Bias V3/V9

User Guide V1.0

Version 1.0

This user guide is subject to change without notice.
For the latest online version, visit: www.voidacoustics.com

Contents

1 Important safety instructions

EXPLANATIONS OF GRAPHICAL SYMBOLS


The triangle with the lightning bolt is used to alert the user to the risk of electric shock.


The triangle with the exclamation point is used to alert the user to important operating or maintenance instructions.


The CE-mark indicates the compliance with the low voltage and electromagnetic compatibility.


Symbol for earth/ground connection.


Symbol indicating that the equipment is for indoor use only.


Symbol for conformity with Directive 2002/96/EC and Directive 2003/108/EC of the European Parliament on waste electrical and electronic equipment (WEEE).


Do not use the unit at altitudes above 2000 m.


Do not use the unit in tropical environment.


WARNING: 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.


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.**


DO NOT EXPOSE THIS EQUIPMENT TO RAIN OR MOISTURE, DRIPPING OR SPLASHING LIQUIDS. OBJECTS FILLED WITH LIQUIDS, SUCH AS VASES, SHOULD NOT BE PLACED ON THIS APPARATUS.


BIAS V3,V9 MUST BE INSTALLED IN RACK CABINETS: INSTEAD OF CONNECTING THE AMPLIFIER TO THE POWER GRID DIRECTLY, PLUG THE AMPLIFIER'S MAINS CONNECTIONS VIA A SECTIONING BREAKER TO A POWER DISTRIBUTION PANEL INSIDE THE RACK CABINET.


WHEN THE UNIT IS INSTALLED IN A CABINET OR A SHELF, MAKE SURE THAT IT HAS SUFFICIENT SPACE ON ALL SIDES TO ALLOW FOR PROPER VENTILATION (50 CM FROM THE FRONT AND REAR VENTILATION OPENINGS).


CONNECTION TO THE MAINS SHALL BE DONE ONLY BY A ELECTROTECHNICAL SKILLED PERSON ACCORDING THE NATIONAL REQUIREMENTS OF THE COUNTRIES WHERE THE UNIT IS SOLD.

Electrical energy can perform many useful functions. This unit has been engineered and manufactured to ensure your personal safety. But IMPROPER USE CAN RESULT IN POTENTIAL ELECTRICAL SHOCK OR FIRE HAZARD.
In order not to defeat the safeguards incorporated into this product, observe the following basic rules for its installation, use and service. Please read these "Important Safeguards" carefully before use.

Important safety instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this equipment near water.
6. Clean only with a dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. 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.
10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.

13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

Numbers 9 and 13 apply only to Bias V3

* Bias V9: interrupt the mains by switching the sectioning breaker off.

** Valid for Bias V3 model only; with Bias V9 a free leads power cord (i.e. without plug) is provided: this solution is intended for connecting the device to a sectioning breaker on the mains. Refer to the installation instruction for selecting the proper sectioning breaker.

2 Regulatory information

FCC COMPLIANCE NOTICE

This device complies with part 15 of the FCC rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

CAUTION: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

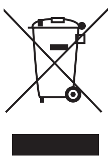
NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

WEEE DIRECTIVE

If the time arises to throw away your product, please recycle all the components possible.

This symbol indicates that when the end-user wishes to discard this product, it must be sent to separate collection facilities for recovery and recycling. By separating this product from other household-type waste, the volume of waste sent to incinerators or land-fills will be reduced and natural resources will thus be conserved.



The Waste Electrical and Electronic Equipment Directive (WEEE Directive) aims to minimise the impact of electrical and electronic goods on the environment. Void comply with the Directive 2002/96/EC and 2003/108/EC of the European Parliament on waste electrical finance the cost of treatment and recovery of electronic equipment (WEEE) in order to reduce the amount of WEEE that is being disposed of in land-fill site.
All of our products are marked with the WEEE symbol; this indicates that this product must NOT be disposed of with other waste. Instead it is the user's responsibility to dispose of their waste electrical and electronic equipment by handing it over to an approved reprocessor, or by returning it to Void for reprocessing. For more information about where you can send your waste equipment for recycling, please contact Void or one of your local distributors.

EC DECLARATION OF CONFORMITY

Manufacturer:

Void Acoustics Research Ltd
Unit 15, Dawkins Road Ind Est.,
Poole,
Dorset, BH15 4JY,
United Kingdom

We declare that under our sole responsibility the products:
Model Names: Bias V3
Bias V9
Intended use: Professional Audio Amplifier

Are in conformity with the provisions of the following EC Directives, including all amendments, and with national legislation implementing these directives:

- 2006/95/EC Low Voltage Directive
- 2004/108/EC Electromagnetic Compatibility Directive
- 2002/95/CE RoHs Directive

The following armonized standards are applied:

- EN 55103-1:2009 /A1:2012
- EN 55014-1:2006 /A1:2009 /A2:2011
- EN 55022:2010 /AC:2011
- EN 61000-3-2:2006 /A1:2009 /A2: 2009
- EN 61000-3-3:2013
- EN 61000-3-11:2000
- EN 61000-3-12:2011
- EN 55103-2:2009 /IS:2012
- EN 61000-4-2:2009
- EN 61000-4-3:2006 /A1:2008 /IS1:2009 /A2:2010
- EN 61000-4-4:2012
- EN 61000-4-5:2006
- EN 61000-4-6:2014
- EN 61000-4-11:2004
- EN 60065:2002 /A1:2006 /A11:2008 /A2:2010 /A12:2011

For compliance questions only: info@voidacoustics.com

3 Bias series user guide

3.1 Welcome

Many thanks for purchasing this Void Acoustics Bias Series amplifier. We truly appreciate your support. At Void, we design, manufacture and distribute advanced professional audio systems for the installed and live sound market sectors. Like all Void products, our highly skilled and experienced engineers have successfully combined pioneering technologies with ground-breaking design aesthetics, to bring you superior sound quality and visual innovation. In buying this product, you are now part of the Void family and we hope using it brings you years of satisfaction. This guide will help you to use this product safely and ensure it performs to its full capability.

3.2 Unpacking & checking for shipping damage

Your Void product has been completely tested and inspected before leaving the factory. Carefully inspect the shipping package before opening it, and then immediately inspect your new product. If you find any damage notify the shipping company immediately.

The box contains the following:

- One Bias Series amplifier
- One AC mains power cord
- This user guide

3.3 Disposal of the packing material

The transport and protective packing has been selected from materials which are environmentally friendly for disposal and can normally be recycled.

Rather than just throwing these materials away, please ensure they are offered for recycling.

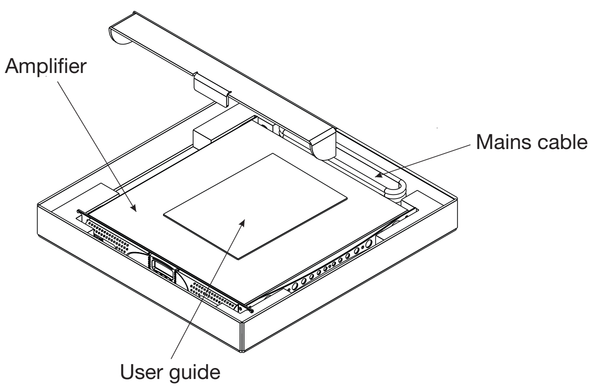


FIGURE 1: Packaging.

3.4 About the amplifier platform

Bias 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 Bias 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.

3.4.1 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 Void products for the same sound pressure level.
Void 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. Void's multi patented application of Pulse Width Modulation (PWM) high frequency sampling techniques is just one of the many factors contributing to the Bias Series' high performance ratings across the audio bandwidth.

3.4.2 The Show Always Goes On

The Bias 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 Mechanical drawings

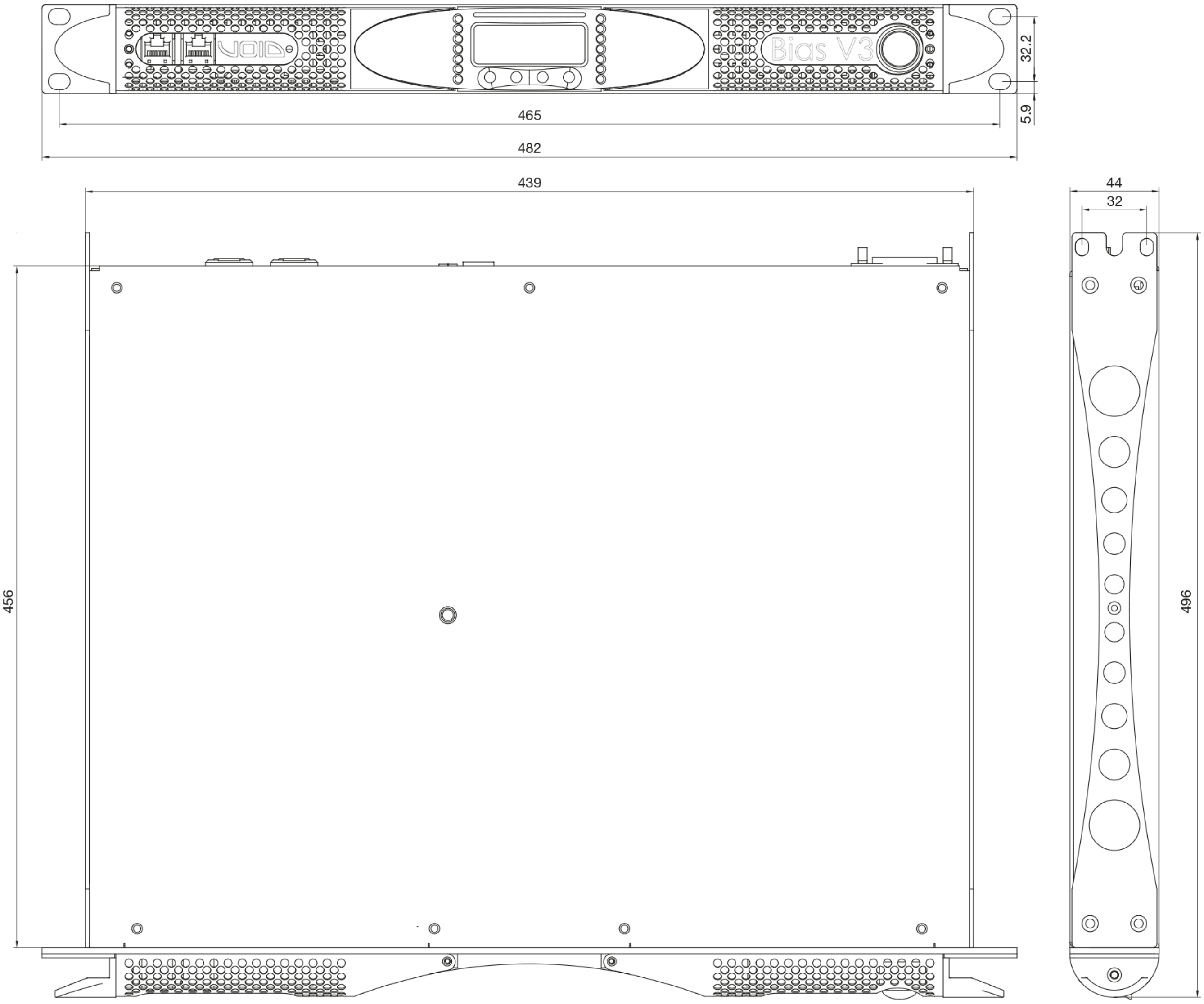


FIGURE 2: Bias V3 mechanical drawings.

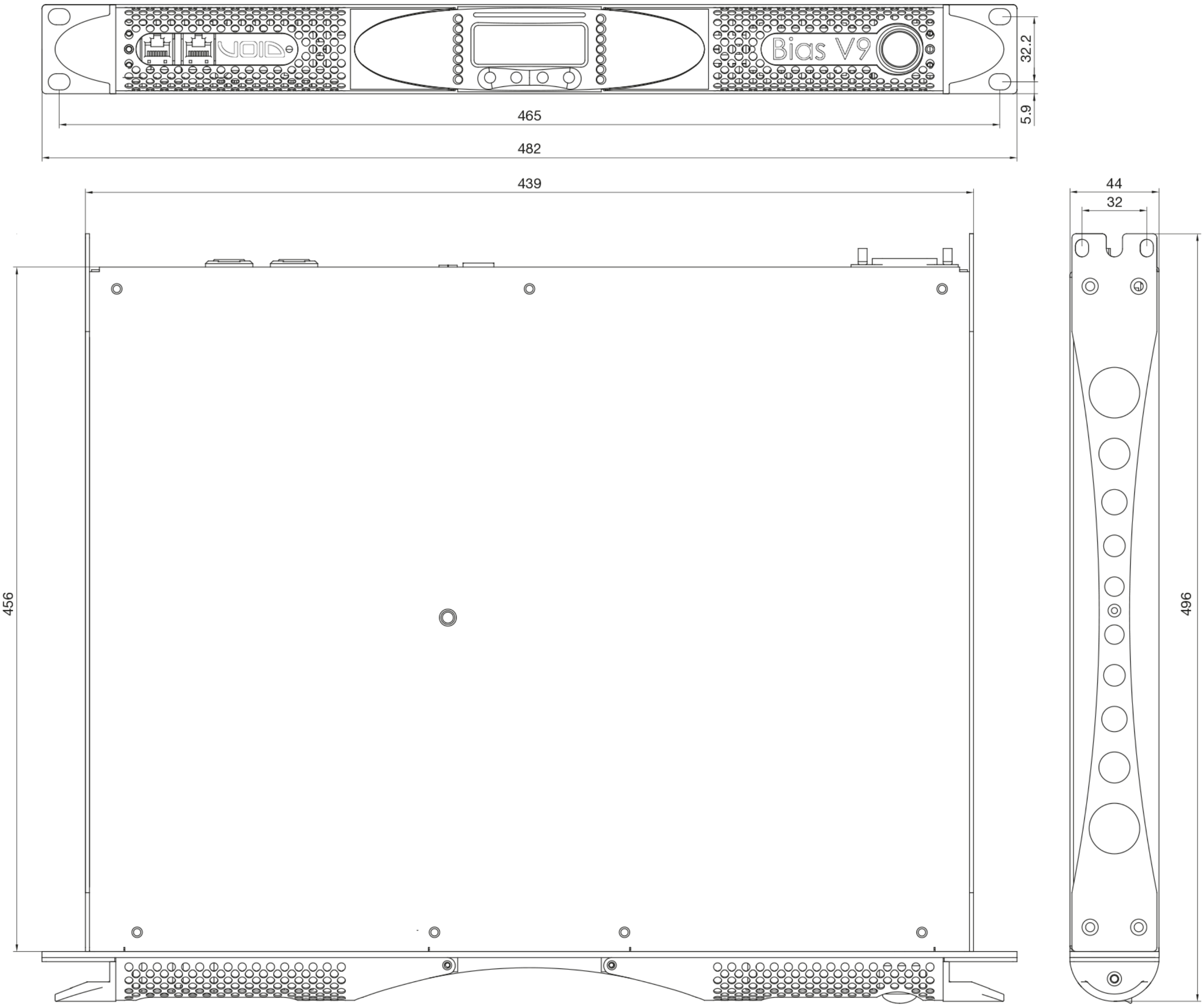
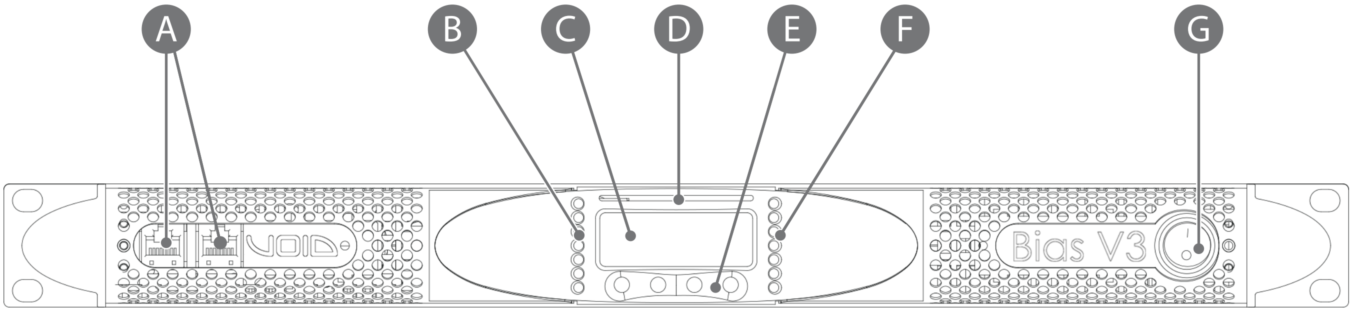


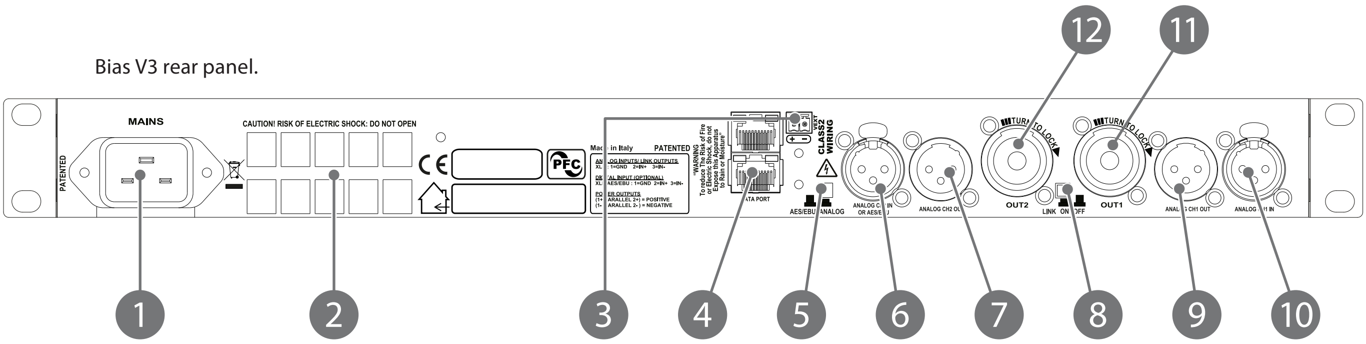
FIGURE 3: Bias V9 mechanical drawings.

