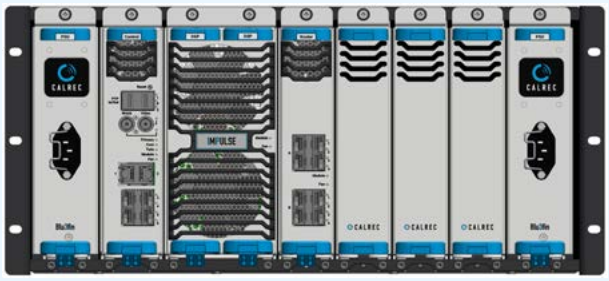
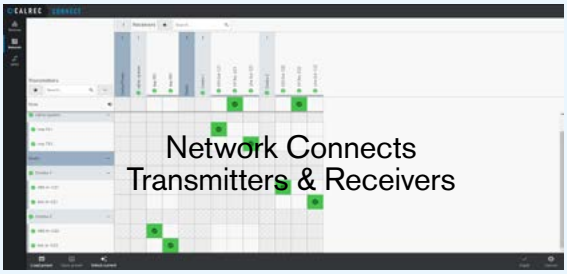
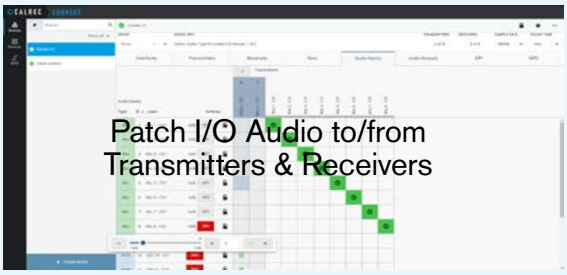
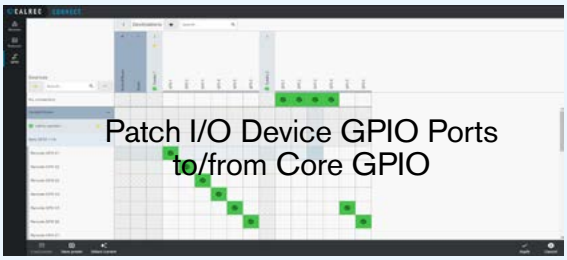
CONNECT GUIDE IMPULSE V2.4 / ARGO V1.0 / TYPE R V2.4



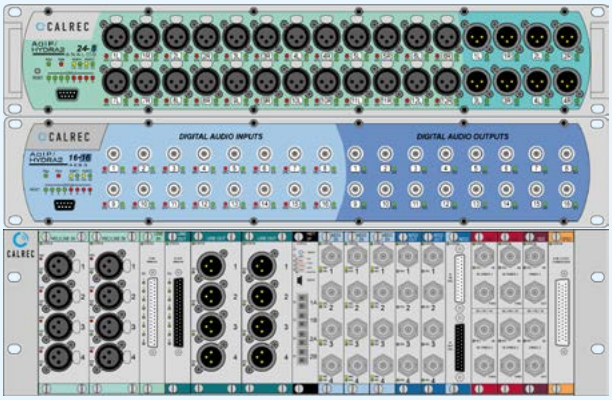
Cores







Connect



Third Party AoIP Interfaces

I/O Devices

AoIP Stream Management System

calrec.com

CALREC

Calrec Audio Ltd
Nutclough Mill
Hebden Bridge
West Yorkshire
England UK
HX7 8EZ

Tel: +44 (0)1422 842159
Fax: +44 (0)1422 845244
Email: enquiries@calrec.com

calrec.com

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CONNECT
CONTENTS

CONNECT
CONFIGURATION

CALREC

OVERVIEW

Connect provides the user with all the facilities needed to view and manage AoIP streams and devices.
These can be connections to and from Calrec AoIP Devices (or other 3rd party AoIP streams from Impulse V1.4/Type R V2.0).

Connect is served by Impulse or Type R processing cores and is accessed via a web browser at a specific IP address, currently this is set to a default of **172.16.255.60** from the Setup port which by default is currently interface **2** on the front of the control module in the Impulse Core, or the setup port on the front of a Type R core, or **172.29.1.21** via the management subnet, or via the media network on **192.168.30.100** , see the **Impulse Configure Guide (926-290).pdf** or **Type R Configure Guide (926-285).pdf** for further information on configuring addresses on the cores.

Note: Interface 2 is classified as a 'Setup port' and whilst it can be used to access the various applications it is not intended to be used as part of a network.

For a quick start up introduction to the Impulse and Type R systems see:-

Impulse Start Up Guide V1.0 (926-291).pdf or **Impulse - Argo Start Up Guide V1.0 (926-321).pdf** or **Type R Start Up Guide (926-282).pdf'**

Connect is optimised to be accessed using a Google Chrome web browser, on a Windows 10 or higher based computer platform.

Windows PC: Minimum specification

- 7th Generation Intel® Core™ i5 7400 CPU Base Frequency 3.00GHz Processor or AMD equivalent
- 8 GB RAM memory
- Windows 10 - 64bit
- Google Chrome Browser - Version 77 or higher
- Ethernet adaptor for remote connectivity/media network interfaces
- Minimum Display Resolution 1366px X 768px

* Impulse and Type R web applications are designed for use with a touch interface where possible.
Calrec recommends the use of a 1920x1080 touch screen monitor.

About this Guide
This guide describes in detail the operation of Connect and the example workflow below provides an overview of how the functions in Connect are used to configure and manage AoIP streams and devices.

