SONIFEX

Manufacturers of Audio Products for AV, Installed Sound, Broadcast Radio & Broadcast TV

avn New Products

- PTP Grandmaster Clock
- AES67 Stream Mix Monitor
- Multi-Channel Audio Interfaces
- Headphone Distribution System
- Talkback Systems Using AES67
- Presenter In-Ear Monitoring System
- Passive Packs

AES67 now!

RAVENNA

EMBER+

AVN-GMCS PTPv2 Grandmaster Clock & GPS Receiver

AVN-GMCS

The AVN-GMCS is a PTPv2 grandmaster clock for use with RAVENNA & AES67 AoIP applications. IEEE1588-2008 PTPv2 (precision time protocol) is used to synchronise all the nodes within a network.

The AVN-GMCS becomes the master clock and distributes time packets using PTPv2 time-stamping to the other nodes on the network, performing this role simply and accurately, enabling sub micro second synchronisation between all nodes.

The unit is configured easily with a responsive embedded 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 three different types: TCXO, OXCO and CSAC (Chip Scale Atomic Clock, Caesium), which vary in both price and accuracy:

AVN-GMCS – TCXO Temp Compensated Oscillator accurate to 1 ppm (1 sec gain/loss per 11.5 days).

AVN-GMCOS – OCXO Oven Controlled Oscillator accurate to 0.1 ppm (1 sec gain/loss per 115 days).

AVN-GMCCS – SAC Quantum Atomic Clock accurate to 0.00050 ppm (1 sec gain/loss per 63 years).

GPS presence and the number of satellites received are shown on the front panel, together with status information on output sample rates, sync type and profile type. Some other features include:

- A screen-saver option which shows the current time.
- It can slave to a separate clock input.
- Clock outputs can be used to provide media clocks for external equipment. (A single AES-3id output and two outputs which can be selected as either word clock or variable PPS).
- The unit can show UTC or 'local time' on the front panel, by adding a time offset. Daylight saving time changes are accommodated.
- A real time clock allows an accurate date and time even after the unit is repowered without GPS access.
- Front panel LEDs show the synchronisation status, GPS lock and the status of the AC and DC power supply inputs.
- OLED display and LED indicators brightness adjustment.
- 4 GPO indicate critical states for GPS lock status, external sync present, AC power present and DC power present.
- Dual redundant power inputs - an IEC mains input and a 12V DC input.
- Low-power sleep mode available.
- In power off situations, a super capacitor keeps the GPS receiver in a low power mode for more than 20 hours, enabling the receiver to regain lock immediately rather than having to 'cold' start.

AVN-PXH12 12 x 2 Channel AES67 Stream Mix Monitor

AVN-PXH12

With audio moving to the AoIP network infrastructure allowing hundreds of audio sources to be available on the network, how do you easily monitor them, in a product which is simple to use and without adding complexity?

The AVN-PXH12 is a 24 x AES67 stream input mixer monitor in a 1U rack frame.

The main benefit over traditional monitors is the speed with which sources can be monitored. Each unit has front panel headphone outputs and a speaker, together with rear panel analogue outputs. There are 12 x mini channel-strips along the front panel, each with a translucent rotary encoder showing confidence monitoring of the input level in the knob itself and output mix level in the LEDs around the encoder. Three buttons for each encoder can be used to select the main/secondary input, to mute the channel and to send the audio of that channel to left, right or stereo mix of the output. Pressing the encoder knob lets you solo the channel. With these simple controls, a mix of any of 24 channels can be made quickly and intuitively, ideal for live news environments where audio sources are changing rapidly and need to be monitored instantly.

As well as monitoring any AES67 AoIP stream, SAP discovery has been added to the unit so that AES67 Dante® streams can also be mixed and monitored. Additionally, Ember+ is used for the control communication allowing remote control of the product using the open Ember+ standard.

