HANDBOOK

AVN-TB10AR

10 Button Advanced Talkback Intercom, AoIP Portal

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

SONIFEX

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This handbook is for use with the following product:
AVN-TB10AR 10 Button Advanced Talkback Intercom, AoIP Portal
AW10847A, Stock Code: 30-296

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Sonifex Ltd, 61, Station Road, Irthlingborough,
Northants, NN9 5QE, England.
Tel: +44 (0)1933 650 700
Fax: +44 (0)1933 650 726
Email: sales@sonifex.co.uk
Website: http://www.sonifex.co.uk

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SONIFEX

Register Online for an Extended 2 Year Warranty

As standard, Sonifex products are supplied with a 1 year back to base warranty.

If you register the product online, you can increase your product warranty to 2 years and we can also keep you informed of any product design improvements or modifications.

Product: _____

_____

Serial No: _____

To register your product, please go online to www.sonifex.co.uk/register

SONIFEX Sonifex Limited • 61 Station Road • Irthlingborough • Northamptonshire • NN9 5QE • United Kingdom
Tel: +44 (0)1933 650 700 • Fax: +44 (0)1933 650 726 • Email: technical.support@sonifex.co.uk • Internet: www.sonifex.co.uk

Product Warranty - 2 Year Extended

As standard, Sonifex products are supplied with a 1 year back to base warranty. In order to register the date of purchase and so that we can keep you informed of any product design improvements or modifications, it is important to complete the warranty registration online. Additionally, if you register the product on the Sonifex website, you can increase your product warranty to 2 years. Go to the Sonifex website at: www.sonifex.co.uk/register to apply for your 2 year warranty.

Sonifex Warranty & Liability Terms & Conditions

1. Definitions

'the Company' means Sonifex Ltd and where relevant includes companies within the same group of companies as Sonifex Limited.

'the Goods' means the goods or any part thereof supplied by the Company and where relevant includes: work carried out by the Company on items supplied by the Purchaser; services supplied by the Company; and software supplied by the Company.

'the Purchaser' means the person or organisation who buys or has agreed to buy the Goods.

'the Price' means the Price of the Goods and any other charges incurred by the Company in the supply of the Goods.

'the Warranty Term' is the length of the product warranty which is usually 12 months from the date of despatch; except when the product has been registered at the Sonifex website when the Warranty Term is 24 months from the date of despatch.

'the Contract' means the quotation, these Conditions of Sale and any other document incorporated in a contract between the Company and the Purchaser.

This is the entire Contract between the parties relating to the subject matter hereof and may not be changed or terminated except in writing in accordance with the provisions of this Contract. A reference to the consent, acknowledgement, authority or agreement of the Company means in writing and only by a director of the Company.

2. Warranty

a. The Company agrees to repair or (at its discretion) replace Goods which are found to be defective (fair wear and tear excepted) and which are returned to the Company within the Warranty Term provided that each of the following are satisfied:

   i. notification of any defect is given to the Company immediately upon its becoming apparent to the Purchaser;

   ii. the Goods have only been operated under normal operating conditions and have only been subject to normal use (and in particular the Goods must have been correctly connected and must not have been subject to high voltage or to ionising radiation and must not have been used contrary to the Company's technical recommendations);

   iii. the Goods are returned to the Company's premises at the Purchaser's expense;

   iv. any Goods or parts of Goods replaced shall become the property of the Company;

   v. no work whatsoever (other than normal and proper maintenance) has been carried out to the Goods or any part of the Goods without the Company's prior written consent;

   vi. the defect has not arisen from a design made, furnished or specified by the Purchaser;

vii. the Goods have been assembled or incorporated into other goods only in accordance with any instructions issued by the Company;

viii. the defect has not arisen from a design modified by the Purchaser;

ix. the defect has not arisen from an item manufactured by a person other than the Company. In respect of any item manufactured by a person other than the Company, the Purchaser shall only be entitled to the benefit of any warranty or guarantee provided by such manufacturer to the Company.

b. In respect of computer software supplied by the Company the Company does not warrant that the use of the software will be uninterrupted or error free.

c. The Company accepts liability:

(i) for death or personal injury to the extent that it results from the negligence of the Company, its employees (whilst in the course of their employment) or its agents (in the course of the agency);

(ii) for any breach by the Company of any statutory undertaking as to title, quiet possession and freedom from encumbrance.

d. Subject to conditions (a) and (c) from the time of despatch of the Goods from the Company's premises the Purchaser shall be responsible for any defect in the Goods or loss, damage, nuisance or interference whatsoever consequential economic or otherwise or wastage of material resulting from or caused by or to the Goods. In particular the Company shall not be liable for any loss of profits or other economic losses. The Company accordingly excludes all liability for the same.

e. At the request and expense of the Purchaser the Company will test the Goods to ascertain performance levels and provide a report of the results of that test. The report will be accurate at the time of the test, to the best of the belief and knowledge of the Company, and the Company accepts no liability in respect of its accuracy beyond that set out in Condition (a).

f. Subject to Condition (e) no representation, condition, warranty or other term, express or implied (by statute or otherwise) is given by the Company that the Goods are of any particular quality or standard or will enable the Purchaser to attain any particular performance or result, or will be suitable for any particular purpose or use under specific conditions or will provide any particular capacity, notwithstanding that the requirement for such performance, result or capacity or that such particular purpose or conditions may have been known (or ought to have been known) to the Company, its employees or agents.

g. (i) To the extent that the Company is held legally liable to the Purchaser for any single breach of contract, tort, representation or other act or default, the Company's liability for the same shall not exceed the price of the Goods.

(ii) The restriction of liability in Condition (g)(i) shall not apply to any liability accepted by the Seller in Condition (c).

h. Where the Goods are sold under a consumer transaction (as defined by the Consumer Transactions (Restrictions on Statements) Order 1976) the statutory rights of the Purchaser are not affected by these Conditions of Sale.

Unpacking Your Product

Each product is shipped in protective packaging and should be inspected for damage before use. If there is any transit damage take pictures of the product packaging and notify the carrier immediately with all the relevant details of the shipment. Packing materials should be kept for inspection and also for if the product needs to be returned.

The product is shipped with the following equipment so please check to ensure that you have all of the items below. If anything is missing, please contact the supplier of your equipment immediately.

Item	Quantity
Product unit	1
IEC mains lead fitted with moulded mains plug	1
Handbook	1

If you require a different power lead, please let us know when ordering the product.

Repairs & Returns

Please contact Sonifex or your supplier if you have any problems with your Sonifex product. Email technical.support@sonifex.co.uk for the repair/upgrade/returns procedure, or for support & questions regarding the product operation.

CE Conformity

The products in this manual comply with the essential requirements of the relevant European health, safety and environmental protection legislation.

The technical justification file for this product is available at Sonifex Ltd.

The declaration of conformity can be found at:
http://www.sonifex.co.uk/declarations

Safety & Installation of Mains Operated Equipment

There are no user serviceable parts inside the equipment. If you should ever need to look inside the unit, always disconnect the mains supply before removing the equipment covers. The cover is connected to earth by means of the fixing screws. It is essential to maintain this earth/ground connection to ensure a safe operating environment and provide electromagnetic shielding.

Voltage Setting Checks

Ensure that the machine operating voltage is correct for your mains power supply by checking the box in which your product was supplied. The voltage is shown on the box label. The available voltage settings are 115V, or 230V. Please note that all products are either switchable between 115V and 230V, or have a universal power supply.

Fuse Rating

The product is supplied with a single fuse in the live conducting path of the mains power input. For reasons of safety it is important that the correct rating and type of fuse is used. Incorrectly rated fuses could present a possible fire hazard, under equipment fault conditions. The active fuse is fitted on the outside rear panel of the unit.

Power Cable & Connection

An IEC power connector is supplied with the product which has a moulded plug attached.

The mains plug or IEC power connector is used as the disconnect device. The mains plug and IEC power connector shall remain readily operable to disconnect the apparatus in case of a fault or emergency.

The mains lead is automatically configured for the country that the product is being sent to, from one of:

Territory	Voltage	IEC Lead Type	Image
UK & Middle East	230V	UK 3 pin to IEC lead	
Europe	230V	European Schuko round 2 pin to IEC lead	
USA, Canada and South America	115V	3 flat pin to IEC lead	
Australia & New Zealand	230V	Australasian 3 flat pin to IEC lead	

Connect the equipment in accordance with the connection details and before applying power to the unit, check that the machine has the correct operating voltage for your mains power supply.

This apparatus is of a class I construction. It must be connected to a mains socket outlet with a protective earthing connection.

Important note: If there is an earth/ground terminal on the rear panel of the product then it must be connected to Earth.

WEEE Directive

The Waste Electrical and Electronic Equipment (WEEE) Directive was agreed on 13 February 2003, along with the related Directive 2002/95/EC on Restrictions of the use of certain Hazardous Substances in electrical and electronic equipment (RoHS). The Waste Electrical and Electronic Equipment Directive (WEEE) aims to minimise the impacts of electrical and electronic equipment on the environment during their life times and when they become waste. All products manufactured by Sonifex Ltd have the WEEE directive label placed on the case. Sonifex Ltd will be happy to give you information about local organisations that can reprocess the product when it reaches its “end of use”, or alternatively all products that have reached “end of use” can be returned to Sonifex and will be reprocessed correctly free of charge.

Atmosphere/Environment

This apparatus should be installed in an area that is not subject to excessive temperature variation (<0°C, >50°C), moisture, dust or vibration.

This apparatus shall not be exposed to dripping or splashing, and no objects filled with water, such as vases shall be placed on the apparatus.

1. AVN-TB10AR 10 Button Advanced Talkback Intercom, AoIP Portal

Introduction

Fig 1-1: The AVN-TB10AR Front Panel

The AVN-TB10AR is a 10 channel talkback intercom control unit from the Sonifex range of IP based AVN (Audio/Video/Network) products. It 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.

The AVN-TB10AR is an advanced feature set unit. Each of the 10 channels can be configured to provide communications with other remote networked intercom units or AoIP streams, or with local 4-wire systems. It has analogue and digital audio inputs that can be used to distribute audio over a network to other RAVENNA/AES67 capable systems, and it has analogue outputs for monitoring of remote sources. It can also be used as a 4 wire bridge, connecting remote AoIP sources with local audio inputs and outputs.

A call back system monitors missed intercom call requests and makes it easy to make a return call, and an independently configurable page function can contact selected units with priority over standard intercom calls if required.

There are 2 independent monitor channels that can route the audio from a local input or AoIP source to the headphones and speaker. This could be used to take an IFB feed, 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 unit and also allows for firmware updates and configuration backup. An Ember+ interface also gives access to the unit's configuration as well as providing remote control and monitoring of the GPIO and virtual GPIO ports.

The front panel OLED display with navigation control gives access to the configuration settings and also provides status and connectivity information.

2. Installation

Network Connection

The AVN-TB10AR has 2 separate gigabit network ports:

Upper Port: Configuration and control interface
Lower Port: Audio over IP (AoIP) interface

In an ideal installation, the lower port should be connected to a dedicated network switch that handles audio and audio related traffic only. All other intercom units, AoIP sources and destinations, and grandmaster clocks should be connected to this network. The upper port can then be connected to a separate network to provide access to the embedded web server. With this setup, potentially heavy network traffic on a company LAN does not affect the performance of the AoIP network.

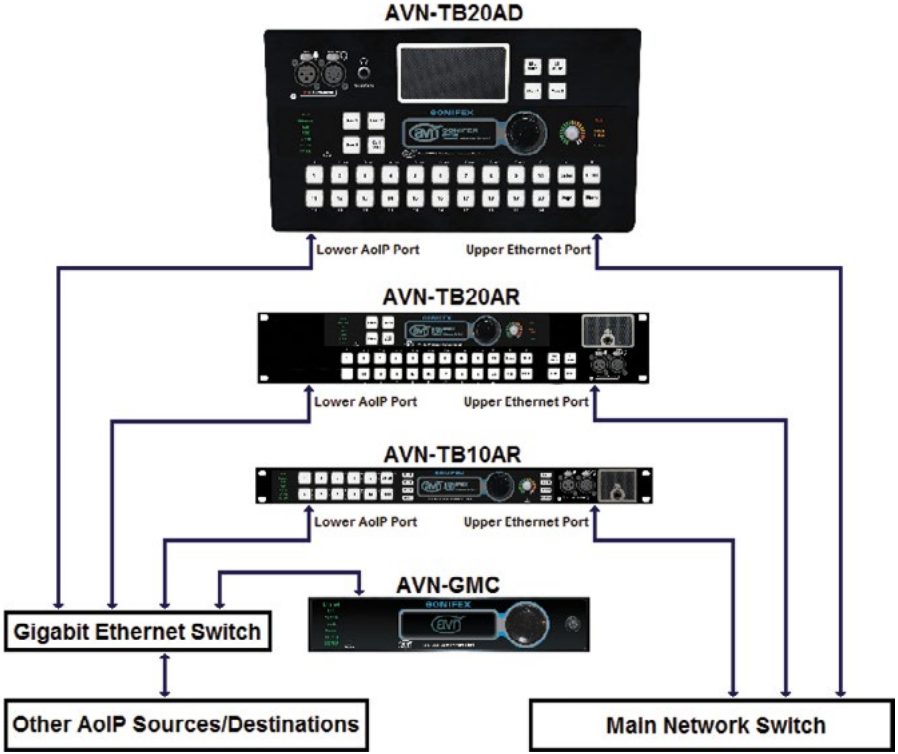

Fig 2-1: Preferred Network Topology

The AVN-TB10AR and any other Sonifex talkback intercom can be used with a single network connection using the lower (AoIP) port as this also provides access to the embedded web server. The talkback intercom units, AoIP sources and destinations, and grandmaster clocks should still ideally be connected to a dedicated network switch.

Microphone

The AVN-TB10AR has connections for a microphone via a 3 pin XLR on the front panel, or a headset via a 5 pin XLR on the front or rear panel. Only one microphone or headset should be connected. If phantom power is required, this can be enabled via the webserver. See Audio - Microphone Settings web page on page 39 for details.

Audio Inputs

Audio inputs connected to the unit can be monitored locally via the headphones or speaker. They can also be routed to the network as an AoIP stream so that other units can listen to the audio. There are 2 un-balanced stereo inputs, 1 balanced stereo input that can also be configured as a mono input/output, and a digital S/PDIF input. Connection details are shown on page 8. Any combination of audio inputs can be used and the routing of audio to an AoIP stream can be muted as required via the webserver. See AoIP Streams web page on page 40 for details.

Audio Outputs

Audio outputs can be sourced from AoIP streams or any of the local audio inputs including the microphone. There is 1 balanced and 1 un-balanced stereo output. There is also a balanced mono output available. Connection details are shown on page 8. The audio routed to the outputs can be muted at any time via the webserver. See Audio - Output Settings web page on page 41 for details.

GPIO

The GPIO connector provides 10 ports, individually configurable as either inputs or outputs, via a 15 way ‘D’ type connector. Connection details are shown on page 8. Outputs are open collector driven, and inputs are active low. See GPIO Settings web page on page 45 for details.

Power Supply

The unit can be powered from the mains via the supplied IEC lead, or by using the optional external 12V DC power supply, part number AVN-DC150. Please contact Sonifex Ltd sales for more details. Both power supplies can be connected and powered at the same time to provide redundancy if required.

3. Front Panel Controls, Connections & Indicators

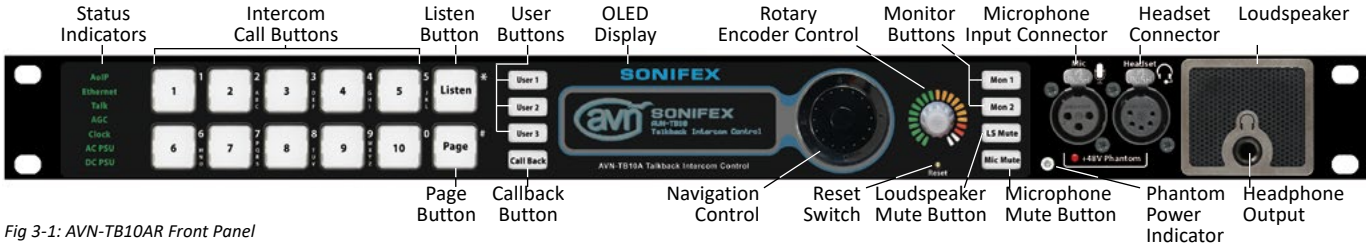

Fig 3-1: AVN-TB10AR Front Panel

Status Indicators

The RGB status indicators are located on the left side of the front panel.

AoIP

Indicates the status of AoIP services on the network as follows:

Green	All transmit AoIP streams are available and at least one external AoIP service is available.
Amber	Either the transmit AoIP streams have not started or there are no external AoIP services available.
Off	The transmit AoIP streams have not started and there are no external AoIP services available. The indicator will also be off if there is no link on the AoIP network port (lower network connector).

The AoIP network interface is configured via the embedded web server. See the Network web page on page 27.

Ethernet

Indicates the link status of the Ethernet network port (upper network connector) as follows:

Green	The link is up.
Off	The link is down.

The Ethernet network interface is configured via the embedded web server. See the Network web page on page 27.

Talk

Illuminates green if an intercom call or page is active.

AGC

Indicates the status of the automatic gain control as follows:

Green	The AGC is enabled.
Red	The audio source level for one of the outputs is too high and is being limited. The monitored outputs are outputs 1-3 on the rear panel, the headphone/headset output and the speaker output.

Clock

Indicates the status of PTP clock as follows:

Green	The unit is a PTP slave with a clock offset of less or equal to ±1µs or the unit is the PTP master.
Amber	The unit is a PTP slave with a clock offset of between ±1µs and ±10µs.

Flash Amber	The unit is in a listening state and is not yet acting as the master or a slave.
Red	The unit is a PTP slave with a clock offset of greater than ±10µs.
Flash Red	The unit is a PTP slave and no master is available.

AC PSU

Indicates the status of the internal AC-DC power supply as follows:

Green	AC-DC generated voltage is within normal operating range of +11V to +13V. This is the normal condition.
Amber	AC-DC generated voltage is less than +11V or greater than +13V. This indicates a warn condition.
Red	AC-DC generated voltage is less than +10V or greater than +14V. This indicates a fault condition.

DC PSU

Indicates the status of the DC power input as follows:

Green	DC voltage is within normal operating range of +11V to +13V. This is the normal condition.
Amber	DC voltage is less than +11V or greater than +13V. This indicates a warn condition.
Red	Not connected or DC voltage is less than +10V or greater than +14V. This indicates a fault condition if a DC power source is connected.

