



GUIDE

**AVN**

A Guide to Media Networking &  
Configuration of the AVN Product Range



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

**SONIFEX**

For the latest Sonifex handbook information please visit the  
Sonifex website at [www.sonifex.co.uk](http://www.sonifex.co.uk)

This is the AVN – A Guide to Media Networking & configuration of the AVN Product Range  
AW10846A, Stock Code: 30-220

Revision 1.03, May 2018  
©Sonifex Ltd, 2018  
All Rights Reserved

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](mailto:sales@sonifex.co.uk)  
Website: <http://www.sonifex.co.uk>

Information in this document is subject to change without notice and does not represent a  
commitment on the part of the vendor. Sonifex Ltd shall not be liable for any loss or damage  
whatsoever arising from the use of information or any error contained in this manual.

No part of this manual may be reproduced or transmitted in any form or by any means, electronic  
or mechanical, including photocopying, recording, information storage and retrieval systems, for any  
purpose other than the purchaser's personal use, without the express written permission of Sonifex  
Ltd. Unless otherwise noted, all names of companies, products and persons contained herein are  
part of a completely fictitious adaptation and are designed solely to document the use of Sonifex  
product.

# Contents

What is AVN?	1
AVN in the Audio World - AES67	2
Discovery	2
Audio Over IP - How it Works	2
Synchronisation - Using PTPv2	3
Buffering	4
Infrastructure	5
Setting the AVN Product Options	6
Information Page	7
Network Settings	8
PTP Profiles	10
AoIP Streams Web Page	13
References and Standards	15

## Figures

Fig 1-1:	AVN Overview Diagram	1
Fig 1-2:	The AVN Hierarchical Structure Diagram	4
Fig 1-3:	The Information Page	7
Fig 1-4:	Network Settings Page	8
Fig 1-5:	PTP Profiles Page	10
Table 1-1:	DSCP Names & Their Corresponding IP Precedence	12
Fig 1-6:	The AoIP Streams Web Page	13
Fig 1-7:	The Auto Multicast Option	14

## What is AVN?

The Sonifex AVN (Audio Video Network) product range is a collection of products designed to take advantage of the latest developments in network technology to provide a packet based system of transmitting live audio data using Internet protocols. This is known as Audio-Over IP (AoIP). AoIP is useful in environments where audio data needs to be transported such as a series of studios and galleries, office buildings, sports stadiums, theatres or houses of worship.

In traditional systems, a continuous flow of digitised data or even analogue data is sent down a single cable via a point-to-point connection between each device. To send the data to more than one destination, a router or distribution amplifier is required.

In the AoIP world, devices connect to a centralised Ethernet switch (or switches) and the media content of any source is chopped into pieces and

