SONIFEX

Dante® & AES67/RAVENNA AoIP Products

Catalogue

SONIFEX
8 AES XLR Stereo Input & Output AoIP Distribution, Dual PoE
Dante
avn AESIO8R
Clock
PoE
Link
Pri
Sec
Reset
AES Lock
1 2 3 4
5 6 7 8
1 2 3 4
1 2 Outputs 3

AoIP – Dante®/AES67 Audio over IP Products

Dante®

Dante® Commentators

AVN-CU2-DANTE	Configurable Dante Commentary Unit for 2 Commentators	2
AVN-CU4-DANTE	Configurable Dante Commentary Unit for 4 Commentators	6

Multi-Channel Dante® Audio Interfaces

AVN-AIO4	4 Input, 4 Output Dante® Interface, PoE	11
AVN-AIO8	8 Input, 8 Output Dante® Interface, PoE	13
AVN-AIO8R	8 Input, 8 Output, Dual Dante® Interface, PoE	15
AVN-AI16	16 Input Dante® Interface, PoE	17
AVN-AI16R	16 Input Dual Dante® Interface, PoE	19
AVN-AO16	16 Output Dante® Interface, PoE	21
AVN-AO16R	16 Output Dual Dante® Interface, PoE	23
AVN-AESIO8	8 AES3 Input, 8 AES3 Output Dante® Interface, PoE	25
AVN-AESIO8R	8 AES3 Input, 8 AES3 Output Dual Dante® Interface, PoE	27

AVN-DIO Audiophile Dante® Audio Interfaces

AVN-DIO01	Dante to Analogue XLR Stereo Output	29
AVN-DIO02	Analogue XLR Stereo Input to Dante®	31
AVN-DIO03	Dante to Headphone Outputs with Volume Control & Limiter	33
AVN-DIO04	Dante® to Analogue Phono Stereo Input & Output	35
AVN-DIO05	Dante® to Analogue Terminal Block Stereo Input & Output	37
AVN-DIO06	Dante® to AES3 XLR Stereo Input & Output	39
AVN-DIO07	Dante® to AES-3id BNC Stereo Input & Output	41
AVN-DIO08	Dante® to AES3 Terminal Block Stereo Input & Output	43
AVN-DIO09	Microphone Input to Dante®	45
AVN-DIO10	Dante® to 3G/HD/SD-SDI Embedder/De-Embedder	47
AVN-DIO19	Dante® to AES3 16 Channel I/O Converter	49
AVN-DIO20	Dante® to MADI AES3 64 Channel I/O Converter	51
AVN-DIORK	19” AVN-DIO Mounting Rack	53

RAVENNA

IEEE1588 PTP Grandmaster Clock with GPS Receiver

AVN-GMCS	IEEE1588 PTP Grandmaster Clock with GPS Receiver	54
AVN-GMCOS	IEEE1588 PTP Grandmaster Clock with GPS Receiver	57

Talkback Intercoms Using Audio Over IP, RAVENNA/AES 67

AVN-TB6	6 Button Talkback Intercom	60
AVN-TB6D	6 Button Talkback Intercom	62
AVN-TB10AR	10 Button Advanced Talkback Intercom, AoIP Portal	64
AVN-TB20AR	20 Button Advanced Talkback Intercom, AoIP Portal	67
AVN-TB20AD	20 Button Advanced Talkback Intercom, AoIP Desktop Portal	70

12 Channel Mix Monitor, AoIP Portal

AVN-PXH12	12 x 2 Channel Mix Monitor, AoIP Portal	73

Multi-Channel Audio Mix Engine Interfaces Using AES67 AoIP - the AVN Portals

AVN-PA8,	8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs	76
AVN-PA8T,	8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs on Terminal Blocks, AES67 Portal	79
AVN-PA8D,	8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs, AES67 Portal, with Detailed Meter Displays	82
AVN-PA8TD	8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs on Terminal Blocks, AES67 Portal, with Detailed Meter Displays	85
AVN-PD8,	8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs, AES67 Portal	88
AVN-PD8T	8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs on Terminal Blocks, AES67 Portal	91
AVN-PD8D,	8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs, AES67 Portal, with Detailed Meter Displays	94
AVN-PD8TD	8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs on Terminal Blocks, AES67 Portal, with Detailed Meter Displays	97
AVN-PM8	8 Mic/Line Inputs, 8 Stereo Analogue Line Outputs, AES67 Portal	100
AVN-PM8T	Advanced Audio Routing, Metering and Equalisation Unit with Terminal Type Analogue Mic/Line Inputs, Terminal Type Analogue Outputs, and RAVENNA AoIP	103
AVN-PM8D	Advanced Audio Routing, Metering and Equalisation Unit with Analogue Mic/Line Inputs, Analogue Outputs, RAVENNA AoIP, and a Detailed Customisable Display	106
AVN-PM8TD	Advanced Audio Routing, Metering and Equalisation Unit with Terminal Type Analogue Mic/Line Inputs, Terminal Type Analogue Outputs, RAVENNA AoIP, and a Detailed Customisable Display	109

Headphone Amplifiers

AVN-HA1	Analogue Headphone Amp for AVN-PA8/D & AVN-PM8/D Portals	112
AVN-HD1	Digital Headphone Amp for AVN-PD8/D Portal	113

Presenter In-Ear Monitoring System Using AES67 AoIP

AVN-MPPR	4 Channel Presenter In-Ear Monitoring Remote Controller, AES67	114
AVN-MPTR	Technician Remote Controller	116
AVN-PX8X4C 8 x 4	Channel Mix Engine, 24 Inputs, 16 Outputs, AoIP AES67	118

Key to Symbols:
When the symbol is active it is gold, when inactive, it is grey.

AES67/Dante AoIP Products

Commercial Infrastructure & Security

Education

Post Production

Pro Audio & Live Sound/PA

Radio Broadcasting

TV Broadcasting & OB Vans/ Trucks

Controlled via RS232 or Ethernet using the free of charge Sonifex Control Interface (SCi) software.

Built-in web server for control via a web browser.

AVN-CU2-DANTE Configurable Dante Commentary Unit for 2 Commentators

Category: Dante Commentary.

Product Function: Commentator unit for two commentary positions with 16 x 16 Dante matrix mixers.

Typical Applications: Used in sports grounds and arenas by commentators to produce commentary & hear talkback, 16 x 16 matrix mixer with configurable controls for live events.

Features:

- Web GUI for configuration.
- Full web GUI remote control.
- Lock-out of all individual controls and/or button row/encoder pages.
- PoE using 2 x Neutrik EtherCON® RJ45s.
- Dual redundant network ports on RJ45 and SFP.
- 2 x locking mic/line inputs with +48V phantom power indication.
- 2 x headphone outputs on locking 6.35mm jack sockets.
- 1 x stereo analogue/digital input and 1 x stereo analogue/digital output on RJ45s.
- 4 x talkback buses.
- On-air buttons with lazy talkback ability.
- Supports up to 16 x input and output streams and 16 input and output multicast flows/streams.
- Configurable headphone source selection.
- 6 x rotary encoders to control volume, or signal panning, and colour coded with the screen.
- 6 x detailed colour displays showing channel name, level metering & limit indication.
- Fully configurable buttons and rotary encoders, for inputs and outputs.
- 4 x pages of 6 x encoder settings, e.g. one for talkback inputs, one for outputs, one to monitor other sources.
- Metering per input/output, with output configured as pre or post level adjustment.
- Built-in line-up tone generator.
- Limiter available on every output.
- Output metering section shows limiter indication, the name of the output & level metering.
- 10 x configurable GPIO with 1 x switched changeover output.
- GPIO & VGPIO matrix programmable with events.

The AVN-CU2-DANTE takes a new approach to provide a multi-purpose configurable tool for commentary teams. Its power lies in the impressive mix engine which overlays the usual Dante® Controller settings. Once Dante® flows have been made, inputs and outputs can be mixed freely to AoIP or physical inputs and outputs, controlled using the programmable buttons and rotary encoders, which control the gain and pan of inputs, outputs or cross-points, allowing total flexibility for different situations.

AVN-CU2-DANTE Iso View.

It can be used in any number of different commentary situations, controlled manually or remotely and controls can be 'locked down' so that they can't be tampered with or altered, to guarantee reliable operation. Housed in a rugged and intuitive, user-friendly package, it's a truly different way of looking at how commentary units should operate.

Traditional commentary units have fixed analogue and digital I/O and fixed controls in fixed positions on the unit. Their inputs and outputs are defined at hardware design and are thus limited by that initial design, including limited routing, mixing and DSP of the audio pathways.

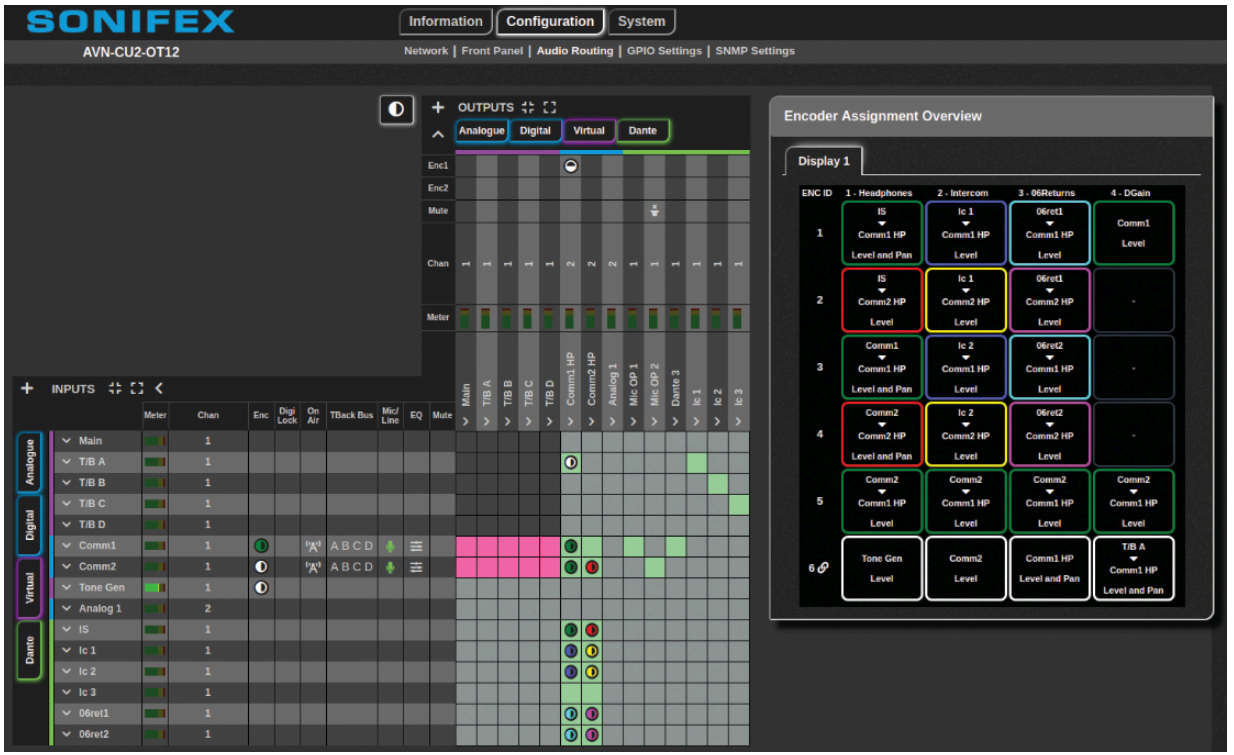
We've taken a different approach with the AVN-CU2-DANTE. It was designed from the ground up to be totally flexible in operation and the use of Dante AoIP means that inputs and outputs can be added as required (up to a max of 16 per unit). Because any physical analogue or digital

input can be mixed and routed with any Dante AoIP input to any physical output or Dante output, you can define your own audio pathways. Additionally you can choose which of those pathways need to be controlled (volume and pan) by the use of rotary encoders.

Using a built-in web GUI, up to 4 nameable pages of 6 rotary encoders (24 in total) can be placed on the mix matrix at inputs, outputs or cross-points. Encoders can control the volume and pan of headphones and volume of inputs, outputs or cross-points. Each rotary encoder has a separate colour-coded meter section showing the channel name, detailed level metering and left/right panning on a bright daylight reading display. Colours can be programmed per encoder to quickly identify particular source groups, so headphone source selection becomes intuitive.

The unit can be fully remotely controlled from the web interface, which mimics the front panel of the unit and includes live metering. All the controls can be operated as if at the unit and there are front-panel lock-out options for every button and encoder, with 'quick lock-out' for the rows of buttons and pages of encoders. This allows you to configure the unit for simple operation by the talent but with more complex setups, say, for mixing the talkback or feeds from an OB truck.

The AVN-CU2-DANTE provides two mic/line inputs with a wide, adjustable gain range and has two stereo headphone outputs with



AVN-CU2-DANTE GUI Audio Routing Page.

lockable jack sockets, suitable for operation by two commentators.

It's powered using Power over Ethernet (PoE), using Neutrik EtherCON® connectors, with primary and secondary ports for power and data redundancy. There's an additional 4 pin XLR 12V DC input. The unit supports up to 16 input and output AoIP channels and up to 16 simultaneous input and output AoIP streams.

The 6 push-button rotary encoders control input and output levels and panning. The 12 key-cap buttons are fully configurable for any button function and can have different colours for the on and off functions.

Metering is available per input/output, with output metering configurable as pre or post level adjustment. The top of the display shows output metering, a limiter indication and the name of the output. An adjustable limiter is available on every output and is applied automatically to prevent signal clipping.

The unit has 2 x locking mic/line inputs with +48V phantom power indication and 2 x headphone outputs on locking 6.35mm jack sockets.

Fourwire I/O on rear panel RJ45 connectors provide an AES3 or analogue input and output that can be assigned as mic outputs ▶

AVN-CU2-DANTE Top View.

AVN-CU2-DANTE GUI SNMP Page.

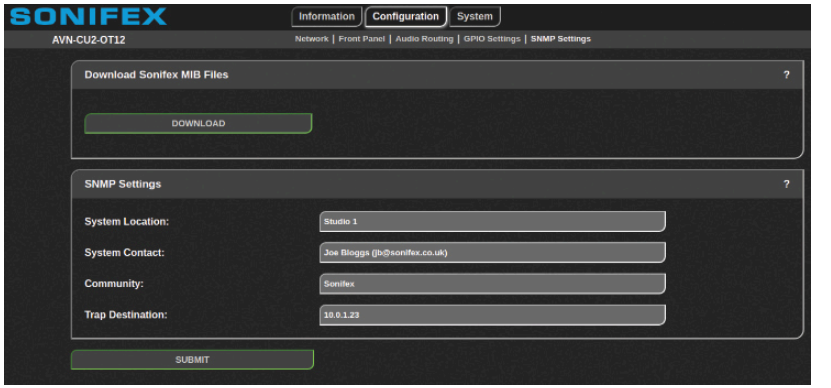
(line level), talkback outputs, programme inputs or talkback inputs as desired. In addition, the AES/analogue connections can be used as an insert or exit point into/out from the AoIP network.

The unit has dual redundant network ports on both RJ45 (PoE using 2 x Neutrik EtherCON® connectors) and SFP cages for long fibre runs.

There are 10 x configurable GPIO on a 15 way D-type connector with 1 x switched changeover output. GPIO & VGPIO (virtual GPIO, i.e. network commands) can be configured on a matrix to visually show actions, combined with button presses and event triggers, such as input/output muting, which enables some automation. Button presses can control buttons and functions on other network connected AVN-CU units using VGPIO.

The complete configuration of the unit, including all encoder settings, button setups and Dante mix matrix audio routes, can be saved off to a text editable .json file. Up to 8 of these files can be stored on the unit and loaded from the main Menu, meaning that the unit can be pre-configured for any venue, or event and with these setups instantly recalled onsite.

"All of the buttons have key-cap text and can be configured for any button in any position. There are some standard operations available:



AVN-CU2-DANTE GUI SNMP Page.

- 2 x On-Air buttons can be used to connect mic audio to the main output, either over AoIP or via the AES digital audio connection. The On-Air buttons can be locked if required.
- A Menu button can be used to access limited setup options on the TFT display.
- 2 x Page buttons change the display and encoders to monitor an additional set of sources, mix points or outputs. Up to 4 pages can be pre-programmed, e.g. one for talkback inputs, one for outputs, one to monitor other sources.
- 2 x Cough buttons take the commentator off-air while pressed.
- A User button can be programmed to perform any function, using the web server.
- 4 x T/B (talkback) buttons can be configured to initiate talkback over AoIP or AES digital audio connection, using 4 x talkback buses. The talkback buttons operate with lazy talkback, taking the commentator off-air when invoked.
- 6 x rotary encoders.
- 4 x pages of 6 rotary encoder positions (24 in total).

The illuminated 'Sonifex' logo acts as a power indication and there are illuminated LEDs to indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary, PoE Secondary and DC power active. ▶

Technical Specification For AVN-CU2-DANTE

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 receive, 16 transmit
Number of Streams:	16 receive, 16 transmit
Sample Rate:	48 kHz
Format:	Linear PCM, 16 or 24 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45 and 2 x SFP
Speed:	1Gbps and 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Mic/Line Inputs

Input Impedance (Mic Mode):	2.5kΩ electronically balanced
Input Impedance (Line Mode):	>10kΩ electronically balanced
Preamp Gain (Mic Mode):	User selectable 16dB to 76dB in 3dB steps (Ref. +18dBu = 0dBFS)
0dBFS Line-up (Mic Mode):	-58dBu @ Max preamp gain, +2dBu @ Min preamp gain
0dBFS Line-up (Line Mode):	User selectable +15dBu / +18dBu / +20dBu / +22dBu / +24dBu = 0dBFS
Frequency Response (Mic Mode):	+0/-0.2dB 20Hz to 20kHz Ref 40dB gain @ 1kHz
Frequency Response (Line Mode):	+0/-0.2dB 20Hz to 20kHz Ref 0dBu @ 1kHz
THD+N (Mic Mode):	<-90dBFS, -64dBFS, 20Hz – 20kHz, 40dB gain, 20kHz BW
THD+N (Line Mode):	<-98dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Equivalent Input Noise (Mic Mode):	125dB Ref. 76dB preamp gain, Rs = 200Ω
Noise (Line Mode):	<-100dBFS, 20kHz BW, Rs = 200Ω
C.M.R.R.	>60dB @ 1kHz
Phantom Power (Mic Mode):	+48V
High Pass Filter:	12dB/octave, user selectable frequency

Stereo Analogue/Digital Input

Input Impedance (Analogue Mode):	>20kΩ electronically balanced
Input Impedance (Digital Mode):	110Ω
Supported Input Rates (Digital Mode):	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz
0dBFS Line-up (Analogue Mode):	User selectable +15dBu / +18dBu / +20dBu / +22dBu / +24dBu = 0dBFS
Frequency Response (Analogue Mode):	+0/-0.2dB 20Hz to 20kHz Ref 0dBu @ 1kHz
THD+N (Analogue Mode):	<-107dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise (Analogue Mode):	<-108dBFS, 20kHz BW, Rs = 200Ω
C.M.R.R (Analogue Mode):	>60dB @ 1kHz

Stereo Analogue/Digital Output

Output Impedance (Analogue Mode):	<50Ω
Output Impedance (Digital Mode):	110Ω
Supported Output Rates (Digital Mode):	48kHz
0dBFS Line-up (Analogue Mode):	User selectable 0dBFS = +15dBu / +18dBu / +20dBu / +22dBu / +24dBu
Frequency Response (Analogue Mode):	+0/-0.5dB 20Hz to 20kHz Ref 0dBu @ 1kHz
THD+N (Analogue Mode):	<-107dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise (Analogue Mode):	<-107dBFS, 20kHz BW

Headphone Outputs

Drive Capability	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-up:	Fixed 0dBFS = +18dBu
Frequency Response:	+0/-0.2dB 20Hz to 20kHz Ref 0dBu @ 1kHz
THD+N:	<-107dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise:	<-110dBFS, 20kHz BW

Main Panel Operational Controls & Indicators

Power LED	Sonifex logo illuminates when power is present
Pushbuttons:	12 x Illuminated buttons that can be assigned to any of the following functions, and configured: On-Air / Talkback / Page # / Page Cycle / GPO / Cough / Menu / Brightness
Display:	480 x 107 pixel colour TFT display showing pan status, level, metering, control type, source/ mix/destination names and main output metering
Status LEDs:	Clock / AoIP status / PoE status and PSU status LEDs to monitor power and connectivity. Can be disabled if required
Rotary Encoders:	6 x robust soft touch rotary encoders with adjacent configurable light bars for group/ source/commentator grouping and identification

Front Panel Connections

Headphone Outputs:	2 x Neutrik ¼" (6.35mm) locking jack sockets
Microphone/Line Inputs:	2 x Neutrik 3 pin latching XLR sockets

Rear Panel Connections

Stereo Analogue/ Digital Input:	RJ45 socket (electronically balanced)
Stereo Analogue/ Digital Output:	RJ45 socket (electronically balanced)
GPIO Port:	15-way 'D'-type socket
DC Input:	Neutrik 4 pin XLR plug, 12VDC, Maximum 12W
Network:	2 x 1Gbps Neutrik EtherCON receptacles, with PoE, maximum 12W 2 x SFP slots for alternative interfaces

PoE Power

Standard:	802.3af
Class:	0
PD Power Range:	0.44 W to 12.94 W
Typical PSE Power Usage:	13W
Max PSE Power Usage:	15.4W

Equipment Type

AVN-CU2-DANTE	Dante® Commentator Unit, 2 Commentators

Physical Specification

Dimensions (Raw):	13.5cm (W) x 23cm (D) x 9.0cm (H - rear) 6.6cm (H - Front) 9.3cm (H - Maximum)
	5.3" (W) x 9" (D) x 3.5" (H - front) x 2.6" (H - rear) 3.7" (H - Maximum)
Dimensions (Boxed):	30.5cm (W) x 21.5cm (D) x 16.3cm (H)
	12" (W) x 8.5" (D) x 6.4" (H)
Weight:	Nett: 1.8kg Gross: 2.1kg Nett: 3.96lbs Gross: 4.62lbs

Accessories

AVN-DCX60	DC Power Supply for AVN Range, 4 pin XLR socket, 60W
AVN-TC3 (Case Only)	Transport Case, AVN-CU2-DANTE & Power Supply Cut-out
AVN-TC3 (Case Only) Dimensions (Raw):	46.5cm (W) x 36cm (D) x 17.6cm (H)
AVN-TC3 (Case Only) Dimensions (Raw):	18.3" (W) x 14.2" (D) x 6.9" (H)
AVN-TC3 (Case Only) Dimensions (Boxed):	52cm (W) x 21.5cm (D) x 43cm (H)
AVN-TC3 (Case Only) Dimensions (Boxed):	20.5" (W) x 8.5" (D) x 16.9" (H)
AVN-TC3 (Case Only) Weight:	Net: 3.0kg Gross: 3.5kg
AVN-TC3 (Case Only) Weight:	Net:6.6lbs Gross: 7.7lbs
CM-GM2:	Professional Gooseneck Condenser Microphone

AVN-CU2-DANTE Front View.

AVN-CU2-DANTE Rear View.

AVN-CU4-DANTE Configurable Dante Commentary Unit for 4 Commentators

The AVN-CU4-DANTE is a configurable portable commentator unit using Dante® AoIP. It is a dual version of the AVN-CU2-DANTE providing four mic/line inputs with a wide, adjustable gain range and four stereo headphone outputs with lockable jack sockets, suitable for operation by three or four commentators.

Fully featured, this unit allows you to handle virtually any commentary situation with both Dante AoIP and 4 wire connections, dual redundant Dante networking and multiple AC/DC/PoE power options. Up to 48 rotary encoders can be used on inputs, outputs or cross-points, allowing talkback feeds, commentary and audio mixing to be handled in one unit.

It's been designed from the ground up to be configurable for different situations with config files that can be saved and recalled for quick setup. Additionally, buttons and rotary encoders can be locked out from use, if the unit is being operated by less-technical commentators.

It has an abundance of 4 wire connections on the rear panel: 4 x analogue line inputs on XLR sockets with latching locks, 6 x analogue line outputs on XLR plugs and an RJ45 AES3 stereo input & output. These connections can act as a simultaneous analogue backup to the Dante AoIP connections.

Comparison with AVN-CU2-DANTE

The featureset is as per the AVN-CU2-DANTE, with the following differences. There are two displays with a doubling of operational controls:

- 4 x On-air buttons.
- 4 x Page buttons, 2 for each half of the display.
- 4 x Cough buttons.
- 8 x Talkback buttons, up to 4 for each user.
- 3 x User buttons.
- 12 x rotary encoders.
- 8 x pages of 6 rotary encoder positions (48 in total).

Similar to the AVN-CU2-DANTE, the illuminated 'Sonifex' logo acts as a power indication and illuminated LEDs indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary, PoE Secondary and AC power active.

The front panel houses 4 x locking mic/line inputs with +48V phantom power indication and 4 x headphone outputs on locking 6.35mm jack sockets.

The unit has dual redundant network ports on both RJ45 (PoE+ using 2 x Neutrik EtherCON® connectors) and SFP cages.

Category: Dante Commentary.

Product Function: Commentator unit for four commentary positions with dual 16 x 16 Dante matrix mixers.

Typical Applications: Used in sports grounds and arenas by commentators to produce commentary & hear talkback, 16 x 16 matrix mixer with configurable controls for live events.

Features:

- Web GUI for configuration.
- Full web GUI remote control.
- Lock-out of all individual controls and/or button row/encoder pages.
- PoE using 2 x Neutrik EtherCON® RJ45s.
- Dual redundant network ports on RJ45 and SFP.
- 4 x locking mic/line inputs with +48V phantom power indication.
- 4 x headphone outputs on locking 6.35mm jack sockets.
- 4 x analogue inputs and 6 analogue outputs on XLR.
- 1 x stereo AES/EBU input and output on RJ45s.
- 4 x talkback buses.
- On-air buttons with lazy talkback ability.
- Supports up to 16 x input and output streams and 16 input and output multicast flows/streams.
- Configurable headphone source selection.
- 12 x rotary encoders to control volume, or signal panning, and colour coded with the screen.
- 12 x detailed colour displays showing channel name, level metering & limit indication.
- Fully configurable buttons and rotary encoders, for inputs and outputs.
- 4 x pages of encoder settings, e.g. one for talkback inputs, one for outputs, one to monitor other sources.
- Metering per input/output, with output configured as pre or post level adjustment.
- Built-in line-up tone generator.
- Limiter available on every output.
- Output metering section shows limiter indication, the name of the output & level metering.
- 10 x configurable GPIO with 1 x switched changeover output.
- GPIO & VGPIO matrix programmable with events.

To power the unit, as well as the dual PoE ports and 12V 4 pin XLR DC input, there is an AC mains input on an IEC inlet, with a universal supply.

Full Description

The unit supports up to 16 input and output AoIP channels and up to 16 simultaneous input and output AoIP streams. Stream setup to and from the unit is initially via Dante® Controller with more detailed configuration performed using the built-in web GUI. The power of this unit lies in the impressive mix engine which overlays the usual Dante® Controller settings. Once Dante® flows have been made, inputs and outputs can be mixed freely to AoIP or physical inputs and outputs, controlled using the programmable buttons and rotary encoders, which control the gain and pan of inputs, outputs or cross-points, allowing total flexibility for different situations.

It can be used in any number of different commentary situations, controlled manually or remotely and controls can be 'locked down' so that they can't be tampered with or altered, to guarantee reliable operation. Housed in a rugged and intuitive, user-friendly package, it's a truly different way of looking at how commentary units should operate.

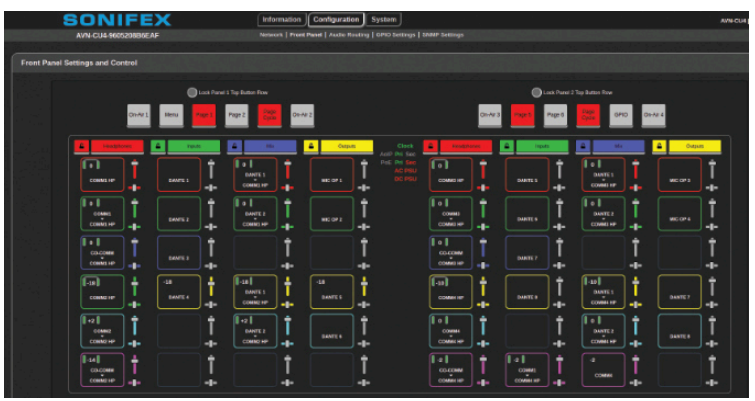
Traditional commentary units have fixed analogue and digital I/O and fixed controls in fixed positions on the unit. Their inputs and outputs are defined at hardware design and are thus limited by that initial design, including limited routing, mixing and DSP of the audio pathways.

We've taken a different approach with the AVN-CU4-DANTE. It was designed from the ground up to be totally flexible in operation and the use of Dante AoIP means that inputs and outputs can be added as required (up to a max of 16 per unit). Because any physical analogue or digital input can be mixed and routed with any Dante AoIP input to any physical output or Dante output, you can define your own audio pathways. Additionally you can choose which of those pathways need to be controlled (volume and pan) by the use of rotary encoders.

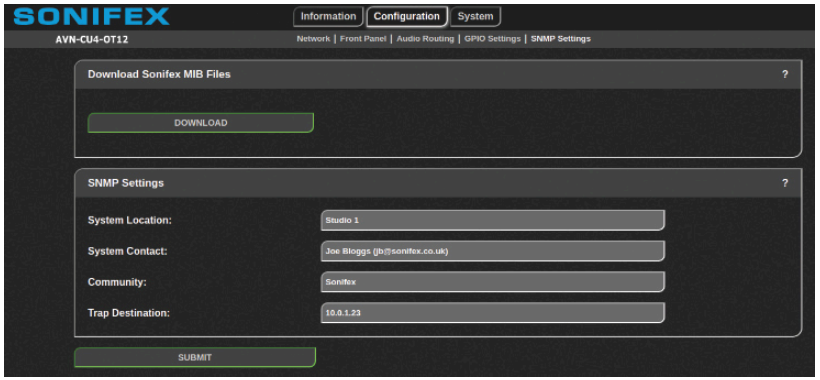
Using a built-in web GUI, up to 8 nameable pages of 6 rotary encoders (48 in total) can be placed on the mix matrix at inputs, outputs or cross-points. Encoders can control the volume and pan of headphones and volume of inputs, outputs or cross-points. Each rotary encoder has a separate colour-coded meter section showing the channel name, detailed level metering and left/right panning on a bright daylight reading display. Colours can be programmed per encoder to quickly identify particular source groups, so headphone source selection becomes intuitive.

The unit can be fully remotely controlled from the web interface, which mimics the front panel of the unit and includes live metering. All the controls can be operated as if at the unit and there are front-panel ▶

AVN-CU4-DANTE Top View.



AVN-CU4-DANTE GUI Front Page.



AVN-CU4-DANTE SNMP Page.

lock-out options for every button and encoder, with 'quick lock-out' for the rows of buttons and pages of encoders. This allows you to configure the unit for simple operation by the talent but with more complex setups, say, for mixing the talkback or feeds from an OB truck.

The AVN-CU4-DANTE provides four locking mic/line inputs, each with +48V phantom power indication and a wide, adjustable gain range. It has four stereo headphone outputs with locking 6.35mm jack sockets, suitable for operation by four commentators.

It's powered using Power over Ethernet (PoE), using Neutrik EtherCON® connectors, with primary and secondary ports for power and data redundancy. There's an additional 4 pin XLR 12V DC input and an AC mains input on an IEC inlet, with a universal supply. The unit supports up to 16 input and output AoIP channels and up to 16 simultaneous input and output AoIP streams.

The 12 x push-button rotary encoders and 24 x key-cap buttons are fully configurable, to control input & output levels and panning. The 12 key-cap buttons can have different colours for the on and off functions.

Metering is available per input/output, with output metering configurable as pre or post level adjustment. The top of the display shows output metering, a limiter indication and the name of the output. An adjustable limiter is available on every output and is applied automatically to prevent signal clipping.

The unit has 4 x locking mic/line inputs with +48V phantom power indication and 4 x headphone outputs on locking 6.35mm jack sockets.

Four wire connections on the rear panel provide a number of ways of linking to the outside world. There are:

- 4 x analogue line inputs on XLR sockets with latching locks
- 6 x analogue line outputs on XLR plugs
- 1 x RJ45 AES3 stereo input
- 1 x RJ45 AES3 stereo output

The I/O can be assigned as programme inputs, talkback inputs, mic outputs (line level), PA outputs, zone outputs or talkback outputs, as desired. In addition, the AES/analogue connections can be used as an insert or exit point into/out from the AoIP network.

The unit has dual redundant network ports on both RJ45 (PoE using 2 x Neutrik EtherCON® connectors) and SFP cages for long fibre runs.

There are 10 x configurable GPIO on a 15 way D-type connector with 1 x switched changeover output. GPIO & VGPIO (virtual GPIO, i.e. network commands) can be

AVN-CU4-DANTE Front View.

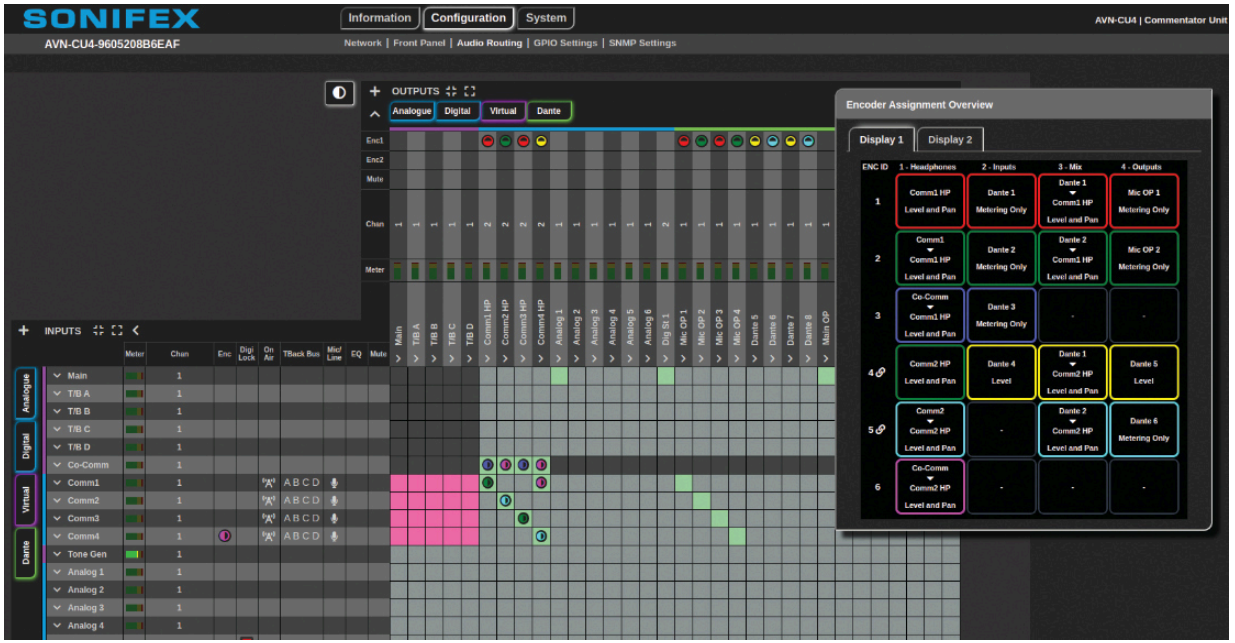
AVN-CU4-DANTE Rear View.

configured on a matrix to visually show actions, combined with button presses and event triggers, such as input/output muting, which enables some automation. Button presses can control buttons and functions on other network connected AVN-CU units using VGPIO.

The complete configuration of the unit, including all encoder settings, button setups and Dante mix matrix audio routes, can be saved off to a text editable .json file. Up to 8 of these files can be stored on the unit and loaded from the main Menu, meaning that the unit can be pre-configured for any venue, or event and with these setups instantly recalled onsite.

