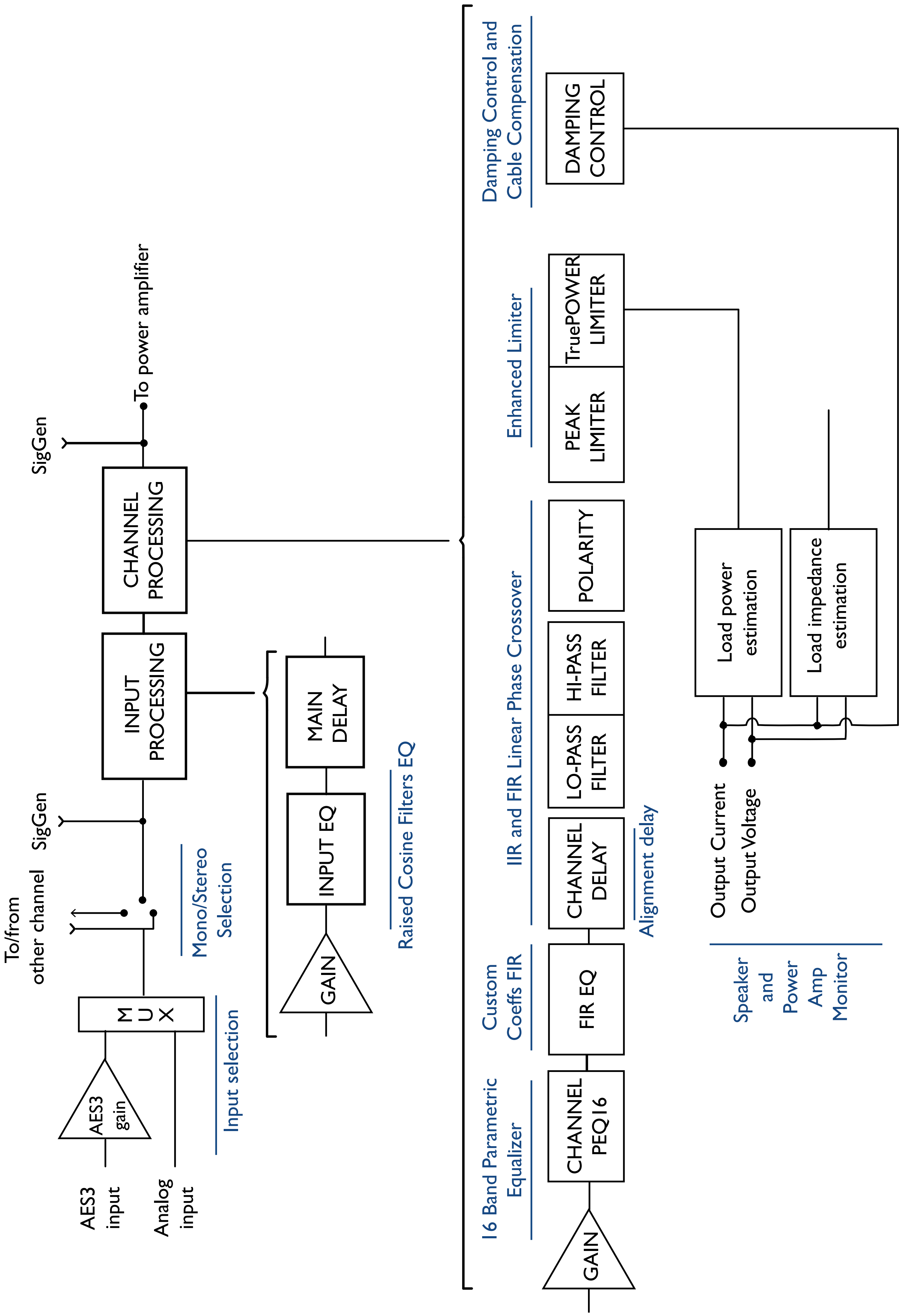


FIGURE 38: DSP processing chain

7. Bandpass
8. Allpass

By pressing the "edit" button, the settings for the selected filter can be changed. The following chart summarizes which parameters can be edited according to the selected filter type.

Parametric Equalizer (PEQ) settings according to filter type:

filter type	Active on/off	Freq (20-20kHz, 1/96 steps)	Gain (-15 to +15dB, 0.1dB steps)	Slope (3-15dB/ oct)	Q factor (0.1-30, 0.1 steps)
Peaking	✓	✓	✓	-	✓
Low Shelving	✓	✓	✓	✓	-
High Shelving	✓	✓	✓	✓	-
Low pass EQ	✓	✓	-	-	✓
High pass EQ	✓	✓	-	-	✓
Bandstop	✓	✓	-	-	✓
Bandpass	✓	✓	✓	-	✓
Allpass	✓	✓	-	-	✓

8.2.2.2 LP Filter (and HP Filter)

This menu allows the user to configure the crossover filters. There are 2 available crossover filters: a lowpass and a highpass. By combining both, the result will be a bandpass response. Both traditional IIRs (Infinite Impulse Response) as well as brickwall linear phase FIRs (Finite Impulse Response) are implemented. If a FIR filter in the EQ section is enabled, a FIR crossover filter cannot be enabled at the same time. The LP or HP filter can be edited by the user via the main LCD screen. The parameters that can be user modified are:

- active status
- frequency
- slope
- filter type

The classic IIR crossover filter shapes that can be selected as a high pass or low pass filter are: Butterworth, Bessel, and Linkwitz-Riley. In the first 2 cases, the frequency parameter in the edit window defines the –3dB point, in the latter, the –6dB point. The slope is freely selectable from a minimum of 6dB/octave (1st order filter) to 48dB/octave (8th order filter).

The FIR filters can be selected as normal (FIR Linear Phase) or enhanced (Hybrid FIR). The enhanced version of the filters gives a higher rejection of out of band signals, at the expense of a small (30°@400Hz) phase modification. In both cases, the minimum working frequency is relative to the desired latency. Standard setting limit this to 400 Hz. For this reason it is advisable to use FIR filters to crossover upper midranges or mid-high drivers for which the phase coherency is a key point.

8.2.2.3 Polarity

This menu allows to reverse the signal polarity. The two selectable modes are:

- In phase: the signal's polarity is not altered
- Reversed: the signal's polarity is reversed.

8.2.2.4 Channel Delay

This menu allows to set a single channel output delay. This is helpful to time-align two different loudspeakers on the two amplifier channels. The selectable delay varies from 0 to 32 ms (about 11 meters), with a single sample step (equal to 1/96000th second or 10.4 us, about 3.5 mm)

8.2.2.5 Gain

This menu changes the channel gain, from –40dB to +15dB, with a 0.1dB step.

8.2.2.6 Limiters

The limiting process in sound reinforcement is a way to protect loudspeakers from accidental damage; therefore, limiters are a safeguard against excessive signal peaks and/or signal power. They not only protect from sudden signal peaks but also they protect against to an over power delivering.

Bear in mind that limiting does not only prevent occasional damage, but it first and foremost guarantees a long component life. The two main purposes of limiting process are:

- **Over-excursion:** an impulsive signal can reach the speakers and cause damage due to over-excursion of the voice coil that is driven out of the magnetic gap (where displacement exceeds Xmax). This can damage the diaphragm (breaking or deforming it).
- **Over-heating:** delivering high power to the voice coil may lead to overheating of the voice coil copper and the relative magnetic gap. This can damage the isolation copper or burn out the copper. Another evident high power driving effect is power compression, noticeable in low frequency speakers.

In order to prevent the two mentioned phenomena two kinds of limiters are provided:

- **Peak limiter:** protects against mechanical damages. The peak limiter may also be used to control amplifier clipping. Designers should set this limiter's parameters as a function of both the maximum displacement (Xmax) of the diaphragm as well as the speaker's maximum tolerated voltage.
- **RMS limiter:** protects speakers against thermal damage when excessive power is applied for extended periods of time, resulting in overheating and, eventually, burning. Designers should be aware of the maximum long term power safely applicable to speakers (AES power rating). An interesting approach to RMS limiting is one that uses coil temperature control. A complete knowledge of the driver's limits allows to keep the temperature level in a safe interval not only to avoid damage but to maintain the speaker in a "linear" zone that avoids power compression.

Peak Limiter

The peak limiter avoids potentially dangerous displacements of the cone (an excursion larger that allowed). It acts by reducing the amplifier gain in order to reduce the measured output peak voltage. Use the declared Peak power or twice the Program power as a loudspeaker safe-zone output power. The peak limiter's setting do not change with the number of parallel speakers connected to the amplifier; this is because the same voltage is applied to all the components in a parallel circuit. When deciding parameters for a peak limiter of an amplifier with many loudspeakers connected to it in parallel, the peak power to be taken into consideration is that reaching only a single speaker.

$$P_{peak} = \frac{V_{peak}^{2}}{R}$$

$$V_{peak} = \sqrt{P_{peak} \cdot R}$$

Where R is the nominal impedance of **only ONE** driver, P_{peak} is the peak power and V_{peak} is the peak output voltage. A peak limiter, used with a very rapid onset (i.e., with a very short attack time), can also be useful in limiting the maximum peak voltage in distributed constant voltage lines.

Powersoft designed the K Series limiters as protective measures; therefore, they are not meant to "color" the sounds such as dynamic compressors can do. With this in mind, time constants for these limiters should be selected so as to limit potentially harmful phenomena which persist for no more than one or two periods of the related signal bandwidth. To limit the dangers of dangerous very fast transient signals, all limiters implement a look ahead time of 0.5s.

The following table gives a few examples of attack and release times with respect to the frequency range of the signal to be limited:

FREQUENCY RANGE (Hz)	ATTACK TIME (ms)	ATTACK/ RELEASE RATIO	RELEASE TIME (ms)
<63	45	x16	720
63-125	16	x16	256
125-250	8	x8	128
250-500	4	x8	32
500-1k	2	x4	8
>1k	1	x2	2

The peak limiter menu allows the user to define the following parameters:

- Active: toggles the power limiter's on/off status
- Threshold (V_{pk}): the peak voltage threshold at which the gain begins to be reduced
- Attack: the attack time,i.e. the response time of the limiter intervention
- Release: the decay time, i.e. the time constant after which the limiter's action is released and the gain restored to the nominal value.

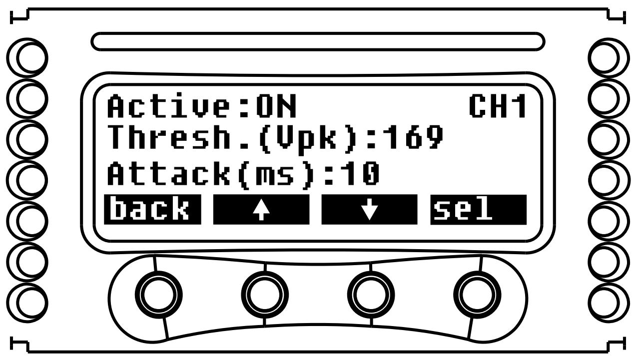


FIGURE 42: Peak limiter main screen

In order to avoid choking the exceptional dynamic range offered by K Series amplifiers, the peak limiter is designed to ignore signal peaks lasting less than the attack time parameter. Moreover, the limiter has an additional lookahead buffer to soften clipping and minimize distortion, effectively yielding superior sonic performance. The lookahead time is 0.5 ms.

When tweaking the peak limiter's levels, it is preferable to first setup the time parameters, and then adjust the threshold voltage. When editing the threshold value, the display shows the gain reduction (GR) in dBs enforced by the limiter. This information, together with the limiting voltage referred to the signal in the input amplifier stage (I) expressed in dBus, is displayed in real time to allow monitoring of the limiting actions as they are performed.

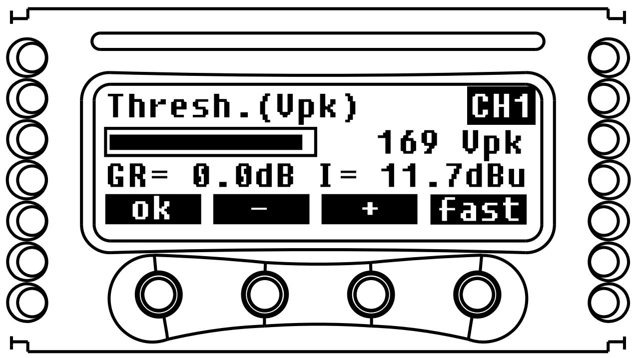


FIGURE 43: Peak limiter threshold value editing screen

RMS Limiter

Given the low efficiency of electromechanical transducers, almost 50% of power reaching the voice coil is transformed into heat. The power limiter is intended to avoid melting the voice coils of drivers while at the same time exploiting their maximum performance. All the power limiter base their operations on the temporal behavior of the voltage and the current, this means that the amplifier can knows the real amount of real power delivered to the load. A correct power limiting is not an easy task and is multifaceted, based on a number of variable, like the knowledge of the component heat dissipation and the goals that must be achieved. Therefore may be difficult and a little bit empirical decide thresholds and constants time. Power limiters behavior base their operations on a mix based on threshold, temporal behavior of the output readings (voltage and/or current) and the type of output readings monitored. The power limiter should be used to protect the drivers from melting. It should NOT be engaged at normal working levels. Check the gain reduction: in order to obtain the optimal sound it should not be greater than 2-4dB even for the loudest piece of music. Please note that a common musical signal has very high peaks, but a rather small average level (high crest factor). A continuous tone has a much higher average power even if it "sounds" less loud to the human ear. This must be taken into account while setting up limiter parameters. The power limiter acts by decreasing the amplifier's gain in order to reduce the power delivered to the load. There are three main operating modes for the K Series power limiters.

TruePower™

In the TruePower operating mode, the amplifier's active output power is estimated by measuring the load current. The TruePower limiter is a Powersoft patent technology useful to avoid overheating of the voice coil; it can however also be used to avoid power compression. The amplifier's DSP provides the measurement of the real power delivered (and then dissipated) to the coil, not the apparent power handled by the line.

Empirical observation yields the following equation:

$$P_{max\ diss} = \frac{P_{AES}}{3}$$

where P_{AES} is the declared AES Power and $P_{max\ diss}$ is the maximum power the speaker can dissipate "in real life".

If the P_{AES} is not available, the P_{RMS} (declared maximum RMS power the loudspeaker can handle) can be used; however, it is important to proceed with caution in evaluating how the P_{RMS} value is obtained. If no other values are declared, this rule of the thumb can be used: the P_{AES} can be estimated as 6dB below the peak power (¼ of the peak power). It is very important to note that, contrary to what happens with the peak limiter, setting the TruePower limiter parameters must take into account the number of speakers connected to the amplifier. This is due to the fact that the real power is calculated not only with the output voltage (which is identical for all speakers connected in parallel) but also with the output current (which changes according to the number of parallel speakers).

Determining the ideal time parameters for TruePower limiters is a very empirical process. As a guide, consider this simple rule:

Larger the coil, larger the thermal inertia, larger the time constant.

The following table summarizes this concept with practical numbers:

Driver **voice coil** size (inches) and application	True Power Threshold (W)	Attack time (ms)	Decay time (ms)
1" tweeter	10 - 20	100	300
1.5" tweeter	20 - 30	150	300
2" horn driver	20 - 40	200	400
3" horn driver	30 - 50	300	500
4" horn driver	40 - 60	500	3000
2" midrange	30 - 100	500	3000
3" midbass	50 - 150	1000	5000
4" woofer	100 - 200	2000	5000
4" woofer	150 - 250	4000	8000
6" woofer	250 - 500	6000	10000

Power vs V @ 8 Ohm

In the Power vs V @ 8 Ohm operating mode, the amplifier's output power is estimated by measuring the RMS value of the output voltage, assuming an 8 ohm load. This mode allows to create settings that work well for any number of speakers connected in parallel. For example, if a "Power @ 8 Ohm" limiter is set to limit the output power to 150W, a single cabinet will be delivered a maximum of 150W with 8 ohm load. Two speaker cabinets connected in parallel will be delivered a maximum of 300W with 4 ohm load (" 8 Ohm loads in parallel) and so on.

This limiter is a pure RMS limiter whose functioning is based solely on the voltage module measured at the amplifier output. Differently from the TruePower limiter, this limiter does not take into account the real part of the power; however, it has the advantage of being independent from the number of cabinets linked together, just as a peak limiter.