Mix Monitor Features:

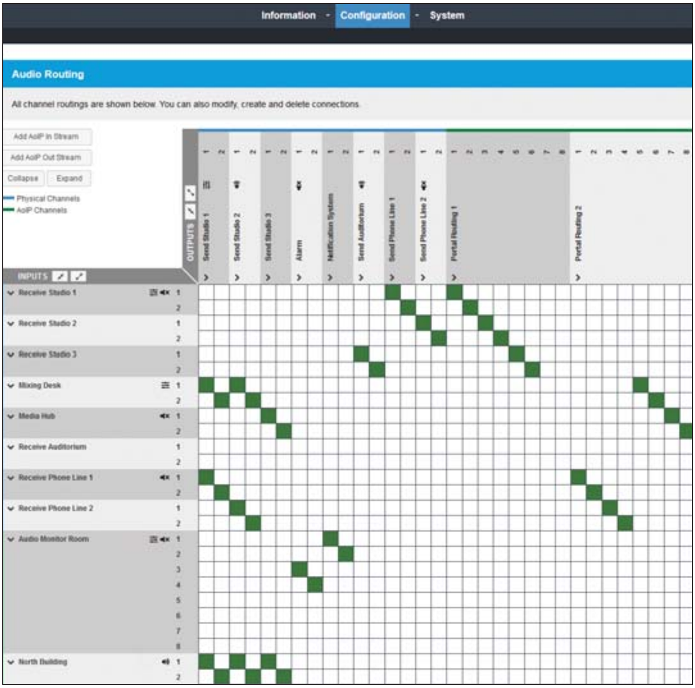
- AES67 is an established AoIP stream format – the unit uses RAVENNA audio to ensure AES67 compatibility.
- A built-in web server is used for all configuration. Sources for all channels are simply assigned on one webpage.
- SAP is used as a discovery mechanism to discover Dante® devices and monitor Dante® 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.
- 6.35mm (1/4") & 3.5mm headphone outputs and a speaker output with separate LS & HP volume controls.
- Speaker mute button.
- Sources from AoIP, balanced or 3 x unbalanced inputs.
- Destinations to AoIP or 3 rear panel balanced outputs.
- 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.
- Dual 1Gb lan ports & 1Gb SFP fibre port.
- 10 user assignable GPIO ports as inputs or outputs.
- Dual redundant AC & DC power supply inputs.

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

These are a range of 3 audio interface portals which mix and route analogue, AES3, microphone & AES67 stream inputs to analogue, AES3 & AES67 outputs.

Analogue	**AVN-PA8, 8 x Stereo Analogue Line Inputs & 8 x Stereo Analogue Outputs**
AES3 Digital	**AVN-PD8, 8 x Stereo AES3 Digital Inputs & 8 x Stereo AES3 Outputs**
Microphone	**AVN-PM8, 8 x Mic/Line Inputs & 8 x Stereo Analogue Outputs**

Each of the portals can be ordered either with the input/output metering displays (D version) or without and there are two options for rear panel connectivity - with XLR/RJ45 connectors or terminal block (T version) connectors.



'Audio Routing' Webpage

The heart of each portal is the web-enabled mix engine. Any physical input and stream input can be mixed or routed to any physical output or stream output, with gain alteration at the input, the mix point or the output.

Up to 16 x AES67 input channels and 64 x AES67 output channels can be created in each portal, supporting the full range of AES67 packet times and channel counts.

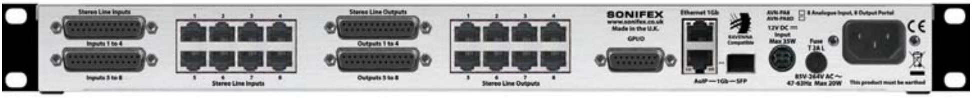
The portals' versatility allows them to be used for any applications where mixing of signals is required, e.g. an 8 channel cleanfeed generator, to send 64 streams of IFB to connected belt-packs, or as an 8 output zone mixer.

AVN Portal Features:

All of the portals have the following feature-set:

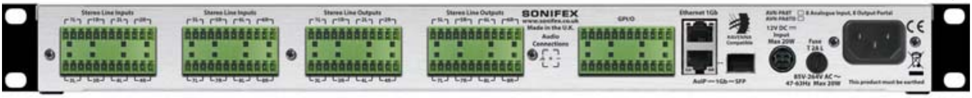
- Responsive webserver software mixer/router to mix any input to any output.
- 'T' version has audio I/O on terminal blocks.
- 'D' version has input & output metering on bright front panel displays, with 9 metering types and channel idents.
- Up to 8 AoIP AES67 input streams with a maximum of 16 input channels to be routed.
- Up to 8 AoIP output channels with a maximum of 8 channels each, providing up to 64 stream outputs.
- Input/output full-scale line-up.
- Input and output gain adjustment.
- Output connections capable of supplying analogue power to satellite headphone amplifiers, the AVN-HA1 and AVN-HD1.
- SNMP V2 - Gets, Sets & sending traps.
- Ember+ remote control.
- Dual 1Gb Ethernet & 1Gb SFP ports.
- Dual AC & DC power supply inputs.
- 10 user assignable GPIO ports and relay output.