5 Front and rear panels

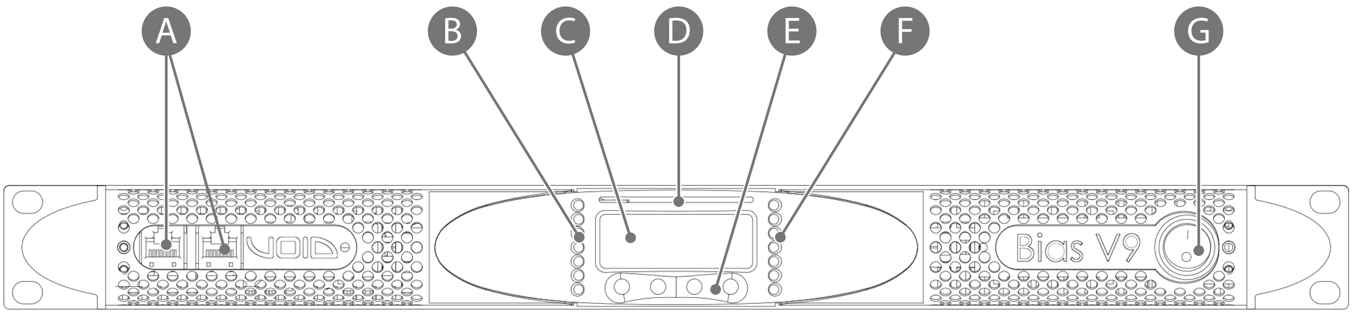


A. RJ45 plugs (either AESOP or RS485 ports according to the amplifier configuration)
B. LED bar: signal metering channel 1
C. Main display
D. Smart Card slot
E. Multifunction buttons
F. LED bar: signal metering channel 2
G. Main switch

Bias V3 rear panel.

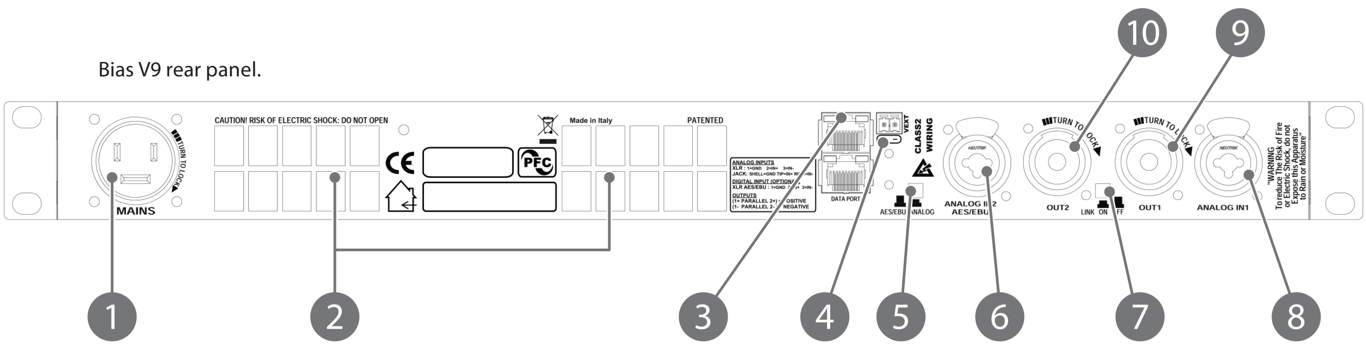


1. Mains plug
2. Air vents
3. Vext: 12 V_{DC}, 1A external voltage input (AESOP version only)
4. Ethernet+AESOP ports (AESOP version only)
5. AES3/analog switch for input 2
6. Input 2: channel 2 analog input in analog mode or AES3 input in AES3 mode, according to the position of the switch in #5
7. Line output channel 2
8. Link button: link input from channels 1 and 2
9. Line output channel 1
10. Input 1: channel 1 analog input
11. Speaker connector: output channel 1
12. Speaker connector: output channel 2



A. RJ45 plugs (either AESOP or RS485 ports according to the amplifier configuration)
B. LED bar: signal metering channel 1
C. Main display
D. Smart Card slot
E. Multifunction buttons
F. LED bar: signal metering channel 2
G. Main switch

Bias V9 rear panel.



1. Mains plug
2. Air vents
3. Ethernet+AESOP ports (AESOP version only)
4. Vext: 12 V_{DC}, 1A external voltage input (AESOP version only)
5. AES3/analog switch for input 2
6. Input 2: channel 2 analog input in analog mode or AES3 input in AES3 mode, according to the position of the switch in #5
7. Link button: link input from channels 1 and 2
8. Input 1: channel 1 analog input
9. Speaker connector: output channel 1
10. Speaker connector: output channel 2

6 Installation

The common installation of the amplifier is in rack cabinets: in order to limit the risk of mechanical damages, the amplifiers must be fixed to the rack using both frontal and rear mounting brackets.

Note: Instead of connecting the amplifier to the power grid directly, plug the amplifier's mains connections to a power distribution panel inside the rack cabinet.

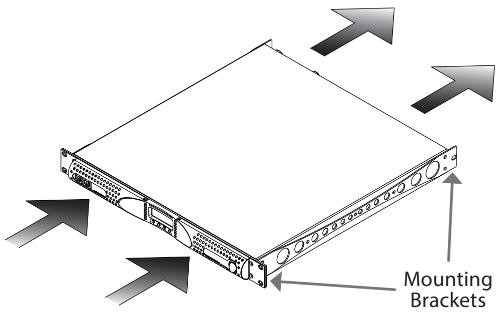


FIGURE 4: Mounting brackets and air flow direction.

6.1 Cooling

Install the amplifier in a well-ventilated location: 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.

All Void amplifiers implement a forced-air cooling system to maintain low and constant operating temperatures. Drawn by the internal fans, air enters from the front panel and is forced over all components, exiting at the back of the amplifier.

The amplifier's cooling system features "intelligent" variable-speed DC fans which are controlled by the heatsink temperature sensing circuits: the fans speed will increase only when the temperature detected 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.

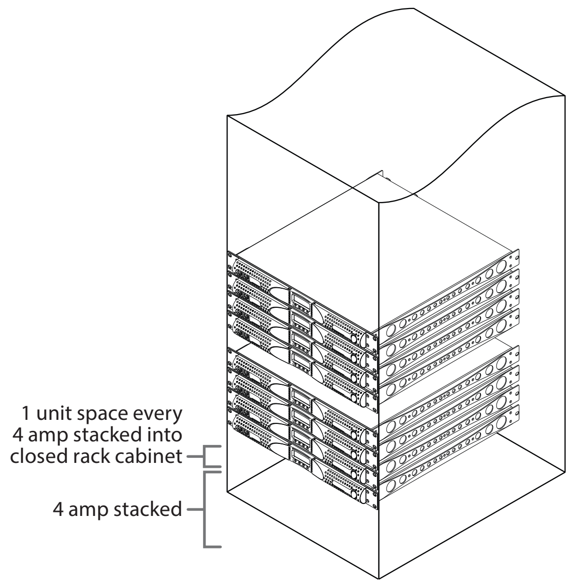


FIGURE 5: How to stack the amplifiers in closed racks.

6.2 Cleaning

Always use a dry cloth for cleaning the chassis and the front panel. Air filter cleaning should be scheduled according to the dust levels in the amplifier's operating environment.

Disconnect the AC main source before attempting to clean any part of the amplifier

In order to clean the vent filters you need to remove the front cover: never attempt to open any other part of the unit.
By means of a screwdriver Phillips PH1, unscrew the screws that lock the left and right cover grills on the front panel (ref. FIGURE 6), gently lift the covers and remove the filters. You may use compressed air to remove the dust from filters, or wash it with clean water: in the latter case ensure that the filters are dry before reassembly.

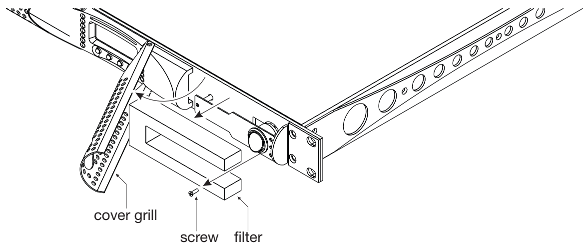


FIGURE 6: Cleaning air filters.

6.3 AC mains supply

The AC Main connection is made via the

- AMP CPC 45A connector in Bias V9;
- IEC C20 connector in Bias V3

The FIGURE 7 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: 115V-230V ±10%.

It is important to connect the ground for safety, do not use adapters that disable the ground connection.

All Bias Series amplifiers have an automatic power factor correction system – PFC – for a perfect mains network interface. The PFC minimizes the reactive power reflected on the network and reduces the harmonic distortion on the voltage/current waveform: in this way the amplifier is seen as a resistive load from the mains network. Furthermore, the system allows performance to be maintained even in case of varying mains voltage.

Connection to the mains shall be done only by a electrotechnical Skilled person according the national requirements of the countries where the unit is sold

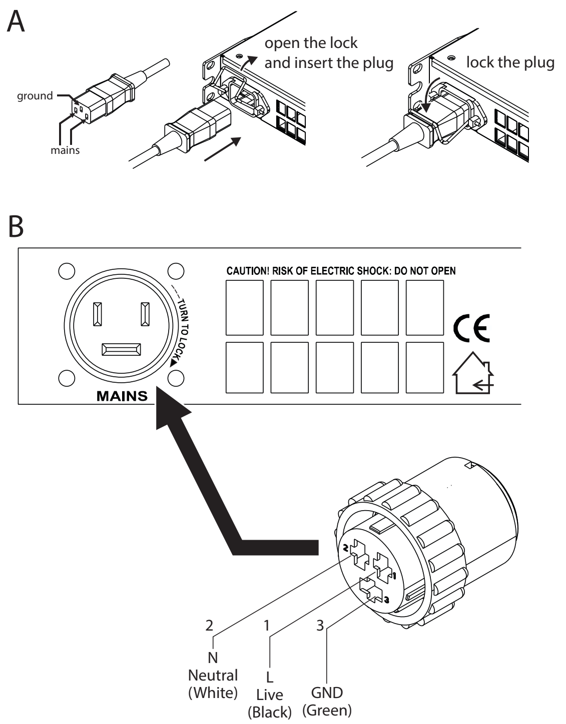


FIGURE 7: Mains connectors; A) IEC C20 in Bias V3; B) AMP CPC 45A in Bias V9.

6.4 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.

The power cord type provided with the amplifier are

- LAPP OLFLEX191 3G6 / SJT 3XAWG10 for Bias V9.
- Bahoing SJT 3x16AWG or I-sheng SGIS 3G 1,5 mm^2 for Bias V3.

WARNING: TO PREVENT FIRE OR ELECTRIC SHOCK

- This device must be powered exclusively by earth connected mains sockets in electrical networks compliant to the IEC 364 or similar rules.
- Install Bias V9 into rack cabinet.
- With Bias V9 a sectioning breaker between the mains connections and the amplifier must be installed inside the rack cabinet. Suggested device is 32A/250VAC, C or D curve, 10kA.
- With Bias V3 provide a sectioning breaker between the mains connections and the amplifier. Suggested device is 16A/250VAC, C or D curve, 10kA.
- 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 use this amplifier if the electrical power cord is frayed or broken.
- Output terminals are hazardous: wiring connection to these terminals require installation by an instructed person and the use of ready made leads.
- Take care to secure the output terminal before switching the device on.
- To avoid electrical shock, do not touch any exposed speaker wiring while the amplifier is operating.
- Do not spill water or other liquids into or on the amplifier.
- No naked flame sources such as lighted candles should be placed on the amplifier.
- Do not remove the cover. Failing to do so will expose you to potentially dangerous voltage.
- 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.
- Contact the authorized service centre for ordinary and extraordinary maintenance.

7 Connections

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

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

7.1 Signal grounding

There is no ground switch or terminal on the Bias 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 (ref. Chapter 6:3.AC mains supply). Never disconnect the ground pin on the AC mains power cord.

7.2 Analog input

Analog input is provided by means of two Neutrik XLR connectors in Bias V3 or a couple of XLR/jack hybrid combo connectors in Bias V9 amplifiers. Signal polarity for XLR and TRS plugs is shown in FIGURE 8.

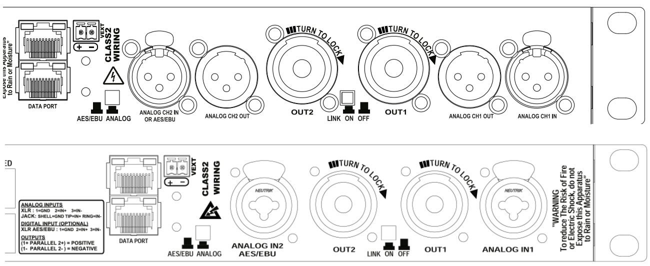


FIGURE 9: Analog input in Bias V3 (top) and Bias V9 (bottom).

7.3 Analog line output

Line out is provided in Bias V3 via a couple of XLR connectors on the rear panel. In DSP equipped models, the output signal is pre-DSP, being a replica of the input signal.

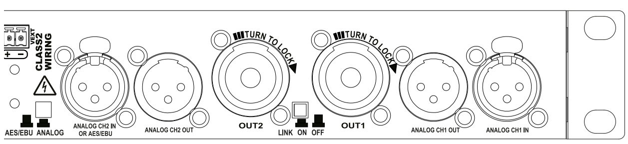
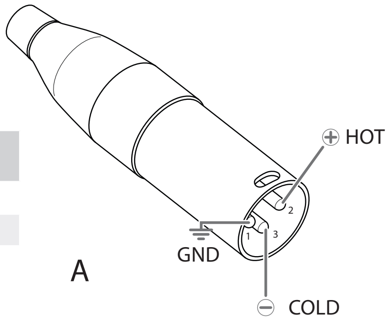
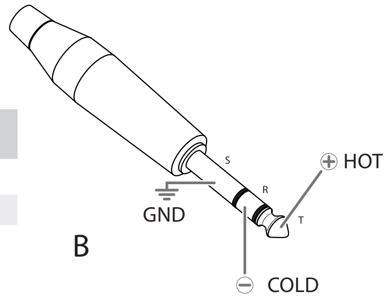


FIGURE 10: Analog line output in Bias V3.