Channel 1 to 10 Intercom Call Buttons

These buttons initiate an intercom call to a pre-configured target intercom unit or AoIP stream. The channels are configured via the embedded web server. See the Channels - Advanced Settings web page on page 34.

Listen Button

This button is used when making an intercom call to change the default call type. See Listen Button on page 13 for more details.

Page Button

The page button initiates a talk only call to a pre-configured list of target intercom units. The button flashes red when active. The target intercoms are selected via the embedded web server. See the Page Settings web page on page 43.

User 1 to 3 Buttons

These buttons are configurable and can be used to control various functions on the unit. To configure these buttons, see the Front Panel - User Buttons web page in on page 38.

Call Back Button

The call back button provides access to the missed call list and allows calls to be easily returned. The button illuminates green when active. See Call Back on page 18 for details.

OLED Display

The OLED display shows the current call and page status of the unit.

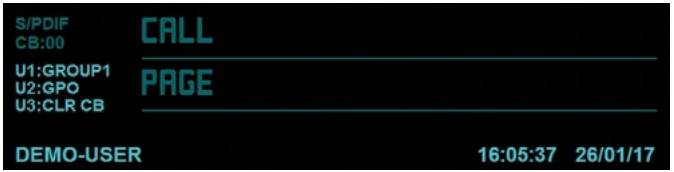

Fig 3-2: Main Screen in the Idle State

It also shows the S/PDIF digital input lock status and the call back count in the upper left side. Below the call back count are the assignments for the 3 user buttons, U1 to U3. The intercom name is shown at the bottom left and the current system time and date is shown at the bottom right. The call and page areas show details of active calls or pages.

Navigation Control

The navigation control, located to the right of the OLED display, is used to navigate through the screens and options shown on the display.

Rotary Encoder Control

The rotary encoder control is used to set the volume levels for the headphones, speaker and the input levels for the various audio inputs. See Volume Control on page 20 for details.

Reset Switch

The recessed reset switch, located under the rotary encoder control, initiates a full hardware reset.

Mon 1 & 2 Buttons

These buttons activate the 2 monitor channels. The monitors can be set to provide audio from any local audio input or from an AoIP source. To configure the monitors, see the Audio - Monitor & Speaker Settings web page on page 42 for details.

LS Mute Button

This button mutes the audio feed to the speaker. The button illuminates red when the mute function is active. The operational mode of the button is user configurable, see the Audio - Monitor & Speaker Settings web page on page 42 for details.

Mic Mute Button

This button can be used to temporarily mute the microphone/headset input. The button illuminates red when active. The operational mode of the button is user configurable, see the Audio - Microphone Settings web page on page 39 for details.

Power Switch

The power switch turns the unit on and off. When the unit is on, the indicator is green. When the unit is off, the indicator is red. The power switch can be disabled if required, see the Miscellaneous Settings web page on page 47 for details.

Microphone Input Connector

This 3 pin XLR socket is the connection for the microphone. The connections are as follows:

Pin 1: Ground
Pin 2: Microphone Phase
Pin 3: Microphone Non-Phase

Phantom power can be supplied if required. To change the mic settings, see the Audio - Microphone Settings web page on page 39 for details.

Headset Connector

This 5 pin XLR socket is the connection for the headset. There is a parallel headset connector on the rear panel. The connections are as follows:

Pin 1:	Microphone Non-Phase
Pin 2:	Microphone Phase
Pin 3:	Ground
Pin 4:	Headphone Left
Pin 5:	Headphone Right

Phantom power can be supplied if required. To change the headset settings, see the Audio - Microphone Settings web page on page 39 for details.

+48V Phantom Power Indicator

This indicator illuminates red when the phantom power supply is enabled. To change the phantom power setting, see the Audio - Microphone Settings web page on page 39 for details.

Headphone Output Connector

The jack socket, located on the right side of the front panel under the speaker, is the headphone connection. The connections are as follows:

Tip:	Headphone Left
Ring:	Headphone Right
Sleeve:	Ground

When headphones are connected, the speaker mute is automatically activated.

4. Rear Panel Connections

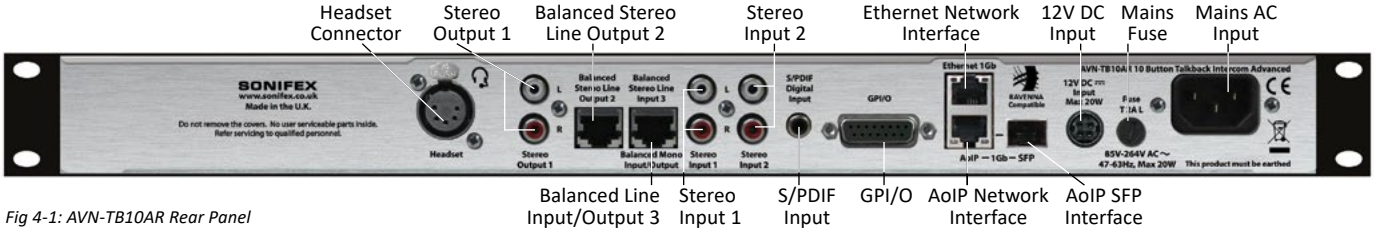

Fig 4-1: AVN-TB10AR Rear Panel

Headset Connector

This 5 pin XLR socket is the connection for the headset. There is a parallel headset connector on the front panel. The connections are as follows:

Pin 1:	Microphone Non-Phase
Pin 2:	Microphone Phase
Pin 3:	Ground
Pin 4:	Headphone Left
Pin 5:	Headphone Right

Phantom power can be supplied if required. To change the headset settings, see the Audio - Microphone Settings web page on page 39 for details.

Stereo Output 1

These RCA female connectors are an un-balanced stereo audio output. The connections are as follows:

Inner:	Signal
Outer:	Ground

The audio source for this output is configurable, see the Audio - Output Settings web page on page 41 for more details.

Balanced Stereo Line Output 2

This RJ45 socket is a balanced stereo audio output. The connections are as follows:

Pin 1:	Left Phase
Pin 2:	Left Non-Phase
Pin 3:	Right Phase
Pin 4:	Ground
Pin 5:	no connection
Pin 6:	Right Non-Phase
Pin 7:	no connection
Pin 8:	no connection

The audio source for this output is configurable, see the Audio - Output Settings web page on page 41 for more details.

Balanced Stereo Line Input 3 or Balanced Mono Input/Output

This RJ45 socket is configurable as either a balanced stereo audio input or a balanced mono input and output. The connections are as follows:

	Balanced Stereo Input	Balanced Mono Input/Output
Pin 1:	Left Phase	Input Phase
Pin 2:	Left Non-Phase	Input Non-Phase
Pin 3:	Right Phase	Output Phase
Pin 4:	Ground	Ground
Pin 5:	no connection	no connection
Pin 6:	Right Non-Phase	Output Non-Phase
Pin 7:	no connection	no connection
Pin 8:	no connection	no connection

See the Audio - Input Settings web page on page 40 for details on changing the function of this audio connection. When configured as a mono input/output, the audio source for this output is configurable, see the Audio - Output Settings web page on page 41 for more details.

Stereo Input 1 & Stereo Input 2

These RCA female connectors are un-balanced stereo audio inputs. The connections are as follows:

Inner:	Signal
Outer:	Ground

The audio presented to these inputs can be routed to an AoIP stream if required. See the AoIP Streams web page on page 29 for more details.

S/PDIF Digital Input

This RCA female connector is a stereo digital input which conforms to the S/PDIF (Sony/Philips Digital Interface) standard. The connections are as follows:

Inner:	Signal
Outer:	Ground

The audio presented to this input can be routed to an AoIP stream if required. See the AoIP Streams web page on page 29 and the Audio - Input Settings web page on page 40 for more details.

GPI/O

This 15 pin ‘D’ type socket provides 10 configurable general purpose inputs or outputs and a voltage free switching relay contact. The connections are as follows:

Pin 1	GPIO Port 1
Pin 2	GPIO Port 2
Pin 3	GPIO Port 3
Pin 4	GPIO Port 4
Pin 5	GPIO Port 5
Pin 6	Relay – Normally Open Contact
Pin 7	Relay – Normally Closed Contact
Pin 8	Relay – Common
Pin 9	GPIO Port 6
Pin 10	GPIO Port 7
Pin 11	GPIO Port 8
Pin 12	GPIO Port 9
Pin 13	GPIO Port 10
Pin 14	Fused (200mA) +12VDC Supply
Pin 15	Ground

GPIO ports configured as an output are open collector type. GPIO inputs are active low. See the GPIO Settings web page on page 45 for details on how to configure the GPIO and relay settings.

Ethernet 1GB & AoIP 1GB Network Interfaces

These RJ45 connectors are the gigabit network ports. The upper connector is the configuration and control port and the lower connector is the audio over IP (AoIP) port.

AoIP SFP Interface

This SFP (small form-factor pluggable) connector allows an alternative interface type to be used for the AoIP port, for example fibre. If a SFP interface is used, the AoIP network port cannot be used at the same time.

12V DC Input

This 4 pin connector allows an external +12V power supply to be used to power the unit. When looking at the rear of the unit, the pin locations and connections are as follows:

	Location	Connection
Pin 1	Upper Left	+12V DC
Pin 2	Upper Right	+12V DC
Pin 3	Lower Left	0V
Pin 4	Lower Right	0V

Both the 12V DC input and the mains AC input can be connected at the same time.

Mains Fuse

This 20mm x 5mm anti-surge mains fuse is rated at 2A.

Mains AC Input

This universally filtered IEC is the mains AC input power supply to the unit.

5. Quick Start Guide

This section outlines the steps required to get the AVN-TB10AR connected to a network and in a state where it can make and receive intercom calls.

1. Connect the upper Ethernet port to a suitable network to access the web server to configure the initial settings. By default this port is configured to use the static IP address 192.168.0.100.
2. Apply power to the unit via the IEC mains connection or via an external 12V DC power supply.
3. Once the unit has started and the main screen is shown on the OLED display, open a web browser and connect to the unit by entering the IP address 192.168.0.100 into the browser address bar. Once connected, the device information web page will be shown.
4. On the web page, select Network from the configuration tab and configure the Ethernet and AoIP ports to the required settings. The IP addresses assigned to the 2 ports must be on different subnets. For example, with the subnet mask on both ports set to 255.255.255.0, if the IP address on the Ethernet port is set to 192.168.0.100, the AoIP port must be set to 192.168.x.y where x is not 0 and y is unique on the chosen subnet.
5. Although not essential at this point, it is a good idea to assign a recognisable intercom name so that other users can easily identify and use services from this unit.
6. To simplify the configuration process, make sure the Auto Multicast option is enabled. This removes the need to set the transmit AoIP multicast IP addresses as they will be configured automatically based on the AoIP port IP address. Once the settings on the network page have been configured, press the submit button to save them to the connected unit. When the network settings have been changed, the unit will automatically restart. If the settings of the network port the browser is connected to have changed, a new connection for the browser will need to be established using the new IP address. If dynamic address mode has been selected, the active IP address for the network port can be found by going to the system info menu item in the main system menu. See System Info menu item on page 50 for details.
7. If a dedicated grandmaster clock, such as the AVN-GMC from Sonifex, is going to be used as the master PTP clock for the network audio system, the PTP profile on the AVN-TB10AR will need to be set to the current setting on the grandmaster device. On the web page, select PTP Profiles from the configuration tab. Select the required profile from the drop down list box and ensure that the domain number for the selected profile matches the grandmaster setting. Press the submit button to save the new settings. If no dedicated grandmaster device is used, the connected intercom units will automatically establish a master clock hierarchy. In this case, the default profile can be used.
8. Connect the lower AoIP port to the audio network.
9. On the web page, select Microphone Settings from the Audio Settings sub tab. Select the required microphone type, with or without phantom power. Now change the microphone or headset gain so that metering LEDs around the front panel rotary control peak in the amber section for normal speech. Press the submit button to save the new settings.
10. The final step is to assign remote intercoms or AoIP streams to the required channels. On the web page, select Basic Settings from the Channels sub tab. The connection type list box for each channel is shown next to the channel number. Ensure the connection type is set to AoIP Stream and then select the required remote intercom or AoIP stream from the Available AoIP Sources list box at the top of the page. This shows a dynamic list of currently available intercom devices including those with IFB, and AoIP streams on the network. To configure a channel for the selected AoIP source, press the corresponding assign button. For sources that are not currently available, the source name can be manually entered in the edit box for each channel. For intercom assignments, a channel on the selected device will need to be assigned to this unit in order to make intercom calls. Press the submit button to save the new settings.

The unit should now have the minimum settings configured to allow intercom calls between devices and AoIP streams.

6. Intercom Calls

Each channel button on the front panel can be assigned to a remote intercom unit in order to make and receive voice calls. They can also be configured to interface to legacy 4-wire talkback units, such as the Sonifex CM-TB8. See the Channels - Advanced Setting web page on page 34 for more details.

Making a Call

To make a call request, press the channel button assigned to the required remote unit. If the called unit has the auto answer option enabled, the call will be connected immediately. The button and display status for an active call is described later in this section. If the called unit's auto answer option is disabled, the selected channel button will slowly flash using the colour assigned to the default function for that channel. For example, if the default function is talk and listen, and the colour assigned to the talk and listen buttons is amber, the button will flash amber. At the same time, the call status indicator on the OLED display will highlight with an upward facing arrow, indicating an outgoing call request, and the channel number and intercom name of the called unit will be shown – see Fig 6-1. If multiple call requests are made at the same time, the display will cycle through the details of each called unit.

Fig 6-1: Outgoing Call Request Screen

If a call request is made to an intercom that doesn't have a valid channel assignment for the calling unit, the error message "NO CHANNEL ON" followed by the intercom name of the target unit, will temporarily be displayed at the bottom of the OLED display. To correct this error, assign the intercom name from the unit attempting to make the call to a channel on the target unit.

Receiving a Call

When an intercom call request is received from a remote unit, the call will be handled by the local channel assigned to the calling intercom unit. The way the call is handled is dependent on the auto answer option setting for the local channel. If the auto answer option is enabled, the call is connected immediately. If the auto answer option is disabled, the channel button that is assigned to the caller will rapidly flash using the colour assigned to the requested call type. For example, if the calling unit has requested a talk only call, the call will be treated locally as listen only. If the local colour assignment for listen buttons is green, the channel button will flash green. At the same time, the call status indicator on the OLED display will highlight with a downward facing arrow, indicating an incoming call request, and the channel number and intercom name of the calling unit will be shown – see Fig 6-2. If multiple call requests are received at the same time, the display will cycle through the details of each calling unit. If the call signal is enabled, a repeated audible signal will be heard in the headphones or speaker. The call is answered by pressing the flashing channel button and the audio from each unit will be routed depending on the callers selected call type.

Fig 6-2: Received Call Request Screen

During a Call

When an intercom call is connected, the channel button is illuminated using the colour assigned to the call type. The intercom unit that initiates the call always dictates the call type. The call status on the OLED display will be highlighted and the channel number and intercom name of the connected unit will be shown – see Fig 6-3. If multiple calls are active at the same time, the display will cycle through the details of each call. If a new call request is made or received, the display will also cycle through these details.

Fig 6-3: Active Call Screen

If a monitor channel is active when an intercom call is connected, the audio routed to the headphones or speaker from the selected monitor source is automatically muted or dimmed according to the selected monitor settings. See the Audio - Monitor & Speaker Settings web page on page 42 for details.

Ending a Call

An active call can be terminated by either party. To end a call, press the illuminated channel button. The call is ended immediately, the button illumination is turned off, and the call details are removed from the OLED display. If there are no other active or requested calls, the display call highlight is returned to its lowlight state. Any active monitor channels will return to their normal state when all calls have ended.

Group Calls

Intercom calls can be made simultaneously to multiple remote units using the group talk function. There are 3 group talk sets available which can include any combination of channels, see the Group Talk web page on page 36 for details. Group calls are selected by assigning a group talk set to one of the user buttons, see the Front Panel - User Buttons web page on page 38 for details. The user button, linked to a group talk set, can then be used to make or end a group call.

Call Signal

The call signal provides an audible indication that an incoming call is waiting to be answered. It can be heard in the headphones and through the speaker. If the call signal is not required, it can be turned off. See the Channels - Advanced Settings web page on page 34 for details.

Auto Answer

The auto answer option is individually configurable for each channel. An incoming call request on a channel that has this option enabled will connect immediately. To change the auto answer settings, see the Channels - Advanced Settings web page on page 34 for details.

Listen Button

Each channel has a default function option that selects the type of call that is initiated when the corresponding channel button is pressed. See Channels - Advanced Settings web page on page 34 for details. The front panel listen button can be used to change the default call function to listen only, talk and listen or talk forced listen. To override the default function for a channel, press the listen button. The first press will select listen only, and the button will illuminate using the colour assigned listen buttons. A second press will select talk and listen, and the button will illuminate using the colour assigned to talk and listen buttons. A third press will select talk forced listen, and the button will flash alternately using the colours assigned to both talk and listen buttons. A fourth press will cancel the listen function and the button illumination will turn off. Once the listen button indicates the required call function, press a channel button to call the assigned unit. If a channel button is not pressed within 3 seconds of the last listen button press, the listen function is cancelled.

Reviewing Channel Assignments

The current channel assignments, and the connection status for each assignment can be monitored via the web server or the front panel OLED display. To use the web server, see the Audio Assignments web page on page 24 for details. To see the information on the OLED display, ensure the main screen is shown and then briefly press the navigation centre button.

Fig 6-4: Channel Assignments Screen

The screen shows the status for each channel in a scrollable list. Use the navigation up and down buttons or the navigation rotary control to scroll through the list. Each entry shows the target intercom or AoIP stream name assigned to the respective channel. If the target is currently available on the network then the source name is highlighted, otherwise it is lowlighted. If a channel has no target assigned, the display shows - . Channels with the connection type set to the local analogue audio input-output will show “AUDIO I/O 3”. To exit this screen, press either the navigation left or centre buttons. The display will then return to the main screen.

Call Functions

The following table shows the corresponding call functions between the calling and receiving unit. The default channel button colours for the call type is also shown.