AVN-PA8D 8 x Stereo Analogue Line Inputs & Outputs, AES67 Display Portal



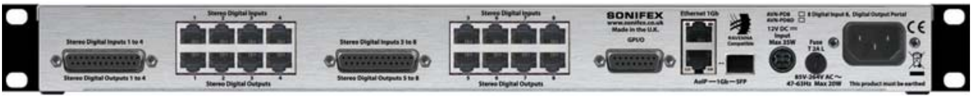
AVN-PA8/D

AVN-PA8T/D



- 8 stereo line inputs and 8 stereo line outputs on D-type sockets with AES59 analogue pinout, paralleled with 16 x RJ45 connectors using StudioHub® pinout.

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

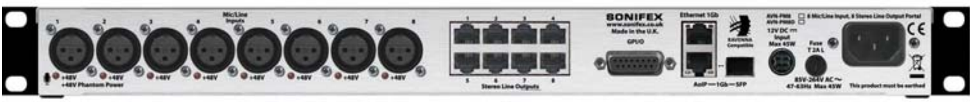


AVN-PD8/D

AVN-PD8T/D

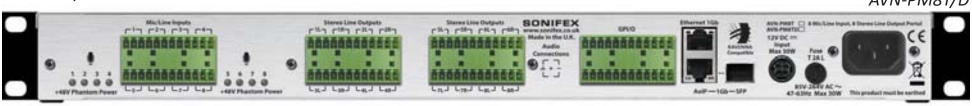
- 8 x stereo digital AES3 inputs and 8 x stereo digital AES3 outputs on D-type sockets with AES59 pinout, paralleled with 16 x RJ45 connectors using StudioHub® pinout.
- Individual input sample rate conversion.

AVN-PM8 8 x Mic/Line Inputs & 8 x Line Outputs, AES67 Portal



AVN-PM8/D

AVN-PM8T/D



- 8 x mic/line inputs on XLR sockets. 8 stereo line outputs on D-type sockets with AES59 analogue pinout, paralleled with 8 x RJ45 connectors using StudioHub® pinout.
- 8 x 3mm red LED phantom presence indications.
- Additional mic pre-amp gain adjustment.

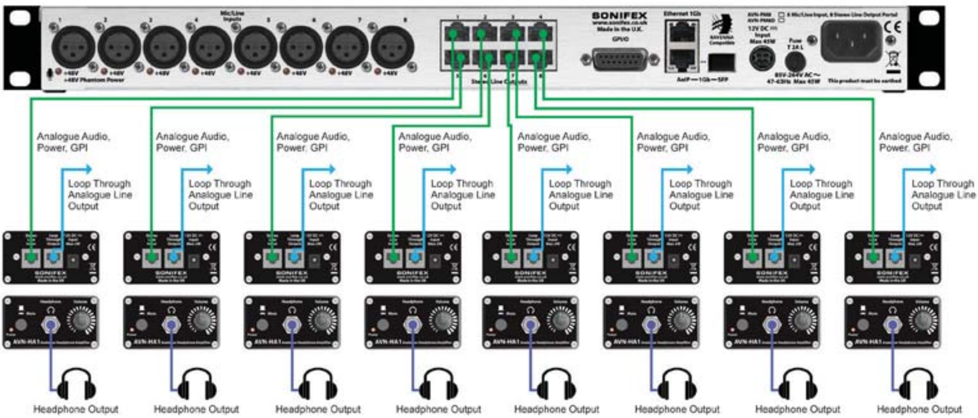
Headphone Distribution System

The AVN Portals can be combined with the Sonifex AVN-HA1 and AVN-HD1 headphone amplifiers to create a headphone distribution system where each headphone amplifier can be sent a separate feed, mixed from any physical or AES67 stream inputs.

On portal units with RJ45 outputs, an AVN-HA1 (for the AVN-PA8 and AVN-PM8) or AVN-HD1 (for the AVN-PD8) headphone amplifier can be used to listen to the outputs, with the portals providing power and audio signals.

The switches on the front panel of the AVN-HA1 and AVN-HD1 can be used as another GPI for example, for muting the output.