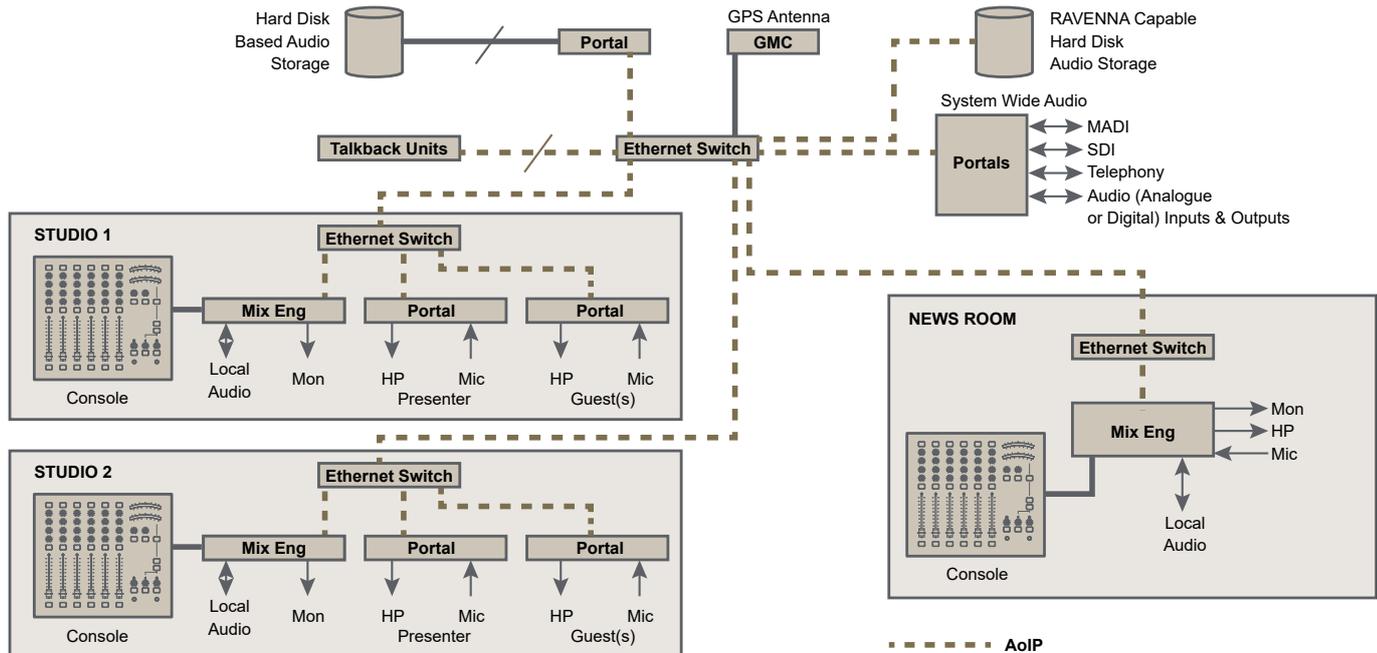


Fig 1-1: AVN Overview Diagram

sent in packets with an address header to determine the destinations. It is quite feasible to label a single packet to go to many destinations (multicasting) and this is handled by the Ethernet switch using IGMP (Internet Group Management Protocol). This means that a single network cable can carry many streams of distinct data in either direction. AVN devices could send audio to hundreds of different destinations and receive audio from hundreds of different sources on a single Ethernet connection.

An AoIP based system has a positive impact on the installation and modification of media systems. Each product only requires one cable to connect it to a network switch and re-routing data is simply arranged in software, rather than requiring possibly extensive rewiring with new cables, connectors and routing equipment. The cabling requirements for all devices is also identical – the ubiquitous Cat6 or Cat5e with RJ45 connectors – which are easily available and inexpensive, anywhere in the world.

Of course, added flexibility for interconnections comes at a management cost, and in addition to the packets containing actual data, there a number of further communication, control and synchronisation protocols that need to be handled for a fully implemented system. Our AVN products use several network protocols including: TCP, UDP, HTTP, RTP, RTSP, SDP, SAP, PTP, mDNS, DNS-SD and IGMP.

### **AVN in the Audio World - AES67**

Audio over IP can be achieved in a variety of ways for example RAVENNA, Dante, Livewire, and Cobranet. The majority of these existed as non-compatible solutions, but there is now an overarching solution through the AES67 standard which helps these competing systems to inter-operate. With the exception of Cobranet from the above, most AoIP solutions claim compatibility with AES67. There are some choices to be made about implementation, and some of the outlying methods are not, in general, being supported, but gradually the use of AoIP in a professional audio environment is becoming standardised on AES67. AVN products are based

on a RAVENNA solution, and we are committed to providing the extra protocols to be able to successfully connect with other AES67 products.

### **Discovery**

Discovery is the act of finding other equipment on the local network, and is essentially an ‘announcement of services supported’ protocol. When a device is first connected to a network, and periodically after that, it will send a broadcast message to the network as an “I’m here and can do this” notification.