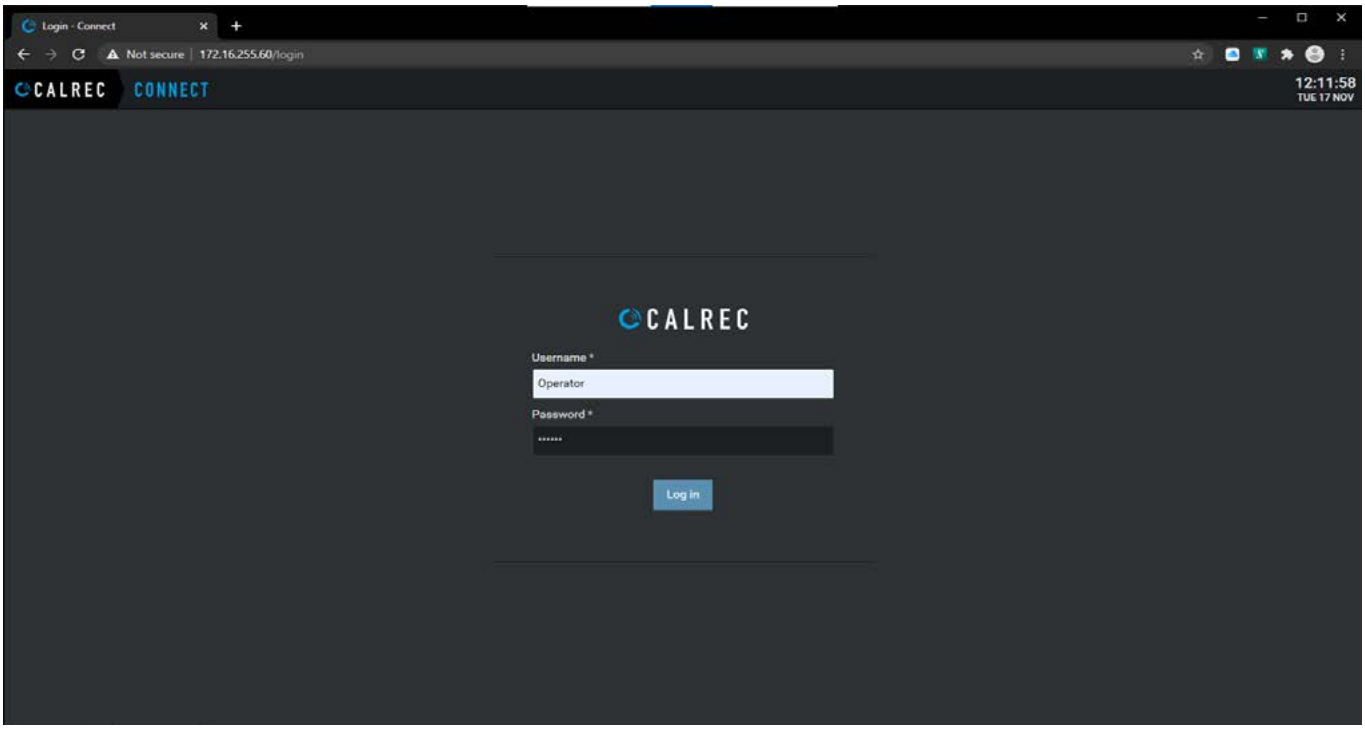
Connect Example Workflow - How to connect audio via AoIP between an AoIP Device and a Core.
From the Devices Menu in Connect:-
Select an AoIP device from the devices list.
Select/Add & configure a transmitter and a receiver for the AoIP Device.
Select the Core from the devices list (shown here as '**calrec-system**').
Select/Add & configure a receiver and a transmitter for the core.
Connect the Audio Inputs from the AoIP devices to its transmitter channels.
From the Network Menu in Connect:-
Connect the AoIP device transmitter to the core receiver.
From the I/O Patching Menu in Console PC for Impulse or in Assist :-
Patch the core receiver channels to the Desk Inputs of the Core.
Using the surface panels:-
Process & Route the Desk Inputs to Desk Outputs as required.
From the I/O Patching Menu in Console PC for Impulse or in Assist:-
Patch the Desk Outputs to the Core transmitter channels.
From the Network Menu in Connect:-
Connect the Core transmitter to the AoIP device receiver.
From the Devices Menu in Connect:-
Connect the AoIP device receiver channels to the Audio Outputs in the AoIP Device.

Note: items shown in BLUE in this guide are from the Console PC and/or Surface or Assist
See the Apollo+ and Artemis+ Operator and Assist Manuals or the Argo Operator and Assist Manuals or the Type R Assist and Console Panel Operation manuals for further information.

STARTING CONNECT

Starting and Logging in to Connect
Open your Web browser e.g. Chrome and browse to the address of the LAN port you are connected to e.g the Setup port on the core of the unit which by default is **172.16.255.60**. This will launch Connect, the system opens on the Connect Login page shown below.

CONNECT LOGIN PAGE



Connect Login Instructions
From version 1.4 of the Impulse applications and version 2.0 of the Type R applications, security passwords have been put in place.

There are currently two fixed Roles/Accounts available, the normal Operator account and an Engineer account which provides technician access to extra configuration functionality. For the Operator account, the Username should be entered as '**Operator**' with the Password '**calrec**'.

Note: both the Username and Password are case sensitive.

In order to access Connect which is for technician access only the user will need to enter the Engineer Username and Password, the user taps on the 'Log' in button and Connect accesses the Home page which is typically the Devices menu page as shown overleaf.

DEVICES

Devices Configuration
On launching Connect the user is taken to the Connect Devices page in the Device Configuration tab as shown above right. If already on another menu item just select Devices from the menu.

The first time that Connect is launched it automatically scans the network for AoIP Devices. Once they have been discovered they are entered into the devices list on the left side of the Devices page.

If no device is selected in the devices list, the Device Configuration sub-view area is blank as shown middle right and the user is prompted to select a device to view.

Note: if Connect has been run before, it will have stored the devices list from its previous session and will display that list.

Devices List
The devices in this column can be accessed by clicking on each device or the user can find the device if they know its name by entering it into the search box shown right and below right.

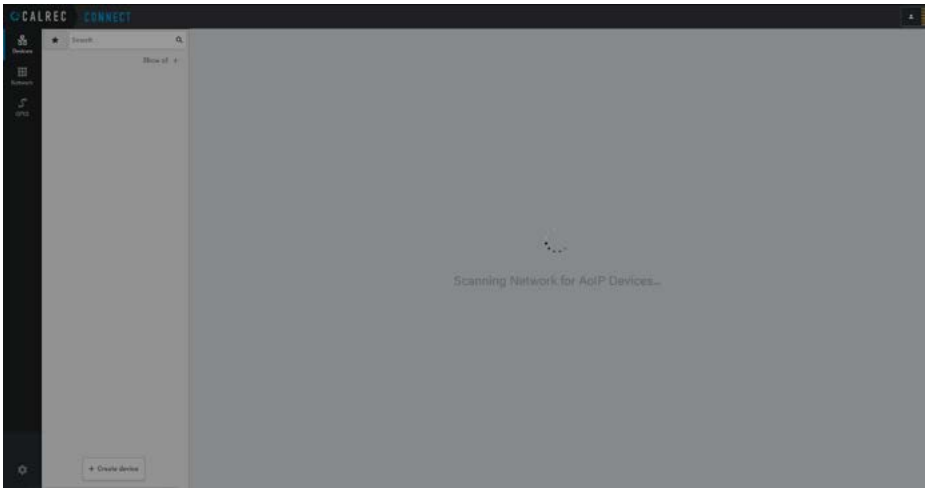
The image below right shows that the Impulse Core has been selected, which is now named as '**calrec-system**'.

Once a device has been selected, the device configuration view is populated with information about the device and its current settings. Across the top of the page is shown a Summary of the device which includes:- Name, Online/ Offline status, Group selector, Device Info, Manufacturer, Software Version, Number of Transmitters/Receivers configured and how many are available, Sample Rate and Packet Time for the selected device.