Some attention is needed to set the power threshold. The P_{AES} can be used if it is available. If no other power rating is declared, the P_{RMS} can be used; however, the RMS parameter is a value related to the maximum manageable power and not the real power. Proceed with caution because the manageable power could be greater than the real power. Some constructors declare the RMS power as the minimum impedance point of the speaker; this, again, may lead to an overestimation of the true power values the speaker can handle. If no other values this rule of the thumb can be used: the P_{RMS} can be estimated as 6dB below the peak power (¼ of the peak power).

In order to preserve the driver in the long term, once the maximum power limit is decided upon, consider a power reduction of up to 3dB of that value.

In order to use this limiter correctly, it is important to recalculate the equivalent power at 8 Ω.

For example:

with an 8 Ω speaker with maximum RMS power of 500W, the threshold power limit is straightforward because the max RMS power is already given with respect to an 8 Ω load. But if, for example, the maximum RMS power is 500W for a 4 Ω speaker, the equivalent power at 8 Ω needs to be calculated.

To calculate the equivalent power at 8 Ω:

1. calculate the RMS voltage value needed to generate the maximum RMS power on the 4 Ω speaker:

$$V_{RMS} = \sqrt{P_{RMS} \cdot R}$$

where V_{RMS} is the RMS voltage of the speaker and P_{RMS} is its maximum RMS power. The RMS voltage of the 4 Ω speaker in the above example is V_{RMS}=44.7 V.

2. calculate the power delivered to a speaker with nominal impedance of 8 Ω with a V_{RMS} voltage:

$$P_{RMS\ equiv} = \frac{V_{RMS}^2}{8}$$

where $P_{RMS\ equiv}$ is the equivalent power on the 8 Ω speaker and V_{RMS} is the RMS voltage value calculated at step 1. In this example this is 250W. This is the threshold power to set in the limiter.

The time constants for the Power vs V @ 8 Ohm limiter can be set in the same way as for the TruePower limiter.

Power vs I @ 8 Ω

This limiter's behavior is similar to the case Power vs V @ 8 Ω with the difference that all calculations are based on the current (and not voltage) measured at the output. In this case the formula to derive the RMS power from the RMS current is:

$$P_{RMS} = \frac{I_{RMS}^2}{R}$$

where P_{RMS} is the RMS power and I_{RMS} is the RMS current. This limiter is particularly useful in situations where the parameter to be controlled is the output current (e.g. for tweeters). It is also useful for special applications such as large coil speakers with current controls. When determining this limiter's parameters, it is necessary to take into account the number of speakers connected in parallel to the amplifier.

Power limiter settings

Power Limiter parameters that can be edited by the user are:

- Mode: allows to determine the power limiter
  1. OFF/ON turns the limiter on or off
  2. TruePower: sets the limiter mode to TruePower
  3. Power vs V @ 8 Ohm
  4. Power vs I @ 8 Ohm
- Soft knee: (ON/OFF)
- Thresh.(W): threshold output power level expressed in Watts at which the gain begins to be reduced
- Attack(ms): the time it takes for the limiter to start reducing the amplifier gain once the output power has exceeded the threshold value
- Release (ms): the time constant after which the gain is restored its nominal value once the output power has returned below the threshold

When editing the power threshold value, the display shows the gain reduction (GR) in dBs enforced by the combined effect of the peak and power limiters. This information, together with the average power truly delivered to the load (Pavg), is displayed in real time to allow monitoring of the limiting actions as they are performed.

8.2.2.7 Damping Control

WARNING: when damping control is enabled, a lowpass filter cutting around 400 Hz is automatically inserted into the amplifier chain. This feature is intended to be used only for subwoofer applications.

This unique and patented feature allows to add a "virtual" series resistor to the amplifier output. This is done to obtain the desired damping factor with any cabling used. For this end, the virtual series resistor can also have a negative value to compensate cabling resistance. For example, using a 10 meter cable to powering the subwoofer means adding a series parasitic resistance of about 0.3 Ohms. By enabling the damping control, a virtual negative series resistance can be added to compensate the cable resistance.

Typical cabling resistance:

AWG	section area	Length (m)	Resistance (Ω)
16	2 x 1.5 mm²	5	0.13
16	2 x 1.5 mm²	10	0.26
16	2 x 1.5 mm²	20	0.52
14	2 x 1.5 mm²	5	0.08
14	2 x 1.5 mm²	10	0.16
14	2 x 1.5 mm²	20	0.32
12	2 x 4 mm²	5	0.05
12	2 x 4 mm²	10	0.10
12	2 x 4 mm²	20	0.20

Another advantage offered by the damping control feature is that in adding the series equivalent output resistance to the amplifier chain, the voice coil resistance increase due to heating can be taken into account. This allows to obtain a correctly damped bass response at average working condition, where the voice coils is subject to heating due to the passage of current. For example, if the subwoofers are going to work at close to full power, an additional negative resistance of 1 to 2 Ohms should be added to compensate the high resistance generated by the heated voice coils to obtain a correctly damped response. On the other hand, if the same subwoofers are working at low power, a smaller negative resistance should be added: in this case the cooler voice coil presents a smaller series resistance to be compensated. Leaving too high an equivalent series resistance results in an overdamped system.

Typical resistance increase due to voice coil heating. Notice the exceptionally high value (3.8 Ohm) when the driver reaches it thermal limit:

Average power/ rated power	Power compression (dB)	Equivalent series resistance for 8 Ω driver (Ω)
10%	1.4	1.0
20%	2.0	1.4
50%	2.8	2.1
100%	4.5	3.8

8.3 CH1/CH2 Setup

8.3.1 Auxiliary Delay

This delay is a further input delay acting on the input EQ This delay is not based on the input eq bypass.

8.3.2 Diagnostics

The diagnostics tool allows the user to program and test the integrity of the input and/or output line. The input test is based on the detection of a pure tone (generated by an external tone generator) on any input line. The output test reliess on the measurement of the impedance at a well defined frequency: the amplifier can generate a pure tone and measure the voltage and current at the generated tone frequency. It is therefore possible to recalculate the impedance at that specific frequency. When an alarm condition is met, the user can be informed of the event via software or directly from the amplifier.

Tone in Alarm

The tone in alarm can measure the integrity of any input line feeding signal into the amplifier. This detector can measure a tone applied by an external generator.

- Tone in Alarm, enable/disable the input tone detection
- Tone in Vmin, the frequency of the tone that has to be detected (range 20 Hz - 24 kHz, step of 10 Hz)
- Tone in Vmax, the minimun thershold value that has been detected (range 0 Vrms - 4 Vrms, step of 10 mVrms)
- The maximum threshold value that has been detected (range 0 Vrms - 4 Vrms, step of 10 mVrms)

Tone out gen

The inner tone generator allows the user to generate a tone that can be used to check the integrity of the output line. This tone should be used outside of the freqeuncy bandwidth of the driven speaker to avoid can be listen.

- Tone out gen, enable/disable the internal generator
- Tone out ampl, the output voltage of th generator (range 0 Vrms - 20 Vrms, step of 1 Vrms
- Tone out freq, the frequency of the tone that has to be generated and eventually detected (range 20 Hz - 24 kHz, step of 10 Hz)

Tone out alarm

The output tone detection can measure the presence of a tone generated by an external or internal generator.

- Tone out Alarm, enable/disable the output tone detection.
- Tone out Vmin, the minimun detected threshold voltage value (range 0 Vrms - 20 Vrms, step of 1 Vrms)
- Tone out Vmax, the maximum detected threshold voltage value (range 0 Vrms - 20 Vrms, step of 1 Vrms)

Load Alarm

The output Load Monitor allows to detect the impedance load at a certain frequency. The high resolution algorithm implemented in this tool allows accurate measures.

- Load Alarm, enable/disable the impedance detection.
- Load Zmin, the minimun allowed impedance threshold value (range 0 Ω - 500 Ω, step of 0.1 Ω)
- Load Zmax, the maximum allowed impedance threshold value (range 0 Ω - 500 Ω, step of 0.1 Ω)

Measures

Pressing the button measures gives access to a sub menu where the various amplifier readings are available.

- Tone in, measurements of the input tone at the selected frequency.
- Tone out, measurements of the output tone at the selected frequency.
- Z load, measurements of the load at the selected frequency.

8.4 Input EQ

This menu allows to turn on / turn off the input processing block. This can be useful when resetting the amplifier to the original “output processing only” behavior without using any software. Turning off the Input EQ, all input processing set up using, for example, the Armonía Audio Suite can be bypassed at once. It is advisable to save amplifier presets with this setting turned off: in this way when loading presets the user can be sure that only the output processing is enabled. The burden of re-enabling and setting up input processing is left to the remote control software.

8.5 Reset Input Section

This operation disables the input processing (input EQ, input gain and delay) and resets the aux delay to zero.

8.6 Reset Output Section

This function disables all output EQ, limiters and damping functions.

Warning: this operation may potentially damage connected speakers. Pay special attention to shutting down any audio source before using this function.

9 Network Operations

Network capabilities and network setting menus are available only for K Series amplifiers equipped with a KAESOP board. KAESOP stands for K (as in Powersoft’s K Series) AES3 and Ethernet Simple

Open Protocol. Powersoft's KAESOP is designed to provide high reliability to live applications in harsh environments where Quality Of Service must be guaranteed. Electromagnetic and radio frequency interference (EMI and RFI) originating from a high power audio and light system must not degrade audio quality or cause a control link interruption. Moreover, a single cable or device failure should not affect the overall system performance.

9.1 User's introduction to AESOP

The AESOP standard can transport a single bidirectional Ethernet 100Mbps control data stream and two separate AES3 digital audio monodirectional streams using one CAT-5 cable. All K Series amplifier with the optional KAESOP board installed are equipped with at least two RJ45 connectors, each a single AESOP port, capable of sending and/or receiving data and audio. If the amplifier has only two RJ45 plugs, these will be on the frontal panel. If four plugs are present, the rear two will be "master" ports, while the two on the frontal panel are "slave" ports. Master ports allow both data and AES3 streams; slave ports, on the other hand, are data-only ports, allowing Ethernet connections only. Ring, daisy chain and a variety of network topologies are possible using the dual port design implemented in all K series amplifiers.

9.1.1 Data stream

The data stream in the AESOP is implemented by a 100 Mbit Ethernet connectivity with auto-sense. The dual port design in K Series amplifiers allows for daisy chain and redundant ring topologies. A fault-bypass built in feature takes into account the possibility of loosing an intermediate device or having a faulty cable link without compromising the ring integrity. Each device can use a static IP address assigned by the user. Alternatively, it can be set to automatically configure itself without user intervention following the Zeroconf protocol. The KAESOP board detects bad quality connections by counting errors on the Ethernet control. Faulty connections are automatically switched from 100Mbit/s to 10Mbit/s to attempt to keep the link active even in the worst case scenarios. Please note that even if crossed Ethernet cables would work control wise, crossed cables are NOT to be used for KAESOP connections: they will not allow the AES3 streams to flow correctly.

9.1.2 Audio

Audio is distributed to devices via the AESOP protocol by 2 independent and separate AES3 streams. These are carried by two CAT-5 wire pairs unused in the 100 Mbit Ethernet protocol. AES3 is a license free and well known standard guaranteeing low-latency, high reliability and excellent audio quality. A single AES3 stream can carry a stereo audio signal. The AESOP protocol can therefore handle four audio channels.

When a K Series amplifier is powered off or if it is unavailable, a passive high frequency relay circuit allows the audio signal to pass through, preserving the network chain connection integrity. When the device is powered up, the internal circuits automatically select the most appropriate AES3 stream direction and bypass the relay, re-buffering actively the AES3 signal. The direction is maintained until errors are detected on the AES3 receiver circuit. When errors or link failure are detected, the direction is swapped, to build-up a new path for the audio. In a fraction of a second (no more than 50ms), some of the devices in a ring will swap to the other direction, restoring the audio streaming.

9.1.3 Network connections: Ethernet, AES3 forwarding and repeater modes

Each K Series amplifier can be configured to handle the pair of AES3 streams embedded in the AESOP protocol in one of two basic network modes: repeater and forwarder. The following section will describe these two different setups in detail. These are true connection "building blocks"; it is therefore important to understand these two modes thoroughly before attempting to create or modify larger and more complex amplifier networks.

The following are definitions of the terms used in this section:

- AES3-A STREAM, AES3-B STREAM: streams from the AESOP CAT-5 network. Each stream can carry a stereo audio signal.
- REAR AES3 STREAM: AES stream from the rear panel CH2 XLR with when the toggle button is in the AES/EBU selected position.
- PORT 1, PORT 2: master RJ45 AES3 and control ports (on the rear panel of amplifiers with four RJ45 ports, on the front panel for amplifiers with only two RJ45 ports).
- PORT 3, PORT 4: slave control data-only ports(on the front panel of amplifiers with four RJ45 ports, not present in amplifiers with only two RJ45 ports).

Ethernet internal switch

All control data streams in the KAESOP system are transported via an Ethernet protocol. Inside all K Series amplifiers is an Ethernet switch connected to each RJ45. This means that the bidirectional data stream can enter/exit one port and exit/enter any other port, either alongside AES3 streams or on its own. Internal routing of Ethernet networking is automatic and not user controllable. An internal switch provides packet flooding block services in order to allow building networks with a ring topology.

KAESOP repeater mode

In the "Repeater" mode, any AES3 stream received on port 1 will be repeated on port 2 and vice-versa: if the AES3 stream is received on port 2 it will be repeated on port 1. This applies to both AES stream A and AES stream B independently. If an AES3 stream (A or B) is present as input at both RJ45 ports (this can happen when a ring network topology is used), the internal AESOP repeater feeds only one of the two identical streams keeping the second stream in standby. If for some reason the first stream fails the second stream is used as a backup audio source.

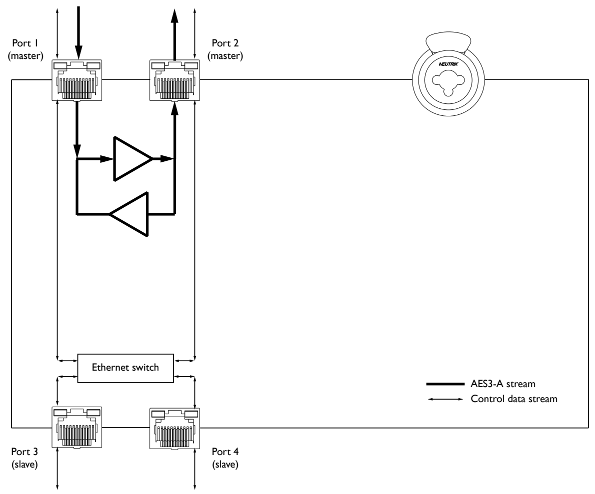


FIGURE 44: This diagram illustrates a simplified internal AES3 and Ethernet data path. The amplifier is set to repeat the AES3-A stream coming from master port 1 to master port 2. For consistency, master ports are placed in the rear of the amp, while slave ports are at the front. Notice that AES3 streams are monodirectional, while data stream is bidirectional.

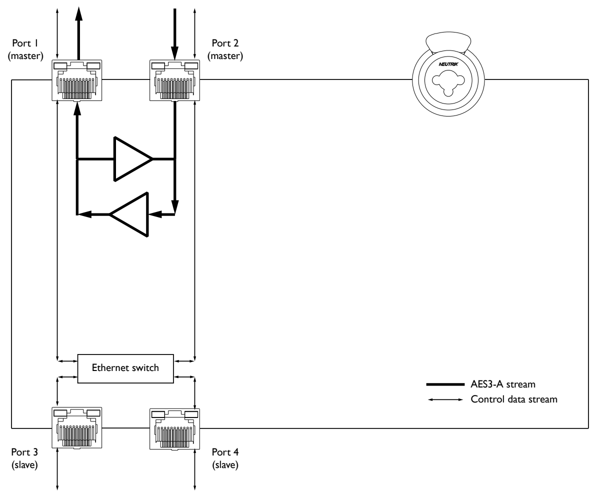


FIGURE 45: This diagram shows the amplifier set to repeat the AES3-A from master port 2 to master port 1.