We use the Bonjour protocol for Discovery – also known as mDNS, DNS-SD or Avahi. Bonjour is an Apple implementation and is embedded in their products. There is a download on our website to use it on PCs. We also have a small PC application – again downloadable from our website - that can report on the various Bonjour enabled network products that are available on your network. SAP is the other major method for discovery used in the professional audio AoIP world, it is simpler than Bonjour, but with probably less PC visibility. Our AVN products can also support SAP for discovering audio streams.

The Discovery mechanism includes an announcement of the services that the device will support, and in our AVN product range this will include the RAVENNA AoIP solution, in both default and AES67 modes; and a Sonifex specific service for network connected GPIO. Audio streams are also announced using SAP.

### **Audio Over IP - How it Works**

We can discover audio streams in 2 ways. The first method is using Bonjour to find compatible devices (AES67 or RAVENNA) which provide some audio streams for us to listen to. At this stage we don’t know what the streams contain or where they are (in IP address terms). So we interrogate the device using RTSP to obtain the SDP (Session Description Protocol) packets for the streams which allows us to find out what streams are available (in

terms of number of channels, sample rates, and sample size) and to what multicast IP address they are assigned. We can choose a suitable stream and again using RTSP we express an interest in using the stream – this can be considered as establishing a connection. AVN products, along with most other similar devices, continually send their streams regardless of the numbers of listeners, but it is entirely feasible that some products will only start sending audio once the RTSP connection has been established.

The second stream discovery method uses SAP (Session Announcement Protocol). Instead of using RTSP, the SDP packets for streams are announced on the network using SAP. Essentially we receive the same information via SAP that we receive using the Bonjour/RTSP method, they are 2 methods that can be used to achieve the same result.

At this point (whether the data was obtained via Bonjour/RTSP or SAP) we now know the stream contents, so can accept the RTP audio packets from the designated multicast address, can de-packetise the data, and then ‘play’ it out. We have successfully connected and received a stream, but that’s not quite the whole story. Unlike the analogue domain where data is effectively real time, or the digital audio domain where clocking is embedded in the data stream, we have no reference between the source device and the receiving device to ensure that the clocking is synchronous.

Early AoIP solutions used 2 main methods. The first relied on a distributed wordclock so audio data clocks were generated locally on both devices from the same source and were essentially synchronous – but this required extra cabling which cancels out a major benefit of AoIP which is the simplification of wiring requirements. The second used the incoming buffering as a crude clock error gauge – so as the buffer got to nearly empty it was assumed that the local clock was running too quickly and conversely as the buffer got to nearly full the local clock was running too slowly. This information was used to either adjust the clock to correct the data flow or, using clever algorithms to minimise audible artefacts, samples were discarded or added to correct the error. The downside is that the buffer needed to be considerably

bigger than the size of the data packets, so this increased latency (the gap between audio being sent by the source and it being played out at the receiver).

## Synchronisation - Using PTPv2

The latest AoIP implementations use the Precision Time Protocol – known as PTPv2 or the IEEE.1588-2008 standard - to synchronise independent clocks running in the source and destination. The PTP system was designed to provide highly accurate real time stamping of IP packets. To achieve this, the source clocks required a GPS receiver to capture and use the GPS satellite time information as a reference. These devices are known as Grand Master Clocks. The AVN-GMC which is part of the Sonifex AVN product range is designed for the particular requirements of the professional audio AoIP environment.

In an AoIP environment what we generally need is not the actual time, but synchronised time, so in small systems most devices can act as a Master Clock with no requirement for the time stamps to reflect the actual time. There is a prioritisation method defined in IEEE.1588 that sets the way that master clocks handle which device is designated as the master clock in a network system.