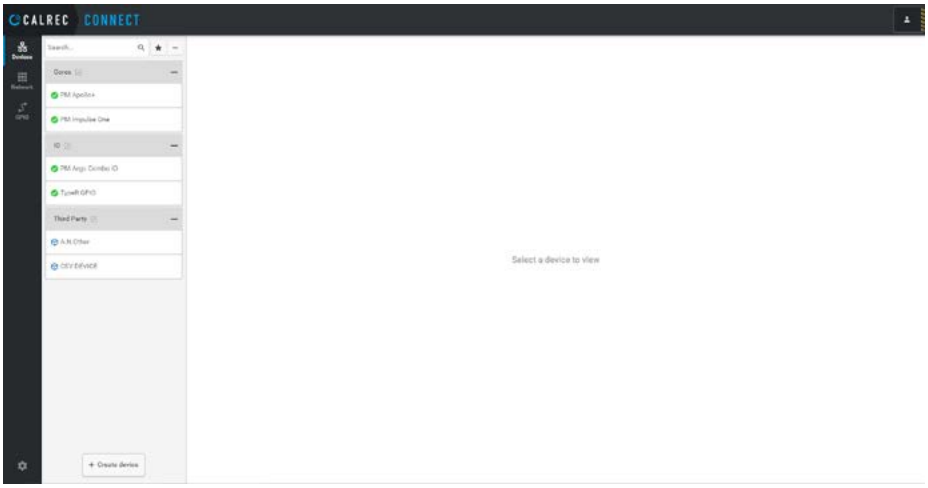
There is also a Lock icon, Favourites selector and an 'Options' button (•••) in the page header. Below this are details of the devices:- Interfaces, Transmitters, Receivers, Sync, Audio Inputs & Outputs, GPI & GPO arranged in Tabs.

Each of the Devices menu features are described in the following pages:-

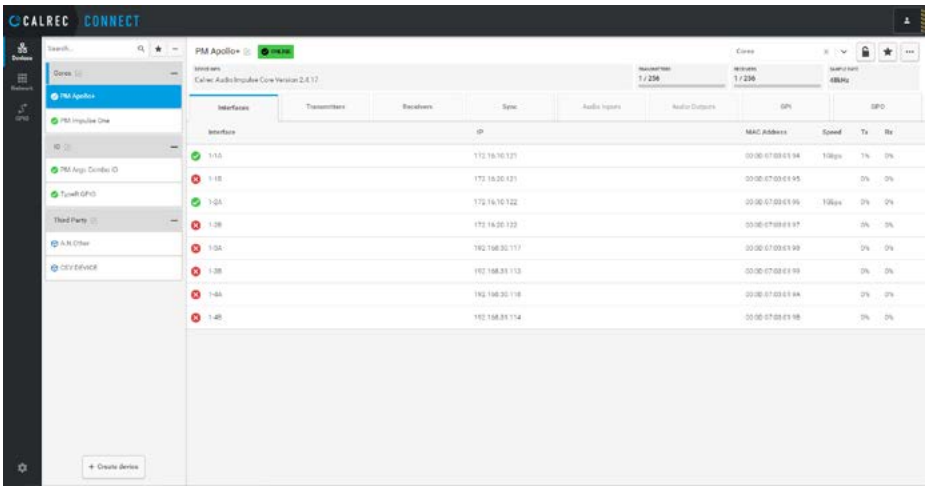
DEVICES-SCANNING NETWORK FOR AOIP DEVICES



DEVICES-NO AOIP DEVICE SELECTED



DEVICES-LANDING PAGE-POST SCAN SHOWING CORE ROUTER INTERFACES



Device Names
Selecting the Device Name from the summary section allows the user to change its label. The image above right is showing the Device Name being edited, the user selects the '✓' symbol to accept or the 'x' symbol to cancel the change.

Group Labels
Selecting the Group Label from the summary section allows the user to add Groups that can be used to organise the various devices. For instance the user may want to list the AoIP devices under an **IO** name and keep the **Consoles** separate. The image right is showing a Group Label being created called '**Cores**', the user enters the Group Label and presses Enter to accept.

If devices have to be in different groups then they require different Group Labels and if they need to be in the same group they would all be placed under the same Group Label. This is shown below right where the Control Room group contains the Core mixer, the Studio group contains Combo 1 and now Combo 2 is going to be placed in a Green Room group.

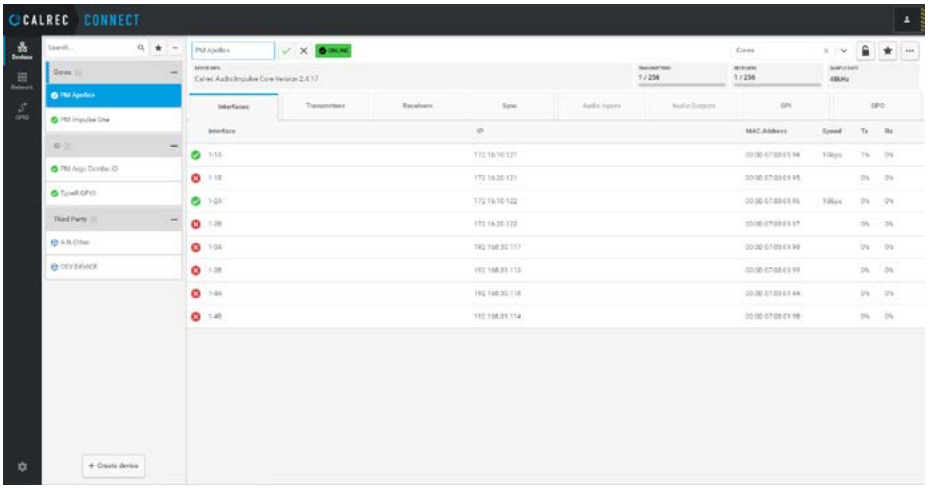
The devices in this column can be hidden or revealed by clicking on each of the group folders such as 'Cores' and 'IO'.

A '+' symbol indicates that there are more items in the collapsed group folder and selecting this will expand the group folder to reveal devices. A '-' symbol when selected will collapse the group folder, hiding the devices. At the top of the Device List to the right of the Search field is a button which allows the user to 'Expand All +' or 'Collapse All -' of the group folders.

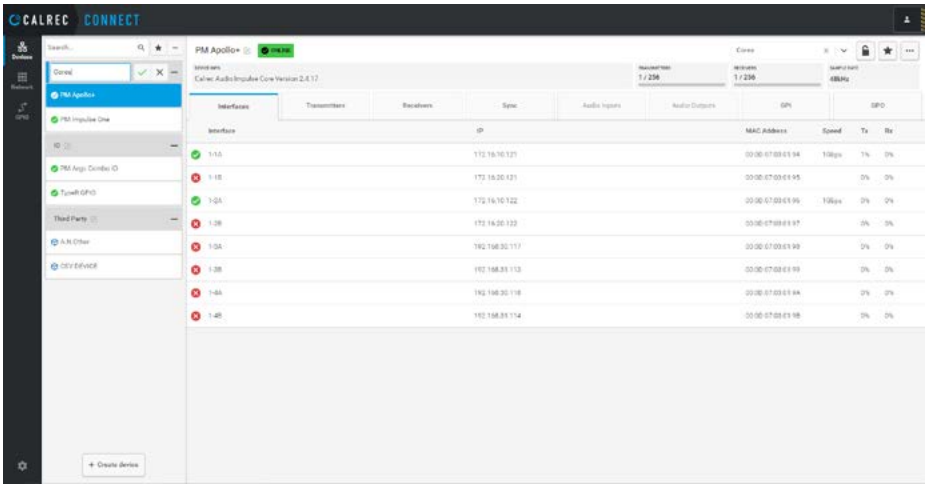
Device Type, Manufacturer label and Software Versions
These are supplied automatically by the device being connected.