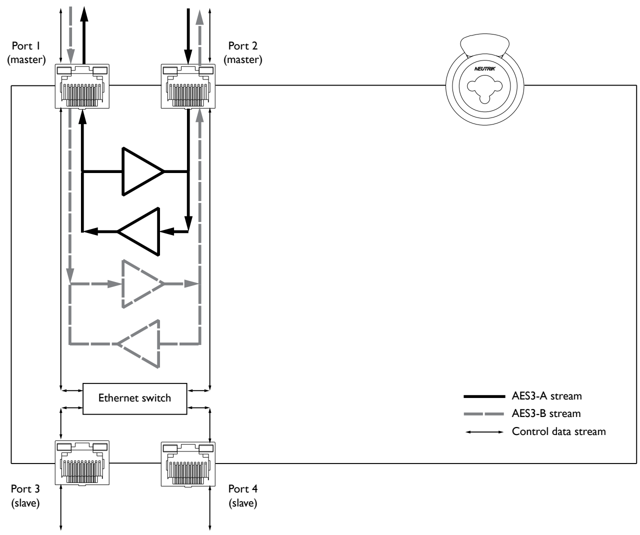


FIGURE 46: This diagram shows both AES3 datapaths in repeater mode. In this example, the AES3-A streams enters port 2 and is repeated out of port 1. At the same time, the AES3-B stream is incoming in port 1 and is repeated outwardly via port 2. All possible permutations are not displayed.

KEASOP forward mode

When the amplifier is set in forward mode, the AES3 signal coming into the amplifier from the rear panel XLR connector is forwarded to both of the master RJ45 ports. The rear panel toggle button next to the CH2 XLR connector must be in the "AES/EBU" position. There are three ways the AES can be forwarded:

- Forward to AES3-A:

the amplifier's rear panel AES input via the XLR connector will be routed to the AES stream A on both master ports 1 and 2. If there is an AES3-B stream incoming from either master ports (1 or 2), this will be repeated on the other master port. For example, the figure below shows the "Forward to AES3-A" function where the AES3 stream coming from the rear XLR connector is forwarded to the AES3-A stream and no AES3-B stream is present.

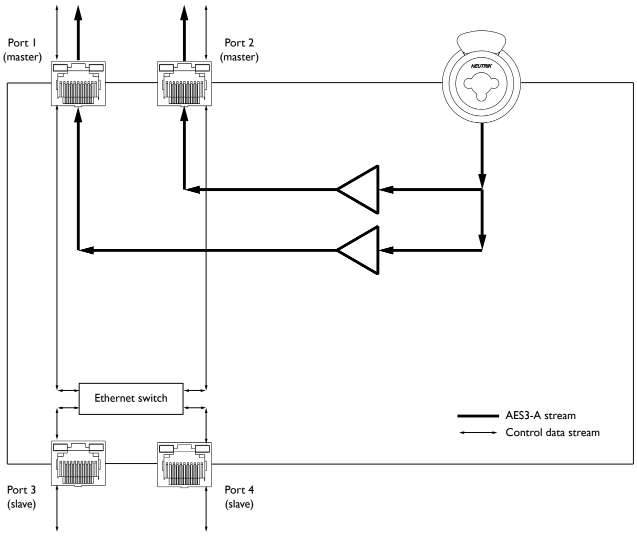


FIGURE 47: Forward to AES3-A signal path. No AES3-B stream present

This figure, on the other hand, illustrates the signal path in "Forward to AES3-A" mode when an AES3-B stream is present; the AES3-B stream is, in this example, incoming through master port 1. The AES3-A stream, if present will be repeated from/to master RJ45 ports 1 and 2.

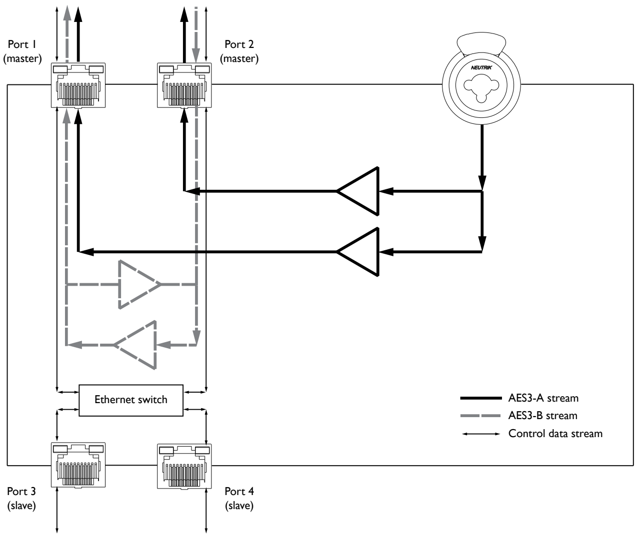


FIGURE 48: Forward to AES3-A signal path and simultaneous AES3-B stream in repeater mode.

- **Forward to AES3-B:**

the amplifier behaves just as in the "forward to AES3-A" mode but with respect to the AES3-B stream. The AES3 stream coming from the rear panel XLR connector will be routed to the AES3-B stream on both RJ45 ports 1 and 2. The AES3-A stream, if present will be repeated from/to master RJ45 ports 1 and 2.

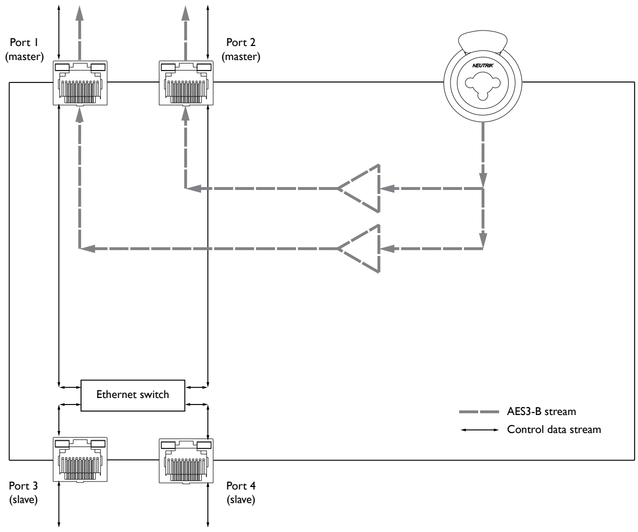


FIGURE 49: Forward to AES3-B signal path. No AES3-A stream present

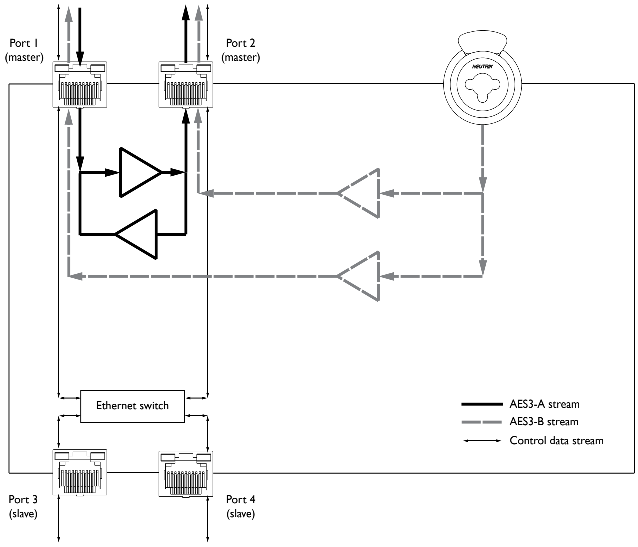


FIGURE 50: Forward to AES3-B signal path and simultaneous AES3-A stream in repeater mode.

- **Forward to both:**

the amplifier's rear panel AES input via the XLR connector will be routed to both AES3 stream A and AES3 stream B on both main ports 1 and 2. Repeater functionality will be disabled.

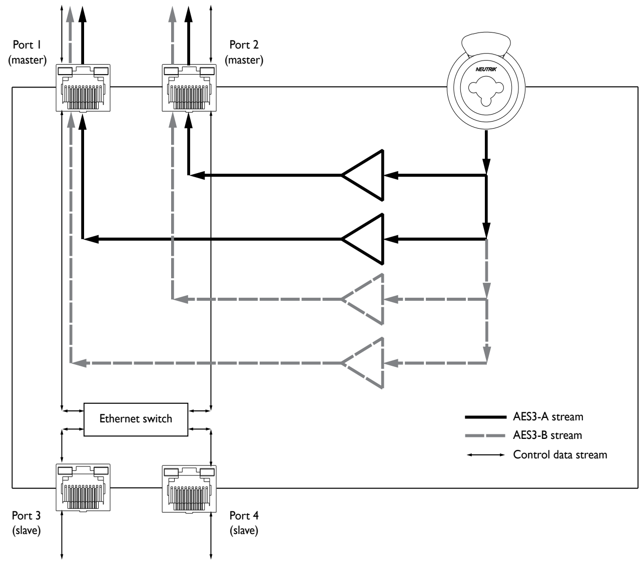


FIGURE 51: AES3 stream coming from the rear XLR stream is routed to both AES3 streams A and B via the master RJ45 ports.

IMPORTANT: when an amplifier is set to forward the XLR AES3 signal to either the AES3-A or AES3-B stream, the amplifier can accept as the sole AES3 input signal the one coming from the XLR connector. The RJ45 ports cannot, when the amplifier is in forwarding mode on both streams, input an AES3 signal to the amplifier.

9.2 Network robustness

K series amplifiers equipped with a KAESOP are capable of being connected each to the other via a network: using a single sound source, each amplifier in the network can be, for example, dedicated to providing power audio signal to a given subsection of a large venue. In dealing with networks of amplifiers, one of the most important aspects to consider, especially when working in a critical application such as large venue sound distribution, is the robustness of the network itself. Data and audio connections can be made "fault proof": this means that if for some reason one audio or data connection should fail, the whole system is not compromised. The degree of redundancy expresses how many network connections can break before sound is interrupted in any one amplifier part of the system. A "zero degree" redundant system is not robust: the first connection to jump (either from a cable failure or even from an amplifier problem) means the whole system goes down. A "one degree" redundancy system, on the other hand, will continue working automatically if one (but no more than one) connection fails. This happens because K series amplifiers can sense a connection failure and automatically (and almost instantaneously) invert the audio feed direction to allow the source signal to remain uninterrupted.

The following section illustrates and analyzes some common amplifier networks divided by redundancy degrees.

9.3 Network connections

- Daisy chain

The following diagrams show a daisy chain connection of 4 amplifiers.

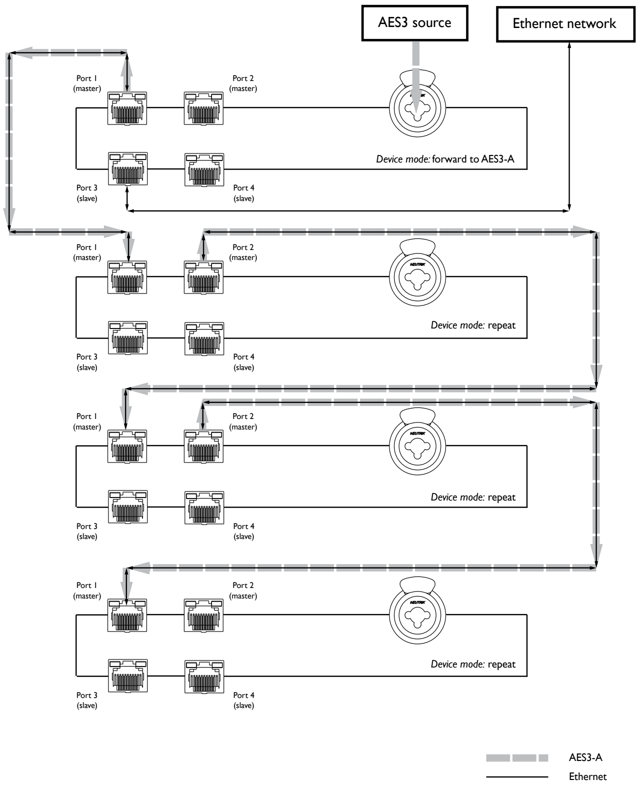


FIGURE 52: Daisy chain connection of four amplifiers with four RJ45 ports each

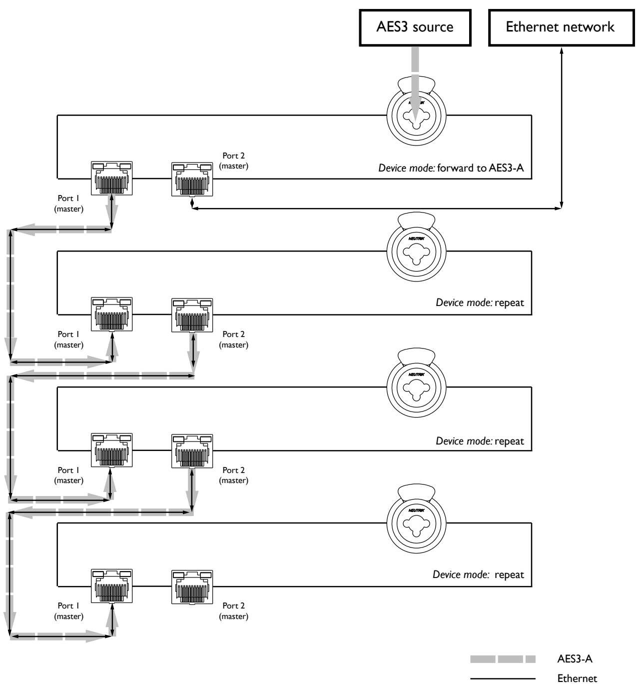


FIGURE 53: Daisy chain connection of four amplifiers with two frontal RJ45 ports each

The first amplifier in the chain receives the AES3 input from the rear panel XLR connector and then forwards it to the AES3-A (or, alternatively, the AES3-B) stream. In order to do so, the first amplifier mode is set to "forward to AES3-A stream". Instructions on how to set the amplifier mode can be found in section "10.1 Device Mode" on page 36. The second amplifier in the chain receives the AES3-A stream from the master port number 1. Set in repeater mode, this amplifier relays the AES3-A signal to the third amplifier in the chain via the RJ45 port number 2. This setup is repeated until the final amplifier in the chain receives its AES3-A signal. The first connection to the Ethernet network is done via a CAT-5 cable inserted in any free RJ45 port (FIGURE 53 shows port number 3 being used, but ports 2 or 4 could have been used instead. In FIGURE 54 the only free port is port number 2). The control data stream, travelling using the Ethernet standard, travels within the chain alongside the AES3-A stream in a bidirectional manner.

The daisy chain topology is not robust. If any single AES3 or Ethernet cable connection is interrupted, the whole system fails. In the diagram below, if the crossed out connection should fail, both amplifiers number 3 as well as 4 would not be able to receive any audio signal to play. Their connection to the Ethernet network would fail as well.

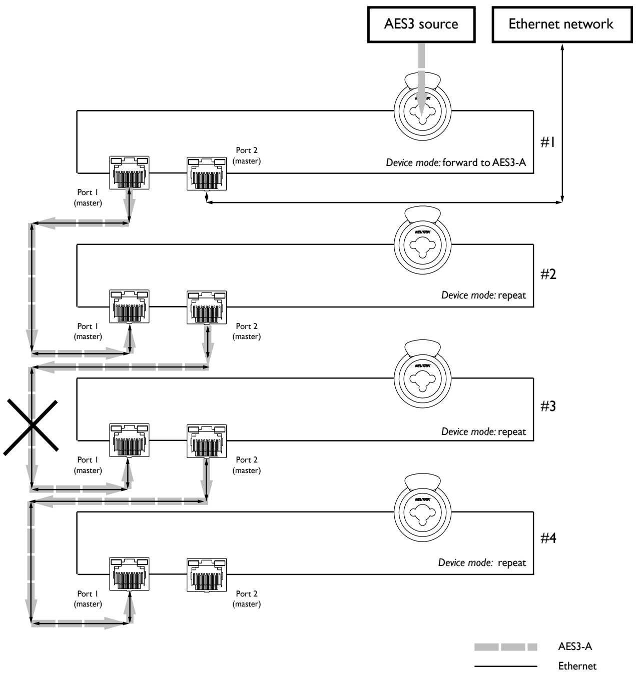


FIGURE 54: Daisy chain connection of four amplifiers with two frontal RJ45 ports each: case of internal connection failure between amps number 2 and 3

▸ **Intermediate audio robust chain**

A slightly more robust network with respect to the audio system is the one illustrated in the following diagram. In this connection, two amplifiers, the first and the last one in the network, are set to work in forward mode. The remaining "central amplifiers" are set to work in repeater mode.