All of the buttons have key-cap text and can be configured for any button in any position. There are some standard operations available:

- 4 x On-Air buttons can be used to connect mic audio to the main output, either over AoIP or via the AES digital audio connection. The On-Air buttons can be locked if required.
- A Menu button can be used to access limited setup options on the TFT display.
- Page buttons change the display and encoders to monitor an additional set of sources, mix points or outputs. Up to 8 pages can be pre-programmed, e.g. one for talkback inputs, one for outputs, one to monitor other sources.
- 4 x Cough buttons take the commentator off-air while pressed.
- 3 x User buttons can be programmed to perform any function, using the web server.
- 8 x T/B (talkback) buttons can be configured to initiate talkback over AoIP, analogue or AES digital audio connections, using 4 x talkback buses. The talkback buttons operate with lazy talkback, taking the commentator off-air when invoked.
- 12 x rotary encoders.
- 8 x pages of 6 rotary encoder positions (48 in total).



AVN-CU4-DANTE GUI Audio Routing Page.

The illuminated 'Sonifex' logo acts as a power indication and there are illuminated LEDs to indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary, PoE Secondary and AC/DC power active.

Technical Specification For AVN-CU4-DANTE

Network and AoIP	
AoIP Standard:	Dante
Number of Channels:	16 receive, 16 transmit
Number of Streams:	16 receive, 16 transmit
Sample Rate:	48 kHz
Format:	Linear PCM, 16 or 24 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45 and 2 x SFP
Speed:	1Gbps and 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Mic/Line Inputs	
Input Impedance (Mic Mode):	2.5kΩ electronically balanced

Input Impedance (Line Mode):	>10kΩ electronically balanced
Preamp Gain (Mic Mode):	User selectable 0dB to 60dB in 3dB steps
0dBFS Line-up (Mic Mode):	-58dBu @ Max preamp gain, +2dBu @ Min preamp gain
0dBFS Line-up (Line Mode):	User selectable +15dBu / +18dBu / +20dBu / +22dBu / +24dBu = 0dBFS
Frequency Response (Mic Mode):	+0/-0.2dB 20Hz to 22kHz
Frequency Response (Mic Mode):	Ref 40dB gain @ 1kHz
Frequency Response (Line Mode):	+0/-0.2dB 20Hz to 22kHz
Frequency Response (Line Mode):	Ref 0dBu @ 1kHz
THD+N (Mic Mode):	<-90dBFS, -64dBFS, 20Hz – 20kHz, 40dB gain, 20kHz BW
THD+N (Line Mode):	<-98dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Equivalent Input Noise (Mic Mode):	127dB Ref. 60dB preamp gain, Rs = 200Ω
Noise (Line Mode):	<-110dBFS, 20kHz BW, Rs = 200Ω
C.M.R.R.	>60dB

Stereo Digital Input

Input Impedance:	110Ω
Supported Input rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz
Output Impedance:	110Ω
Supported output rates:	48kHz

Analogue Line Inputs

Input Impedance:	>20kΩ electronically balanced
0dBFS Line-up:	User selectable +15dBu / +18dBu / +20dBu / +22dBu / +24dBu = 0dBFS
Frequency Response:	+0/-0.2dB 20Hz to 22kHz
Frequency Response:	Ref 0dBu @ 1kHz
THD+N:	<-110dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise:	<-110dBFS, 20kHz BW, Rs = 200Ω
C.M.R.R.	>60dB

Analogue Line Outputs

Output Impedance:	<50Ω
0dBFS Line-up:	User selectable 0dBFS = +15dBu / +18dBu / +20dBu / +22dBu / +24dBu
Frequency Response:	+0/-0.5dB 20Hz to 22kHz
Frequency Response:	Ref 0dBu @ 1kHz
THD+N:	<-110dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise:	<-110dBFS, 20kHz BW

Headphone Outputs

Drive Capability	Drives 150mW into 32Ω to 600Ω Headphones
0dBFS Line-up:	Fixed 0dBFS = +18dBu
Frequency Response:	+0/-0.2dB 20Hz to 22kHz
Frequency Response:	Ref 0dBu @ 1kHz
THD+N:	<-108dBFS, -30dBFS, 20Hz – 20kHz, 20kHz BW
Noise:	<-110dBFS, 20kHz BW

Main Panel Operational Controls & Indicators

Power LED	Sonifex logo illuminates when power is present
Pushbuttons:	24 x Illuminated buttons that can be assigned to any of the following functions, and configured:
Pushbuttons:	On-Air / Talkback / Page # / Page Cycle / GPO / Cough / Menu / Brightness
Display:	2 x 480 x 107 pixel colour TFT display showing pan status, level, metering, control type, source/mix/destination names and main output metering
Status LEDs:	Clock / AoIP status / PoE status and PSU status LEDs to monitor power and connectivity. Can be disabled if required
Rotary Encoders: configurable	12 x robust soft touch rotary encoders with adjacent light bars for group/source/commentator grouping and identification

Front Panel Connections

Headphone Outputs: jack	4 x Neutrik ¼" (6.35mm) locking sockets
Microphone/Line Inputs:	4 x Neutrik 3 pin latching XLR sockets

Rear Panel Connections

Stereo Digital Input: balanced)	RJ45 socket (electronically
Stereo Digital Output: balanced)	RJ45 socket (electronically
Analogue Line Inputs:	4 x Neutrik locking XLR sockets
Analogue Line Outputs:	6 x Neutrik locking XLR plugs
GPIO Port:	15-way 'D'-type socket
Network:	2 x 1Gbps Neutrik EtherCON receptacles, with PoE, Maximum 12W
Network:	2 x SFP slots for alternative interfaces
DC Input:	Neutrik 4 pin XLR plug, 12VDC, Maximum 12W
Mains Input:	Filtered IEC, 85 – 264VAC, 47-63 Hz, 60W max
Fuse Rating:	Anti-surge fuse 2A 20 x 5mm

PoE Power

Standard:	802.3at Type 2
Class:	4
PD Power Range:	12.95 W to 25.50 W
Typical PSE Power Usage:	20W
Max PSE Power Usage:	30W

Equipment Type

AVN-CU4-DANTE	AVN Dante commentator unit, 4 commentators

Physical Specification

Dimensions (Raw):	25.7cm (W) x 22.9cm (D) x 9.2cm (H - max), 6.5cm (H - front), 8.9cm (H - rear)
Dimensions (Raw):	10.1" (W) x 9" (D) x 3.6" (H - max), 2.6" (H - front), 3.5" (H - rear)
Dimensions (Boxed):	37cm (W) x 35.5cm (D) x 19cm (H)
Dimensions (Boxed):	14.6" (W) x 14" (D) x 7.5" (H)

Accessories

AVN-DCX60	DC Power Supply for AVN Range, 4 pin XLR socket, 60W
AVN-TC1 (Case Only)	Transport Case, AVN-CU2-DANTE & Power Supply Cut-out
AVN-TC1 (Case Only) Dimensions (Raw):	46.5cm (W) x 36cm (D) x 17.6cm (H)
AVN-TC1 (Case Only) Dimensions (Raw):	18.3" (W) x 14.2" (D) x 6.9" (H)
AVN-TC1 (Case Only) Dimensions (Boxed):	52cm (W) x 21.5cm (D) x 43cm (H)
AVN-TC1 (Case Only) Dimensions (Boxed):	20.5" (W) x 8.5" (D) x 16.9" (H)
Weight:	Net: 3.0kg Gross: 3.5kg
Weight:	Net:6.6lbs Gross: 7.7lbs
CM-GM2:	Professional Gooseneck Condenser Microphone

AVN-CU4-DANTE Iso View.

AVN-AIO4 4 Input, 4 Output Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts up to four analogue inputs and four analogue outputs to and from the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 4 x balanced analogue inputs on XLR.
- 4 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (100Mb Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor.

The AVN-AIO4 audio converter and interface converts up to four analogue inputs and four analogue outputs to and from the Dante Audio-over-IP networking standard. This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue inputs and outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 4 x balanced analogue inputs on XLR.
- 4 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (100Mb Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor. ▶

Technical Specification For AVN-AIO4

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	4 receive, 4 transmit
Number of Streams:	4 receive, 4 transmit
Sample Rates:	44.1kHz, 48 kHz, 88.2kHz or 96kHz

(Note: 2 channels I/O only when >48kHz. Either 4 inputs or 4 outputs at 96kHz, can be configured on request with a separate firmware download).

Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	1 x RJ45
Speed:	100Mbps
Dante Domain Manager Ready:	Yes

Audio Input Performance Specifications

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100 dB
Common Mode Rejection:	>60dB @ 1kHz

Audio Output Performance Specifications

Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N:	<-107dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-107dBFS, 20kHz BW
Crosstalk:	<100dB

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Input Phase/Positive
3	Input Non-Phase/Negative

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Output Phase/Positive
3	Output Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	4.5 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AIO4:	4 Input, 4 Output Dante® Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19" (W) x 4.3" (D) x 1.7" (H)
Weight:	Nett: 0.9kg Gross: 1.4kg Nett: 1.9lbs Gross: 3.1lbs

AVN-AIO4 Front View.

AVN-AIO4 Rear View.

AVN-AIO8 8 Input, 8 Output Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts up to eight analogue inputs and eight analogue outputs to and from the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 8 x balanced analogue inputs on XLR.
- 8 x balanced analogue outputs on XLR.
- 1 x RJ45 Dante connector (1Gb/s Ethernet Port).
- POE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor.

The AVN-AIO8 audio converter and interface converts up to eight analogue inputs and eight analogue outputs to and from the Dante Audio-over-IP networking standard. This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue inputs and outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 8 x balanced analogue inputs on XLR.
- 8 x balanced analogue outputs on XLR.
- 1 x RJ45 Dante connector (1Gb/s Ethernet Port).
- POE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor.
- (The AVN-AIO8R is available with dual redundant Ethernet ports). ▶

Technical Specification For AVN-AIO8

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	8 receive, 8 transmit
Number of Streams:	8 receive, 8 transmit
Sample Rates:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	1 x RJ45
Speed:	1Gbps or 100Mbps
Dante Domain Manager Ready:	Yes

Audio Input Performance Specifications

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N: 20kHz BW	<-110dBFS, -30dBFS, 20Hz to 20kHz,
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100 dB
Common Mode Rejection:	>60dB @ 1kHz

Audio Output Performance Specifications

Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N:	<-107dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise (Mic=EIN):	-107dBFS, 20kHz BW
Crosstalk:	<100dB

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Input Phase/Positive
3	Input Non-Phase/Negative

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Output Phase/Positive
3	Output Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6.5 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AIO8:	8 Input, 8 Input, 8 Output Dante® Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19" (W) x 4.3" (D) x 1.7" (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AIO8 Front View.

AVN-AIO8 Rear View.

AVN-AIO8R 8 Input, 8 Output, Dual Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts up to eight analogue inputs and eight analogue outputs to and from the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 8 x balanced analogue inputs on XLR.
- 8 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19” rack-mount form factor.

The AVN-AIO8R audio converter and interface converts up to eight analogue inputs and eight analogue outputs to and from the Dante Audio-over-IP networking standard. Dual Ethernet ports allow the unit to operate in redundant mode, ensuring audio routing is maintained in the event of loss of link on either of the network connections.

This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue inputs and outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 8 x balanced analogue inputs on XLR.
- 8 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19” rack-mount form factor.

(The AVN-AIO8 is available with a single Ethernet port).

Technical Specification For AVN-AIO8R

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	8 receive, 8 transmit
Number of Streams:	8 receive, 8 transmit
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Audio Input Performance Specifications

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100 dB
Common Mode Rejection:	>60dB @ 1kHz

Audio Output Performance Specifications

Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N:	<-107dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-107dBFS, 20kHz BW
Specifications	Crosstalk: <100dB

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Input Phase/Positive
3	Input Non-Phase/Negative

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Output Phase/Positive
3	Output Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	7.5 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AIO8R:	8 Input, 8 Output, Dual Dante® Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19" (W) x 4.3" (D) x 1.7" (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AIO8R Front View.

AVN-AIO8R Rear View.

AVN-AI16 16 Input Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts sixteen analogue inputs to the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 16 x balanced analogue inputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (1 Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19" rack-mount form factor.

All analogue inputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 16 x balanced analogue inputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (1 Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19" rack-mount form factor.
- (The AVN-AI16R is available with dual redundant Ethernet ports).

The AVN-AI16 audio converter and interface converts up to sixteen analogue inputs to the Dante Audio-over-IP networking standard. This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

Technical Specification For AVN-AI16

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 transmit
Number of Streams:	16 transmit
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	1 x RJ45
Speed:	1Gbps or 100Mbps
Dante Domain Manager Ready:	Yes

Audio Input Performance Specifications

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N: 20kHz BW	<-110dBFS, -30dBFS, 20Hz to 20kHz,
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100 dB
Common Mode Rejection:	>60dB @ 1kHz

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Input Phase/Positive
3	Input Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AI16:	16 Input Dante® Interface, PoE

Physical Specification

Dimensions (Raw): (H)(1U)	48cm (W) x 11cm (D) x 4.3cm 19” (W) x 4.3” (D) x 1.7” (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AI16 Front View.

AVN-AI16 Rear View.

AVN-AI16R 16 Input Dual Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts sixteen analogue inputs to the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 16 x balanced analogue inputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/ +24dBu).
- 2 x RJ45 Dante connector (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19" rack-mount form factor.

The AVN-AI16R audio converter and interface converts up to sixteen analogue inputs to the Dante Audio-over-IP networking standard. Dual Ethernet ports allow the unit to operate in redundant mode, ensuring audio routing is maintained in the event of loss of link on either of the network connections.

This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue inputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 16 x balanced analogue inputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 2 x RJ45 Dante connector (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19" rack-mount form factor.

(The AVN-AI16 is available with a single Ethernet port).

Technical Specification For AVN-AI16R

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 transmit
Number of Streams:	16 transmit
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45
Speed:	1Gbps and 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Audio Input Performance Specifications

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100 dB
Common Mode Rejection:	>60dB @ 1kHz

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Input Phase/Positive
3	Input Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	7 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AI16R:	16 Input Dual Dante® Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19" (W) x 4.3" (D) x 1.7" (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AI16R Front View.

AVN-AI16R Rear View.

AVN-AO16 16 Output Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts sixteen analogue outputs to the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 16 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (1 Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19" rack-mount form factor.

The AVN-AO16 audio converter and interface converts up to sixteen analogue outputs from the Dante Audio-over-IP networking standard. This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 16 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 1 x RJ45 Dante connector (1 Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19" rack-mount form factor.

(The AVN-AO16R is available with dual redundant Ethernet ports).

Technical Specification For AVN-AO16

Network and AoIP	
AoIP Standard:	Dante
Number of Channels:	16 receive
Number of Streams:	16 receive
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	1 x RJ45
Speed:	1Gbps or 100Mbps
Dante Domain Manager Ready:	Yes

Audio Output Performance Specifications	
Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N: 20kHz BW	<-107dBFS, -30dBFS, 20Hz to 20kHz,
Noise:	-107dBFS, 20kHz BW
Crosstalk:	<100dB

Balanced Line Output XLR Pinout	
Pin	Function
1	Chassis Ground/Screen
2	Output Phase/Positive
3	Output Non-Phase/Negative

PoE Power	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	7 W
Max PSE Power Usage	15.4 W

Equipment Type	
AVN-AO16:	16 Output Dante® Interface, PoE

Physical Specification	
Dimensions (Raw): (H)(1U)	48cm (W) x 11cm (D) x 4.3cm 19” (W) x 4.3” (D) x 1.7” (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AO16 Front View.

AVN-AO16 Rear View.

AVN-AO16R 16 Output Dual Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts sixteen analogue outputs to the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 16 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19" rack-mount form factor.

The AVN-AO16R audio converter and interface converts up to sixteen analogue outputs from the Dante Audio-over-IP networking standard. Dual Ethernet ports allow the unit to operate in redundant mode, ensuring audio routing is maintained in the event of loss of link on either of the network connections.

This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up, except line-up levels, done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All analogue outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link, and Clock. Global 0dBFS line-up can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch.

- 16 x balanced analogue outputs on XLR.
- Adjustable global 0dBFS line up selection (+12dBu/+18dBu/+24dBu).
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19" rack-mount form factor.

(The AVN-AO16 is available with a single Ethernet port).

Technical Specification For AVN-AO16R

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 receive
Number of Streams:	16 receive
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Audio Output Performance Specifications

Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	User Selectable +12dBu/+18dBu/ +24dBu balanced
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N: 20kHz,	<-107dBFS, -30dBFS, 20Hz to 20kHz BW
Noise:	-107dBFS, 20kHz BW
Crosstalk:	<100dB

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground/Screen
2	Output Phase/Positive
3	Output Non-Phase/Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	8 W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AO16R:	16 Output Dual Dante®Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19” (W) x 4.3” (D) x 1.7” (H)
Weight:	Nett: 1.1kg Gross: 1.6kg Nett: 2.4lbs Gross: 3.5lbs

AVN-AO16R Front View.

AVN-AO16R Rear View.

AVN-AESIO8 8 AES3 Input, 8 AES3 Output Dante® Interface, PoE

Category: Dante Audio Interfaces.

Product Function: Converts up to eight digital stereo AES3 inputs and eight digital stereo AES3 outputs to and from the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 8 x balanced digital stereo AES3 inputs on XLR, supporting input rates of 32kHz – 192kHz.
- Sample rate conversion on physical inputs to Dante system sample rate.
- 8 x balanced digital stereo AES3 outputs on XLR, output rate matches Dante system sample rate.
- 1 x RJ45 Dante connector (1Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- AES3 Lock LED status indicators for each input.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor.

The AVN-AESIO8 audio converter and interface converts up to eight digital stereo AES3 inputs and eight digital stereo AES3 outputs to and from the Dante Audio-over-IP networking standard. Each input can accept sample rates from 32kHz to 192kHz, which will be sample rate converted to the Dante system sample rate. A valid input signal is confirmed by front panel AES3 lock LEDs for each input. All outputs follow the Dante system sample rate.

This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All digital AES3 inputs and outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link and Clock.

- 8 x balanced digital stereo AES3 inputs on XLR, supporting input rates of 32kHz – 192kHz.
- Sample rate conversion on physical inputs to Dante system sample rate.
- 8 x balanced digital stereo AES3 outputs on XLR, output rate matches Dante system sample rate.
- 1 x RJ45 Dante connector (1Gb/s Ethernet Port).
- PoE, Link, and Clock LED status indicators.
- AES3 Lock LED status indicators for each input.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE.
- 1U 19” rack-mount form factor.

(The AVN-AESIO8R is available with dual redundant Ethernet ports).

Technical Specification For AVN-AESIO8

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 receive, 16 transmit
Number of Streams:	16 receive, 16 transmit
Sample Rates:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	1 x RJ45
Speed:	1Gbps or 100Mbps
Dante Domain Manager Ready:	Yes

AES3 Input Specifications

Input Impedance:	110Ω balanced
Support Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to Dante system sample rate).
Signal Level:	AES3-2009 compliant
Bit Depth:	Up to 24bits
Output Impedance:	110Ω balanced
Support Output Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (set as per Dante system sample rate).
Signal Level:	AES3-2009 compliant
Bit Depth:	24 bits

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground
2	Input Positive
3	Input Negative

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground
2	Output Positive
3	Output Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AESIO8: Interface, PoE	8 AES3 Input, 8 AES3 Output Dante®

Physical Specification

Dimensions (Raw): (H)(1U)	48cm (W) x 11cm (D) x 4.3cm 19" (W) x 4.3" (D) x 1.7" (H)
Weight:	Nett: 1.0kg Gross: 1.5kg Nett: 2.2lbs Gross: 3.3lbs

AVN-AESIO8 Front View.

AVN-AESIO8 Rear View.

AVN-AESIO8R 8 AES3 Input, 8 AES3 Output Dual Dante® Interface, PoE

The AVN-AESIO8R audio converter and interface converts up to eight digital stereo AES3 inputs and eight digital stereo AES3 outputs to and from the Dante Audio-over-IP networking standard. Each input can accept sample rates from 32kHz to 192kHz, which will be sample rate converted to the Dante system sample rate. A valid input signal is confirmed by front panel AES3 lock LEDs for each input. All outputs follow the Dante system sample rate.

Dual Ethernet ports allow the unit to operate in redundant mode, ensuring audio routing is maintained in the event of loss of link on either of the network connections. This cost effective 1U rack-mount unit offers an easy solution for AV professionals and system integrators. It is simple to configure and operate, with all set-up done via the standard Dante Controller software and power via PoE (Power Over Ethernet).

All digital AES3 inputs and outputs are on high-quality Neutrik XLR connectors and there are front panel status/confidence LEDs for PoE, Link and Clock.

- 8 x balanced digital stereo AES3 inputs on XLR, supporting input rates of 32kHz – 192kHz.
- Sample rate conversion on physical inputs to Dante system sample rate.

Category: Dante Audio Interfaces.

Product Function: Converts up to eight digital stereo AES3 inputs and eight digital stereo AES3 outputs to and from the Dante Audio-over-IP networking standard.

Typical Applications: System integration to Dante.

Features:

- 8 x balanced digital stereo AES3 inputs on XLR, supporting input rates of 32kHz – 192kHz.
- Sample rate conversion on physical inputs to Dante system sample rate.
- 8 x balanced digital stereo AES3 outputs on XLR, output rate matches Dante system sample rate.
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- AES3 Lock LED status indicators for each input.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19” rack-mount form factor.

- 8 x balanced digital stereo AES3 outputs on XLR, output rate matches Dante system sample rate.
- 2 x RJ45 Dante connectors (1Gb/s Ethernet Port) allowing the unit to operate in redundant or switched modes.
- PoE and Link LED status indicators for each Ethernet port.
- Clock LED status indicator.
- AES3 Lock LED status indicators for each input.
- Configuration using Dante Controller.
- AES67 operation & Dante Domain Manager compliant.
- Powered by PoE on either (or both) Ethernet ports, offering power supply redundancy.
- 1U 19” rack-mount form factor.

(The AVN-AESIO8 is available with a single Ethernet port).

Technical Specification For AVN-AESIO8R

Network and AoIP

AoIP Standard:	Dante
Number of Channels:	16 receive, 16 transmit
Number of Streams:	16 receive, 16 transmit
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

AES3 Input Specifications

Input Impedance:	110Ω balanced
Support Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to Dante system sample rate).
Signal Level:	AES3-2009 compliant
Bit Depth:	Up to 24bits
Output Impedance:	110Ω balanced
Support Output Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (set as per Dante system sample rate).
Signal Level:	AES3-2009 compliant
Bit Depth:	24 bits

Balanced Line Input XLR Pinout

Pin	Function
1	Chassis Ground
2	Input Positive
3	Input Negative

Balanced Line Output XLR Pinout

Pin	Function
1	Chassis Ground
2	Output Positive
3	Output Negative

PoE Power

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	7W
Max PSE Power Usage	15.4 W

Equipment Type

AVN-AESIO8R:	8 AES3 Input, 8 AES3 Output Dual Dante® Interface, PoE

Physical Specification

Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)(1U) 19” (W) x 4.3” (D) x 1.7” (H)
Weight:	Nett: 1.0kg Gross: 1.5kg Nett: 2.2lbs Gross: 3.3lbs

AVN-AESIO8R Front View.

AVN-AESIO8R Rear View.

AVN-DIO01 Dante to Analogue XLR Stereo Output

The AVN-DIO01 is a Dante AoIP network to analogue XLR stereo output converter in the Sonifex DIO range of Dante input/output devices. It features two balanced analogue XLR outputs and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante to Analogue XLR Stereo Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 2 x balanced XLR analogue outputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A conversion, >120dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

The superior A/D and D/A circuitry used in this product is designed for optimal audio performance and offers 120dB of dynamic range - ten times better than similar competing products. All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 2 x balanced XLR analogue outputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.

- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A conversion, >120dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO01

Analogue XLR Output Pin-out:

Pin	Function
1	Chassis Ground
2	Output Phase
3	Output Non Phase

Line Output - XLR/Terminal:

Parameter:	Description
Output Impedance:	<200Ω balanced
0dBFS Line-Up:	+18dBu
Frequency Response:	20Hz to 20kHz, +0/-0.5dB (600Ω load, ref 1kHz)
THD+N:	<-100dBu, -30dBFS, 20Hz to 20kHz, 20kHz BW
Dynamic Range:	120dB, 20kHz BW
Crosstalk:	<-110dB

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO01:	Dante® to Analogue XLR Stereo Output

Physical Specification:

Dimensions: (Raw)	10.6cm (W) x 7.3cm (D) x 4.3cm (H) 4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO01 Front View.

AVN-DIO01 Rear View.

AVN-DIO02 Analogue XLR Stereo Input to Dante®

The AVN-DIO02 is an analogue XLR stereo input to Dante AoIP network converter in the Sonifex DIO range of Dante input/output devices. It features two balanced analogue XLR inputs and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Analogue XLR Stereo Input to Dante®.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 2 x balanced XLR analogue inputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range A/D conversion, >120dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

The superior A/D and D/A circuitry used in this product is designed for optimal audio performance and offers 120dB of dynamic range - ten times better than similar competing products. All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 2 x balanced XLR analogue inputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range A/D conversion, >120dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack. ▶

Technical Specification For AVN-DIO02

Analogue XLR Input Pin-out:

Pin	Function
1	Chassis Ground
2	Input Phase
3	Input Non Phase

Line Input - XLR/Terminal:

Parameter	Description
Input Impedance	>10kΩ balanced
0dBFS Line-Up	+18dBu
Frequency Response	20Hz to 20kHz, +0/-0.2dB (600Ω load, ref 1kHz)
THD+N	<-118dBFS, -12dBu, 20Hz to 20kHz, 20kHz BW
Dynamic Range	120dB, 20kHz BW, Rs=200Ω
Crosstalk	<-110dB
Common Mode Rejection	>60dB @ 1kHz

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	3 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO02:	Analogue XLR Stereo Input to Dante®

Physical Specification:

Dimensions: (Raw)	10.9cm (W) x 7.3cm (D) x 4.3cm (H) 4.3” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO02 Front View.

AVN-DIO02 Rear View.

AVN-DIO03 Dante to Headphone Outputs (1/4” & 3.5mm Jacks) with Volume Control & Limiter

Category: Dante Audio Interfaces.

Product Function: Dante to Headphone Outputs (1/4” & 3.5mm Jacks) With Volume Control & Limiter.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1/4-inch and 3.5mm jack analogue headphone outputs.
- Headphones volume control.
- Limiter on/off, threshold control and LED indicator.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES 67 compatible.
- Ultra-high quality, wide dynamic range D/A conversion.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

The AVN-DIO03 is Dante AoIP network-to-headphones convertor in the Sonifex DIO range of Dante input/output devices. It features a single stereo headphone output available on two connections for ¼” and 3.5mm jacks and a volume control for headphones level. A limiter with threshold setting and LED indication can be switched in to prevent over-shooting and protect hearing.

The AVN-DIO03 is Dante AoIP network-to-headphones convertor in the Sonifex DIO range of Dante input/output devices. It features a single stereo headphone output available on two connections for ¼” and 3.5mm jacks and a volume control for headphones level. A limiter with threshold setting and LED indication can be switched in to prevent over-shooting and protect hearing.

The front panel potentiometer adjusts headphone volume from mute (fully anticlockwise) to +6dB of gain when fully clockwise. This is useful if the Dante stream level is low and requires boosting.

A simple headphone limiter is included to prevent hearing damage by limiting the audio level sent to the headphones. The limit level can be set using a trimmer adjustment tool, or small flat blade screwdriver, between approximately -12dBu (fully anticlockwise) and +6dBu (fully clockwise).

When limiting, the blue limit LED illuminates. This should prompt you to turn the headphone volume down until the blue LED extinguishes, as audio quality will be reduced whilst the limiter is active.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

The superior A/D and D/A circuitry used in this product is designed for optimal audio performance and offers 120dB of dynamic range - ten times better than similar competing products. All DIO products use ▶

Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

** **Note:** Because the headphone output of this device is unbalanced, performance is 6dB lower than for balanced AVN-DIO products.*

- 1/4-inch and 3.5mm jack analogue headphone outputs.
- Headphones volume control.
- Limiter on/off, threshold control and LED indicator.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES 67 compatible.
- Ultra-high quality, wide dynamic range D/A conversion.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO03

Headphone Jack Output Pin-out:

Pin	Function
Tip	Left
Ring	Right
Screen	0V Common

Headphone Output - Jack:

Parameter	Description
Output Impedance:	Capable of driving 150mW into 32Ω to 600Ω headphones
Maximum Output Level:	+16dBu
D to A Line-Up:	-6dBFS = +16dBu (max volume)
Gain:	Mute (min Volume) to +6dB (max volume)
Frequency Response:	20Hz to 20kHz, +0/-0.5dB (ref 1kHz)
THD+N:	<0.005%, -14dBFS input, max volume, 20Hz to 20kHz, 20kHz BW
Dynamic Range:	114dBu, 20kHz BW, 0dB gain
Crosstalk:	<100dB
Audio Limiter Range:	-12dBu to +6dBu

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	4 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO03:	Dante® to Headphone Output

Physical Specification:

Dimensions: (Raw)	10.6cm (W) x 7.3cm (D) x 4.3cm (H) 4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO03 Front View.

AVN-DIO03 Rear View.

AVN-DIO04 Dante® to Analogue Phono Stereo Input & Output

The AVN-DIO04 is a Dante to analogue phono stereo input & output convertor in the Sonifex DIO range of Dante input/output devices. It features stereo analogue input and output phono connections and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante® to Analogue Phono Stereo Input & Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 2 x analogue phono-type inputs.
- 2 x analogue phono-type outputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A and A/D conversion.
- Powered via PoE (Power over Ethernet).

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

The superior A/D and D/A circuitry used in this product is designed for optimal audio performance and offers 114dB * of dynamic range - ten times better than similar competing products. All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

* Note: Because the inputs & outputs of this device are unbalanced, performance is 6dB lower than for balanced AVN-DIO products.

- 2 x analogue phono-type inputs.
- 2 x analogue phono-type outputs.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A and A/D conversion.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO04

Analogue Phono Input & Output Pin-out:

Pin	Function
Inner	Signal (White - Left, Red - Right)
Outer	Chassis Ground

Line Input - RCA Phono:	
Parameter	Description
Input Impedance	>5kΩ unbalanced
0dBFS Line-Up	+12dBu
Frequency Response	20Hz to 20kHz, +0/-0.5dB (ref 1kHz)
THD+N	<-114dBFS, -18dBu, 20Hz to 20kHz, 20kHz BW
Dynamic Range	>114dB A-weighted, 20kHz BW, Rs=200Ω
Crosstalk	<-100dB
Line Output - RCA Phono:	
Output Impedance	<200Ω balanced
0dBFS Line-Up	+12dBu
Frequency Response	20Hz to 20kHz, +0/-0.5dB (ref 1kHz)
THD+N	<-114dBu, -30dBFS, 20Hz to 20kHz, 20kHz BW
Dynamic Range	>114dB A-weighted, 20kHz BW
Crosstalk	<-100dB
Dante:	
Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	5 W
Max PSE Power Usage	15.4 W
Equipment Type:	
AVN-DIO04:	Dante® to Analogue Phono Stereo Input & Output
Physical Specification:	
Dimensions: (Raw) (H)	11.1cm (W) x 7.3cm (D) x 4.3cm 4.4” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed): (H)	17.4cm (W) x 9.5cm (D) x 5.6cm 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO04 Front View.

AVN-DIO04 Rear View.

AVN-DIO05 Dante® to Analogue Terminal Block Stereo Input & Output

The AVN-DIO05 is a Dante to analogue terminal block input and output convertor in the Sonifex DIO range of Dante input/output devices. It features balanced stereo analogue inputs and outputs on a terminal block connector and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante® to Analogue Terminal Block Stereo Input & Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 12 x terminal block connections (balanced stereo inputs and outputs).
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES 67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A and A/D conversion.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

The superior A/D and D/A circuitry used in this product is designed for optimal audio performance and offers 120dB of dynamic range - ten times better than similar competing products. All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 12 x terminal block connections (balanced stereo inputs and outputs).
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.

- AES 67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality, wide dynamic range D/A and A/D conversion.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO05

Analogue Terminal Stereo Input & Output Pin-out:

Pin	Function
1	Chassis Ground
2	Left Input Phase
3	Left Input Non Phase
4	Chassis Ground
5	Right Input Phase
6	Right Input Non Phase
7	Chassis Ground
8	Left Output Phase
9	Left Output Non Phase
10	Chassis Ground
11	Right Output Phase
12	Right Output Non Phase

Line Input – Terminal Block:

Parameter	Description
Input Impedance	>10kΩ balanced
0dBFS Line-Up	+18dBu
Frequency Response	20Hz to 20kHz, +0/-0.2dB (600Ω load, ref 1kHz)
THD+N	<-118dBFS, -12dBu, 20Hz to 20kHz, 20kHz BW
Dynamic Range	120dB, 20kHz BW, Rs=200Ω
Crosstalk	<-110dB
Common Mode Rejection	>60dB @ 1kHz

Line Output – Terminal Block:

Output Impedance	<200Ω balanced
0dBFS Line-Up	+18dBu
Frequency Response ref 1kHz)	20Hz to 20kHz, +0/-0.5dB (600Ω load,
THD+N	<-100dBu, -30dBFS, 20Hz to 20kHz, 20kHz BW
Dynamic Range	120dB, 20kHz BW
Crosstalk	<-110dB

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	5 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO05:	Dante® to Analogue Terminal Block Stereo Input & Output

Physical Specification:

Dimensions: (Raw)	10.6cm (W) x 7.3cm (D) x 4.3cm (H) 4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO05 Front View.

AVN-DIO05 Rear View.

AVN-DIO06 Dante® to AES3 XLR Stereo Input & Output

The AVN-DIO06 is a Dante to AES3 digital input and output audio convertor in the Sonifex DIO range of Dante input/output devices. It features stereo AES3 digital audio inputs and outputs on Neutrik XLR connectors, and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante® to AES3 XLR Stereo Input & Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1 x stereo AES3 XLR input.
- 1 x stereo AES3 XLR output.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 1 x stereo AES3 XLR input.
- 1 x stereo AES3 XLR output.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO06

AES3 XLR Stereo Input Pin-out:

Pin	Function
1	Chassis Ground
2	Input Phase
3	Input Non Phase

AES3 XLR Stereo Output Pin-out:

Pin	Function
1	Chassis Ground
2	Output Phase
3	Output Non Phase

Power Supply - Class 0 802.3af PoE:

Device	Power Consumption (Watts)
DIO06:	< 3 W
Parameter	Description

AES3 Input - XLR:

Output Impedance	110Ω balanced
Output Format	AES3
Supported Sample Rates	44.1kHz, 48kHz, 88.2kHz, 96kHz

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	2 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO06:	Dante® to AES3 XLR Stereo Input & Output

Physical Specification:

Dimensions: (Raw)	10.6cm (W) x 7.3cm (D) x 4.3cm (H)
	4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO06 Front View.

AVN-DIO06 Rear View.

AVN-DIO07 Dante® to AES-3id BNC Stereo Input & Output

The AVN-DIO07 is a Dante to AES-3id digital input and output audio convertor in the Sonifex DIO range of Dante input/output devices. It features stereo AES-3id digital audio inputs and outputs on BNC connectors, and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante® to AES-3id BNC Stereo Input & Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1 x stereo AES-3id BNC input.
- 1 x stereo AES-3id BNC output.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 1 x stereo AES-3id BNC input.
- 1 x stereo AES-3id BNC output.
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO07

AES-3id BNC Input & Output Pin-out:

Pin	Function
Inner	Signal
Outer	Chassis Ground

AES-3id Input - BNC:

Parameter	Description
Output Impedance	75Ω unbalanced
Output Format	AES-3id
Supported Sample Rates	44.1kHz, 48kHz, 88.2kHz, 96kHz

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
PoE Power:	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	2 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO07:	Dante® to AES-3id BNC Stereo Input & Output

Physical Specification:

Dimensions: (Raw)	11.6cm (W) x 7.3cm (D) x 4.3cm (H) 4.6” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO07 Front View.

AVN-DIO07 Rear View.

AVN-DIO08 Dante® to AES3 Terminal Block Stereo Input & Output

The AVN-DIO08 is a Dante to AES3 digital input and output audio convertor in the Sonifex DIO range of Dante input/output devices. It features stereo AES3 digital audio inputs and outputs on terminal block connectors, and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

Category: Dante Audio Interfaces.