Calling Unit		Receiving Unit	
Call Type	**Default Channel Button Colour**	**Call Type**	**Default Channel Button Colour**
Talk Only	Red	Listen Only	Green
Listen Only	Green	Talk Only	Red
Talk & Listen	Amber	Talk & Listen	Amber
Talk Forced Listen - Talk inactive - Talk Active	 Green with Red Flash Red with Green Flash	Talk Forced Listen Talk Only Talk & Listen	 Red Amber

Table 6-1: Call Functions

The Talk Forced Listen call works differently to the standard talk/listen functions. When this type of call is initiated, the call starts as a listen only call. The callers channel button is illuminated using the colour assigned to listen buttons, with a brief flash using the colour assigned to talk buttons. The channel button on the receiving unit is illuminated to indicate a talk call. The calling unit can then repeatedly press the channel button to enable and disable talk. When talk is enabled, the channel button illuminates using the colour assigned to talk buttons, with a brief flash using the colour assigned to listen buttons. The channel button on the receiving unit is illuminated to indicate a talk & listen call when the callers talk is enabled. The receiving unit can end the call by pressing the channel button as per other call functions. The calling unit ends the call by briefly holding the channel button until the illumination has turned off.

7. Intercom Calls with IFB

Several models in the AVN Talkback range, including the AVN-TB10AR, can generate specific IFB AoIP streams. The content of the IFB feed is fully configurable as is the talk mode which determines how the audio in the IFB feed is modified when a call is active. As the content of the IFB stream is controlled by the unit generating the feed, the stream can be used by remote AoIP receiving devices without the need for them to control and mix multiple audio streams. See the AoIP Streams Settings web page on page 29 for details. These IFB streams from remote units can by assigned to the channel buttons on the AVN-TB10AR by enabling the channel IFB mode. See the Channels - Advanced Setting web page on page 34 for more details. With IFB mode enabled, the selected IFB feed from the remote unit will be heard in the headphones and through the speaker when the channel is idle, and the channel button will flash briefly in the colour assigned to listen buttons. Once a call is initiated and connected, the remote unit will modify the audio content of the IFB stream and the receiver will hear the required interrupt audio. Intercom calls with IFB are controlled in exactly the same way as normal intercom calls.

8. Non-Intercom Calls

The AVN-TB10AR allows any AoIP stream to be assigned to the channel buttons. When a call is initiated to a non AVN Talkback device, the unit does not attempt to communicate with the remote device for call control. It simply routes the AoIP stream assigned to the channel to the headphones and loudspeaker and unmutes the microphone audio routed to the microphone AoIP transmit stream. If simply call indication is required between devices, the GPIO ports on the unit can be used to provide control and tally functions. See the GPIO Settings web page on page 44 for details on how to configure the GPIO and relay settings.

9. Intercom Mode

The AVN-TB10AR has an intercom mode setting that controls the routing of audio to and from remote AoIP devices. The setting has 2 options, Normal Mode and 4W Bridge Mode and it applies to all channels. See the Audio - Advanced Settings web page on page 34 for more details.

Normal Mode

This is the normal operating mode of the unit. When one or more calls are connected, the remote devices hear the local microphone audio, and the audio received from the remote devices is mixed and routed to the headphone and loudspeaker.

4W Bridge Mode

This is an alternative mode of operation which is designed to link remote AoIP devices with local audio inputs and outputs, essentially bridging the 2 domains. When one or more calls are connected, the remote devices hear the audio presented at the selected local audio input. The audio received from the remote devices is mixed and routed to the selected local audio output.

10. Call Back

The AVN-TB10AR maintains a list of the last 18 unanswered intercom calls so that they can be returned if required. The main display shows the current number of missed calls in the upper left side of the screen. With no missed calls, the display shows “CB:00” and the text is not highlighted. When a call is missed, this indicator is highlighted, the number is incremented to show the total number of missed calls, and the call back button starts to flash green.

Call Back List

The call back list is only available when missed calls have been registered. To access the call back list, press the call back button. The button illuminates red and the display changes to show the call back list information – see Fig 10-1.

Fig 10-1: Call Back List Screen

Each missed call entry shows the callers intercom name and the time and date. The last 2 items shown are clear list, which deletes all entries from the list, and exit which returns to the main screen. Pressing the call back button whilst viewing the list also exits back to the main screen.

Returning a Missed Call

To return a missed call, highlight the required entry from the call back list using the navigation up and down buttons or the navigation rotary control then press the navigation centre button to select. The display returns to the main screen and a call to the selected unit is requested. If the call is answered, the selected call back list entry is deleted.

Deleting the Call Back List

To delete all entries from the call back list, scroll down to the clear list option using the navigation up and down buttons or the navigation rotary control then press the navigation centre button. The list is cleared and the display returns to the main screen. There is also a user button option to clear the call back list, see the Front Panel - User Buttons web page on page 38.

11. Page

The page function provides an easy method of making a high priority talk only call to a preselected group of target intercoms. Up to 20 intercom units can be selected as page targets and each unit can be configured to treat received page calls as a priority over normal intercom calls. See the Page Settings web page on page 43 for details.

Making a Page

To make a page call, press the page button. The button will slowly flash green and the page status indicator on the OLED display will highlight with an upward facing arrow, indicating an outgoing page call. The display also shows the number of configured page targets that are actually receiving the page – see Fig 11-1.

Fig 11-1: Outgoing Page Screen

A configured target may not be receiving the page for a number of reasons. These include:

- The target unit may be switched off or not connected to the network.
- The target unit may be already involved in a page call, either outgoing or incoming.
- The target unit may be involved in an intercom call and the page priority option on that unit is disabled, meaning that the intercom call takes priority.

Receiving a Page

When a page is received and accepted, the page button fast flashes red and the page status indicator on the OLED display will highlight with a downward facing arrow, indicating a received page call. The display will also show the intercom name of the paging unit and the local channel assignment, if one exists – see Fig 11-2.

Fig 11-2: Received Page Screen

If the paging unit is not assigned to a local intercom call channel, no channel number is shown. If the page priority option is disabled, and an intercom call is currently active, the page will be ignored.

During a Page

During a page call, the microphone audio signal from the paging unit will be heard by the units that accepted the page. The page targets do not need to respond to the page to hear the audio. Any current intercom calls on the units receiving the page will be suspended and any active monitor channels on the paging and receiving units will be muted for the duration of the page call.

Ending a Page

A page call can only be terminated by the unit that initiated it. To end a page call, press the green flashing page button. The page is ended immediately, the button stops flashing, and the page details are removed from the OLED display. Any suspended intercom calls on the target page units are resumed and any active monitor channels on the paging and receiving units are unmuted.

12. Volume Control

The rotary encoder control on the front panel is used to set the volume levels of the speaker, the headphones and the input levels for the various audio sources.

Speaker & Headphone Volume

The volume control automatically switches from speaker to headphones when a headphone jack plug is connected. To adjust the volume, turn the rotary encoder control. The front panel display switches immediately to the speaker or headphone volume screen which shows the current level as a sliding scale between minimum (muted) and maximum (+12dB gain). The volume screen will automatically close if no adjustments have been made for 1.5 seconds. Alternatively, the screen can be closed manually by pressing the rotary encoder.

Call Channel Volume

The volume levels of the audio from remote units assigned to the call channel buttons can be individually adjusted to compensate for varying talk levels. This can be done during a call if required. To adjust the channel volume, press the rotary encoder control which will then flash to indicate that volume adjustment mode has been initiated. This mode will automatically cancel if no further action is taken within 3 seconds. Then select a channel to adjust by pressing the associated channel button. The rotary encoder control and the selected channel button will now flash using the colour assigned to the default function for that channel. The front panel display immediately switches to the channel volume screen which shows the current level as a sliding scale. Turn the rotary encoder control to adjust the volume level. The volume screen will automatically close if no adjustments have been made for 1.5 seconds. Alternatively, the screen can be closed manually by pressing the rotary encoder.

Monitor Volume

The volume levels of the audio from the selected monitor sources can be individually adjusted to compensate for varying levels. This can be done regardless of whether the monitor channel is currently active or not. To adjust the monitor volume, press the rotary encoder control which will then flash to indicate that volume adjustment mode has been initiated. This mode will automatically cancel if no further action is taken within 3 seconds. Then select a monitor channel to adjust by pressing the associated monitor button. The rotary encoder control and the selected monitor button will now flash green. The front panel display immediately switches to the volume screen for the currently selected monitor source which shows the current level as a sliding scale. Turn the rotary encoder control to adjust the monitor level. The volume screen will automatically close if no adjustments have been made for 1.5 seconds. Alternatively, the screen can be closed manually by pressing the rotary encoder.

Call Signal Volume

The volume level of the call signal routed to the speaker and headphones can be adjusted to provide a comfortable mix level with other active audio sources. The call signal volume control automatically switches from speaker to headphones when a headphone jack plug is connected. To adjust the call signal volume, press and hold the rotary encoder control until it starts to flash blue. The front panel display switches to the speaker or headphone call signal volume screen which shows the current level as a sliding scale. The volume screen will automatically close if no adjustments have been made for 1.5 seconds. Alternatively, the screen can be closed manually by pressing the rotary encoder.

13. Embedded Web Server

The AVN-TB10AR has an embedded web server which provides easy access to all the configuration options through a web browser. It also gives access to system information and allows the firmware to be easily updated when new releases are made available.

To access the embedded web server, the unit needs to be connected to a network via either of the 2 network ports on the rear panel. The upper port is the general access Ethernet port and the lower port is the Audio over IP (AoIP) port. By default, both ports are set to static address mode with the upper port IP address set to 192.168.0.100 and the lower port IP address set to 192.168.1.100. If the network address mode for the port to be used has been set to dynamic, the unit will attempt to acquire an IP address from a DHCP server or auto configuration if no DHCP server is found. The current IP address of these ports can be accessed by going to the main menu via the front panel navigation control, and selecting System Info – see page 50 for details. If a port has yet to acquire an IP address or has failed to link, the corresponding IP address and subnet mask will show 0.0.0.0. Once the IP address of the required port is known, simply type this into the address bar of a web browser. The Device Information page of the connected AVN-TB10AR will be displayed. This is the default page and will always be displayed first when connecting to the embedded web server. Each page of the web server shows the intercom name in the upper right hand corner, under the AVN-TB10AR label. This makes it easier to identify the connected intercom unit, especially when configuring multiple units at the same time. The right hand side of each page has a brief help section that describes the content of each section.

Information Tab

The pages under the Information tab provide device information as well as a summary of the audio and GPIO assignments.

Device Information Web Page

The top part of this page shows details of the connected unit as shown:

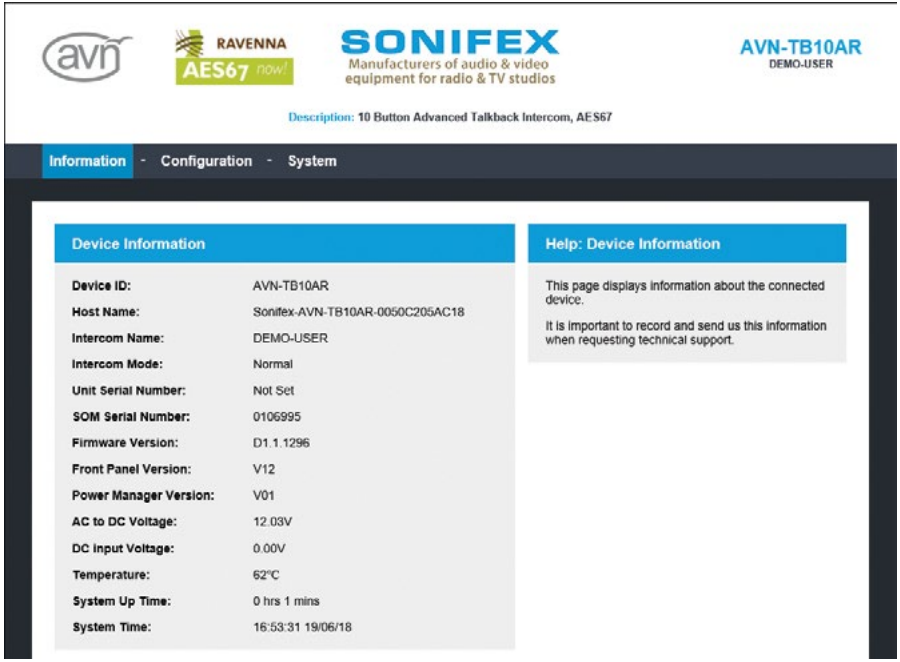

Fig 13-1: Upper Section of Device Information Web Page

This information shows the current status of the unit as well as the software versions of the various modules running on the unit. When contacting

Sonifex technical support, it is important to provide the information shown on this page.

The lower half of the page shows the status of the Precision Time Protocol (PTP) clock, as well as the configuration of the network ports as shown:

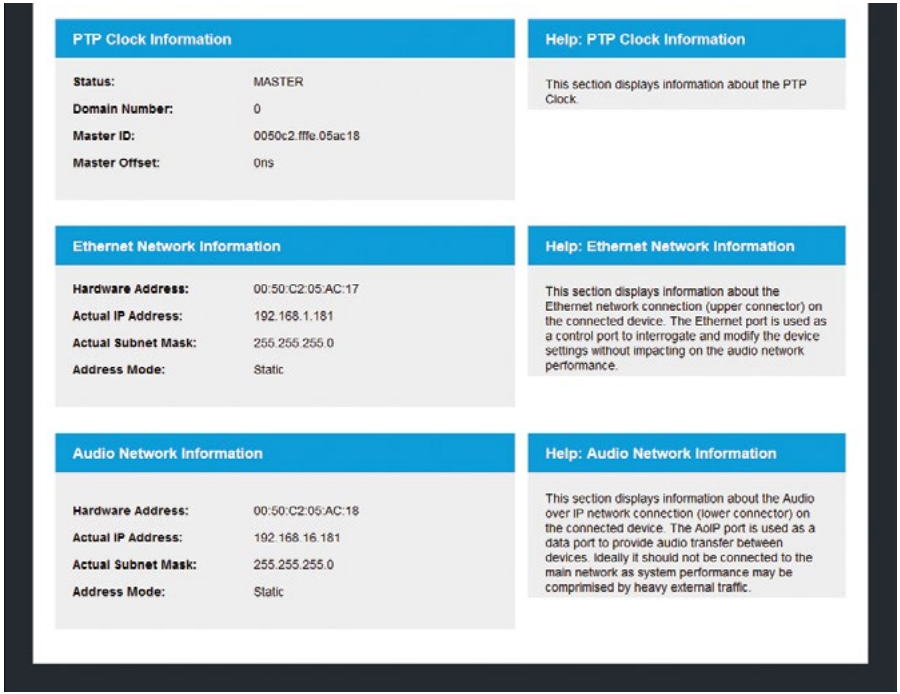

Fig 13-2: Lower Section of Device Information Web Page

The network IP addresses and subnet masks shown are the actual values currently in use.

Two additional pages under the information tab, provide a configuration summary of the audio and GPIO assignments.

Audio Assignments Web Page

The Audio Assignments page shows the current channel and monitor configuration as shown:

Each channel is listed with the intercom name or AoIP stream name assignment or local audio I/O connection – if any. Channels that are assigned to remote sources indicate whether that source is currently online. If it is online, the assignment is shown in black and the AoIP port IP address of the source is also shown. If the target assignment is an AVN talkback device, an AVN logo is shown next to the channel number. Unavailable sources or un-assigned channels are shown in grey.

The Monitor Assignments show the current configuration and status of the 2 monitor channels. If the monitor channel is assigned to an AoIP stream and the stream is currently available, the assignment is shown in black. Unavailable streams are shown in grey.

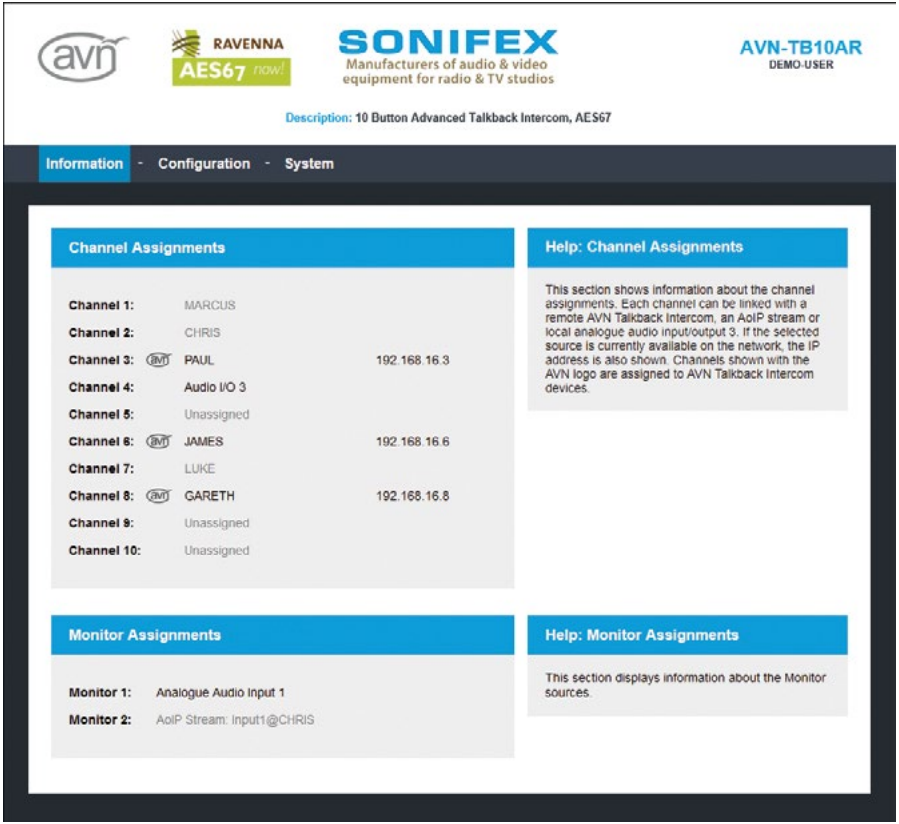

Fig 13-3: Audio Assignments Web Page

GPIO Assignments Web Page

The GPIO Assignments page shows the current GPIO, virtual GPIO and relay configuration as shown:

Each physical GPIO can be configured as an input, output or it can be disabled. Any configuration errors that will stop the GPIO functioning are shown in red. For example, GPIO 6 has been assigned as an input that activates a virtual output. However, none of the virtual GPIO have been set to an output that is driven by GPI6. At the bottom of the GPIO assignments is the list of functions that trigger the relay – if any.