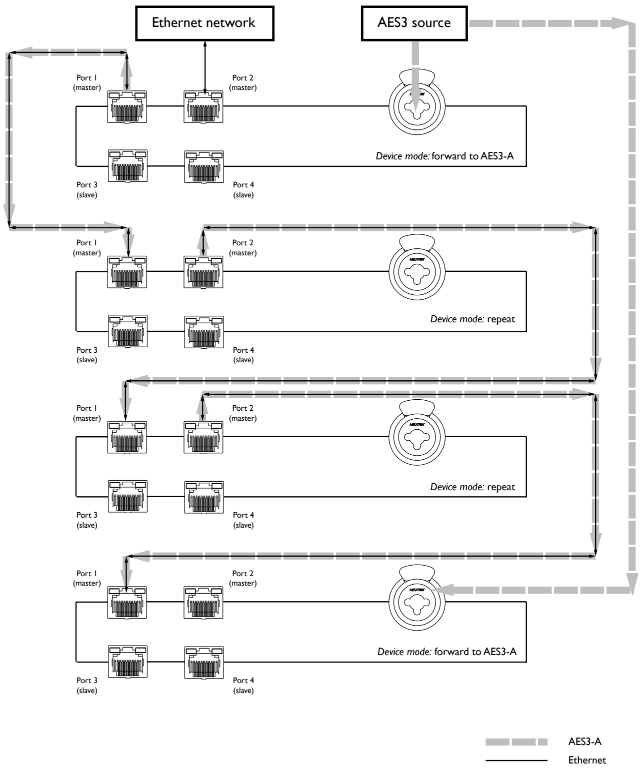


FIGURE 55: Intermediate connection, internally robust with respect to the AES3 stream. Four-port-amplifier diagram

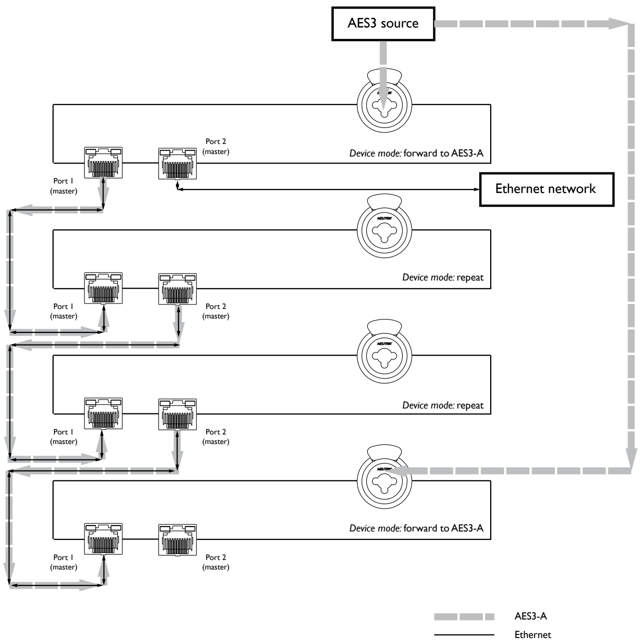


FIGURE 56: Intermediate connection, internally robust with respect to the AES3 stream. Two-port-amplifier diagram

The fourth amplifier's audio input is the AES3 stream coming from the XLR connector because it is in forward mode; the AES3-A stream coming from amplifier number 3 via master port 1 is redundant, meaning it is not necessary for the fourth amplifier to produce sound. The reason for this connection is to improve the robustness of the audio connection of amplifiers 2 and 3.

The system's connections could be interrupted in the following ways:

If the connection between amplifiers 2 and 3 should fail, the Ethernet network connection would be interrupted but not the audio stream. As a result of the interruption of the amplifier 2-amplifier 3 link, amplifier number 3 would stop receiving an incoming AES3 stream from amplifier number 2; amplifier number 4, however, would continue forwarding the AES3 stream into amplifier 3. This means that amplifier 3 would automatically sense a backup AES3 feed coming from amplifier 4.

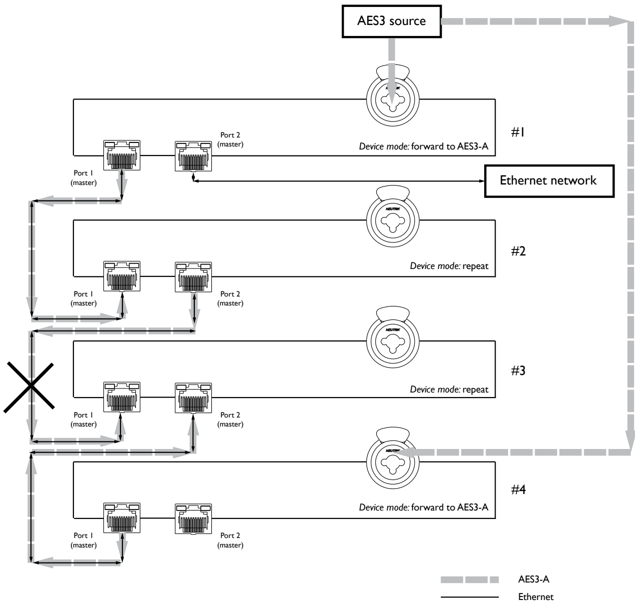


FIGURE 57: If the connection between amplifiers number 2 and 3 should fail, amplifier number 4 forwards a backup AES3 stream towards amplifier number 3 so no audio interruption can be heard.

If the connection between amplifiers 1 and 2 should fail, no sound interruption would be heard. In this case, amplifier number 4 would still forward its AES3 stream direction to amplifier 3. Amplifiers 3 would invert its repeater stream and feed the AES3 stream to amplifier number 2. Amplifier number 2 would therefore be able to continue to reproduce the AES3 stream this time coming from amplifier number 3 instead of amplifier number 1. Amplifier number 1 reproduces sound from the XLR rear panel AES3 source.

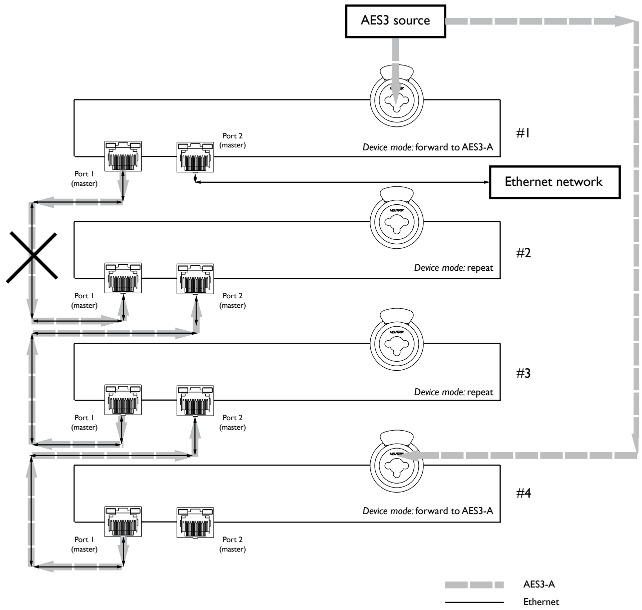


FIGURE 58: If the connection between amplifiers number 1 and 2 should fail, amplifiers number 4 and 3 would automatically act as a backup AES3 stream that can reach amplifier number 2 so no audio interruption can be heard.

The Ethernet network, however, would still be compromised.

If the connection between amplifier number 3 and 4 should fail, no audio interruption would be heard. Amplifier number 3 receives its incoming AES3 stream from amplifier number 2. The fourth amplifier reproduces sound coming directly from the AES3 source fed in its rear panel XLR connector.

The robustness of this network is guaranteed for AES3 signals only, and for a single cable failure at a time. If two or more connections should fail, one or more amplifiers (depending on where the interruption occurs) would be muted.

‣ Intermediate data robust chain

The audio signal robustness of the previous intermediate chain example is guaranteed by the double AES3 forward mode of the top and bottom amplifiers in the chain. The same cane be done for the Ethernet data connection, using an external switch capable of managing two (and more) Ethernet streams. The diagram below shows this configuration for both 4 and 2 -RJ45 ports amplifiers.

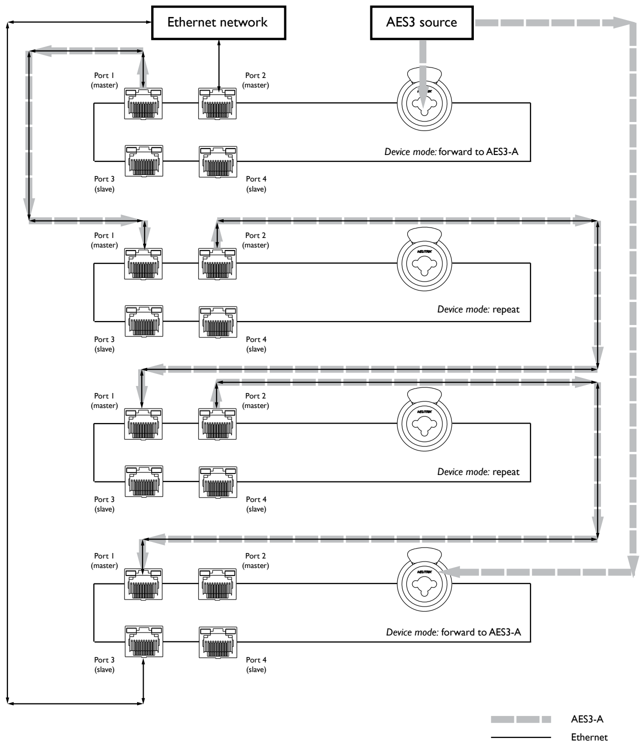


FIGURE 59: Intermediate connection, internally robust with respect to the AES3 stream. Four-port-amplifier diagram

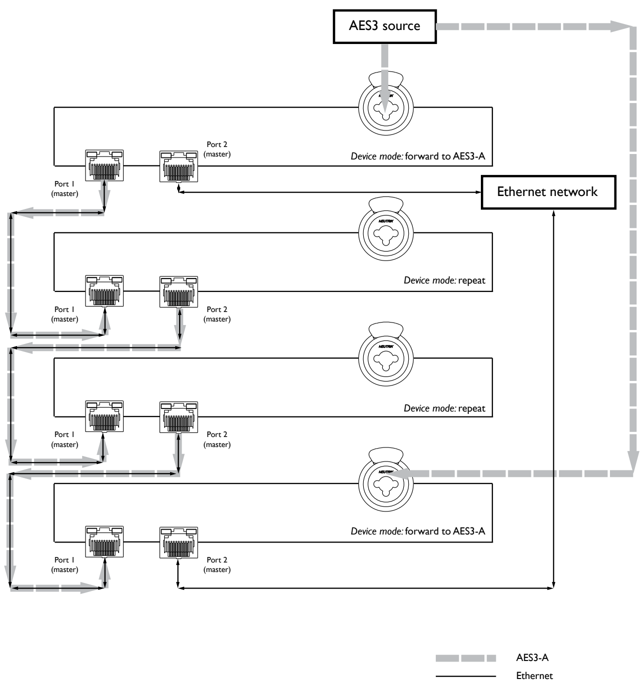


FIGURE 60: Intermediate connection, internally robust with respect to the AES3 stream. Two-port-amplifier diagram

10 KAESOP Network settings menu

In all the menus illustrating a series of possible alternatives that a specific amplifier parameter can have, a diamond shape will mark the value set for that specific parameter.

Many of the menus in this section require the user to select one functioning mode from a set of possible alternatives. These alternatives are all presented in a list. A black diamond shape next to a specific item in the list indicates that that is the selected option.

In FIGURE 61, for example, the selected device mode is "Forward to AES3-A" because the diamond shape appears next to it in the Device mode list.

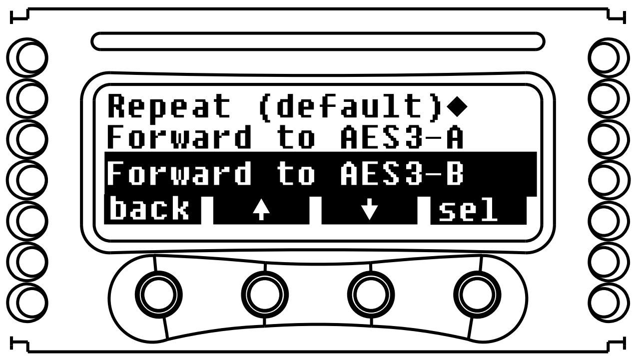


FIGURE 61: The black diamond next to the "Forward to AES3-A" item indicates that it is the currently set device mode

10.1 Device Mode

This parameter sets the amplifier mode with respect to the AES3 stream

- Repeater: the AES3 stream received on port 1 is repeated to port 2 and vice versa
- Forward to AES3-A: the AES stream from the rear panel XLR is rerouted to exit both RJ45 master ports as an AES3-A stream. The amplifier will be in repeater mode with respect to the AES3-B stream
- Forward to AES3-B: the AES stream from the rear panel XLR is rerouted to exit both RJ45 master ports as an AES3-B stream. The amplifier will be in repeater mode with respect to the AES3-A stream
- Forward to both: the AES stream from the rear panel XLR is rerouted to both AES3-A and AES3-B streams out both RJ45 master ports. The repeater mode is disabled in this configuration.

Note: when an amplifier is in forward mode (either to AES3-A, AES3-B or both) the amplifier can only accept the AES signal coming from the rear panel XLR connector. AES3 streams incoming from any other RJ45 port are ignored.

10.2 Addressing Mode

This parameter controls the IP addressing assignment strategy:

- "Manual" requires the user to set a valid static address and subnet mask (and, optionally, the default gateway). The PC should be on the same subnet of the amplifier if no routers are present between the PC and amplifier.
- "Automatic" lets the amplifier ask and obtain a network configuration from a DHCP server. Starting from power-on, the amplifier tries to obtain a valid IP address from a DHCP server. After a timeout of 30 seconds, if an IP address is not obtained, the amplifier takes an automatic private address in the range 169.254.x.y, but continues to search for a DHCP server. When the DHCP becomes available, the address is updated. If no DHCP server is available, the amplifier obtains an IP address by Automatic IP (local link addressing or

ZeroConf).

The amplifier behavior complies with RFC 3927, guaranteeing the interoperability with any host PC supporting this standard.

10.3 Set Address

This menu allows to manually set the amplifier's IP address, subnet mask and default gateway.

10.4 Show Net Config

This menu shows the current networking configuration, either set by the user via the "Set address" menu or obtained automatically if the automatic addressing mode is selected.

10.5 Audio

10.5.1 Source Selection

This menu allows the user to select the AES3 stream source to feed the output power stage of the amplifier. The AES3 signal can come from either:

- AES3 XLR: the rear panel XLR connector, while the "AES/EBU-Analog" pushbutton is in the "EAS/EBU" selected position
- AES3-A: the AES3-A stream coming from one of the two master RJ45 ports (either the two in the back of the amplifier for 4-port amps, or the two in front for amps with only two RJ45 jacks)
- AES3-B: the AES3-B stream coming from on of the two master RJ45 ports (either the two in the back of the amplifier for 4-port amps, or the two in front for amps with only two RJ45 jacks)

10.5.2 Source Mode

This menu allows to selects the channel(s) contained the selected AES stream to be forwarded to the output power stage of the amplifier. The possibilities are:

- Parallel from L: the left channel from the selected AES3 stream (see "8.2.1.1 Source Selection" on page 22) is forwarded to both amplifier channels
- Parallel from R: the right channel from the selected AES3 stream (see "8.2.1.1 Source Selection" on page 22) is forwarded to both amplifier channels
- Stereo: the right channel from the selected AES3 stream goes to channel 1 or the amplifier; the right channel from the AES3 stream goes to the amplifier's left channel.

10.5.3 Gain Trim

This parameter trims the digital level of the AES3 stream. The gain trim scale goes from +5dB to -40dB with 0.5dB steps. The 0dB gain trim level has an analog level equivalent of +13.5dBu. A 0dBFS level in the AES3 stream corresponds to an absolute analog level of +18.5dBu (with a +5dB gain trim level).

Note: please note that when using a digital input, the amplifier will keep a fixed 32dB gain.

10.5.4 If no link

This parameter allows the user to choose the behavior of the amplifier when the digital audio stream is missing and the Input Selection is set as KAESOP=>OUT. The two possible alternatives are:

- Mute: in this case the amplifier output is muted
- Analog: in this case the amplifier automatically switches to CH1/CH2 analog input if the digital stream is missing, returning to the digital stream in case this should become available again. This mode could be used to implement an analog backup connection for the digital stream.