Analog input XLR-M pinout	
Pin 1	GND
Pin 2	HOT ⊕
Pin 3	COLD ⊖



Analog input TRS Jack pinout	
Tip	HOT ⊕
Ring	COLD ⊖
Sleeve	GND



Analog line output XLR-F pinout	
Pin 1	GND
Pin 2	HOT ⊕
Pin 3	COLD ⊖

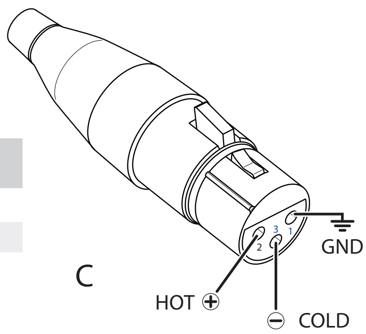


FIGURE 8: Signal polarity in balanced connections;
A) XLR-M plug; B) TRS jack; C) XLR-F plug.

7.4 Digital Input

On DSP equipped models, the XLR input for channel 2 can switch to an AES3 digital input. The AES3/analog push-button located nearby the channel 2 XLR input connector toggles the XLR between analog and digital input.
In AES3 mode

- the channel 2 analog line out is off;
- the channel 1 analog input can be used as redundant input if the digital input fails.

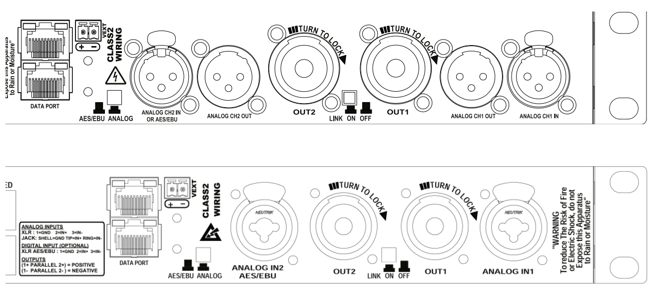


FIGURE 11: Digital input in Bias V3 (top) and Bias V9 (bottom).

The AES3 connection carries a channel pair through a 110 Ω nominal impedance wire in the form of a balanced (differential) digital signal: in AES3 XLR connectors the identification of hot and cold pins is not an issue; take care to never tie pin 2 or pin 3 (balanced signals) to pin 1 (ground). Avoid the use of microphone cables in AES connections: impedance mismatch can result in signal reflections and jitter, causing bit errors at the receiver.

7.5 AESOP

The AESOP standard can transport a single bidirectional Fast Ethernet (IEEE 802.3u, 100 Mbit/s) control data stream and two independent separate AES3 digital audio monodirectional streams using one Cat5 cable.

All Bias Series amplifier with the optional AESOP board installed are equipped with at least two RJ45 connectors, each of them being 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 front panel. If four plugs are present, the rear two will be “primary” ports, while the two on the front panel are “secondary” ports.

Primary ports allow both data and AES3 streams; secondary ports, on the other hand, are data-only ports, allowing Ethernet connections only.

Cat5 standard twisted pair cables shall be used for connections up to 100 meters (328 ft). RJ45 pinout must comply to TIA/EIA-568-B and adopt the T568B scheme pinout, as show in TABLE 1.

For more details about networking and AESOP please refer to Section: Network operations (p. 25).

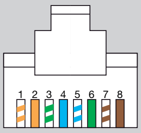
RJ45 connector seen from the front end		
	Colour code (TIA/EIA-568-B)	Pin
	ORANGE / WHITE	1
	ORANGE	2
	GREEN / WHITE	3
	BLUE	4
	BLUE / WHITE	5
	GREEN	6
	BROWN / WHITE	7
	BROWN	8

TABLE 1: EtherCON/RJ45 T568B scheme pinout.

7.6 Loudspeaker connections

CLASS3 WIRING V9 CLASS3 WIRING

CLASS2 WIRING V3 CLASS2 WIRING

Output terminals are hazardous: wiring connection to these terminals require installation by an instructed person and the use of ready made leads.

Take care to secure the output terminal before switching the device on.

Two Neutrik NL4MD speakON connectors are located on the rear panel, each of them being a single output to loudspeaker.

Pins 1+ and 2+ are physically bridged to the positive pole; pins 1– and 2– are physically bridged to the negative pole.

In order to remain within safe operating conditions, when using low impedance loads – i.e. 4 Ω or less (8 Ω or less in bridge mode) –, connections must be made with a four wire cable. Use suitable wire gauges to minimize power and damping factor losses in speaker cables.

7.6.1 Bridge-tied load

Bridge-tied load connection can be achieved as described in FIGURE 12. In analog mode, only the input of channel 1 needs to be wired: link channel 2 to channel 1 by means of the link push-button located on the rear panel.

When operating with digital inputs – i.e. AES3 and AESOP – link the channels via software: do not switch the link push-button.

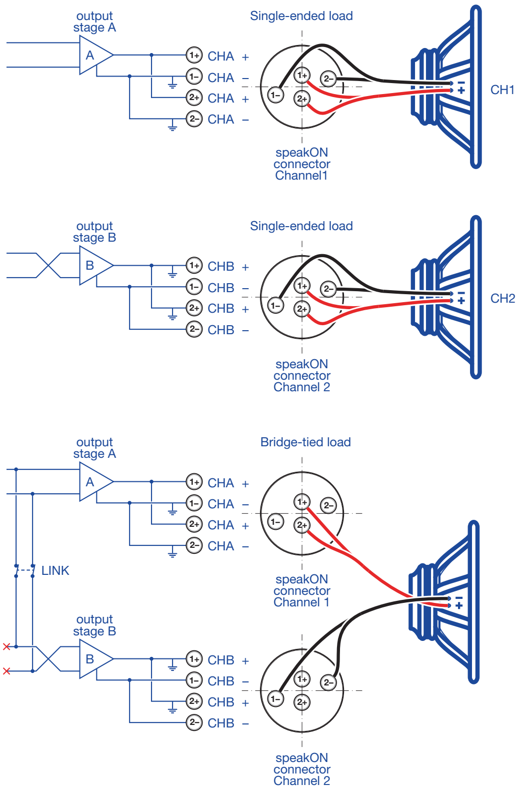


FIGURE 12: Loudspeaker connections: single-ended loads (top), bridged-tied load (bottom).

7.6.2 Internal signal path polarity

In order to increase the power's supply energy storage efficiency, signals coming from each channel pairs are polarity reversed, one with respect to the other within the pair, 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 at the output connectors to ensure that both channels output with the same polarity as their corresponding input signals.

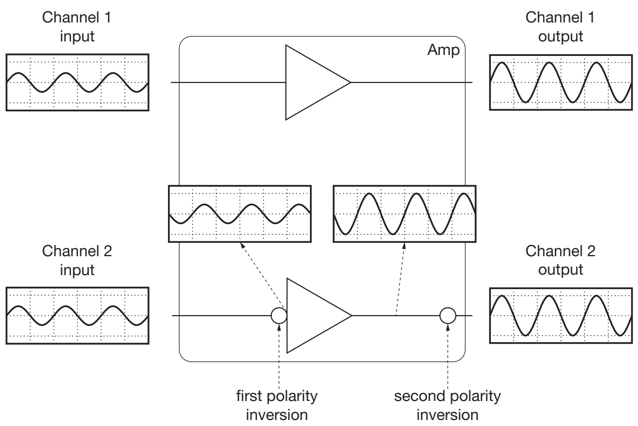


FIGURE 13: Internal signal path polarity with example input signals. Both channels 1 and 2 are fed the same sine signal.

7.7 V Ext

The V Ext terminal is used to remotely manage the DSP in Bias Series DSP amplifier and enable remote on/off.

Bias Series provided with a AESOP board have a dedicated 2 pin Phoenix connector MCV 1,5/ 2-G-3,81 - 1803426 located near the rear Ethernet ports. K Series with the RS-485 serial port implement the V Ext connection on pin 2 (pin 7) of the RJ45 rear connector (ref. FIGURE 15).

When the V Ext port is powered by and external 12 V_{DC} (1 A max) power supply, the internal controller allows to control the DSP – if present – even without AC mains supply, and allows serial communication – via RS-485 or ethernet communication in AESOP equipped models – for remote on/off via the Armonía Pro Audio Suite software.

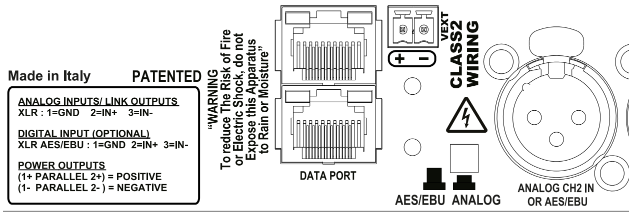


FIGURE 14: V Ext phoenix connector MCV 1,5/ 2-G-3,81.

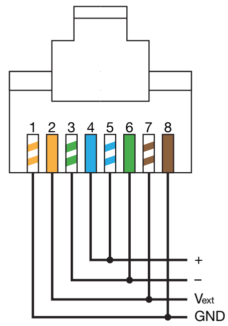


FIGURE 15: Front view of the RJ45 connector with T568 B wiring: RS-485 pinout.

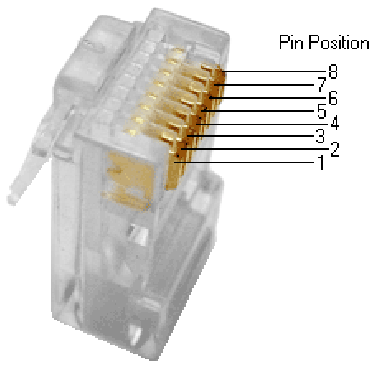


FIGURE 16: RJ45 (8P8C) plug.

8 LEDs and display menu

In all Bias 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.

All the setup and settings functions described in this section can be also accessed through a computer with Void's Armonía Pro Audio Suite software. Armonía is a software environment that offers an easy to use end user remote control interface and signal processing capabilities.

Armonía Pro Audio Suite is available for free on the Armonía forum:

http://forum.voidaudio.com/uploads/public/Armonia.zip

Please note that when an Armonía client is connected to the amplifier, any local operation is overridden by the software. Please note the it can also be found on the shared Google Drive resource.

8.1 LED chart

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.

Colour		Solid	Blinking
	RED	Signal clipping OR channel muted for protection[1]	Tone detection problem
	YELLOW	Temperature above 85°C OR output level[2] -2 dB	Critical temperature (80° - 85°C)
	GREEN	output level[2] -3 dB	
	GREEN	output level[2] -6 dB	
	GREEN	output level[2] -9 dB	
	GREEN	output level[2] -15 dB	
	GREEN	input signal is above -60 dBV OR output level[2] -18 dB	

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

TABLE 2: LED chart.

8.2 Front display

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

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 operating 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.

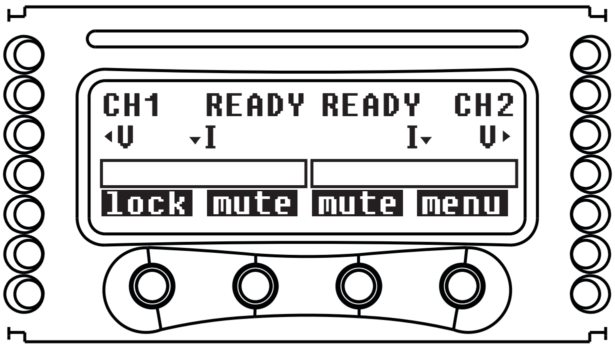


FIGURE 17: Bias Series front display.

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 shadow will appear in the software workspace view, signalling local access to the amplifier.

8.2.1 How to navigate the main menu

The Bias Series main menu can be accessed by pressing the first button on the right, underneath the LCD label "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.

Some submenus in the Bias 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.

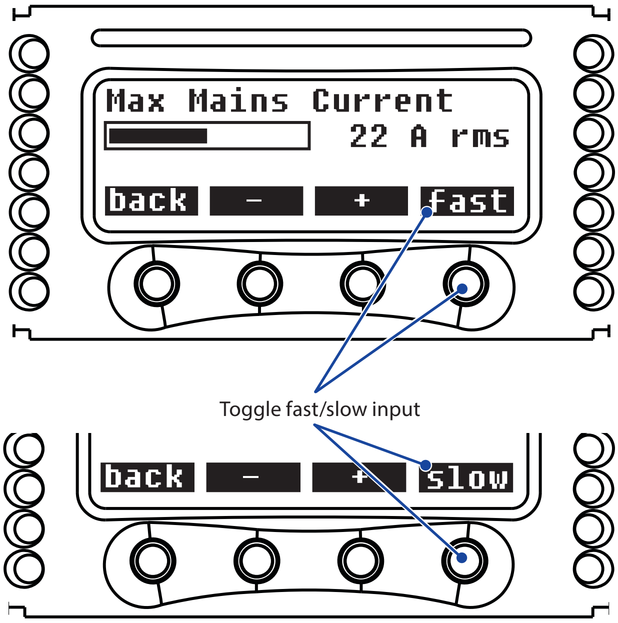


FIGURE 18: Fast/slow data emission.

For example: in “slow” mode a single “+” button press will increase the Max mains current from 22 A to 23 A; in “fast” mode a single “+” button press will increase the Max mains current from 22 A to 32 A.

8.2.2 Menu diagrams

On the following pages you will find two diagrams providing an overview of the structure of the Main menu (FIGURE 20) and DSP settings menu (FIGURE 21) accessible via the front panel on Bias Series amplifiers.