Product Function: Dante® to AES3 Terminal Block Stereo Input & Output.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 6 x terminal block connections (balanced stereo inputs and outputs).
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

- 6 x terminal block connections (balanced stereo inputs and outputs).
- Neutrik EtherCon® Ethernet connection.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality digital audio.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification ForAVN-DIO08

AES3 Terminal Block Stereo Input & Output Pin-out:

1	Chassis Ground
2	Input Phase
3	Input Non Phase
4	Chassis Ground
5	Output Phase
6	Output Non Phase

AES3 Input – Terminal Block:

Parameter	Description
Input Impedance	110Ω balanced
Input Format	AES3
Supported Sample Rates	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz

AES3 Output - Terminal Block:

Parameter	Description
Output Impedance	110Ω balanced
Output Format	AES3
Supported Sample Rates	44.1kHz, 48kHz, 88.2kHz, 96kHz

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32

PoE Power:

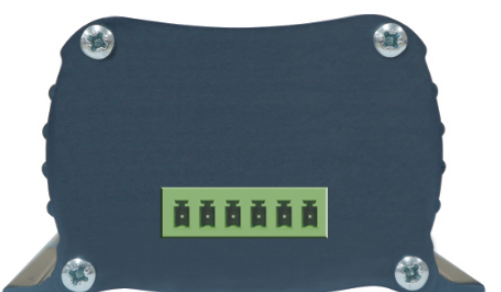
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	2 W
Max PSE Power Usage	15.4 W

Equipment Type:

AVN-DIO08:	Dante® to AES3 Terminal Block Stereo Input & Output

Physical Specification:

Dimensions: (Raw)	10.6cm (W) x 7.3cm (D) x 4.3cm (H) 4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed):	17.4cm (W) x 9.5cm (D) x 5.6cm (H) 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs



AVN-DIO08 Front View.

AVN-DIO08 Rear View.

AVN-DIO09 Microphone Input to Dante®

Category: Dante Audio Interfaces.

Product Function: Microphone Input to Dante®.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1 x balanced microphone input on XLR socket with latch lock.
- Neutrik EtherCon® Ethernet connection.
- Single turn pot setting fine mic gain (0dB – 36dB).
- Coarse mic gain switch (+20db/+50dB).
- High pass filter on/off switch.
- Phantom power on/off switch.
- Phantom power LED indicator.
- Level LED indicator.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality E.I.N. of >129dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

The AVN-DIO09 is a Microphone input to Dante converter in the Sonifex DIO range of Dante input/output devices with A/D circuitry offering a world-class E.I.N. of 129dB. It features a single high quality mic preamp with balanced XLR input, coarse and fine gain controls, high pass filter, phantom power, tri-colour level LED and one Neutrik EtherCon® connector for direct connection to a Dante AoIP network.

All Sonifex DIO interfaces provide a simple, convenient, and elegant plug and play method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

All DIO products use Dante Controller for configuration, are AES67 and Dante Domain Manager compliant and are powered by PoE (Power over Ethernet).

They use rugged aluminium boxes with side slots for screw-mounting and five units can be fitted into the optional AVN-DIORK 1U rack. All feature rugged Neutrik EtherCon® connectors and Neutrik lockable audio connectors for ultra-reliable connectivity.

The AVN-DIO09 has coarse and fine mic gain with the coarse gain set using a toggle switch, providing 20dB/50dB of gain, and the fine gain set using a trimmer adjustment tool, or small flat blade screwdriver, adding between 0dB and 36dB of additional gain. An on/off toggle switch turns the high pass filter on or off and when enabled, it acts on frequencies below 125Hz at a roll off of 6dB/octave.

Phantom power is enabled/disabled via a toggle switch on the front panel and when enabled, a 48V DC supply is provided to power an appropriate microphone. A red LED illuminates to show when phantom is enabled.

A front panel audio level LED helps to set the gain and shows the audio level being sent to the Dante network.

Note: *If using a phantom powered microphone, it may be necessary to earth the unit using the rear panel earth tag to eliminate mains hum, depending on the nature of your PoE supply.*

- 1 x balanced microphone input on XLR socket with latch lock.
- Neutrik EtherCon® Ethernet connection.
- Single turn pot setting fine mic gain (0dB – 36dB).
- Coarse mic gain switch (+20db/+50dB).

- High pass filter on/off switch.
- Phantom power on/off switch.
- Phantom power LED indicator.
- Level LED indicator.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Ultra-high quality E.I.N. of >129dB.
- Powered via PoE (Power over Ethernet).
- Five units fit into an AVN-DIORK 1U rack.

Technical Specification For AVN-DIO09

Microphone XLR Input Pin-out :

Pin	Function
1	Chassis Ground
2	Input Phase
3	Input Non Phase

Earthing Point:

Single earth tag.

Level LED Thresholds:

Off	Level is under -38dBFS (-20dBu)
Green	Level is between -38dBFS (-20dBu) and -18dBFS (0dBu)
Amber	Level is between -18dBFS (0dBu) and -10dBFS (+8dBu)
Red	Level is above -10dBFS (+8dBu).

Microphone Input - XLR:

Parameter	Description
Input Impedance	2.2kΩ balanced
Maximum Input Level	-68dBu (max gain) to -8dBu (min gain)
Gain	Adjustable 20dB to 86dB
Frequency Response	20Hz to 20kHz, +0/-0.5dB (ref 1kHz)
High Pass Filter Response	Fc = 125Hz @ 6dB per octave
THD+N	<0.01%, -32dBu input, 40dB gain, 20Hz to 20kHz, 20kHz BW
E.I.N.	129dBu, 20kHz BW, max gain, Rs=200Ω
Common Mode Rejection	>60dB @ 1kHz
Phantom Power	+48V ± 4V

Dante:

Parameter	Description
Sample Rates:	4.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
PoE Power:	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	4 W
Max PSE Power Usage	15.4 W
Equipment Type:	
AVN-DIO09:	Microphone Input to Dante®

Physical Specification:

Dimensions: (Raw) (H)	10.6cm (W) x 7.3cm (D) x 4.3cm 4.2” (W) x 2.9” (D) x 1.7” (H)
Dimensions (Boxed): (H)	17.4cm (W) x 9.5cm (D) x 5.6cm 6.9” (W) x 3.7” (D) x 2.2” (H)
Weight:	Nett: 0.2kg Gross: 0.3kg Nett: 0.44lbs Gross: 0.66lbs

AVN-DIO09 Front View.

AVN-DIO09 Rear View.

AVN-GPIO AVN-GPIO GPIO to LAN Transceiver (PTP, EMBER+ & UDP)

Category: Dante Audio Interfaces.

Product Function: GPIO to LAN Transceiver

Typical Applications: converts GPIO (General Purpose Inputs & Outputs) to network commands to control, and be controlled by, other equipment or software across a standard network.

Features:

- 10 assignable GPIOs, 8 of which are PTP enabled.
- PTPv2 keeps the hardware clock in sync with a PTP master to achieve a sub 10ns accuracy.
- Single voltage free relay contact to operate external equipment.
- Virtual GPIO for sending physical GPIO action commands across a network.
- A single cable can be used for both power and control.
- The built-in webserver makes the AVN-GPIO easy to configure and allows it to be controlled by any device with a web browser.
- It can provide power to other devices via DC outputs.
- Uses Ember+ and UDP for communication, allowing programs to be written to interface with it.

The AVN-GPIO is part of the AVN range of network interface boxes, which converts GPIO (General Purpose Inputs & Outputs) to network commands to control, and be controlled by, other equipment or software across a standard network.

It has 10 configurable GPIO's, 8 of which can be used for PTP based programming, together with a relay. It provides virtual GPIO that allow the device to trigger or be triggered by other Sonifex devices on the network using virtual GPIO without the need for extra wiring (virtual GPIO communication occurs via the ethernet connection). Units can either be interfaced together across a network, or you can use Ember+ and UDP commands to accept commands from, or drive, the AVN-GPIO.

The AVN-GPIO is a PTP (Precision Time Protocol) enabled GPIO device. IEEE1588-2008 PTPv2 is used to keep a hardware clock in sync with a PTP master, such as the AVN-GMCS Grandmaster Clock, to achieve a sub 10ns synchronisation to the master reference. This means that the AVN-GPIO can be used to accurately timestamp input events and to trigger outputs at configured times. The AVN-GPIO supports Default and AES67 Media profiles, and also provides a Custom profile which can be configured by the user.

It is housed in a rugged aluminium box with side slots for screw-mounting and is powered by PoE (Power over Ethernet).

The device is configured via a built-in webserver. This allows the configuration of PTP as well as live monitoring of its status. A GPIO routing webpage is provided which allows physical, timed, and virtual inputs to be routed to physical, timed, and virtual outputs.

The AVN-GPIO provides a simple UDP messaging system that allows other devices on the network to query the device status information, for example to retrieve the time at which a change in input occurred. Custom applications can also be written to query this information via UDP.

The device has 8 'timed' GPIOs - when used as inputs these can detect the rising and falling edges and will generate a timestamp synced to the hardware PTP clock. This means that the recorded timestamps will be synced within 10ns of the PTP master in a correctly setup system. When setup as outputs, a signal can be generated precisely at a time chosen by you and the time at which the output toggles can be configured down to the nanosecond. Alternatively, these timed GPIOs ▶

can be configured to act as 'normal' GPIOs depending on your application.

There are also two 'normal' GPIOs. When normal GPIOs are configured as inputs they can be set to either momentary or latching mode. When setup as outputs, they pull the signal on the GPO pin down to ground when active.

The AVN-GPIO has a voltage free relay contact that can be used to operate external equipment, and also provides 6 x DC output voltage pins that can be used to power external equipment such as signage and various sensors and actuators. When powered via PoE (Power over Ethernet) the AVN-GPIO outputs 12V at 500mA total on these pins. When powered via the DC input the AVN-GPIO provides 12V at 1.5A total on these pins. The DC outputs are fused to prevent drawing more current than the device can provide and these fuses are automatically reset when the device is power cycled.

Applications

- Motor racing timing: PTP enabled time outputs can be used to start a race at a specified time, then PTP enabled time inputs can be used to accurately record the time when cars pass the finish line with a nanosecond resolution. The results can be queried via UDP messages with a history of previous recordings also stored for querying.
- To connect legacy equipment, with only hardware I/O, to a network for remote control purposes. There are many Redbox units with hardware remotes that could now be controlled over a network, for example, the RB-DD4, RB-SD1 or RB-OA3.
- Button presses can be converted to network commands, for example to act as inputs to software for event handling, or to control remote equipment.
- Inputs from mechanical buttons & switches, or devices, can control lights and equipment connected to the AVN-GPIO across a network.
- Timed control switching, e.g. switching lights on/off at specific times of the day, reconfiguring which light switch turns on/off which light and/or controlling light on/off via a webpage.
- Time control of power on/off to test equipment and heating systems, together with providing times at which tests are passed.
- Controlling the power for other equipment at unattended/remote locations, whilst logging precise event data. (e.g. unattended weather stations).
- Configuring automated systems, controlling conveyor belts, turning on/off machinery.
- Notification system for a control room, to indicate an area that needs inspection which can be connected to alarm/bell.
- Signal detection on a Sonifex AVN-CU2/4 commentary unit can be linked to a virtual GPO which is then used as an input to the AVN-GPIO.
- A physical input to an AVN-GPIO can be routed to a virtual GP output, which could be a virtual input on the AVN-CU2/4. This could be used to change the commentator on/off air status, turn on/off ducking, or mute/unmute commentator as examples.
- A physical input to an AVN-GPIO can be routed to a virtual output, to act as a virtual input on a Sonifex AVN-Portal which can be used to mute/unmute a mix-point.

Technical Specification For AVN-GPIO

PTP Timing Specification	
Profile Support	IEEE 1588 Default Profile, AES67 Media & Customs Profile
Timing Protocol	PTPv2, IEEE 1588-2008
Timing Accuracy	PTP time stamping resolution 8 nanoseconds

Connections	
Network	1 x 100 Mbit/s Ethernet (RJ45/100BASE-TX) with Power over Ethernet (PoE)
GPIO, Power Out and Relay Terminal Block	1 x 24-Pin Phoenix Style
DC Power In	1 x 2.5mm locking DC inlet (Max 24V DC)
GPIO, PTP Enabled	8
GPIO Normal	2
Voltage Free Relay Contact	1 NC, NO, Wiper, 12V

Power over Ethernet (PoE)	
Standard	IEEE 802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	TBC 2.3W
Maxl PSE Power Usage	15.4 W

DC Power Out (Using PoE)	
Max Voltage	12V DC
Max Currentl:	TBC 1.5 A Total

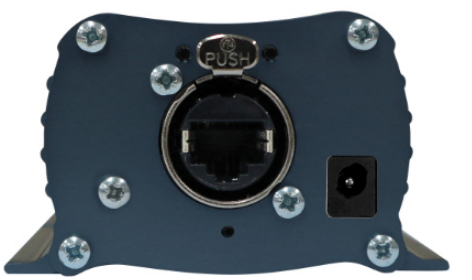
DC Power Out (Using DC Power in)	
Max Voltage	12V DC
Max Current	

Ember+ Interface Connection	
Interface Type	Provider and Consumer
Port	9000

UDP Interface Connection	
Port	31780

Equipment Type
AVN-GPIO GPIO to LAN Transceiver (PTP, EMBER+ & UDP)

Physical Specification:
Dimensions (Raw)
10.6cm (W) x 7.3cm (D) x 4.3cm (H)
4.2" (W) x 2.9" (D) x 1.7" (H)
Dimensions (Boxed)
17.4cm (W) x 9.5cm (D) x 5.6cm (H)
6.9" (W) x 3.7" (D) x 2.2" (H)
Weight
Nett: 0.2kg Gross: 0.3kg
Nett: 0.44lbs Gross' 0.66lbs



AVN-DIO10 Dante® to 3G/HD/SD-SDI Embedder/ De-Embedder

Category: Dante Audio Interfaces.

Product Function: Dante® to 3G/HD/SD-SDI Embedder/De-Embedder.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1 x 3G/HD/SD-SDI input.
- 1 x reclocked 3G/HD/SD-SDI output.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCon® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, SDI Lock, PoE and Sync LEDs.
- DIPSwitch selection of embed channel pairs.
- Overwrite or insert into existing SDI audio groups.
- Test tones available on embedded outputs.
- SDI audio sample rate support at 48kHz.
- All available Dante sample rates supported.
- Sample rate conversion of audio between Dante and SDI.
- Dante clock domain can be optionally synchronised from the SDI clock.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and underdesk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

The easiest way to connect legacy SDI equipment to the Dante® network, the AVN-DIO10 can be used for simultaneous embedding and de-embedding. This simple plug and play audio/video interface provides a convenient and elegant method of connecting legacy 3G/HD/SD-SDI equipment to the Dante® AoIP audio network.

The AVN-DIO10 takes an SDI feed, de-embeds the 16 audio channels and places them on channels 1-16 of the Dante network, mapped using Dante Controller. It simultaneously takes the 16 input channels mapped to the device on Dante Controller and re-embeds them onto the SDI output.

Switches on the unit allow embedding of Dante channels onto the SDI output per channel pair and there are two modes of operation: Insert Mode enabled allows embedding to overwrite existing SDI audio selectively per channel pair. Insert Mode disabled clears any incoming audio channels on the SDI output and then allows selective embedding onto the SDI output per channel pair.

A Test Tone Mode allows 1kHz, 2kHz, 3kHz and 4kHz signals to be output on channels 1 to 4 respectively, for any group where embedding is enabled. This is so that downstream SDI audio outputs can be tested without the need of Dante sources.

It's powered using Power over Ethernet (PoE), using Neutrik EtherCON connectors, with primary and secondary ports for power and data redundancy. The AVN-DIO10 uses the latest Audinate Dante ® chipsets so is AES67 and Dante Domain Manager® compliant.

There are front panel LEDs to indicate network clock status, SDI lock status, AoIP Primary and AoIP Secondary link status, PoE Primary power and PoE Secondary power active.

A web interface is available for firmware updates, status information and network settings.

The AVN-DIO10 is supplied with a belt-clip, shoulder strap hooks (but not the strap) and an underdesk mounting plate. Up to 3 of the AVN-DIO10 units can be rackmount in the 1U AVN-DIORK.

- 1 x 3G/HD/SD-SDI input.
- 1 x reclocked 3G/HD/SD-SDI output.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCon® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, SDI Lock, PoE and Sync LEDs.
- DIPSwitch selection of embed channel pairs.
- Overwrite or insert into existing SDI audio groups.
- Test tones available on embedded outputs.
- SDI audio sample rate support at 48kHz.
- All available Dante sample rates supported.
- Sample rate conversion of audio between Dante and SDI.
- Dante clock domain can be optionally synchronised from the SDI clock.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and underdesk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

Technical Specification For AVN-DIO10

Parameter SDI Input:	
Input Impedance:	75Ω Unbalanced
SDI Supported Standards:	270Mbps SMPTE-259M-C (SD)
SDI Supported Standards:	1.485 or 1.4835Gbps SMPTE-292M (HD)
SDI Supported Standards:	2.97 or 2.967Gbps SMPTE-424M (3G)
Supported Video Formats:	525/59.94 (SMPTE-125M)
Supported Video Formats:	625/50 (ITU-R BT.656)
Supported Video Formats:	720p/23.98, 24, 25, 29.97, 30, 50, 59.94, 60 (SMPTE-296M)
Supported Video Formats:	1035i/59.94, 60 (SMPTE-260M)
Supported Video Formats:	1080i/50, 59.94, 60 (SMPTE-274M)
Supported Video Formats:	1080p/23.98, 24, 25, 50, 59.94, 60 (SMPTE-274M)
Supported Video Formats:	1080pSF/23.98, 24, 25, 29.97, 30 (RP-211)
Supported Video Formats:	1080i/50 (SMPTE-295M)
Supported Video Formats:	1080p/50 (SMPTE-295M)
Embedded Audio:	48kHz, synchronous
Embedded Audio:	SMPTE-272M-ABC
Embedded Audio:	SMPTE-299M
Supported Image Mapping:	SMPTE-425M-AB

Parameter SDI Output:	
Output Impedance:	75Ω Unbalanced
Alignment Jitter:	<0.2UI
Output Level:	800mV ±10%
Return Loss:	<15dB @ 1.5GHz
SDI Supported Standards:	Output follows input
Supported Video Formats:	Output follows input
Embedded Audio:	48kHz, synchronous
Embedded Audio:	SMPTE-272M-C
Embedded Audio:	SMPTE-299M

Network and AoIP	
AoIP Standard:	**Dante**
Channels:	16 receive, 16 transmit
Flows:	16 receive, 16 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support	Yes
Connectivity:	2 x etherCON (RJ45 compatible)
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader), Network PTP Leader or from SDI input (Sync to External)

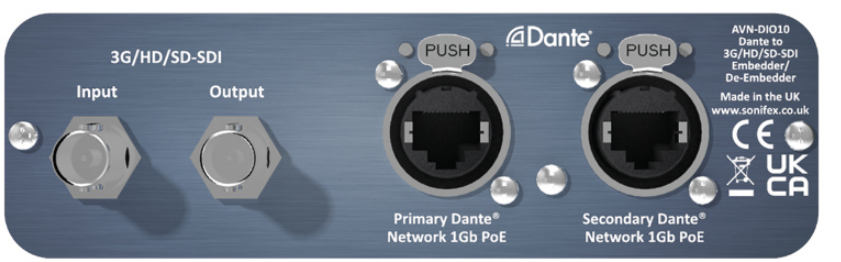
PoE Power	
Standard	802.3af
Redundancy	Yes
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6 W
Max PSE Usage	15.4 W

Equipment Type:	
AVN-DIO10:	Dante to 3G/HD/SD-SDI Embedder/De-Embedder

Physical Specification:	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H) 5.5” (W) x 5.4” (D) x 1.7” (H)
Weight:	Nett: 0.5kg Gross: 0.3kg Nett: 1.1lbs Gross: 0.7lbs

Accessories:	
AVN-DIORK	AVN-DIO 1U 19" Rack Kit (5 x Small DIO or 3 x Large DIO)

AVN-DIO10 Front View.



AVN-DIO10 Rear View.

AVN-DIO12 Dual Microphone Input to Dante® with Mic Gain Converter

Category: Dante Audio Interfaces.

Product Function: Dante® to 3G/HD/SD-SDI Embedder/De-Embedder.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- Neutrik EtherCon® Ethernet connection.
- Single turn pots setting fine mic gain (0dB – 36dB).
- Coarse mic gain switches (+20db/ +50dB).
- High pass filter on/off switches.
- Ultra-high quality, wide dynamic range A/D conversion.
- Phantom power on/off switches.
- Phantom power LED indicators.
- Level LED indicators.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Powered via PoE (Power over Ethernet).
- Clock, PoE and Sync LEDs.
- 3 x units rackmount in the AVN-DIORK

The AVN-DIO12 is a dual microphone input to Dante® converter with adjustable mic gain in the Sonifex DIO range of Dante® input/output devices. It's effectively 2 x AVN-DIO09 units in a single chassis, still offering A/D circuitry with a world-class E.I.N. of 129dB.

It features 2 balanced analogue XLR inputs and one Neutrik etherCON connector for direct connection to a Dante® AoIP network. Each mic has coarse and fine gain controls, high pass filter, phantom power & tri-color level LED and can be routed via Dante Controller to any Dante receiver.

It's powered using Power over Ethernet (PoE), using a Neutrik etherCON connector for power and data redundancy. The AVN-DIO12 uses the latest Audinate Dante chipset so is AES67 and Dante Domain Manager compliant.

Gain Adjustment

The AVN-DIO012 has coarse and fine mic gain for each microphone input. The coarse gain is set using the toggle switch, which provides 20dB/50dB of gain. The fine gain can be set using a trimmer adjustment tool, or small flat blade screwdriver, and adds between 0dB and 36dB of additional gain.

High Pass Filter

Each microphone input has an on/off toggle switch which turns the high pass filter on or off. When enabled, it acts on frequencies below 125Hz at a roll-off of 6dB/octave.

Phantom Power

Phantom power is enabled/disabled via a toggle switch on the front panel for each

microphone input. When enabled, a 48V DC supply is provided to power an appropriate microphone. The red LED next to each input will illuminate to indicate phantom power is enabled.

Audio Level LED

To help you set the mic gain, a level LED is provided on the front panel for each microphone input to display the audio level being sent to the Dante network. The indicator thresholds are as follows:

Off: Level is under -38dBFS (-20dBu)

Green: Level is between -38dBFS (-20dBu) and -18dBFS (0dBu)

Amber:
Level is between -18dBFS (0dBu) and -10dBFS (+8dBu)

Red: Level is above -10dBFS (+8dBu)

Note: When using a phantom powered microphone, it may be necessary to earth the unit using the rear panel earth tag to eliminate mains hum.

- 2 x balanced microphone input on XLR sockets with latch locks.
- Neutrik EtherCon® Ethernet connection.
- Single turn pots setting fine mic gain (0dB – 36dB).
- Coarse mic gain switches (+20db/+50dB).
- High pass filter on/off switches.
- Ultra-high quality, wide dynamic range A/D conversion.
- Phantom power on/off switches.
- Phantom power LED indicators.
- Level LED indicators.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Powered via PoE (Power over Ethernet).
- Clock, PoE and Sync LEDs.
- 3 x units rackmount in the AVN-DIORK.

Technical Specification For AVN-DIO12

Microphone XLR Input Pin-out	
Pin	Function
1	Chassis Ground
2	Input Phase
3	Input Non Phase

Microphone Input - XLR	
Input Impedance:	2.2kΩ balanced
Maximum Input Level:	-68dBu (max gain) to -8dBu (min gain)
Gain:	Adjustable 20dB to 86dB
Frequency Response: 1kHz)	20Hz to 20kHz, +0/-0.5dB (ref
High Pass Filter Response:	Fc = 125Hz @ 6dB per octave
THD+N:	<0.01%, -32dBu input, 40dB gain, 20Hz to 20kHz, 20kHz BW
E.I.N.:	129dBu, 20kHz BW, max gain, Rs=200Ω
Common Mode Rejection:	>60dB @ 1kHz
Phantom Power:	+48V ± 4V

Network and AoIP	
AoIP Standard:	Dante
Channels:	2 transmit
Flows:	2 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support	Yes
Connectivity:	EtherCON (RJ45 compatible)
Speed:	100Mbps
Dante Domain Manager Ready:	Yes
Clock Source: Network PTP Leader	Internal (PTP Leader) or

PoE Power	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	TBD
Max PSE Power Usage	15.4 W

Equipment Type	
AVN-DIO12:	Dual Microphone Input to Dante® with Mic Gain

Physical Specification	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H)
5.5” (W) x 5.4” (D) x 1.7” (H)	
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H)
7.0” (W) x 6.7” (D) x 2.2” (H)	
Weight:	Nett: 0.5kg Gross: 0.7kg
	Nett: 1.1lbs Gross: 1.4lbs

Accessories
AVN-DIOBT
AVN-DIOMT
AVN-DIORK
AVN-DIO Large Unit Belt Clip Kit
AVN-DIO Large Unit Underdesk Mount
AVN-DIO 1U 19" Rack Kit (5 x Small DIO or 3 x Large DIO)

AVN-DIO14 Dante® to XLR Analogue Stereo Input & Output

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Category: Dante Audio Interfaces.

Product Function: Dante® to XLR Analogue Stereo Input & Outpu

The easiest way to connect legacy SDI equipment to the Dante® network, the AVN-DIO10 can be used for simultaneous embedding and de-embedding. This simple plug and play audio/video interface provides a convenient and elegant method of connecting legacy 3G/HD/SD-SDI equipment to the Dante® AoIP audio network.

The front panel provides a global 0dBFS line-up which can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch. There are also front panel LEDs to indicate network clock status, AoIP link status and PoE power active.

It's powered using Power over Ethernet (PoE), using a Neutrik etherCON connector. The AVN-DIO14 uses the latest Audinate Dante chipset so is AES67 and Dante Domain Manager compliant.

Technical Specification For AVN-DIO14

Analogue XLR Input / Output Pin-out	
Pin	Function
1	Chassis Ground
2	Phase
3	Non Phase

Line Inputs - XLR	
Input Impedance:	5kΩ Unbalanced
0dBFS Line-Up:	Selectable +12/+18/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB (ref 1kHz)
THD+N:	<-118dBFS, -12dBu (+18dBu=0dBFS mode), 20Hz to 20kHz, 20kHz BW
Dynamic Range	120dB, 20kHz BW, Rs=200Ω
Cross Talk	<-110dB
Common Mode Rejection	>60dB @ 1kHz

Line Outputs - XLR	
Output Impedance:	<200Ω balanced
0dBFS Line-Up:	Selectable +12/+18/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.5dB (ref 1kHz)
THD+N:	<-100dBu, -30dBFS, 20Hz to 20kHz, 20kHz BW
Dynamic Range	120dB, 20kHz BW
Cross Talk	<-110dB

Network and AoIP	
AoIP Standard:	Dante
Channels:	2 transmit, 2 receive
Flows:	2 transmit, 2 receive
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support	Yes
Connectivity:	etherCON (RJ45 compatible)
Speed:	100Mbps
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader) or Network PTP Leader

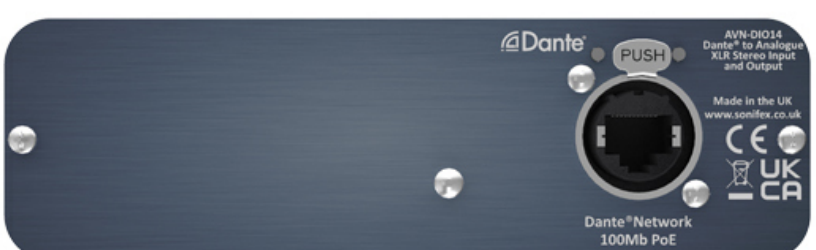
PoE Power	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6 W
Max PSE Usage	15.4 W

Equipment Type	
AVN-DIO14:	Dante® to XLR Analogue Stereo Input & Output converter

Physical Specification	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H)
	5.5” (W) x 5.4” (D) x 1.7” (H)
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H)
	7.0" (W) x 6.7" (D) x 2.2" (H)
Weight:	Nett: 0.42kg Gross: 0.56kg
	Nett: 0.95lbs Gross: 1.25lbs

Accessories	
AVN-DIOBT	AVN-DIO Large Unit Belt Clip Kit
AVN-DIOMT	AVN-DIO Large Unit Underdesk Mount
AVN-DIORK	AVN-DIO 1U 19" Rack Kit (5 x Small DIO or 3 x Large DIO)

AVN-DIO14 Front View.



AVN-DIO14 Rear View.

AVN-DIO15 4 Analogue XLR Inputs to Dante®

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Category: Dante Audio Interfaces.

Product Function: 4 Analogue XLR Inputs to Dante®

The AVN-DIO15 is a 4 analogue XLR input to Dante converter in the Sonifex DIO range of Dante input/output devices. It features four balanced analogue XLR inputs and one Neutrik etherCON connector for direct connection to a Dante AoIP network.

The front panel provides a global 0dBFS line-up which can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch. There are also front panel LEDs to indicate network clock status, AoIP link status and PoE power active.

It's powered using Power over Ethernet (PoE), using a Neutrik etherCON connector. The AVN-DIO15 uses the latest Audinate Dante chipset so is AES67 and Dante Domain Manager compliant.

Technical Specification For AVN-DIO15

Analogue XLR Input Pin-out	
Pin	Function
1	Chassis Ground
2	Phase
3	Non Phase

Line Inputs - XLR	
Input Impedance:	5kΩ balanced
0dBFS Line-Up:	Selectable +12/+18/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB (ref 1kHz)
THD+N:	
<-118dBFS, -12dBu (+18dBu=0dBFS mode), 20Hz to 20kHz, 20kHz BW	
Dynamic Range	120dB, 20kHz BW, Rs=200Ω
Cross Talk	<-110dB
Common Mode Rejection	>60dB @ 1kHz

Network and AoIP	
AoIP Standard:	Dante
Channels:	4 transmit
Flows:	2 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support	Yes
Connectivity:	etherCON (RJ45 compatible)
Speed:	100Mbps
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader), Network PTP Leader or from SDI input (Sync to External)

PoE Power	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W

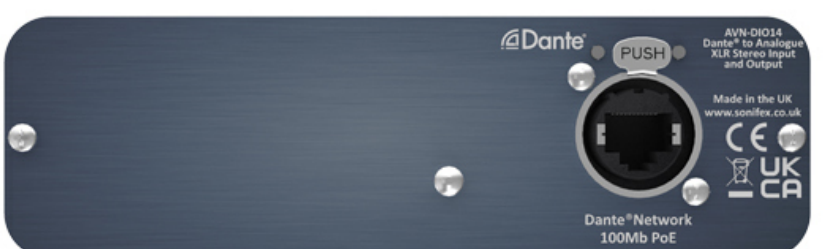
Typical PSE Power Usage	**6 W**
Max PSE Usage	15.4 W

Equipment Type	
AVN-DIO15:	4 Analogue XLR Inputs to Dante®

Physical Specification	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H)
	5.5” (W) x 5.4” (D) x 1.7” (H)
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H)
	7.0" (W) x 6.7" (D) x 2.2" (H)
Weight:	Nett: 0.42kg Gross: 0.56kg
	Nett: 0.95lbs Gross: 1.25lbs

Accessories	
AVN-DIOBT	AVN-DIO Large Unit Belt Clip Kit
AVN-DIOMT	Large Unit Underdesk Mount
AVN-DIORK	AVN-DIO 1U 19" Rack Kit (5 x Small DIO or 3 x Large DIO)

AVN-DIO15 Front View.



AVN-DIO15 Rear View.

AVN-DIO16 to 4 Analogue XLR Outputs

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Category: Dante Audio Interfaces.

Product Function: Dante® to 4 Analogue XLR Outputs

The AVN-DIO16 is a 4 analogue XLR output to Dante converter in the Sonifex DIO range of Dante input/output devices. It features four balanced analogue XLR outputs and one Neutrik etherCON connector for direct connection to a Dante AoIP network.

The front panel provides a global 0dBFS line-up which can be set to +12dBu, +18dBu or +24dBu to meet your specific requirement via the front panel recessed toggle switch. There are also front panel LEDs to indicate network clock status, AoIP link status and PoE power active.

It's powered using Power over Ethernet (PoE), using a Neutrik etherCON connector. The AVN-DIO16 uses the latest Audinate Dante chipset so is AES67 and Dante Domain Manager compliant.

Technical Specification For AVN-DIO16

Analogue XLR Input Pin-out	
Pin	Function
1	Chassis Ground
2	Phase
3	Non Phase

Line Inputs - XLR	
Input Impedance:	5kΩ balanced
0dBFS Line-Up:	Selectable +12/+18/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB (ref 1kHz)
THD+N:	
<-118dBFS, -12dBu (+18dBu=0dBFS mode), 20Hz to 20kHz, 20kHz BW	
Dynamic Range	120dB, 20kHz BW, Rs=200Ω
Cross Talk	<-110dB
Common Mode Rejection	>60dB @ 1kHz

Network and AoIP	
AoIP Standard:	Dante
Channels:	4 transmit
Flows:	2 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support	Yes
Connectivity:	etherCON (RJ45 compatible)
Speed:	100Mbps
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader), Network PTP Leader or from SDI input (Sync to External)

PoE Power	
Standard	802.3af
Class	0
PD Power Range	0.44 W to 12.94 W

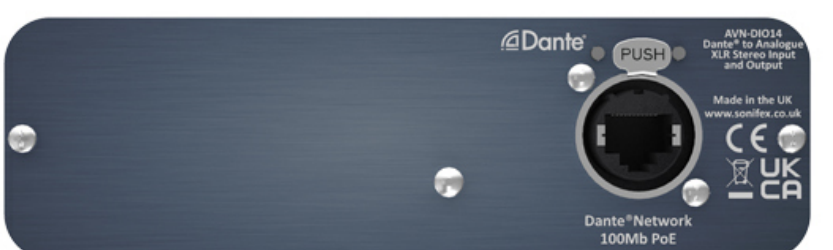
Typical PSE Power Usage	6 W
Max PSE Usage	15.4 W

Equipment Type	
AVN-DIO15:	4 Analogue XLR Inputs to Dante®

Physical Specification	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H)
5.5” (W) x 5.4” (D) x 1.7” (H)	
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H)
7.0" (W) x 6.7" (D) x 2.2" (H)	
Weight:	Nett: 0.42kg Gross: 0.56kg
	Nett: 0.95lbs Gross: 1.25lbs

Accessories	
AVN-DIOBT	AVN-DIO Large Unit Belt Clip Kit
AVN-DIOMT	Large Unit Underdesk Mount
AVN-DIORK	AVN-DIO 1U 19" Rack Kit (5 x Small DIO or 3 x Large DIO)

AVN-DIO16 Front View.



AVN-DIO16 Rear View.

AVN-DIO19 Dante® to AES3 16 Channel I/O Converter

Category: Dante Audio Interfaces.

Product Function: Dante® to AES3 16 Channel I/O Converter.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 8 x balanced digital stereo AES3 inputs and outputs on 2 x 25-way D-types.
- Sample rate conversion of physical inputs to Dante system sample rate.
- Physical output sample rate matches Dante system sample rate.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCON® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, PoE, Network link and AES3 input lock LEDs.
- All available Dante sample rates supported.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and under desk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

The AVN-DIO19 audio converter and interface converts up to eight digital stereo AES3 inputs and eight digital stereo AES3 outputs to and from the Dante Audio-over-IP networking standard. Each input can accept sample rates from 32kHz to 192kHz, which will be sample rate converted to the Dante system sample rate. All outputs follow the Dante system sample rate.

It's powered using Power over Ethernet (PoE), using Neutrik EtherCON® connectors, with primary and secondary ports for power and data redundancy. The AVN-DIO19 uses the latest Audinate Dante® chipsets so is AES67 and Dante Domain Manager® compliant.

There are front panel LEDs to indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary power and PoE Secondary power active. In addition to these there are 8 AES3 input lock status LEDs.

A web interface is available for firmware updates, status information, network and device settings.

The AVN-DIO19 is supplied with a belt-clip, shoulder strap hooks (but not the strap) and an under desk mounting plate. Up to 3 of the AVN-DIO19 units can be rackmount in the 1U AVN-DIORK.