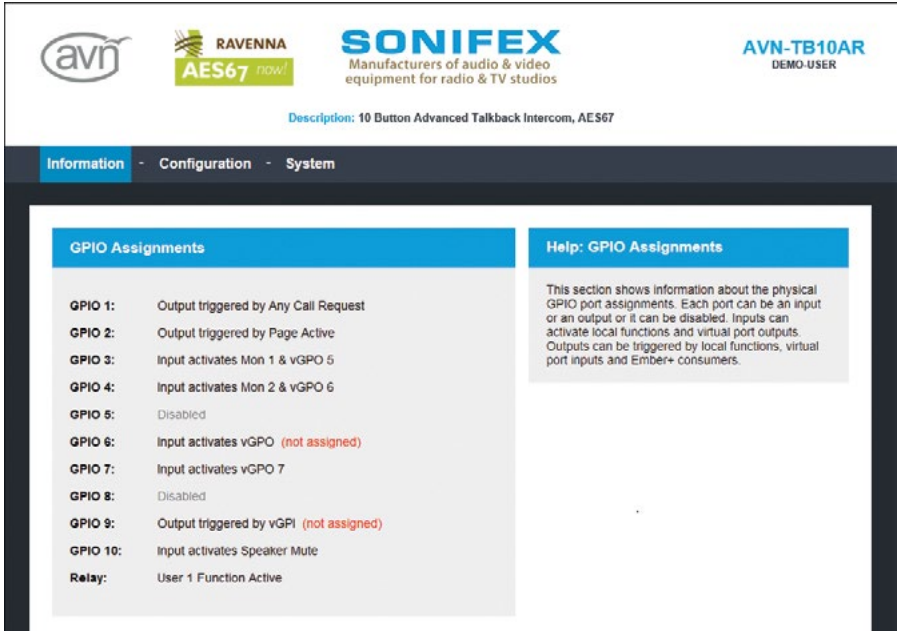

Fig 13-4: Upper Section of GPIO Assignments Web Page

Each virtual GPIO can also be configured as an input, output or it can be disabled. As with the GPIO assignments, configuration errors are shown in red. Virtual inputs require a remote virtual output driver to trigger the input. The driver details are shown below the virtual input description and will be grey if the driver service is not available. If the service is available, it will be shown in black.

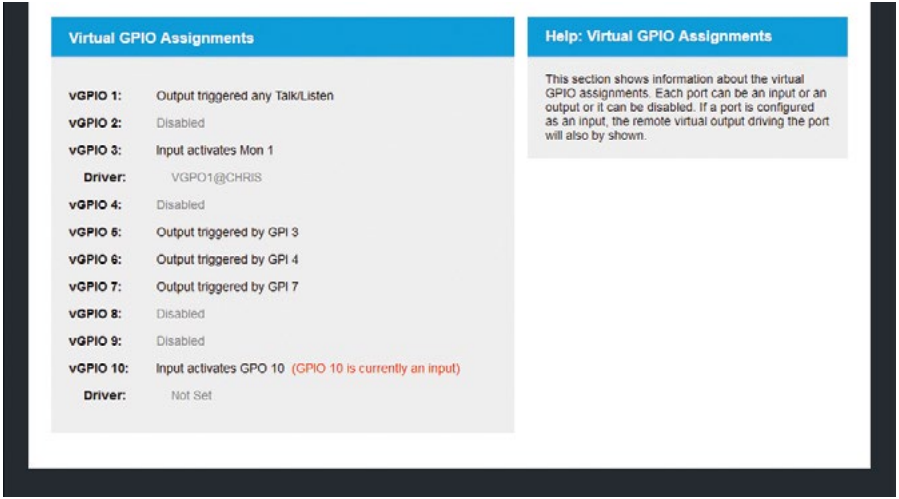

Fig 13-5: Lower Section of GPIO Assignments Web Page

Configuration Tab

The pages under the Configuration tab are split into groups and all of the unit's configuration settings can be edited here. Each page has a SUBMIT button that sends the changes to the connected unit. The SUBMIT button remains disabled until changes have been made on the page. Once the SUBMIT button has been pressed, a busy indicator will be shown and the page will be temporarily disabled until the changes have been written.

Network Web Page

The Network page shows the current configuration of the Ethernet and AoIP network ports. The intercom name is also set here.

Intercom Name – The intercom name is the user friendly name that identifies the unit on the network. It is a good idea to assign a user name or location as this is easily recognised by other users. The default intercom name is made from the device ID and the 7 digit product serial number i.e. AVN-TB10AR-1234567. The intercom name can only contain letters, numbers and hyphens although it cannot start or end with a hyphen.

Address Mode – Each network port has its own independent address mode which determines how the port obtains its IP address. When set to dynamic, the unit will attempt to acquire an IP address automatically from either a DHCP server or via auto configuration if no DHCP server is found. The actual IP address will be shown on the device information page. When static mode is used, the IP address and subnet mask values entered will be assigned to the corresponding network port.

Static IP Address – This is the IP address that will be assigned to the corresponding network port when static address mode is selected. It is important to ensure that this IP address is not currently in use on the network. These values are not used when the address mode is dynamic.

Static Subnet Mask – This is the subnet mask that will be used for the corresponding network port when static address mode is selected. These values are not used when the address mode is dynamic.

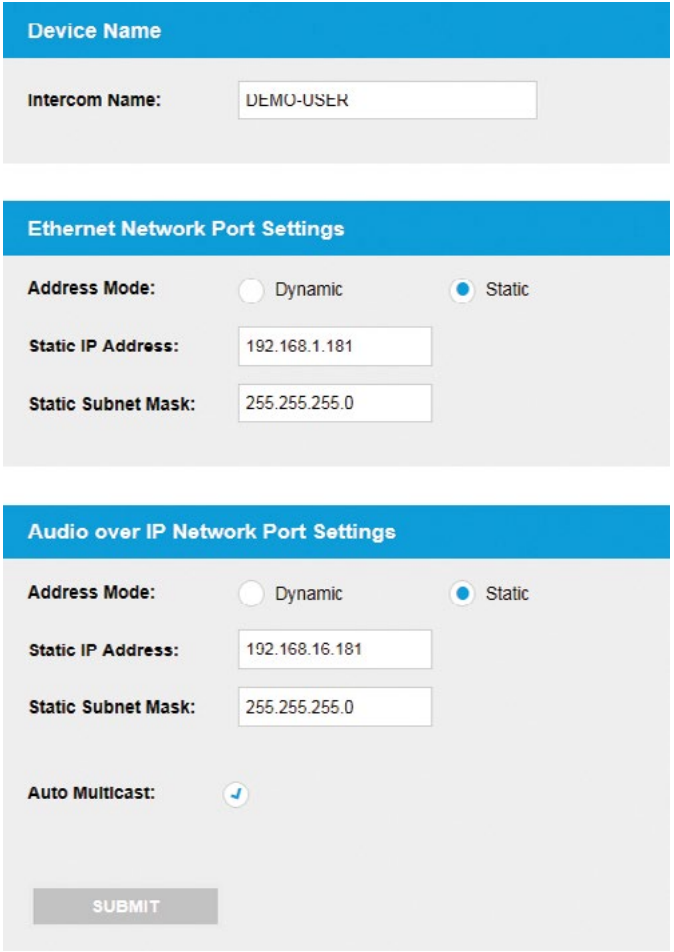

Fig 13-6: Network Web Page

Auto Multicast – When this option is enabled and the address mode of the AoIP port is set to static, the AoIP transmit multicast IP addresses, shown on the AoIP Streams Web Page, are automatically configured based on the AoIP port IP address. If the IP address of the AoIP port is changed, the multicast addresses are updated. This simplifies configuration and ensures the multicast addresses are unique on the network.

If any of the network configuration options are changed, the unit will automatically restart to implement the new settings. If the IP address of the network port that is currently being used to access the web server is changed, a new connection will need to be made once the unit has restarted. Otherwise, the new page will be shown automatically once the restart is complete.

Network Defaults

Intercom Name:	AVN-TB10AR-xxxxxxx
	Where xxxxxxx is the product serial number
Ethernet Port:	
Address Mode:	Static
Static IP Address:	192.168.0.100
Static Subnet Mask:	255.255.255.0
Audio over IP Port:	
Address Mode:	Static
Static IP Address:	192.168.1.100
Static Subnet Mask:	255.255.255.0
Auto Multicast:	Enabled

AoIP Streams Web Page

The AoIP Streams page shows the current configuration of the AoIP transmit options.

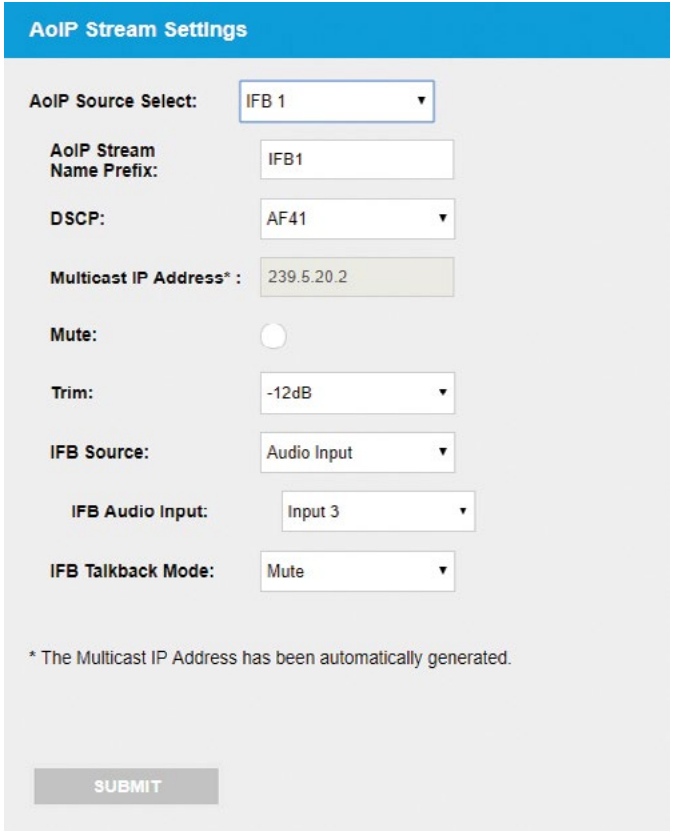

Fig 13-7: AoIP Streams Web Page

AoIP Source Select – There are 8 transmit sources on the AVN-TB10AR. These are for the microphone/headset input (mic), the 3 analogue audio inputs on the rear panel (Input 1-3), the S/PDIF input which is also on the rear panel (S/PDIF) and the 3 IFB streams that can be sourced from a local audio input or from a remote AoIP stream (IFB 1-3). These AoIP source streams are always active and cannot be disabled. This drop down list selects the transmit source to view and edit.

AoIP Stream Name Prefix – The AoIP stream name prefix forms part of the AoIP stream name which will be displayed on remote devices when searching for AoIP streams. This stream name is comprised of the AoIP stream name prefix, set here for the selected transmit source, and the intercom name. For example, if the default input 1 AoIP stream name prefix “Input1” is used, and the intercom name is set to “DEMO-USER”, the advertised AoIP stream name would be “Input1@DEMO-USER”. The AoIP stream name prefix must be unique for each transmit source.

DSCP – The DSCP or Differentiated Services Code Point value is used by DiffServ to control the precedence of outgoing AoIP data packets for the selected transmit source. Network hardware can use this value to ensure audio data has higher priority over other network traffic. See Table 17-1 DSCP Names & Their Corresponding IP Precedence on on page 68 for more information.

Multicast IP Address – The multicast IP address is the IPv4 multicast destination address for the selected transmit source. Remote network devices that are configured to receive this source will use this IP address to access the audio data. It is important to ensure that the IP address entered here is unique on the network as AoIP stream sources cannot share the same multicast IP address. Multicast IP addresses should be within the range 224.0.0.0 to 239.255.255.255. If the Auto Multicast option on the Network page is enabled and the AoIP port address mode is set to static, these multicast IP addresses are automatically configured and cannot be changed.

Mute – This checkbox mutes the audio on the selected transmit source. This option is not available for the mic source.

Trim – The audio level of the selected transmit source can be adjusted by ±12dB. This option is not available for the mic source.

IFB Source – This option selects the audio source for the selected IFB transmit stream. This can be a local audio input or an AoIP stream. This option is only available for IFB sources.

IFB Audio Input – This option selects the local audio input for the selected IFB transmit stream when the source is set to audio input. This option is unavailable when the IFB source is set AoIP stream.

IFB AoIP Stream – This setting is the name of the AoIP stream that will be routed to the selected IFB transmit stream when the IFB source is set to AoIP stream. For streams sourced from other AVN intercom devices, the stream name is comprised of the AoIP stream name prefix and the intercom name. For example, if the required AoIP stream, called “Input1”, is being transmitted from an intercom unit with the name “DEMO-SOURCE”, the AoIP stream name would be “Input1@DEMO-SOURCE”. This name can be entered manually, or it can be selected from the dynamic list of available AoIP streams detailed below. This option is unavailable when the IFB source is set to audio input.

Available AoIP Streams – This dynamic drop down list shows the AoIP streams that are currently visible on the network. Selecting an AoIP stream from this list places the name into the IFB AoIP Stream edit box. This option is unavailable when the IFB source is set to audio input.

IFB Talkback Mode – This setting controls how the audio level of the selected IFB source is modified when an intercom channel is active. There are 3 options, the level of the IFB source is either muted, reduced by 75% or reduced by 50%.

AoIP Streams Defaults

Mic Transmit Source:	
AoIP Stream Name Prefix:	Mic
DSCP:	AF41
Multicast IP Address:	239.0.1.100
Input 1 Transmit Source:	
AoIP Stream Name Prefix:	Input1
DSCP:	AF41
Multicast IP Address:	239.1.1.100
Mute:	Enabled
Trim:	0dB
Input 2 Transmit Source:	
AoIP Stream Name Prefix:	Input2
DSCP:	AF41
Multicast IP Address:	239.2.1.100
Mute:	Enabled
Trim:	0dB
Input 3 Transmit Source:	
AoIP Stream Name Prefix:	Input3
DSCP:	AF41
Multicast IP Address:	239.3.1.100
Mute:	Enabled
Trim:	0dB
S/PDIF Transmit Source:	
AoIP Stream Name Prefix:	SPDIF
DSCP:	AF41
Multicast IP Address:	239.4.1.100
Mute:	Enabled
Trim:	0dB
IFB 1 Transmit Source:	
AoIP Stream Name Prefix:	IFB1
DSCP:	AF41

Multicast IP Address:	239.5.1.100
Mute:	Enabled
Trim:	0dB
IFB Source:	Audio Input
IFB Audio Input:	Input 1
IFB AoIP Stream:	not set
IFB Talkback Mode:	Mute
IFB 2 Transmit Source:	
AoIP Stream Name Prefix:	IFB2
DSCP:	AF41
Multicast IP Address:	239.6.1.100
Mute:	Enabled
Trim:	0dB
IFB Source:	Audio Input
IFB Audio Input:	Input 1
IFB AoIP Stream:	not set
IFB Talkback Mode:	Mute
IFB 3 Transmit Source:	
AoIP Stream Name Prefix:	IFB3
DSCP:	AF41
Multicast IP Address:	239.7.1.100
Mute:	Enabled
Trim:	0dB
IFB Source:	Audio Input
IFB Audio Input:	Input 1
IFB AoIP Stream:	not set
IFB Talkback Mode:	Mute

PTP Profiles Web Page

The PTP Profiles page shows the currently selected PTP profile. It also allows the parameters in each profile to be edited.

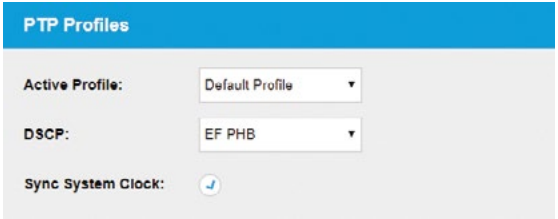

Fig 13-8: Upper Section of PTP Profiles Web Page

Active Profile – There are 3 PTP profiles available: Default, AES67 Media and Custom. This drop down list selects which of the profiles will be active.

DSCP – The DSCP or Differentiated Services Code Point value is used by DiffServ to control the precedence of outgoing PTP packets. Network hardware can use this value to ensure PTP data has higher priority over other network traffic. See Table 17-1 – DSCP Names & Their Corresponding IP Precedence for more information.

Sync System Clock – This option allows the system date and time, shown on the OLED display main screen, to be synchronised to the date and time of the PTP master device.

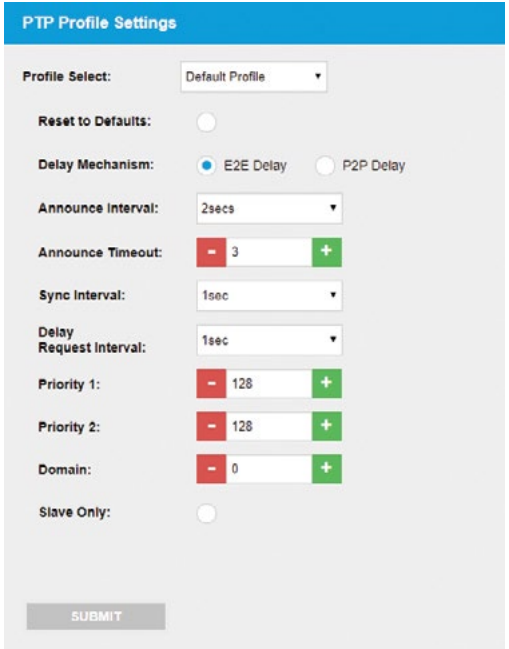

Fig 13-9: Lower Section of PTP Profiles Web Page

Profile Select – This drop down list selects the profile to view and change settings. All of the available profiles can be edited.

Reset to Defaults – Selecting this checkbox and pressing the submit button resets the selected profile to the default settings for that profile.

Delay Mechanism – This selects the method for calculating and maintaining the clock synchronisation communication delay between master and slave devices. Both P2P (Peer to Peer) and E2E (End to End) methods are supported. The best method for a particular installation will depend on the type of network hardware used. If all the switches are known to be IEEE 1588 capable, P2P should be used. Otherwise it is best to use E2E.

Announce Interval – This is the rate at which the announce message is sent, one per the selected time. The announce message forms part of the Best Master Clock Algorithm (BMCA) and contains the properties of the clock which sends it. If a device receives an announce message from a better clock, it will enter the slave state.

Announce Timeout – This is the time the unit will wait for an announce message and is a multiple of announce intervals. If no announce messages are received within the timeout interval, the unit will assume the role of grandmaster.

Sync Interval – This is the rate at which the sync message is sent, one per the selected time. The sync and follow up messages are sent from the master to the slave to determine the difference in clock frequency. This information is used in conjunction with the network delay to synchronise the clocks.