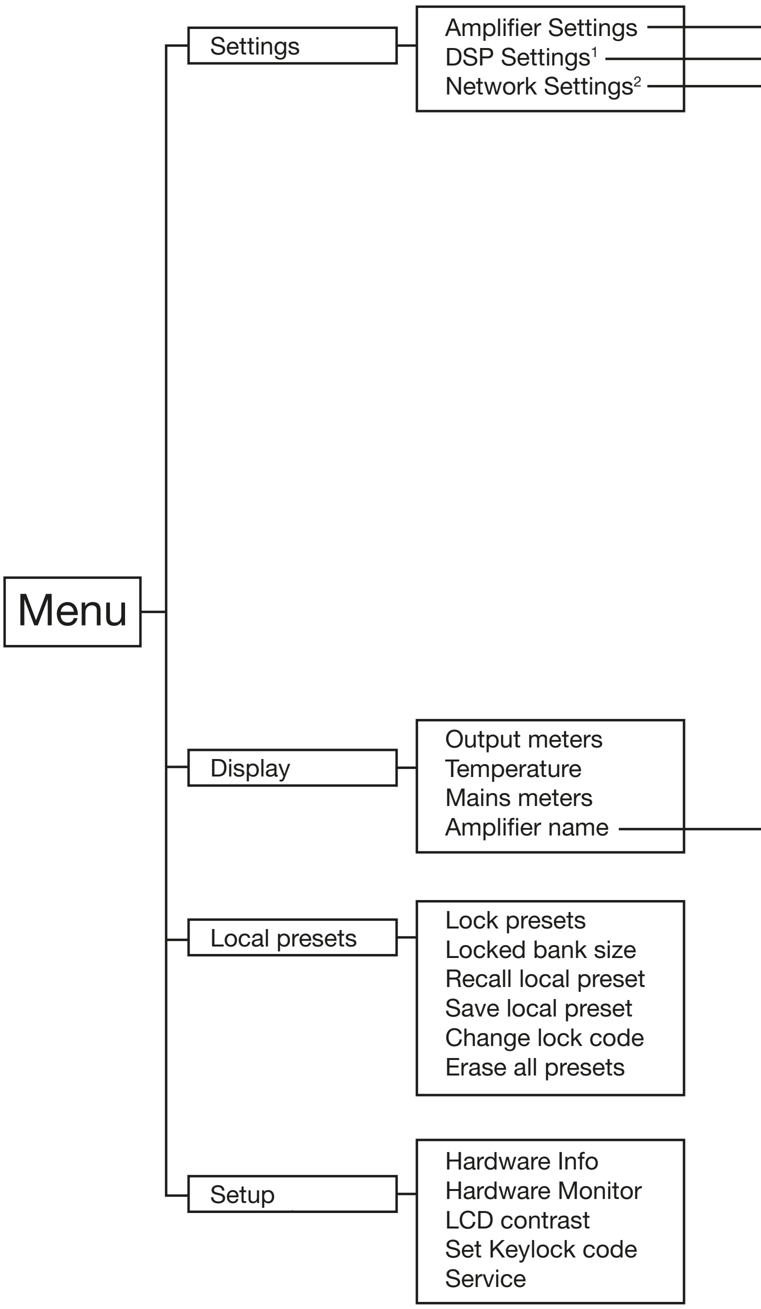
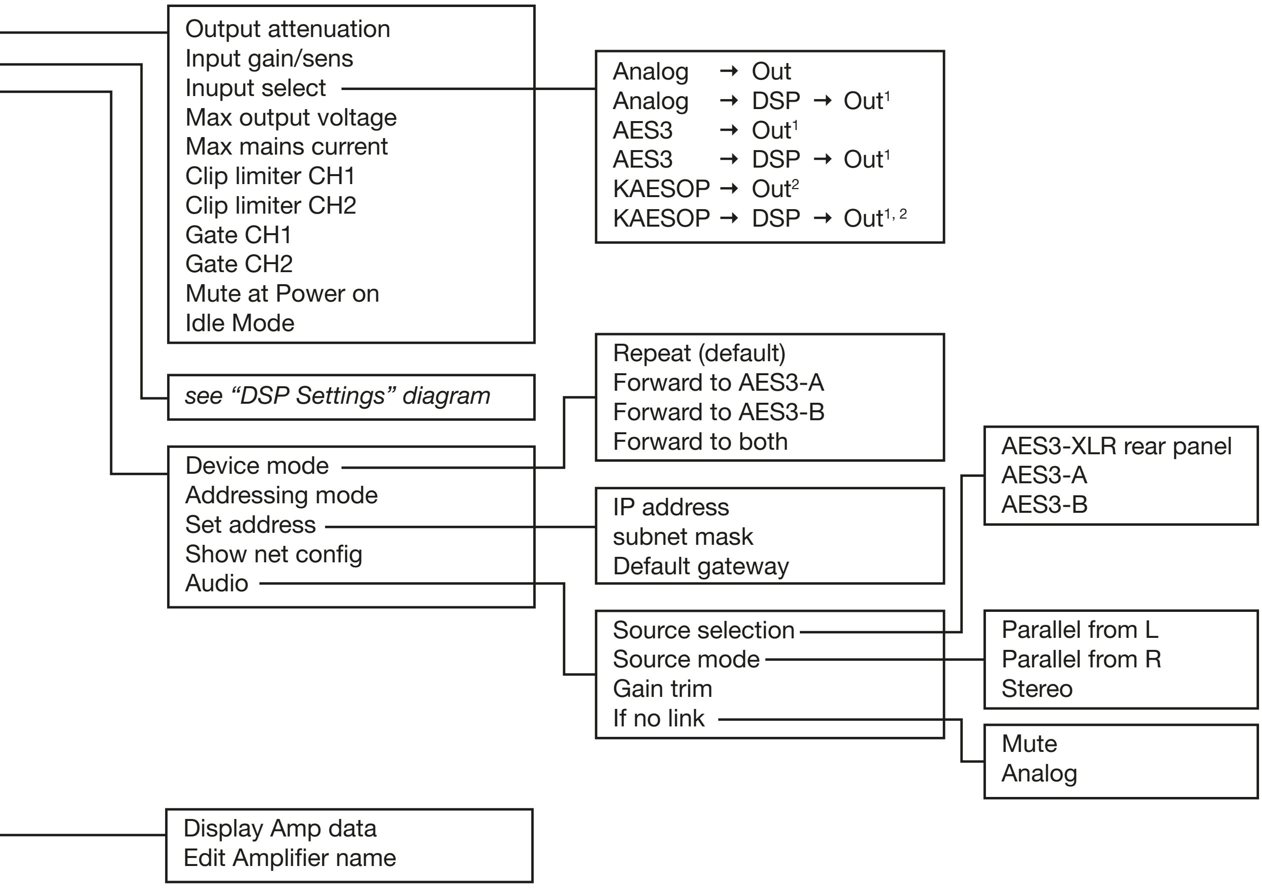


FIGURE 19: Main menu diagram.



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

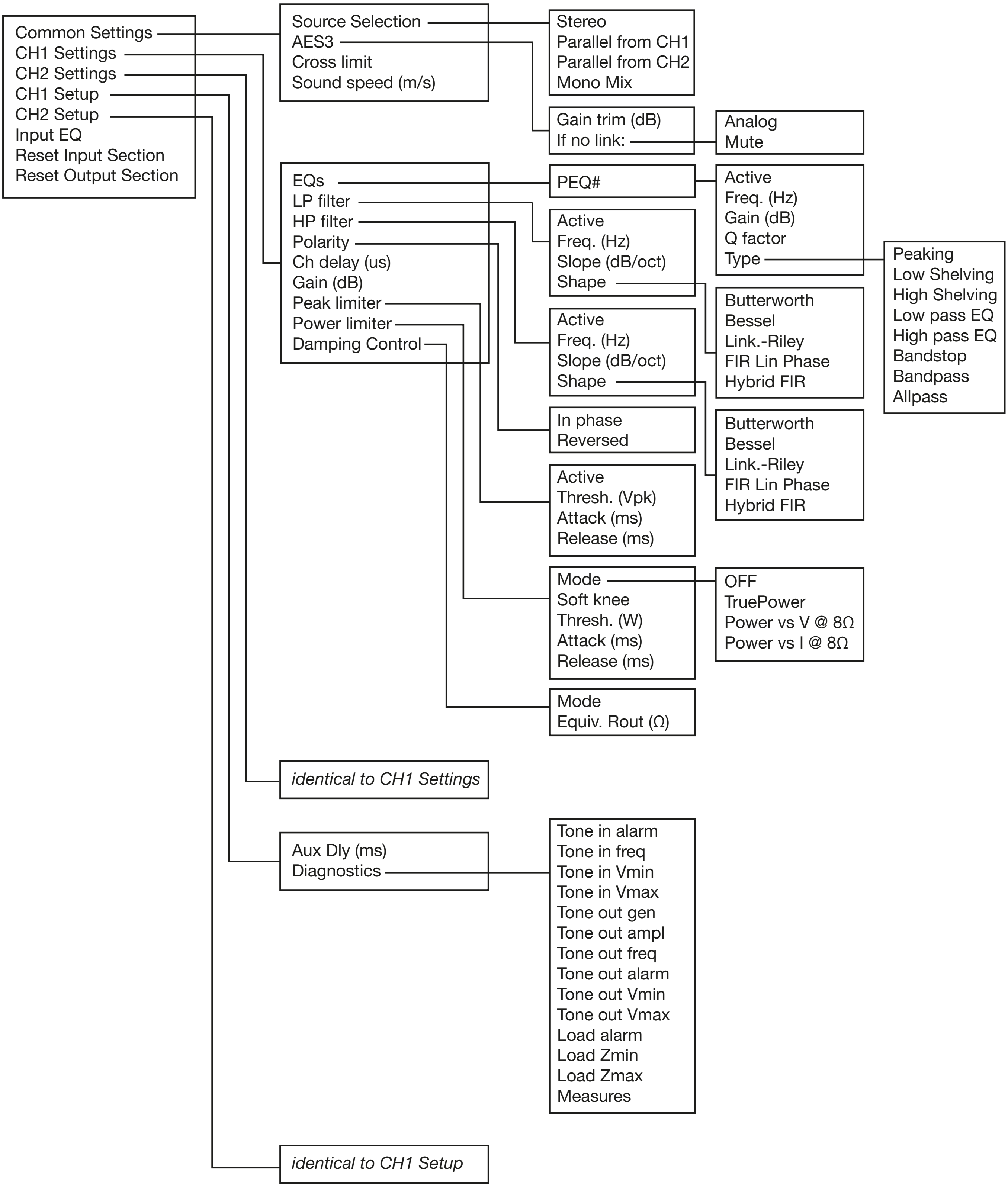


FIGURE 20: DSP settings diagram.

9.1 Amplifier settings: 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 -30 dB. A single "+" or "-" button press will increase or decrease the output attenuation by 1 dB.

Note: for ideal sonic performance, select a 0 dB output attenuation (meaning no attenuation), and select the proper gain/sensitivity level as explained in the next paragraph.

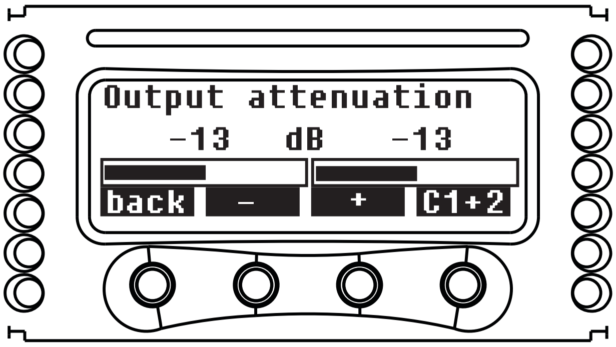


FIGURE 21: Output attenuation.

9.2 Amplifier settings: Input Gain/ Sensitivity

All Bias 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 26 dB, 29 dB, 32 dB and 35 dB.

TABLE 3 shows the input sensitivity values for the Bias Series amplifiers. These are the maximum RMS voltage values of a

Gain	V3	V9
26 dB	5.30 V	7.37 V
29 dB	3.75 V	5.22 V
32 dB	2.66 V	3.68 V
35 dB	1.88 V	2.62 V

TABLE 3: Input sensitivity (in RMS volt) @ 1 kHz vs gain.

1 kHz sine wave input before clipping occurs at the output stage. These values are reported with respect to the amplifier's gain.

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 TABLE 4.

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

TABLE 4: Maximum balanced input signal vs gain.

9.3 Amplifier settings: Input select

Bias Series amplifiers allow the user to choose three different input modes (if available): Analog, AES3[1] and/or AESOP[2].

Each of these inputs can either be processed by the internal DSP (if installed) 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 available signal routing path configurations are:

- Analog → Out
  Analog input and direct output
- Analog → DSP → Out[1]
  Analog input routed to the internal DSP
- AES3 → Out
  AES3 input, direct output
- AES3 → DSP → Out[1]
  AES3 input routed to the internal DSP
- AESOP → Out[2]
  AESOP input, direct output
- AESOP → DSP → Out[1,2]
  AESOP input routed to the internal DSP

9.4 Amplifier settings: Max output voltage

The max output peak voltage of Bias 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. Available voltage ranges for each model are shown in TABLE 5.

V3	V9
40/165 V	40/225 V

TABLE 5: Maximum output voltage (V_{peak}).

[1] Available only with optional KDSP board
[2] Available only with optional AESOP board

9.5 Amplifier settings: 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.

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 Bias V3 and from 15 A to 32 A range for Bias V9.

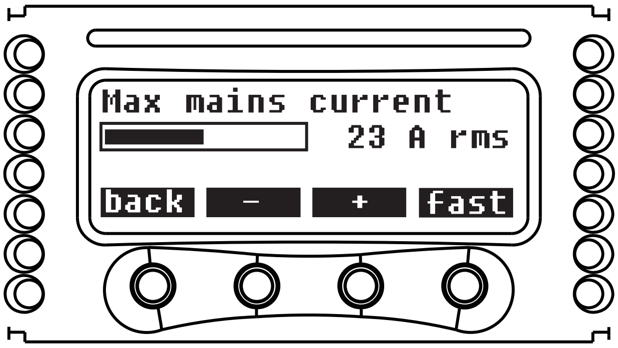


FIGURE 22: Max mains current.

Setting the maximum mains current determines the current threshold at which a C-Type current breaker will trip.

9.6 Amplifier settings: Clip limiter CH1/CH2

The clip function can be used to prevent distortion caused by clipping of the output signal.

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

CAUTION: disabling clip limiters can potentially damage loudspeakers.

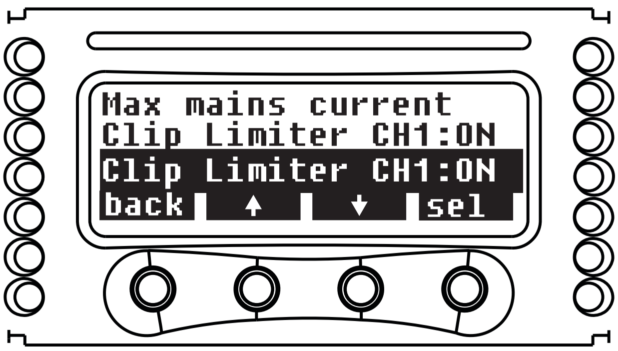


FIGURE 23: Clip limiters.

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.

9.7 Amplifier settings: Gate CH1/CH2

This function allows to mute the amplifier channels individually if the input signal amplitude falls below the threshold shown in

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.

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

TABLE 6: Gate threshold vs gain.

9.8 Amplifier settings: 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.

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.

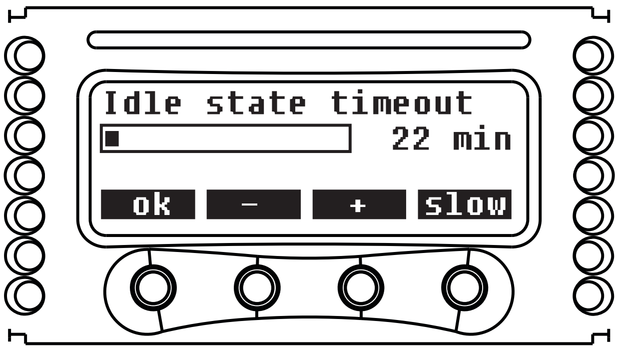


FIGURE 24: Idel state timeout.

9.9 Amplifier settings: 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 a selected threshold is detected for a user selectable amount of time, saving about 40 W of power per channel. 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 rightmost 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.

9.10 DSP Settings: Common settings

The DSP is an advanced digital sound processor board based on a floating point SHARC® DSP.

DSP can be used to optimise 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 DSP board.

9.10.1 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 routed to both output channels in order to maintain a consistent output level.

9.10.2 AES3

This menu controls the AES3 input stream options. The AES3 source can enter the amplifier from the rear XLR connector or from the AESOP board (if present) based on the type of input selection (ref. Chapter 9:3.Amplifier settings: Input select).

9.10.2.1 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 0 dB gain makes the full-scale digital signal equivalent to a 20 dBu analog input signal.

9.10.2.2 If no link

This menu controls the amplifier's behaviour 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 configuration is an AES3 fed via the rear XLR (AES3 → DSP → Out ref. Chapter 7:4.Digital Input). The backup analog cable, with an analog signal identical to that provided by AES3, should be plugged in the channel 1 XLR input. If the AES3 feed should fail, the amplifier will automatically fall back to channel 1 analog input (we suggest to set the DSP source selection to "Parallel from CH1"). The signal levels of both primary AES3 and backup analog signals should be carefully matched. This can be done using the gain trim parameter or by adjusting the analog signal level.

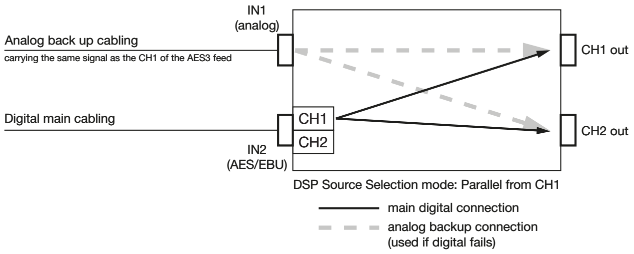


FIGURE 25: AES3 from XLR.

- AES3 from ASEOP – the primary audio signal for this configuration is an AES3 fed via the RJ45 port (AESOP → DSP → Out ref. Chapter 7:4.Digital Input). The backup analog cable, with an analog signal identical to that provided by the AESOP, should be plugged in the channel 1 XLR and channe 2 XLR (set to analog) connectors. The DSP's source selection can be set to any possible input. If the AESOP feed should fail, the amplifier will automatically fall back to the analog input on the channels 1 and 2. The signal levels of both primary AESOP and backup analog signals should be carefully matched. This can be done using the gain trim parameter or by adjusting the analog signal level.

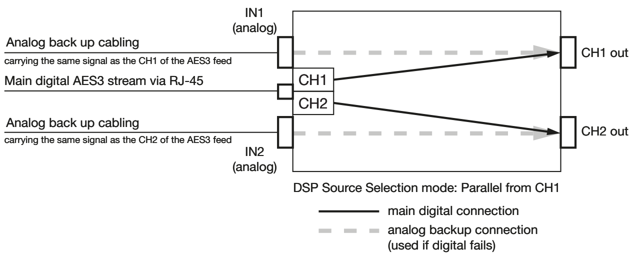


FIGURE 26: AES3 from AESOP.