There are various flavours of PTP – called profiles – and these determine the regularity of the various requests and announcements made by and expected by the Master Clock. There are a number of pre-defined profiles that have been designed to meet certain requirements

There is also a hierarchical structure available, whereby a Master Clock supplies signals to what is known as a boundary clock, which in turn supplies PTP packets to other devices. Many top-end Ethernet switches have such an option and this has the benefit of reducing network overhead on both the network infrastructure and on the Master Clock. Usage of PTP requires a good network infrastructure and really only works across a LAN where IP packet deliveries are consistent – see the ‘Infrastructure’ section

for best practice. In solutions where you might want to use AoIP across a WAN or leased line connection or even broadband, then the 2 ends of the conversation would need independent Grand Master Clocks that are synchronised to the common GPS time.

### Buffering

AoIP is affected by the packetized and negotiated nature of IP, so we cannot expect the data to arrive at consistent time intervals. This means that we need to buffer data at the source device – to fill the designated packet size,

and at the receiver device to allow for the packet nature and the variation in timing of packets being received. There is a straightforward trade-off between latency and network overhead, when the packet size is larger, the buffering requirement increases and therefore latency is higher, but fewer packets are required so the network overhead is reduced – and vice-versa for smaller packet sizes. The various arrangements are again known as profiles and we always support the middle ground ‘Default’ profile as a norm, with other profiles available according to the device and its design parameters.

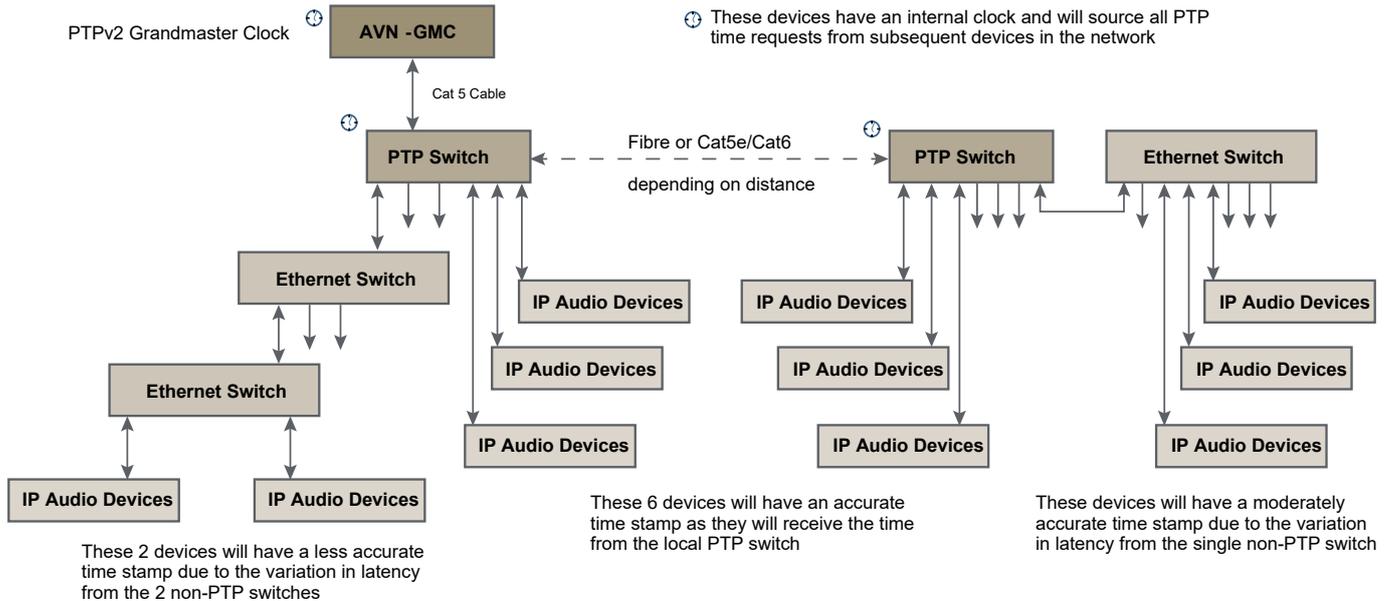


Fig 1-2: The AVN Hierarchical Structure Diagram

## Infrastructure

The infrastructure requirements of an AoIP solution are relatively simple – cabling, Ethernet Switches, and clock source. Cabling is straightforward and you should use Cat6 or Cat5e cables with standard pin to pin RJ45 connectors. Ethernet Switches are available with many different capabilities and the necessity for these depends on the complexity of your network. For a rough idea of the switch requirements based on the type of installation see the table below:

No Switch required	Just 2 devices connected to each other
10/100 unmanaged switch or hub	2 or 3 devices with low network overhead – some devices are 1Gbe only so will need a faster Ethernet switch
10/100 managed switch with L3 QoS (DiffServ) and IGMP Snooping	Multiple (<15) devices all with 10/100 connectivity (by definition these should be low network overhead)
10/100/1000 managed switch with L3 QoS (DiffServ) and IGMP Snooping	Multiple (<15) devices, some with 1Gbe connectivity – this is the minimum we would recommend for a non-laboratory installation
10/100/1000 managed switch with L3 QoS (DiffServ), IGMP Snooping and IEEE.1588 transparent time stamping	Multiple (>10) devices, or several devices with 1Gbe connectivity
10/100/1000 managed switch with L3 QoS (DiffServ), IGMP Snooping, and Boundary Clock	Multiple (>10) devices, or several devices with 1Gbe connectivity and with a number of intermediate switch elements to the Master Clock

As a rule we recommend that AoIP networks are kept separate from normal office networks - most managed switches these days have VLAN capability and running the AoIP and office on different VLANS would provide sufficient separation. The nature of office network with file transfers and live streaming can be network traffic heavy and may affect the successful audio transmission, and, conversely, office transfers may well be slowed during periods of high audio connectivity.

This might be an issue if you need to manage or monitor the status of AVN devices from office PCs. Consequently the higher spec AVN equipment has 2 Ethernet ports, the first for command and control (C & C) and the second for the AoIP connectivity (and C&C). For AVN devices with only one port then we suggest using a bridge or router to connect between the office and AoIP networks, which will minimise the traffic between them; or use a second network card, connected to the AoIP network; or attach a laptop to the AoIP network on occasions where you need to set up or monitor the AVN devices.

## Setting the AVN Product Options

All AVN products will have an embedded web server, which means that you can use a browser - Chrome, Firefox, Safari, Internet Explorer etc – to access the devices and check their status, change settings, or update firmware. The following web pages, with minor differences to suit specific products requirements, and the explanations, are applicable to all AVN products.

To access the embedded web server, the unit needs to be connected to a network via the network ports on the rear panel – where more than 1 port is available either port may be used and in this case, generally, 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 or only port IP address set to 192.168.0.100 and the second, lower port IP address set to 192.168.1.100 with both ports using a subnet mask of 255.255.255.0.

Ethernet Port IP Address: 192.168.0.100

AoIP Port IP Address: 192.168.1.100

Subnet Mask: 255.255.255.0.

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. Where the unit has a front panel with an OLED screen and rotary navigation control then the current IP address of these ports can be accessed by going to the main menu and selecting 'System Info' 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 Information page of the connected product 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 device identifier in the upper right hand corner, with the unique host name underneath. This makes it easier to identify the connected device, 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 Page

This page will show general information about the device – this is a typical page for connected equipment. The information may change from product to product, according to the power supply used by the device and the number of Ethernet ports available.

Device Information shows the identifiers for the device, the serial numbers of the relevant components, the firmware versions of software running on the device, the voltage status, a temperature reading and system timers.

PTP Clock Information provides information about the source of the PTP clock that is used to synchronise audio play-out. The source can be local (MASTER) or remote (SLAVE) and the domain and ID will identify the source and type. The offset quantifies the accuracy of the internally generated clock with respect to the source. All AVN devices can be a MASTER PTP source, but we would recommend that the priorities are set up so that the MASTER is a GMC or a higher powered central unit.