Number of Transmitters and Receivers In Use/Available
These give the user a summary of the transmitter and receiver streams in use and the total number of available transmitters and receivers for each device.

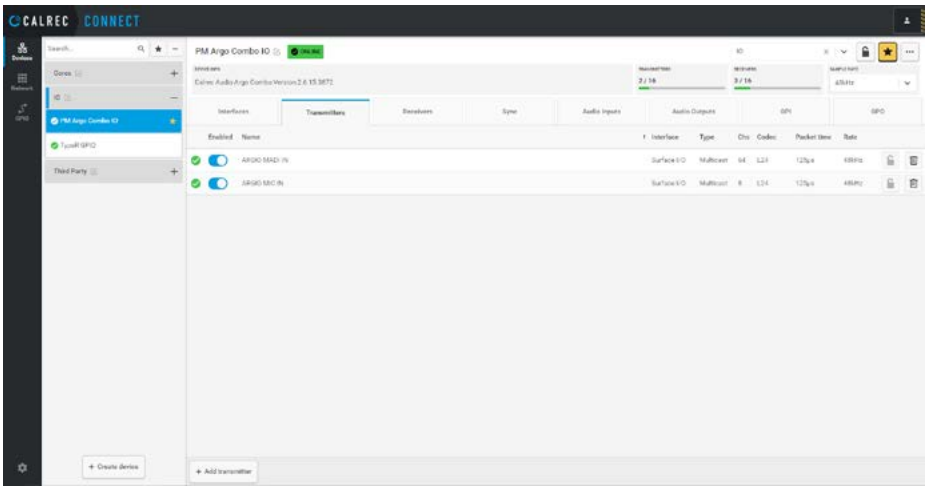
DEVICES-EDIT DEVICE NAME



DEVICES-EDIT GROUP LABEL



DEVICES-LIST EXPAND/COMPACT



Add to/Remove from Favourites
The 'Favourites' option works in conjunction with the devices list to apply a favourites filter allowing the user to only view the devices selected as favourites.

The user clicks on a device, and clicks on the Yellow Star on the far right of the page header, this adds an icon to the device showing that it is a favourite see '**Combo 1**' above right.

Once a selection of favourites have been made the user can click on the view favourites star at the top left of the devices list next to the search box to apply the favourites filter. This button allows the user to toggle between viewing all devices and viewing favourite devices.

To remove a favourite, click on the device marked as a favourite and click on the Yellow star on the far right of the page header this will remove the device from the favourites selection.

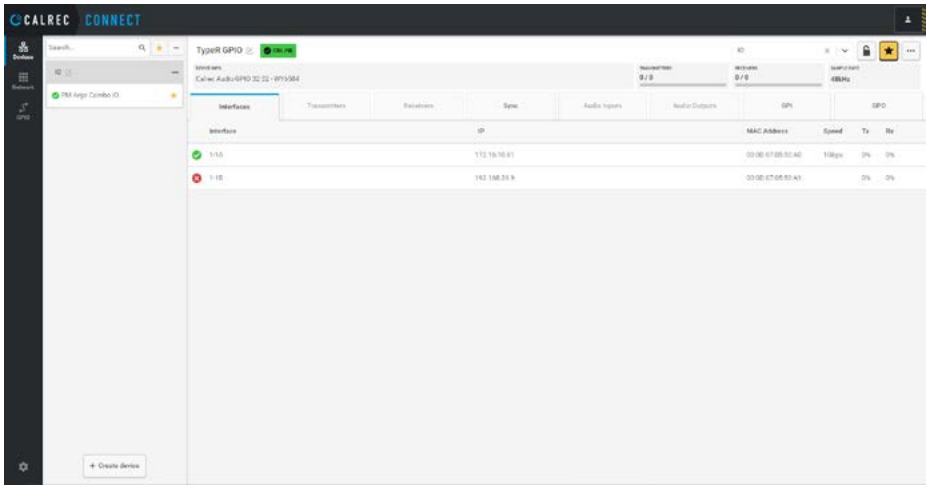
Search Devices
In a large devices network there can be many device entries in the devices list. At the very top of the devices list is a 'search' field which allows the user to type in the name of the device. As more characters are entered into the search field the devices list is progressively filtered until the required device appears.

For example typing 'Com' into the search field (shown right), only shows Combo 1 & Combo 2 in the Studio group.

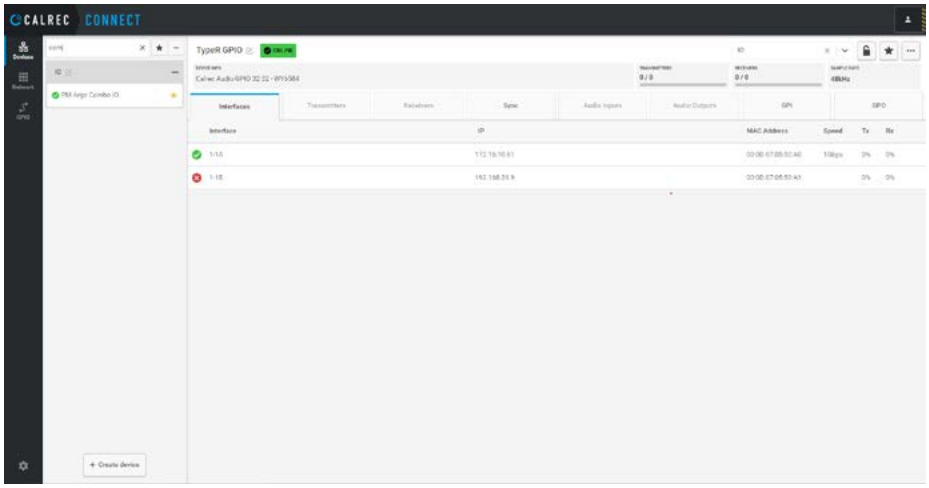
Change Sample Rate
The user clicks on a device, opens and selects the 'Sample Rate' option for the I/O Devices noting that the current sample rate for that device is shown in the dropdown.

A selection of available sample rates appear with the current sample rate highlighted, the user may now select a different sample rate as shown below right.