11 Display

11.1 Output Meters

The output meters screen shows important output signal information for the amplifier. By pressing the right most front panel button, the screen view is toggled between information relative to channel 1, channel 2 or relative to the sum of channels 1 and 2[1] The top line in this screen displays the RMS voltage value of the output, both as a number as well as a horizontal meter bar. The second and third line display the output RMS current and power level respectively. The output power reported is a peak value reading taken every 200 ms. The bottom line of the screen displays the load impedance as "Zload". The minimum output voltage is stored internally and available to remote clients connected to the amplifier. The load impedance is indirectly inferred by a successive approximations. Time between single output impedance approximations depends on the output signal: the greater the amplitude of the signal, the shorter the time interval between measurements needed to approximate the output impedance, the faster the successive approximation method will converge to the true impedance value.

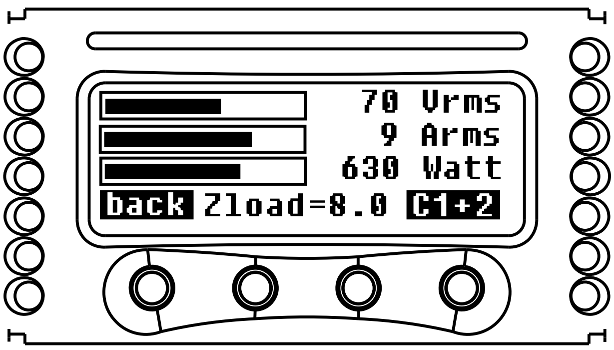


FIGURE 62: Output meters screen for a bridged channel 1/channel 2 connection. Measured load impedance in this example is 8 Ohms.

11.2 Temperature

This screen displays the current amplifier temperature.

1 Note: in the "C1+2" mode, the RMS voltage and power readings displayed are the average RMS voltage and peak power of each channel. The RMS current value, on the other hand, is the sum of each single channel's RMS current level.

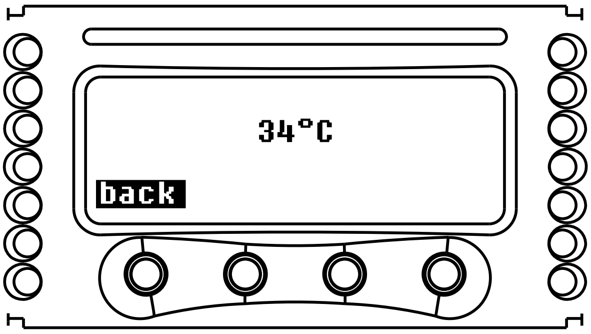


FIGURE 63: Current amplifier temperature

11.3 Mains meters

This screen displays the updated mains RMS voltage and RMS current levels. Values are displayed in numbers and as progress bars.

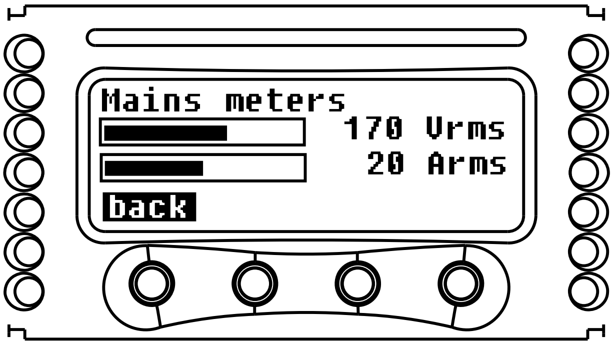


FIGURE 64: Updated mains RMS current and voltage values

PLEASE NOTE: the current and voltage levels displayed in this screen are approximate values: which serve the purpose of giving a general indication of the mains levels. Please refer to other sources (such as calibrated multimeters) for reliable and exact mains voltage and current measurements.

11.4 Amplifier Name

The "Amplifier Name" menu gives access to two menu branches: the "Display Amp data" function and the "Edit Amplifier name" menu.

When the "Display Amp data" function is activated, the main amplifier screen shows the amplifier name (20 characters, bold) blinking to a second screen showing the current selected preset name (40 characters). If the preset has been altered in any way, the displayed preset name will have a "Modified" prefix to indicate this.

The amplifier name can be assigned by entering the "Edit amplifier name" menu. For information regarding on-screen text editing, please see "12.4 Save local preset" on page 39.

12 Local presets

All K Series amplifiers have an on board memory capable of storing up to 50 presets. An amplifier preset is a snapshot of the current amplifier status, including the basic amplifier settings and the KDSP board settings if a DSP board is present.

12.1 Locked presets

When the "locked presets" function is active, a number of presets, determined by the "Locked bank size" menu, is not over writable. This function's status can be toggled on/off by entering the Lock code. For instructions on how to enter and edit text, please see section "12.4 Save local preset" on page 39.

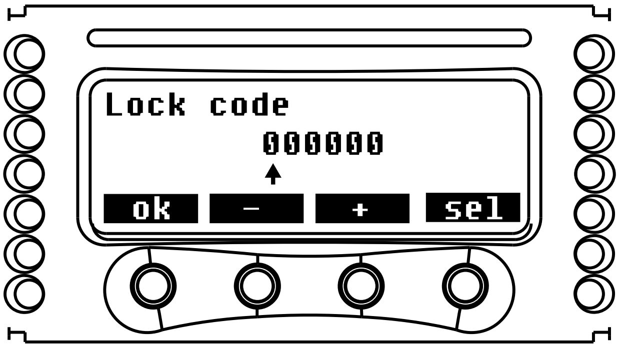


FIGURE 65: Enter the lock code. Select correct digit by using "+" and "-" buttons. Move to the digit to the right by pressing the "sel" button

If the wrong code is entered, the system simply returns to the previous local presets menu.

12.2 Locked bank size

This menu allows the user to set the number of locally stored presets that cannot be overwritten. Either all (50) or none (0) of the presets can be locked. After entering the correct lock code, select the number of presets to be write protected.

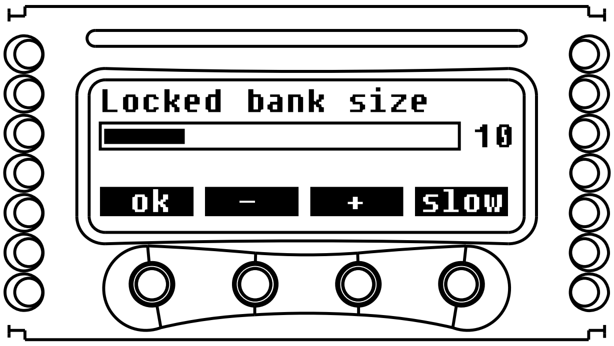


FIGURE 66: Change the number of locked presets by pressing the "+" or the "-" buttons. The slow editing mode uses one preset steps; the fast mode uses 10 presets steps.

When done, press the left most button labelled "ok" to return to the previous screen.

12.3 Recall local preset

In order to recall on of the 50 locally stored presets, press ok when the "Recall local preset" line is highlighted. Then use the middle buttons to navigate forwards or backwards in the existing presets list. If a preset number is not used, it is labelled <empty>. Once the desired preset has been found, press the right most button labelled "ok" to load it.

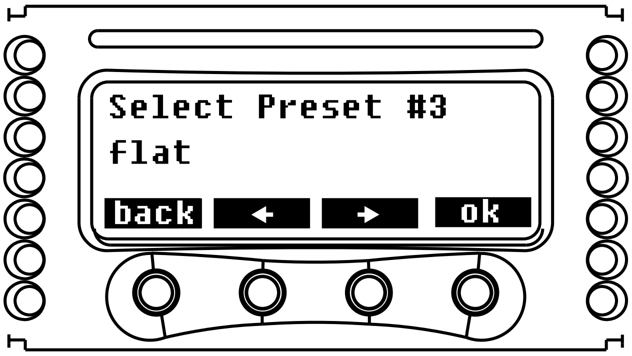


FIGURE 67: Preset number 3, named "flat" is selected. To load it, press "ok". Press "back" to return to the previous screen

Once the preset has been loaded correctly, press the left most button labelled "back" to return to the local presets menu.

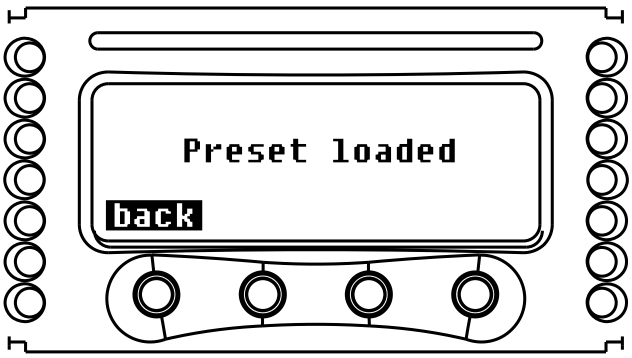


FIGURE 68: The chosen preset has been loaded correctly. The amplifier's current settings match those store in the loaded preset

12.4 Save local preset

Save to an empty slot

To save a current amplifier setup as a preset to the local memory, enter the "Save local preset" menu. Select an non used preset which is labelled empty:

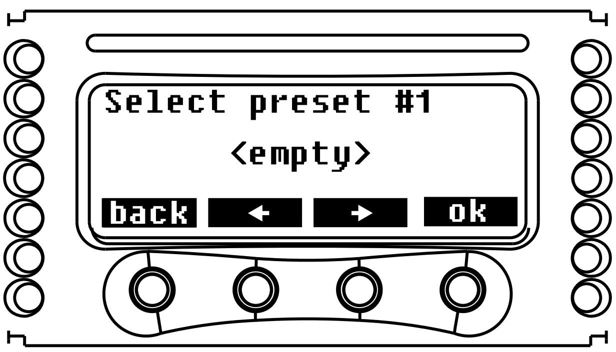


FIGURE 69: Preset memory location number 1 is empty

After pressing "ok", the user is asked whether to keep the current preset name or change it. The current preset name will be "PRESET" followed by the selected memory slot name if no other preset has been loaded in the amplifier either via remote control or using a SmartCard.

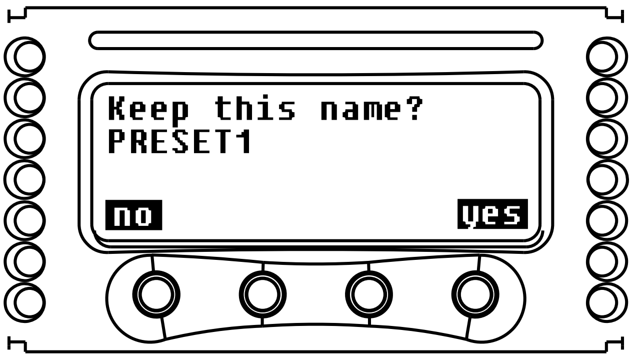


FIGURE 70: Pressing "yes" will write the current setup to preset number one and will name it "PRESET 1". Pressing "no" will allow the user to change the preset name

By pressing "no", the preset name can be edited. The preset name can be edited one character at a time. The arrow points towards the active character that is currently being edited. To move from one character to the next, press the "sel" button. The "+" and "-" buttons allow to navigate within a standard set of capital letters and basic punctuation marks.

```
ABCDEFGHIJKLMNOPQRST
UVWXYZ !"#$%&'()*+,-
./0123456789:;<=>?@
```

FIGURE 71: The list of available characters

When the preset has been correctly saved with the name entered by the user entered, a confirmation screen will appear (see FIGURE 72).

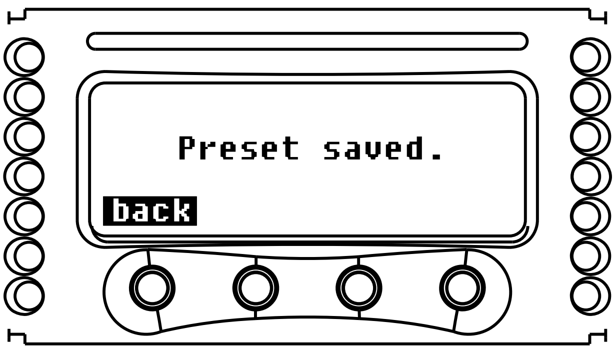


FIGURE 72: Preset saved confirmation screen

Overwriting an existing preset

If the preset location is not empty, the amplifier will ask the user confirmation to overwrite the file. Note that if you have already input a preset name, or if you have loaded a preset from local memory or a SmartCard, the name is used as starting point for a new save preset operation. For example, suppose that a preset named "18IN SUB 1" has been loaded from a SmartCard with the purpose of saving it in the amplifier's local memory in the preset slot number 3, as show in FIGURE 73:

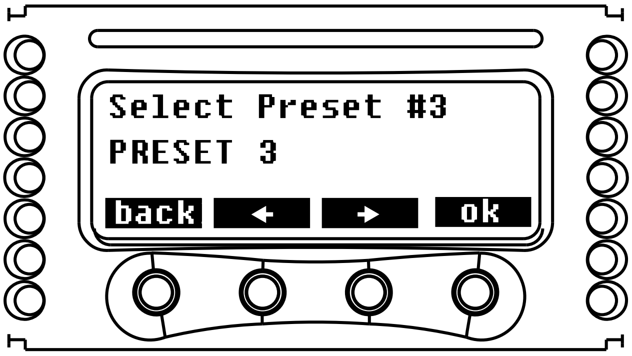


FIGURE 73: Preset slot number 3 being selected to accommodate the preset loaded from the SmartCard.

In this case the amplifier asks the user whether to keep the preset's name as loaded from the SmartCard or change it. This is useful for copying presets from/to SmartCard.

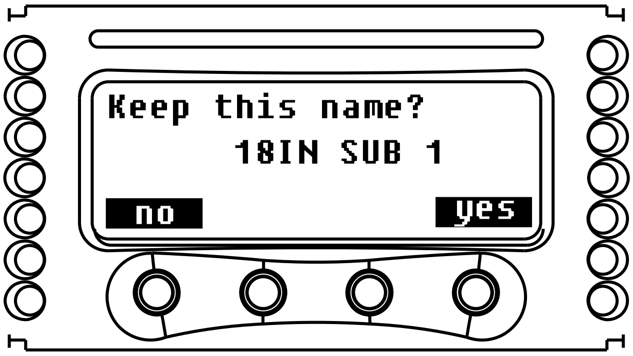


FIGURE 74: Keep this name confirmation screen

By pressing "no" the system will enter in a text editing mode, allowing the user to choose a preset name. For details on text editing, see section "Save local preset" on page 39. By pressing "yes", the user is prompted to confirm the intention of overwriting the preset.

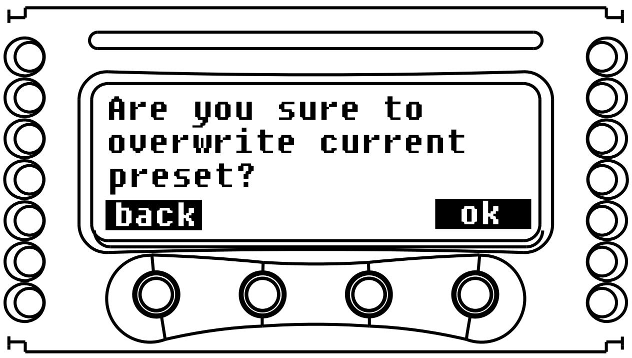


FIGURE 75: This screen prompts the user to confirm overwriting an existing preset slot

Pressing "ok" will confirm the overwrite. Once the preset has been correctly saved a screen will inform the user of this (see FIGURE 72). Pressing "back" will return the user to the previous screen to select another memory slot in which to save the current preset.

12.5 Change Lock Code

In order to change the lock code used to activate the "Lock presets" function, the old user code must be entered. Enter the code by following the text editing procedure described in the "Save Local Preset" section. Press "ok" when the code has been completely entered. If the entered code is correct, another screen will prompt the user to enter then new lock code. If the entered code is incorrect, the system returns to the previous screen. There is no limit on the number of times that an incorrect lock code can be entered.

12.6 Erase all presets

This function allows to erase all non write protected presets in the amplifier's internal memory. After having selected this function's submenu by pressing "ok", a confirmation screen will appear.