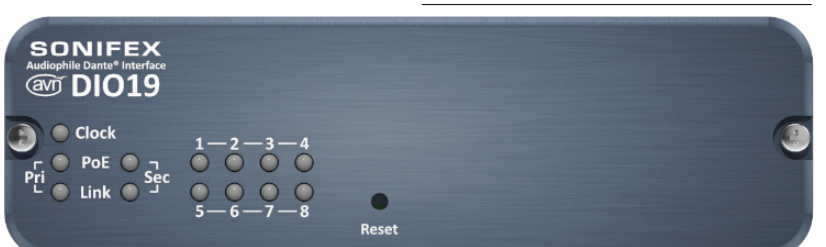
- 8 x balanced digital stereo AES3 inputs and outputs on 2 x 25-way D-types.
- Sample rate conversion of physical inputs to Dante system sample rate.

- Physical output sample rate matches Dante system sample rate.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCON® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, PoE, Network link and AES3 input lock LEDs.
- All available Dante sample rates supported.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and under desk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

Technical Specification For AVN-DIO19

AES3 I/O	
Input & Output Impedance:	110Ω Balanced
Sample Rates:	32 – 192kHz
Network and AoIP	
AoIP Standard:	**Dante**
Channels:	16 receive, 16 transmit
Flows:	16 receive, 16 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Encoding:	PCM 16, PCM 24, PCM 32
Connectivity:	2 x etherCON (RJ45 compatible)
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
AES67 Support:	Yes
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader) or from Network PTP Leader
PoE Power	
Standard	802.3af
Redundancy	Yes
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	4.6W
Max PSE Usage	15.4 W
Equipment Type	
AVN-DIO19:	Dante to AES3 16 channel I/O converter
Physical Specification	
Dimensions (Raw): (H)	14.0cm (W) x 13.6cm (D) x 4.2cm 5.5” (W) x 5.4” (D) x 1.7” (H)
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H) 7.0” (W) x 6.7” (D) x 2.2” (H)
Weight:	TBD

Accessories	
AVN-DIORK:	AVN-DIO 1U 19” rack kit (5 x small DIO or 3 x large DIO)
CBL-D25-4XI4XO:	AES3 balanced cable, DB25 to 4 x XLR3M and 4 x XLR3F, 3m
CBL-D25-D25:	AES3 balanced cable, DB25 to DB25, 3m



AVN-DIO19 Front View.

AVN-DIO19 Rear View.

AVN-DIO20 Dante® to MADI AES3 64 Channel I/O Converter

Category: Dante Audio Interfaces.

Product Function: Dante® to MADI AES3 64 Channel I/O Converter.

Typical Applications: These simple plug and play audio interfaces provide a convenient and elegant method of connecting legacy analogue and digital audio equipment to the Dante AoIP audio network.

Features:

- 1 x AES10 MADI input and output, on either SFP or coaxial BNC (up to 64 channels of MADI I/O).
- 8 x stereo AES3 inputs and outputs on 2 x 25-way D-types, using AES59 digital pinout.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCON® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, PoE, Network link, AES3 input lock and MADI active LEDs.
- MADI 64, 56, 28 and 16 channel support and audio sample rate support up to 192kHz.
- MADI SFP to BNC automatic failover.
- All available Dante sample rates supported (44.1kHz to 192kHz).
- Sample rate conversion of audio between Dante and MADI/AES3.
- Dante clock domain can be optionally synchronised from the MADI source.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and under desk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

The AVN-DIO20 is a MADI and AES3 to Dante bridging device allowing MADI to connect bidirectionally with AoIP, together with 8 stereo channels of AES3. This simple plug and play audio interface provides a quick and convenient method of connecting legacy MADI equipment to the Dante AoIP audio network.

The AVN-DIO20 takes a MADI feed, sample rate converts all 64 channels and places them on the Dante network, mapped using Dante Controller. It simultaneously takes the 64 channels mapped to the device on Dante Controller, optionally sample rate converts them, and transmits them on the MADI output. The unit accepts 8 stereo AES3 inputs and place them on the Dante network, replacing 16 of the selected MADI input channels. The unit also sends 8 stereo AES3 outputs which can be sourced from the Dante channels, in blocks of 16.

The unit accepts MADI signals from either a compatible SFP module or the coaxial BNC input. The input can be selected via the built-in web GUI or, by default, it will use an SFP module if it is inserted into the unit.

The coaxial BNC output is disabled while the SFP is in use. There is also an additional automatic failover mode which, when selected, allows for automatic switching to the BNC input when the SFP signal is lost, providing glitch free audio redundancy.

The MADI output audio can be clocked from either the Dante network, in which case the output SRC is bypassed, or the recovered clock obtained from the MADI input.

It's powered using Power over Ethernet (PoE), using Neutrik EtherCON® connectors, with primary and secondary ports for power and data redundancy. The AVN-DIO20 uses the latest Audinate Dante™ chipsets so is AES67 and Dante Domain Manager™ compliant.

There are front panel LEDs to indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary power and PoE Secondary power active. In addition there are 8 AES3 input lock status LEDs and an active MADI input indication.

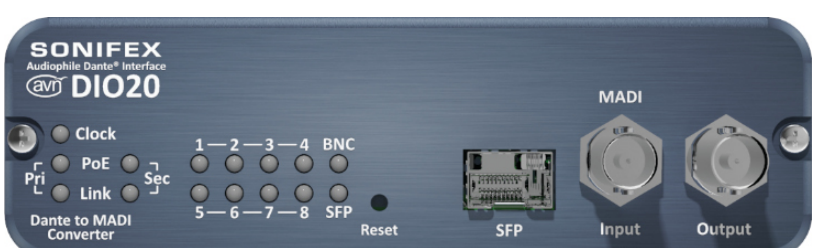
A web interface is available for network and device settings, status information and firmware updates.

The AVN-DIO20 is supplied with a belt-clip, shoulder strap hooks (but not the strap) and an under desk mounting plate. Up to 3 of the AVN-DIO20 units can be rackmount in the 1U AVN-DIORK.

- 1 x AES10 MADI input and output, on either SFP or coaxial BNC (up to 64 channels of MADI I/O).
- 8 x stereo AES3 inputs and outputs on 2 x 25-way D-types, using AES59 digital pinout.
- Dual redundant Primary and Secondary Dante network ports using Neutrik EtherCON® Ethernet connectors.
- Powered via PoE (Power over Ethernet) with PoE dual redundancy.
- Fully Dante compliant device.
- AES67 compatible.
- Dante Domain Manager compliant.
- Web interface for configuration.
- Clock, PoE, Network link, AES3 input lock and MADI active LEDs.
- MADI 64, 56, 28 and 16 channel support and audio sample rate support up to 192kHz.
- MADI SFP to BNC automatic failover.
- All available Dante sample rates supported (44.1kHz to 192kHz).
- Sample rate conversion of audio between Dante and MADI/AES3.
- Dante clock domain can be optionally synchronised from the MADI source.
- Supplied with belt-clip, shoulder strap hooks (but not the strap) and under desk mounting plate.
- 3 x units rackmount in the AVN-DIORK.

Technical Specification For AVN-DIO20

MADI I/O	
BNC Input & Output Impedance:	75Ω Unbalanced
MADI Input Sample Rate/Channels:	48kHz – 56/64 channels 96kHz – 28/32 channels 192kHz – 16 channels
MADI Output Sample Rate/Channels:	Follow input, Dante sample rate
Connections:	Coaxial BNC input and output SFP - LVDS SFP 100MB/s
AES3 I/O	
Input& Output Impedance:	110Ω Balanced
Sample Rates:	32 – 192kHz
Connections:	8 x stereo AES3 inputs and outputs on 2 x 25-way D-types using AES59 pinout
Network and AoIP	
AoIP Standard:	**Dante**
Channels:	64 receive, 64 transmit
Flows:	32 receive, 32 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Encoding:	PCM 16, PCM 24, PCM 32
AES67 Support:	Yes
Connectivity:	2 x etherCON (RJ45 compatible)
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader), Network PTP Leader or from MADI input)
PoE Power	
Standard	802.3af
Redundancy	Yes
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	6.1W
Max PSE Usage	15.4 W
Equipment Type	
AVN-DIO20:	Dante to MADI/AES3 64 channel I/O converter
Physical Specification	
Dimensions (Raw):	14.0cm (W) x 13.6cm (D) x 4.2cm (H) 5.5” (W) x 5.4” (D) x 1.7” (H)
Dimensions (Boxed):	17.8cm (W) x 17cm (D) x 5.6cm (H) 7.0” (W) x 6.7” (D) x 2.2” (H)
Weight:	Nett: 0.5kg Gross: 0.9kg Nett: 1.1lbs Gross: 2.0lbs
Accessories	
AVN-DIORK:	AVN-DIO 1U 19” rack kit (5 x small DIO or 3 x large DIO)
CBL-D25-4XI4XO:	AES3 balanced cable, DB25 to 4 x XLR3M and 4 x XLR3F, 3m
CBL-D25-D25:	AES3 balanced cable, DB25 to DB25, 3m



AVN-DIO20 Front View.

AVN-DIO20 Rear View.

AVN-DIORK 19” AVN-DIO Mounting Rack

Category: Accessories.

Product Function: A 1U rackmount kit for AVN-DIO Audiophile boxes.

Typical Applications: A 1U rackmount kit, providing a tray to hold AVN-DIO Audiophile boxes.

Features:

- 1U rackmount tray.
- Can take up to 5 x AVN-DIO01-09 units.
- Alternatively, can take up to 3 x AVN-DIO10 sized boxes.
- Supplied with 5 sets of AVN-DIO mounting screws and nuts..

The AVN-DIORK is a 1U rackmount kit that can accept up to 5 of the smaller AVN-DIO01-9 boxes, or 3 of the AVN-DIO10 boxes. It is supplied complete with AVN-DIO box fixings.

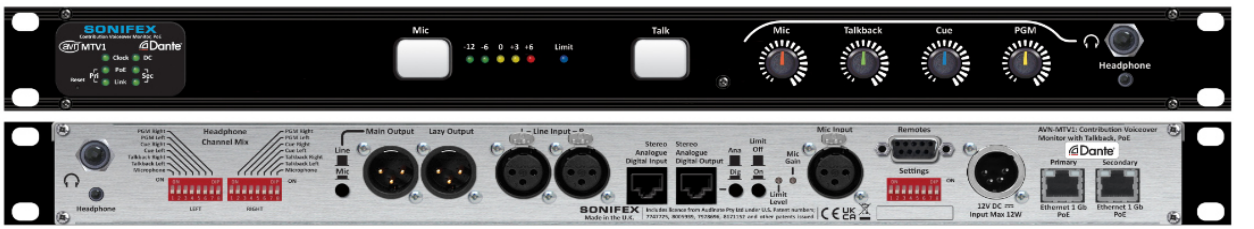
Please note: rack comes empty.

AVN-MTV1 Dante Contribution Voiceover Monitor with Talkback

Category: Dante AES67 Audio Over IP

Product Function: Contribution Voiceover Monitor with Talkback

Typical Applications: Applications where voice needs to be added to programme content and then monitored



AVN-MTV1 Front View.

The AVN-MTV1 is an upgrade of the RB-MTV1, adding Audinate Dante AoIP capability to the Redbox voiceover monitor. It is a 1U rack-mount unit designed to be used in voice-over booths, news booths, commentary locations, for continuity announcements and for any other similar applications where voice needs to be added to programme content and then monitored. Talkback, Cue and Program feeds can be taken from the Dante network and monitored.

The AVN-MTV1 has three physical inputs and three physical outputs. It has a mono microphone input on an XLR with physically or remote switched coarse gain and physical variable fine fain control using a multi-turn preset potentiometer to give an overall gain range from +20dB to +80dB. There is also an LF rumble filter, +48V phantom power and level limiting control. A rear-panel multi-turn preset potentiometer allows adjustment of the threshold at which the limiter begins to operate, from -8dBu to +26dBu. There is an indication of the limiter activity using a blue LED and the microphone level is monitored by a simple 5 LED meter. The meter can be configured to either show the MIC signal activity in normal operation (i.e. when TALK is pressed the meter is off) or it can permanently show the MIC activity even when the TALK button is on or the MIC button is off.

There is a balanced XLR stereo input and an auto-sensing balanced analogue/ digital input on an RJ45 (StudiohubTM pinout). Each of these inputs are sent to the Dante network.

The Talkback, Cue and Program streams come from the Dante network, as well as the Mic input, and can be mixed and monitored into the front and rear ¼"/ 3.5mm headphone jack outputs with individual volume controls. Which of the inputs is presented to the headphone outputs can be configured using two banks of DIP switches on the rear of the unit (allowing control over left and right headphone mixes). If a presenter doesn't like to hear themselves in their headphones when using the TALK button, there is also an option to mute the mic signal to the headphones.

The processed microphone signal is fed to both the main mono balanced output on an XLR and the Main Output Dante stream when the latching front panel MIC button is active – the button illuminates when active.
The main output XLR can be independently switched to be at "Line" or "Mic" level outputs using the rear panel push switch. Setting the output to a microphone level allows the unit to be inserted into the microphone input of a mixing desk. There is an option to permanently enable the MIC button, even when remotely controlled, for

occasions when you always want the MIC channel left open. Additionally, there is an option to mix the Cue input as a mono feed to the outputs permanently.

The front panel TALK button is a momentary push switch that routes the processed microphone signal to both the rear balanced XLR Lazy Output and the Lazy Output Dante stream, whilst disconnecting it from the main outputs, allowing the operator to talk to a colleague. This enables the unit to be used as a talkback intercom between two or more studios.

There is a switchable digital/analogue balanced output on an RJ45 which can be used as an exit point from the Dante network.

There is a rear panel remotes connector giving remote control of the two front panel MIC and TALK buttons and opto-isolated tallies of their status.

The unit is powered using Power over Ethernet (PoE) with primary and secondary ports for power and data redundancy. The AVN-MTV1 uses the latest Audinate Dante chipsets so is AES67 and Dante Domain Manager compliant.

There are front panel LEDs to indicate network clock status, AoIP Primary and AoIP Secondary link status, PoE Primary power and PoE Secondary power active.

Technical Specification For AVN-MTV1

Network and AoIP	
AoIP Standard:	Dante
Number of channels:	8 receive, 6 transmit
Number of streams:	8 receive, 6 transmit
Sample Rate:	44.1kHz, 48 kHz, 88.2kHz, 96kHz, 176.4kHz or 192kHz
Format:	Linear PCM, 16, 24 or 32 bit
AES67 Support:	Yes
Connectivity:	2 x RJ45
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
Dante Domain Manager Ready:	Yes

Analogue Input Performance Specifications	
Parameter	**Mic Input**
Input Impedance:	2kΩ nominal balanced
Maximum Input Level:	-10dBu
Mic Gain Range:	Adjustable +20dB to +80dB
Frequency Response:	+0/-0.5dB 20Hz to 20kHz, 40dB gain @ 1kHz
E.I.N.:	<-125dB ref 80dB gain, Rs=200Ω
Common Mode Rejection:	>60dB @ 1kHz
Low Frequency Roll-Off:	125Hz @ 6dB/octave
THD+N:	<-90dBFS, -64dBFS, 20Hz to 20kHz, 40dB gain, 20kHz BW
Phantom Power:	48V
Limiter Threshold:	Adjustable +8dBu to +18dBu

Parameter	**Line Input**
Input Impedance:	>20kΩ balanced
0dBFS Line-up	+18dBu
Frequency Response:	+0/-0.5dB 20Hz to 20KHz
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise	<-110dBFS, 20kHz BW, Rs=200Ω
Common Mode Rejection:	>60dB @ 1kHz
Output Impedance:	<50Ω balanced

Analogue Output Performance Specifications	
Parameter	Line Output
0dBFS Line-up:	+18dBu
Frequency Response:	+0/-0.5dB 20Hz to 20KHz
THD+N:	<-107dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	<-107dBFS, 20kHz BW

Parameter	Headphone Output
Drive capability: headphones	Drives 150mW into 32Ω to 600Ω
0dBFS Line-up:	+18dBu
Frequency Response:	+0/-0.5dB 20Hz to 20KHz
THD+N:	<-107dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	<-107dBFS, 20kHz BW

AES3 Input Specifications	
Input Impedance:	110Ω balanced
Supported Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to Dante system sample rate)
Signal Level:	AES3-2009 Compliant
Bit Depth:	Up to 24 bits

AES3 Output Specifications	
Output Impedance:	110Ω balanced
Supported Output Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (set as per Dante system sample rate)
Signal Level:	AES3-2009 Compliant
Bit Depth:	Up to 24 bits

Rear Panel Connections	
Mic Input:	1 x XLR 3 pin female (Balanced)
Line/Digital Input:	2 x XLR 3 pin female (Balanced) 1 x RJ45 (Balanced, Analogue/Digital Auto-sensing)
Main Output:	1 x XLR 3 pin male (Balanced)
TALK (LAZY) Output:	1 x XLR 3 pin male (Balanced)
Headphone Outputs:	2 x ¼" (6.35mm)/3.5mm A-gauge 3-pole stereo jack sockets
Line/Digital Output:	1 x RJ45 (Balanced, Analogue/Digital Switchable)
Remote I/O Port:	9-way 'D'-type socket
DC Input:	1 x 4 Pin XLR, 12V nominal, 12W Max

Network and AoIP	
AoIP Standard:	Dante
Channels:	8 receive, 6 transmit
Flows:	8 receive, 6 transmit
Sample Rates:	44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz
Encoding:	PCM 16, PCM 24, PCM 32
Connectivity:	2 x RJ45
Speed:	1Gbps or 100Mbps
Network Modes:	Switched or redundant
AES67 Support:	Yes
Dante Domain Manager Ready:	Yes
Clock Source:	Internal (PTP Leader) or from Network PTP Leader

PoE Power	
Standard	802.3af
Redundancy	Yes
Class	0
PD Power Range	0.44 W to 12.94 W
Typical PSE Power Usage	TBD
Max PSE Usage	15.4 W

Equipment Type	
AVN-MTV1:	Dante Contribution Voiceover Unit With Talkback

Physical Specification	
Dimensions (Raw):	48cm (W) x 11cm (D) x 4.3cm (H)
19" (W) x 4.3" (D) x 1.7" (H) (1U)	
Weight:	Nett: 1.3kg Gross: 1.9kg
	Nett: 2.9lbs Gross: 4.2lbs

AVN-GMCS IEEE1588 PTP Grandmaster Clock with GPS Receiver

Category: AES67 PTP Grandmaster Clocks.

Product Function: IEEE1588 PTP Grandmaster Clock with GPS Receiver.

Typical Applications: A PTPv2 grandmaster clock for use with AoIP applications.

Features:

- AES67/RAVENNA/AES-R16 compatible.
- GPS satellites received indicator.
- Master and slave sync modes.
- 8ns PTPv2 time stamping resolution.
- Holdover drift < 1ppm, with an option for < 0.01ppm.
- AES-3id, wordclock & variable PPS outputs.
- Analogue master input.
- Dual power supply inputs.
- Front panel display.
- GPS sync and power supply present LED indicators.
- Real time clock for accurate date/time.
- Responsive design Ethernet webserver and front panel control & configuration.

The AVN-GMCS is a PTPv2 grandmaster clock for use with AoIP applications. IEEE1588-2008 PTPv2 (precision time protocol) is used to synchronise all the nodes within a network. To achieve this one of the nodes must become the master clock and distribute time packets to the others. The AVN-GMCS is designed to perform this role simply and accurately, enabling sub micro second synchronisation between all nodes.

RAVENNA (of which AES67 is a subset) allows for the distribution of audio across a network. For this to be possible, each of the nodes needs to be time synchronised with one another. RAVENNA uses PTP time stamping to achieve this, which distributes the network time but also works out the latency involved in the delivery and adjusts the time at each node accordingly.

Unit configuration is achieved easily either with the front panel controls or the webserver, including the setup of the PTP profiles.

The AVN-GMCS supports the Default (RAVENNA), Media (AES67) and AES-R16-2016 (SMPTE-ST 2059-2 & AES67 compatible) profiles and has a 'Custom' profile page for you to define your own.

In normal operation, the unit has PTPv2 time stamping resolution to 8nsec. It uses a combination of a GPS receiver, a PLL (phase lock loop) and a specialist on-board clock device to create the precise, low jitter clock signals required to drive the physical transceiver's time stamping circuitry, also providing holdover if the GPS signal is lost.

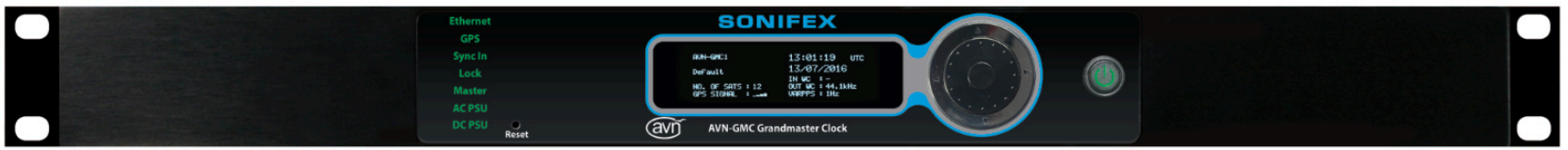
The specialist on board clock is available in two different types: TCXO & OXCO, which vary in both price and accuracy:

AVN-GMCS – TCXO Temperature Compensated Oscillator accurate to 1 part per million (worst case 1 sec gain/loss every 11.5 days). *

AVN-GMCOS – OCXO Oven Controlled Oscillator accurate to 0.1 parts per million (worst case 1 sec gain/loss every 116 days). *

GPS presence and the number of satellites received is shown on the front panel, together with status information on output sample rates, sync type and profile type. The unit also has a screen-saver option which shows the current time.

Although designed as a grandmaster clock, a separate clock input can act as an alternative



AVN-GMCS Front View.

AVN-GMCS Rear View.

reference source to GPS which the unit can 'slave' to. Clock outputs, driven from the physical transceiver, can be used to provide media clocks for external equipment local to the AVN-GMCS when it is in both 'master' and 'slave' states. The clock outputs are available as a single AES-3id output and two outputs which can be selected as either word clock or variable PPS. The wordclock can operate at 32, 44.1, 48, 96, 176.4 and 192kHz. When set as a variable PPS output, the unit can act as a clock master to distribute a reference frequency to test and measurement equipment.

The unit shows UTC as standard, but can be set to show 'local time' on the front panel, by adding a time offset. Daylight saving time changes can be accommodated by entering Spring Forward and Fall Back dates. It has a real time clock so that accurate date and time is available even after the unit is repowered without GPS access.

The built-in webserver, or front panel OLED display, can be used to configure the unit. The webserver is a responsive design meaning that it can be used with small screens on smartphones and tablets.

Front panel LEDs show the synchronisation status, GPS lock and the status of the AC and DC power supply inputs.

The brightness of the OLED display and LED indicators can be adjusted for low or high lighting conditions 4 general purpose outputs indicate critical states for the unit using a 9 way D-type connector mounted on the rear panel. Pull down when active pins are supplied for GPS lock status, external sync present, AC power present and DC power present.

The unit has a front panel power button and dual power connectors - an IEC mains input and a 12V DC input, which allows the AVN-GMCS to be used for both studio and mobile installations. Moreover this allows for a secondary power source to reduce the effect of power down events. In any case, the unit monitors the status of both power sources and displays this on the front panel.

The unit can be put into a low-power sleep mode when not in use, with an instant start when power is re-applied. In power off situations, a super capacitor is used to keep the GPS receiver powered in a low power mode for more than 20 hours, enabling the receiver to regain lock immediately rather than having to 'cold' start.

- AES67/RAVENNA/AES-R16 compatible.
- GPS satellites received indicator.
- Master and slave sync modes.
- 8ns PTPv2 time stamping resolution.
- Holdover drift <1ppm, with an option for <0.01ppm.
- AES-3id, wordclock & variable PPS outputs.
- Analogue master input.
- Dual power supply inputs.
- Front panel display.
- GPS sync and power supply present LED indicators.
- Real time clock for accurate date/time.
- Responsive design Ethernet webserver and front panel control & configuration.

Technical Specification For AVN-GMCS

Timing Specification:	
Profile Support:	Default (RAVENNA), Media (AES67), AES-R16-2016 (SMPTE-ST 2059-2 & AES67 compatible), Custom profile
Timing Protocol:	PTPv2, IEEE1588-2008
Timing Accuracy:	PTP time stamping resolution 8ns
Holdover Drift: TCXO:	<90ms
Holdover Drift:	These figures are over 24 hours at constant temperature
GPS Performance:	50 channel GPS receiver
GPS Frequency:	1575.42MHz, L1 band
Clock Specification:	
Input Impedance:	75Ω
Output Impedance:	<75Ω
Antenna Impedance:	50Ω
Connections:	
Clocking Input:	BNC female
Clocking Outputs:	3 x BNC female Wordclock, AES-3id @ 32, 44.1, 48, 96, 176.4 & 192kHz Variable PPS (1, 10, 100, 1000) TTL
GPS Input:	SMA socket
GPIO:	D-Type female 9 way
Ethernet Port:	RJ45 socket, 100 BASE-T
Mains AC Input:	Universal filtered IEC socket, continuously rated 85-264 VAC @47-63Hz, max 10W
DC Input:	1 x 12V, KPJX-45 socket, positive pins 1 and 3
Maximum Operating Range (DC):	10.3V to 13.2V DC
Equipment Type:	
AVN-GMCS	Grandmaster clock for PTP systems, GPS, IP, TCXO, 1ppm, rackmount
Physical Specification:	
Dimensions (Raw):	4.4cm (H) x 48.3cm (W) x 17.8cm (D(1U) 1.8” (H) x 19” (W) x 7” (D) (1U)
Dimensions (Boxed):	6.8cm (H) x 58.8cm (W) x 27cm (D) 2.7” (H) x 23” (W) x 10.6” (D)
Weight:	Nett: 1.5kg Gross: 2.2kg Nett: 3.2lbs Gross: 4.8lbs
AVN-DC150	150W DC power supply with KPJX-4S plug

AVN-GMCOS IEEE1588 PTP Grandmaster Clock with GPS Receiver

Category: AES67 PTP Grandmaster Clocks.

Product Function: IEEE1588 PTP Grandmaster Clock with GPS Receiver.

Typical Applications: A PTPv2 grandmaster clock for use with AoIP applications.

Features:

- AES67/RAVENNA/AES-R16 compatible.
- GPS satellites received indicator.
- Master and slave sync modes.
- 8ns PTPv2 time stamping resolution.
- Holdover drift < 1ppm, with an option for < 0.01ppm.
- AES-3id, wordclock & variable PPS outputs.<
- Analogue master input.
- Dual power supply inputs.
- Front panel display.
- GPS sync and power supply present LED indicators.
- Real time clock for accurate date/time.
- Responsive design Ethernet webserver and front panel control & configuration.

The AVN-GMCOS is a PTPv2 grandmaster clock for use with AoIP applications which uses a temperature controlled crystal for GPS holdover accuracy. IEEE1588-2008 PTPv2 (precision time protocol) is used to synchronise all the nodes within a network. To achieve this one of the nodes must become the master clock and distribute time packets to the others. The AVN-GMCOS is designed to perform this role simply and accurately, enabling sub micro second synchronisation between all nodes.

RAVENNA (of which AES67 is a subset) allows for the distribution of audio across a network. For this to be possible, each of the nodes needs to be time synchronised with one another. RAVENNA uses PTP time stamping to achieve this, which distributes the network time but also works out the latency involved in the delivery and adjusts the time at each node accordingly.

Unit configuration is achieved easily either with the front panel controls or the webserver, including the setup of the PTP profiles.

The AVN-GMCOS supports the Default (RAVENNA), Media (AES67) and AES-R16-2016 (SMPTE-ST 2059-2 & AES67 compatible) profiles and has a 'Custom' profile page for you to define your own.

In normal operation, the unit has PTPv2 time stamping resolution to 8nsec. It uses a combination of a GPS receiver, a PLL (phase lock loop) and a specialist on-board clock device to create the precise, low jitter clock signals required to drive the physical transceiver's time stamping circuitry, also providing holdover if the GPS signal is lost.

The specialist on board clock is available in two different types: TCXO & OXCO, which vary in both price and accuracy:

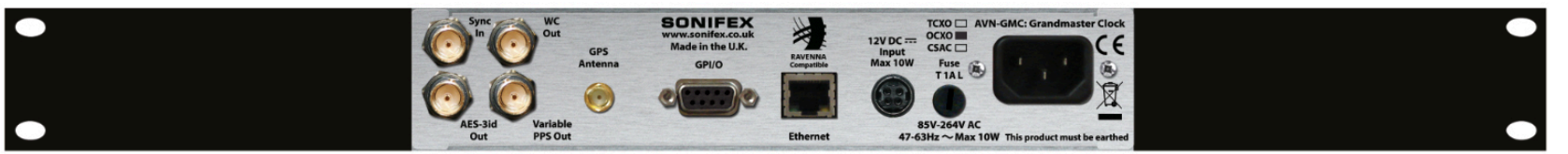
AVN-GMCS – TCXO Temperature Compensated Oscillator accurate to 1 part per million (worst case 1 sec gain/loss every 11.5 days). *

AVN-GMCOS – OCXO Oven Controlled Oscillator accurate to 0.1 parts per million (worst case 1 sec gain/loss every 116 days).

GPS presence and the number of satellites received is shown on the front panel, together with status information on output sample rates, sync type and profile type. The unit also has a screen-saver option which shows the current time.

Although designed as a grandmaster clock, a separate clock input can act as an alternative ▶

AVN-GMCOS Front View.



AVN-GMCOS Rear View.

reference source to GPS which the unit can 'slave' to. Clock outputs, driven from the physical transceiver, can be used to provide media clocks for external equipment local to the AVN-GMCOS when it is in both 'master' and 'slave' states. The clock outputs are available as a single AES-3id output and two outputs which can be selected as either word clock or variable PPS. The wordclock can operate at 32, 44.1, 48, 96, 176.4 and 192kHz. When set as a variable PPS output, the unit can act as a clock master to distribute a reference frequency to test and measurement equipment.

The unit shows UTC as standard, but can be set to show 'local time' on the front panel, by adding a time offset. Daylight saving time changes can be accommodated by entering Spring Forward and Fall Back dates. It has a real time clock so that accurate date and time is available even after the unit is repowered without GPS access.

The built-in webserver, or front panel OLED display, can be used to configure the unit. The webserver is a responsive design meaning that it can be used with small screens on smartphones and tablets.

Front panel LEDs show the synchronisation status, GPS lock and the status of the AC and DC power supply inputs.

The brightness of the OLED display and LED indicators can be adjusted for low or high lighting conditions 4 general purpose outputs indicate critical states for the unit using a 9 way D-type connector mounted on the rear panel. Pull down when active pins are supplied for GPS lock status, external sync present, AC power present and DC power present.

The unit has a front panel power button and dual power connectors - an IEC mains input and a 12V DC input, which allows the AVN-GMCOS to be used for both studio and mobile installations. Moreover this allows for a secondary power source to reduce the effect of power down events. In any case, the unit monitors the status of both power sources and displays this on the front panel.

The unit can be put into a low-power sleep mode when not in use, with an instant start when power is re-applied. In power off situations, a super capacitor is used to keep the GPS receiver powered in a low power mode for more than 20 hours, enabling the receiver to regain lock immediately rather than having to 'cold' start.

- AES67/RAVENNA/AES-R16 compatible.
- GPS satellites received indicator.
- Master and slave sync modes.
- 8ns PTPv2 time stamping resolution.
- Holdover drift < 1ppm, with an option for < 0.01ppm.

- AES-3id, wordclock & variable PPS outputs.<
- Analogue master input.
- Dual power supply inputs.
- Front panel display.
- GPS sync and power supply present LED indicators.
- Real time clock for accurate date/time.
- Responsive design Ethernet webserver and front panel control & configuration.

Technical Specification For AVN-GMCOS

Timing Specification:	
Profile Support:	Default (RAVENNA), Media (AES67), AES-R16-2016 (SMPTE-ST 2059-2 & AES67 compatible), Custom profile
Timing Protocol:	PTPv2, IEEE1588-2008
Timing Accuracy:	PTP time stamping resolution 8ns
Holdover Drift: OCXO:	<900µs
Holdover Drift:	These figures are over 24 hours at constant temperature
GPS Performance:	50 channel GPS receiver
GPS Frequency:	1575.42MHz, L1 band
Clock Specification:	
Input Impedance:	75Ω
Output Impedance:	<75Ω
Antenna Impedance:	50Ω
Connections:	
Clocking Input:	BNC female
Clocking Outputs:	3 x BNC female Wordclock, AES-3id @ 32, 44.1, 48, 96, 176.4 & 192kHz Variable PPS (1, 10, 100, 1000) TTL
GPS Input:	SMA socket
GPIO:	D-Type female 9 way
Ethernet Port:	RJ45 socket, 100 BASE-T
Mains AC Input:	Universal filtered IEC socket, continuously rated 85-264 VAC @47-63Hz, max 10W
DC Input:	1 x 12V, KPJX-45 socket, positive pins 1 and 3
Maximum Operating Range (DC):	10.3V to 13.2V DC
Equipment Type:	
AVN-GMCOS GPS,	Grandmaster clock for PTP systems, IP, OCXO, 0.01ppm, rackmount
Physical Specification:	
Dimensions (Raw): (D)(1U)	4.4cm (H) x 48.3cm (W) x 17.8cm 1.8” (H) x 19” (W) x 7” (D) (1U)
Dimensions (Boxed):	6.8cm (H) x 58.8cm (W) x 27cm (D) 2.7” (H) x 23” (W) x 10.6” (D)
Weight:	Nett: 1.5kg Gross: 2.2kg Nett: 3.2lbs Gross: 4.8lbs
AVN-DC150	150W DC power supply with KPJX-4S plug

AVN-TB6 6 Button Talkback Intercom

Category: AES67 Talkback, IFB, Intercom & Commentary.

Product Function: Provides broadcast quality audio communication using RAVENNA/AES67 as the transport mechanism, allowing simple CAT 5 cabling and expansion.

Typical Applications: Ideal for broadcast quality audio communication between studios, offices and different areas in a facility or building complex.

Features:

- 6 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Dual 1Gb Ethernet & 1Gb SFP port.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Loudspeaker & Mic Mute buttons.
- Dual AC & DC power supply inputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- 10 user assignable GPIO ports.
- Responsive design Ethernet webserver.
- AVN-TB6RK 19” rack kit available.

AVN-TB6 Front View.

The AVN-TB6 is a 6 channel talkback intercom control unit from the Sonifex AVN range of IP based products. The AVN-TB6 is a freestanding version which can be rackmounted with the AVN-TB6RK 1U rack kit. There is also a desktop version, the AVN-TB6D which can be fitted flush into a work surface if required. Both models offers similar functionality.

This unit provides broadcast quality audio communication between studios, offices and different areas in a facility or building complex, using RAVENNA/AES67 as the transport mechanism, allowing simple CAT 5 cabling and expansion. RAVENNA (of which AES67 is a subset) allows for the distribution of audio across a network. The AVN range use RAVENNA as the communication method providing compatibility with other AES67 systems.

Each of the 6 channels on the AVN-TB6 can be configured to provide communications with other remote networked units, and an independently configurable 'page' function can contact selected units with priority over standard intercom calls if required.

There is a monitor channel that can route the audio from an AoIP source to the headphones and speaker. This could be used to take an IFB feed or an off-air transmission signal or simply to listen to any audio source.

A user configurable GPIO system, with 10 physical ports and 10 virtual ports, can be used to control operational functions on local or networked units, or drive outputs as selected states change, and a voltage free relay contact can be used to operate external equipment.

A built-in web server provides complete configuration control of the units and also allows for firmware updates and configuration backup. An Ember+ interface also gives access to the configuration options as well as providing remote control and monitoring of the GPIO and virtual GPIO ports.