Delay Request Interval – This defines the rate at which a slave clock sends delay request messages to the master. The delay request message allows a slave device to calculate the network delay from the slave to the master. This option is only valid when using the E2E delay mechanism.

Peer Delay Request Interval – This is the rate at which a device exchanges peer delay measurement messages. This allows each unit to track the network delays between itself and its connected neighbours. This option is only valid when using the P2P delay mechanism.

Priority 1 & Priority 2 – These values define a precedence setting used by the best master clock algorithm when selecting a grandmaster clock. Priority 2 defines a fine tune setting when multiple clock sources have similar ordering criteria.

Domain – This defines the group of PTP devices that the unit will communicate with.

Slave Only – This option limits the unit to being a slave PTP clock only.

PTP Profiles Defaults

Active Profile:	Default
DSCP:	AF PHB
Sync System Clock:	Enabled
Default Profile:	
Delay Mechanism:	E2E
Announce Interval:	2 secs
Announce Timeout:	3 secs
Sync Interval:	1 sec
Delay Req Interval:	1 sec
Peer Delay Req Interval:	1 sec
Priority 1:	128
Priority 2:	128
Domain:	0
Slave Only:	Disabled
AES67 Media Profile:	
Delay Mechanism:	E2E
Announce Interval:	2 secs
Announce Timeout:	3 secs
Sync Interval:	1/8 sec
Delay Req Interval:	1 sec
Peer Delay Req Interval:	1 sec
Priority 1:	128
Priority 2:	128
Domain:	0
Slave Only:	Disabled
Custom Profile:	
Delay Mechanism:	E2E
Announce Interval:	2 secs
Announce Timeout:	3 secs
Sync Interval:	1 sec
Delay Req Interval:	1 sec
Peer Delay Req Interval:	1 sec
Priority 1:	128
Priority 2:	128
Domain:	0
Slave Only:	Disabled

Channels - Basic Settings Web Page

The Basic Settings page is located under the Channels sub tab. It allows the connection type and AoIP stream source assignments to be set for all 10 channel buttons on the front panel. The other channel options are shown on the Channels - Advanced Settings web page – see page 34 for details.

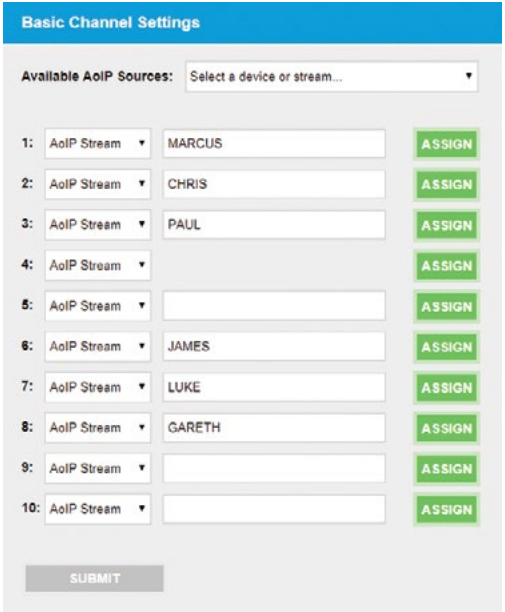

Fig 13-10: Channels - Basic Settings Web Page

Available AoIP Sources – This dynamic drop down list shows the 3 types of AoIP sources that are currently visible on the network and available for selection as a target for a channel. These types are AVN Intercom Devices, AVN Intercom Devices with IFB and AoIP Streams. To make an assignment, select a source from the list and then press the assign button on the required channel. This option is unavailable when the connection type is set to Audio IO 3.

Channels 1 - 10 – Each row shows the basic settings for the corresponding channel. The connection type is selected from the drop down list. This setting selects the type of audio connection that is initiated when the corresponding channel button is pressed. There are 2 options, AoIP Stream which connects to a remote intercom device or AoIP stream and Audio IO 3 which allows connection to a legacy 4-wire analogue talkback channel, such as the Sonifex CM-TB8, via audio input port 3 on the rear panel. When selecting this connection type, the mode for balanced stereo input 3 will need to be changed to provide a balanced mono input/output. The audio input and output signals can then be routed to the 4-wire talkback channel. See the Audio - Input Settings web page on page 40 for details. As there is no network control of the call handling in this scenario, a GPIO input port could be used to automatically enable the relevant channel when a talk button on the 4-wire unit is activated. See the GPIO Settings web page on page 45 for details.

When the connection type is set to AoIP Stream, the current stream name will be shown. To change the stream name, select a source from the Available AoIP Sources list and press the assign button on the required channel. Alternatively, the stream name can also be entered manually for remote sources that are currently offline.

Channels - Basic Settings Defaults

Channels 1 – 10:

Connection:	AoIP Stream
AoIP Source:	not set

Channels - Advanced Settings Web Page

The Advanced Settings page is located under the Channels sub tab. It shows the current assignments for each of the 10 channel buttons on the front panel. Other options that control intercom calls are also set here.

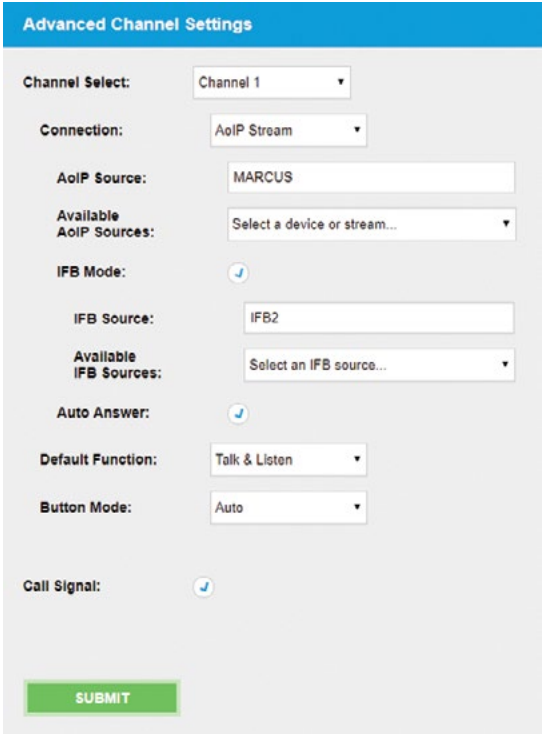

Fig 13-11: Channels - Advanced Settings Web Page

Channel Select – This drop down list selects the channel to view and change settings.

Connection – This setting selects the type of audio connection that is initiated when the corresponding channel button is pressed. There are 2 options, AoIP Stream which connects to a remote intercom device or AoIP stream and Audio IO 3 which allows connection to a legacy 4-wire analogue talkback channel, such as the Sonifex CM-TB8, via audio input port 3 on the rear panel. When selecting this connection type, the mode for balanced stereo input 3 will need to be changed to provide a balanced mono input/output. The audio input and output signals can then be routed to the 4-wire talkback channel. See the Audio - Input Settings web page on page 40 for details. As there is no network control of the call handling in this scenario, a GPIO input port could be used to automatically enable the relevant channel when a talk button on the 4-wire unit is activated. See the GPIO Settings web page on page 45 for details.

AoIP Source – This setting assigns the intercom name or AoIP stream name to connect to when the intercom channel is active. If the source is an intercom device, this name corresponds to the intercom name that is assigned on the network web page. This name can be entered manually, or it can be selected from the dynamic list of available AoIP sources detailed below. This option is unavailable when the connection type is set to Audio IO 3.

Available AoIP Sources – This dynamic drop down list shows the 3 types of AoIP sources that are currently visible on the network and available for selection as a target for a channel. These types are AVN Intercom Devices, AVN Intercom Devices with IFB and AoIP Streams. Selecting an AoIP source from this list places the name into the AoIP source edit box. This option is unavailable when the connection type is set to Audio IO 3.

IFB Mode – This option allows the selected channel to be assigned to an IFB AoIP stream from a remote intercom device. This stream type will have a designated audio feed that will be replaced by or mixed with the microphone audio when the channel is active. If IFB mode for the channel is enabled and the AoIP source is an available intercom device, but the configured IFB stream does not exists, a non IFB mode call will be connected instead. This option is unavailable when the connection type is set to Audio IO 3.

IFB Source – This is the AoIP stream name prefix of the IFB stream on the selected AoIP Source. This name can be entered manually, or it can be selected from the dynamic list of available IFB sources detailed below. This option is only available if IFB mode is enabled.

Available IFB Sources – This dynamic drop down list shows the names of the available IFB streams on the selected AoIP source. If the selected AoIP source is not an IFB enabled intercom device, this list will be empty. This option is only available if IFB mode is enabled.

Auto Answer – This option allows an incoming call on the selected channel to be answered automatically without the need to press the associated channel button.

Default Function – The default function option sets the call type that is initiated when the corresponding channel button is pressed. Talk & Listen calls allow both the caller and receiver hear each other. In a Talk Only call, only the receiver hears the caller. In a Listen Only call, only the caller hears the receiver. Talk Forced Listen calls allow the caller to hear the receiver, with the option to enable or disable talk to the receiver by repeatedly pressing the channel button. If IFB mode is enabled on the selected channel, Talk Forced Listen is not available. The default function can be overridden when initiating call by using the listen button on the front panel.

Button Mode – Selects the operational mode of the selected channel talk button. In momentary mode, the channel is active whilst the button is pressed. In latching mode, a separate button press is required to enable and disable the channel. In auto mode, the button has a momentary action that will automatically latch on or off if pressed briefly. The button can also be disabled if required. Momentary mode is not available if auto answer is enabled.

Call Signal – This option enables an audible indication that an incoming call is waiting to be answered. The call signal is heard through the loudspeaker, or the headphones if connected. This setting applies to all channels.

Channels - Advanced Settings Defaults

Channels 1 – 10:

Connection:	AoIP Stream
AoIP Source:	not set
IFB Mode:	Disabled
IFB Source:	not set
Auto Answer:	Disabled
Default Function:	Talk & Listen
Button Mode:	Auto
Call Signal:	Enabled

Group Talk Web Page

The Group Talk page shows the channels that are included in the 3 group talk sets.

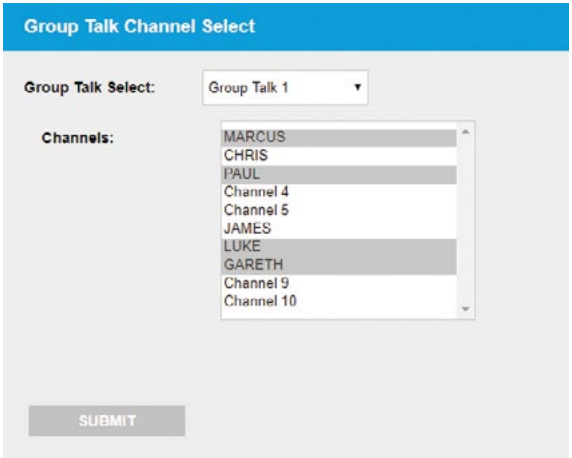

Fig 13-12: Group Talk Web Page

Group Talk Select – This drop down list selects the group talk set to view and change settings.

Channels – Click on this list to select the channels that are to be included in the selected group talk set. To select multiple channels, hold down the Ctrl key on the keyboard and then make a selection.

Group Talk Defaults

Group Talk 1:

Channels: 1, 3, 5, 7, 9

Group Talk 2:

Channels: 2, 4, 6, 8, 10

Group Talk 3:

Channels: 1, 2, 3, 4, 5, 6, 7, 8, 9, 10

Front Panel - Display Settings Web Page

The Display Settings page is located under the Front Panel sub tab. It allows the OLED display contrast and the LED brightness to be adjusted.

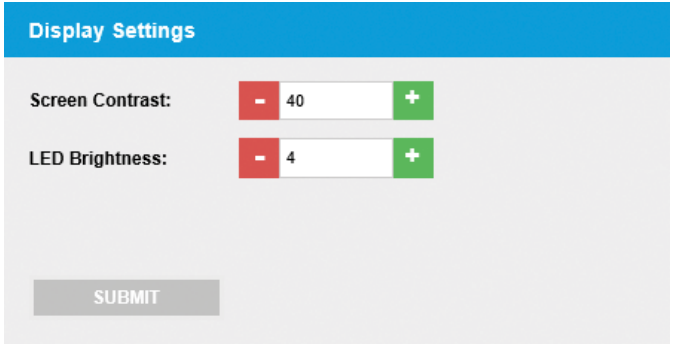

Fig 13-13: Front Panel - Display Settings Web Page

Screen Contrast – Click on the + or – buttons to change the screen contrast in the range 1 to 64.

LED Brightness – Click on the + or – buttons to change the brightness of the LEDs on the front panel in the range 1 to 7.

Front Panel - Display Settings Defaults

Screen Contrast: 40

LED Brightness: 4

Front Panel - Colour Settings Web Page

The Colour Settings page is located under the Front Panel sub tab. It shows the colours that are assigned to the intercom call and user buttons as shown:

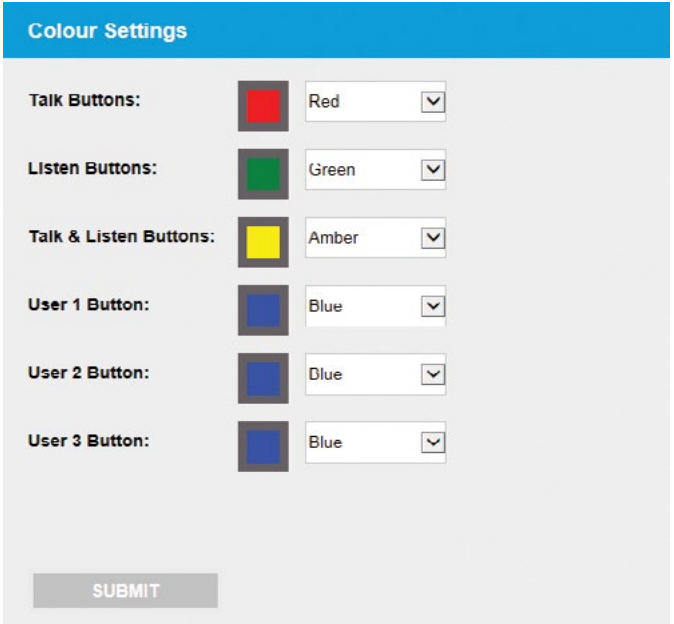

Fig 13-14: Front Panel - Colour Settings Web Page

Talk Buttons – This setting defines the colour of the channel buttons when a talk only call is active.

Listen Buttons – This setting defines the colour of the channel buttons when a listen only call is active.

Talk & Listen Buttons – This setting defines the colour of the channel buttons when a talk & listen call is active.

User 1 – 3 Buttons – This setting defines the colour of the 3 user assignable function buttons.

Front Panel - Colour Settings Defaults

Talk Buttons:	Red
Listen Buttons:	Green
Talk & Listen Buttons:	Amber
User 1 Button:	Blue
User 2 Button:	Blue
User 3 Button:	Blue

Front Panel - User Button Settings Web Page

The User Button Settings page is located under the Front Panel sub tab. It shows the configuration of the 3 user buttons.

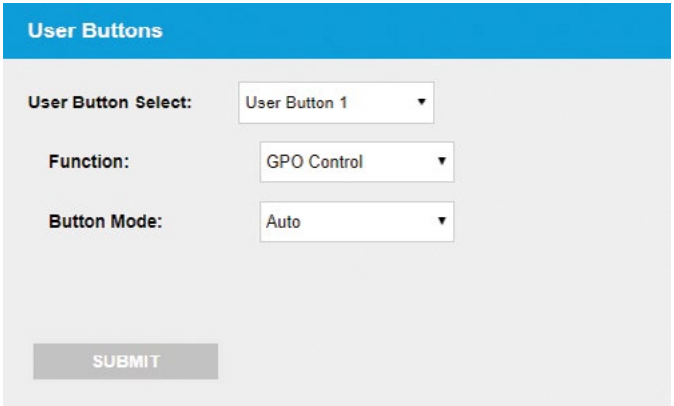

Fig 13-15: Front Panel - User Button Settings Web Page

User Button Select – This drop down list selects the user button to view and change settings.

Function – This option sets the function for the selected user button. Group Talk 1-3 initiates a group talk call to the members of the corresponding group talk set. Phone control* provides an interface to external telephone equipment. Clear Call Back List deletes all the entries from the missed call log. GPO Control allows a selection of physical and virtual GPO ports to be asserted. The GPO and virtual GPO ports that are under user button control are assigned in the GPIO and virtual GPIO settings pages.

Button Mode – Selects the operational mode of the selected user button. In momentary mode, the selected user function is active whilst the button is pressed. In latching mode, a separate button press is required to enable and disable the selected user function. In auto mode, the button has a

momentary action that will automatically latch on or off if pressed briefly. The button can also be disabled if required.

*this functionality is not currently available

Front Panel - User Button Defaults

User Button 1:

Function:	Group Talk 1
Button Mode:	Auto

User Button 2:

Function:	Group Talk 2
Button Mode:	Auto

User Button 3:

Function:	Clear Call Back List
Button Mode:	Auto

Audio - Microphone Settings Web Page

The Microphone Settings page is located under the Audio sub tab. The microphone input type and associated options are set here.

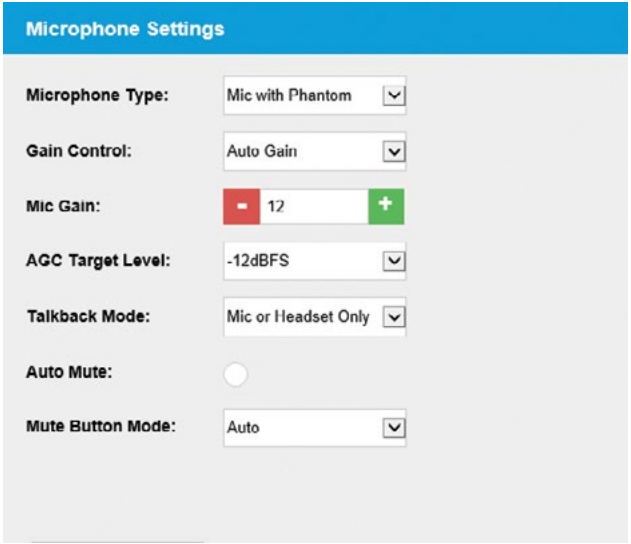

Fig 13-16: Audio - Microphone Settings Web Page

Microphone Type – This drop down list selects the required microphone input type and also enables phantom power if required.

Gain Control – This option selects between fixed or auto gain control. When auto gain is selected, the gain levels for the mic and headset inputs are automatically adjusted to compensate for varying talk levels.