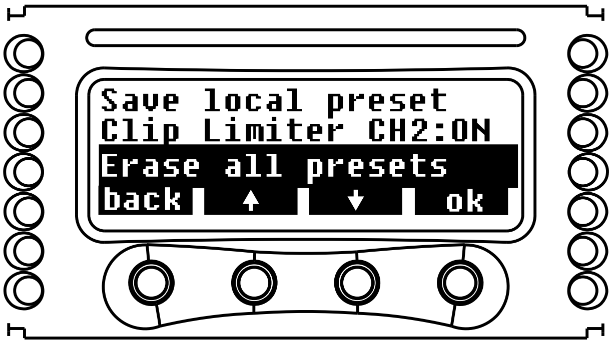


FIGURE 76: Press "ok" to select the "erase all presets" submenu

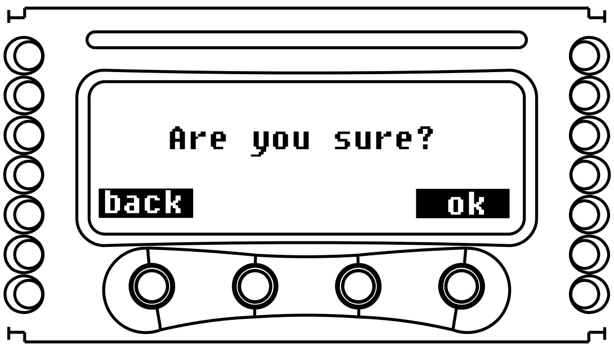


FIGURE 77: Erase all preset confirmation request

Pressing "ok" will erase all non protected presets. Pressing "back" will return the user to the previous screen. When all non write protected presets have been erased, a screen confirming this will appear.

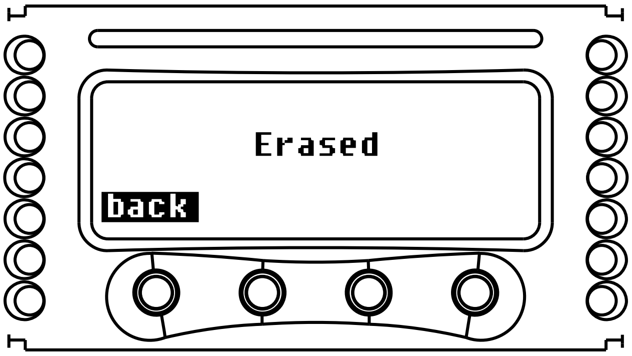


FIGURE 78: All presets have been erased

Please press "back" to return to the local presets menu.

13 Setup

13.1 Hardware info

This menu allows the user to access various information about the amplifier's hardware. The first screen shows the amplifier name followed by:

- S/N: serial number of the amplifier
- Hw ID: hardware ID, selectable via the rotary encoders on the back panel

Pressing the "more" button on the screen allows to cycle through a greater number of pages containing more information; the "back" button will bring the user back to the previous setup menu.

- KFRNT: this is the front panel version
- KCTRL: controller version number
- KDSP: DSP board version number (available only for models with the optional DSP board)
- KAESOP: network board version number (available only for models with the optional KAESOP board)
- Lifetime: operating hours of the amplifier.

13.2 Hardware monitor

This menu allows the user to access information about the current amplifier system parameters. These are:

- PWRBSCH1: amplifier's power supply voltage for channel 1
- PWRBSCH2: amplifier's power supply voltage for channel 2

Pressing the "more" button on the screen allows to cycle through a greater number of pages containing more information; the "back" button will bring the user back to the previous setup menu.

- VAUX: internal auxiliary voltage
- +5VAN: auxiliary analog voltage
- VEXT: external remote control voltage
- VAUX: indicates if the power supply auxiliary voltage is correct
- IGBTCONV: indicates the DC/DC converter monitor status
- VBOOST: internal post PFC voltage
- 192KHZ: system clock frequency status

13.3 LCD contrast

This screen allows the user to set the LCD display contrast using the "+" and "-" buttons.

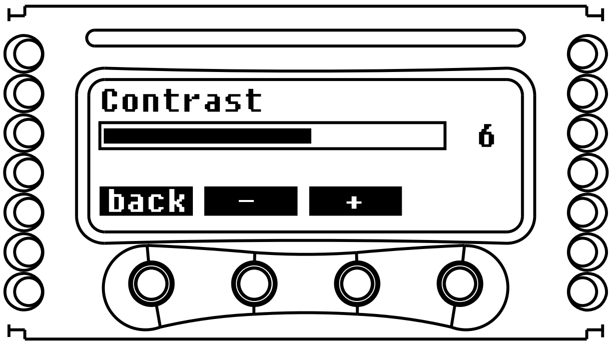


FIGURE 79: Setting screen for the LCD display contrast

13.4 Key Locking and Setting The Keylock Code

In order to prevent the amplifier's settings from being altered by acting on the front panel commands, the "lock" function can be activated if the corresponding button is held pressed for more than 1 second; in this case all other buttons are locked. Unlocking buttons is done in the same way, but an unlock code is required for security reasons. In order to enter an unlock code for the amplifier, select the "Set Keylock Code" from the Setup Menu. Please not that this screen can also be accessed by pressing the "unlock" button in the main screen when the amp is in locked key mode. Using the two central buttons, chose and set an unlock code. Pressing the right most key (labelled "sel") allows to select the desired digit.

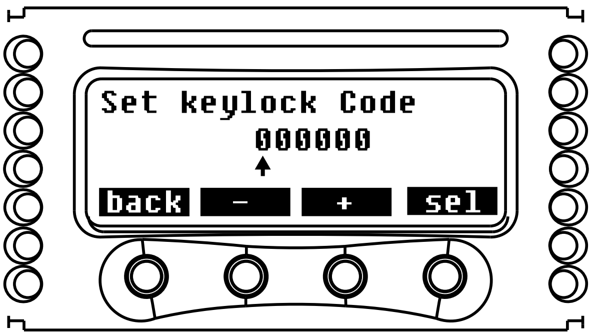


FIGURE 80: Setting the keylock code screen

13.5 Single Channel Muting

Muting of one channel at a time can be done via the "mute" function directly from the amplifier's front panel. Pressing the button directly below the "mute" label can mute each channel individually; in this case, the on screen channel-specific parameters are replaced by the "muted" label. Unmuting the channels is done by pressing the "mute" button again.

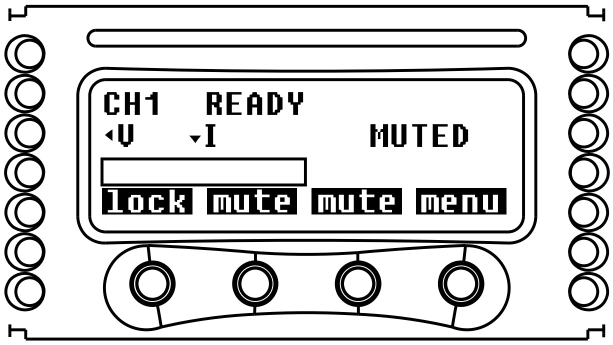


FIGURE 81: Example of a muted right channel. To unmute the channel, press the right most "mute" button again

14 Protection

In order to protect your device and your speakers from accidental damage, K Series amplifiers include an extensive automatic protection system. In the following sections, potentially dangerous scenarios and the amplifiers' corresponding protective response are explained in detail.

14.1 Turn-On/Turn-Off muting

For about four seconds after turn-on, and immediately at turn-off, the amplifier outputs are muted. Class D amplifiers may cause severe speaker damage at power up due to the high voltage levels at the output stage. In order to avoid this, the outputs are muted for about 4 seconds after turn on. Similarly, turning off the amplifier can cause the same problem: outputs are muted immediately at turn off.

14.2 Short circuit protection

Short circuits or very low impedance loads may destroy the output stage of any amplifier. In order to protect the amplifier from the dangerously high current surges arising from accidental output short circuits or low impedance loads, all K series amps block channel activity when the current drawn from the load rises above a set value. In case of short circuit, the topmost front panel red LED will light-up. At the same time, the "PROT" warning appears in the first line of the LCD display. The channel is muted for 2 seconds after which the amplifier will check whether the current draw is still over the safety threshold. Should this be the case, the amplifier will mute the channel for another 2 seconds and the procedure will reiterate. The amplifier will therefore automatically self-reset the channel every 2 seconds. Once the amplifier channel has undergone 50 resets and the output current draw is still above safe limits, the channel enters a permanent protection mode: an on/off cycle is needed to restart the unit and restore it to full functioning mode. The red LED will be turned off and the amplifier will return to normal operating conditions only when the output current draw returns to acceptable levels.

14.3 Thermal protection

All K Series amplifiers use a continuously variable speed fan to assist cooling (the fan speed changes in response to the amplifier's cooling needs). If the heat sink temperature reaches approximately 80°C, the yellow front panel LED starts blinking. If the temperature should rise above 85°C the thermal sensing circuitry will mute each power section channels, the yellow LED will be steadily on, and the power supply will be cut off. At the same time, the "PROT" warning appears in the first line of the LCD display. Once the heatsink has cooled down, the amplifier will automatically reset and the yellow LED will go off. One possible way to reduce the temperature is to reduce the output power.

14.4 DC fault protection

In order to protect your speakers from mechanical damage caused by a DC signal coming from the amplifier's output, a DC detection circuit is placed between the amplifier's output stage and power supply. If a DC signal or excessive subsonic energy appears at a channel output an instantaneous protection circuit will cut off the power supply to both channels. Power supply shutdown is used instead of speaker relays in order to improve the damping factor and reliability of K Series amplifiers. At the same time, the "PROT" warning appears in the first line of the LCD display.

14.5 Input/Output protection

An ultrasonic network decouples radio frequencies from the outputs keeping the amplifier stable with reactive loads and protects the loudspeakers against strong very high frequency non-musical signals above the audible range.

15 User Maintenance

15.1 Cleaning

Before attempting to clean any part of the amplifier, first disconnect the AC main source. Use a soft cloth and mild nonabrasive solution to clean the faceplate and chassis. **WARNING! Never let any liquid reach the internal parts of the amplifier.**

15.2 Service

There are no user-serviceable parts in your amplifier. Refer servicing to qualified technical personnel. In addition to having an in-house service department, Powersoft supports a network of authorized service centers. If your amplifier needs repair contact your Powersoft dealer (or distributor). You can also contact the Powersoft Technical Service department to obtain the location of the nearest authorized service center.

15.3 Dust Removal

In dusty environments, the front side air filters clog with dust after prolonged use. The dust gathered in the filters will interfere with cooling. You may use compressed air to remove the dust from filters. To remove air filters:

1. first unscrew the 2 M2 5X8 screws at the sides of the frontal panel
2. rotate the covering grill

3. repeat the same operation for the other covering grill

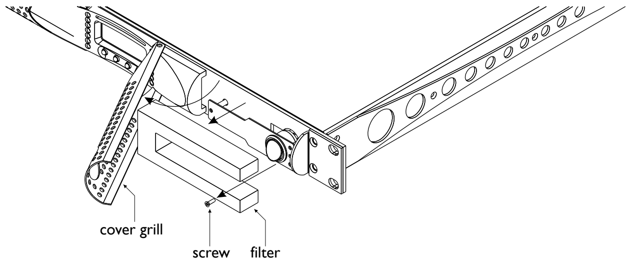


FIGURE 82: Air filter removal

Air filter cleaning should be scheduled according to the dust levels in the amplifier's operating environment.

16 Warranty

Product Warranty:

Powersoft guarantees its manufactured products to be free from defective components and factory workmanship for a period of 48 (forty eight) months, starting from the date of purchase printed on Powersoft's (or any of its Authorized Dealer's) invoice to the end customer. All warranty repairs and retrofits must be performed at Powersoft facilities or at an Authorized Service Center at no cost for the purchaser. Warranty exclusion: Powersoft's warranty does not cover product malfunctioning or failure caused by: misuse, abuse, repair work or alterations performed by non-authorized personnel, incorrect connections, exposure to harsh weather conditions, mechanical damages(including shipping accidents), and normal wear and tear. Powersoft will perform warranty services provided that the product is not damaged during transportation.

Return of Goods:

Goods can be returned to Powersoft only after they have been granted a Return Merchandise Authorization (RMA) number to be attached to the external packaging. Powersoft (or its Authorized Service Center) has the right to refuse any returned good without a RMA number.

Repair or replacement:

Powersoft reserves the right to repair or replace any defective goods covered by product warranty at its sole discretion and as it deems best. Cost and responsibility of transport: The purchaser (or end user/customer) is solely responsible for all transportation costs and risks associated with sending warranty covered goods to Powersoft or its Authorized Service Center. Powersoft will assume full responsibility and cover all costs incurred to send the goods back to the purchaser (or end user/ customer).

17 Assistance

Even though most product malfunctioning can be solved at your premises through Powersoft Customer Care or your direct knowledge, occasionally, due the nature of the failure, it might be necessary to return defective products to Powersoft for repair. In the latter case, before shipping, you are kindly asked to follow step by step the procedure described below: Obtain the "Defect Report Form" contacting our Customer Care Department via email:

service@powersoft.it

or download the "Defect Report Form". Fill out one "Defect Report form" for each returned item (the form is an editable tab guided document) and save as your name, amp model and serial Number (for example: distributornamek10sn17345.doc) providing all required information except the RMA code/s and send it to service@ powersoft.it for Powersoft approval. In case of defect reports approved by the Powersoft Customer Service Representative you will receive an RMA authorization code (one RMA code for each returning device). Upon receiving the RMA code you must package the unit and attach the RMA code outside the pack, protected in a waterproof transparent envelope so it is clearly visible. All returning items must be shipped to the following address:

Powersoft srl

Via Enrico Conti, 13-15

50018 Scandicci (FI) Italy

In case of shipment from countries NOT belonging to the European Community make sure you have also followed the instructions described in the document available for download at the following link:

http://www.powersoft-audio.com/en/component/docman/doc_download/298-temporary-export-import-procedure.html?Itemid=111

TEMPORARY EXPORTATION / IMPORTATION PROCEDURE

Thank you for your understanding and cooperation and continued support as we work to improve our partnership.

18 Appendix

18.1 Custom Ethernet/AES3 combo box

It is possible to build a custom box to combine the Ethernet signal and AES3 signal/s in a single RJ45 connector. This makes it possible to avoid the using amplifiers in a network in forwarding mode. This increases system robustness, as an amplifier in forward mode can receive input only from the rear panel XLR; on the other hand, the repeater mode allows the amplifier to reroute its incoming signal automatically from either one of two master ports (see "Network Operations" on page 29).

Following the AESOP standard RJ45 pin out illustrated in the "Ethernet Connection" on page 15, the following diagram shows the pin-out of the adapter box.

Please note that the maximum cable length from the AES3 source that allows reliable connections is:

- 90m for Ethernet standard
- 250m for AES3 standard

on a Cat 5e/Cat 6 cable. If a mixed AES3/Ethernet standard is used, the maximum cable length is the more restrictive of the two standards (90m).

18.2 Amplifier Error Codes

The error code value displayed in the main screen is the sum of the single error code values. An "error" occurs when the following voltage values or power conditions fall outside normal ranges.