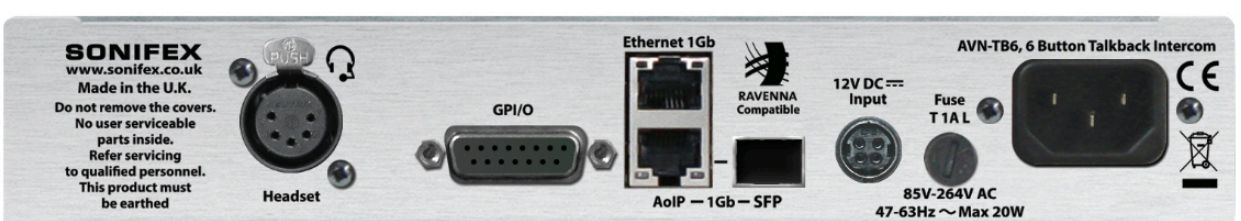
Webserver Software

The AVN-TB6 has a built-in webserver for setup and configuration. The webserver is responsive, and resizes depending on the size of your screen, meaning that it can be used on large monitors or small handheld devices such as smart-phones. Help information is shown on the right hand side of the screen so it's a good place to go to find out how the unit operates."

- 6 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Dual 1Gb Ethernet & 1Gb SFP port.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Loudspeaker & Mic Mute buttons.
- Dual AC & DC power supply inputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- 10 user assignable GPIO ports.
- Responsive design Ethernet webserver.
- AVN-TB6RK 19” rack kit available.

Technical Specification For AVN-TB6

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD)
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/Webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Transit Streams:	1 (fixed)
Sample Rate:	48 kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Microphone and Headset Input	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain
Headphone Output	
Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW
Loudspeaker	
Power Output:	4W
Volume:	Mute to full volume via front panel control
Connections	
Microphone:	XLR-3 pin female (electronically balanced)
Headphone:	¼ inch (6.35mm) stereo jack socket
Headset:	1 x XLR-5 pin female (electronically balanced input)
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45 1 x SFP
Mains AC Input:	Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W
DC Input:	4 pin 7.5A power jack socket, 10-14VDC
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm
Equipment Type	
AVN-TB6:	6 channel freestanding talkback intercom control unit with RAVENNA AoIP
Physical Specification	
Dimensions: (Raw)	25.5cm (W) x 19.4cm (D) x 4.4cm (H) (1U) 10.1" (W) x 7.7" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	33cm (W) x 26cm (D) x 17cm (H) 13" (W) x 10.3" (D) x 6.7" (H)
Weight:	Nett: 2kg Gross: 3kg Nett: 4.4lbs Gross: 6.6lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-TB6RK	1U Rack Kit
CM-GM2:	Professional Gooseneck Condenser Microphone



AVN-TB6 Rear View.

AVN-TB6D 6 Button Talkback Intercom

Category: AES67 Talkback, IFB, Intercom & Commentary.

Product Function: Provides broadcast quality audio communication using RAVENNA/AES67 as the transport mechanism, allowing simple CAT 5 cabling and expansion.

Typical Applications: Ideal for broadcast quality audio communication between studios, offices and different areas in a facility or building complex.

Features:

- 6 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Dual 1Gb Ethernet & 1Gb SFP port.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Loudspeaker & Mic Mute buttons.
- Dual AC & DC power supply inputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- 10 user assignable GPIO ports.
- Responsive design Ethernet webserver.

AVN-TB6D Front View.

The AVN-TB6D is a 6 channel talkback intercom control unit from the Sonifex AVN range of IP based products. The AVN-TB6D is a desktop version which can be fitted flush into a work surface if required. There is also a freestanding version which can be rackmounted with the AVN-TB6RK 1U rack kit. Both models offers similar functionality.

This unit provides broadcast quality audio communication between studios, offices and different areas in a facility or building complex, using RAVENNA/AES67 as the transport mechanism, allowing simple CAT 5 cabling and expansion. RAVENNA (of which AES67 is a subset) allows for the distribution of audio across a network. The AVN range use RAVENNA as the communication method providing compatibility with other AES67 systems.

Each of the 6 channels on the AVN-TB6D can be configured to provide communications with other remote networked units, and an independently configurable 'page' function can contact selected units with priority over standard intercom calls if required.

There is a monitor channel that can route the audio from an AoIP source to the headphones and speaker. This could be used to take an IFB feed or an off-air transmission signal or simply to listen to any audio source.

A user configurable GPIO system, with 10 physical ports and 10 virtual ports, can be used to control operational functions on local or networked units, or drive outputs as selected states change, and a voltage free relay contact can be used to operate external equipment.

A built-in web server provides complete configuration control of the units and also allows for firmware updates and configuration backup. An Ember+ interface also gives access to the configuration options as well as providing remote control and monitoring of the GPIO and virtual GPIO ports.

Webserver Software

The AVN-TB6D has a built-in webserver for setup and configuration. The webserver is responsive, and resizes depending on the size of your screen, meaning that it can be used on large monitors or small handheld devices such as smart-phones. Help information is shown on the right hand side of the screen so it's a good place to go to find out how the unit operates.

- 6 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Dual 1Gb Ethernet & 1Gb SFP port.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Loudspeaker & Mic Mute buttons.
- Dual AC & DC power supply inputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- 10 user assignable GPIO ports.
- Responsive design Ethernet webserver.

Technical Specification For AVN-TB6D

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD)
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/Webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Transit Streams:	1 (fixed)
Sample Rate:	48 kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Microphone and Headset Input	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain
Headphone Output	
Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW
Loudspeaker	
Power Output:	4W
Volume:	Mute to full volume via front panel control

Connections	
Microphone:	XLR-3 pin female (electronically balanced)
Headphone:	¼ inch (6.35mm) stereo jack socket
Headset:	1 x XLR-5 pin female (electronically balanced input)
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45 1 x SFP
Mains AC Input:	Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W
DC Input:	4 pin 7.5A power jack socket, 10-14VDC
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm
Equipment Type	
AVN-TB6D:	6 channel desktop talkback intercom control unit with RAVENNA AoIP

Physical Specification	
Dimensions: (Raw) (H) (1U)	22.2cm (W) x 17cm (D) x 11.5cm 8.7" (W) x 6.7"(D) x 4.5" (H) (1U)
Dimensions (Boxed):	41cm (W) x 31cm D) x 23cm (H) 16" (W) x 12" (D) x 9" (H)
Weight:	Nett: 1.8kg Gross:2.6kg Nett: 4.0llbs Gross:5.7llbs
Accessories	
AVN-DC060: range	60W DC power supply for AVN with KPJX-4S plug

AVN-TB6D Rear View.

AVN-TB10AR 10 Button Advanced Talkback Intercom, AoIP Portal

AVN-TB10AR Front View.

Category: AES67 Talkback, IFB, Intercom & Commentary.

Product Function: An advanced talkback/listening/paging intercom unit to enable voice/audio communication between different areas in a facility or building complex.

Typical Applications: OB truck comms, theatre comms, inter-studio comms in a TV or radio station.

Features:

- 10 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Sources from AoIP, balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- Dual AC & DC power supply inputs.
- Front panel display providing source & destination information.
- Ethernet webserver and front panel control & configuration.
- Speaker & microphone mute buttons.
- Callback button with source display.

The AVN-TB products are IP audio based talkback intercom units with an advanced feature set, allowing them to be used in multiple applications. With both 4-wire analogue inputs and outputs, as well as AoIP network audio connectivity, the AVN-TB units can be used with existing legacy 4-wire systems and with new AES67 AoIP networked audio infrastructure. Additionally, all audio is at 48kHz sample rate, meaning that it's broadcast quality audio as standard.

RAVENNA (of which AES67 is a subset) allows for the distribution of audio across a network. The AVN range use RAVENNA as the communication method providing compatibility with other AES67 systems.

The AVN-TB10AR is a 10 button intercom meaning that 10 other 'stations' can be defined, one per button, for communication. Comms can be made as a Talk action, a Listen action or a duplex Talk/Listen action to/from each station. Coloured LEDs in the buttons help to show which action is being used and there is also a Callback button for when you're unavailable to receive a call.

The stations can be from anywhere on the AoIP network and the use of Bonjour Device Discovery means that other stations can be found quickly and sometimes automatically.

The Page button is used to speak to all stations (or a defined list of stations) and Group Talk functions can be enabled to page particular groups of stations.

Two monitor buttons allow for routing audio directly to the speaker e.g. to take an IFB feed or an off-air transmission signal. Signals can be ducked or mixed when a talkback input is received to the speakers or headphones.

Three user defined buttons can be programmed for different functions, such as for Group Talk.

The speaker mutes automatically when headphones are inserted and the volume level of headphones, speaker and incoming sources can all be controlled with one front ▶

AVN-TB10AR Rear View.

panel rotary encoder volume control knob, which shows the level using RGB LEDs around the outside of the knob.

Advanced acoustic echo cancellation & built-in microphone AGC (automatic gain control) ensure that there's no acoustic feedback between microphone and speaker.

Buttons are available for microphone mute (cough) and speaker mute actions and these can be controlled remotely by GPI or network commands.

Each unit has a built-in webserver which is where the majority of settings and configurations are made. The front panel OLED display can also be used to configure the unit, although more functionality is available by using the webserver. The webserver is a responsive design meaning that it can be used with small screens on smartphones and tablets.

The unit can act as a PTP masterclock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

Front panel LEDs show the AoIP network status, synchronisation status, whether AGC is being used and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be adjusted for low or high lighting conditions.

The unit has a front panel power button and dual power connectors - an IEC mains input and a 12V DC input, which allows the AVN-TB10AR to be used for both studio and mobile installations. Also, a secondary power source reduces the effect of power down events. In any case, the unit monitors the status of both power sources and displays this on the front panel.

10 GPIOs (general purpose inputs/outputs) and a programmable relay output can be configured to indicate critical states for the unit, via the 15 way D-type connector, for example, to show loss of DC power, or to show a button press action.

Webserver Software

Each AVN-TB talkback unit has a built-in webserver for setup and configuration. The webserver is responsive, and resizes depending on the size of your screen, meaning that it can be used on large monitors or small handheld devices such as smart-phones. Help information is shown on the right hand side of the screen so it's a good place to go to find out how the unit operates.

- 10 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Sources from AoIP, balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- Dual AC & DC power supply inputs.
- Front panel display providing source & destination information.
- Ethernet webserver and front panel control & configuration.
- Speaker & microphone mute buttons.
- Callback button with source display.

Technical Specification For AVN-TB10AR

Audio-Over-IP Specification

Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD)
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Maximum Streams:	RX 6, TX 5 (fixed)
Sample Rate:	48 kHz

Timing Synchronisation

Profile Support:	Default, media & custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Technical Specification

Microphone and Headset Input

Input Impedance:	>2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain

Unbalanced Line Inputs

Input Impedance:	>20kΩ
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-97dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-97dB

Balanced Line Inputs

Input Impedance:	>20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100dB
Common Mode Rejection:	>70dB @ 1kHz

Headphone Output

Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW

Unbalanced Line Outputs

Output Impedance:	<50Ω
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-95dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW

Balanced Line Outputs

Output Impedance:	<50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

Loudspeaker

Power Output:	4W
Volume:	Mute to full volume via front panel control

Connections

Microphone:	XLR-3 pin female (electronically balanced)
Headphone:	¼ inch (6.35mm) stereo jack socket
Headset:	2 x XLR-5 pin female (front/rear electronically balanced input)
Audio Inputs:	2 x unbalanced stereo, RCA phono
Audio Inputs:	1 x S/PDIF, RCA phono
Audio Outputs:	1 x balanced stereo, RJ45
Audio Outputs:	1 x unbalanced stereo, 2 x RCA phono
Audio Outputs:	1 x loudspeaker output
Audio Input/Output:	1 x balanced stereo input or mono input/output, RJ45
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45
Network:	1 x SFP fibre
Mains AC Input:	Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W
DC Input:	4 pin 7.5A power jack socket, 9.5-14VDC
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm

Equipment Type

AVN-TB10AR	10 channel rackmount talkback intercom control unit with RAVENNA AoIP

Physical Specification

Dimensions: (Raw)	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	59cm (W) x 28cm (D) x 11cm (H) 23" (W) x 11" (D) x 4.3" (H)
Weight:	Nett: 2.4kg Gross: 3.1kg Nett: 5.3lbs Gross: 6.8lbs

Accessories

AVN-DC150	150W DC power supply with KPJX-4S plug
CM-GM2:	Professional Gooseneck Condenser Microphone

AVN-TB20AR 20 Button Advanced Talkback Intercom, AoIP Portal

AVN-TB20AR Front View.

Category: AES67 Talkback, IFB, Intercom & Commentary.

Product Function: An advanced talkback/listening/paging intercom unit to enable voice/audio communication between different areas in a facility or building complex.

Typical Applications: OB truck comms, theatre comms, inter-studio comms in a TV or radio station.

Features:

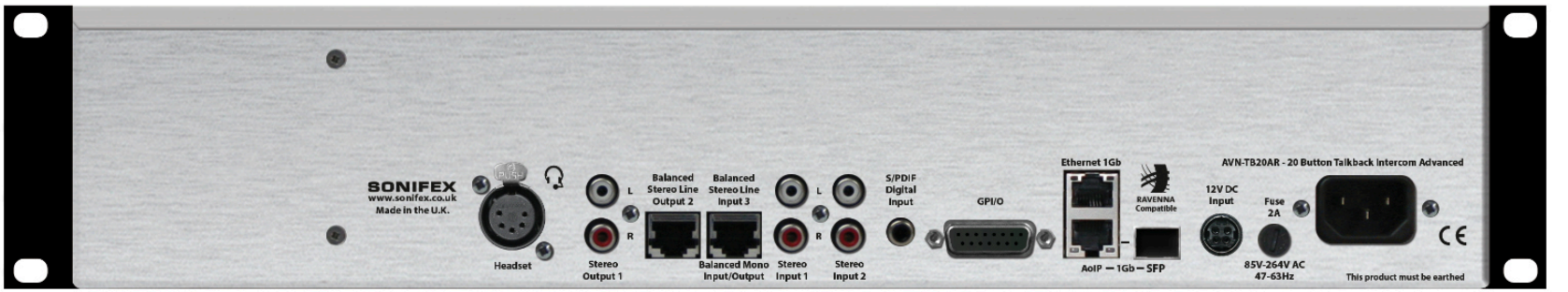
- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Phone button for remote dialling and control of an external telephone hybrid.
- Page button and Group Talk facilities.
- Callback button with callback source display.
- Three user definable buttons.
- Speaker & microphone mute buttons.
- Mic & headset inputs (front & rear panel headset connection), headphone & speaker outputs.
- Front panel volume control which operates on speaker/headphone outputs and incoming source levels.
- +48V phantom power for the mic inputs.
- Ethernet webserver and front panel control & configuration.
- Front panel display providing source & destination information.
- Sources from AoIP, 1 x balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- GPI/O button for triggering external events, via physical GPIO or network commands.
- Front panel LEDs for network audio presence, Talk activity, AGC activity, clock sync and power supply activity.
- Two front panel monitor buttons for routing audio directly to the speaker e.g. to take an IFB feed or off-air transmission signal.
- Ducking or mixing of inputs to speaker/headphones.
- Dual AC & DC power supply inputs.

The AVN-TB20AR is a 2U rackmount 20 button version of the AVN-TB10AR with the same specification, but more station buttons allowing greater communication for larger facilities. It also has 'GPI/O' and 'Phone' buttons for remote control of external equipment and Sonifex DHY-04 telephone hybrids, useful in an OB truck or production gallery.

"The AVN-TB20AD is a desktop version of the rackmount AVN-TB20AR intercom with a smaller form factor and an elegant sloped front. It has the same feature set and connectivity *.

** Except for an additional headset connector on the rear of the AVN-TB20AR.*

- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.



AVN-TB20AR Front View.

- Phone button for remote dialling and control of an external telephone hybrid.
- Page button and Group Talk facilities.
- Callback button with callback source display.
- Three user definable buttons.
- Speaker & microphone mute buttons.
- Mic & headset inputs (front & rear panel headset connection), headphone & speaker outputs.
- Front panel volume control which operates on speaker/headphone outputs and incoming source levels.
- +48V phantom power for the mic inputs.
- Ethernet webserver and front panel control & configuration.
- Front panel display providing source & destination information.
- Sources from AoIP, 1 x balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- GPI/O button for triggering external events, via physical GPIO or network commands.
- Front panel LEDs for network audio presence, Talk activity, AGC activity, clock sync and power supply activity.
- Two front panel monitor buttons for routing audio directly to the speaker e.g. to take an IFB feed or off-air transmission signal.
- Ducking or mixing of inputs to speaker/ headphones.
- Dual AC & DC power supply inputs.

Technical Specification For AVN-TB20AR

Audio-Over-IP Specification

Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD)
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Maximum Streams:	RX 6, TX 5 (fixed)
Sample Rate:	48 kHz

Timing Synchronisation

Profile Support:	Default, media & custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Technical Specification

Microphone and Headset Input

Input Impedance:	>2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain

Unbalanced Line Inputs

Input Impedance:	>20kΩ
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-97dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-97dB

Balanced Line Inputs

Input Impedance:	>20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100dB
Common Mode Rejection:	>70dB @ 1kHz

Headphone Output

Output Impedance: headphones	Drives 150mW into 32Ω to 600Ω
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW

Unbalanced Line Outputs

Output Impedance:	<50Ω
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-95dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW

Balanced Line Outputs

Output Impedance:	<50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

Loudspeaker

Power Output:	4W
Volume:	Mute to full volume via front panel control

Connections

Microphone:	XLR-3 pin female (electronically balanced)
Headphone:	¼ inch (6.35mm) stereo jack socket
Headset:	2 x XLR-5 pin female (front/rear electronically balanced input)
Audio Inputs:	2 x unbalanced stereo, RCA phono
Audio Inputs:	1 x S/PDIF, RCA phono
Audio Outputs:	1 x balanced stereo, RJ45
Audio Outputs:	1 x unbalanced stereo, 2 x RCA phono
Audio Outputs:	1 x loudspeaker output
Audio Input/Output:	1 x balanced stereo input or mono input/output, RJ45
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45
Network:	1 x SFP fibre
Mains AC Input:	Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W
DC Input:	4 pin 7.5A power jack socket, 9.5-14VDC
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm

Equipment Type

AVN-TB20AR	20 channel rackmount talkback intercom control unit with RAVENNA AoIP

Physical Specification

Dimensions: (Raw)	48.3cm (W) x 17.5cm (D) x 8.8cm (H)(2U) 19" (W) x 6.9" (D) x 3.6" (H) (2U)
Dimensions (Boxed):	59cm (W) x 28cm (D) x 15cm (H) 23" (W) x 11" (D) x 5.9" (H)
Weight:	Nett: 3.7kg Gross: 4.4kg Nett: 8.1lbs Gross: 9.7lbs

Accessories

AVN-DC150	150W DC power supply with KPJX-4S plug
CM-GM2:	Professional Gooseneck Condenser Microphone

AVN-TB20AD 20 Button Advanced Talkback Intercom, AoIP Desktop Portal

Category: AES67 Talkback, IFB, Intercom & Commentary.

Product Function: An advanced talkback/ listening/paging intercom unit to enable voice/audio communication between different areas in a facility or building complex.

Typical Applications: OB truck comms, theatre comms, inter-studio comms in a TV or radio station.

Features:

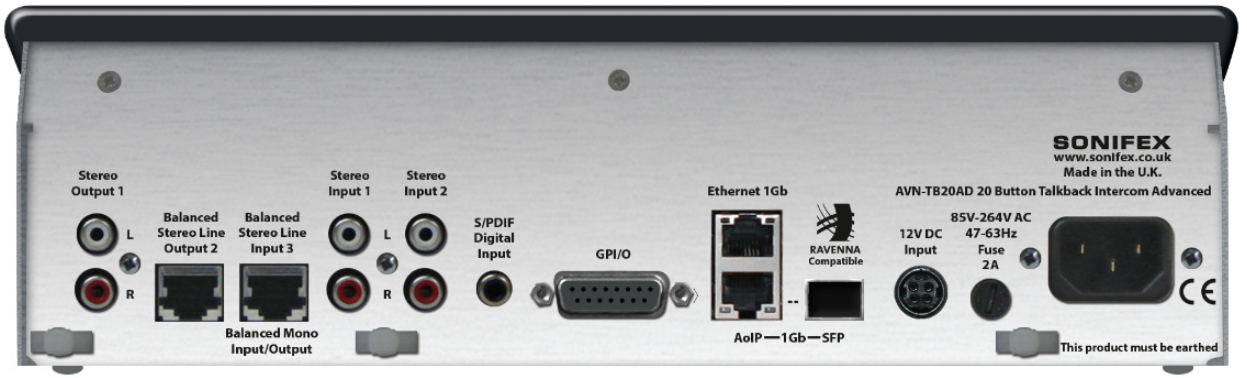
- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Phone button for remote dialling and control of an external telephone hybrid.
- Page button and Group Talk facilities.
- Callback button with callback source display.
- Three user definable buttons.
- Speaker & microphone mute buttons.
- Mic & headset inputs, headphone & speaker outputs.
- Front panel volume control which operates on speaker/headphone ` outputs and incoming source levels.
- +48V phantom power for the mic inputs.
- Ethernet webserver and front panel control & configuration.
- Front panel display providing source & destination information.
- Sources from AoIP, 1 x balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.<
- GPIO button for triggering external events, via physical GPIO or network commands.
- Front panel LEDs for network audio presence, Talk activity, AGC activity, clock sync and power supply activity.
- Two front panel monitor buttons for routing audio directly to the speaker e.g. to take an IFB feed or off-air transmission signal.
- Ducking or mixing of inputs to speaker/ headphones.
- Dual AC & DC power supply inputs.

AVN-TB20AD Front View.

The AVN-TB20AD is a desktop version of the rackmount AVN-TB20AR intercom with a smaller form factor and an elegant sloped front. It has the same feature set and connectivity *.

* Except for an additional headset connector on the rear of the AVN-TB20AR.

- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Phone button for remote dialling and control of an external telephone hybrid.
- Page button and Group Talk facilities.
- Callback button with callback source display.
- Three user definable buttons.
- Speaker & microphone mute buttons.
- Mic & headset inputs, headphone & speaker outputs.
- Front panel volume control which operates on speaker/headphone outputs and incoming source levels.
- +48V phantom power for the mic inputs.



AVN-TB20AD Rear View.

- Ethernet webserver and front panel control & configuration.
- Front panel display providing source & destination information.
- Sources from AoIP, 1 x balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.<
- GPIO button for triggering external events, via physical GPIO or network commands.
- Front panel LEDs for network audio presence, Talk activity, AGC activity, clock sync and power supply activity.
- Two front panel monitor buttons for routing audio directly to the speaker e.g. to take an IFB feed or off-air transmission signal.
- Ducking or mixing of inputs to speaker/ headphones.
- Dual AC & DC power supply inputs.

Technical Specification For AVN-TB20AD

Audio-Over-IP Specification

Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD)
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Maximum Streams:	RX 6, TX 5 (fixed)
Sample Rate:	48 kHz

Timing Synchronisation

Profile Support:	Default, media & custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Technical Specification

Microphone and Headset Input

Input Impedance:	>2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain
Input Impedance:	>20kΩ

Unbalanced Line Inputs

0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-97dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-97dB

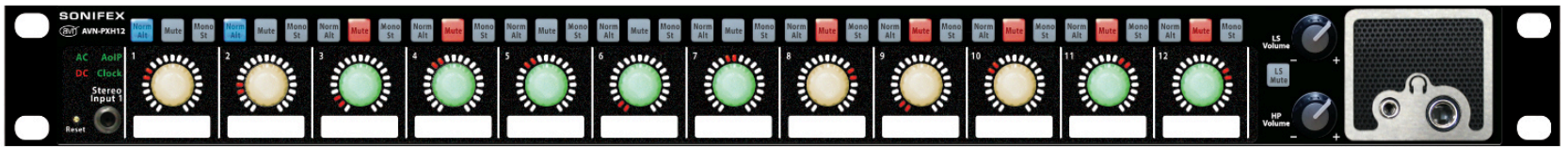
Balanced Line Inputs

Input Impedance:	>20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW

Technical Specification For AVN-TB20AD continued...

Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100dB
Common Mode Rejection:	>70dB @ 1kHz
Headphone Output	
Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW
Unbalanced Line Outputs	
Output Impedance:	<50Ω
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-95dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW
Balanced Line Outputs	
Output Impedance:	<50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Loudspeaker	
Power Output:	4W
Volume:	Mute to full volume via front panel control
Connections	
Microphone:	XLR-3 pin female (electronically balanced)
Headphone:	¼ inch (6.35mm) stereo jack socket
Headset:	1 x XLR-5 pin female (front electronically balanced input)
Audio Inputs:	2 x unbalanced stereo, RCA phono
Audio Inputs:	1 x S/PDIF, RCA phono
Audio Outputs:	1 x balanced stereo, RJ45
Audio Outputs:	1 x unbalanced stereo, 2 x RCA phono
Audio Outputs:	1 x loudspeaker output
Audio Input/Output:	1 x balanced stereo input or mono input/output, RJ45
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45
Network:	1 x SFP fibre
Mains AC Input:	Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W
DC Input:	4 pin 7.5A power jack socket, 9.5-14VDC
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm
Equipment Type	
AVN-TB20AD	20 channel desktop talkback intercom control unit with RAVENNA AoIP
Physical Specification	
Dimensions: (Raw)	29.4cm (W) x 16.5cm (D) x 8.5cm (H) 11.6" (W) x 6.5" (D) x 3.3" (H)
Dimensions (Boxed):	40cm (W) x 28cm (D) x 15cm (H) 16" (W) x 11" (D) x 5.9" (H)
Weight:	Nett: 2.5kg Gross: 3.2kg Nett: 5.5lbs Gross: 7.0lbs
Accessories	
AVN-DC150	150W DC power supply with KPJX-4S plug
CM-GM2:	Professional Gooseneck Condenser Microphone

AVN-PXH12 12 x 2 Channel Mix Monitor, AoIP Portal



AVN-PXH12 Front View.

The Sonifex AVN-PXH12 is a monitor-mixer primarily designed for monitoring Audio Over IP audio channels in a compact 1U format rack. Any AES67 audio channels on your network can be assigned to input channels on the unit and mixed down to analogue outputs, a headphone output and built-in speaker.

It is a stereo monitoring device that allows you to monitor up to 12 audio sources, from an input total of 24, at any one time. The 24 audio sources can be selected from 4 discrete stereo analogue audio inputs (1 x front panel 3.5mm jack socket, 2 x rear panel 3.5mm jack sockets and 1 x rear panel stereo XLR input pair) or from any RAVENNA, AES67 or AES67-enabled Dante® AoIP connected streams.

These stereo signals are routed to the 12 x control channels on the front panel, each of which have a 'Normal' and an 'Alternate' input selection. Each channel has three buttons: one for input selection, another to Mute the channel and the third to select whether the channel input is routed to the left, right or stereo output legs.

The knob for each channel controls the level of the input routed to the output and the knob also illuminates either green, amber or red to show input level. Pressing the knob 'Solos' the channel input to the output.

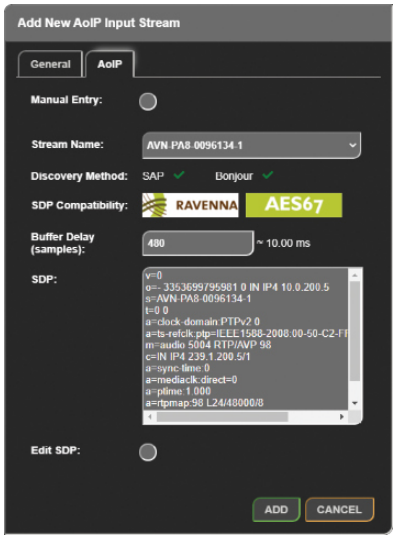
Category: AES67 Mix Monitor

Product Function: A mix monitor for monitoring 2 x 12 AES67 Audio Over IP audio channels in a compact 1U format rack to analogue outputs, a headphone output and built-in speaker.

Typical Applications: Production gallery or control room channel monitoring positions, OB truck monitoring, confidence channel monitoring of any AoIP installation.

Features:

- AES67 as a format has been established, providing compatibility with most other AoIP products – the unit uses RAVENNA audio to ensure AES67 compatibility.
- A built-in web server is used for all configuration. Sources for all channels are simply assigned on one webpage and can be freely selected and altered at will.
- SAP can be used as a discovery mechanism to discover Dante devices and monitor Dante® AES67 streams. Dante® is a trademark of Audinate Pty Ltd.
- Confidence monitoring on the translucent volume knob for each channel so you've got ' at-a-glance' monitoring available.
- The front panel Mute button and the Solo feature on the control knob allow a single channel, or a handful of, channels to be auditioned quickly.
- For each channel, 'Normal' and 'Alternate' inputs can be switched quickly (with <1msec accuracy) for direct comparison.
- Each channel can be directed to headphone left ear, right ear, or a stereo mix, providing you with the ability to set up a familiar headphone mix.
- 6.35mm (1/4") & 3.5mm headphone outputs and a speaker output with separate LS & HP volume controls.
- Sources from AoIP, balanced or 3 x unbalanced inputs.
- Destinations to AoIP or rear panel balanced outputs.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- Dual AC & DC power supply inputs.
- Ethernet webserver control & configuration.
- Speaker mute button.



AVN-PXH12 GUI - AoIP Tab of Add New AoIP Stream Window.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs.

The front panel has 3 outputs: paralleled stereo headphones on 6.35mm (¼") jack and 3.5mm jack sockets, each with their own individual attenuation settings, and a mono-mix speaker output. There are discrete volume controls for the headphones and the speaker, and the latter also has a mute button.

The rear panel has an additional 3 line level XLR-3 audio outputs, which can be designated as mono mix or left or right channel outputs of the mixed audio content (similar to the speaker and headphone outputs respectively), or any of the physical inputs or AoIP input sources.

The unit also sends to the network, as AoIP AES67 streams, the 8 channels of the 4 physical stereo inputs, together with a stereo mix of the speaker output.

The unit can act as a PTP masterclock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

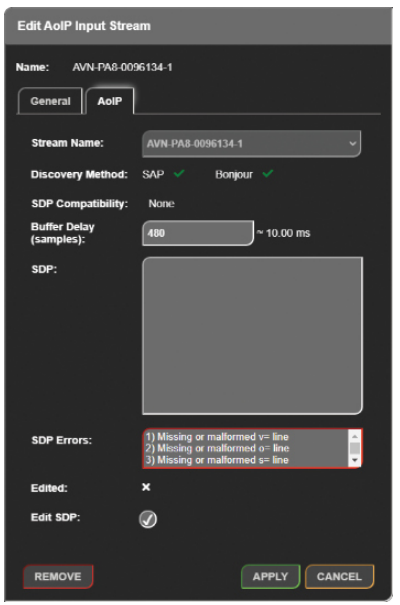
The rear panel contains IEC mains and secondary DC power inputs which provide power redundancy to the product. There are two Ethernet RJ45 connections (control and AoIP) and there is an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software controlled functionality. A voltage free relay contact can be used to operate external equipment.

A built-in web server provides complete configuration control of the unit including source assignment to each channel and also allows for firmware updates and configuration backup. The unit can be controlled by suitable Ember+ commands.

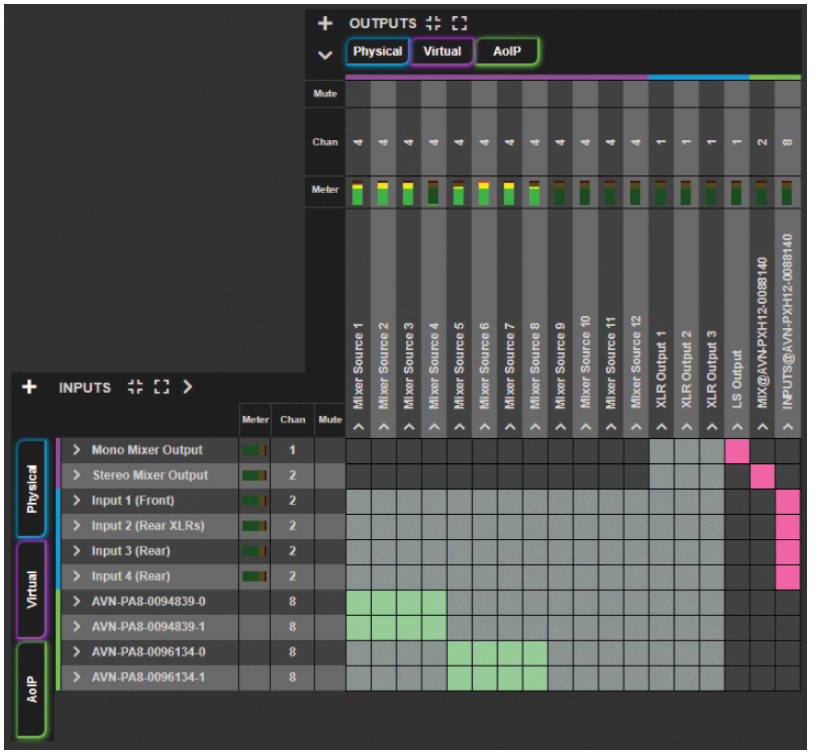
Features:

- AES67 as a format has been established, providing compatibility with most other AoIP products – the unit uses RAVENNA audio to ensure AES67 compatibility.
- A built-in web server is used for all configuration. Sources for all channels are simply assigned on one webpage and can be freely selected and altered at will.
- SAP can be used as a discovery mechanism to discover Dante devices and monitor Dante® AES67 streams. Dante® is a trademark of Audinate Pty Ltd.
- Confidence monitoring on the translucent volume knob for each channel so you've got 'at-a-glance' monitoring available.
- The front panel Mute button and the Solo feature on the control knob allow a single channel, or a handful of, channels to be auditioned quickly.
- For each channel, 'Normal' and 'Alternate' inputs can be switched quickly (with <1msec accuracy) for direct comparison.



AVN-PXH12 GUI - Edit AoIP Input Stream Window AoIP Tab SDP Errors Field.

AVN-PXH12 Front View.



AVN-PXH12 GUI - The Audio Routing Grid on the 'Audio Routing' Web Page.

- Each channel can be directed to headphone left ear, right ear, or a stereo mix, providing you with the ability to set up a familiar headphone mix.
- 6.35mm (1/4") & 3.5mm headphone outputs and a speaker output with separate LS & HP volume controls.
- Sources from AoIP, balanced or 3 x unbalanced inputs.
- Destinations to AoIP or rear panel balanced outputs.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports.
- Dual AC & DC power supply inputs.
- Ethernet webserver control & configuration.
- Speaker mute button.

Webserver Software

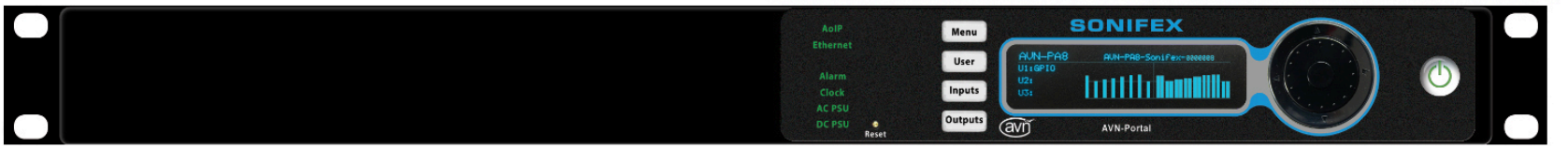
The mix monitor has a built-in webserver for setup and configuration.

The webserver is responsive, and resizes depending on the size of your screen, meaning that it can be used on large monitors or small handheld devices such as smart-phones. Help information is shown on the right hand side of the screen so it's a good place to go to find out how the unit operates.