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

9.10.3 Cross limit

In case of power limiting of only one channel (ref. Chapter 9:6. Amplifier settings: Clip limiter CH1/CH2), the gain reduction on one channel is mirrored to the other channel in order to maintain consistent signal 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.

9.10.4 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.

9.11 DSP Settings: 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”.

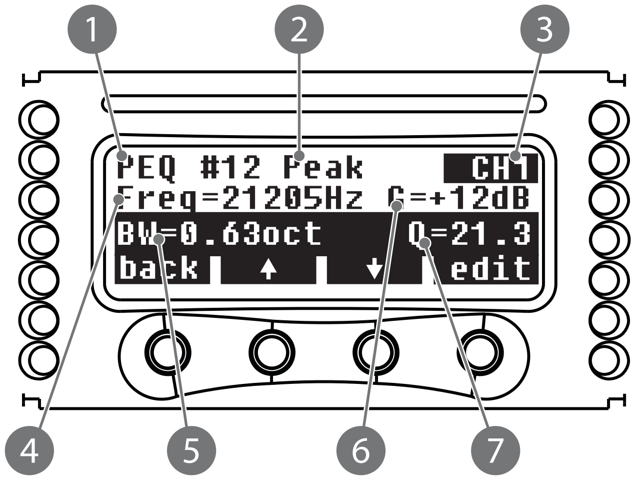


FIGURE 27: EQ settings: 1) Filter number; 2) Filter type; 3) Channel; 4) Frequency; 5) Bandwidth; 6) Gain; 7) Q.

9.11.1 EQs

This menu gives access to the parametric output equalizer interface. This menu lists the 16 parametric filters one by one. The current selected filter number is shown on the left of the

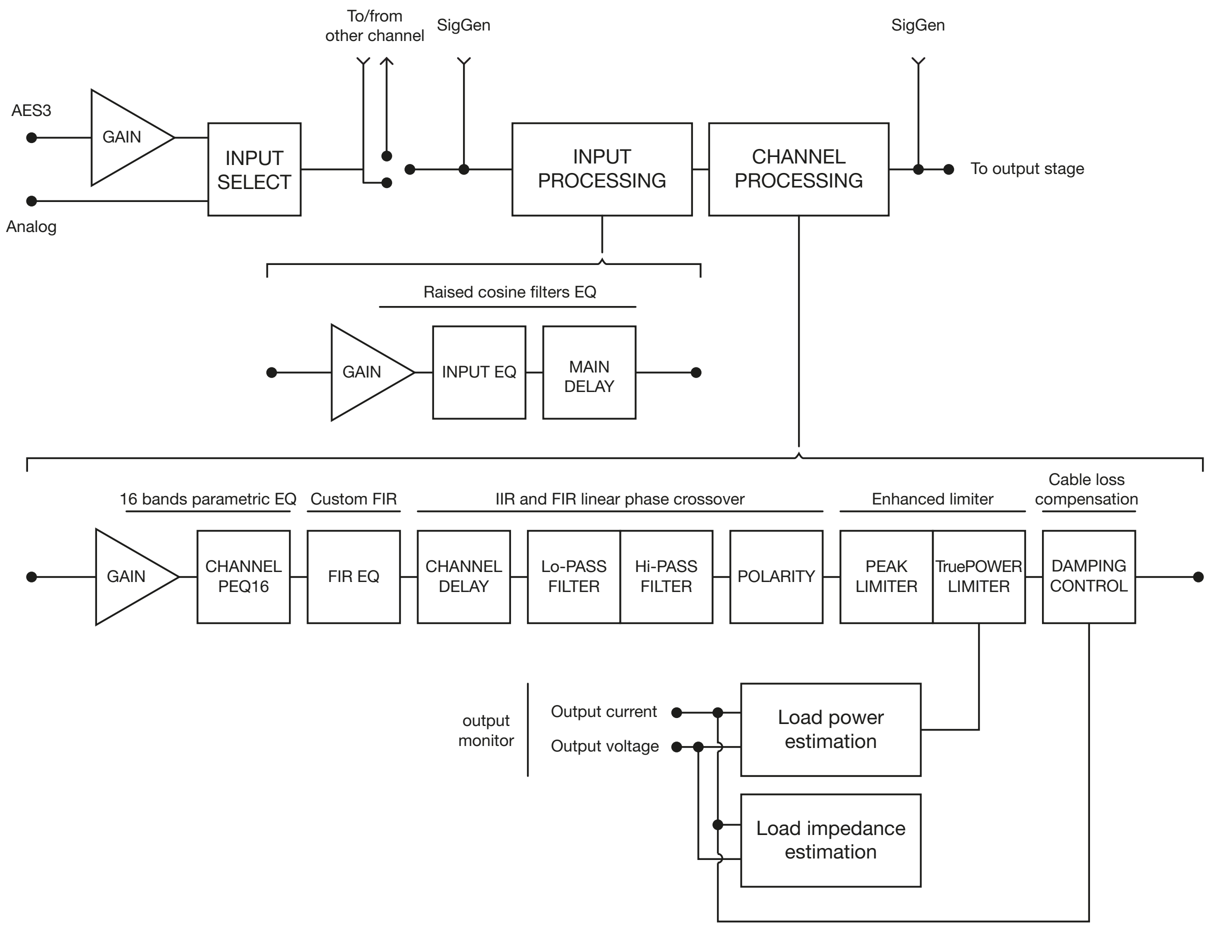


FIGURE 28: DSP processing diagram.

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.

- 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 +15 dB in 0.1 dB steps.
- Q factor: quality factor of the filter. This can be 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 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
  7. Bandpass
  8. Allpass

	Frequency 20Hz-20kHz	Gain ±15 dB	Slope 3-15dB/oct	Q 0.1-30
Peaking	✓	✓		✓
Lo-Shelv	✓	✓	✓	
Hi-Shelv	✓	✓	✓	
Lo-pass	✓			✓
Hi-pass	✓			✓
Band-stop	✓			✓
Band-pass	✓	✓		✓
All-pass	✓			✓

TABLE 7: Filters parameters.

By pressing the “edit” button, the settings for the selected filter can be modified. TABLE 7 summarizes which parameters can be edited according to the selected filter type.

9.11.2 Lo-pass/Hi-pass filters

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 Infinite Impulse Response as well as brickwall linear phase Finite Impulse Response filters are implemented. If a FIR filter in the EQ section is enabled, a FIR crossover filter cannot be enabled at the same time. The low pass and high pass filters can be edited (active status, frequency, slope, filter type) by the user via the main LCD screen.

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 -3 dB point, in the latter, the -6 dB point. The slope is freely selectable from a minimum of 6 dB/octave (1st order filter) to 48 dB/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 phase modification (30°@400Hz). 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.

9.11.3 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.

9.11.4 Channel Delay

This menu allows to set a single channel output delay.

This is helpful to time-align two different loudspeakers on the two output stages. The selectable delay varies from 0 to 32 ms (about 11 meters at 344 m/s sound speed), with a single sample step (equal to 1/96000th second or 10.4 us, about 3.5 mm)

9.11.5 Gain

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

9.11.6 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:

- Limit 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. This can damage the diaphragm (breaking or deforming it).
- Limit over-heating: delivering high power to the voice coil may lead to overheating. This can damage the isolation or burn out the voice coil. Another evident high power driving effect is power compression, noticeable in low frequency speakers.

In order to prevent the 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.
- Power 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.

9.11.6.1 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. To limit the dangers of dangerous very fast transient signals, all limiters implement a look ahead time of 0.5 ms.

As a rule of thumb, 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.

You can refer to the following formulas:

$$P_{peak} = \frac{V^2_{peak}}{Re}$$

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

Where Re 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 short attack time (i.e., with a very rapid onset), can also be useful in limiting the maximum peak voltage in distributed constant voltage lines.

Void designed the Bias Series limiters as protective measures; therefore, they are not meant to "colour" 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.

TABLE 8 gives a few examples of attack and release times with respect to the frequency range of the signal to be limited.

The peak limiter menu allows the user to define the following

Octave band (Hz)	Attack time (ms)	Release time (ms)	Atk/Rel ratio
63	45	720	x16
125	16	256	x16
250	8	128	x8
500	4	32	x8
1000	2	8	x4
> 1000	1	2	x2

TABLE 8: Attack and release times per octave bands.

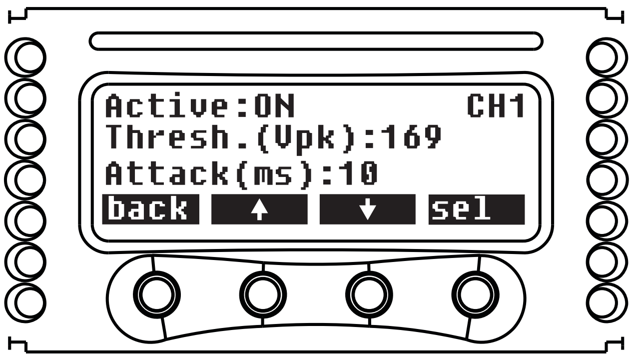


FIGURE 29: Peak limiter settings.

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.

In order to avoid choking the exceptional dynamic range offered by Bias 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 (0.5 ms) to soften clipping and minimize distortion, effectively yielding superior sonic performance.

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 dB enforced by the limiter.

This information, together with the limiting voltage referred to the signal in the input amplifier stage (I) expressed in dBu, is displayed in real time to allow monitoring of the limiting actions as they are performed.

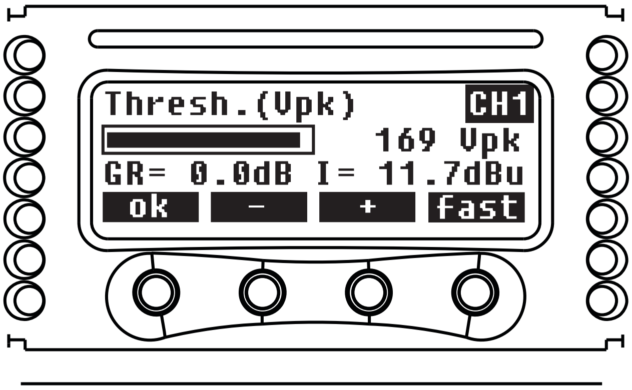


FIGURE 30: Peak limiter: threshold voltage and gain reduction.

9.11.6.2 Power 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, therefore the power limiter should not be engaged at normal working levels. The power limiter acts by decreasing the amplifier's gain in order to reduce the 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 behaviour base their operations on a mix based on threshold, dynamic behaviour of the output readings (voltage and current) and the type of output readings monitored.

Check the gain reduction: in order to obtain the optimal sound it should not be greater than 2-4 dB 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 stationary tone has a much higher average power (e.g. a stationary sine wave has 3 dB crest factor) even if it "sounds" less loud to the human ear.

There are three main operating modes for the Bias Series power limiters.

- TruePower™: the amplifier's active output power is estimated by measuring the load current. The TruePower limiter is a Void patent technology useful to avoid overheating of the voice coil; it can however also be used to avoid power compression. The DSP provides the measurement of the real power delivered (and then dissipated) to the coil, ignoring the apparent power handled by the line.

Empirical observation yields the following equation

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

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

If the P_{AES} is not available, the average or continuous power, known as P_{rms} can be used as well; 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 6 dB 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 (ref. TABLE 9).

Voice coil size (inches)		Threshold (W)	Attack time (ms)	Release time (ms)
1″	tweeter	10-20	100	300
1.5″	tweeter	20-30	150	300
2″	comp. driver	20-40	200	400
3″	comp. driver	30-50	300	500
4″	com. driver	40-60	500	3000
2″	midange	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

TABLE 9: Filters parameters.

- Power vs voltage @ 8 ohm: 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 150 W, a single cabinet will be delivered a maximum of 150 W with 8 ohm load. Two speaker cabinets connected in parallel will be delivered a maximum of 300 W 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 as well; 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 at 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 are available, the following rule of the thumb can be used: the P_{rms} can be estimated as 6 dB 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 3 dB of that value.

In order to use this limiter correctly, it is important to recalculate the equivalent power at 8 ohm. For example, with an 4 ohm speaker with 500 W maximum RMS power, the equivalent power at 8 ohm needs to be calculated as follow:

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

$$V_{rms} = \sqrt{Re \cdot P_{rms}}$$

   where V_{rms} is the RMS voltage of the speaker, P_{rms} is its average or continuous power and Re the nominal impedance. In the above example the RMS voltage of the 4 ohm speaker is $V_{rms} = 44.7$ V.

2. calculate the power delivered to a speaker with nominal impedance of 8 ohm with that V_{rms} voltage:

$$P_{eqiv} = \frac{V^2_{rms}}{8}$$

   where P_{equiv} is the equivalent power on the 8 ohm speaker and V_{rms} is the RMS voltage value calculated at step 1. In this example (4 ohm speaker) this is 250 W. This is the threshold power to set in the limiter.

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

- Power vs current @ 8 ohm: similar to the case power vs voltage @ 8 ohm, but based on the current measured at the output.

In this case the formula to derive the average or continuous power, known as P_{rms} from the RMS current is:

$$P_{rms} = I^2_{rms} \cdot Re$$

where 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.

The power limiter menu allows to set the following parameters:

- Mode: allows to determine the power limiter
  - OFF/ON: toggle le limiter on or off;
  - TruePower: sets the limiter mode to TruePower
  - Power vs V @ 8 Ω: sets the limiter mode to Power vs voltage @ 8 ohm
  - Power vs I @ 8 Ω: sets the limiter mode to Power vs current @ 8 ohm
- Soft knee: toggle ON/OFF
- Thresh.(W): threshold output power level expressed in watt 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 to 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 dB enforced by the combined effect of the peak and power limiters. This information, together with the average power truly delivered to the load (P_{avg}), is displayed in real time to allow monitoring of the limiting actions as they are performed.

9.11.7 Damping Control

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.

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.

Another advantage offered by the damping control feature is that in adding the series equivalent output resistance to the amplifier chain, the variation of the voice coil resistance 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.

Section area (mm² - AWG)	Nominal resistance	Length (ms)	Resistance (ohm)
2 x 1.5 - 16 AWG	R = 12 Ω/km	5	0,12
		10	0,24
		20	0,48
2 x 2.5 - 13 AWG	R = 7.4 Ω/km	5	0,07
		10	0,15
		20	0,30
2 x 4 - 11 AWG	R = 4.5 Ω/km	5	0,05
		10	0,09
		20	0,18

TABLE 10: Typical speaker cabling resistance.

On TABLE 11 notice the exceptionally high value (3.8 ohm) when the driver reaches it thermal limit.

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

TABLE 11: Typical resistance increase due to voice coil heating.

9.12 DSP Settings: Channel setup

9.12.1 Auxiliary delay

This delay is a further input delay: it acts before the input EQ and is independent from the input EQ stage.

9.12.2 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 relies 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.

9.12.2.1 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 freq: the frequency of the tone that has to be detected (range 20 Hz - 24 kHz, step of 10 Hz).
- Tone in Vmin: the minimum threshold value that has been detected (range 0 V_{rms} - 4 V_{rms}, step of 10 mV_{rms}).
- Tone in Vmax: the maximum threshold value that has been detected (range 0 V_{rms} - 4 V_{rms}, step of 10 mV_{rms}).

9.12.2.2 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 frequency 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 the generator (range 0 V_{rms} - 20 V_{rms}, step of 1 V_{rms}).
- 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).

9.12.2.3 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 minimum detected threshold voltage value (range 0 V_{rms} - 20 V_{rms}, step of 1 V_{rms}).
- Tone out Vmax: the maximum detected threshold voltage value (range 0 V_{rms} - 20 V_{rms}, step of 1 V_{rms}).

9.12.2.4 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 minimum 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).

9.12.2.5 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.

9.13 DSP Settings: 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.

9.14 DSP Settings: Reset input section

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

9.15 DSP Settings: 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.

10 Network operations

Network capabilities and network setting menus are available only for Bias Series amplifiers equipped with a AESOP board.

AESOP stands for AES3 and Ethernet Simple Open Protocol. Void's AESOP 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.

10.1 Introduction to AESOP

The AESOP standard can transport a single bidirectional Ethernet 100 Mbps control data stream and two separate AES3 digital audio monodirectional streams using one Cat5 cable.