Mic Gain – This option sets the input gain for the mic in the range 0dB to +60dB in 3dB steps.

Headset Gain – This option sets the input gain for the headset in the range 0dB to +60dB in 3dB steps.

AGC Target Level – This option sets the required output level for the microphone/headset automatic gain control. This option is available only when the gain control is set to auto gain.

Talkback Mode – This option selects what a call recipient will hear during an active intercom call. This can be the selected microphone or headset only, or a mix with one of the audio inputs on the rear panel.

Auto Mute – This option selects whether the microphone or headset AoIP stream is automatically muted when no calls are active. Please note, if this option is not enabled, the audio from the microphone or headset will be available to any other devices on the network.

Mute Button Mode – Selects the operational mode of the mic mute button on the front panel. In momentary mode, the mic mute is active whilst the button is pressed. In latching mode, a separate button press is required to enable and disable the mic mute function. In auto mode, the button has a momentary action that will automatically latch on or off if pressed briefly. The button can also be disabled if required.

Audio - Microphone Settings Defaults

Microphone Type:	Normal Mic
Gain Control:	Fixed Gain
Mic Gain:	30
Headset Gain:	30
AGC Target Level:	-18dBFS
Talkback Mode:	Mic or Headset Only
Auto Mute:	Enabled
Mute Button Mode:	Auto

Audio - Input Settings Web Page

The Input Settings page is located under the Audio sub tab and it shows the configuration of audio input 3 and the audio line up level.

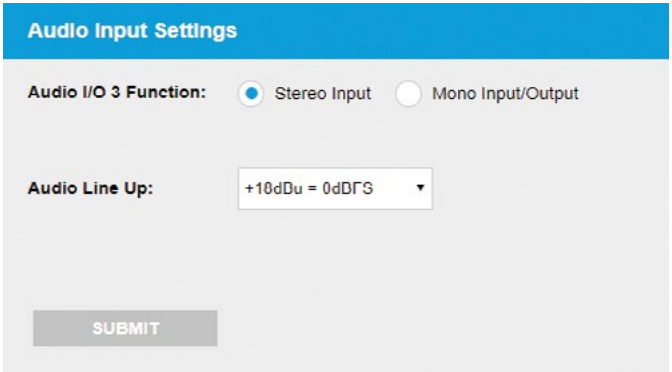

Fig 13-17: Audio - Input Settings Web Page

Audio Input 3 Function – Analogue audio input 3 on the rear panel can be configured as either a balanced stereo input or a balanced mono input and output. In this mode, the mono output is routed to the right channel of the connector.

Audio Line Up – The input audio line up applies to the balanced stereo line input 3 or balanced mono input when audio input function is set to mono input/output. It sets the analogue audio level that corresponds to full scale digits when converted to digital audio. It is recommended to use a similar value on all equipment that offers variable input line up.

Audio Input Settings Defaults

Audio Input 3 Function:	Stereo Input
Audio Line Up:	+18dBu = 0dBFS

Audio - Output Settings Web Page

The Output Settings page is located under the Audio sub tab and it shows the current source and state of the 3 analogue audio outputs. The output audio line level is also set here.

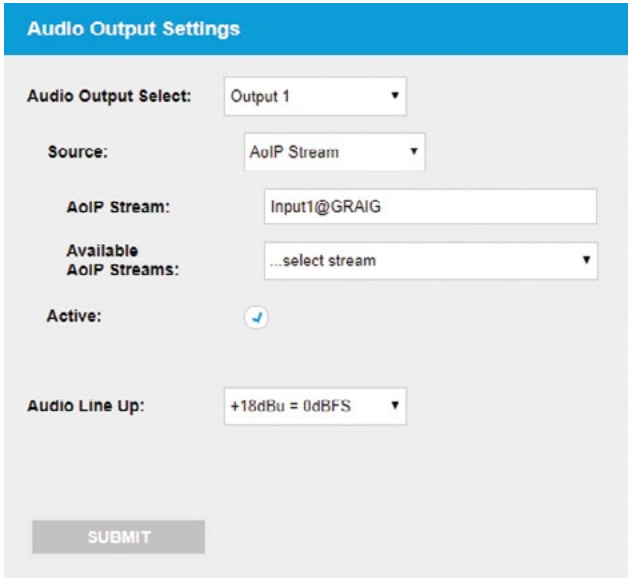

Fig 13-18: Audio - Output Settings Web Page

Audio Output Select – This drop down list selects the audio output to view and change settings.

Source – This option selects the audio source for the selected output. This can be any of the local audio inputs including the microphone, or an AoIP stream.

AoIP Stream – This setting is the name of the AoIP stream that will be routed to the selected output, if the output is active. The stream name is comprised of the AoIP stream name prefix and the intercom name. For example, if the required AoIP stream, called “Input1”, is being transmitted from an intercom unit with the name “DEMO-SOURCE”, the AoIP stream name would be “Input1@DEMO-SOURCE”. This name can be entered manually, or it can be selected from the dynamic list of available AoIP streams detailed below. This option is unavailable when the source is set to one of the local audio inputs.

Available AoIP Streams – This dynamic drop down list shows the AoIP streams that are currently visible on the network. Selecting an AoIP stream from this list places the name into the AoIP stream edit box. This option is unavailable when the source is set to one of the local audio inputs.

Active – This option routes the selected source to the associated output. If the output source is set to an AoIP stream and that stream is not currently available, the output will be muted.

Audio Line Up – The output audio line up applies to the balanced stereo line output 2 and the balanced mono input/output. It sets the analogue audio output level that corresponds to full scale digits when converted from digital audio. It is recommended to use a similar value on all equipment that offers variable output line up.

Audio - Output Settings Defaults

Output 1 – 3:

Source:	Input 1
AoIP Stream:	not set
Active:	Disabled
Audio Line Up:	+18dBu = 0dBFS

Audio - Monitor & Speaker Settings Web Page

The Monitor & Speaker Settings page is located under the Audio sub tab and it shows the current assignments for the 2 monitor channels. The loudspeaker options are also set here.

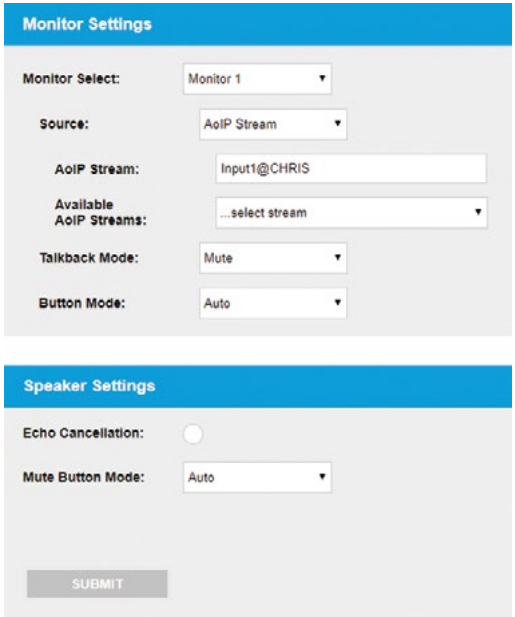

Fig 13-19: Audio - Monitor & Speaker Settings Web Page

Monitor Select – This drop down list selects the monitor channel to view and change settings.

Source – This option selects the audio source for the selected monitor channel. This can be a local audio input or an AoIP stream.

Audio Input – This option selects the local audio input for the selected monitor channel when the source is set to audio input. This option is unavailable when the source is set AoIP stream.

AoIP Stream – This setting is the name of the AoIP stream that will be routed to the selected monitor channel when the source is set to AoIP stream. The stream name is comprised of the AoIP stream name prefix and the intercom name. For example, if the required AoIP stream, called "Input1", is being transmitted from an intercom unit with the name "DEMO-SOURCE", the AoIP stream name would be "Input1@DEMO-SOURCE". This name can be entered manually, or it can be selected from the dynamic list of available AoIP streams detailed below. This option is unavailable when the source is set to audio input.

Available AoIP Streams – This dynamic drop down list shows the AoIP streams that are currently visible on the network. Selecting an AoIP stream from this list places the name into the AoIP stream edit box. This option is unavailable when the source is set to audio input.

Talkback Mode – This setting controls how the audio level of the selected monitor is modified when an intercom channel is active. There are 3 options, the level of the monitor is either muted, reduced by 75% or reduced by 50%.

Button Mode – Selects the operational mode of the selected monitor channel button on the front panel. In momentary mode, the monitor channel is active whilst the button is pressed. In latching mode, a separate button press is required to enable and disable the monitor channel. In auto mode, the button has a momentary action that will automatically latch on or off if pressed briefly. The button can also be disabled if required.

Echo Cancellation – This setting enables the acoustic echo cancellation algorithm which removes audio content, originating from the speaker, from the microphone input. This prevents a remote user from hearing a delayed echo of themselves.

Mute Button Mode – This setting selects the operational mode of the speaker mute button on the front panel. In momentary mode, the speaker is muted whilst the button is pressed. In latching mode, a separate button press is required to enable and disable the speaker mute. In auto mode, the button has a momentary action that will automatically latch on or off if pressed briefly. The button can also be disabled if required with the mute permanently on or off.

Audio - Monitor & Speaker Settings Defaults

Monitor 1:	
Source:	Audio Input
Audio Input:	Input 1
AoIP Stream:	not set
Talkback Mode:	Mute
Button Mode:	Auto
Monitor 2:	
Source:	Audio Input
Audio Input:	Input 2
AoIP Stream:	not set
Talkback Mode:	Mute
Button Mode:	Auto
Echo Cancellation:	Disabled
Mute Button Mode:	Auto

Audio - Advanced Settings Web Page

The Advanced Settings page is located under the Audio sub tab and it shows the current operating mode of the unit.

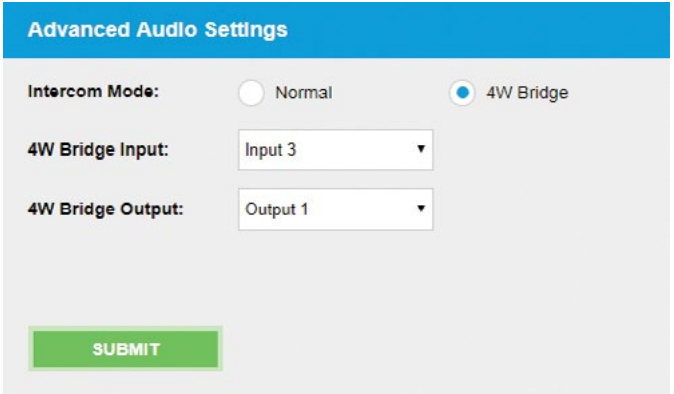

Fig 13-20: Audio - Advanced Settings Web Page

Intercom Mode – For connected calls in normal mode, the remote devices hear the local microphone audio, and the audio received from the remote devices is mixed and routed to the headphone and loudspeaker. In 4W bridge mode, remote devices hear the audio presented at the selected local audio input. The audio received from the remote devices is mixed and routed to the selected local audio output.

4W Bridge Input – This option selects the local audio input that is the source for calls to remote units.

4W Bridge Output – This option selects the local audio output where call audio from remote units is routed.

Audio - Advanced Settings Defaults

Intercom Mode:	Normal
4W Bridge Input:	Input 3
4W Bridge Output:	Output 3

Page Settings Web Page

The Page Settings page shows the current list of target intercoms for the page function. The page priority setting is also set here.

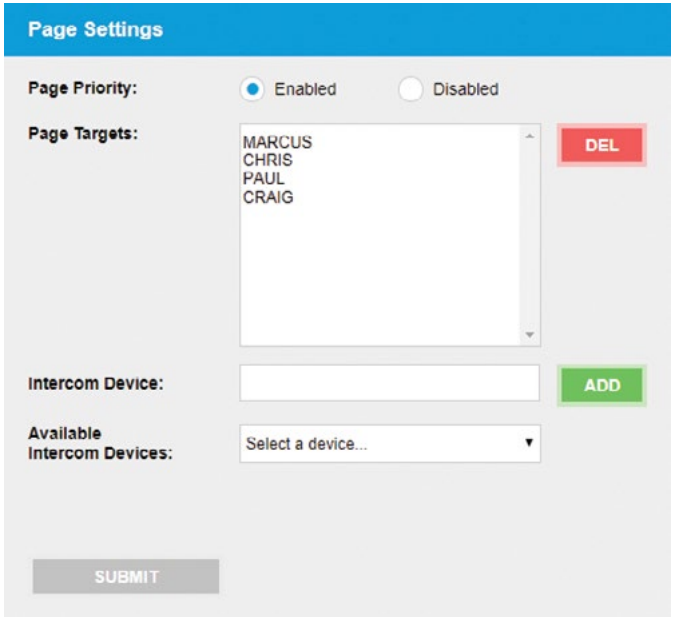

Fig 13-21: Page Settings Web Page

Page Priority – This option selects whether received page calls have priority over current intercom calls. If this option is enabled, all current intercom calls will be suspended during a received page call. These calls will then resume once the page call has ended. If a page request is received and this option is disabled, the page will be ignored.

Page Targets – This is the list of up to 20 target intercom devices that will be called during a page. To add to the list, select a new device from the available intercom devices select box or manually enter the name in the intercom devices edit and then press the ADD button. To remove one or more targets from the list, highlight the required page targets and press the DEL button. Multiple targets can be selected by clicking the required options whilst holding down the Ctrl key on the keyboard.

Intercom Device – A new page target intercom name can be added to the list by entering the name in this edit box and pressing the ADD button. Alternatively, it can be selected from the dynamic list of available intercom devices detailed below.

Available Intercom Devices – This dynamic drop down list shows the intercom units that are currently visible on the network and available for selection as a page target. Selecting an intercom from this list places the intercom name into the intercom device edit box.

Page Settings Defaults

Page Priority:	Enabled
Page Targets:	none set

GPIO Settings Web Page

The GPIO Settings page shows the current configuration of the 10 GPIO ports on the rear panel mounted GPIO connector. The functions, inputs and states that control the output relay are also set here.

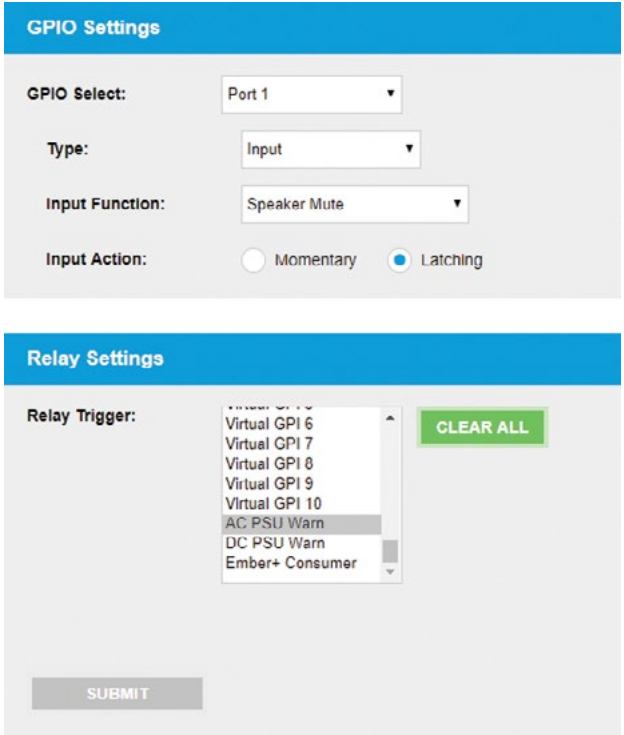

Fig 13-22: GPIO Settings Web Page

GPIO Select – This drop down list selects the GPIO port to view and change settings.

Type – This option selects whether the selected GPIO port is an input, an output or whether it is disabled. Ports configured as inputs are active low and they have a weak internal pullup. Ports configured as outputs are open collector driven and are therefore active low.

Input Function – This option selects the required function that the selected GPIO input controls. GPIO inputs can be set to control a local function and a virtual GPIO output if required. If no local function control is needed, but control of a virtual GPIO output is required, select Virtual GPO Only. The assignment for which GPIO input triggers a virtual GPIO output is made on the virtual GPIO settings page. This option is only available when the type is set to input.

Input Action – This option selects the operational mode of the selected GPIO input. In momentary mode, the selected function is active whilst the input is asserted low. In latching mode, a separate active low pulse is required to enable and disable the selected function. This option is only available when the type is set to input.

Output Trigger – This option selects the function that triggers the GPIO output. GPIO outputs can be triggered by a local function, a virtual GPIO input and an Ember+ consumer if required. If a local function trigger is not needed, but a virtual GPIO input or Ember+ consumer trigger is required, select Virtual GPI Only or Ember+ Consumer Only. The assignment for which virtual GPIO input triggers a GPIO output is made on the virtual GPIO settings page. This option is only available when the type is set to output.

Relay Trigger – The output relay, which provides both normally open and normally closed contacts on the GPIO connector on the rear panel, can be triggered by multiple functions, inputs and states. When any of the selected triggers become active, the relay is switched on. To select multiple triggers, click on the required options whilst holding down the Ctrl key on the keyboard. The CLEAR ALL button provides an easy way to remove all current relay trigger assignments.

GPIO Settings Defaults

GPIO Port 1 – 10:

Type:	Disabled
Input Function:	Talk/Listen 1
Input Action:	Latching
Output Trigger:	Talk/Listen 1

Virtual GPIO Settings Web Page

The Virtual GPIO Settings page shows the current configuration of the 10 virtual GPIO ports. These virtual ports provide a method of controlling functions on remote units by linking a virtual output on one unit to a virtual input on another unit.

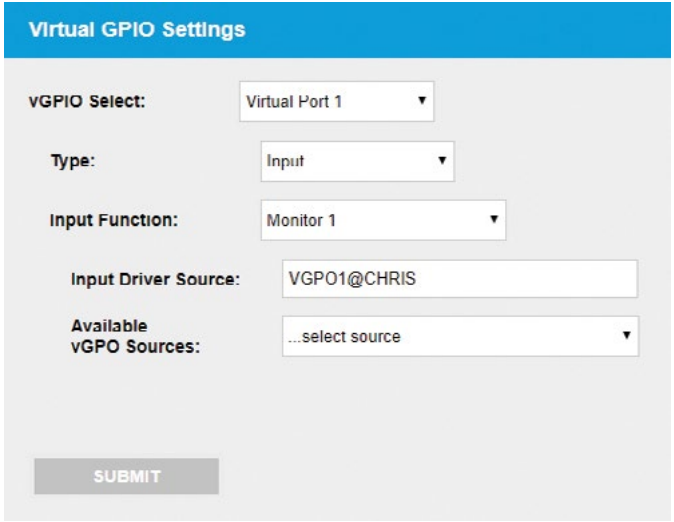

Fig 13-23: Virtual GPIO Settings Web Page

vGPIO Select – This drop down list selects the virtual GPIO port to view and change settings.