Error Code	Error Description
1	192 kHz clock not present
2	Positive 15V aux
4	Negative 15V aux
8	Positive 5V analog
100	Negative power bus CH1
200	Negative power bus CH2
2000	Positive power bus CH1
4000	Positive power bus CH2
8000	External auxiliary voltage
Check rail fuses	Check rail fuses CH1 and CH2

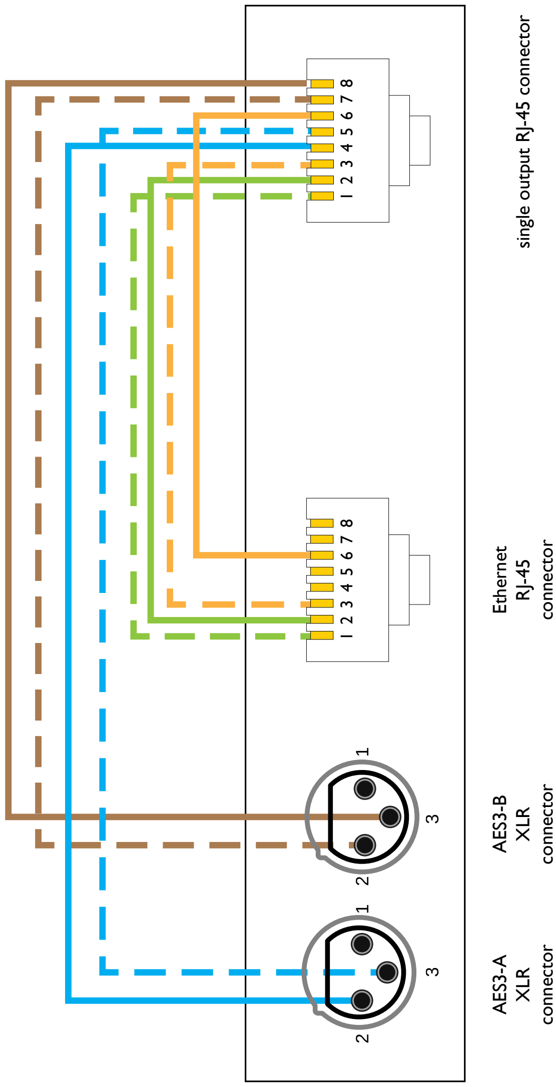
For example:

4301=4000+200+100+1 means there are 4 distinct errors

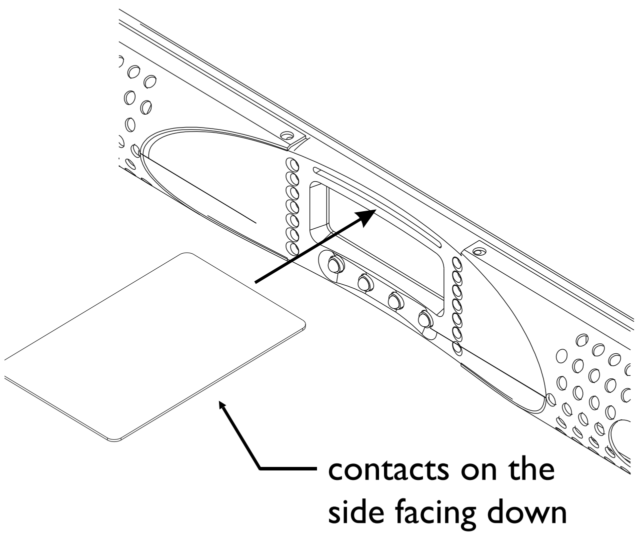
- Positive power bus CH2
- Negative power bus CH2
- Negative power bus CH1
- 192KHz clock not present

18.3 SmartCard function

A maximum of 150 presets can be easily stored in a standard-sized SmartCard. Please note that SmartCards for K Series presets must be initialized by Powersoft. In order to access the SmartCard menu, simply insert the card in the amplifier as shown in APPENDIX FIGURE 2.

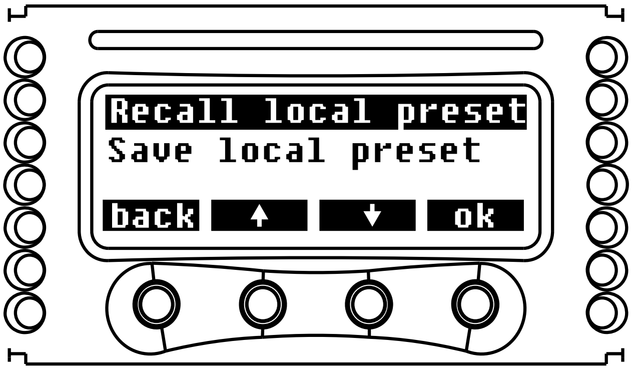


APPENDIX FIGURE 1: Pin out diagram for female connectors in a custom Ethernet/AES3-A/B box



APPENDIX FIGURE 2: How to insert the SmartCard in the amplifier's front slot

The main SmartCard menu will allow the user to save or recall presets stored on the card. Please see "12 Local presets" on page 38 for instructions on how to store and load presets in K series amplifiers.



APPENDIX FIGURE 3: The main SmartCard menu is visible when the card is inserted in the amplifier's frontal slot

Please note: if the inserted card is blank, the "Recall local preset" option will not be displayed in the main SmartCard menu.

18.4 Control Software

18.4.1 Powersoft's Armonía Pro Audio Suite

Armonía is Powersoft's application software specifically designed to be used with K Series amplifiers. Communication between the software and the amplifier is established via an RS-485 or Ethernet connection, depending on the available ports.

Armonía
Pro Audio Suite™

Armonía provides control and monitoring of a wide range of amplifier functions, such as attenuation, mute, internal temperature and voltage rail monitoring. Amplifiers in the K Series may also be fitted with the optional Powersoft DSP card allowing Armonía to control a greater range of features, including input and output equalization, alignment delays, FIR filters and load impedance monitoring.

Armonía is scalable: it allows control of a single Powersoft amplifier or a very large system containing many amplifiers. For large fixed or touring installations, Armonía gives the operator the ability to monitor and control all amplifiers in the system from a single location, regardless of the amplifiers' positions.

This software has been designed to accept software plug-ins to enable third-party product control. Further information is available on the website (www.armoniasuite.com) as plug-ins become available.

Powersoft amplifiers can connect to a PC running Armonía in two ways: with an RS-485 serial connection or via Ethernet. K Series amplifiers can be equipped with either or both methods of connectivity. Please bear in mind that Ethernet is a faster communications protocol than serial RS-485. Systems employing both categories of amplifiers may use both methods simultaneously: an Ethernet network being implemented for some amplifiers, and RS-485 for the others. The range of network topologies which can be used in wiring a real system varies between the two communications methods. Ethernet provides a slightly greater degree of freedom, as standard IT network switches may be used to create multiple hub-and-spoke systems as well as a single hub-and-spoke and linear daisy-chaining. A looped Ethernet topology is also permissible, which will provide redundancy in the event of a network failure. An amplifier system using an RS-485 network can either be daisy-chained throughout or use the Powersoft PowerHub as a local switch.

Armonía is free. It can be downloaded after signing up for our user forum: see the "Armonía Support Forum" section at

www.powersoft-audio.com

18.4.2 Third Party Controls

The K Series provides plug-ins for different third party controls, developed for Powersoft by independent consultants specialized in systems integration designs. These plug-ins provide integrated monitoring and control of K Series amplifiers, leading to the possibilities of setting up complex integrated systems. These plug-ins can be downloaded from www.powersoft-audio.com at our "Software/Third Party Plug-Ins" section.

19 Technical Specifications

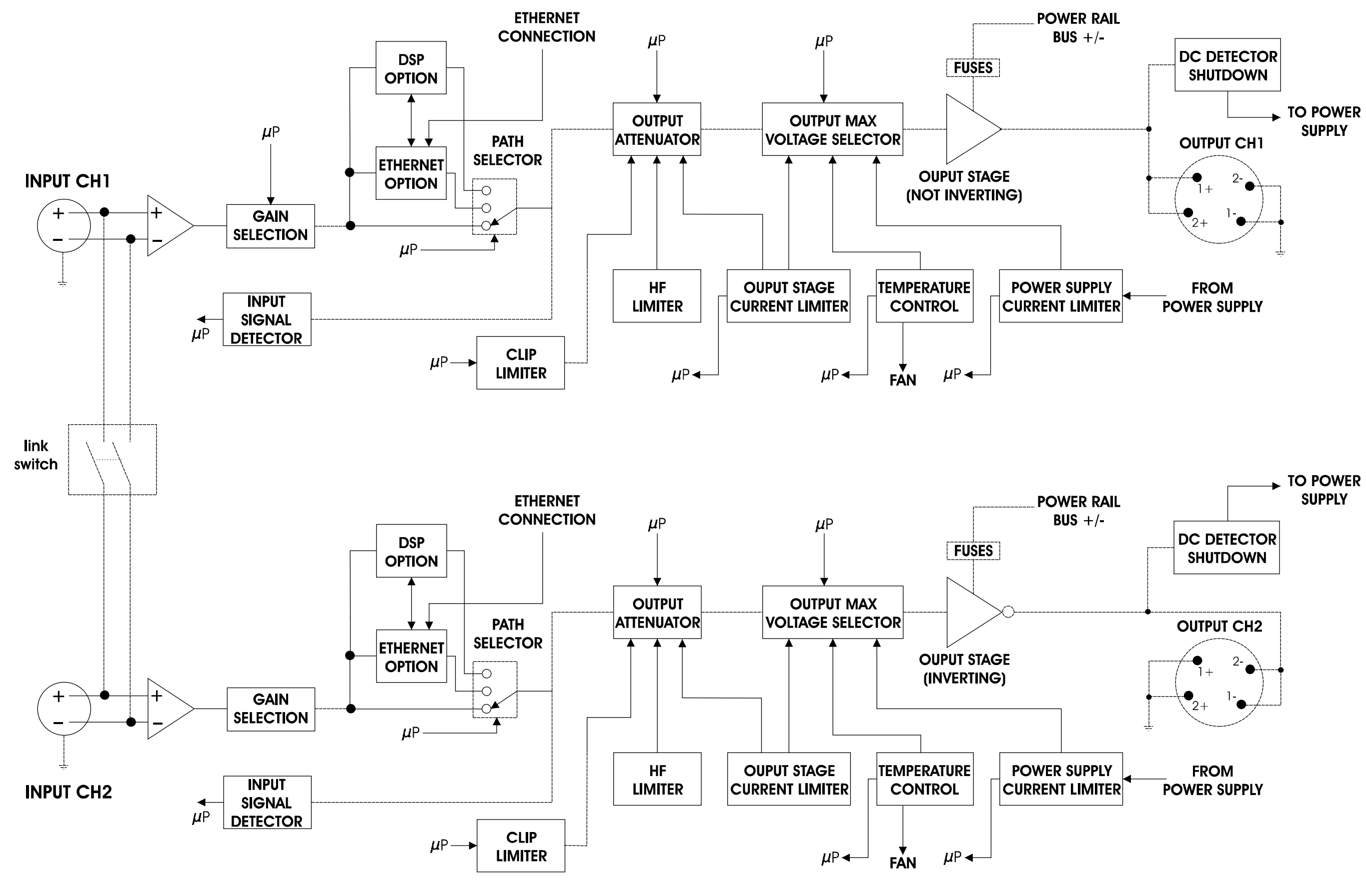


FIGURE 83: Output stage block diagram for all K Series amplifiers

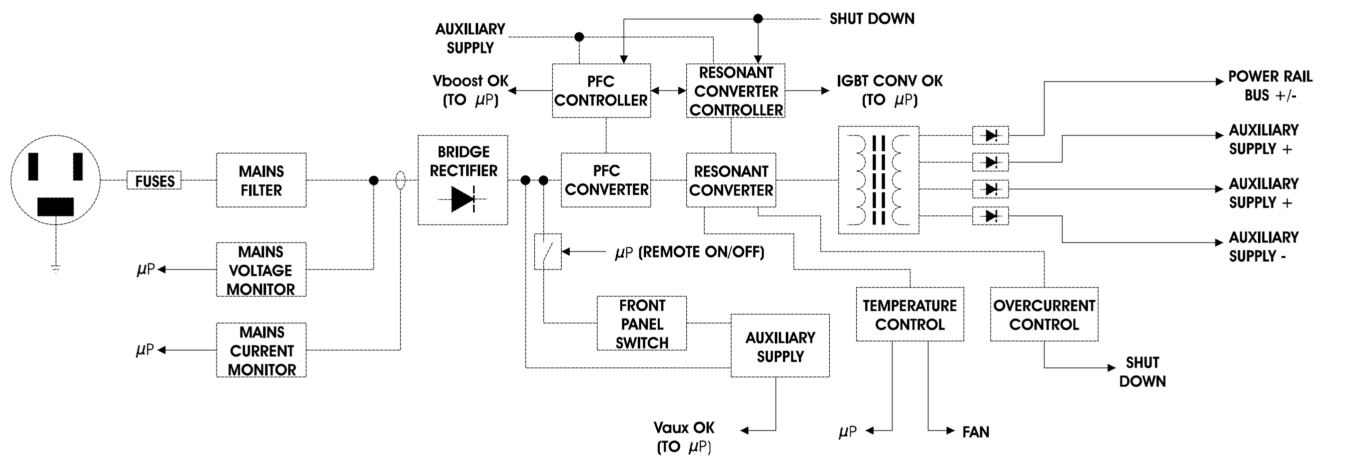


FIGURE 84: Power supply block diagram for all K Series amplifiers

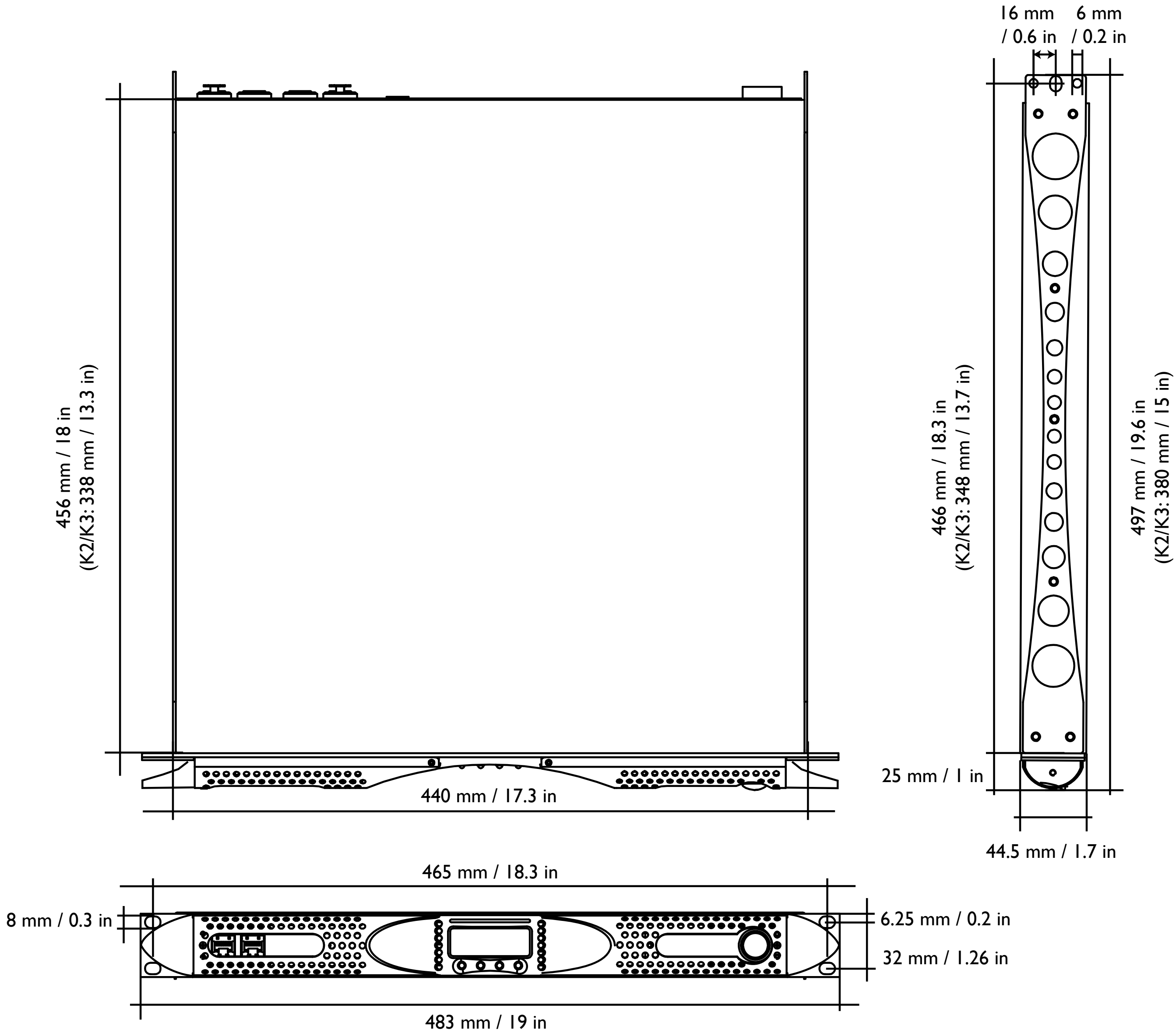


FIGURE 85: K Series dimensions

19.1 K2

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		2,400 W	1,950 W	1,000 W	4,800 W	3,900 W
	Max output voltage / current	140 V_{peak} / 102 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	75 W	1.3 A	64 W	1.12 A	
	1/8 of max output power @ 4 Ω	609 W	3.1 A	609 W	6.3 A	
	1/4 of max output power @ 4 Ω	1,219 W	5.7 A	1,219 W	11.4 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	382 BTU/h	96 kcal/h			
	1/8 of max output power @ 4 Ω	722 BTU/h	182 kcal/h			
	1/4 of max output power @ 4 Ω	1,062 BTU/h	268 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	4.48 V	3.17 V	2.47 V	1.59 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>106dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 70dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.2% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/µs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu; mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easy accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® XLR female; AES3: use channel 2 XLR				
	Audio signal output connectors	Analog: 2 x balanced Neutrik® XLR male				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	IEC20A with IEC20A Schuko for EU, IEC20A/American 15 A pin plug				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 380 mm / 15"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	8 kg (17.7 lb)				