All Bias Series amplifier with the optional AESOP board installed are equipped with at least two RJ45 connectors, each of them being 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 front panel. If four plugs are present, the rear two will be "primary" ports, while the two on the front panel are "secondary" ports.

Primary ports allow both data and AES3 streams; secondary ports, on the other hand, are data-only ports, allowing Ethernet connections only.

Cat5 standard twisted pair cables shall be used for connections up to 100 meters (328 ft). RJ45 pinout must comply to TIA/EIA-568-B and adopt the T568B scheme pinout, as show in TABLE 1 p. 23.

Please note that even if crossed Ethernet cables would work control wise, crossed cables are not to be used for AESOP connections: they will not allow the AES3 streams to flow correctly.

10.1.1 Data stream

The data stream in the AESOP is implemented by a 100 Mbit Ethernet connectivity with auto-sense.

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 dual port design in Bias 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.

The AESOP board detects bad quality connections by counting errors on the Ethernet control. Faulty connections are automatically switched from 100 Mbit/s to 10 Mbit/s to attempt to keep the link active even in the worst case scenarios.

10.1.2 Audio

Audio is distributed to devices via the AESOP protocol by 2 independent and separate AES3 streams labelled AES3-A stream, AES3-B stream. These are carried by two Cat5 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 Bias 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.

10.1.3 Ethernet internal switch

All control data streams in the AESOP system are transported via an Ethernet protocol. Inside all Bias 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.

10.1.4 Forwarding and repeater modes

Each Bias 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.
These are true connection "building blocks"; it is therefore important to understand these two modes thoroughly be-

fore 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: AESOP digital audio stream A (two channels)
- AES3-B stream: AESOP digital audio stream B (two channels)
- AES3-XLR stream: AES3 digital audio stream via the rear panel XLR connector.
- PORT 1, PORT 2: primary RJ45 AESOP ports
- PORT 3, PORT 4: secondary RJ45 Ethernet ports

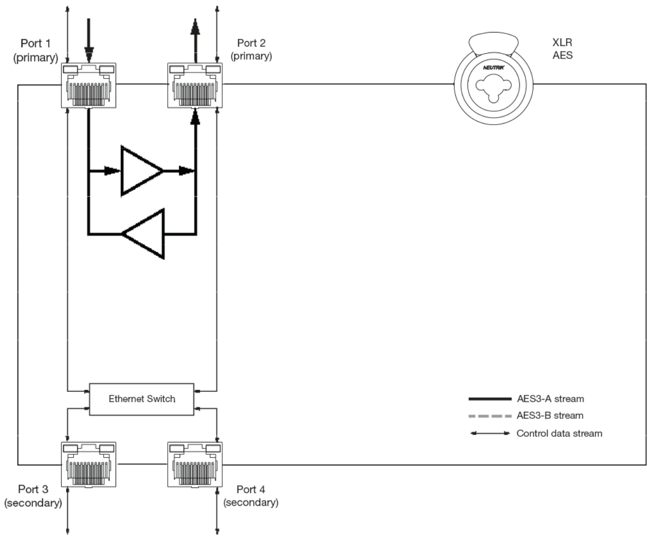


FIGURE 31: Repeat AES3-A from PORT 1 to PORT 2.

For consistency, primary ports are placed in the rear of the amp, while secondary ports are at the front. Notice that AES3 streams are monodirectional, while data stream is bidirectional.

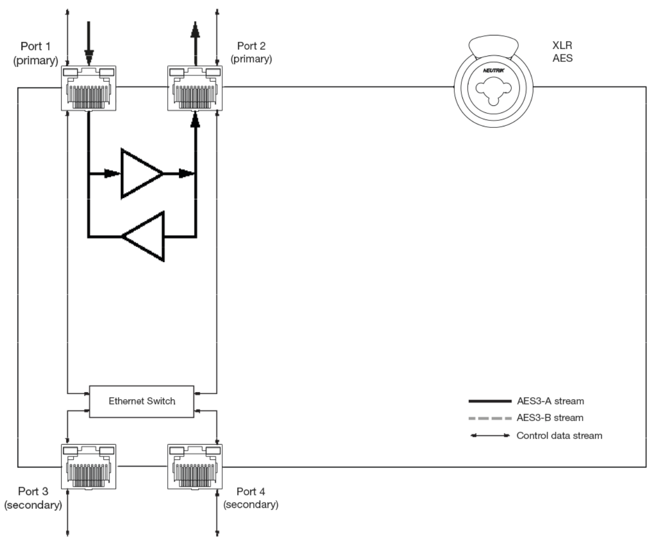


FIGURE 32: Repeat AES3-A from PORT 2 to PORT 1.

10.2 AESOP 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.

The Repeater mode is the default device mode setting. This applies to both AES3-A stream and AES-B stream 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.

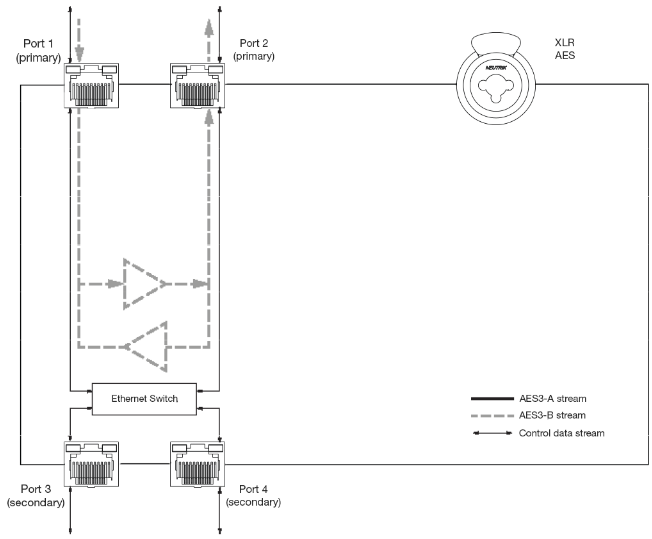


FIGURE 33: Repeat AES3-B from PORT 1 to PORT 2.

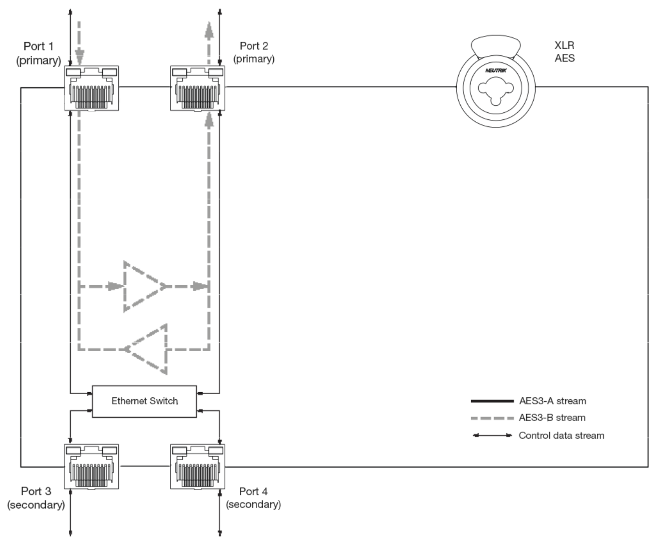


FIGURE 34: Repeat AES3-B from PORT 2 to PORT 1.

10.3 AESOP forward mode

When the amplifier is set in forward mode, the AES3 signal coming into the amplifier from the AES3-XLR connector is forwarded to both of the primary RJ45 ports.

The rear panel toggle button near to the channel 2 XLR connector must be in the "AES/EBU" position. There are three ways the AES can be forwarded: forward to AES3-A, forward to AES3-B, forward to both.

10.3.1 Forward to AES3-A

The amplifier's AES3-XLR connector will be routed to the AES3-A stream on both primary PORT 1 and 2 (FIGURE 36). If there is an AES3-B stream incoming from either primary RJ45 ports (1 or 2), this will be repeated on the other primary port (FIGURE 39).

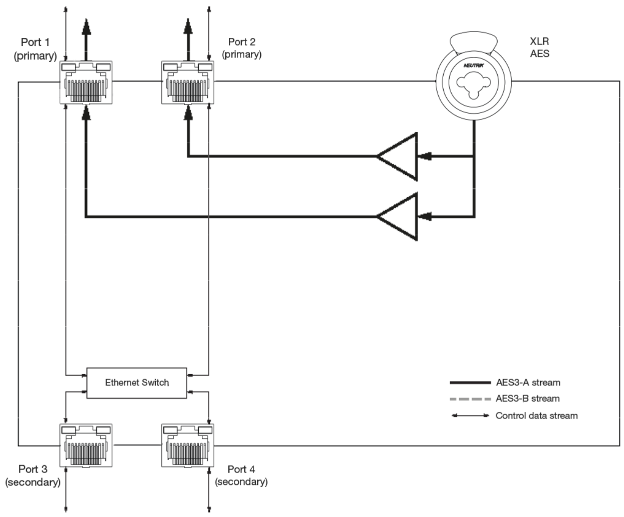


FIGURE 35: Forward AES3-XLR to AES3-A.

For consistency, primary ports are placed in the rear of the amp, while secondary ports are at the front. Notice that AES3 streams are monodirectional, while data stream is bidirectional.

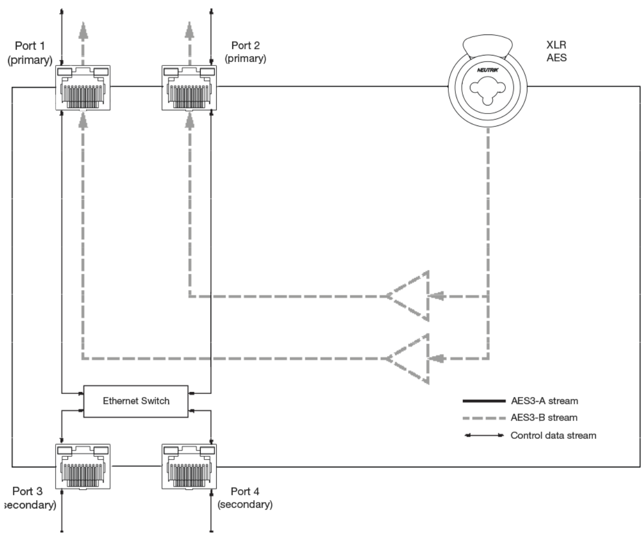


FIGURE 36: Forward AES3-XLR to AES3-B.

10.3.2 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-XLR stream will be routed to the AES3-B stream on both primary PORTS 1 and 2 (FIGURE 37). The AES3-A stream, if present will be repeated from/to primary RJ45 ports 1 and 2.

10.3.3 Forward to both

The amplifier's AES3-XLR stream will be routed to both AES3-A and AES3-B streams on both primary PORTS 1 and 2 (FIGURE 38).

Repeater functionality will be disabled.

IMPORTANT: In any forward mode, the amplifier can accept as the sole AES3 input signal the one coming from the AES3-XLR connector. The RJ45 ports cannot, input AES3 signals to the amplifier.

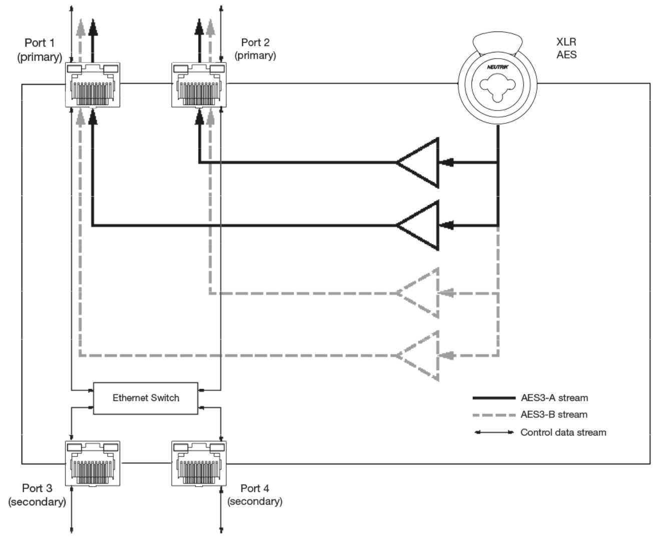


FIGURE 37: Forward AES3-XLR to both.
Repeat is disabled.

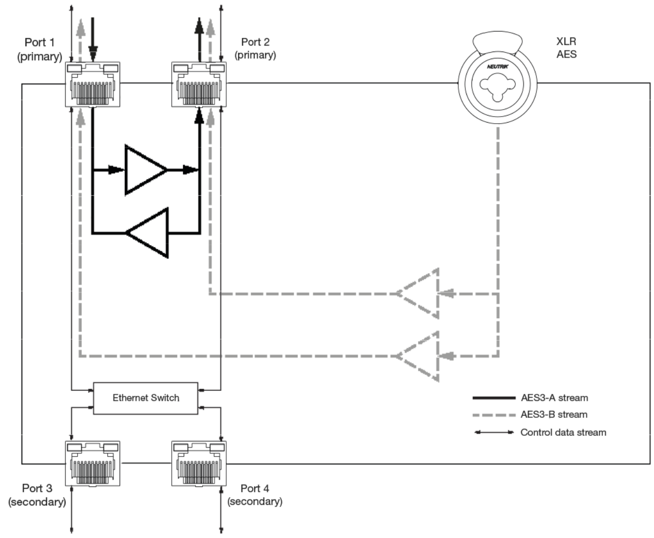


FIGURE 38: Forward AES3-XLR to AES3-B.
Repeat AES3-A from PORT 1 to PORT 2.

10.4 Network robustness

Bias Series amplifiers equipped with an AESOP are capable of being networked routing both data and audio streams to each other.

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 by means of some level of redundancy.

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.

Bias Series amplifiers support up to two degrees redundancy thanks to the adoption of proper networking topology: by detecting any connection failure on both analog and digital input Bias Series amplifiers are capable to automatically (and almost instantaneously) modify the audio feed direction to allow the output signal to remain uninterrupted.
The following section illustrates and analyses some common amplifier network topologies with different degree of redundancy.

10.4.1 Daisy chain

Daisy chain is a wiring scheme in which multiple devices are networked together in sequence, i.e. in series.

FIGURE 40 and FIGURE 41 show the diagrams of daisy chain connections of 4 amplifiers with four and two AESOP ports respectively.

In both configuration, only the first amplifier – feeded with the digital signal via the AES3-XLR – is set in forward mode: any other amplifier in the chain is a repeater.

Ethernet data are fed through any free port, either primary or secondary, and conveyed to the AESOP with the AES3.
This daisy chain topology is not robust (zero degree of redundancy). If any single AES3 or Ethernet cable connection is interrupted, the whole system fails.

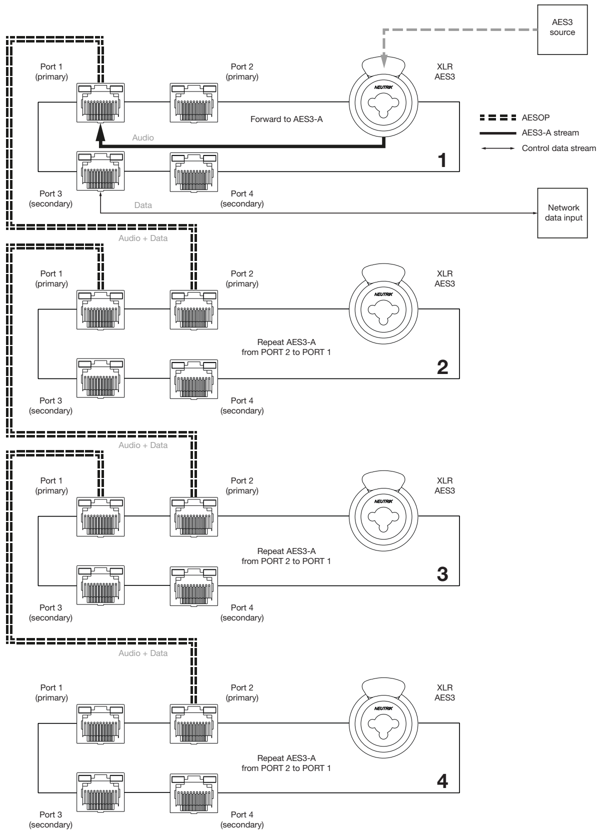


FIGURE 39: Daisy chain - four port models.