Network Information is displayed for each Ethernet port fitted and shows the MAC address and the currently used IP settings of that port.

<p><b>Device Information</b></p> <p><b>Device ID:</b> AVN-TB10AR  <b>Host Name:</b> AVN-TB10AR - 6089746  <b>Friendly Name:</b> DEMO-USER  <b>Unit Serial Number:</b> 6089746  <b>SOM Serial Number:</b> 4072051  <b>Firmware Version:</b> V1.0.0  <b>Front Panel Version:</b> V01  <b>Power Manager Version:</b> V01  <b>AC to DC Voltage:</b> 11.91V  <b>DC Input Voltage:</b> 0.0V  <b>Temperature:</b> 56°C  <b>System Up Time:</b> 1 hrs 54 mins  <b>System Time:</b> 22/12/16 11:33:22</p>	<p><b>Help: Device Information</b></p> <p>This page displays information about the connected device.</p> <p>It is important to record and send us this information when requesting technical support.</p>
<p><b>PTP Clock Information</b></p> <p><b>Status:</b> MASTER  <b>Domain Number:</b> 0  <b>Master ID:</b> 0050c2.ffe.05ac18  <b>Master Offset:</b> 0ns</p>	<p><b>Help: PTP Clock Information</b></p> <p>This section displays information about the PTP Clock.</p>
<p><b>Ethernet Network Information</b></p> <p><b>Hardware Address:</b> 00:50:C2:05:AC:17  <b>Actual IP Address:</b> 192.168.1.181  <b>Actual Subnet Mask:</b> 255.255.255.0  <b>Address Mode:</b> Static</p>	<p><b>Help: Ethernet Network Information</b></p> <p>This section displays information about the Ethernet network connection (upper connector) on the connected device. The Ethernet port is used as a control port to interrogate and modify the device settings without impacting on the audio network performance.</p>
<p><b>Audio Network Information</b></p> <p><b>Hardware Address:</b> 00:50:C2:05:AC:18  <b>Actual IP Address:</b> 192.168.16.181  <b>Actual Subnet Mask:</b> 255.255.255.0  <b>Address Mode:</b> Static</p>	<p><b>Help: Audio Network Information</b></p> <p>This section displays information about the Audio over IP network connection (lower connector) on the connected device. The AoIP port is used as a data port to provide audio transfer between devices. Ideally it should not be connected to the main network as system performance may be compromised by heavy external traffic.</p>

Fig 1-3: The Information Page

## Network Settings

- **Friendly Name** – The friendly name is a local 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 friendly name is the host name which is made from the device ID and the 7 digit product serial number i.e. AVN-TB10AR-1234567. The host name is unique, but the friendly is user settable and does not have to be unique – though, generally, this is a GOOD idea. When the device is addressed across the internet the host name is used, but for user interaction the host name is aliased to the device’s friendly name to ease identification.
- **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 response from a DHCP server is received. The actual IP address will be shown on the device information page or in the front panel menu if an OLED screen is fitted to the device. 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. This value is 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. This value is not used when the address mode is dynamic.

If any of the network configuration options are changed and submitted, 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.

**Device Name**

**Friendly Name:**

**Ethernet Network Port Settings**

**Address Mode:**  Dynamic  Static

**Static IP Address:**

**Static Subnet Mask:**

**Audio over IP Network Port Settings**

**Address Mode:**  Dynamic  Static

**Static IP Address:**

**Static Subnet Mask:**

Fig 1-4: Network Settings Page

**Network Defaults**

Friendly Name: DDDDDD-xxxxxxx  
Where DDDDDD is the device type  
and xxxxxx is the product serial  
number

Ethernet Port:  
Address Mode: Static  
Static IP Address: 192.168.0.100  
Static Subnet Mask: 255.255.255.0

AoIP Port or shared port: (where fitted)  
Address Mode: Static  
Static IP Address: 192.168.1.100  
Static Subnet Mask: 255.255.255.0

## PTP Profiles

The PTP Profiles web page show the currently selected PTP profile and allows the parameters in each profile to be edited. Similar changes can be made from the front panel when an OLED display and rotary navigation control is present.