Technical Specification For AVN-PXH12

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) & SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Ember+/webserver
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Maximum of 8
Frames Per Packet:	48
Maximum Streams:	RX 15, TX 2
Sample Rate:	48 kHz
Timing Synchronisation	
Profile Support:	Default, media & custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Technical Specification	
Unbalanced Line Inputs	
Input Impedance:	>20kΩ
0dBFS Line-Up:	+12dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-97dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-100dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-97dB
Balanced Line Inputs	
Input Impedance:	>20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100dB
Common Mode Rejection:	>70dB @ 1kHz
Headphone Output	
Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
0dBFS Line-Up:	+20dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-110dBFS, 20kHz BW
Balanced Line Outputs	
Output Impedance:	<50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Loudspeaker	
Power Output:	4W
Volume:	Mute to full volume via front panel control
Connections	
Headphone:	¼ inch (6.35mm) stereo jack socket
Headphone:	3.5mm stereo jack socket
Audio Inputs:	3 x unbalanced 3.5mm stereo jack socket
Audio Inputs:	2 x balanced XLR-3 female socket
Audio Outputs:	3 x balanced XLR-3 male plug
Audio Outputs:	1 x loudspeaker output
GPIO:	15-way 'D'-type socket
Network:	2 x gigabit Ethernet, RJ45
Network:	1 x SFP fibre
Equipment Type	
AVN-PXH12:	2 x 12 Channel mix monitor, AoIP portal
Physical Specification	
Dimensions: (Raw)	48.3cm (W) x 20.12cm (D) x 4.4cm (H)(1U) 19" (W) x 7.92" (D) x 1.8" (H) (1U)
Dimensions (Boxed): (H)	55.2cm (W) x 29.5cm (D) x 16.5cm 21.7" (W) x 11.6" (D) x 6.5" (H)
Weight:	Nett: 2.9kg Gross: 3.5kg Nett: 6.38lbs Gross: 7.7lbs
Accessories	
AVN-DC150:	150W DC power supply with KPJX-4S plug

AVN-PA8 8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs, AES67 Portal



AVN-PA8 Front View.

The AVN-PA8 is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight stereo analogue line inputs, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PA8 becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PA8 unit is supplied with the standard OLED front panel display, which provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, version

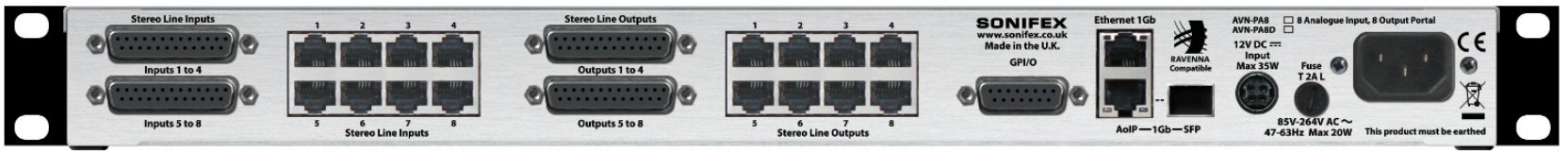
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight stereo analogue inputs and eight stereo analague outputs on D-type sockets with AES59 pinout, paralleled with eight RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.<
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PA8 Rear View.

information, and basic metering (The **'D'** version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

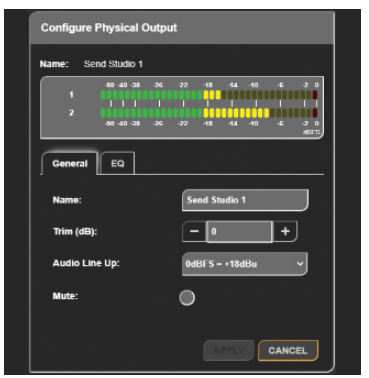
For physical audio connections, the AVN-PA8 uses D-type sockets with AES59 analogue pinout, paralleled with eight RJ45 connectors using the StudioHub® pinout.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

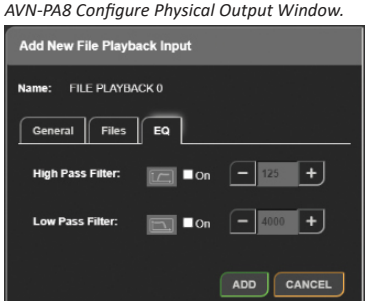
A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage **AVN-HA1** headphone amplifiers to provide a headphone distribution system. Output connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HA1 can be used as another GPI for muting the output.

The AVN-PA8 is part of the Sonifex **AVN-Portal** series, which includes the **AVN-PA8** (analogue physical I/O), the **AVN-PD8** (digital physical I/O), and the **AVN-PM8** (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The **AVN-PA8D** features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the **AVN-PA8T** features rear panel terminal block connectors for all physical inputs and outputs; the **AVN-PA8TD** features both the TFT displays and terminal block connections.

- Eight stereo analogue inputs and eight stereo analague outputs on D-type sockets with AES59 pinout, paralleled with eight RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and



AVN-PA8 Configure Physical Output Window.



AVN-PA8 Add New File Playback Input Window.

1Gb SFP ports.

- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

Technical Specification For AVN-PA8

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48 kHz

Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

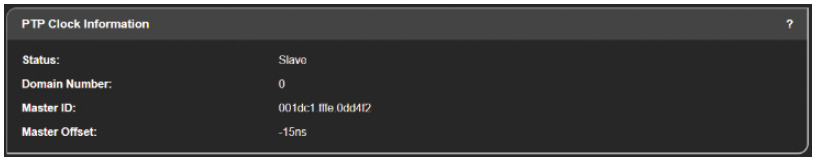
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N: 20kHz,	< -110dBFS, -30dBFS, 20Hz to 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB
Common Mode Rejection:	> 70dB @ 1kHz

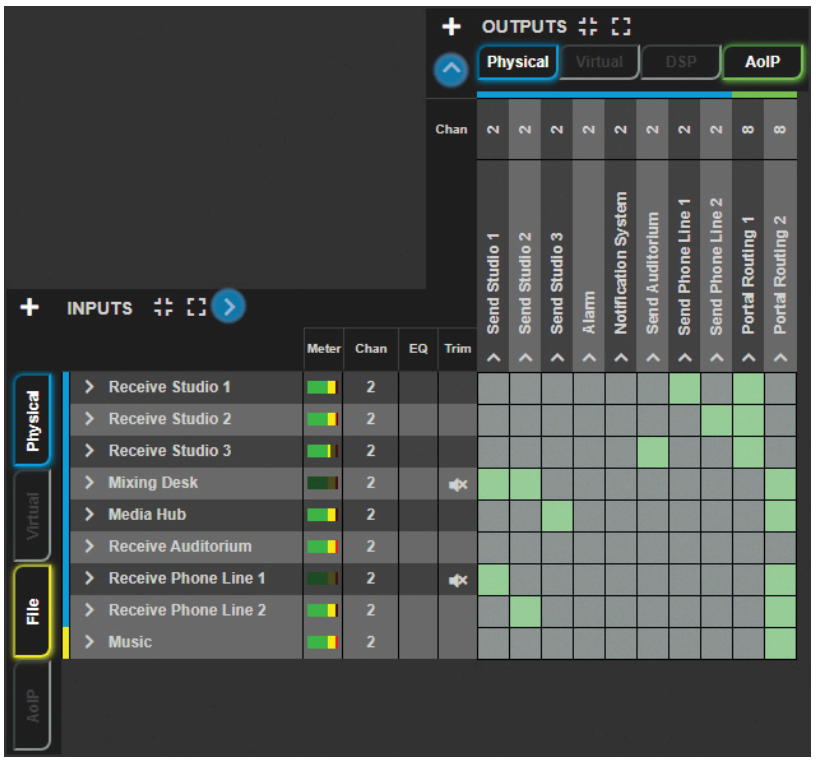
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

Connections	
Inputs:	2 x D-Sub (DB-25) connections (TASCAM AES-59 analogue pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
Outputs:	2 x D-Sub (DB-25) connections (TASCAM AES-59 analogue pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
GPIO:	1 x D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10 14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PA8:	8 Stereo analogue line inputs & 8 stereo analogue line outputs, AES67 portal
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 3.12 kg Gross:4.16 kg Nett: 6.86 lbs Gross:9.15 lbs

Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.

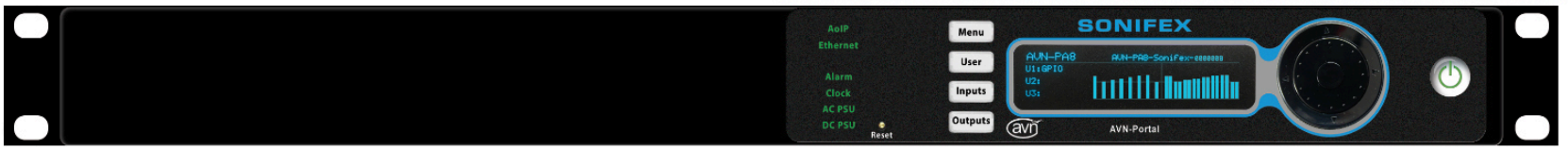


AVN-PA8 PTP Clock Information.



AVN-PA8 Show or Hide Status Icons

AVN-PA8T 8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs on Terminal Blocks, AES67 Portal



AVN-PA8T Front View.

The AVN-PA8T is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight stereo analogue line inputs, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding. Physical audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PA8T becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PA8T unit is supplied with the standard OLED front panel display, which provides detailed status information on ▶

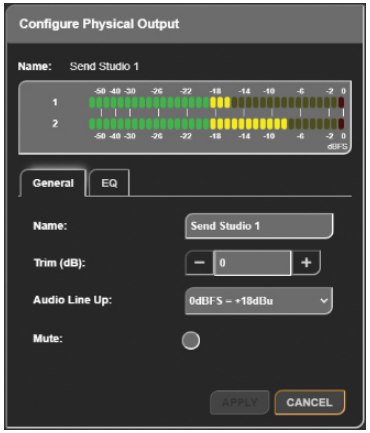
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight stereo analogue inputs and eight stereo analague outputs on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).



AVN-PA8T Configure Physical Output Window.

AVN-PA8T Add New File Playback Input Window.

device name, network addresses, PTP clocking info, power status/voltages, version information, and basic metering (The 'D' version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections and GPIO the AVN-PA8 uses terminal block connections.

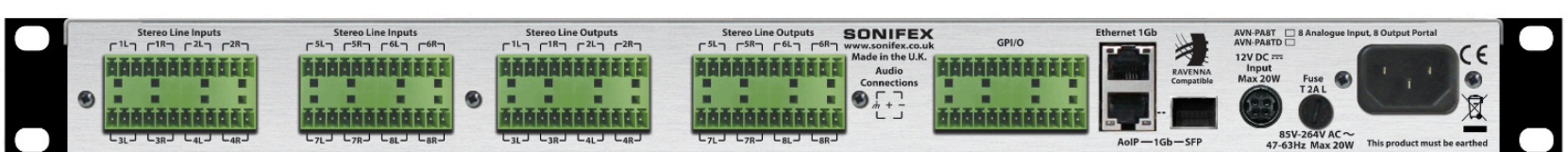
There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

The rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

The AVN-PA8T is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo analogue inputs and eight stereo analague outputs on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).



AVN-PA8T Rear View.

Technical Specification For AVN-PA8T

Audio-Over-IP Specification

Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48 kHz

Ember+ Interface Connection

Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

Timing Synchronisation

Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Balanced Line Inputs

Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB
Common Mode Rejection:	> 70dB @ 1kHz
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

Connections

Inputs:	2 x 24-Pin Phoenix style terminal blocks (Analogue pinout).
Outputs:	2 x 24-Pin Phoenix style terminal blocks. (Analogue pinout).
GPIO:	1 x 24-Pin Phoenix style terminal blocks.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input: 14VDC.	4-pin 7.5A power jack socket, 10-
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.

Equipment Type

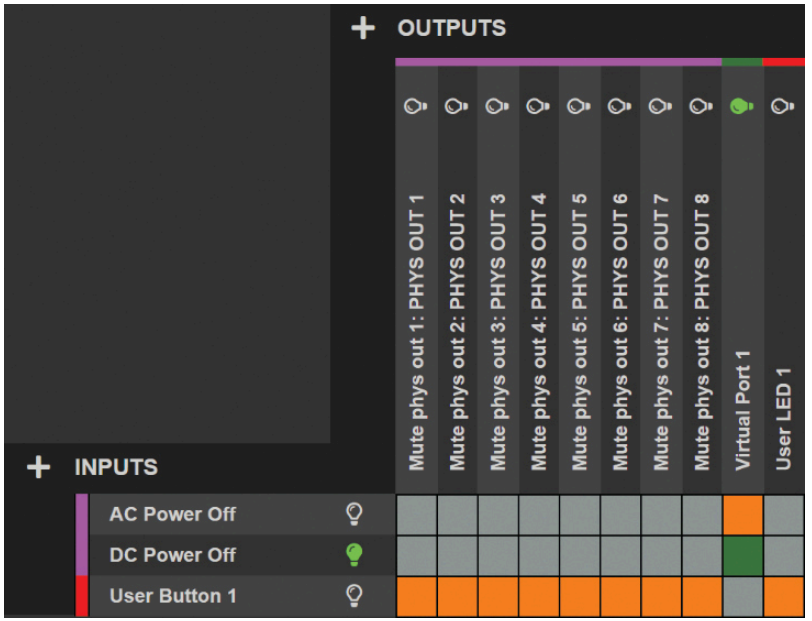
AVN-PA8T:	8 Stereo analogue line inputs & 8 stereo analogue line outputs on terminal blocks, AES67 portal

Physical Specification

Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.96 kg Gross: 4.00kg Nett: 6.51lbs Gross: 8.80lbs

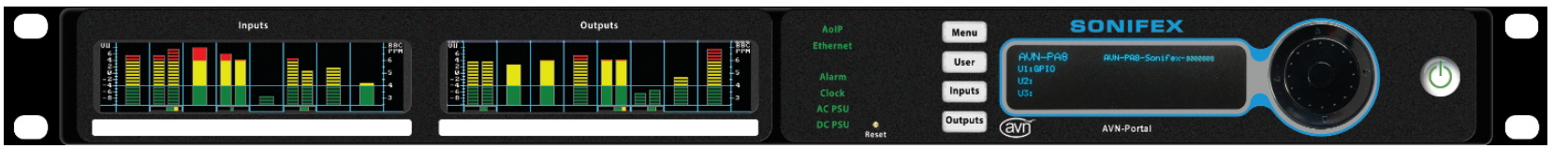
Accessories

AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.



AVN-PA8T GPIO Routing Window.

AVN-PA8D 8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs, AES67 Portal, with Detailed Meter Displays



AVN-PA8D Front View.

The AVN-PA8D is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight stereo analogue line inputs, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PA8D becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PA8D has the standard OLED front panel display, and the additional twin TFT

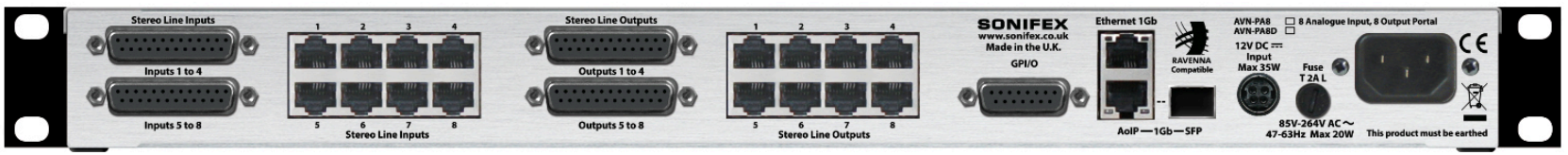
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight stereo analogue on D-type sockets with AES59 pinout, paralleled with eight RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PA8D Rear View.

meter displays. The OLED display provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view."

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

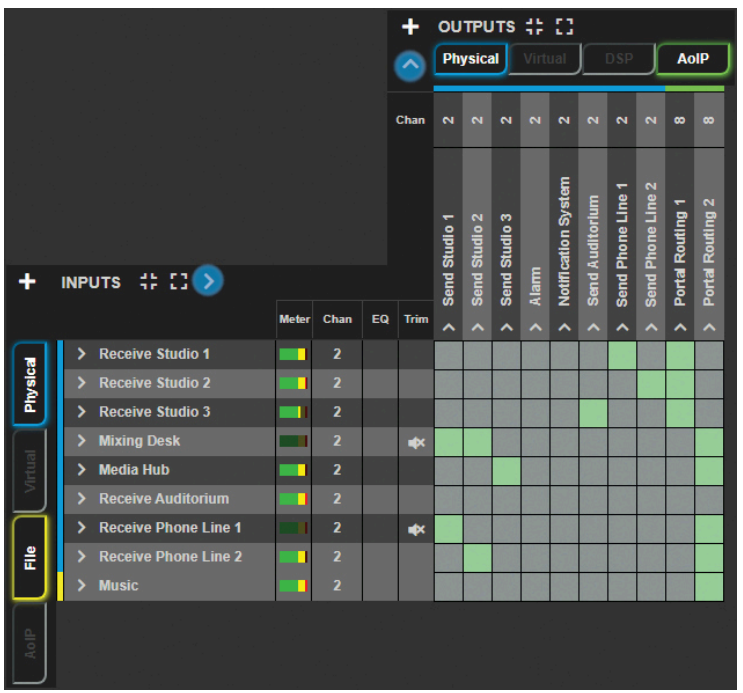
The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PA8D uses D-type sockets with AES59 analogue pinout, paralleled with eight RJ45 ▶



AVN-PA8D Show or Hide Status Icons

connectors using the StudioHub® pinout."

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

8-Way Analogue Headphone Distribution System

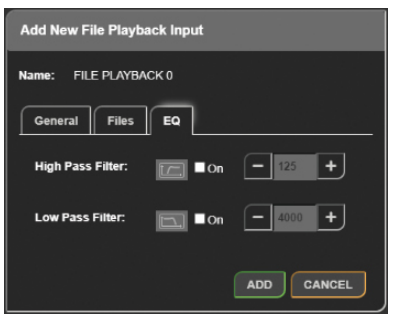
The AVN-PA8D StudioHub® outputs can be combined with multiple AVN-HA1 headphone amplifiers to provide a headphone distribution system. Output connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HA1 can be used as another GPI for muting the output."

The AVN-PA8D is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo analogue on D-type sockets with AES59 pinout, paralleled with eight RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

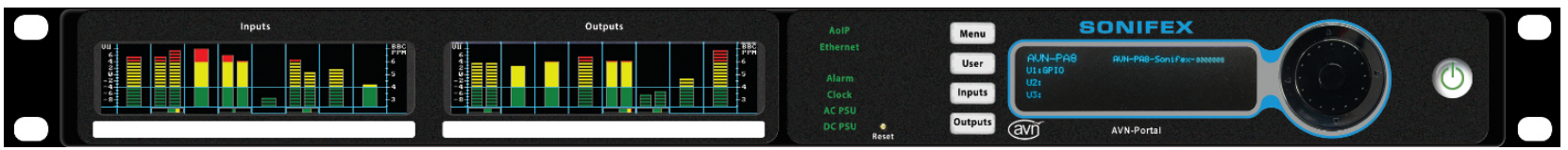
Technical Specification For AVN-PA8D

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB
Common Mode Rejection:	> 70dB @ 1kHz
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Connections	
Inputs:	2 x D-Sub (DB-25) connections (TASCAM AES-59 analogue pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
Outputs:	2 x D-Sub (DB-25) connections (TASCAM AES-59 analogue pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
GPIO:	1 x D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10 14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PA8D:	8 Stereo analogue line inputs & 8 stereo analogue line outputs, AES67 portal, with detailed meter displays
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.98 kg Gross: 4.02kg Nett: 6.56 lbs Gross:8.84 lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.



AVN-PA8D EQ Tab of ADD New File Playback Input.

AVN-PA8TD 8 Stereo Analogue Line Inputs & 8 Stereo Analogue Line Outputs on Terminal Blocks, AES67 Portal, with Detailed Meter Displays



AVN-PA8TD Front View.

The AVN-PA8TD is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight stereo analogue line inputs, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding. Physical audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PA8TD becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

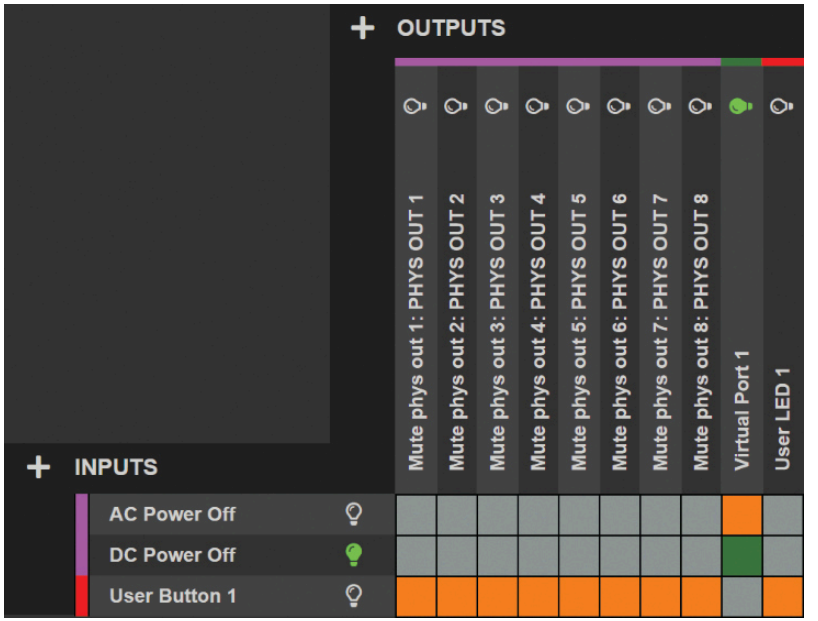
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight stereo analogue inputs and eight stereo analague outputs on terminal block connections
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PA8TD GPIO Routing Window.

Front Panel Displays, Metering & Controls

The AVN-PA8TD has the standard OLED front panel display and the additional twin TFT meter displays. The OLED display provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view.

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

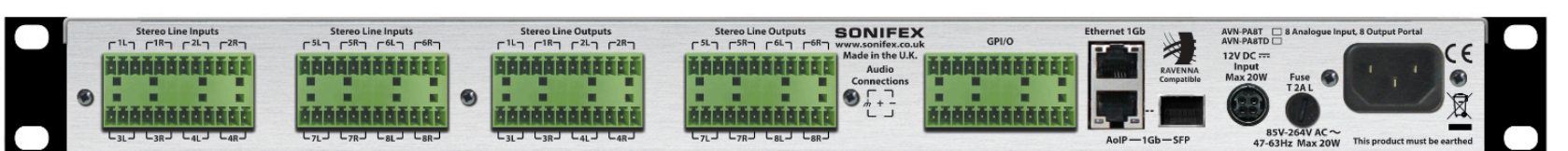
Physical Inputs & Outputs

For physical audio connections and GPIO the AVN-PA8TD uses terminal block connections.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

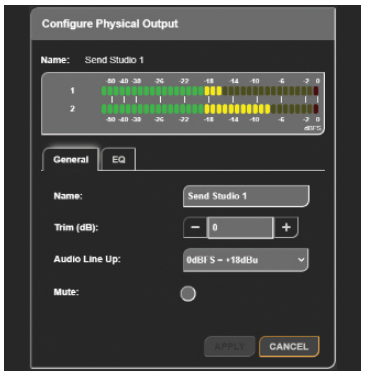
For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.



AVN-PA8TD Rear View.

The AVN-PA8TD is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo analogue inputs and eight stereo analague outputs on terminal block connections
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

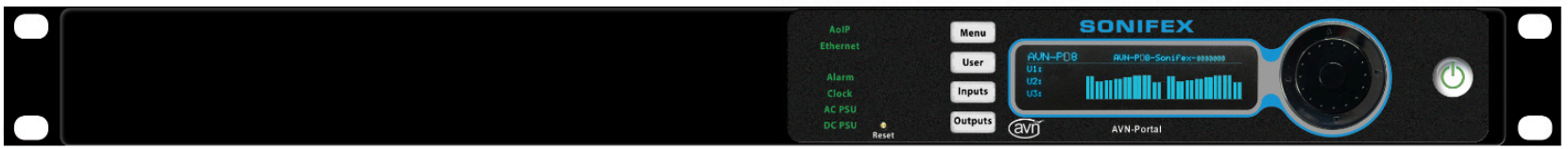


AVN-PA8 Configure Physical Output Window.

Technical Specification For AVN-PA8TD

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB
Common Mode Rejection:	> 70dB @ 1kHz
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Connections	
Inputs:	2 x 24-Pin Phoenix style terminal blocks (Analogue pinout).
Outputs:	2 x 24-Pin Phoenix style terminal blocks. (Analogue pinout).
GPIO:	1 x 24-Pin Phoenix style terminal blocks.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PA8TD:	8 Stereo analogue line inputs & 8 stereo analogue line outputs on terminal blocks, AES67 portal, with detailed meter displays
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.90kg Gross: 3.94 kg Nett: 6.38lbs Gross: 8.67lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.

AVN-PD8 8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs, AES67 Portal



AVN-PD8 Front View.

The AVN-PD8 is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight stereo AES3 digital inputs, eight stereo AES3 digital outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PD8 becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PD8 unit is supplied with the standard OLED front panel display, which provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, version information, and basic metering (The 'D' version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain

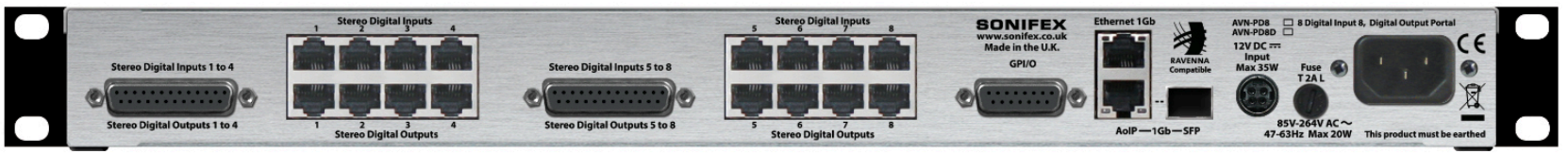
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful AES3 & AES67 mix engine which allows for multiple applications: 8 output AES3 zone mixer, 8 channel digital mixer, 64 channel AES67 stream distribution amplifier from digital sources, 8 channel headphone distribution system (with AVN-HD1 units).

Features:

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on D-type sockets with AES59 pinout, paralleled with 8 x RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PD8 Rear View.

functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PD8 uses D-type sockets with AES59 digital pinout, paralleled with eight RJ45 connectors using the StudioHub® pinout.

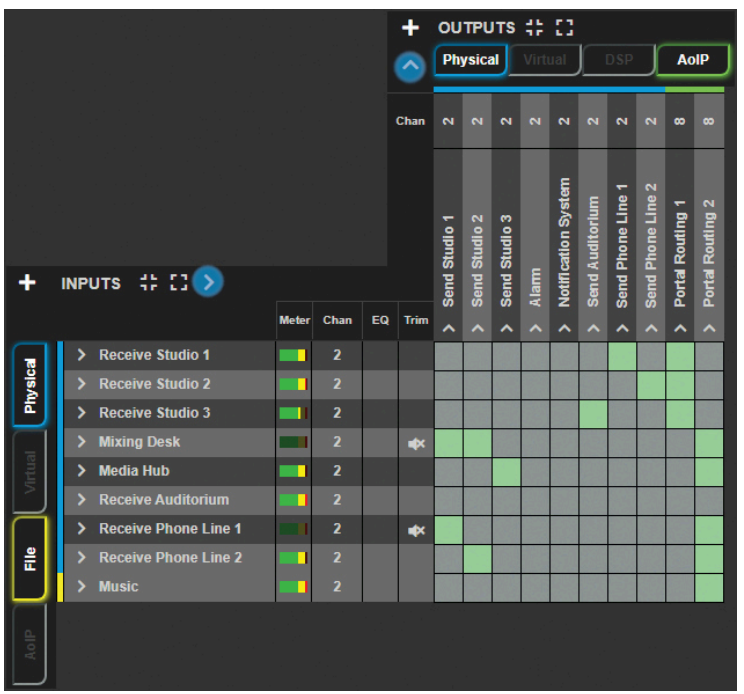
There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

8 Way Digital Headphone Distribution System

The AVN-PD8 StudioHub® outputs can be combined with multiple AVN-HD1 headphone amplifiers to provide a headphone distribution system. Output ▶



AVN-PD8 Show or Hide Status Icons

connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HD1 can be used as another GPI for muting the output.

The AVN-PD8 is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on D-type sockets with AES59 pinout, paralleled with 8 x RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

AVN-PD8 Network Port Settings.

Technical Specification For AVN-PD8

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Stereo Digital Input	
Input Impedance:	110Ω balanced
Supported Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to 48kHz)
Output Impedance:	110Ω balanced
Supported Output Rates:	48kHz
Connections	
Inputs:	1 x D-Sub (DB-25) connection (TASCAM AES-5 digital pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
Outputs:	1 x D-Sub (DB-25) connection (TASCAM AES-59 digital pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
GPIO:	1 x D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10 14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PD8:	8 Stereo AES3 digital inputs & 8 stereo AES3 digital outputs, AES67 portal

Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.88 kg Gross:3.92 kg Nett: 6.34 lbs Gross:8.62 lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HD1:	Digital Headphone Amplifier.

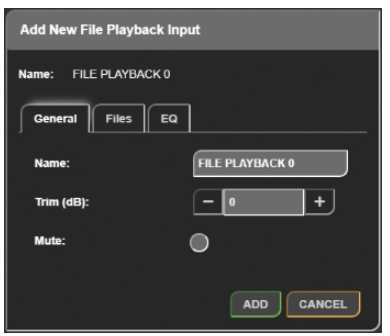
AVN-PD8T 8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs on Terminal Blocks, AES67 Portal

AVN-PD8T Front View.

The AVN-PD8T is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight stereo AES3 digital inputs, eight stereo AES3 digital outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding. Physical digital audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles. ▶



AVN-PD8T General Tab of the Add New File Playback Input.

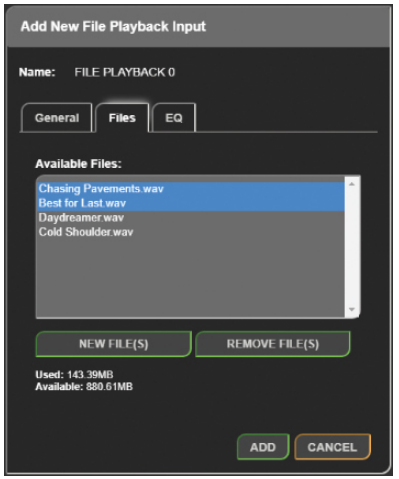
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful AES3 & AES67 mix engine which allows for multiple applications: 8 output AES3 zone mixer, 8 channel digital mixer, 64 channel AES67 stream distribution amplifier from digital sources, 8 channel headphone distribution system (with AVN-HD1 units).

Features:

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).



AVN-PD8T Files Tab of the Add New File Playback Input.

With this flexibility, the AVN-PD8T becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PD8T unit is supplied with the standard OLED front panel display, which provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, version information, and basic metering (The 'D' version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

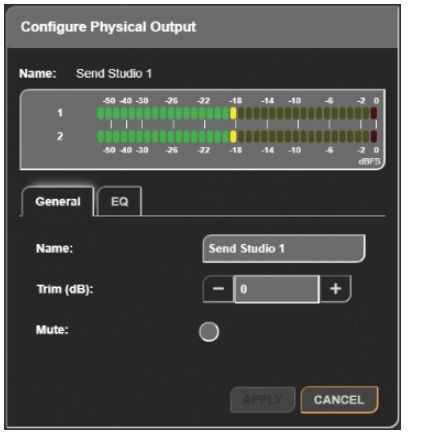
For physical digtial audio connections and GPIO the AVN-PD8T uses terminal block connections.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

The AVN-PD8T is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs



AVN-PD8T Configure Physical Output.

and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software

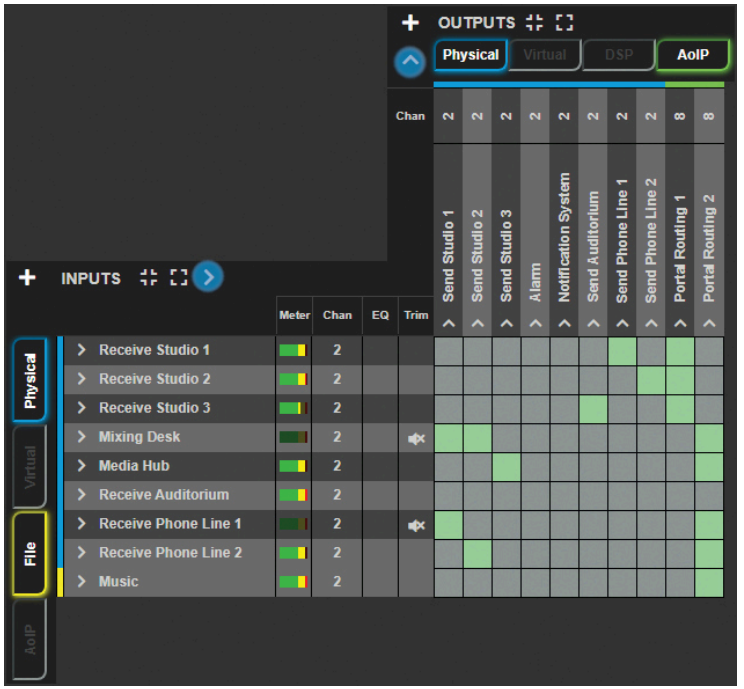
AVN-PD8T Rear View.

controlled router/mixer.

- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).

Technical Specification For AVN-PD8T

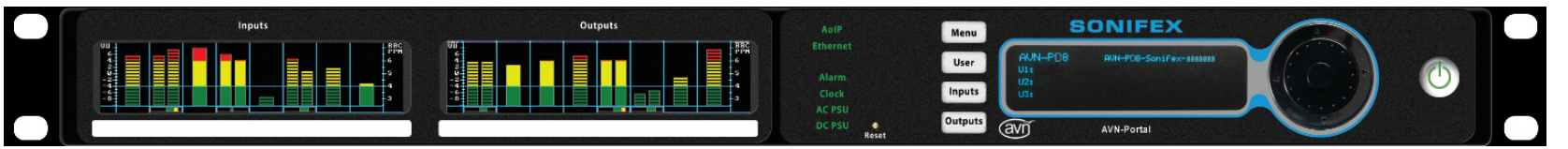
Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Stereo Digital Input	
Input Impedance:	110Ω balanced
Supported Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to 48kHz)
Output Impedance:	110Ω balanced
Supported Output Rates:	48kHz
Connections	
Inputs:	2 x 24-Pin Phoenix style terminal blocks (Digital pinout).
Outputs:	2 x 24-Pin Phoenix style terminal blocks. (Digital pinout).
GPIO:	1x 24-Pin Phoenix style terminal blocks.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PD8T:	8 Stereo AES3 digital inputs & 8 stereo AES3 digital outputs on terminal blocks, AES67 portal
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.86 kg Gross: 3.90kg Nett: 6.29 lbs Gross: 8.58lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HD1:	Digital Headphone Amplifier.



AVN-PD8T Show or Hide Status Icons

AVN-PD8T Network Port Settings.

AVN-PD8D 8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs, AES67 Portal, with Detailed Meter Displays



AVN-PD8D Front View.

The AVN-PD8D is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight stereo AES3 digital inputs, eight stereo AES3 digital outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/ recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PD8D becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote

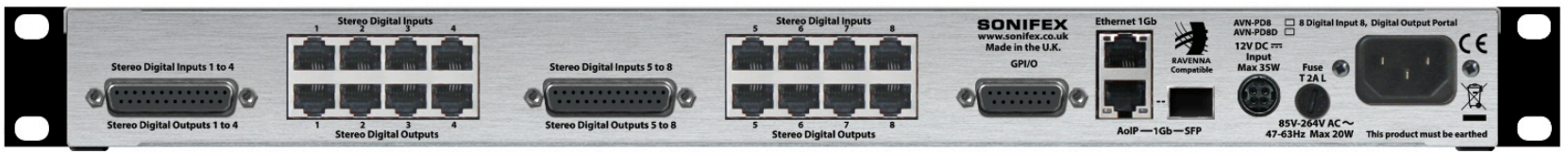
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful AES3 & AES67 mix engine which allows for multiple applications: 8 output AES3 zone mixer, 8 channel digital mixer, 64 channel AES67 stream distribution amplifier from digital sources, 8 channel headphone distribution system (with AVN-HD1 units).

Features:

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on D-type sockets with AES59 pinout, paralleled with 8 x RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PD8D Front View.

handling via GPIO, VGPIO, SNMP and Ember+ support.