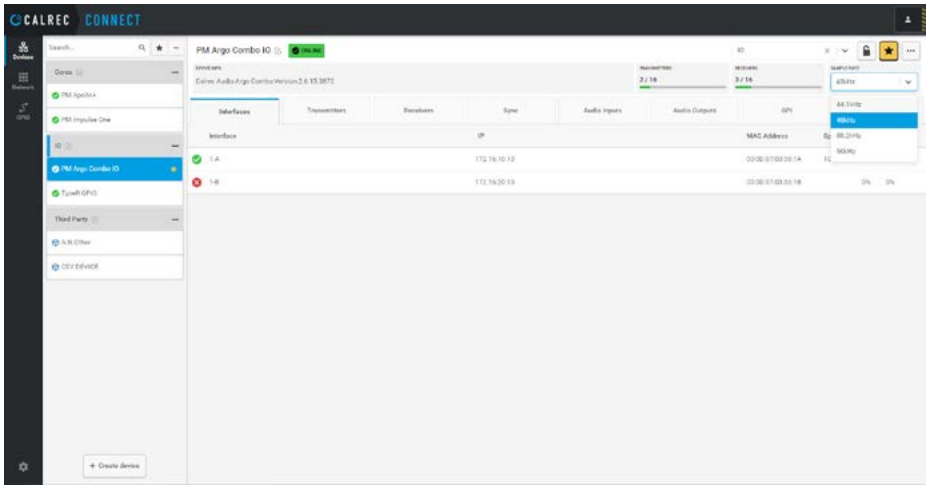
DEVICES-FAVOURITE SELECTION



DEVICES-SEARCH



DEVICES-CHANGE SAMPLE RATE



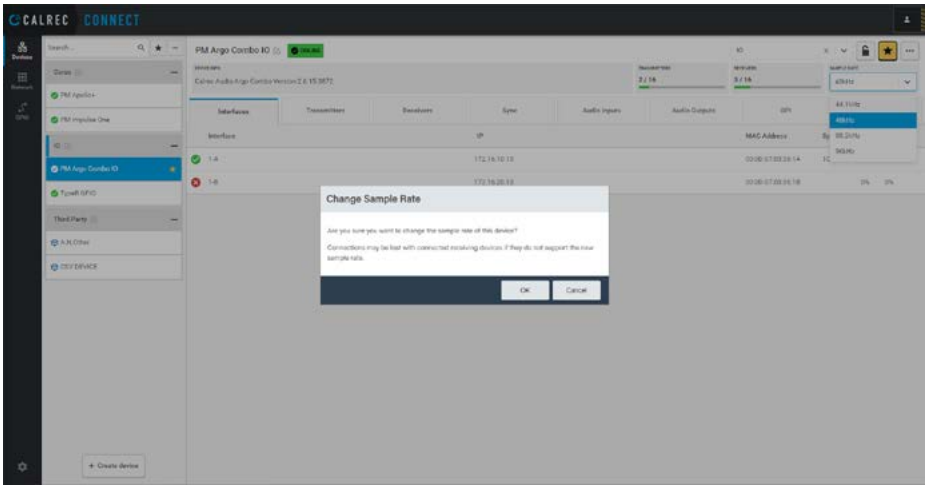
Change Sample Rate Confirmation
Once the user has selected the new sample rate to change to for the device, the 'Change Sample Rate' confirmation message appears as shown above right, warning that connections may be lost with receiving devices if they are not set to operate at the new sample rate. On accepting this new sample rate by clicking on 'OK', any existing streams at the old sample rate will be suspended. Alternatively the user can 'Cancel' the request to change the sample rate and the existing streams will be unaffected. The user will need to ensure that both the transmitting and receiving devices are operating at the same sample rate in order to successfully pass streaming audio between them.

Change Packet Time (Covaloz AoIP Interface Cards US6493/US6525)
The user clicks on a device opens and selects the 'Packet Time' option noting that the current packet time for that device is shown in the dropdown. A selection of available packet times appear with the current packet time highlighted as shown right, the user may now select a different packet time as required. The preferred packet time for live broadcast is 125us.

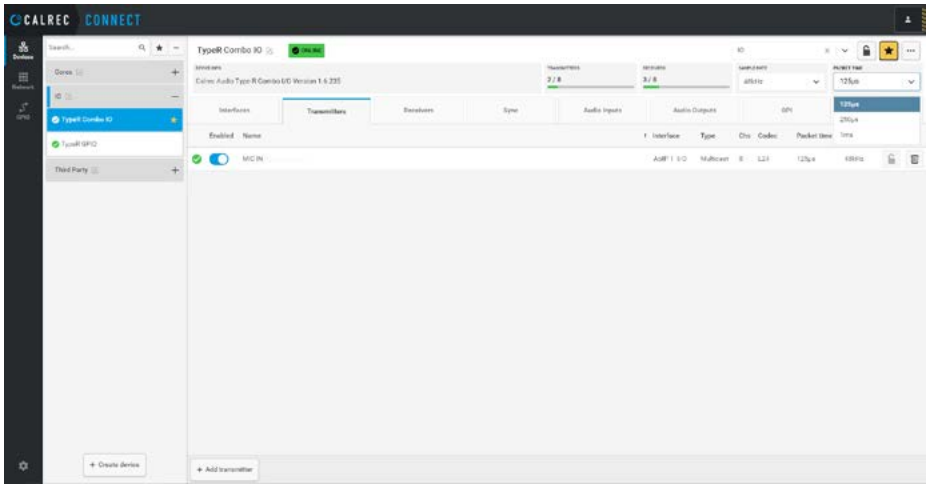
Note: all the streams on the Covaloz AoIP interface card will be set to the same packet time.

Change Packet Time Confirmation
Once the user has selected the new packet time to change to for the device the 'Change Packet Time' confirmation message appears as shown below right, warning that connections may be lost with receiving devices if they are not set to operate at the new packet time. On accepting this new packet time by clicking on 'OK', any existing streams using the old packet time setting will be suspended. Alternatively the user can 'Cancel' the request to change the packet time and the existing streams will be unaffected. The user will need to ensure that both the transmitting and receiving devices are operating using the same packet time in order to successfully pass streaming audio between them.

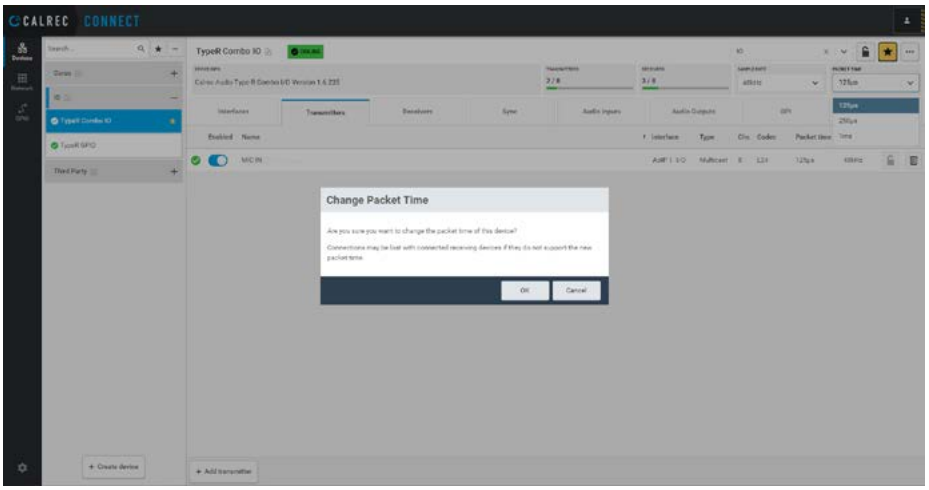
DEVICES-CHANGE SAMPLE RATE CONFIRMATION



DEVICES-CHANGE PACKET TIMES PER DEVICE (COVALOZ CARDS)



DEVICES-PACKET TIMES CONFIRMATION (COVALOZ CARDS)