Headphone Distribution System Using Analogue Portal & 8 x AVN-HA1 Units



AVN-HA1

Analogue Headphone Amplifier for AVN-PA8/D & AVN-PM8/D Portals

- Front panel 6.35mm (1/4”) headphone socket and volume control knob, with 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.

AVN-HD1

Digital Headphone Amplifier for AVN-PD8/D Portals

- Front panel 6.35mm (1/4”) headphone socket and volume control knob, with mute/GPO push button.
- 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).
- Locking DC power connector if a portal is not being used to supply the unit with power.

Talkback Intercom System Using AES67 AoIP

The new range of AVN talkback/listening/paging intercoms aid communication between studios, stages, theatres, offices and different areas in a facility or building complex.

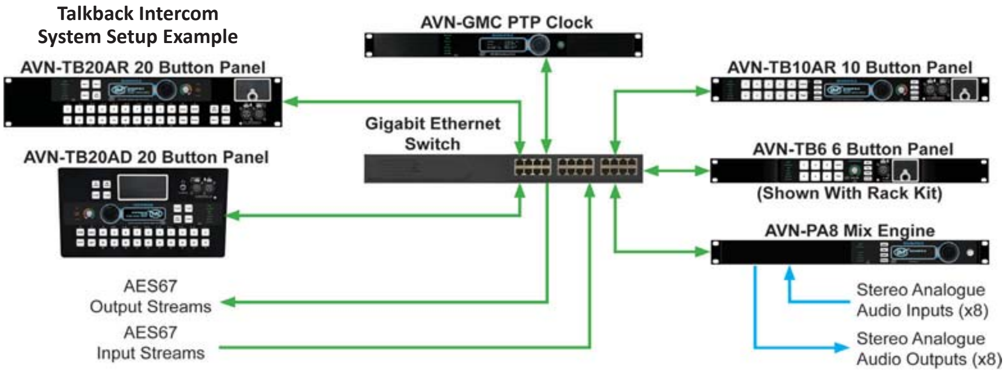
The system doesn't use a central router – each unit is intelligent and can talk to other intercoms connected to the same network.

With both 4-wire analogue inputs and outputs, as well as AES67 AoIP network audio connectivity, the AVN-TB units can be used with existing legacy 4-wire systems and with new AES67 AoIP infrastructure.

All audio is at 48kHz sample rate, meaning that it's broadcast quality audio as standard.

Also, connections can be made to AES67 streams, as well as Sonifex devices, with GPIO or VGPIO tallies used for signalling.

Simple responsive webserver configuration with Ember+ remote control.



AVN-TB6 6 Button Talkback Intercom

This is a 6 station version for smaller talkback systems, or for areas with reduced communications needs.

- 6 illuminated key-cap Talk buttons plus Listen & Page buttons.
- 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.
- Dual 1Gb Ethernet & 1Gb SFP port.

AVN-TB6

- Responsive design Ethernet webserver.
- AVN-TB6RK 19" rack kit available.

AVN-TB6MC 6 Button Talkback Intercom With Monitor Controller

The AVN-TB6MC is similar to the AVN-TB6 but with the addition of a configurable audio monitor output and dedicated front panel DIM and MUTE monitor controls. The DIM and MUTE buttons are controllable remotely via GPI, VGPI or Ember+ commands.

AVN-TB6MC

AVN-TB6D 6 Button Desktop Talkback Intercom

The AVN-TB6D is a desktop version of the AVN-TB6 intercom with a smaller form factor and an elegant sloped front. It has the same feature set and connectivity.

AVN-TB6D

- 6 illuminated key-cap Talk buttons.
- Listen & Page buttons.
- User definable button.
- Speaker & microphone mute buttons.
- Mic & headset inputs, with +48V phantom power for the mic inputs.
- Headphone & speaker outputs.
- Front panel volume control which operates on speaker headphone outputs and incoming source levels.
- Sources and destinations from/to AoIP.
- Advanced echo cancellation & mic AGC to prevent acoustic feedback.
- Front panel monitor button for routing audio directly to the speaker e.g. to take an IFB feed or off-air transmission signal.
- Ducking or mixing of inputs.

AVN-TB20AR 20 Button Advanced Rackmount Talkback Intercom

The AVN-TB20AR is a 2U rackmount version of the AVN-TB10AR with the same specification, but 20 station buttons allowing greater communication for larger facilities.