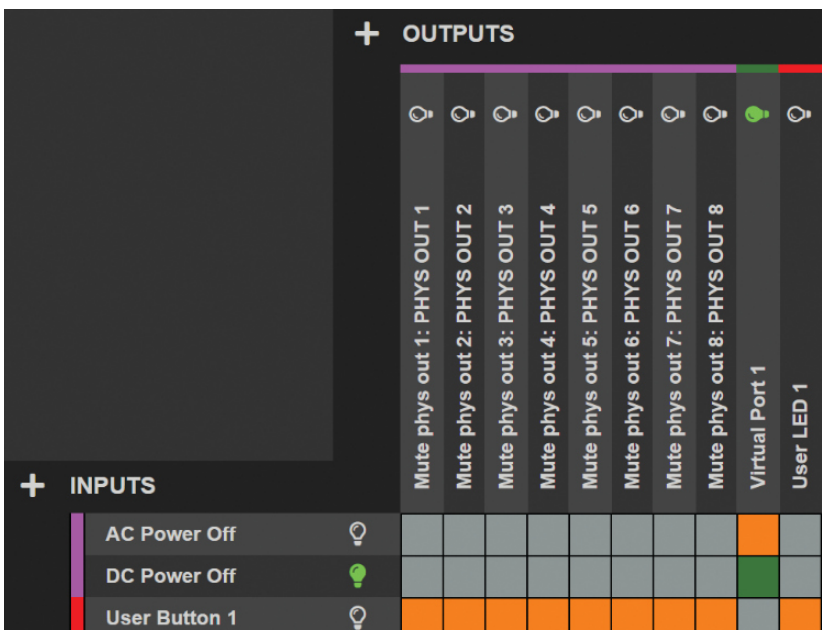
Front Panel Displays, Metering & Controls

The AVN-PD8D has the standard OLED front panel display, and the additional twin TFT meter displays. The OLED display provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view.

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.



AVN-PD8D GPIO Routing Window.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PD8D uses D-type sockets with AES59 digital pinout, paralleled with eight RJ45 connectors using the StudioHub® pinout.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

8 Way Digital Headphone Distribution System

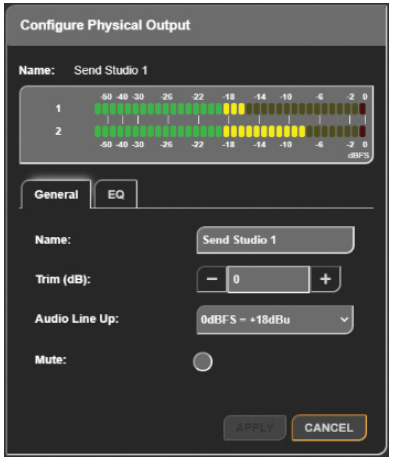
The AVN-PD8 StudioHub® outputs can be combined with multiple AVN-HD1 headphone amplifiers to provide a headphone distribution system. Output connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HD1 can be used as another GPI for muting the output.

The AVN-PD8D is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on D-type sockets with AES59 pinout, paralleled with 8 x RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition
- Optional multi-channel Opus codec encoding and decoding
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

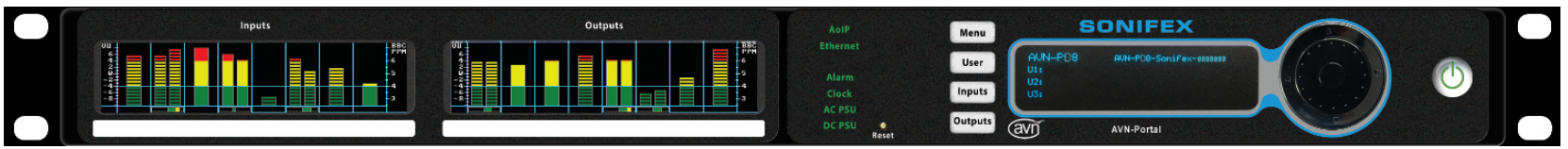
Technical Specification For AVN-PD8D

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Stereo Digital Input	
Input Impedance:	110Ω balanced
Supported Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to 48kHz)
Output Impedance:	110Ω balanced
Supported Output Rates:	48kHz
Connections	
Inputs:	1 x D-Sub (DB-25) connection (TASCAM AES-5 digital pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
Outputs:	1 x D-Sub (DB-25) connection (TASCAM AES-59 digital pinout) paralleled with 8 RJ45 connections (StudioHub+ pinout).
GPIO:	1 x D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10 14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PD8D:	8 Stereo AES3 digital inputs & 8 stereo AES3 digital outputs, AES67 portal, with detailed meter displays
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.82 kg Gross: 3.86kg Nett: 6.20 lbs Gross:8.49lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HD1:	Digital Headphone Amplifier.



AVN-PD8D Configure Physical Output Window.

AVN-PD8TD 8 Stereo AES3 Digital Inputs & 8 Stereo AES3 Digital Outputs on Terminal Blocks, AES67 Portal, with Detailed Meter Displays



AVN-PD8TD Front View.

The AVN-PD8TD is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight stereo AES3 digital inputs, eight stereo AES3 digital outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding. Physical audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PD8TD becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PD8TD has the standard OLED front panel display, and the additional twin TFT meter displays. The OLED display provides ▶

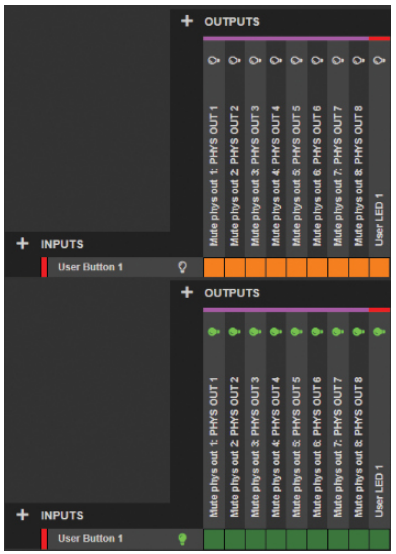
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful AES3 & AES67 mix engine which allows for multiple applications: 8 output AES3 zone mixer, 8 channel digital mixer, 64 channel AES67 stream distribution amplifier from digital sources, 8 channel headphone distribution system (with AVN-HD1 units).

Features:

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on terminal black connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PD8TD GPIO Routing Window.

detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view.

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

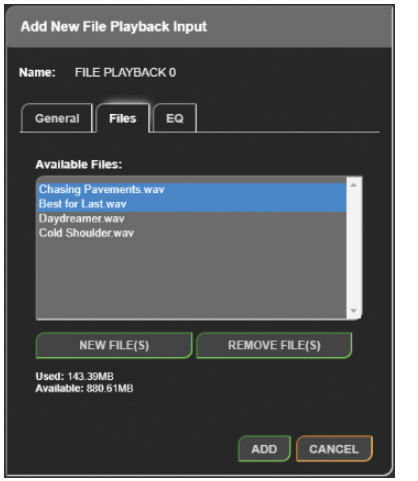
For physical AES3 digital audio connections and GPIO, the AVN-PD8TD uses terminal block connections.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

The AVN-PD8TD is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8



AVN-PD8TD Add New File Playback Input Window.

(digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

AVN-PD8TD Rear View.

- Eight stereo digital AES3 inputs and eight stereo digital AES3 outputs on terminal black connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

Technical Specification For AVN-PD8TD

Audio-Over-IP Specification

Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz

Ember+ Interface Connection

Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

Timing Synchronisation

Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Stereo Digital Input

Input Impedance:	110Ω balanced
Supported Input Rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz, 176.4kHz and 192kHz (sample rate converted to 48kHz)
Output Impedance:	110Ω balanced
Supported Output Rates:	48kHz

Connections

Inputs:	2 x 24-Pin Phoenix style terminal blocks (Digital pinout).
Outputs:	2 x 24-Pin Phoenix style terminal blocks. (Digital pinout).
GPIO:	1x 24-Pin Phoenix style terminal blocks.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.

Equipment Type

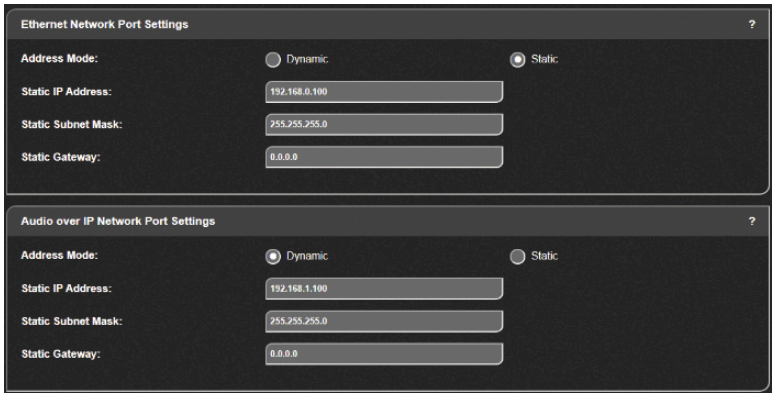
AVN-PD8TD:	8 Stereo AES3 digital inputs & 8 stereo AES3 digital outputs on terminal blocks, AES67 portal, with detailed meter displays

Physical Specification

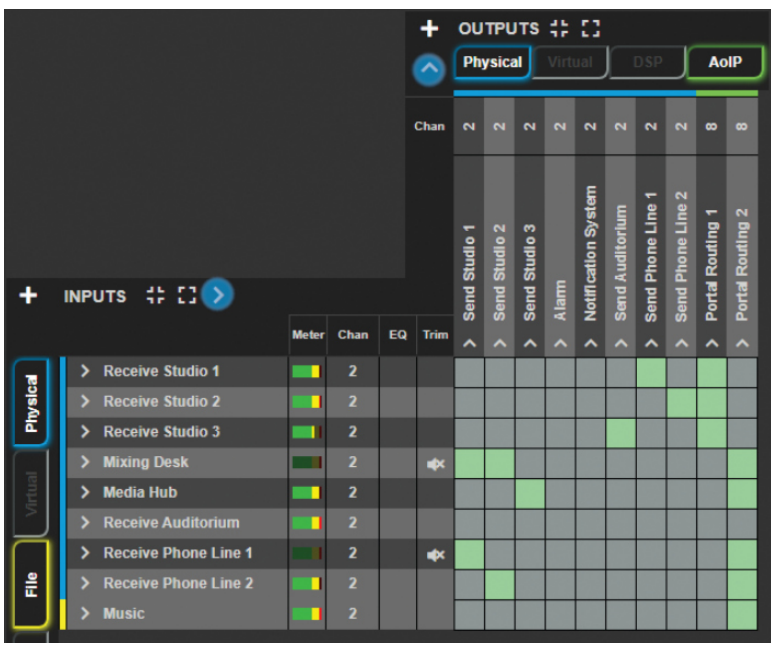
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.80kg Gross: 3.84kg Nett: 6.16 lbs Gross: 8.45lbs

Accessories

AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HD1:	Digital Headphone Amplifier.

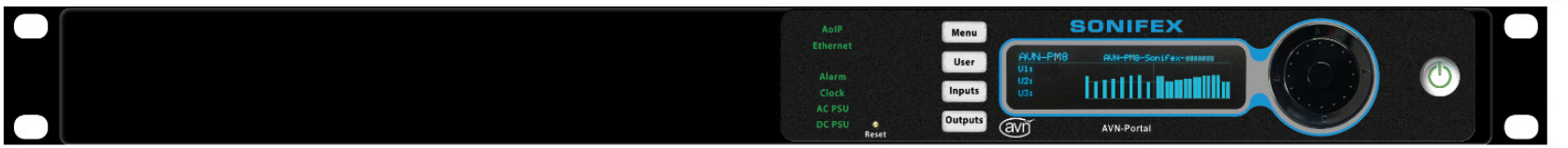


AVN-PD8TD Network Port Settings.



AVN-PD8TD Show or Hide Status Icons

AVN-PM8 8 Mic/Line Inputs, 8 Stereo Analogue Line Outputs, AES67 Portal



AVN-PM8 Front View.

The AVN-PM8 is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight mic/line inputs with switchable phantom power, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PM8 becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PM8 unit is supplied with the standard OLED front panel display, which provides detailed status information on device name, network addresses, PTP

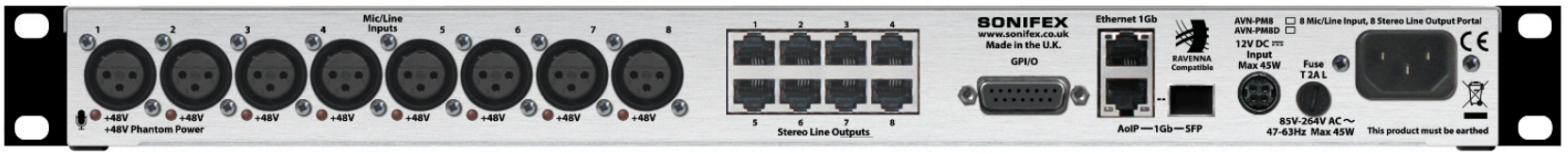
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

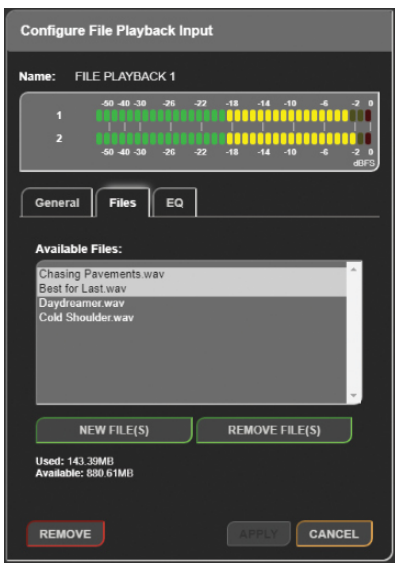
Features:

- Eight balanced XLR mic/line inputs and eight stereo line outputs on RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PM8 Rear View.

clocking info, power status/voltages, version information, and basic metering (The 'D' version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain



AVN-PM8 Configure File Playback Input.

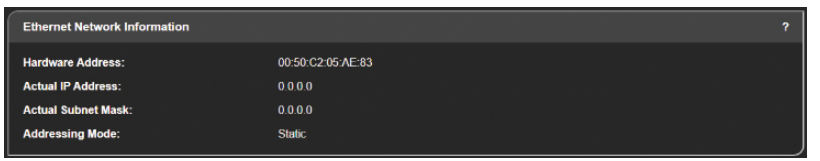
functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PM8 uses eight mic/line XLR sockets for the inputs and eight RJ45 connectors using StudioHub® pinout for the stereo analogue line outputs. +48V phantom power is



AVN-PM8 Ethernet Network Information.

available for each microphone input with a red LED presence indication.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

8-Way Digital Headphone Distribution System

The AVN-PM8 StudioHub® outputs can be combined with multiple AVN-HA1 headphone amplifiers to provide a headphone distribution system. Output connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HA1 can be used as another GPI for muting the output.

The AVN-PM8D is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 ▶

(digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight balanced XLR mic/line inputs and eight stereo line outputs on RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

Technical Specification For AVN-PM8

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz

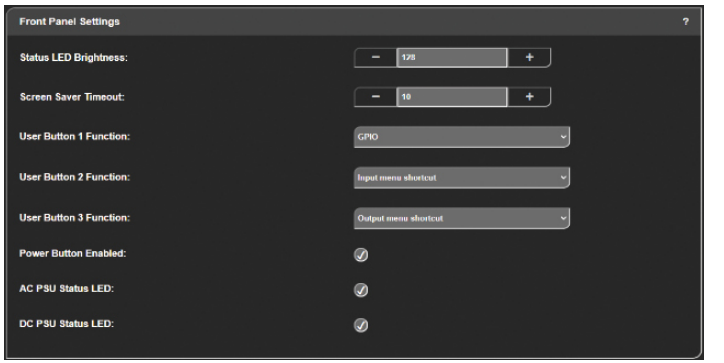
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Microphone Inputs	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain

Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
	0dBFS Line-Up: Adjustable +15/+18/+20/+22/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB

Common Mode	
Rejection:	> 70dB @ 1kHz
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/+24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω



AVN-PM8 Front Panel settings.

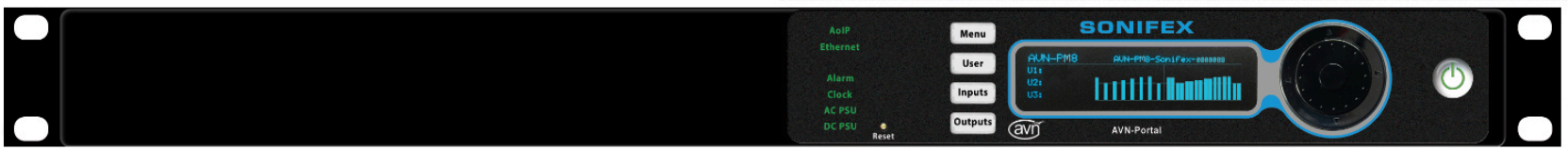
Connections	
Inputs:	XLR connectors with phantom power (toggle).
Outputs:	RJ45 connections (StudioHub+ pinout).
GPIO:	1 D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.

Equipment Type	
AVN-PM8:	Advanced audio routing, metering and equalisation unit with analogue mic/line inputs, analogue outputs, and RAVENNA AoIP.

Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.98 kg Gross:4.02 kg Nett: 6.56 lbs Gross:8.84 lbs

Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.

AVN-PM8T Advanced Audio Routing, Metering and Equalisation Unit with Terminal Type Analogue Mic/ Line Inputs, Terminal Type Analogue Outputs, and RAVENNA AoIP



AVN-PM8T Front View.

The AVN-PM8T is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, and web-enabled control and configuration. It features eight mic/line inputs with switchable phantom power, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/ recognition, and optional Opus codec encoding & decoding. Physical digital audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PM8T becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP ▶

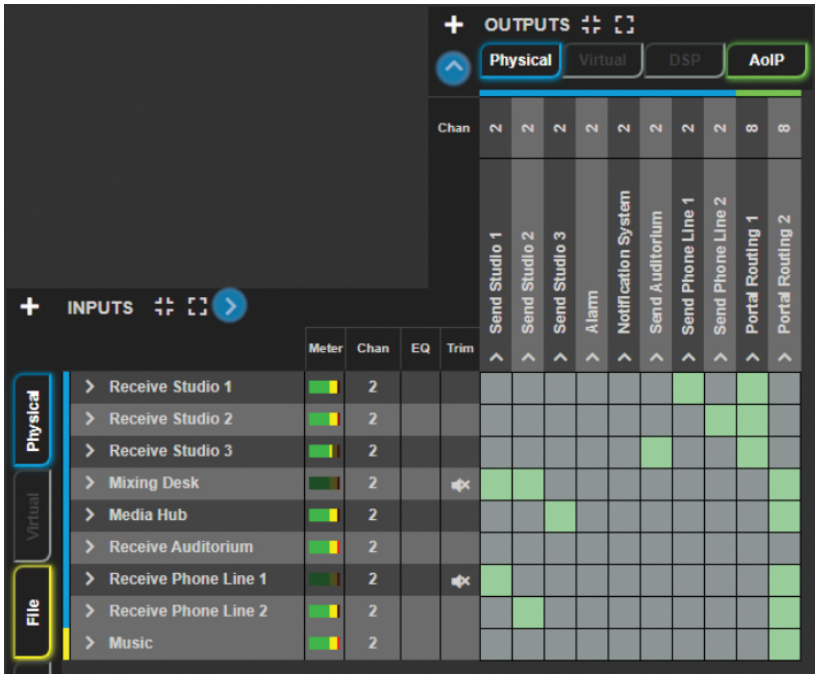
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route analogue & AES67 stream inputs to analogue & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight balanced mic/line inputs and eight stereo line outputs on on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).



AVN-PM8T Show or Hide Status Icons

and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PM8T unit is supplied with the standard OLED front panel display, which provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, version information, and basic metering (The 'D' version adds twin TFT metering with configurable ballistics). The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PM8T uses terminal block connections for the eight mic/line XLR inputs, the eight stereo analogue line outputs, and the GPIO. +48V phantom power is available for each microphone input with a red LED presence indication.

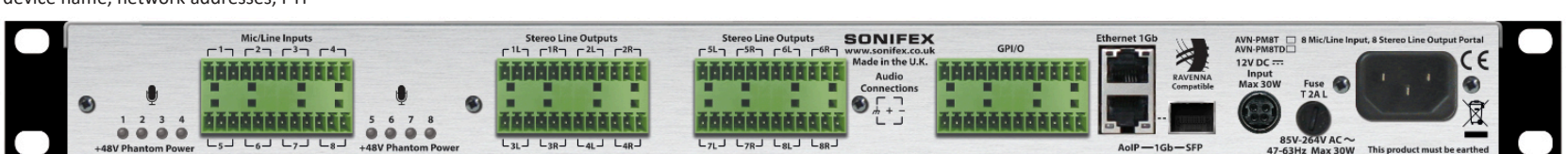
There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

The AVN-PM8T is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

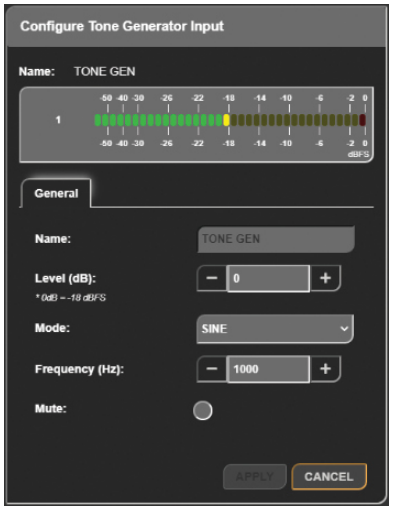
- Eight balanced mic/line inputs and eight



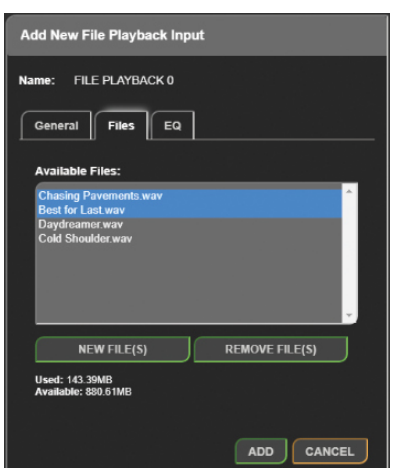
AVN-PM8T Rear View.

stereo line outputs on on terminal block connections.

- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports (terminal block connectors).



AVN-PM8T Configure Tone Generator Input.



AVN-PM8T Add New File Playback Input.

Technical Specification For AVN-PM8T

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz

Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Microphone Inputs	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain

Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB

Common Mode	
Rejection:	> 70dB @ 1kHz
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

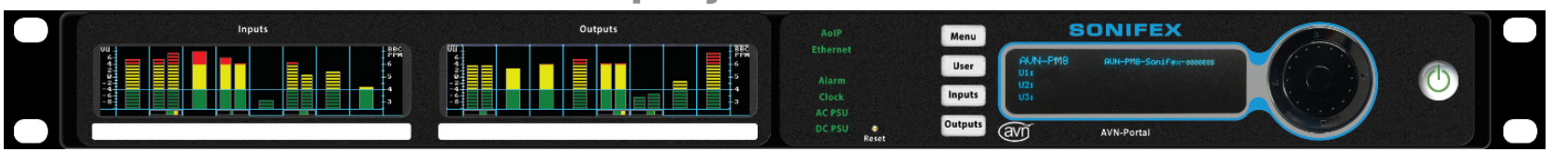
Connections	
Inputs:	1 24-Pin Phoenix style terminal blocks. (Analogue pinout)
Outputs:	2 24-Pin Phoenix style terminal blocks. (Analogue pinout)
GPIO:	1 24-Pin Phoenix style terminal blocks
Network:	2 x Gigabit Ethernet, RJ45's
Network:	1 x SFP fibre
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.

Equipment Type	
AVN-PM8T:	Advanced audio routing, metering and equalisation unit with terminal type analogue mic/line inputs, terminal type analogue outputs, and RAVENNA AoIP.

Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.96 kg Gross: 4.00kg Nett: 6.51 lbs Gross: 8.80lbs

Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.

AVN-PM8D Advanced Audio Routing, Metering and Equalisation Unit with Analogue Mic/Line Inputs, Analogue Outputs, RAVENNA AoIP, and a Detailed Customisable Display



AVN-PM8D Front View.

The AVN-PM8D is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight mic/line inputs with switchable phantom power, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PM8Dbecomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, multiple stream outputs can be provided, combined with remote

Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

Features:

- Eight balanced XLR mic/line inputs and eight stereo line outputs on RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PM8D has the standard OLED front panel display, and the additional twin TFT meter displays. The OLED display provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view.

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

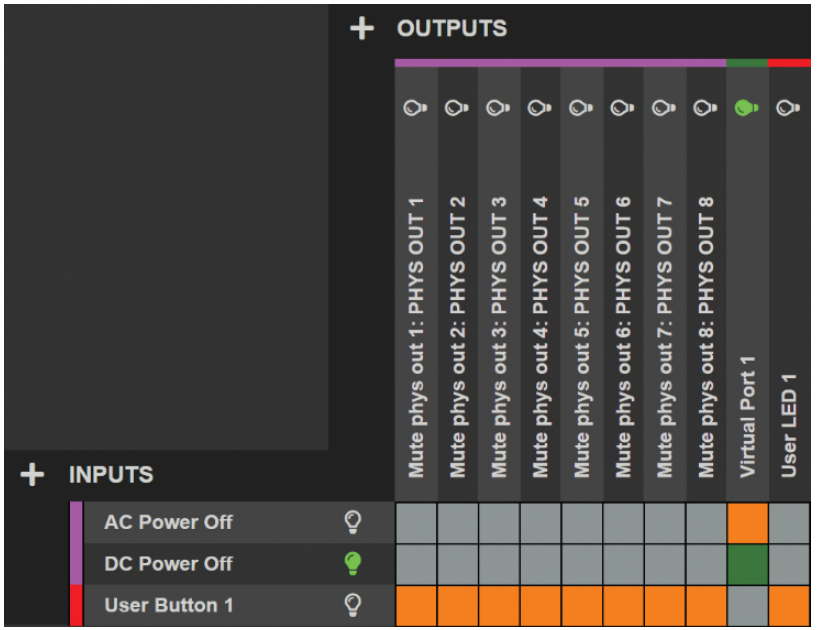
A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

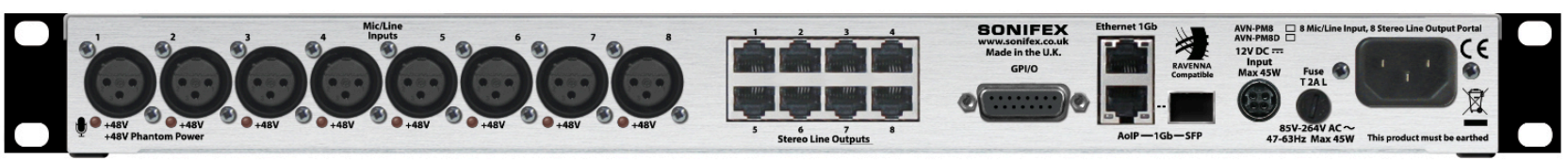
For physical audio connections, the AVN-PM8D uses eight mic/line XLR sockets for the inputs and eight RJ45 connectors using StudioHub® pinout for the stereo analogue line outputs. +48V phantom power is available for each microphone input with a red LED presence indication.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.



AVN-PM8D GPIO Routing Window.



AVN-PM8D Rear View.

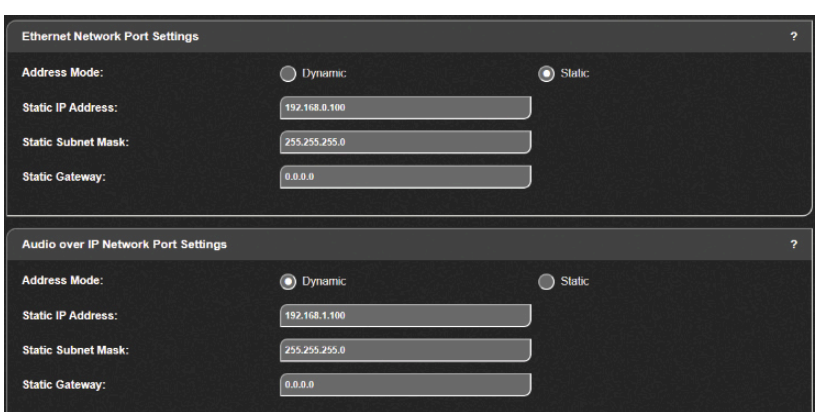
For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

8-Way Digital Headphone Distribution System

The AVN-PM8D StudioHub® outputs can be combined with multiple AVN-HA1 headphone amplifiers to provide a headphone distribution system. Output connections can supply analogue power to the satellite headphone amplifiers. Each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs. The switches on the front panel of the AVN-HA1 can be used as another GPI for muting the output.

The AVN-PM8D is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight balanced XLR mic/line inputs and eight stereo line outputs on RJ45 connectors using StudioHub® pinout.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

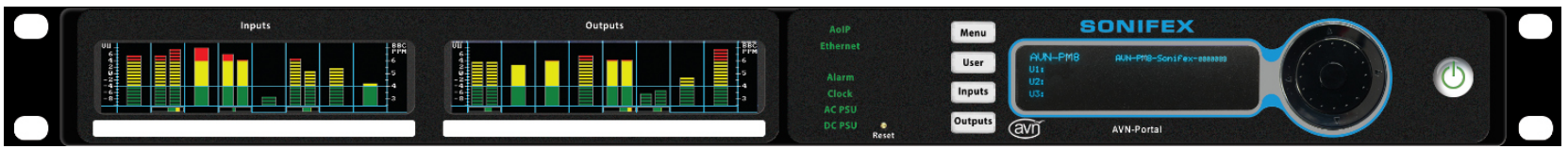


AVN-PM8D Network Port Settings.

Technical Specification For AVN-PM8D

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz
Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000
Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008
Microphone Inputs	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain
Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB
Common Mode	
Rejection:	> 70dB @ 1kHz
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Connections	
AVN-PM8/D	
Inputs:	XLR connectors with phantom power (toggle).
Outputs:	RJ45 connections (StudioHub+ pinout).
GPIO:	1 D-Sub (DA-15) connections.
Network:	2 x Gigabit Ethernet, RJ45's.
Network:	1 x SFP fibre.
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.
Equipment Type	
AVN-PM8D:	Advanced audio routing, metering and equalisation unit with analogue mic/line inputs, analogue outputs, RAVENNA AoIP, and a detailed customisable display.
Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.92 kg Gross: 3.96kg Nett: 6.42 lbs Gross:8.71 lbs
Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.

AVN-PM8TD Advanced Audio Routing, Metering and Equalisation Unit with Terminal Type Analogue Mic/ Line Inputs, Terminal Type Analogue Outputs, RAVENNA AoIP, and a Detailed Customisable Display



AVN-PM8TD Front View.

The AVN-PM8TD is a powerful audio mix engine and AES67 interface with built-in mixer matrix, assignable DSP functions, GPIO, audio playback, web-enabled control and configuration, and twin TFT meter displays. It features eight mic/line inputs with switchable phantom power, eight stereo analogue line outputs, plus a dedicated AES67 Audio over IP (AoIP) portal, available via RJ45 Ethernet connection or an SFP (small form-factor pluggable) port. Recent firmware additions include tone generation, audio file playback, DTMF generation/recognition, and optional Opus codec encoding & decoding. Physical audio inputs and outputs, and GPIO provided on terminal block connectors.

At the heart of this unit is a powerful mix engine with web-enabled configuration and control. Any physical audio input and AES67 stream input can be mixed or routed to any physical audio output or AES67 stream output, with gain adjustment at the input, the mix point or the output. DSP functions, such as gain and filtering, can be added at inputs, outputs and cross-points.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created, supporting the full range of AES67 packet times and channel counts. It can send or receive RAVENNA and AES67 AoIP streams, or AES67-enabled Dante® flows (discovered using SAP). Input and output AES67 streams can be individually added/ modified and the SDP of each stream can be checked and edited. The unit can act as a PTP master clock or slave clock and supports IEEE1588-2008 PTPv2 media and default profiles.

With this flexibility, the AVN-PM8TD becomes an advanced problem-solving box for any applications where monitoring of inputs and outputs and mixing of signals is required. With native AES67 support, ▶

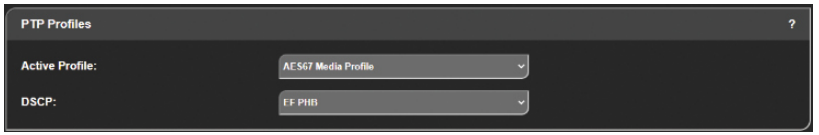
Category: AES67 Multi-Channel Audio Interfaces & Mix Engines.

Product Function: Mix and route AES3 & AES67 stream inputs to AES3 & AES67 stream outputs.

Typical Applications: A powerful analogue & AES67 mix engine which allows for multiple applications: 8 output analogue zone mixer, 8 channel clean-feed generator, 64 channel AES67 stream distribution amplifier, IFB distribution to analogue belt-packs & AES67 streams, 8 channel headphone distribution system (with AVN-HA1 units).

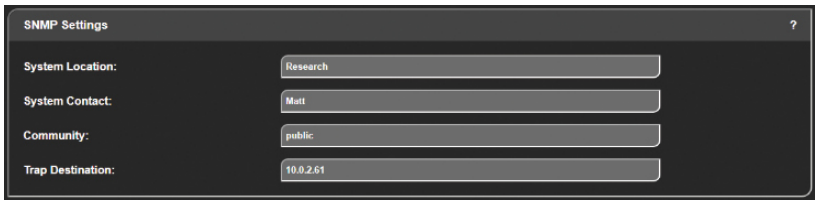
Features:

- Eight balanced mic/line inputs and eight stereo line outputs on on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.



AVN-PM8TD PTP Profiles Window.

AVN-PM8TD PTP Clock Information Window.



AVN-PM8TD SNMP Settings Window.

multiple stream outputs can be provided, combined with remote handling via GPIO, VGPIO, SNMP and Ember+ support.

Front Panel Displays, Metering & Controls

The AVN-PM8TD has the standard OLED front panel display, and the additional twin TFT meter displays. The OLED display provides detailed status information on device name, network addresses, PTP clocking info, power status/voltages, and version information. The TFT screens provide live display physical input and output levels. A rotary navigation control can be used to select a single input or output and view its metering data in a more detailed horizontal view."

The metering scale used is user configurable to one of nine different metering scales, with relevant ballistics. The metering scales available are: Dual PPM + Standard VU, EBU PPM, BBC PPM, Nordic PPM, AES Digital PPM, DIN PPM, German PPM, SMPTE RP.0155, Standard VU & Extended VU. Metering can be set to be either 'Discrete' or 'Continuous', which changes the appearance of the meter bar. Phase metering can be displayed per stereo channel, and channel idents can be shown either above or below the metering to identify each input/output.

The display and navigation controls allow editing of certain functions, limited to networking (IP addresses, friendly name, etc) and display (brightness and contrast). The front panel controls also include user configurable buttons which can be set-up to perform actions such as activating a GPIO or as a shortcut button to jump to a specified menu on the OLED display.

Front panel LEDs show the AoIP network status, synchronisation status and the status of the AC and DC power supply inputs. The brightness of the OLED display and LED indicators can be continuously adjusted for low or high lighting conditions.

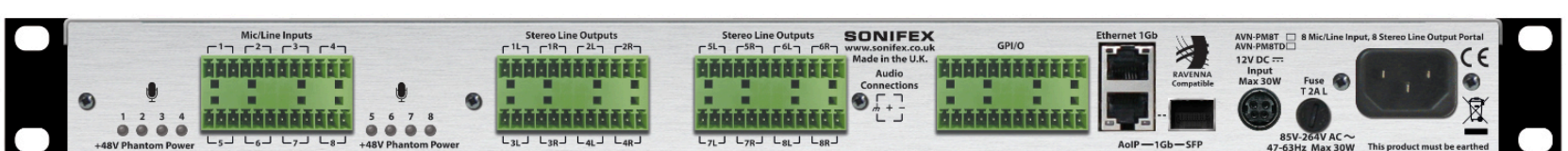
A front panel power button is available to turn the unit on and off. The power button is disabled by default but can be enabled through the 'Display Settings' web page.