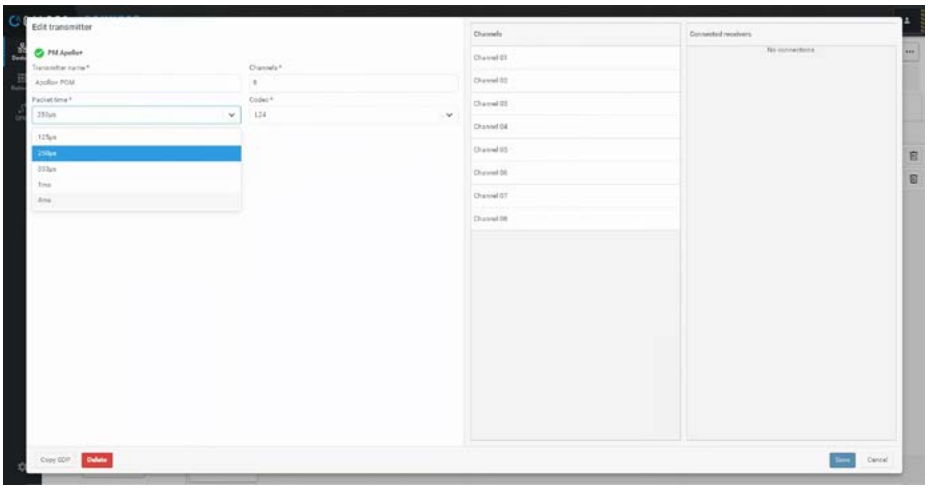


Change Packet Time (Medium Capacity AoIP Interface Cards US6493-2 & US6525-2)

The medium capacity interface cards allow the user to set different packet times for streams in the same AoIP device rather than limiting all the streams to operate at the same packet time in the original interface cards.

The user clicks on a device opens and selects the required stream. Select the 'Packet Time' option noting that the current packet time for that stream is shown in the dropdown. A selection of available packet times appear with the current packet time highlighted, the user may now select a different packet time as required. The preferred packet time for live broadcast is 125us, however the packet time for this stream has been changed to 250us as shown in the image above right. This allows each of the streams in the device to be set to different packet times as required.

DEVICES-CHANGE PACKET TIMES PER STREAM (MEDIUM CAPACITY CARDS)



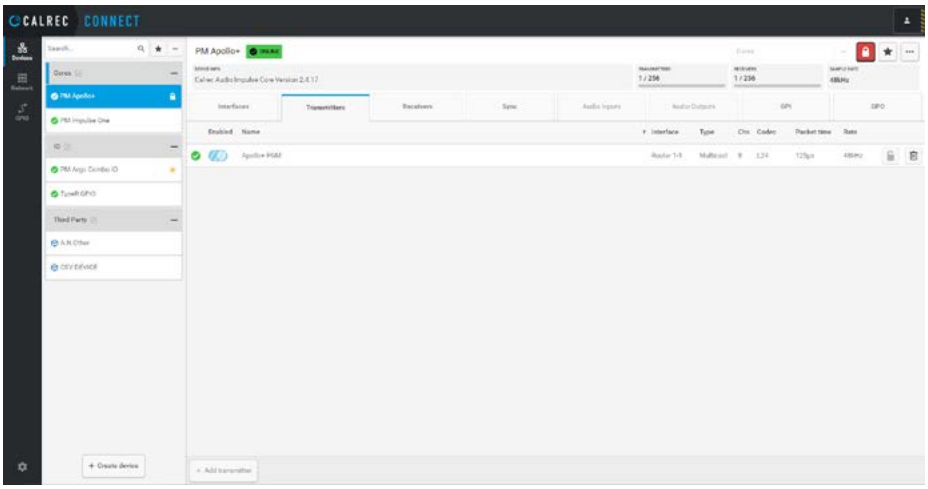
Lock Device

The user clicks on a device and selects the Padlock on the right side of the page header. A padlock icon is added to the device and the summary entry indicating it is locked.

If a device is locked, users cannot:-

- Change device name or group label.
- Edit AoIP interface configuration.
- Edit the configuration of existing transmitters.
- Edit the configuration of existing receivers.
- Add new transmitters or receivers.
- Delete transmitters or receivers.
- Change sync settings (from Connect).
- Change the routing within the device to/ from streams - specifically, I/O routing that can be performed by Connect. Locking the device in Connect should not prevent changes to a console's router from console UI).
- Relabel I/O box ports.
- Change the sample rate of the device (from Connect).
- Audio ports settings (but only from Connect, they can still be adjusted by mixing console operators).

DEVICES-LOCK DEVICE



If a device is locked, users can:-

- View all configuration settings, including transmitter/receiver detail & SDP copy.
- Add or remove the device from users' favourites lists.
- Remove the device from their Device listing.
- Manually add the device to their Device listing.

Unlocking Devices

As can be seen from the image above right, if the user selects a device that is already locked and the user selects the padlock, then the padlock icon is removed from the device indicating that it is unlocked and full functionality is restored.

DEVICE OPTIONS

Connect Options Menu
Clicking on the 3 dots icon in the top right corner of the page header opens the options drop down shown above right.

This allows the user to perform various functions, these include:-
Get Logs from Connect,
Set the Analogue level for 0dBFS,
Set the Mic Input Headroom,
Enable/Disable System status messages from sources on the selected device,
Update AoIP devices software,
Download and Restore system backups
Forget Device if no longer required
Advanced Options for using NMOS 4, NMOS 5 and Ember+.

Options-Analogue Settings
There are 2 Audio related functions:-
On selecting **Analogue level at 0dBFS** a drop down appears as shown at the top right of the page on the right. Select the operating level required. The Calrec default for this is 0dBFS=18dBu.

On selecting **Mic input headroom** a drop down appears as shown at the right of the page on the right. Select the operating mic input headroom required. The Calrec default for this is 28dB.

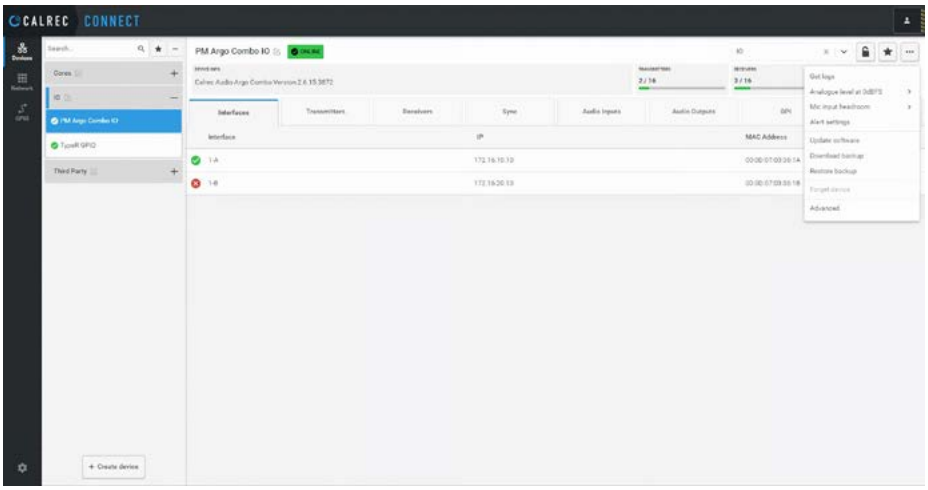
Options-Get Logs
This is an engineering option which allows the user to collect Logs from the Connect Container, see below right. Clicking on **Get Logs** from the Options dropdown starts a collection process with a progress bar showing.