AVN-TB20AR

- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Group Talk with up to 3 user defined groups.
- Phone button for remote dialling and control of an external telephone hybrid.
- Dual 1Gb Ethernet & 1Gb SFP port.
- Mic & headset inputs (front & rear panel headset connection), headphone & speaker outputs.
- Dual AC & DC power supply inputs.
- Sources from AoIP, 1 x balanced, 2 x unbalanced and S/PDIF digital inputs.

AVN-TB20AD 20 Button Advanced Desktop Talkback Intercom

AVN-TB20AD

The AVN-TB20AD is a 20 channel desktop version of the AVN-TB20AR talkback intercom.

- 20 illuminated key-cap Talk buttons plus Listen & Page buttons.
- 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.
- Advanced echo cancellation & mic AGC to prevent feedback.

AVN-TB10AR 10 Button Advanced Rackmount Talkback Intercom

AVN-TB10AR

The AVN-TB10AR is the flagship of the AVN-TB range providing an advanced feature-set with unparalleled audio quality. A superb acoustic echo cancellation algorithm allows units to be placed next to each other with open mics and no feedback. Low latency, broadcast quality audio comes as standard, using RAVENNA AES67.

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, a Talk with Forced 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 talkback source can also be any stream on the network, using GPIO and VGPIO for call control, if required. Also, a new 4W Bridge Mode offers an alternative to normal intercom use, allowing audio equipment connected to local inputs and outputs to communicate with remote devices assigned to the channel buttons.

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 panel rotary encoder volume control.

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

Microphone mute (cough) and speaker mute buttons can be controlled remotely by GPI or Ember+.

Each unit has a built-in webserver which is where the majority of settings and configurations are made. The webserver is a responsive design meaning that it can be used with small screens on smartphones and tablets. Front panel LEDs show the AoIP network status, synchronisation status, whether AGC is being used and the status of the AC and DC power supplies.

AVN-TB10AR Features:

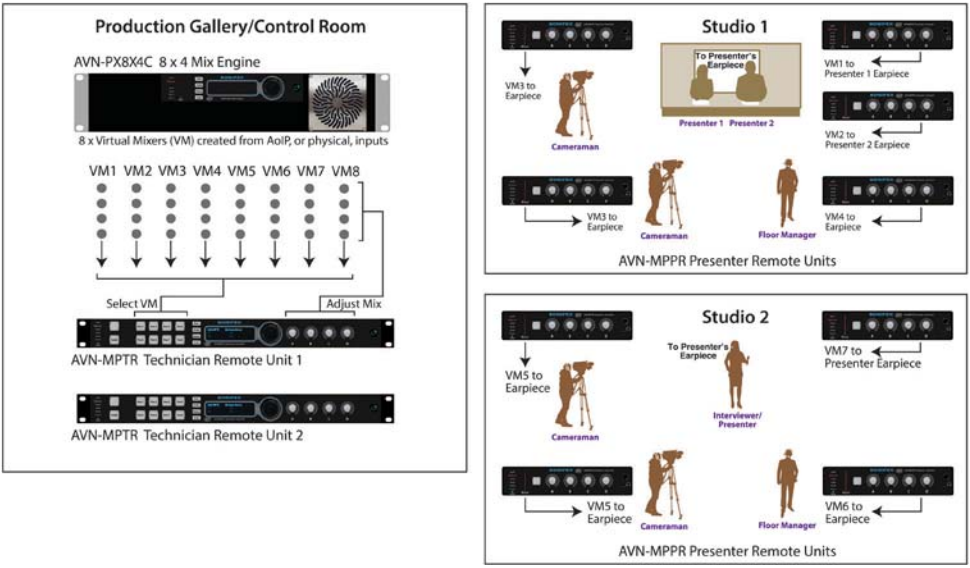
- 10 illuminated key-cap Talk buttons plus Listen & Page buttons.
- Mic & headset inputs, headphone & speaker outputs with volume control.
- Sources from AoIP, balanced, 2 x unbalanced or S/PDIF digital inputs.
- Destinations to AoIP or rear panel balanced & unbalanced outputs.
- Advanced echo cancellation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

Presenter In-Ear Monitoring System Using AES67 AoIP

An 8 x 4 channel mix engine forms the central core of a powerful presenter in-ear monitoring system, allowing up to 32 Presenter Remote controllers to alter the mix output levels of 1 of 8 virtual mixers (VMs), each with 4 mono channels.