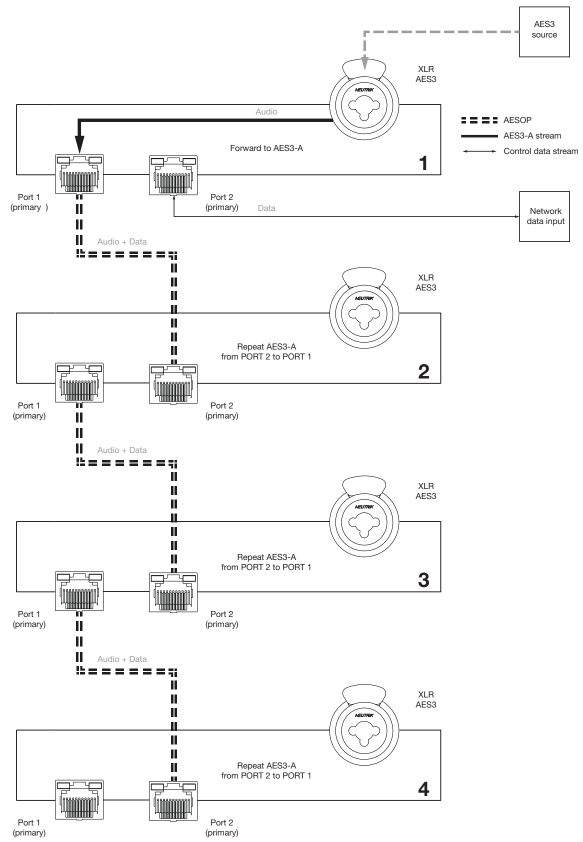


FIGURE 40: Daisy chain - two port models.

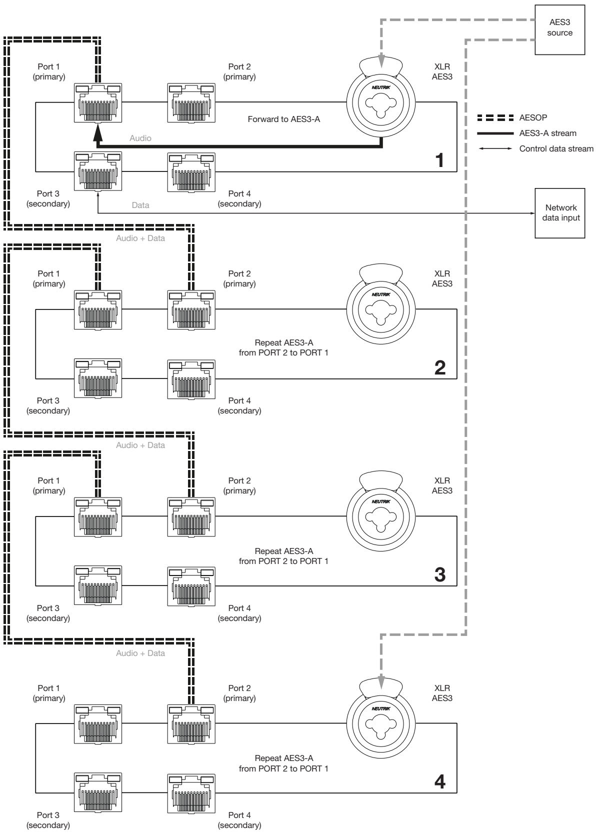


FIGURE 41: Daisy chain with AES3 redundancy - four port models.

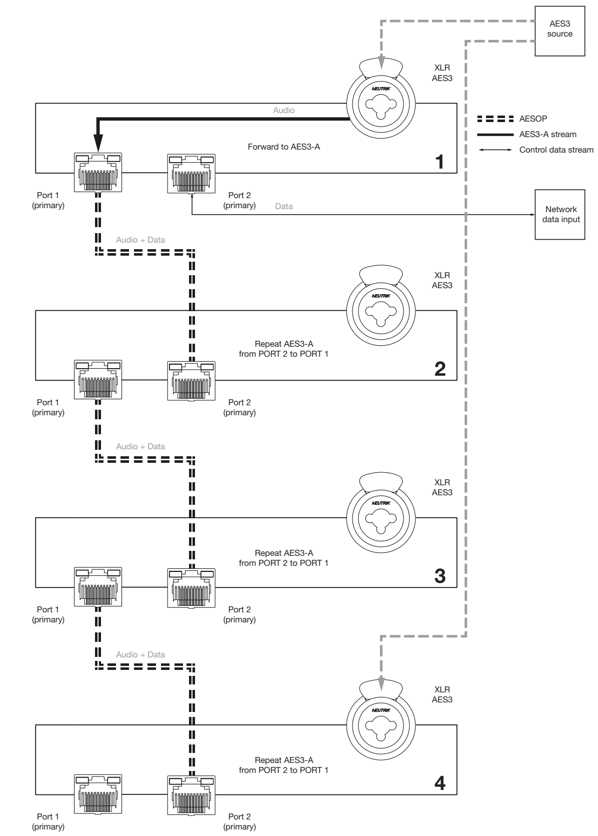


FIGURE 42: Daisy chain with AES3 redundancy - two port models.

10.4.2 Daisy chain with redundant AES3

A slightly more robust network with respect to the audio system is the one illustrated in FIGURE 42 and FIGURE 43.
Two amplifiers, the first and the last one in the chain, are set to work in forward mode. The remaining amplifiers are set to work in repeater mode.

Even if both the leading and the trailing amplifier forward the AES3 stream through the AESOP, there is no risk of data collision; furthermore, all amplifiers are capable to switch in real time to the best signal source in case of connection failures.

This configuration implies the use of an AES3 patch bay in order to feed with the same digital signal the leading and trailing amplifiers.

Thanks to the auto-sync features implemented in Bias Series amplifiers, no synchronization mismatch occur between the two AES3 streams.

Failure cases:

- damaged AESOP connection between amp n and n+1: Ethernet network connection would be interrupted but not the audio stream. The audio continuity is preserved thanks to the real-time switch of the AESOP stream toward the uncorrupted source coming from the trailing amplifier. The amplifier n+1 and the following lose the data connectivity.
- damaged AES3 input connection: no sound interruption would be heard because the failured input is immediately replaced by the AESOP stream. Ethernet connectivity is not affected by this kind of failure.

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

10.4.3 Daisy chain with AES3 and Ethernet redundancy

Similarly to the previous schema, both AES3 and Ethernet connection are fed to the leading and the trailing amplifiers in the daisy chain network.

The AESOP protocol can handle data conflict and manage real-time witching to a safe signal and/or data source.
This configuration implies the use of an AES3 patch bay and an ethernet switch.

Failure cases:

- damaged AESOP connection between amp n and n+1: the audio and ethernet continuity is preserved thanks to the real-time switch of the AESOP stream toward the uncorrupted source coming from the trailing amplifier.
- damaged AES3 input connection: no sound interruption would be heard because the failured input is immediately replaced by the AESOP stream. Ethernet connectivity is not affected by this kind of failure.
- damaged Etehrnet input connection: no data corruption would occur because the failured input is immediately replaced by the AESOP stream. The audio stream is not affected by this kind of failure.

Even if this network configuration implements both data and audio redundancy, its robustness is the same as that of the previous topology: Connectivity of data and audio is garanteed in case of 1 single cable failure only (redundancy of first failure type).

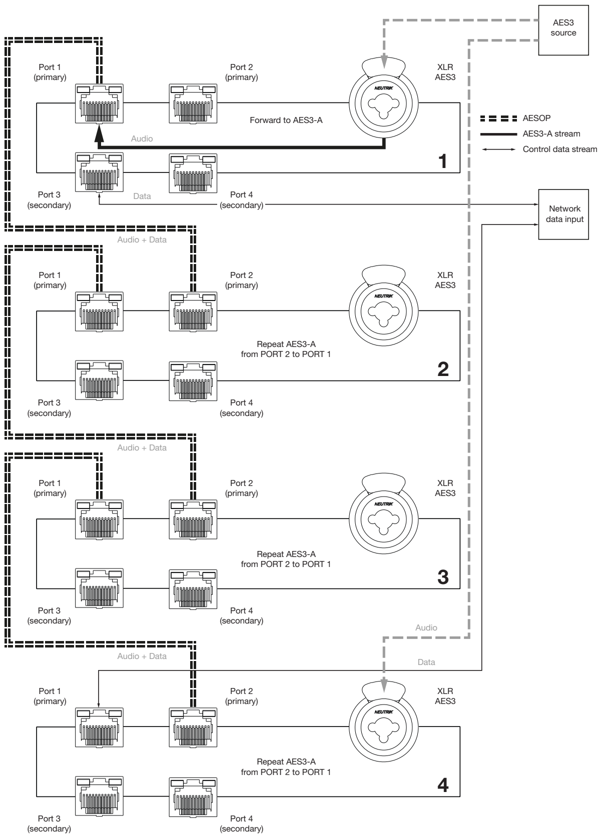


FIGURE 43: Daisy chain with AES3 and Ethernet redundancy - four port models.

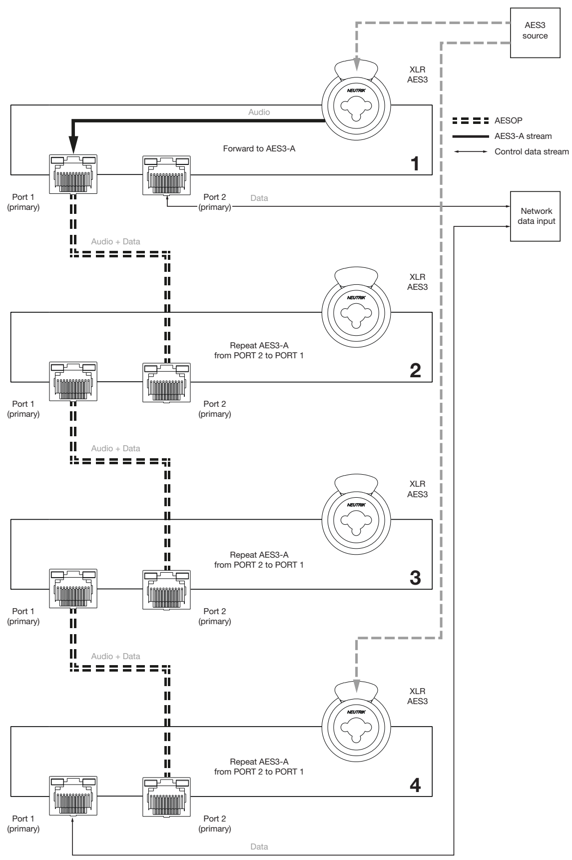


FIGURE 44: Daisy chain with AES3 and Ethernet redundancy - two port models.

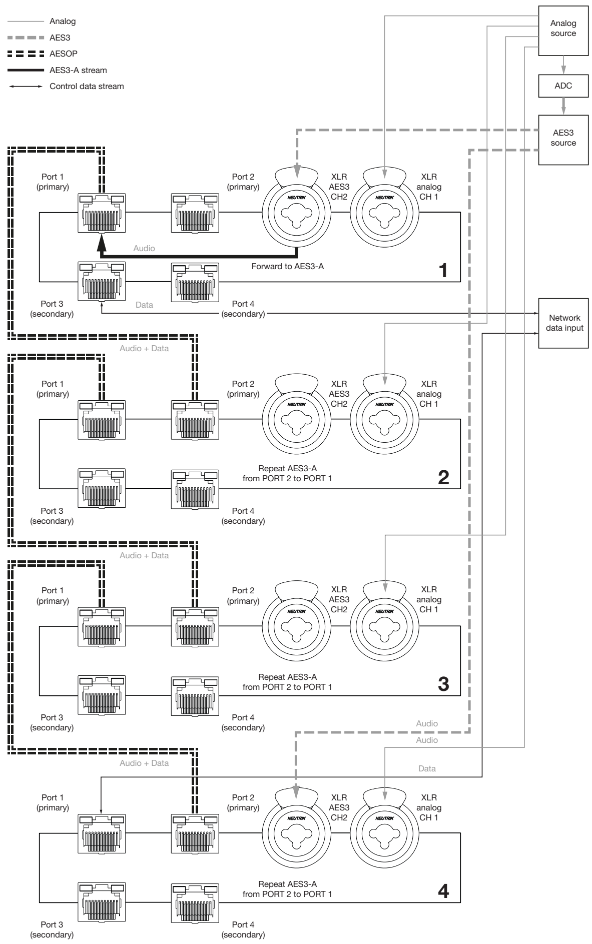


FIGURE 45: Two degree redundant daisy chain via AES3, Ethernet and Analog input.

10.4.4 Two degree redundant daisy chain

If the amplifiers in the daisy chain are fed with mono signal and the channel of each unit are linked – so that to use the same input signal (ref. Chapter 9:10.1. Source selection) –, a two degree redundant connection topology can be achieved.

Taking advantage of the "if no link" features in the Network settings menu (ref. Chapter 10:5.Network settings menu), the K Series can switch to the analog input when the AESOP stream fails. Bearing this in mind, it is possible to achieve high degree of redundancy exploiting both digital and analog inputs.

Remember that when operating with digital inputs – i.e. AES3 and AESOP – channel link must be achieved via software: do not switch the link pushbutton.

The network topology is described in FIGURE 46.

10.5 Network settings menu

The Network settings menu become available when the AESOP board is installed (ref. FIGURE 20).

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 is the selected option.

- Device mode: this parameter sets the amplifier mode with respect to the AES3 stream (ref. Chapter 10:2. AESOP repeater mode and Chapter 10:3.EASOP forward mode). Available options are:
  - Repeater (default); Forward to AES3-A;
  - Forward to AES3-B;
  - Forward to both.

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 AES3-XLR connector. AES3 streams incoming from any other RJ45 port are ignored.

- 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 behaviour complies with RFC 3927, guaranteeing the interoperability with any host PC supporting this standard.

- Set address: this menu allows to manually set the amplifier's IP address, subnet mask and default gateway.
- 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.
- Audio
  - Source selection: this menu allows the user to select the AES3 stream source to feed the output power stage. The AES3 signal can come from either: AES3-XLR, AES3-A or AES3-B.
  - 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 is forwarded to both amplifier channels), Parallel from R (the right channel from the selected AES3 stream 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).
  - Gain trim: this parameter trims the digital level of the AES3 stream. The gain trim scale goes from +5 dB to -40 dB with 0.5 dB steps with respect to 0 dB equivalent of +13.5 dBu. A 0 dBFS level in the AES3 stream corresponds to an absolute analog level of +18.5 dBu when a +5 dB gain trim level is applied.
  - If no link: this parameter allows the user to choose the behaviour of the amplifier when the digital audio stream is missing and the "Input selection" is set as AESOP → OUT (or AESOP → DSP →OUT). The two possible alternatives are: Mute and Analog. In Analog mode 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 (ref. Chapter 10:4.4. Two degree redundant daisy chain).

11 Display

The Display menu allows the user to monitor the system status and performance.

11.1 Display: 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.

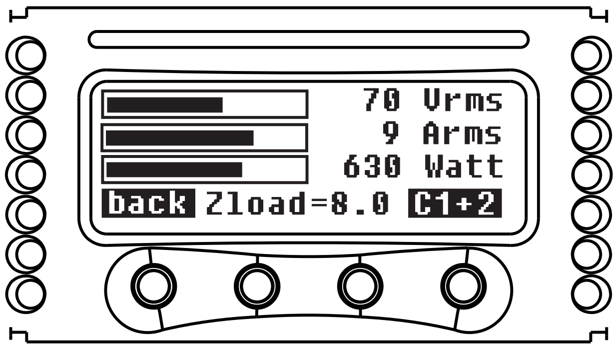


FIGURE 46: Display: Output meters.

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 200ms.

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.

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.

11.2 Display: Temperature

This screen displays the current amplifier temperature.

11.3 Display: Mains meters

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

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.

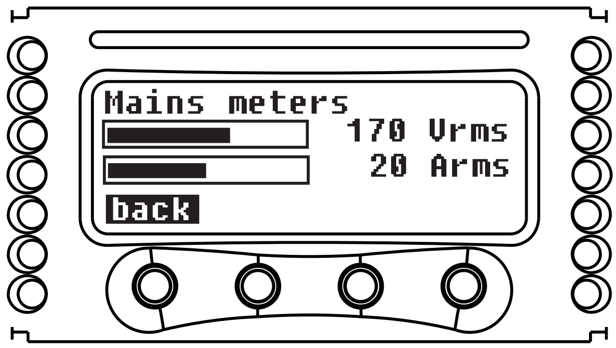


FIGURE 47: Display: Mains meters.