19.2 K2 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		2,400 W	1,950 W	1,000 W	4,800 W	3,900 W
	Max output voltage / current	140 V_{peak} / 102 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	88 W	1.35 A	69 W	1.2 A	
	1/8 of max output power @ 4 Ω	609 W	3.1 A	609 W	6.3 A	
	1/4 of max output power @ 4 Ω	1,219 W	5.7 A	1,219 W	11.4 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	382 BTU/h	96 kcal/h			
	1/8 of max output power @ 4 Ω	722 BTU/h	182 kcal/h			
	1/4 of max output power @ 4 Ω	1,062 BTU/h	268 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	4.48 V	3.17 V	2.47 V	1.59 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>106dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 70dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.2% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu; mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® XLR female; AES3: use channel 2 XLR				
	Audio signal output connectors	Analog: 2 x balanced Neutrik® XLR male				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port Ethernet	2 x RJ45 with activity LEDs				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	IEC20A with IEC20A Schuko for EU, IEC20A/American 15 A pin plug				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 380 mm / 15"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	8 kg (17.7 lb)				

19.3 K3

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		2,800 W	2,600 W	1,400 W	5,600 W	5,200 W
	Max output voltage / current	165 V_{peak} / 102 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	75 W	1.3 A	64 W	1.12 A	
	1/8 of max output power @ 4 Ω	813 W	4 A	813 W	8 A	
	1/4 of max output power @ 4 Ω	1,625 W	7.4 A	1,625 W	14.8 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	382 BTU/h		96 kcal/h		
	1/8 of max output power @ 4 Ω	836 BTU/h		211 kcal/h		
	1/4 of max output power @ 4 Ω	1,390 BTU/h		326 kcal/h		
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	5.30 V	3.75 V	2.66 V	1.88 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>106dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 70dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.3% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu; mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® XLR female; AES3: use channel 2 XLR				
	Audio signal output connectors	Analog: 2 x balanced Neutrik® XLR male				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	IEC20A with IEC20A Schuko for EU, IEC20A/American 15 A pin plug				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 380 mm / 15"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	8 kg (17.7 lb)				

19.4 K3 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		2,800 W	2,600 W	1,400 W	5,600 W	5,200 W
	Max output voltage / current	165 V_{peak} / 102 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	88 W	1.35 A	69 W	1.2 A	
	1/8 of max output power @ 4 Ω	813 W	4 A	813 W	8 A	
	1/4 of max output power @ 4 Ω	1,625 W	7.4 A	1,625 W	14.8 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	382 BTU/h	96 kcal/h			
	1/8 of max output power @ 4 Ω	836 BTU/h	211 kcal/h			
	1/4 of max output power @ 4 Ω	1,390 BTU/h	326 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	5.30 V	3.75 V	2.66 V	1.88 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>106dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 70dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.3% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters total				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu; mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® XLR female; AES3: use channel 2 XLR				
	Audio signal output connectors	Analog: 2 x balanced Neutrik® XLR male				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port Ethernet	2 x RJ45 with activity LEDs				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	EC20A with IEC20A Schuko for EU, IEC20A/American 15A pin plug				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 380 mm / 15"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	8 kg (17.7 lb)				

19.5 K6

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		3,600 W	2,500 W	1,300 W	7,200 W	5,000 W
	Max output voltage / current	153 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	84 W	1.03 A	91 W	1.11 A	
	1/8 of max output power @ 4 Ω	781 W	4.1 A	781 W	8.2 A	
	1/4 of max output power @ 4 Ω	1,563 W	7.4 A	1,563 W	14.8 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	546 BTU/h	138 kcal/h			
	1/8 of max output power @ 4 Ω	982 BTU/h	248 kcal/h			
	1/4 of max output power @ 4 Ω	1,419 BTU/h	358 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	5.11 V	3.62 V	2.56 V	1.81 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/µs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm^2 (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	12 kg (26.5 lb)				

19.6 K6 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		3,600 W	2,500 W	1,300 W	7,200 W	5,000 W
	Max output voltage / current	153 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	84 W	1.17 A	91 W	1.3 A	
	1/8 of max output power @ 4 Ω	781 W	4.1 A	781 W	8.2 A	
	1/4 of max output power @ 4 Ω	1,563 W	7.4 A	1,563 W	14.8 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	546 BTU/h	138 kcal/h			
	1/8 of max output power @ 4 Ω	982 BTU/h	248 kcal/h			
	1/4 of max output power @ 4 Ω	1,419 BTU/h	358 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	5.11 V	3.62 V	2.56 V	1.81 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port Ethernet	2 x RJ45 with activity LEDs				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm^2 (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	12 kg (26.5 lb)				

19.7 K8

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		4,800 W	3,000 W	1,500 W	9,600 W	6,000 W
	Max output voltage / current	169 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	84 W	1.03 A	91 W	1.11 A	
	1/8 of max output power @ 4 Ω	938 W	4.8 A	938 W	9.5A	
	1/4 of max output power @ 4 Ω	1,875 W	8.7 A	1,875 W	17.4 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	546 BTU/h	138 kcal/h			
	1/8 of max output power @ 4 Ω	1,069 BTU/h	270 kcal/h			
	1/4 of max output power @ 4 Ω	1,593 BTU/h	402 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	5.50 V	3.90 V	2.75 V	1.95 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm^2 (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				

19.8 K8 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		4,800 W	3,000 W	1,500 W	9,600 W	6,000 W
	Max output voltage / current	169 V_{peak} / 125 A_{peak}				

AC Mains Power					
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)			
	Operating voltage	100-240 V ±10%, 50/60 Hz			
	Power factor cos (φ)	>0.95 @ >500 W			
	Consumption / current draw	@ 230 V		@ 115 V	
	Idle	84 W	1.17 A	91 W	1.3 A
	1/8 of max output power @ 4 Ω	938 W	4.8 A	938 W	9.5A
	1/4 of max output power @ 4 Ω	1,875 W	8.7 A	1,875W	17.4 A
Thermal					
	Environmental operating temperature	0° - 45° C / 32° - 113° F			
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow			
	Idle	546 BTU/h		138 kcal/h	
	1/8 of max output power @ 4 Ω	1,069 BTU/h		270 kcal/h	
	1/4 of max output power @ 4 Ω	1,593 BTU/h		402 kcal/h	
Audio					
	Gain, selectable	26dB	29dB	32dB	35dB
	Input Sensitivity @ 8 Ω	5.50 V	3.90 V	2.75 V	1.95 V
	Max input level	27dBu	24dBu	21dBu	18dBu
	Gate	-52dBu	-55dBu	-58dBu	-61dBu
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)			
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)			
	Crosstalk separation	> 66dB @ 1 kHz			
	Input Impedance	10 k Ω balanced			
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)			
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed			
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz			

DSP		
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)
	Memory	8 MB (RAM) plus 2 MB (flash for presets)
	Presets	50 stored locally + 150 stored on a smartcard
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter
Front Panel		
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel
	Controls	4 pushbuttons, function depending on user menu
	Power switch	Mains switch
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers
Rear Panel		
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD
	Network data port Ethernet	2 x RJ45 with activity LEDs
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm² (10AWG) cable
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch
Construction		
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support
	Weight	12 kg (26.5 lb)

19.9 K10

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		6,000 W	4,000 W	2,000 W	12,000 W	8,000 W
	Max output voltage / current	200 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	84 W	1.03 A	91 W	1.11 A	
	1/8 of max output power @ 4 Ω	1,250 W	6.1 A	1,250 W	12.2 A	
	1/4 of max output power @ 4 Ω	2,500 W	11.3 A	2,500 W	22.6 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	546 BTU/h		138 kcal/h		
	1/8 of max output power @ 4 Ω	1,244 BTU/h		314 kcal/h		
	1/4 of max output power @ 4 Ω	1,943 BTU/h		491 kcal/h		
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	6.34 V	4.49 V	3.18 V	2.25 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm² (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	12 kg (26.5 lb)				

19.10 K10 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		6,000 W	4,000 W	2,000 W	12,000 W	8,000 W
	Max output voltage / current	200 V_{peak} / 125 A_{peak}				

AC Mains Power					
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)			
	Operating voltage	100-240 V ±10%, 50/60 Hz			
	Power factor cos (φ)	>0.95 @ >500 W			
	Consumption / current draw	@ 230 V		@ 115 V	
	Idle	84 W	1.17 A	91 W	1.3 A
	1/8 of max output power @ 4 Ω	1,250 W	6.1 A	1,250 W	12.2 A
	1/4 of max output power @ 4 Ω	2,500 W	11.3 A	2,500 W	22.6 A

Thermal			
	Environmental operating temperature	0° - 45° C / 32° - 113° F	
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow	
	Idle	546 BTU/h	138 kcal/h
	1/8 of max output power @ 4 Ω	1,244 BTU/h	314 kcal/h
	1/4 of max output power @ 4 Ω	1,943 BTU/h	491 kcal/h

Audio					
	Gain, selectable	26dB	29dB	32dB	35dB
	Input Sensitivity @ 8 Ω	6.34 V	4.49 V	3.18 V	2.25 V
	Max input level	27dBu	24dBu	21dBu	18dBu
	Gate	-52dBu	-55dBu	-58dBu	-61dBu
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)			
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)			
	Crosstalk separation	> 66dB @ 1 kHz			
	Input Impedance	10 k Ω balanced			
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)			
	Slew rate	50 V/µs @ 8 Ω, input filter bypassed			
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz			

DSP		
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)
	Memory	8 MB (RAM) plus 2 MB (flash for presets)
	Presets	50 stored locally + 150 stored on a smartcard
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter

Front Panel		
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel
	Controls	4 pushbuttons, function depending on user menu
	Power switch	Mains switch
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers

Rear Panel		
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD
	Network data port Ethernet	2 x RJ45 with activity LEDs
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm^2 (10AWG) cable
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch

Construction		
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support
	Weight	12 kg (26.5 lb)

19.11 K20

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		9,000 W	5,200 W	2,700 W	18,000 W	10,400 W
	Max output voltage / current	225 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	88 W	1.17 A	90 W	1.15 A	
	1/8 of max output power @ 4 Ω	1,625 W	7.9 A	1,625 W	15.8 A	
	1/4 of max output power @ 4 Ω	3,250 W	14.7 A	3,250 W	29.3 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	682 BTU/h		172 kcal/h		
	1/8 of max output power @ 4 Ω	1,590 BTU/h		402 kcal/h		
	1/4 of max output power @ 4 Ω	2,498 BTU/h		631 kcal/h		
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	7.37 V	5.22 V	3.68 V	2.62 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP (optional)						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port RS485	1 x RJ45 with 2 recessed rotary encoders for ID selection				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm² (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	12 kg (26.5 lb)				

19.12 K20 DSP+AESOP

Specifications

General						
	Number of channels	2				
	Output power	stereo mode			mono-bridged mode	
	EIAJ Test Standard, 1 kHz, 1% THD	2 Ω/ch	4 Ω/ch	8 Ω/ch	4 Ω	8 Ω
		9,000 W	5,200 W	2,700 W	18,000 W	10,400 W
	Max output voltage / current	225 V_{peak} / 125 A_{peak}				
AC Mains Power						
	Power supply	Universal, regulated switch mode with PFC (Power Factor Correction)				
	Operating voltage	100-240 V ±10%, 50/60 Hz				
	Power factor cos (φ)	>0.95 @ >500 W				
	Consumption / current draw	@ 230 V		@ 115 V		
	Idle	90 W	1.31 A	92 W	1.34 A	
	1/8 of max output power @ 4 Ω	1,625 W	7.9 A	1,625 W	15.8 A	
	1/4 of max output power @ 4 Ω	3,250 W	14.7 A	3,250 W	29.3 A	
Thermal						
	Environmental operating temperature	0° - 45° C / 32° - 113° F				
	Thermal dissipation	Fan, continuously variable speed, temperature controlled, front to rear airflow				
	Idle	682 BTU/h	172 kcal/h			
	1/8 of max output power @ 4 Ω	1,590 BTU/h	402 kcal/h			
	1/4 of max output power @ 4 Ω	2,498 BTU/h	631 kcal/h			
Audio						
	Gain, selectable	26dB	29dB	32dB	35dB	
	Input Sensitivity @ 8 Ω	7.37 V	5.22 V	3.68 V	2.62 V	
	Max input level	27dBu	24dBu	21dBu	18dBu	
	Gate	-52dBu	-55dBu	-58dBu	-61dBu	
	Frequency response	20 Hz - 20 kHz (1W @ 8 Ω, ±0.5dB)				
	S/N ratio (amplifier section)	>110dBA (20 Hz - 20 kHz, A weighted)				
	Crosstalk separation	> 66dB @ 1 kHz				
	Input Impedance	10 k Ω balanced				
	THD+N/SMPTE IMD/DIM 100 IMD	<0.5% from 1W to full power (typically <0.05%)				
	Slew rate	50 V/μs @ 8 Ω, input filter bypassed				
	Damping factor @ 8 Ω	>5000 @ 20-200 Hz				
DSP						
	A/D converter	Dual 24bit 96 kHz Tandem® architecture with 127dBA of dynamic range and THD <0.005% (20 Hz - 20 kHz)				
	D/A converter	Dual 24bit 96 kHz Tandem® architecture with 122dBA of dynamic range and THD <0.003% (20 Hz - 20 kHz)				
	Memory	8 MB (RAM) plus 2 MB (flash for presets)				
	Presets	50 stored locally + 150 stored on a smartcard				
	Digital audio input	AES3 (glitchless fallback to analog audio selectable)				
	Delay for time alignment	up to 4 s on the input section, up to 32 ms per output, sample-by-sample stepping				
	Crossover filters	Butterworth, Linkwitz-Riley, Bessel, Arbitrary Asymmetric, 6dB/oct to 48dB/oct (IIR), linear phase (FIR), hybrid (FIR+IIR)				
	Output equalizer	16 fully parametric filters per channel, IIR: peaking, hi/lo shelving, hi/lo pass eq, band pass, band stop, all pass. Custom FIR up to 384 taps @ 48 or 96 kHz				
	Input equalizer	Three layers (PEQ, raised cosine, shelving), 32 filters each + group filters, up to 256 filters per channel				
	Cable compensation network	up to 2 Ω negative/positive wire compensation (Active DampingControl™)				
	Limiters	Power limiter (TruePower™, RMS voltage, RMS current) + Peak Limiter				
Front Panel						
	Indicators	7 meter LEDs: 5 x green, 1 x yellow, 1 x red, top yellow and red show alarm with protect description on LCD panel				
	Controls	4 pushbuttons, function depending on user menu				
	Power switch	Mains switch				
	Network data port AESOP incl. AES3	2 x RJ45 with activity LEDs				
	Maintenance	SmartCard reader/writer for firmware updates and preset storage. Easily accessible dust filter foam behind two steel covers				
Rear Panel						
	Audio signal input connectors	Analog: 2 x balanced Neutrik® Combo XLR female/1/4" jack; AES3: use channel 2 XLR				
	Loudspeaker output connectors	2 x Neutrik® Speakon NL4MD				
	Network data port Ethernet	2 x RJ45 with activity LEDs				
	Aux voltage	1 x 2-pin Phoenix P. 3.81mm				
	AC mains	AMP CPC 45A on rear panel; AMP CPC 45A connector mounted on a 3 x 5mm^2 (10AWG) cable				
	Controls	1 x link switch, linking analog inputs 1 and 2; AES3/analog input switch				
Construction						
	Dimensions	W 483 mm / 19", H 44.5 mm / 1 RU, D 475 mm / 18.7"				
	Chassis	1 mm / 0.04" steel chassis and removable dust cover ; 3 mm / 0.12" steel front panel, screw hole protection, side reinforcement & rear support				
	Weight	12 kg (26.5 lb)				