**Active Profile** – There are 3 PTP profiles available: Default, AES 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 on page 12 – DSCP Names & Their Corresponding IP Precedence for more information.

**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.

The screenshot displays the PTP Profiles configuration interface. It is divided into two main sections: 'PTP Profiles' and 'PTP Profile Settings'.

**PTP Profiles Section:**

- Active Profile:** A dropdown menu currently set to 'Default Profile'.
- DSCP:** A dropdown menu currently set to 'EF PHB'.

**PTP Profile Settings Section:**

- Profile Select:** A dropdown menu currently set to 'Default Profile'.
- Reset to Defaults:** An unchecked checkbox.
- Delay Mechanism:** Two radio buttons: 'E2E Delay' (selected) and 'P2P Delay'.
- Announce Interval:** A dropdown menu set to '2secs'.
- Announce Timeout:** A numeric input field with a red minus button on the left and a green plus button on the right, currently set to '3'.
- Sync Interval:** A dropdown menu set to '1sec'.
- Delay Req Interval:** A dropdown menu set to '1sec'.
- Priority 1:** A numeric input field with red minus and green plus buttons, set to '128'.
- Priority 2:** A numeric input field with red minus and green plus buttons, set to '128'.
- Domain:** A numeric input field with red minus and green plus buttons, set to '0'.
- Slave Only:** An unchecked checkbox.

At the bottom of the settings section is a grey 'SUBMIT' button.

Fig 1-5: PTP Profiles Page

**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 Req 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 Req 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
<b>Default Profile:</b>	
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

**AES 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
Slave Only:	Disabled

DSCP Name	DS Field Value	IP Precedence (Description)
CS0	0	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 1-1: DSCP Names & Their Corresponding IP Precedence

## AoIP Streams Web Page

The AoIP Streams page shows the current configuration of the AoIP transmit and receive options. This allows the user to select the various output streams available on the device and edit their settings. This shows a typical page, but the options may vary according to the actual device in question

**Transmit Source Select** – There may be any number of streams that a device can provide. 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 the AVN-TB product range. 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”. Often the prefixes are preset and not editable, in which case the stream name will be displayed as a fixed string.

**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 on page 12 – ‘DSCP Names & Their Corresponding IP Precedence’ 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. No two AoIP stream sources can share the same multicast IP address.

Fig 1-6: The AoIP Streams Web Page

AES67 requires audio multicast streams to be within the ‘administratively scoped range’, which is restricted to 239.0.0.0 to 239.255.255.255.

Our recommendation to avoid address conflict is to setup the unit to use addresses derived from the current IP address set for that port and a session count. So if the AoIP port has an address of 192.168.123.45 then the multicast addresses should be of the form 239.s.123.45 where s is the session count. The session count ‘s’ is simply a number that is incremented for every AoIP stream that the user creates.

This means the first stream should have a multicast address of 239.0.123.45, the next stream should be 239.1.123.45 and so on. Where the unit creates non-audio streams (eg metering info) then the full range of multicast addresses are available (224.0.0.0 to 239.255.255.255) and it makes sense to use a first octet outside the AoIP value of 239 so for instance 238.0.123.45.

At all times it is important to recognise that other equipment on the network may not use the same rules and so the user will need to be fully aware of such equipment, and its IP addresses, before choosing any IP address for AVN equipment.

**Auto Multicast** - Where 'Auto Multicast' is available in the 'AoIP Network Port Settings', enabling this will automatically generate the Multicast IP addresses for the transmit AoIP streams. This option is enabled by default.