Type – This option selects whether the selected virtual GPIO port is an input, an output or whether it is disabled.

Input Function – This option selects the required function that the selected virtual GPIO input controls. This list includes GPO 1 – 10 which allows a virtual GPIO input to control a physical GPIO output. Please note, if a GPO port is selected, the type option for that GPIO port must be set to output. This option is only available when the type is set to input.

Input Driver Source – This setting is the name of the virtual GPIO output on a remote unit that will drive the virtual GPIO input on this unit. These source names are generated automatically based on the source type and the intercom name. For example, for virtual output port 1 on an intercom unit with the name “DEMO-SOURCE”, the source name would be “VGPO1@DEMO-SOURCE”. This name can be entered manually, or it can be selected from the dynamic list of available vGPO sources detailed below. This option is only available when the type is set to input.

Available vGPO Sources – This dynamic drop down list shows the virtual GPIO outputs that are currently visible on the network and available for selection as an input driver source. Selecting a source from this list places the virtual GPIO output name into the input driver source edit box. This option is only available when the type is set to input.

Output Trigger – This option selects the function that triggers the virtual GPIO output. This list includes GPI 1 – 10 which allows a physical GPIO input to trigger a virtual GPIO output. Please note, if a GPI port is selected, the type option for that GPIO port must be set to input otherwise the virtual output will not be triggered. This option is only available when the type is set to output.

Virtual GPIO Settings Defaults

Virtual GPIO Port 1 – 10:

Type:	Disabled
Input Function:	Talk/Listen 1
Input Driver Device:	not set
Output Trigger:	Talk/Listen 1

Miscellaneous Settings Web Page

The Miscellaneous Settings page shows general settings for the connected unit.

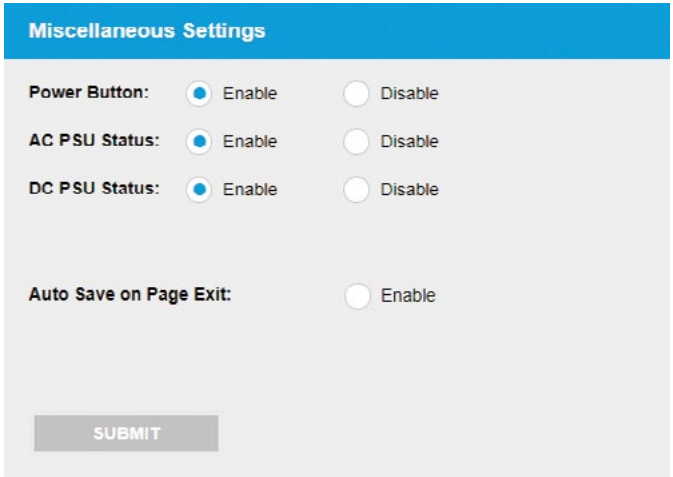

Fig 13-24: Miscellaneous Settings Web Page

Power Button – This option enables the power button on the front panel of the unit. When enabled, pressing the power button will switch the unit on and off.

AC PSU Status – This option enables the AC PSU status LED on the front panel.

DC PSU Status – This option enables the DC PSU status LED on the front panel.

Auto Save on Page Exit – With this option enabled, any changes made on a page are automatically saved when leaving the page. This option is initially cleared when connecting to a device.

Miscellaneous Settings Defaults

Power Button:	Enabled
AC PSU Status:	Enabled
DC PSU Status:	Enabled

System Tab

The System Tab allows the firmware on the unit to be updated. It also has options for saving and loading the configuration settings to and from a local file.

Update Firmware

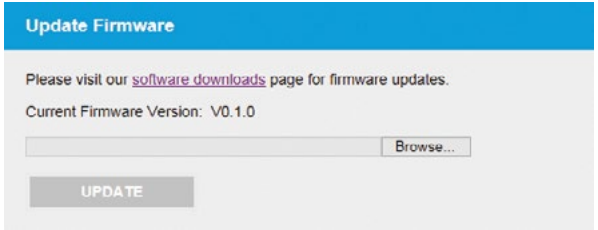

Fig 13-25: Update Firmware Section of Update Web Page

The current firmware version on the connected unit is shown. New versions of firmware will be released as new features are added, and when any bug fixes are completed. Click on the software downloads link to visit the Sonifex web page. If an update is available, download the latest file. The file will have the '.SWU' extension. Click the select button and browse your computer to locate the new firmware file and then click Update. Once the file has been successfully uploaded to the unit, the firmware update process will begin. This takes approximately 1 minute during which time a progress page will be shown. Once the update is complete, the device information page will be shown automatically.

Save Configuration To A File

Fig 13-26: Save Configuration Section of Update Web Page

The current configuration of the connected unit can be saved to a local xml file by clicking the SAVE CONFIG button. The filename will be "avn_tb_" followed by the intercom name. It is a good idea to save the configuration settings of all units on the network once they have been setup as this provides a quick and easy way of returning the units to a known working condition.

Load Configuration From A File

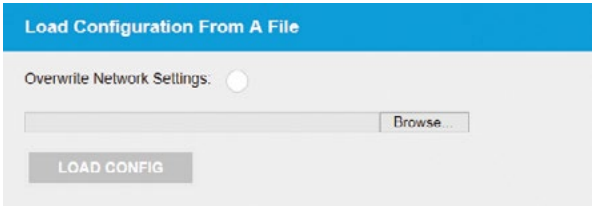

Fig 13-27: Load Configuration Section of Update Web Page

The current configuration of the connected unit can be overwritten from a previously saved xml file. The settings on the network page will only be updated if the Overwrite Network Settings check box is ticked. Click the select button and browse your computer to locate the required configuration file and then click the LOAD CONFIG button. Once the file has been successfully uploaded, the unit will automatically reboot to load

the new configuration settings. This takes approximately 25 seconds during which time a progress page will be shown. If the settings of the network port the browser is connected to have changed, a new connection will need to be made once the unit has restarted. Otherwise, the device information page will be shown automatically.

Save System Log To A File

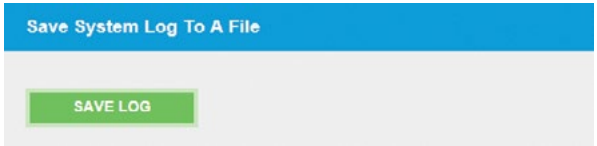

Fig 13-28: Save System Log Section of Update Web Page

The AVN-TB10AR maintains an internal log that records system events and errors. This information may be useful for resolving configuration issues. To download the log, click the SAVE LOG button.

14. System Menu

The easiest method for reviewing and changing the system configuration options is via the web server. The system menu, accessed via the front panel, offers alternative access to a limited set of options and information. Each layer of the menu structure, apart from the main menu, has a “BACK” option which ascends to the parent menu. All layers have a “CLOSE” option which exits the system menu and returns to the main screen.

Fig 14-1: Main System Menu

To access the system menu, press and hold the navigation centre button until the menu appears. The left side of the screen shows the current position within the menu structure. The right side shows the available menu selections or the options related to the current menu item. To navigate through the menu, use either the navigation outer up/down buttons or the inner rotary control to highlight the required item and then press either the centre button or the right button. The configuration settings within the menu structure are presented either as a list of options with a tick to show the current selection, or in a dialog which allows the value to be changed. For listed options, highlight the required setting and press the centre button to select. Options presented in a dialog give guidance on how to change the current setting. To ascend up the menu structure, press either the navigation outer left button, or select the option “back”.

The following section gives a brief description of the options within each branch of the menu tree. This is a condensed list such that menu items with similar associated options have only been listed once.

System Info

The read only System Info menu item shows important information about the unit. In particular, it shows the current IP addresses for the 2 network ports which will be required when accessing the embedded web server. The PTP clock information is dynamic and shows the current status, whether Listening, Master or Slave. The ID of the current master is shown as well as the clock offset from the master clock if the unit is currently a slave. The state of the AC-DC power supply and DC power input is also dynamic and shows the voltage level of each supply.

System Info →
- Intercom Name
- Intercom Mode
- Ethernet Port
  - Network Mode
  - IP Address
  - Subnet Mask
  - MAC Address
- AoIP Port
  - Network Mode
  - IP Address
  - Subnet Mask
  - MAC Address
- PTP Clock
  - Status
  - Domain Number
  - Master ID
  - Master Offset
- AC-DC Voltage
- DC Voltage
- Version
- FP Version
- PM Version
- Serial
- SOM Serial

Network

The Network menu area allows the settings for the Ethernet and AoIP network ports to be edited. These settings correspond to the options shown on the Network web page on page 27. For a description of these items, please refer to that section.

- Network
  - → Ethernet Port
  - → AoIP Port
    - → Address Mode
    - → Static Settings
      - → IP Address
      - → Subnet Mask
      - → Back
      - → Close
    - → Back
    - → Close
  - → Auto Multicast
  - → Back
  - → Close

AoIP Setup

The AoIP Setup menu area allows the transmit AoIP streams to be edited. These settings correspond to the transmit options shown on the AoIP Streams web page on page 29. For a description of these items, please refer to that section.

- AoIP Setup
  - → Mic Stream
    - → DSCP
    - → IP Address
    - → Back
    - → Close
  - → Input 1 Stream
  - → Input 2 Stream
  - → Input 3 Stream
  - → S/PDIF Stream
    - → DSCP
    - → IP Address
    - → Mute
    - → Trim
    - → Back
    - → Close
  - → IFB 1 Stream
  - → IFB 2 Stream
  - → IFB 3 Stream
    - → DSCP
    - → IP Address
    - → Mute
    - → Trim
    - → Source
    - → Audio Input
    - → Talkback Mode
    - → Back
    - → Close
  - → Back
  - → Close

PTP Setup

The PTP Setup menu area allows all of the PTP options to be edited. These settings correspond to the options shown on the PTP Profiles web page on page 31. For a description of these items, please refer to that section.

- PTP Setup
  - → DSCP
  - → Sync Clock
  - → Active Profile
  - → Profiles
    - → Default
    - → AES67 Media
    - → Custom
      - → Edit
        - → Domain Number
        - → Priority 1
        - → Priority 2
        - → Announce Int
        - → Announce Timeout
        - → Sync Int
        - → Min Dly Req Int
        - → Min PDly Req Int
        - → Slave Only
        - → Delay Mechanism
        - → Back
        - → Close
      - → Reset To Defaults
      - → Back
      - → Close
    - → Back
    - → Close
  - → Back
  - → Close

Display

The Display menu area allows the screen contrast and LED brightness to be adjusted and the colours assigned to the listed front panel buttons to be changed. These settings correspond to the options shown on the Front Panel - Display Settings web page and the Front Panel - Colour Settings web page on page 37. For a description of these items, please refer to those sections. For the screen contrast and LED brightness options, a dialog will be shown on the display and the current setting is changed by turning the navigation rotary control. The value is represented on screen with a sliding scale.

- Display
  - → Screen Contrast
  - → LED Brightness
  - → Function Colours
    - → Talk
    - → Listen
    - → Talk+Listen
    - → User1
    - → User2
    - → User3
    - → Back
    - → Close
  - → Back
  - → Close

Mic Setup

The Mic Setup menu area allows all of the microphone related options to be edited. These settings correspond to the options shown on the Audio - Microphone Settings web page on page 39. For a description of these items, please refer to that section.

Mic Setup → Type
→ Gain Control
→ Mic Gain
→ Headset Gain
→ AGC Target Level
→ Talkback Mode
→ Auto Mute
→ Mute Button Mode
→ Back
→ Close

Audio Inputs

The Audio Inputs menu sets the mode for the third audio input and the audio line up level. These settings correspond to the options shown on the Audio - Input Settings web page on page 40. For a description of these items, please refer to that section.

Audio Inputs → Audio I/O 3
→ Audio Lineup
→ Back
→ Close

Audio Outputs

These settings correspond to the options shown on the Audio - Output Settings web page on page 41. For a description of these items, please refer to that section.

Audio Outputs → Output 1-3 → Source
→ Active
→ Back
→ Close
→ Audio Lineup
→ Back
→ Close

Advanced Audio

The Advanced Audio menu area selects the intercom mode of the unit. These settings correspond to the options shown on the Audio - Advanced Settings web page on page 43. For a description of these items, please refer to that section.

Advanced Audio → Intercom Mode
→ 4W Bridge Input
→ 4W Bridge Output
→ Back
→ Close

Channels

The Channels menu area selects the connection type and intercom assignments for the 10 channel buttons on the front panel. These settings correspond to the options shown on the Channels - Advanced Settings web page on page 34. For a description of these items, please refer to that section.

Channels → Channel 1-10 → Connection
→ AoIP Stream → IFB Mode
→ Auto Answer
→ Back
→ Close
→ Default Function
→ Button Mode
→ Back
→ Close
→ Call Signal
→ Back
→ Close

Group Talk

The Group Talk menu area selects the members of the 3 group talk sets. These settings correspond to the options shown on the Group Talk web page on page 36. For a description of these items, please refer to that section.

Group Talk → Group Talk 1
→ Group Talk 2
→ Group Talk 3
→ Back
→ Close

Page Priority

The Page Priority menu item corresponds to the same option shown on the Page Settings web page on page 44. For a description of this item, please refer to that section. The page targets option is not available in the system menu.

Page Priority

GPIO

The GPIO menu area selects the type and functionality of the 10 GPIO ports. These settings correspond to the options shown on the GPIO Settings web page on page 45. For a description of these items, please refer to that section.

- GPIO
  - → Port 1-10
    - → Type
    - → Input Function
    - → Input Action
    - → Output Trigger
    - → Back
    - → Close
  - → Relay
  - → Back
  - → Close

Virtual GPIO

The Virtual GPIO menu area selects the type and functionality of the 10 virtual GPIO ports. These settings correspond to the options shown on the Virtual GPIO Settings web page on page 46. For a description of these items, please refer to that section.

- Virtual GPIO
  - → Virtual Port 1-10
    - → Type
    - → Input Function
    - → Output Trigger
    - → Back
    - → Close
  - → Back
  - → Close

User Buttons

The User Buttons menu area selects the functions controlled by the 3 front panel user button. These settings correspond to the options shown on the Front Panel - User Buttons web page on page 38. For a description of these items, please refer to that section.

- User Control
  - → User 1-3
    - → Function
    - → Button Mode
    - → Back
    - → Close
  - → Back
  - → Close

Monitor Control

The Monitor Control menu area selects the sources and functionality of the 2 monitor channels. These settings correspond to the options shown on the Audio - Monitor & Speaker Settings web page on page 42. For a description of these items, please refer to that section.

- Monitor Control
  - → Mon 1 Setup
  - → Mon 2 Setup
    - → Source
    - → Audio Input
    - → Talkback Mode
    - → Button Mode
    - → Back
    - → Close
  - → Speaker
    - → Echo Cancel
    - → Mute Button Mode
    - → Back
    - → Close
  - → Back
  - → Close

System

The System menu area allows the system date and time to be set. Selecting these options will display an edit dialog with ok and cancel options. To edit the date or time, press the navigation control up button. The highlight moves up to the left hand field. Use the navigation control left/right buttons to select the field to edit and then use the navigation rotary control to edit the value. Once editing is complete, use the down button to move the highlight back to the ok and cancel options. Highlight the required option using the left/right buttons and press the central button to select. The power button and the PSU status items corresponds to the same options shown on the Miscellaneous Settings web page on page 47. For a description of these items, please refer to that section.

System
- → Set Date
- → Set Time
- → Power Button
- → AC PSU Status
- → DC PSU Status
- → Back
- → Close

15. Ember+ Interface

The AVN-TB10AR has an Ember+ provider interface that exposes all of the available configuration options. It also allows the GPIO, virtual GPIO and the relay to be controlled and monitored by an external Ember+ consumer.

Connecting to the Ember+ Interface

The Ember+ interface is available on both the Ethernet and AoIP network connections using port 9000. The structure of the provider tree is similar to the system menu on the front panel. Each of the configuration options are arranged in branches grouped by function. Fig 15-1 shows a graphical representation of the root structure.

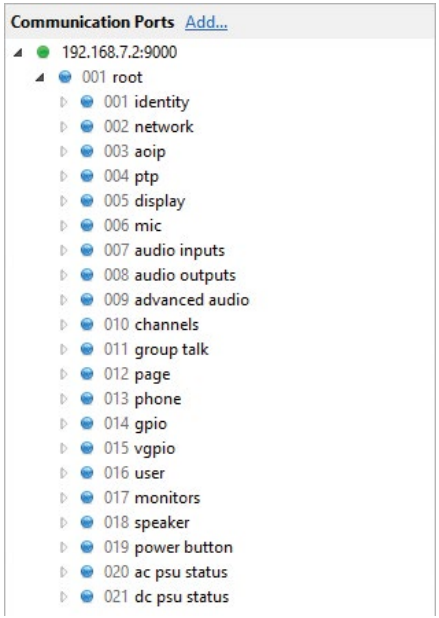

Fig 15-1: Ember+ Root Structure

The configuration options are presented as either string or integer types. Where possible, integer type options are enumerated to indicate what each of the setting values are. A full list of the options and values associated with each of the Ember+ parameters on page 60.

GPIO Control

The GPIO branch of the Ember+ provider interface allows monitoring and control of the GPIO ports. Fig 15-2 shows the structure of each GPIO child node. For further details on the configuration options, refer to page 45.

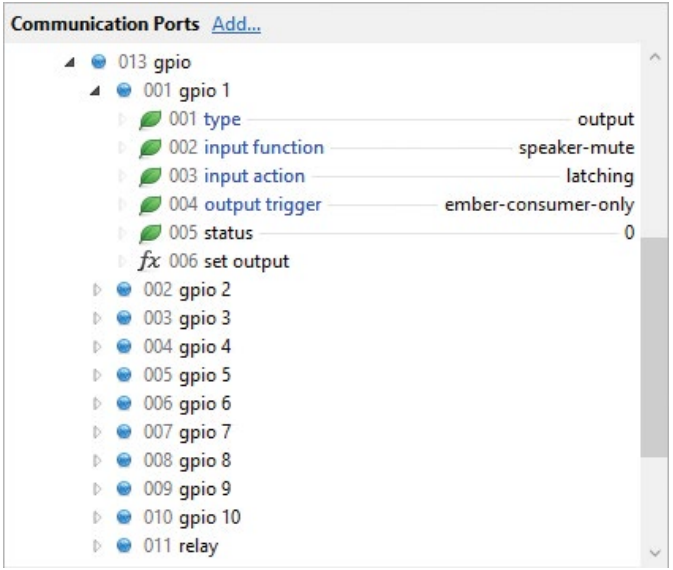

Fig 15-2: Ember+ GPIO Branch Structure

Each GPIO node has a similar set of parameters and functions:

Type Parameter

This enumerated integer parameter is the current port type.