Technician Remote units can oversee all VMs, adjusting the level of the 4 mono inputs of each VM in case presenters are busy, or for total remote operation.

In-Ear Monitoring System Overview 1



AVN-MPPR 4 Channel Presenter In-Ear Monitoring Remote Controller, AES67

AVN-MPPR

The presenter remote controller 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 can switch between visual feedback for both mix level of the source and actual input level by pressing the encoder.

Up to 32 units can be connected to a mix engine. 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 units using AES67 AoIP with a single PoE Neutrik ™ Ethercon network connection. A front panel button can be used to actuate one of the ten GPIO pins.

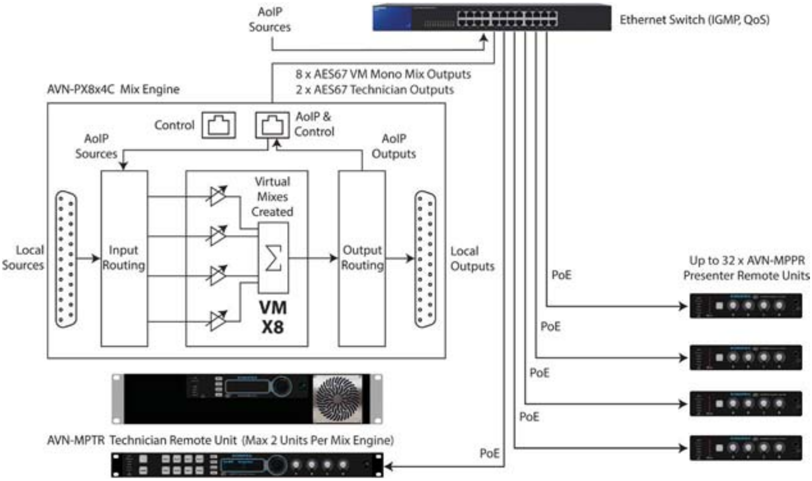
Units can be mounted on a standard microphone screw thread or under a desktop using optional accessories.

AVN-MPTR Technician Remote Controller

AVN-MPTR

- Allows selection of each of 8 x VMs and remote adjustment of mix output volumes.
- 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.
- 2 x AVN-MPTR allowed per mix engine.

In-Ear Monitoring System Overview 2



AVN-PX8X4C 8 x 4 Channel Mix Engine, 24 Inputs, 16 Outputs, AES67

AVN-PX8X4C

The mix engine houses the hardware where the complex routing, mixing and DSP functions are performed. This resilient 2U rack has dual hot-swappable AC power supplies and both 1Gb Ethernet & SFP ports.

- 8 virtual mixers, each with 4 mono channel inputs to one mono output AES67 stream.
- Virtual mixers controlled by connected technician and presenter remote controls.
- 24 analogue inputs and 16 analogue outputs on D-type connectors.
- 32 logical inputs & 10 logical outputs using AoIP.
- Sophisticated configuration via webserver.
- 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.

Passive Packs

The passive packs are a range of small utility products which use transformers to change audio input & output impedances and levels and to distribute audio signals.

Headphone Amplifiers

CM-HPR1 Headphone Volume Control

This allows the connection of a stereo balanced line input via an RJ45 with StudioHub+™ pinout and creates a headphone output, with level control. It is transformer balanced, and can be used with any headphones above 150Ω impedance.

CM-HPR1

CM-HPX1 Headphone Volume Control

This allows the connection of a stereo balanced line input via 2 x female XLRs and creates a headphone output, with level control.

CM-HPX1

Matching Amplifiers

CM-LUR1 Balanced to Unbalanced Audio Converter

This allows the connection of a stereo balanced line input via RJ45 with StudioHub+™ pinout to a stereo unbalanced output on stereo phono connectors.

CM-LUR1

CM-LUX1 Balanced to Unbalanced Audio Converter

This allows the connection of a stereo balanced line input via 2 x female XLRs to a stereo unbalanced output on phono connectors.

CM-LUX1

CM-ULR1 Unbalanced to Balanced Audio Converter

This allows the connection of a stereo unbalanced line input on phono connectors to a balanced line output via a RJ45 with StudioHub+™ pinout.

CM-ULR1

CM-ULX1 Unbalanced to Balanced Audio Converter

This allows the connection of a stereo unbalanced line input on phono connectors to a balanced line output via 2 x male XLRs.

CM-ULX1

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