This takes a few minutes and as with the other options should not be used On-Air. Once the Download is completed the 'Logs downloaded' message will appear as shown and the file can be saved as shown top left.

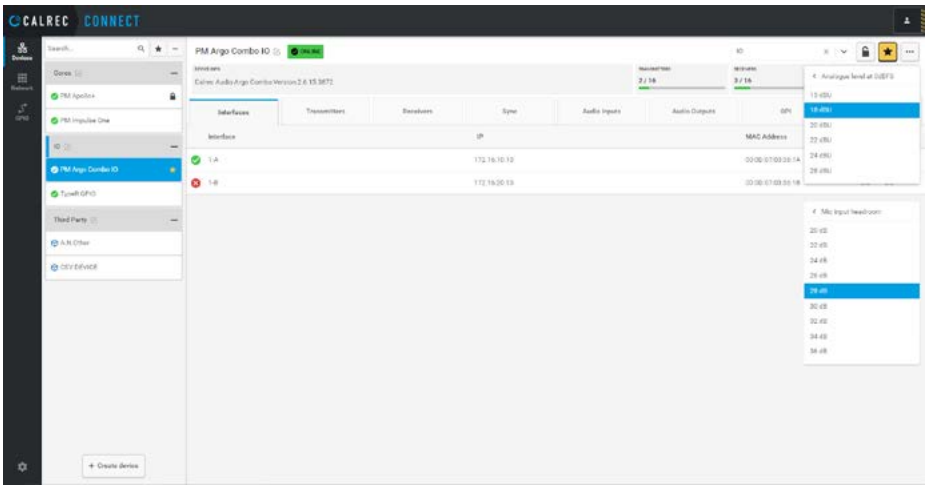
Note: once the ConnectLogs.zip file has been created and saved (see bottom left of footer) the file can then be accessed and sent off to Calrec for analysis.

File references in the footer can be cleared of this message by clicking on the X at the bottom right of the footer.

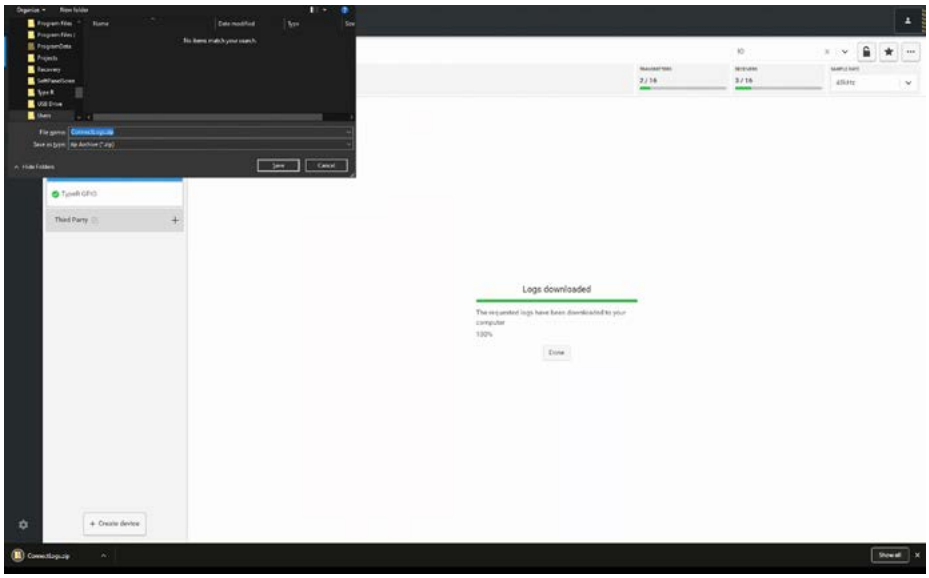
DEVICES-OPTIONS MENU



DEVICES-OPTIONS ANALOGUE LEVEL & MIC INPUT HEADROOM



DEVICES-OPTIONS-GET LOGS



Options-Alert settings
This option allows the user to enable or disable System status messages from the selected device for the sources enabled in the list by toggling the appropriate slider switches as shown above right.

Options-Update software
This option allows the user to update the software associated with AoIP devices in the system.

Note: this software update is independent of the Core software update that is managed in Configure.

On clicking on **Update software** from the Options dropdown the dialogue box opens as shown above right.

This shows which AoIP devices are connected with their type and the software version last loaded.

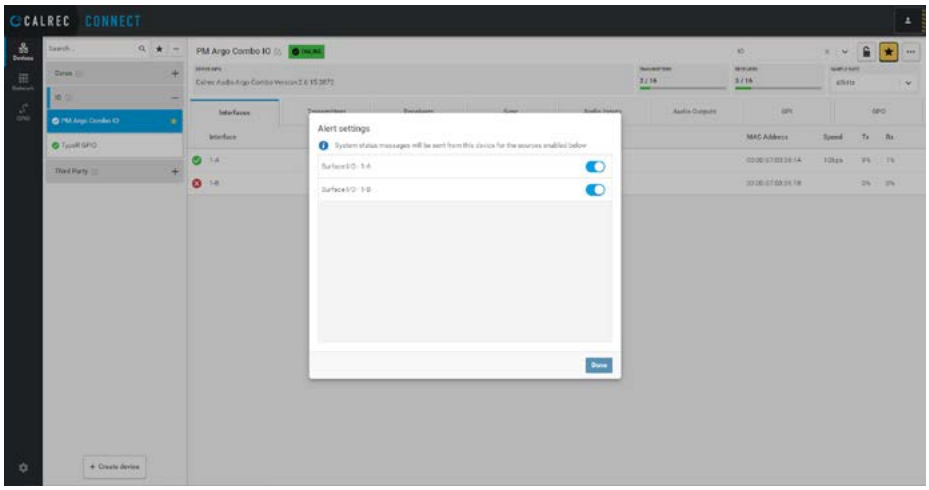
The user can check if there is a later version available on the Calrec FTP site where the file name takes the form '**aoip-2.6.12.3855.calrec**', where 2.6 is the version and 12.3855 is the build number.

Once downloaded from the FTP server, the user clicks on the button labelled '**Upload new software**' button, which then asks the user to browse and find the file to upload. Once selected the file is uploaded ready for use as shown right.