Since we use the default of 192.168.0.100 and 192.168.1.100 for the IP addresses of the Ethernet and AoIP ports respectively, for units that have a fixed number of sessions the Multicast address is 239.x.1.100 - where x is the session number.

For units that have dynamic sessions, the Multicast address is 239.x.y.z - where x is the session number, y and z are the 3rd and 4th octets of the IP address respectively. For example if the unit's IP address is set to 192.168.16.200 the sessions will be created with the default multicast address of 239.x.16.200

**Note that when this option is enabled, changing of the session addresses is NOT allowed.** This option is only active while the addressing mode of AoIP port of the device is set to 'static'. When in dynamic mode this option is ignored and the session addresses can be changed. If the addressing mode is then changed back to static this will in turn revert the session addresses back to their defaults as per the above rules.

The screenshot shows a configuration interface for 'Audio over IP Network Port Settings'. It features a blue header bar with the title. Below the header, there are four main settings: 'Address Mode' with radio buttons for 'Dynamic' and 'Static' (the latter is selected); 'Static IP Address' with a text input field containing '10.0.20.2'; 'Static Subnet Mask' with a text input field containing '255.0.0.0'; and 'Auto Multicast' with a checked checkbox icon. At the bottom of the form is a grey 'SUBMIT' button.

Fig 1-7: The Auto Multicast Option

**Auto Multicast Summary** - When the 'Auto Multicast' option is enabled and the addressing mode is 'static' the session addresses will always be in the form of 239.x.y.z

where x is the session number, y and z are the 3rd and 4th octets of the AoIP port's IP address respectively.

## References and Standards

### Free download from [datatracker.ietf.org](http://datatracker.ietf.org) – search by RFC number

- IP - Internet Protocol. RFC 791
- UDP - User Datagram Protocol. RFC 768
- TCP - Transmission Control Protocol. RFC 793
- IGMP - Internet Group Management Protocol (Version2). RFC 2236
- DHCP - Dynamic Host Configuration Protocol. RFC 2131
- Auto-IP - Dynamic Configuration of IPv4 Link-Local Addresses. RFC 3927
- RTSP - Real Time Streaming Protocol. RFC 2326
- SDP - Session Description Protocol. RFC 4566
- RTP - A Transport Protocol for Real Time Applications. RFC 3550
- RTCP - Real Time Transport Control Protocol. RFC 3550
- mDNS - Multicast DNS. RFC 6762
- DNS-SD - DNS-Based Service Discovery. RFC 6763
- HTTP/1.1 - Hypertext Transfer Protocol. RFC 2616
- URL - Uniform Resource Locators. RFC 1738
- DiffServ - Definition of the Differentiated Services in IPv4 and IPv6 headers. RFC 2474  
Configuration Guidelines for DiffServ Service Classes. RFC 4594

### Purchase from [standards.ieee.org](http://standards.ieee.org)

- PTPv2 - IEEE Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems. IEEE 1588-2008

### Purchase from [www.aes.org/publications/standards/](http://www.aes.org/publications/standards/)

- AES67-2015 - AES standard for audio applications of networks - High-performance streaming audio-over-IP interoperability

### View at [www.ravenna-network.com/about-ravenna/resources](http://www.ravenna-network.com/about-ravenna/resources)

- RAVENNA Network Principles 1.0
- RAVENNA Network Requirements 0.2
- RAVENNA and AES67 1.0

### Sonifex & Ember+ Apps

- [www.sonifex.co.uk/downloads/servdisc.zip](http://www.sonifex.co.uk/downloads/servdisc.zip)
- [www.sonifex.co.uk/downloads/tinyEmber.zip](http://www.sonifex.co.uk/downloads/tinyEmber.zip)

**SONIFEX**

[www.sonifex.co.uk](http://www.sonifex.co.uk)

t:+44 (0)1933 650 700

f:+44 (0)1933 650 726

[sales@sonifex.co.uk](mailto:sales@sonifex.co.uk)