Input Function Parameter

This enumerated integer parameter is the function that will be activated when the GPIO port, when configured as an input, is asserted.

Input Action Parameter

This enumerated integer parameter is the operational mode of the GPIO port when it is configured as an input.

Output Trigger Parameter

This enumerated integer parameter is the function that triggers the GPIO port when it is configured as an output.

Status Parameter

The status parameter shown in Fig 15-2 represents the current state of the GPIO port regardless of whether the port is set as an input or an output. A value of 0 indicates the port is inactive and a 1 indicates the port is active. This parameter has an Ember+ stream identifier and a consumer can subscribe to it to automatically receive updates when the status changes.

Set Output Function

The set output leaf shown in Fig 15-2 is a function that allows control of the GPIO port when it is configured as an output. The function exposes 1 argument called enable, which is an integer with a value of non-zero for assert the output, or zero for de-assert the output. The function also returns an integer result which reflects the current state of the GPIO output similar to the status parameter. A return value of -1 indicates there was an error invoking the function. If the GPIO port is configured as an input, invoking this function has no effect.

Relay

The relay node of the GPIO branch of the Ember+ provider interface allows the relay to be controlled by an Ember+ consumer. Fig 15-3 shows the structure of the relay node. For further details on the configuration options, refer to page 45.

Fig 15-3: Ember+ Relay Node Structure

Trigger Parameter

This integer parameter is the combined value of the functions that trigger the operation of the relay. As multiple trigger functions can be selected, the value shown is the sum of the selected items corresponding values.

Status Parameter

The status parameter shown in Fig 15-3 represents the current state of the relay. A value of 0 indicates the relay is inactive and a 1 indicates the relay is active. This parameter has an Ember+ stream identifier and a consumer can subscribe to it to automatically receive updates when the status changes.

Active Function

The active leaf shown in Fig 15-3 is a function that allows control of the relay. The function exposes 1 argument called enable, which is an integer with a value of non-zero to activate the relay, or zero to de-activate the relay. The function also returns an integer result which reflects the current state of the relay similar to the status parameter. A return value of -1 indicates there was an error invoking the function.

Virtual GPIO Control

The virtual GPIO branch of the Ember+ provider interface allows monitoring and control of the virtual GPIO ports. Fig 15-4 shows the structure of each virtual GPIO child node. For further details on the configuration options, refer to page 46.

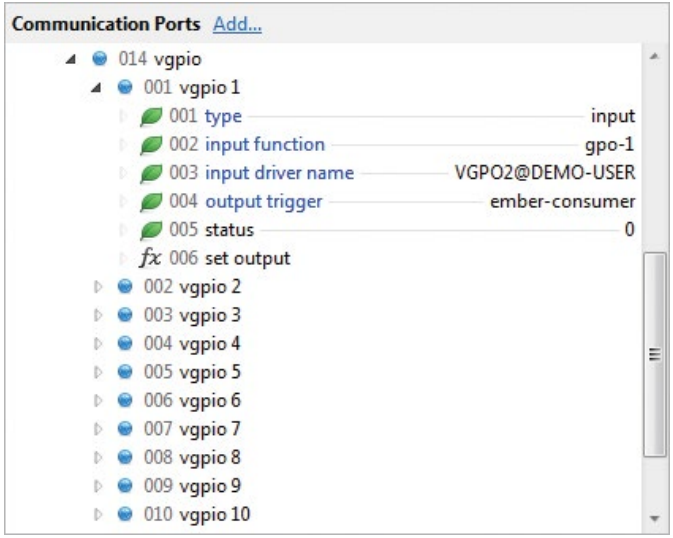

Fig 15-4: Ember+ Virtual GPIO Branch Structure

Type Parameter

This enumerated integer parameter is the current virtual GPIO port type.

Input Function Parameter

This enumerated integer parameter is the function that will be activated when the virtual GPIO port, when configured as an input, is asserted.

Input Driver Name

This string parameter is the name of the virtual output on a remote unit that drivers the virtual GPIO port when it is configured as an input.

Output Trigger Parameter

This enumerated integer parameter is the function that triggers the GPIO port when it is configured as an output.

Status Parameter

The status parameter shown in Fig 15-4 represents the current state of the virtual GPIO port regardless of whether the port is set as an input or an output. A value of 0 indicates the port is inactive and a 1 indicates the port is active. This parameter has an Ember+ stream identifier and a consumer can subscribe to it to automatically receive updates when the status changes.

Set Output Function

The set output leaf shown in Fig 15-4 is a function that allows control of the virtual GPIO port when it is configured as an output. To use this function, the output trigger must be set to ember-consumer, and at least one remote unit must have this output selected as the driver for a virtual input. The function exposes 1 argument called enable, which is an integer with a value of non-zero for assert the output, or zero for de-assert the output. The function also returns an integer result which reflects the current state of the virtual GPIO output similar to the status parameter. A return value of -1 indicates there was an error invoking the function. If the virtual GPIO port is configured as an input, invoking this function has no effect.

Parameter Value Specification

This section provides a full list of the possible parameter values for the Ember+ interface. Integer values with the '0x' prefix are in hexadecimal format.

Identity

This is a read only leaf that describes the unit.

Network

Intercom Name (string):	Max length = 23 characters
Address Mode (integer):	0 = Dynamic, 1 = Static
IP Address (string):	Max length = 15 characters
Subnet Mask (string):	Max length = 15 characters
Auto Multicast (integer):	0 = Disable, 1 = Enable

AoIP

Stream Name (string):	Max length = 12 characters
DSCP (integer):	Min = 0, Max = 56
IP Address (string):	Max length = 15 characters
Mute (integer):	0 = Disable, 1 = Enable
Trim (integer):	0 - 24 = -12dB to +12dB
Source (integer):	0 = Audio Input 1 = AoIP Stream
Audio Input (integer):	0 - 2 = Input 1 - 3 3 = S/PDIF
AoIP Stream (string):	Max length = 36 characters
Talkback Mode (integer):	0 = Mute 1 = Dim 75% 2 = Dim 50%

PTP

DSCP (integer):	Min = 0, Max = 56
Sync Clock (integer):	0 = Disable, 1 = Enable
Profile (integer):	0 = Default 1 = AES67 Media 2 = Custom
Domain Number (integer):	Min = 0 Max = 127 (Default Profile) Max = 255 (AES67 Media Profile) Max = 255 (Custom Profile)
Priority 1 (integer):	Min = 0, Max = 255
Priority 2 (integer):	Min = 0, Max = 255
Announce Interval (integer):	Setting value is 2integer Min = 0 (Default Profile) Min = 0 (AES67 Media Profile) Min = -3 (Custom Profile) Max = 4
Announce Timeout (integer):	Min = 2, Max = 10
Sync Interval (integer):	Setting value is $2^{integer}$ Min = -1, Max = 1 (Default Profile) Min = -4, Max = 1 (AES67 Media Profile) Min = -7, Max = 6 (Custom Profile)
Min Delay Request Interval (integer):	Setting value is $2^{integer}$ Min and Max values in following read only parameters
Min Delay Request Interval Min (integer):	Value is read only Greater of 0 or Sync Interval (Default Profile) Greater of -3 or Sync Interval (AES67 Media Profile) Sync Interval (Custom Profile)
Min Delay Request Interval Max (integer):	Value is read only Lesser of 5 or Sync Interval+5 (Default Profile) Lesser of 5 or Sync Interval+5 (AES67 Media Profile) Sync Interval+5 (Custom Profile)
Min Peer Delay Request Interval (integer):	Setting value is $2^{integer}$ Min = 0, Max = 5 (Default Profile) Min = 0, Max = 5 (AES67 Media Profile) Min = -1, Max = 5 (Custom Profile)
Slave Only (integer):	0 = Disable, 1 = Enable
Delay Mechanism (integer):	0 = E2E, 1 = P2P
Reset Defaults (integer):	1 = Reset Profile to Default Values

Display

Screen Contrast (integer):	Min = 1, Max = 64
LED Brightness (integer):	Min = 1, Max = 7
Talk Colour, Listen Colour, Talk Listen Colour, User 1 Colour, User 2 Colour, User 3 Colour (integer):	0 = Red 1 = Green 2 = Amber 3 = Blue 4 = Purple 5 = Light Blue 6 = White

Mic

Type (integer):	0 = Mic 1 = Phantom Mic 2 = Headset 3 = Phantom Headset
Gain Control (integer):	0 = Fixed, 1 = Automatic
Mic Gain (integer):	Actual Gain is value x 3 Min = 0, Max = 20
Headset Gain (integer):	Actual Gain is value x 3 Min = 0, Max = 20
AGC Target Level (integer):	0 = -24dBFS 1 = -21dBFS 2 = -18dBFS 3 = -15dBFS 4 = -12dBFS 5 = -9dBFS 6 = -6dBFS
Talkback Mode (integer):	0 = Mic Only 1 = Input 1 Mix 2 = Input 2 Mix 3 = Input 3 Mix 4 = SPDIF Mix
Auto Mute (integer):	0 = Disable, 1 = Enable
Mute Button Mode (integer):	0 = Auto 1 = Momentary 2 = Latching 3 = Disabled On 4 = Disabled Off

Audio Inputs

Audio In Out 3 (integer):	0 = Input 1 = Input & Output
Audio Lineup (integer):	0 = 0dBFS = +15dBu 1 = 0dBFS = +18dBu 2 = 0dBFS = +20dBu 3 = 0dBFS = +22dBu 4 = 0dBFS = +24dBu

Audio Outputs

Source (integer):	0 = Mic 1 - 3 = Input 1 - 3 4 = SPDIF 5 = AoIP Stream
AoIP Stream (string):	Max length = 36 characters
Active (integer):	0 = Disable, 1 = Enable
Audio Lineup (integer):	0 = 0dBFS = +15dBu 1 = 0dBFS = +18dBu 2 = 0dBFS = +20dBu 3 = 0dBFS = +22dBu 4 = 0dBFS = +24dBu

Advanced Audio

Intercom Mode (integer):	0 = Normal, 1 = 4W Bridge
Bridge Input (integer):	0 - 2 = Input 1 - 3 3 = SPDIF
Bridge Output (integer):	0 - 2 = Output 1 - 3

Channels

Connection (integer):	0 = AoIP Stream, 1 = Audio IO 3
AoIP Source (string):	Max length = 36 characters
IFB Mode (integer):	0 = Disable, 1 = Enable
IFB Source (string):	Max length = 12 characters
Auto Answer (integer):	0 = Disable, 1 = Enable
Default Function (integer):	0 = Talk & Listen 1 = Talk Only 2 = Listen Only 3 = Talk Forced Listen
Button Mode (integer):	0 = Auto 1 = Momentary 2 = Latching 3 = Disabled
Call Signal (integer):	0 = Disable, 1 = Enable

Group Talk

Group Talk 1, Group Talk 2, Group Talk 3 (integer):	Value is sum of required options 0x1 = Channel 1 0x2 = Channel 2 0x4 = Channel 3 0x8 = Channel 4 0x10 = Channel 5 0x20 = Channel 6 0x40 = Channel 7 0x80 = Channel 8 0x100 = Channel 9 0x200 = Channel 10

Page

Priority (integer):	0 = Disable, 1 = Enable
Target 1, Target 2, Target 3, Target 4, Target 5, Target 6, Target 7, Target 8, Target 9, Target 10, Target 11, Target 12, Target 13, Target 14, Target 15, Target 16, Target 17, Target 18, Target 19, Target 20 (string):	Max length = 23 characters

GPIO

Type (integer):
- 0 = Input
- 1 = Output
- 2 = Disabled

Input Function (integer):
- 0 - 9 = Talk/Listen 1 - 10
- 10 = Page
- 11 - 13 = User Button 1 - 3
- 14 - 15 = Monitor 1 - 2
- 16 = Speaker Dim
- 17 = Speaker Mute
- 18 = Microphone Mute
- 19 = Virtual GPO Only

Input Action (integer):
- 0 = Momentary
- 1 = Latching

Output Trigger (integer):
- 0 - 9 = Talk/Listen 1 - 10
- 10 = Any Talk/Listen
- 11 - 20 = Call Request 1 - 10
- 21 = Any Call Request
- 22 = Missed call
- 23 = Page
- 24 - 26 = User Button 1 - 3
- 27 - 28 = Monitor 1 - 2
- 29 = Speaker Dim
- 30 = Speaker Mute
- 31 = Microphone Mute
- 32 = Virtual GPI Only
- 33 = Ember+ Consumer Only

Relay

Trigger (integer): Value is sum of required options
- 0x1 = Talk/Listen 1
- 0x2 = Talk/Listen 2
- 0x4 = Talk/Listen 3
- 0x8 = Talk/Listen 4
- 0x10 = Talk/Listen 5
- 0x20 = Talk/Listen 6
- 0x40 = Talk/Listen 7
- 0x80 = Talk/Listen 8
- 0x100 = Talk/Listen 9
- 0x200 = Talk/Listen 10
- 0x400 = Call Request 1
- 0x800 = Call Request 2
- 0x1000 = Call Request 3
- 0x2000 = Call Request 4
- 0x4000 = Call Request 5
- 0x8000 = Call Request 6
- 0x10000 = Call Request 7
- 0x20000 = Call Request 8
- 0x40000 = Call Request 9
- 0x80000 = Call Request 10
- 0x100000 = Missed Call
- 0x200000 = Page
- 0x400000 = User Button 1
- 0x800000 = User Button 2
- 0x1000000 = User Button 3
- 0x2000000 = Monitor 1
- 0x4000000 = Monitor 2
- 0x8000000 = Speaker Dim
- 0x10000000 = Speaker Mute
- 0x20000000 = Mic Mute
- 0x40000000 = GPI 1

0x80000000 = GPI 2
0x100000000 = GPI 3
0x200000000 = GPI 4
0x400000000 = GPI 5
0x800000000 = GPI 6
0x1000000000 = GPI 7
0x2000000000 = GPI 8
0x4000000000 = GPI 9
0x8000000000 = GPI 10
0x10000000000 = Virtual GPI 1
0x20000000000 = Virtual GPI 2
0x40000000000 = Virtual GPI 3
0x80000000000 = Virtual GPI 4
0x100000000000 = Virtual GPI 5
0x200000000000 = Virtual GPI 6
0x400000000000 = Virtual GPI 7
0x800000000000 = Virtual GPI 8
0x1000000000000 = Virtual GPI 9
0x2000000000000 = Virtual GPI 10
0x4000000000000 = AC PSU Warn
0x8000000000000 = DC PSU Warn
0x10000000000000 = Ember+ Consumer

vGPIO

Type (integer):	0 = Input
	1 = Output
	2 = Disabled
Input Function (integer):	0 - 9 = Talk/Listen 1 - 10
	10 - 12 = User Button 1 - 3
	13 - 14 = Monitor 1 - 2
	15 = Speaker Dim
	16 = Speaker Mute
	17 = Microphone Mute
	18 - 27 = GPO 1 - 10
Input Driver Name (string):	Max length = 30 characters
Output Trigger (integer):	0 - 9 = Talk/Listen 1 - 10
	10 = Any Talk/Listen
	11 - 20 = Call Request 1 - 10
	21 = Any Call Request
	22 = Missed call
	23 = Page
	24 - 26 = User Button 1 - 3
	27 - 28 = Monitor 1 - 2
	29 = Speaker Dim
	30 = Speaker Mute
	31 = Microphone Mute
	32 - 41 = GPI 1 - 10
	42 = Ember+ Consumer

User

Function (integer):	0 - 2 = Group Talk 1 - 3 3 = Phone Control 4 = Clear Call Back 5 = GPO Control
Button Mode (integer):	0 = Auto 1 = Momentary 2 = Latching 3 = Disabled

Monitors

Source (integer):	0 = Audio Input 1 = AoIP Stream
Audio Input (integer):	0 - 2 = Input 1 - 3 3 = SPDIF
AoIP Stream (string):	Max length = 36 characters
Talkback Mode (integer):	0 = Mute 1 = Dim 75% 2 = Dim 50%
Button Mode (integer):	0 = Auto 1 = Momentary 2 = Latching 3 = Disabled

Speaker

Echo Cancel (integer):	0 = Disable, 1 = Enable
Mute Button Mode (integer):	0 = Auto 1 = Momentary 2 = Latching 3 = Disabled On 4 = Disabled Off

Power Button

Enable (integer):	0 = Disable, 1 = Enable

AC PSU Status

Enable (integer):	0 = Disable, 1 = Enable

DC PSU Status

Enable (integer):	0 = Disable, 1 = Enable

16. Technical Specifications

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:	Web server/Ember+
Format:	Linear PCM 24-bit (L24)
Channels Per Stream:	2
Frames Per Packet:	48
Transmit Streams:	5 (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

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 1 x S/PDIF, RCA phono
Audio Outputs:	1 x balanced stereo, RJ45 1 x unbalanced stereo, 2 x RCA phono 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 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, 10-14VDC
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

17. Additional Information

DSCP Name	DS Field Value	IP Precedence (Description)
CS0		0: Best effort
CS1, AF11-AF13	8, 10, 12, 14	1: Priority
CS2, AF21-AF23	16, 18, 20, 22	2: Immediate
CS3, AF31-AF33	24, 26, 28, 30	3: Flash (mainly used for voice signalling)
CS4, AF41-AF43	32, 34, 36, 38	4: Flash Override
CS5, EF PHB	40, 46	5: Critical (mainly used for RTP)
VOICE ADMIT	44	5: Critical (uses Call Admission Control)
CS6	48	6: Internetwork Control
CS7	56	7: Network Control

Table 17-1: DSCP Names & Their Corresponding IP Precedence

Table 17-1 shows the corresponding IP precedence for the DSCP names used in the PTP Profiles and AoIP Stream Settings. For more information on the types of DSCP name, see the following documents:

CS: Class Selector (RFC 2474)
AFxy: Assured Forwarding (x= class, y = Drop Precedence) (RFC 2597)
EF: Expedited Forwarding (RFC 3246)

SONIFEX
www.sonifex.co.uk
t:+44 (0)1933 650 700
f:+44 (0)1933 650 726
sales@sonifex.co.uk