11.4 Display: 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.

12 Local presets

All Bias 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 DSP board settings if a DSP board is present.

12.1 Local preset: Locked presets

When the “locked presets” function is active, a number of presets, determined by the “Locked bank size” menu, is not overwritable. This function’s status can be toggled on/off by entering the Lock code.

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

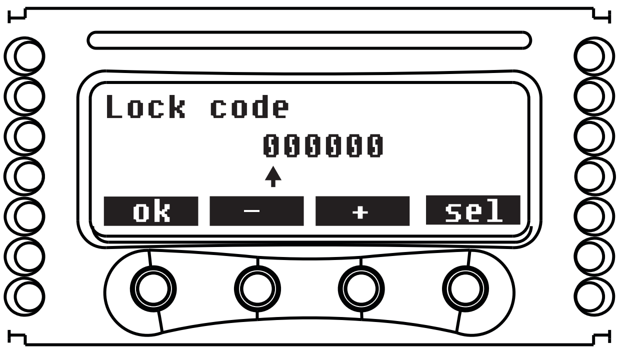


FIGURE 48: Lock code.

12.2 Local preset: 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.

12.3 Local preset: Recall local preset

In order to recall one of the 50 locally stored presets, press ok when the “Recall local preset” line is highlighted.
Navigate forwards or backwards in the existing presets list: press the right most button labelled “ok” to load the desired preset. If a preset location is not used, it is labelled <empty>.

12.4 Local preset:Save local preset

12.4.1Save to an empty slot

To save the current amplifier setup as a preset to the local memory, enter the “Save local preset” menu. Select an empty preset location, press “ok”, and define a proper name.
If no other preset has been loaded in the amplifier either via remote control or using a SmartCard, by default the current preset name will be “PRESET” followed by the selected memory slot name.

The preset name can be edited one character at a time.

12.4.2 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.

12.5 Local preset: Change lock code

In order to change the lock code used to activate the “Lock presets” function, the old user code must be entered.
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.

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

FIGURE 49: The “+” and “-” buttons allow to navigate within a standard set of capital letters and basic punctuation marks

12.6 Local preset: 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.

Pressing “ok” will erase all non protected presets; when all non write protected presets have been erased, a screen confirming this will appear.

13 Setup

13.1 Setup: 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: front panel firmware version.
- KCTRL: controller firmware version number.
- KDSP: DSP board firmware version (available only for models with the optional DSP board).
- AESOP: network board firmware version (available only for models with the optional AESOP board).
- Lifetime: operating hours of the amplifier (by default any brand new amplifier has 50 operating hours spent during the factory burn-in and initialization process).

13.2 Setup: 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 Setup; LCD contrast

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

13.4 Setup: Set 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

To activate the lock function, keep pressed for more than 1 second the button corresponding to the lock label: all other buttons will be 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 note 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.

13.5 Setup: 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.

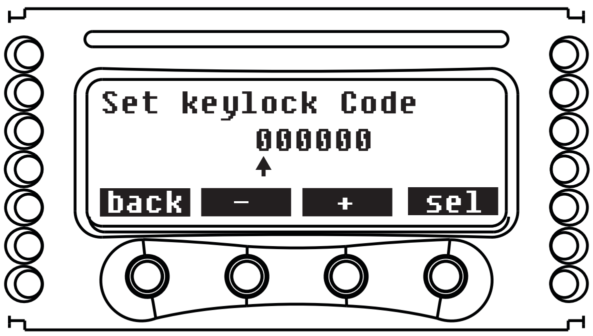


FIGURE 50: Setup: Set the keylock code.

14 System and signal protections

In order to protect your device and your speakers from accidental damage, Bias Series amplifiers include an extensive automatic protection system.

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 stop channel activity when the current drawn from the load rises above a set threshold.

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 maintain muted the channel and the procedure will reiterate every 2 seconds.

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 Bias 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 Software

15.1 Armonía Pro Audio Suite

Armonía Pro Audio Suite™ has been specifically designed to be used with Bias Series amplifiers as an easy to use configuring interface that allows system setting and customization.

Communication between the software and the amplifier is established via an RS-485 or Ethernet connection, depending on the available ports on the units.

Armonía provides control and monitoring of a wide range of amplifier functions, such as attenuation, mute, internal temperature and voltage rail monitoring.

On Bias Series equiped with the DSP board Armonía offers full control on all signal processing features, including input and output equalization, alignment delays, FIR filters and load impedance monitoring, etc.

Armonía is scalable: it allows control of a single Void 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.

Armonía
Pro Audio Suite™

This software has been designed to accept software plug-ins to enable third-party product control.

Armonía is freely available on Void website: it can be downloaded after signing up for the user forum: further information is available on the software section of Void website www.voidacoustics.com.

15.1.1 Networking

Void amplifiers can connect to a PC running Armonía in two ways: with an RS-485 serial connection or via Ethernet.

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 (bear in mind that Ethernet is a faster communications protocol than serial RS-485).

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 systems as well as a single hub 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 Void PowerHub as a local switch.

16 Warranty and assistance

16.1 Warranty

16.1.1 Product warranty

Void 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 Void's (or any of its Authorized Dealer's) invoice to the end customer. All warranty repairs and retrofits must be performed at Void facilities or at an Authorized Service Centre at no cost for the purchaser. Warranty exclusion: Void'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. Void will perform warranty services provided that the product is not damaged during transportation.

16.1.2 Return of Goods

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

16.1.3 Repair or replacement

Void reserves the right to repair or replace any defective goods covered by product warranty at its sole discretion and as it deems best.

16.1.4 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 Void or its Authorized Service Centre. Void will assume full responsibility and cover all costs incurred to send the goods back to the purchaser (or end user/customer).

16.2 Assistance

All servicing and repairs for Void Bias series amplifiers is handled by Powersoft Worldwide. Please follow the instructions below in case of any difficulties.

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 centres. 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 centre.

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" by contacting our Customer Care Department via email: service@powersoft.it or download the "Defect Report Form" from Powersoft's website (http://www.powersoft-audio.com/en/support/service).

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
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 TEMPORARY EXPORTATION / IMPORTATION PROCEDURE link at http://www.powersoft-audio.com/en/support/service.

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

Bias V9

Channel Handling	
Number of output channels	2 mono, bridgeable per ch. pair
Number of input channels:	
Analog	2x Combo XLR/TRS
AES3	1x XLR[1, 2]
	AESOP via 2x RJ45[2]
Number of output channels:	
Speaker	2x NL4MD speakON

Audio				
Gain	26 dB	29 dB	32 dB	35 dB
Input sensitivity @ 8 Ω	7.37 V	5.22 V	3.68 V	2.62 V
Max input level	27 dB	24 dB	21 dB	18 dB
Gate	-52 dBu	-55 dBu	-58 dBu	-61 dBu
Frequency Response (±0.5 dB , 1 W @ 8 Ω)	20 Hz - 20 kHz			
Crosstalk (1 kHz)	> 66 dB			
S/N ratio (20 Hz - 20 kHz A-Weighted @ 8 Ω)	> 110 dB			
Input impedance	10 kΩ balanced			
THD+N (from 0.1 W to Full Power)	< 0.5% (typical < 0.05%)			
DIM (from 0.1 W to Full Power)	< 0.5% (typical < 0.05%)			
Slew Rate (input filter bypassed @ 8 Ω)	> 50 V/μs			
Damping Factor @ 8 Ω, 20 Hz - 200 Hz	> 5000			

DSP[3]	
AD converters	24 Bit Tandem™ @ 96 kHz 127 dB-A Dynamic Range - 0.005 % THD+N
DA converters	24 Bit Tandem™ @ 192 kHz 122 dB-A Dynamic Range - 0.003 % THD+N
Sample rate converter	24 Bit @ 44.1 kHz to 192 kHz 140 dB Dynamic Range - 0.0001 % THD+N
Internal precision	40 bit floating point
Latency	6.0 ms fixed latency architecture
Memory/Presets	8 MB (RAM) plus 2 MB flash for presets: 50 stored locally + 150 stored on SmartCard
Delay	4 s (input) + 32 ms (output) for time alignment
Equalizer	Raised-cosine, custom FIR, parametric IIR: peaking, hi/lo-shelving, all-pass, band-pass, band-stop, hi/lo-pass
Crossover	linear phase (FIR), hybrid (FIR-IIR), Butterworth, Linkwitz-Riley, Bessel: 6 dB/oct to 48 dB/oct (IIR)
Limiters	TruePower™, RMS voltage, RMS current, Peak limiter
Damping control	Active DampingControl™ and LiveImpedance™ measurement

Networking	
Standards compliance	RS-485 serial connection or auto-sensing 10/100 Mbps UTP ports + AESOP[2]
Supported topologies	star, daisy-chain, closed loop[2]
Remote interface	Armonía Pro Audio Suite™
Ports	
Non AESOP models	Rear: 1 x Rj45 (RS-485 + V Ext)
2 port AESOP[2] models	Front: 2 x Rj45 (Ethernet + AESOP connection)
4 port AESOP[2] models	Front: 2 x Rj45 (Ethernet) Rear: 2 x Rj45 (Ethernet + AESOP connection)
Auxiliary supply[3]	12 V / 1 A max for DSP management and remote on/off via RJ45 or 2 pin Phoenix[2] MCV 1,5/ 2-G-3,81

Output Stage	
Maximum output power per channel @ 8 Ω	2700 W
Maximum output power per channel @ 4 Ω	5200 W
Maximum output power per channel @ 2 Ω	9000 W
Maximum output power @ 8 Ω Bridged	10400 W
Maximum output power @ 4 Ω Bridged	18000 W
Peak total output, all channels driven	18000 W
Maximum unclipped output voltage	225 V_{peak}
Maximum output current	125 A_{peak}

The power figure is calculated by driving and loading symmetrically all the channels: uneven loads allow to achieve highest performance.

AC Mains Power				
Power supply	Universal regulated switch more with PFC			
Nominal voltage (±10%)	100-240 V @ 50-60Hz			
Power factor (> 500 W ouput)	> 0.95			
Consumption/current draw	@ 115 V		@ 230 V	
Idle	91 W	1.3 A	88 W	1.17 A
1/8 Max Output Power @ 4 Ω	1650 W	15.8 A	1625 W	7.9 A
1/4 Max Output Power @ 4 Ω	3250 W	29.3 A	3250 W	14.7 A
AC Mains connector	AMP CPC 45A connector - 45 A max (region-specific power cord provided)			

Thermal		
Operating temperature	0° - 45° C / 32° - 113° F	
Cooling	Fan, continuously variable speed, temperature controlled, front to rear airflow	
Thermal dissipation		
Idle	682 BTU/h	171.9 kcal/h
1/8 Max Output Power @ 4 Ω	1590 BTU/h	400.7 kcal/h
1/4 Max Output Power @ 4 Ω	2498 BTU/h	629.5 kcal/h

Construction	
Dimensions	483 mm x 44.5 mm x 475 mm (19.0 in x 1.8 in x 18.7 in)
Weight	12 kg (26.5 lb)

[1] Common to channel 2 XLR analog input, either analog or AES3 depending on system configuration.
[2] Available only in AESOP equipped models.
[3] Only for DSP equipped model

Bias V3

Channel Handling

Number of output channels	2 mono, bridgeable per ch. pair
Number of input channels:	
Analog	2x Combo XLR/TRS
AES3	1x XLR[1,2]
	AESOP via 2x RJ45[2]
Number of output channels:	
Line out (through)	2x XLR
Speaker	2x NL4MD speakON

Audio

Gain	26 dB	29 dB	32 dB	35 dB
Input sensitivity @ 8 Ω	5.30 V	3.75 V	2.66 V	1.88 V
Max input level	27 dB	24 dB	21 dB	18 dB
Gate	-52 dBu	-55 dBu	-58 dBu	-61 dBu
Frequency Response (±0.5 dB , 1 W @ 8 Ω)	20 Hz - 20 kHz			
Crosstalk (1 kHz)	> 70 dB			
S/N ratio (20 Hz - 20 kHz A-Weighted @ 8 Ω)	> 106 dB			
Input impedance	10 kΩ balanced			
THD+N (from 0.1 W to Full Power)	< 0.3% (typical < 0.05%)			
DIM (from 0.1 W to Full Power)	< 0.3% (typical < 0.05%)			
Slew Rate (input filter bypassed @ 8 Ω)	> 50 V/µs			
Damping Factor @ 8 Ω, 20 Hz - 200 Hz	> 5000			

DSP[3]

AD converters	24 Bit Tandem™ @ 96 kHz 127 dB-A Dynamic Range - 0.005 % THD+N
DA converters	24 Bit Tandem™ @ 192 kHz 122 dB-A Dynamic Range - 0.003 % THD+N
Sample rate converter	24 Bit @ 44.1 kHz to 192 kHz 140 dB Dynamic Range - 0.0001 % THD+N
Internal precision	40 bit floating point
Latency	6.0 ms fixed latency architecture
Memory/Presets	8 MB (RAM) plus 2 MB flash for presets: 50 stored locally + 150 stored on SmartCard
Delay	4 s (input) + 32 ms (output) for time alignment
Equalizer	Raised-cosine, custom FIR, parametric IIR: peaking, hi/lo-shelving, all-pass, band-pass, band-stop, hi/lo-pass
Crossover	linear phase (FIR), hybrid (FIR-IIR), Butterworth, Linkwitz-Riley, Bessel: 6 dB/oct to 48 dB/oct (IIR)
Limiters	TruePower™, RMS voltage, RMS current, Peak limiter
Damping control	Active DampingControl™ and LiveImpedance™ measurement

Networking

Standards compliance	RS-485 serial connection or auto-sensing 10/100 Mbps UTP ports + AESOP[2]
Supported topologies	star, daisy-chain, closed loop[2]
Remote interface	Armonía Pro Audio Suite™
Ports	
Non AESOP models	Rear: 1 x Rj45 (RS-485 + V Ext)
2 port AESOP[2] models	Front: 2 x Rj45 (Ethernet + AESOP connection)
4 port AESOP[2] models	Front: 2 x Rj45 (Ethernet) Rear: 2 x Rj45 (Ethernet + AESOP connection)
Auxiliary supply[3]	12 V / 1 A max for DSP management and remote on/off via RJ45 or 2 pin Phoenix[2] MCV 1,5/ 2-G-3,81

Output Stage

Maximum output power per channel @ 8 Ω	1400 W
Maximum output power per channel @ 4 Ω	2600 W
Maximum output power per channel @ 2 Ω	2800 W
Maximum output power @ 8 Ω Bridged	5200 W
Maximum output power @ 4 Ω Bridged	5600 W
Peak total output, all channels driven	5600 W
Maximum unclipped output voltage	165 V_{peak}
Maximum output current	102 A_{peak}

The power figure is calculated by driving and loading symmetrically all the channels: uneven loads allow to achieve highest performance.

AC Mains Power

Power supply	Universal regulated switch more with PFC			
Nominal voltage (±10%)	100-240 V @ 50-60Hz			
Power factor (> 500 W ouput)	> 0.95			
Consumption/current draw	@ 115 V		@ 230 V	
Idle	64 W	1.12 A	75 W	1.3 A
1/8 Max Output Power @ 4 Ω	813 W	8 A	813 W	4 A
1/4 Max Output Power @ 4 Ω	1625 W	14.8 A	1625 W	7.4 A
AC Mains connector	AMP CPC 45A connector - 45 A max (region-specific power cord provided)			

Thermal

Operating temperature	0° - 45° C / 32° - 113° F	
Cooling	Fan, continuously variable speed, temperature controlled, front to rear airflow	
Thermal dissipation		
Idle	382 BTU/h	96.3 kcal/h
1/8 Max Output Power @ 4 Ω	836 BTU/h	210.7 kcal/h
1/4 Max Output Power @ 4 Ω	1390 BTU/h	350.3 kcal/h

Construction

Dimensions	483 mm x 44.5 mm x 380 mm (19.0 in x 1.8 in x 15 in)
Weight	8 kg (17.7 lb)

[1] Common to channel 2 XLR analog input, either analog or AES3 depending on system configuration.
[2] Available only in AESOP equipped models.
[3] Only for DSP equipped model

North America

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