Physical Inputs & Outputs

For physical audio connections, the AVN-PM8TD uses terminal block connections for the eight mic/line XLR inputs, the eight stereo analogue line outputs, and the GPIO. +48V phantom power is available for each microphone input with a red LED presence indication.

There are two Ethernet RJ45 connections - one for control and one for the AoIP network. There is also an Ethernet SFP module that, when used, replaces the AoIP RJ45 connection, e.g. for a 1Gbit/s copper or optical SFP transceiver. When an SFP is used, this replaces the AoIP RJ45 connection.

A rear panel GPIO connector provides 10 local ports which can be user configured as inputs or outputs and provide software-controlled functionality. A voltage free relay contact can be used to operate external equipment. There are virtual GPIO ports which can be used to trigger events over the network between devices.



AVN-PM8TD Rear View.

For remote operation and monitoring, SNMP V2 is supported and the units can be controlled using Ember+ commands.

The AVN-PM8TD is part of the Sonifex AVN-Portal series, which includes the AVN-PA8 (analogue physical I/O), the AVN-PD8 (digital physical I/O), and the AVN-PM8 (mic inputs, line outputs). Each of those has three additional variants indicated by letter suffixes: The AVN-PA8D features two bright TFT meter displays which provide a live display of the levels of the physical inputs and outputs respectively; the AVN-PA8T features rear panel terminal block connectors for all physical inputs and outputs; the AVN-PA8TD features both the TFT displays and terminal block connections.

- Eight balanced mic/line inputs and eight stereo line outputs on on terminal block connections.
- AES67 portal on 1Gb Ethernet (RJ45) and 1Gb SFP ports.
- Gain and DSP functions available at inputs, outputs, and crosspoints.
- Twin TFT Meter displays with selectable ballistics, plus phase and channel ident display.
- Responsive webserver software controlled router/mixer.
- Up to eight AoIP input streams with a maximum of 16 channels to be routed.
- Up to eight AoIP output streams with a maximum of eight channels each (64 channels).
- Audio file playback.
- DTMF tone generation/recognition.
- Optional multi-channel Opus codec encoding and decoding.
- Dual AC and DC power supply inputs.
- 10 user assignable GPIO ports.

Technical Specification For AVN-PM8TD

Audio-Over-IP Specification	
Open Standards:	RAVENNA, AES67
Device Discovery:	Bonjour (mDNS / DNS-SD) or SAP
Audio Delivery:	RTP/UDP over IPv4 multicast
QoS:	DiffServ
Stream Management:	RTSP/SDP
Control:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	Up to 8
Frames Per Packet:	48
Transmit Streams:	Up to 8
Sample Rate:	48kHz

Ember+ Interface Connection	
Interface Type:	Provider
Network Interface:	Ethernet port and AoIP port
Port:	9000

Timing Synchronisation	
Profile Support:	Default, AES67 Media & Custom profiles
Timing Protocol:	PTPv2, IEEE1588-2008

Microphone Inputs	
Input Impedance:	> 2.5kΩ balanced
Gain Range:	0dB to +60dB
0dBFS Line-Up:	Adjustable in steps of 3dB from -58dBu to +2dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
Noise:	-127dBu, 20kHz BW, Rs=200Ω ref. 76dB gain

Balanced Line Inputs	
Input Impedance:	> 20kΩ balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	< -100dB

Common Mode	
Rejection:	> 70dB @ 1kHz

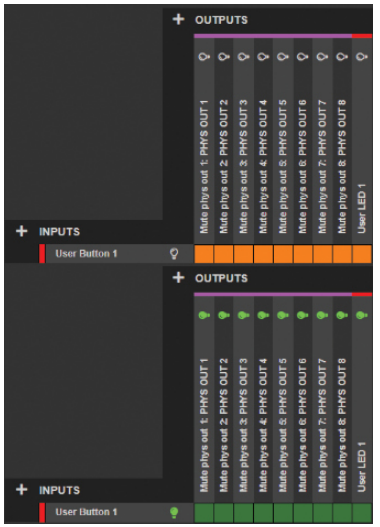
Balanced Line Outputs	
Output Impedance:	< 50Ω balanced
0dBFS Line-Up:	Adjustable +15/+18/+20/+22/ +24dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	< -110dBFS, -30dBFS, 20Hz to 20kHz, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

Connections	
Inputs:	1 24-Pin Phoenix style terminal blocks. (Analogue pinout)
Outputs:	2 24-Pin Phoenix style terminal blocks. (Analogue pinout)
GPIO:	1 24-Pin Phoenix style terminal blocks
Network:	2 x Gigabit Ethernet, RJ45's
Network:	1 x SFP fibre
Power:	Mains AC Input: Universal filtered IEC, continuously rated 85-264VAC, 47-63Hz, 20W.
DC Input:	4-pin 7.5A power jack socket, 10-14VDC.
Fuse Rating:	Anti-surge fuse 2A 20mm x 5mm.

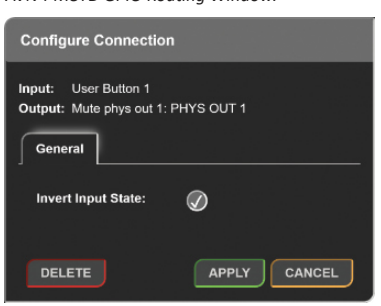
Equipment Type	
AVN-PM8TD:	Advanced audio routing, metering and equalisation unit with terminal type analogue mic/line inputs, terminal type analogue outputs, RAVENNA AoIP, and a detailed customisable display.

Physical Specification	
Dimensions (Raw):	48.3cm (W) x 17.5cm (D) x 4.4cm (H)(1U) 19" (W) x 6.9" (D) x 1.8" (H) (1U)
Dimensions (Boxed):	55.5cm (W) x 29cm (D) x 18cm (H) 22" (W) x 11.4" (D) x 7" (H)
Weight:	Nett: 2.90kg Gross: 3.94kg Nett: 6.38 lbs Gross: 8.67lbs

Accessories	
AVN-DC060:	60W DC power supply for AVN range with KPJX-4S plug
AVN-HA1:	Analogue Headphone Amplifier.



AVN-PM8TD GPIO Routing Window.



AVN-PM8TD Configure Connection Window (GPIO Settings) Window.

AVN-HA1 Analogue Headphone Amp for AVN-PA8/D & AVN-PM8/D Portals

Category: Headphone Distribution Amplifiers.

Product Function: Simple headphone amplifier connected to the output of the AVN-PA8/PM8 portals, with volume control and button tally. Up to 8 can be connected to each AVN portal.

Typical Applications: Headphone amplifiers in a radio studio, a talks studio of a radio station/TV station, a voice-over booth or news booth.

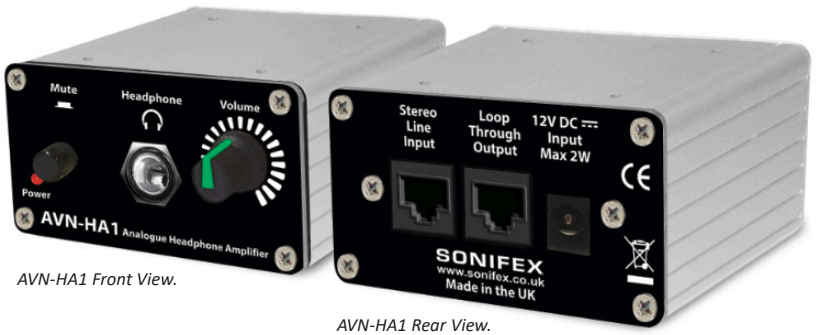
- Front panel 6.35mm (1/4") headphone socket and volume control knob, with mute/GPO push button.
- Locking DC power connector if a portal is not being used to supply the unit with power.
- Analogue audio input on RJ45 (the connector provides power to the unit and a GPO back to the portal).
- Loop through audio output on RJ45 (power and GPO signal are not connected).

The AVN-PA8/D and the AVN-PM8/D Portals can be combined with multiple AVN-HA1 headphone amplifiers to provide a headphone distribution system – the portal output connections can supply analogue power to satellite headphone amplifiers and the unit can be mounted using the CM-MNT1 desk mount panel.

The AVN-PA8/D and AVN-PM8/D can be combined with multiple Sonifex **AVN-HA1** headphone amplifiers to provide 8 separate headphone signals where each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs.

- Front panel 6.35mm (1/4") headphone socket and volume control knob.
- Front panel Mute/GPO push button.
- Analogue audio input on RJ45 (the connector provides power to the unit and a GPO back to the portal).
- Loop through audio output on RJ45 (power and GPO signal are not connected).
- Locking DC power connector (if a portal is not being used to supply the unit with power).

Note: The AVN-HA1 is an analogue input product taking an analogue audio feed from the AVN portals. It can be used independently of the portals by using the separate DC input for power and a separate analogue input.



AVN-HA1 Front View.

AVN-HA1 Rear View.

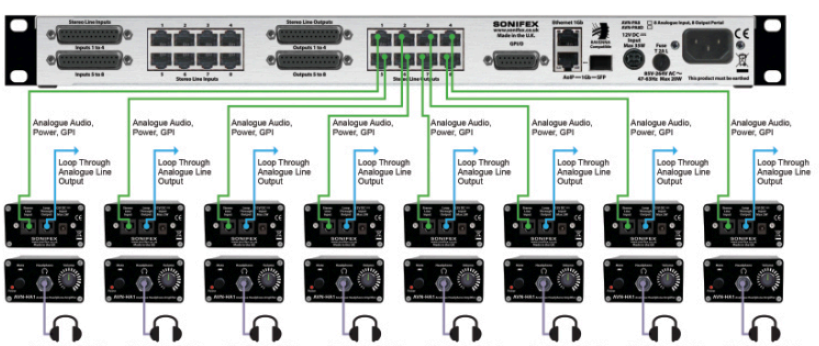
Technical Specification For AVN-HA1

Parameter	
Output Impedance: headphones.	Drives 150mW into 32Ω to 600Ω
Max Input Level:	+24dBu
Max Output Level:	+18dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-90dBu, -12dBu, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-92dBu, A weighted
Crosstalk:	-90dB, 1kHz, +4dBu
Common Mode:	-40dB, 1kHz
Cable:	Cat5/Cat5e/Cat6, Maximum length of 20m between Portal and AVN-HA1 if no external PSU is used

Equipment Type	
AVN-HA1:	Analogue Headphone Amp for AVN-PA8/D & AVN-PM8/D Portals

Physical Specification	
Dimensions: (Raw)	7.7cm (W) x 8.3cm (D) x 4.2cm (H) 3.0" (W) x 3.3" (D) x 1.7" (H)
Dimensions (Boxed):	12.7cm (W) x 22.9cm (D) x 7.6cm (H) 5" (W) x 9" (D) x 3" (H)
Weight:	Nett: 0.22kg Gross: 0.33kg Nett: 0.49lbs Gross: 0.73lbs

Accessories	
CM-MNTI	Desk Mount Pane



AVN-PA8 & Multiple AVN-HA1 Diagram

AVN-HD1 Digital Headphone Amp for AVN-PD8/D Portal

The AVN-PD8/DPortal can be combined with multiple AVN-HD1 headphone amplifiers to provide a headphone distribution system – the portal output connections can supply analogue power to satellite headphone amplifiers and the unit can be mounted using the CM-MNT1 desk mount panel.

Category: Headphone Distribution Amplifiers.

Product Function: Simple AES3 digital headphone amplifier connected to the output of the AVN-PD8/D portals, with volume control and button tally. Up to 8 can be connected to each AVN portal.

Typical Applications: Headphone amplifiers in a radio studio, a talks studio of a radio station/TV station, a voice-over booth or news booth.

- Front panel 6.35mm (1/4";) headphone socket and volume control knob, with mute/GPO push button.
- Locking DC power connector if a portal is not being used to supply the unit with power.
- AES3 digital input on RJ45 (the connector provides power to the unit and a GPO back to portal).
- AES digital output on RJ45 (power and GPO signal are not connected).

The AVN-PD8/D can be combined with multiple Sonifex **AVN-HD1** headphone amplifiers to provide 8 separate headphone signals where each headphone amplifier can be sent a separate feed, mixed from any physical or stream inputs.

- Front panel 6.35mm (¼") headphone socket and volume control knob.
- Front panel Mute/GPO push button.
- AES3 digital input on RJ45 (the connector provides power to the unit and a GPO back to the portal).
- AES3 digital output on RJ45 (power and GPO signal are not connected).
- Locking DC power connector if a portal is not being used to supply the unit with power.

Note: The AVN-HD1 is an AES3 digital input product taking an AES3 audio feed from the AVN portals. It can be used independently of the portals by using the separate DC input for power and a separate AES3 audio input.

Technical Specification For AVN-HD1

Parameter	
Output Impedance:	Drives 150mW into 32Ω to 600Ω headphones
D to A Line-up:	-6dBFS = +18dBu
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-90dBu, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise:	-92dBu, A weighted
Crosstalk:	-90dB, 1kHz, +14dBFS
Cable:	Cat5/Cat5e/Cat6, Maximum length of 20m between Portal and AVN-HD1 if no external PSU is used

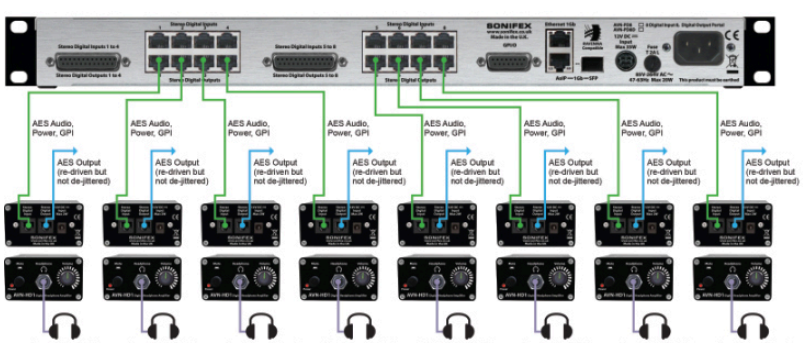
AVN-HD1 Front View.

AVN-HD1 Rear View.

Equipment Type	
AVN-HD1:	Digital Headphone Amp for AVN-PD8/D Portal

Physical Specification	
Dimensions: (Raw)	7.7cm (W) x 8.3cm (D) x 4.2cm (H) 3.0" (W) x 3.3" (D) x 1.7" (H)
Dimensions (Boxed):	12.7cm (W) x 22.9cm (D) x 7.6cm (H) 5" (W) x 9" (D) x 3" (H)
Weight:	Nett: 0.22kg Gross: 0.33kg Nett: 0.49lbs Gross: 0.73lbs

Accessories	
CM-MNTI	Desk Mount Panel



AVN-PD8 & Multiple AVN-HD1 Diagram

AVN-MPPR 4 Channel Presenter In-Ear Monitoring Remote Controller, AES67

Category: AES67 Presenter In Ear Monitoring System.

Product Function: The AVN-MPPR presenter remote provides the mixed audio to the presenter's earpiece via front panel ¼" & 3.5mm headphone outputs, together with a rear panel XLR output.

Typical Applications: Production galleries, control rooms and studios.

- 4 rotary encoders control the 4 channels of a selected virtual mixer.
- 1, XLR, plus ¼";& 3.5mm headphone outputs.
- Software-configurable output level control (one for each headphone output) to set the attenuation of the signal.
- Virtual GPO.
- IP address reset.
- Can be mounted on a standard microphone screw thread or under a desktop using optional accessories.

AVN-MPPR Front View.

The presenter mixer system allows on-air presenters to adjust the audio levels fed to their radio or wired IEM systems, whilst also allowing Sound Control Room operators to adjust the levels remotely. The system consists of 3 types of unit all connected on a common Ravenna Audio over IP (AoIP) network with added audio sources derived from Ravenna based AoIP mixers or input portals. Setup and control can be provided via webpages and/or 3rd party applications. The system uses mono audio throughout.

The 3 units are:
AVN-PX8x4C - a 2U rack mounted Mix Engine Rack.
AVN-MPPR - a small desk, or microphone stand, mounted Presenters' Remote Unit.
AVN-MPTR - a 1U rack mounted Technicians' Remote Unit.

The AVN-MPPR presenter remote provides the mixed audio to the presenter's earpiece via front panel ¼" & 3.5mm headphone outputs, together with a rear panel XLR output.

The four rotary encoders control the 4 channels of a selected virtual mixer, and give both visual feedback for mix level of the source and actual input level.

Multiple units can simultaneously control any of the 8 virtual mixers in the main mix engine with the VM number displayed on the front panel.

Audio is transported from the mix engine to the remote using AES67 AoIP with a single PoE Neutrik ™ Ethercon network connection.

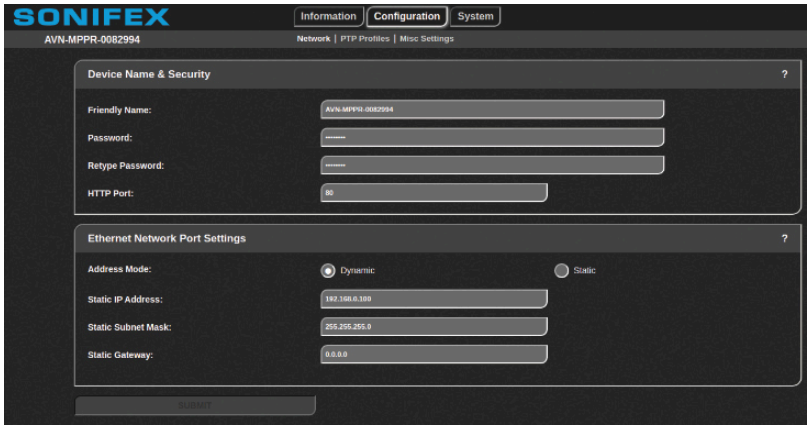
Units can be mounted on a standard microphone screw thread or under a desktop using optional accessories.

By default, the audio on the XLR output on the rear of the unit is not muted when headphones are connected to one of the sockets on the front of the unit. However, there is a "Headphone XLR Mute" configuration setting in the "Misc Settings" section of the webpages to enable the muting of the XLR output on headphone connection if required.

The headphones will have an additional software-configurable output level control (one for each headphone output) to set the attenuation of the signal to the headphone driver for comfortable levels at normal signal level from the output.

GPO Button - The button between the STATUS LEDs and the rotary controls provides a virtual GPO from the unit that can be used on the Mix Engine to trigger a GPO on its remote port. This button will illuminate red when it has been selected by this MPPR and will be yellow when initiated by a paralleled MPPR or MPTR.

IP Address Reset - As well as being able to set the IP address of the unit via the web



AVN-MPPR Network Window.

pages, it is also possible to reset the IP address of the unit from the front panel to a factory default.

At the rear of the AVN-MPPR there are three connectors and two controls:

Network Connector - This is a Neutrik etherCON RJ45 CAT5e connector with latching lock with PoE plus for unit power.

Mix Output Mono Analogue Output - The mix output is a XLR 3 pin XLR male output which is electronically balanced.

Standard GPIO Connector - This is a 15-way female 'D'-type connector with the same pin connects as for the AVN-PX8x4C Mix Engine.

- 4 rotary encoders control the 4 channels of a selected virtual mixer.
- 1, XLR, plus ¼" & 3.5mm headphone outputs.
- Software-configurable output level control (one for each headphone output) to set the attenuation of the signal.
- Virtual GPO.
- IP address reset.
- Can be mounted on a standard microphone screw thread or under a desktop using optional accessories.

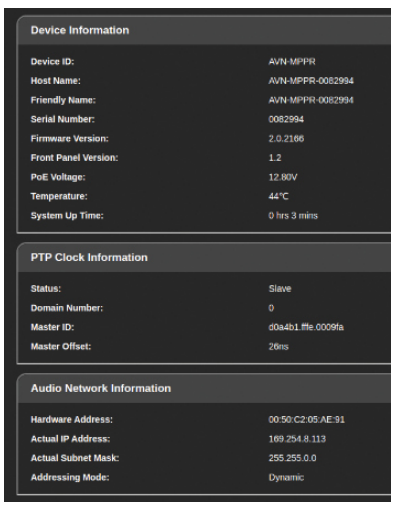
Technical Specification For AVN-MPPR

Parameter	
Output Impedance Line Input:	<50Ω balanced
Output Impedance Headphone Output:	Drives 150mW into 8Ω to 600Ω headphones
0dBFS Line-up Line Input:	+15/+18/+20/+22/+24dBu Balanced*
0dBFS Line-up Headphone Output:	+20dBu Maximum when no attenuation factor applied.
Frequency Response Line Input:	20Hz to 20kHz, +0/-0.5dB
Frequency Response Headphone Output:	20Hz to 20kHz, +0/-0.2dB
THD+N Line Input:	<-110dBFS, -30dBFS, 20Hz to 20kHz, All input gain settings, 20kHz BW
THD+N Headphone Output:	<-108dBFS, -30dBFS, 20Hz to 20kHz, unity gain, 20kHz BW
Noise Line Input:	-110dBFS, 20kHz BW, Rs=200Ω
Noise Headphone Output:	-110dBFS, 20kHz BW

PoE Power	
Standard:	802.3af
Class:	3
PD Power Range:	6.49W to 12.95W
Typical PSE Power Usage:	14W
Max PSE Power Usage:	15.4W

Equipment Type	
AVN-MPPR	4 Channel Presenter In-Ear Monitoring Remote Controller, AES67

Physical Specification	
Dimensions: (Raw)	20cm (W) x 15cm (D) x 4.2 (H) (1U) 7.9" (W) x 5.9" (D) x 1.7" (H) (1U)
Dimensions (Boxed):	28cm (W) x 19cm(D) x 9cm(H) 11" (W) x 7.5(D) x 3.5(H)
Weight:	Nett: 0.6kg Gross: 1.13kg Nett: 1.3lbs Gross: 2.45lbs



AVN-MPPR Device Information Window.

AVN-MPPR Rear View.

AVN-MPTR Technician Remote Controller

Category: AES67/Dante AoIP Products.

Product Function: Allows selection of each of 8 VMs and remote adjustment of volume controls.

Typical Applications: Production galleries, control rooms and studios.

- Allows selection of each of 8 VMs and remote adjustment of volume controls.
- 8 virtual mixer select buttons.
- 4 x rotary encoders showing input level metering and output volume control.
- GPO enable and activation buttons.
- 1Gb PoE Ethernet and power using Neutrik Ethercon.
- Status LEDs and OLED display.
- 10 x GPIO ports.
- Configured via webserver.

The presenter mixer system allows on-air presenters to adjust the audio levels fed to their radio or wired IEM systems, whilst also allowing Sound Control Room operators to adjust the levels remotely. The system consists of 3 types of unit all connected on a common Ravenna Audio over IP (AoIP) network with added audio sources derived from Ravenna based AoIP mixers or input portals. Setup and control can be provided via webpages and/or 3rd party applications. The system uses mono audio throughout.

The 3 units are:
AVN-PX8x4C - a 2U rack mounted Mix Engine Rack.
AVN-MPPR - a small desk, or microphone stand, mounted Presenters' Remote Unit.
AVN-MPTR - a 1U rack mounted Technicians' Remote Unit.

The AVN-MPTR Technician Remote Unit allows a technician to have control of a Virtual Mixer, paralleling the Presenter Remote Unit controls.

The MPTRs are assigned to a gallery and hence are normally tied to a Mix Engine. Note that no audio is present on the Technician's Remote Unit – the monitor audio for this is provided by an output on the Mix Engine routed to the technician's area, but this product has similar connectivity to the MPPR and we can supply audio enabled versions of the MPTR if required.

The Virtual Mixer (VM) to be controlled is selectable using the Virtual Mixer buttons labelled Pres 1 – Pres 8 on the front panel. The mixer selection buttons are only active when the AVN-MPTR is grabbed by a mix engine (as indicated by a green Auth LED). When a VM is selected, the corresponding

AVN-MPTR Front View.

AVN-MPTR Rear View.

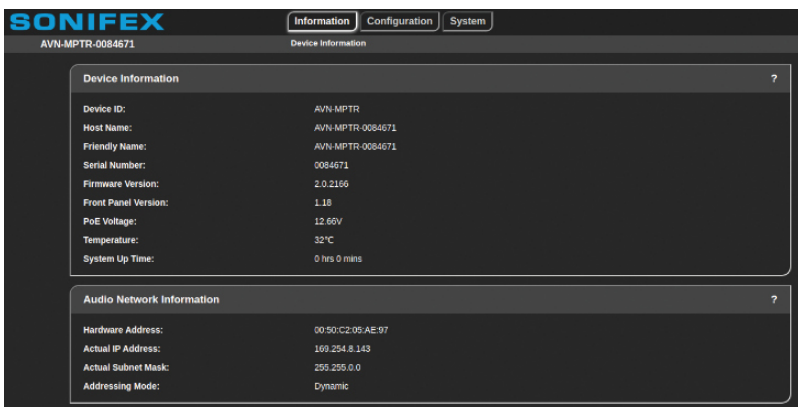
button will be illuminated green (or blue if MPPR lock out is set for that VM). If no VM is selected, then all the mixer select button LEDs will be off. Once a VM has been selected, the rotary encoders act in the same way as on the Presenter Unit.

The AVN-MPTR is able stop MPPRs that are assigned to a particular VM from controlling the source levels.

- Allows selection of each of 8 VMs and remote adjustment of volume controls.
- 8 virtual mixer select buttons.
- 4 x rotary encoders showing input level metering and output volume control.
- GPO enable and activation buttons.
- 1Gb PoE Ethernet and power using Neutrik Ethercon.
- Status LEDs and OLED display.
- 10 x GPIO ports.
- Configured via webserver.

Technical Specification For AVN-MPTR

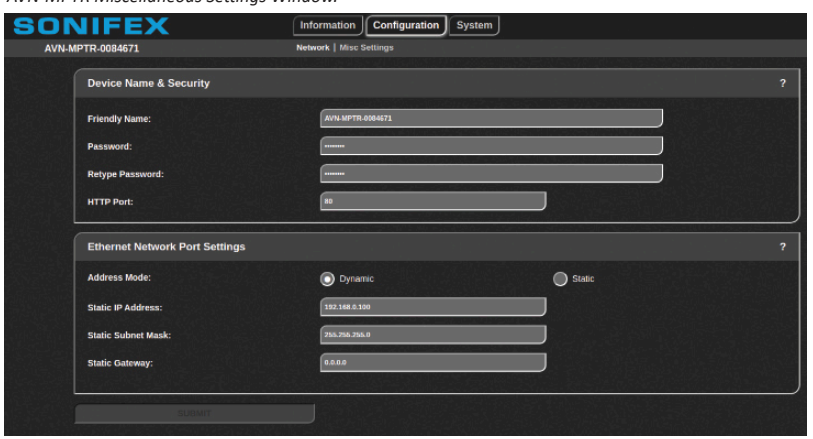
PoE Power	
Standard:	802.3af
Class:	3
PD Power Range:	0.44W to 12.94W
Typical PSE Power Usage:	7W
Max PSE Power Usage:	15.4W
Equipment Type	
AVN-MPTR	Technician Remote Controller
Physical Specification	
Dimensions: (Raw)	48cm (W) x 15.0cm(D) x4.2cm (H) (1U) 19"(W) x 5.9" (D) x1.7" (H) (1U)
Dimensions (Boxed):	56cm (W) x 29.5cm (D) x 16.5cm (H) 22" (W) x 11.6" (D) x 6.5" (H)
Weight:	Nett: 1.9kg Gross: 2.8kg Nett: 4.2lbs Gross: 6.2llbs



AVN-MPTR Device Information Window.

AVN-MPTR System Window.

AVN-MPTR Miscellaneous Settings Window.



AVN-MPTR Network Settings Window.

AVN-PX8X4C 8 x 4 Channel Mix Engine, 24 Inputs, 16 Outputs, AoIP AES67

Category: AES67/Dante AoIP Products.

Product Function: The mix engine houses the hardware where the complex routing, mixing and DSP functions are performed.

Typical Applications: Production galleries, control rooms and studios.

- 8 virtual mixers, each with 4 mono channel inputs to one mono output.
- Virtual mixers controlled by connected technician and presenter remote controls.
- Sophisticated configuration via webserver.
- 24 analogue inputs and 16 analogue outputs on D-type connectors.
- 32 logical inputs & 10 logical outputs using AoIP.
- Dual hot-swappable AC power supplies.
- 2 x 1Gb Ethernet ports & 1 x SFP port.
- Status LEDs and OLED display.
- 20 x configurable GPIO ports.

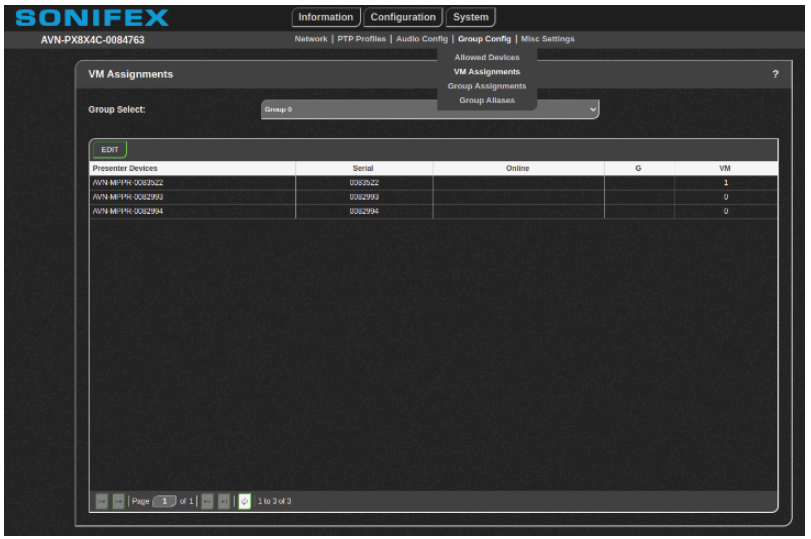
The presenter mixer system allows on-air presenters to adjust the audio levels fed to their radio or wired IEM systems, whilst also allowing Sound Control Room operators to adjust the levels remotely. The system consists of 3 types of unit all connected on a common Ravenna Audio over IP (AoIP) network with added audio sources derived from Ravenna based AoIP mixers or input portals. Setup and control can be provided via webpages and/or 3rd party applications. The system uses mono audio throughout.

The 3 units are:
AVN-PX8x4C - a 2U rack mounted Mix Engine Rack.
AVN-MPPR- a small desk, or microphone stand, mounted Presenters' Remote Unit.
AVN-MPTR - a 1U rack mounted Technicians' Remote Unit.

The AVN-PX8X4C Mix Engine, as well as using AES67 audio streams, accepts 24 analogue mono input channels on 3 x DB25 connectors and output 16 analogue mono channels on 2 x DB25 connectors. Each mixer unit is able to create and control 8 virtual mixers, each capable of mixing any 4 sources (analogue or AoIP) to a single mixed output that is available locally at an output and via an AoIP stream (to the Presenter Unit and other destinations). The analogue audio line-up level will default to +18dBU = 0dBFS.

A built-in webserver is used for configuration settings, input/output routing and for defining which Presenter and Technician Remote Units are assigned to the particular Mix Engine and Virtual Mixers. Up to 4 MPPRs can be assigned to each Virtual Mixer on a first-come first-served basis. It's also possible that both MPTRs could also select the same VM so a

AVN-PX8X4C Front View.



AVN-PX8X4C VM Assignments Window.

maximum of 6 devices can control the mix levels of a VM.

The Mix Engine will allow a maximum of 32 audio over IP input streams (each stream can be up to 8 channels). There are 8 virtual mixer outputs available. Each Virtual Mixer has 4 sources and 1 mixed output. There are also 2 more monitor outputs which play out the audio of the VM selection made on each of the 2 connected Technician Remote Units.

The Mix Engine has dual universal AC power supplies which are hot-swappable in the event of failure and have a green LED to indicate within spec operation. The front panel has LED indications and GPOs to warn of power supply failure. There are also front panel LEDs for network and sync/clock failure.

- 8 virtual mixers, each with 4 mono channel inputs to one mono output.
- Virtual mixers controlled by connected technician and presenter remote controls.
- Sophisticated configuration via webserver.
- 24 analogue inputs and 16 analogue outputs on D-type connectors.
- 32 logical inputs & 10 logical outputs using AoIP.
- Dual hot-swappable AC power supplies.
- 2 x 1Gb Ethernet ports & 1 x SFP port.
- Status LEDs and OLED display.
- 20 x configurable GPIO ports.

Technical Specification For AVN-PX8x4C

AVN-PX8x4C Audio Inputs – Gain setting 0dBFS = +18dBu unless otherwise stated

Parameter	Line Input
Input Impedance:	>20kΩ balanced
0dBFS line-up:	+15/+18/+20/+22/+24dBu Balanced*
Frequency Response:	20Hz to 20kHz, +0/-0.2dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, All input gain settings, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω
Crosstalk:	<-100dB
Common Mode Rejection:	>70dB @ 1kHz, All input gain settings

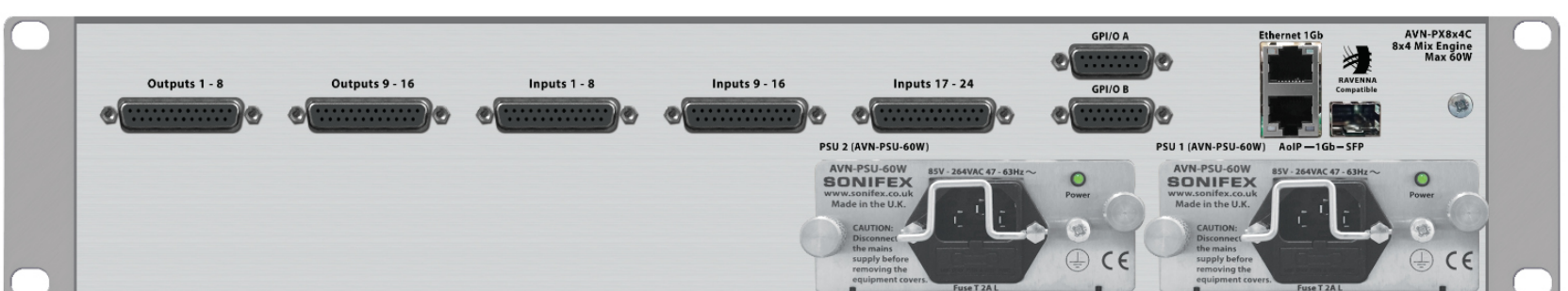
AVN-PX8x4C Audio Outputs – Gain setting 0dBFS = +18dBu unless otherwise stated

Parameter	Line Output
Output Impedance:	<50Ω balanced
0dBFS line-up:	+15/+18/+20/+22/+24dBu Balanced*
Frequency Response:	20Hz to 20kHz, +0/-0.5dB
THD+N:	<-110dBFS, -30dBFS, 20Hz to 20kHz, All input gain settings, 20kHz BW
Noise:	-110dBFS, 20kHz BW, Rs=200Ω

**+18dB, +22dB and +24dB are handled in hardware, +15dB and +20dB in DSP*

AVN-PX8x4C Power – Dual redundant IEC 85 - 264VAC 47 – 63Hz Max 60W 2Amp Fused type T 2A L

Equipment Type	
AVN-PX8X4C	8 x 4 Channel Mix Engine, 24 Inputs, 16 Outputs, AoIP AES67
Physical Specification	
Dimensions: (Raw)	48cm (W) x 27cm (D) x 8.8cm (H) (1U) 19" (W) x 10.6"(D) x 3.5" (H) (1U)
Dimensions (Boxed):	59cm (W) x 49cm (D) x 38cm (H) 23" (W) x 19" (D) x 15" (H)
Weight:	Nett: 5.7kg Gross:7.2kg Nett: 12.5lbs Gross:15.8llbs



AVN-PX8X4C Rear View.

SONIFEX

www.sonifex.co.uk

UK Office:
Sonifex Ltd
61 Station Road, Irthlingborough,
Northants, NN9 5QE, UK
Tel: +44 (0) 1933 650700
Fax: +44 (0) 1933 650726
Email: sales@sonifex.co.uk

Australian Office:
Sonifex Pty Ltd
12/6 Leighton Place,
Hornsby NSW 2077, Australia
Tel: +61 (2) 9987 0499
Fax: +61 (2) 9476 4950
Email: sales@sonifex.com.au

February 2023