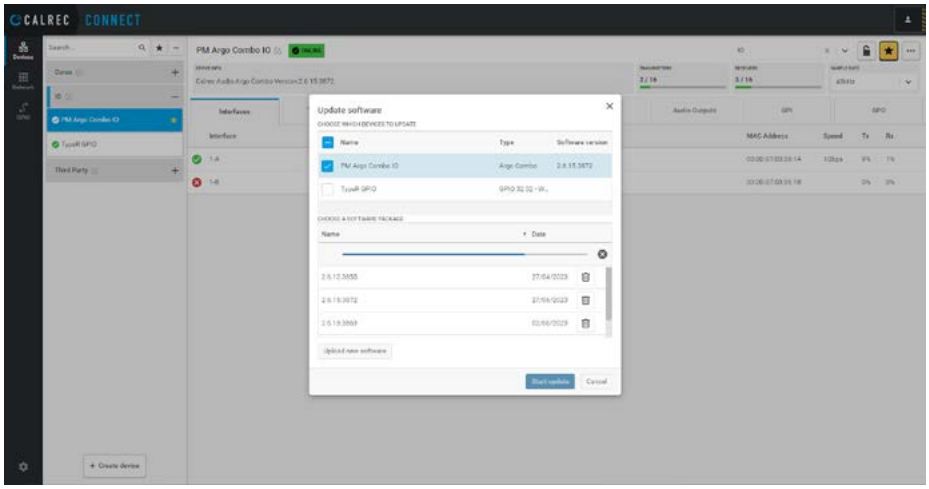
To update AoIP devices, the user chooses which devices to update by clicking on their tick boxes as shown for Combo 2 below right. Then clicks '**Start Update'**.

This process takes a few minutes and on completion the new software version number is shown in the dialogue box against each device as shown below right.

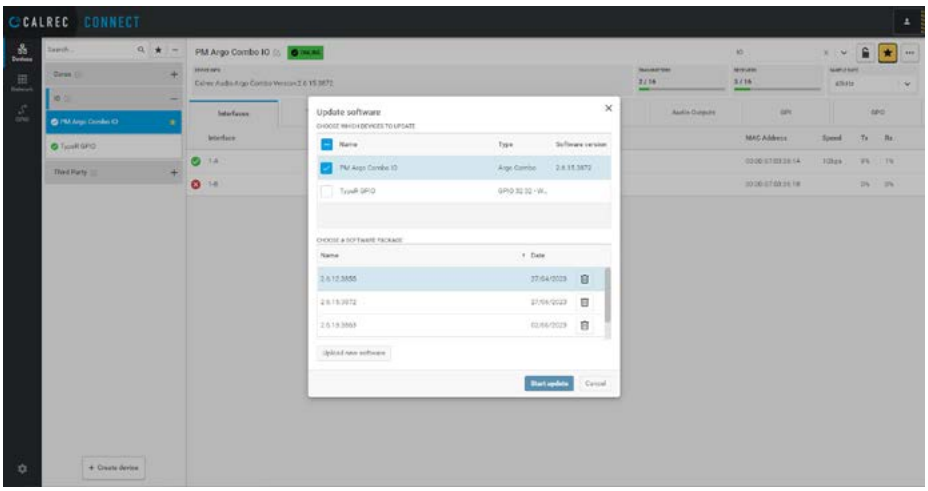
OPTIONS ALERT SETTINGS



OPTIONS UPLOAD NEW SOFTWARE



OPTIONS-START SOFTWARE UPDATE



Options-Backup Devices
The ability to backup all the parameters of the connected devices is provided by clicking on the **Download Backup** option, this starts a collection process with a progress bar showing.

Once the Backup process has completed the 'Backup downloaded' message will appear as shown above right.

Also note that an option in Chrome has been set which asks for a destination to Save the backup to.

This is also used in the Restore process after the 'Choose file' button is pressed in the Import backup dialogue box to select a backup to open.

Note: a ConnectDevice.zip file has been created (see bottom left of footer) which can then be accessed for later recall when required.

The footer can be cleared of this message by clicking on the X at the bottom right of the footer

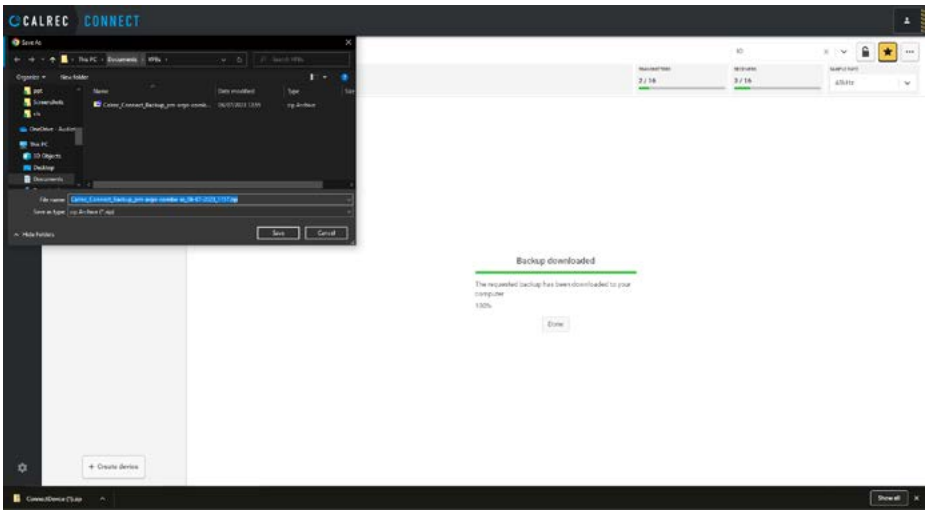
Options-Restore Devices
The ability to restore all the parameters of the connected devices is provided by clicking on the **Restore backup** option.

This opens the dialogue box shown right advising that restoring a backup will overwrite the current configuration.

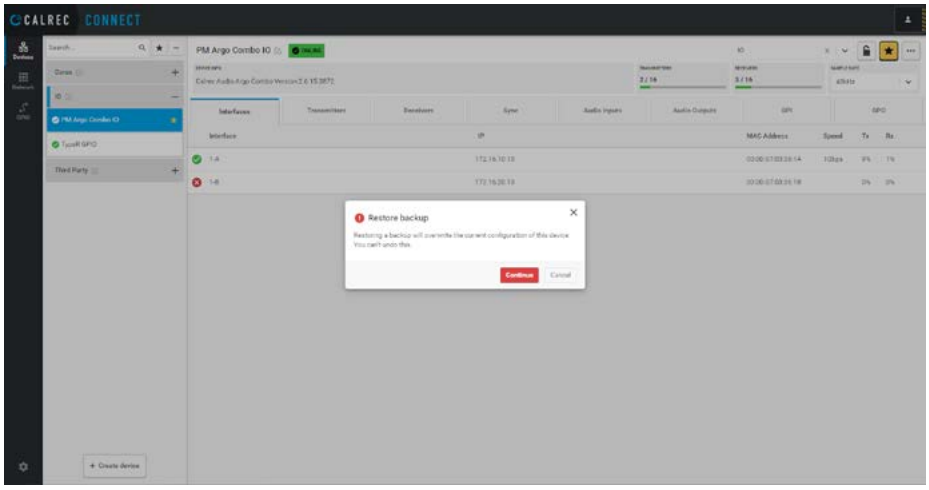
Clicking on Continue opens the Import Dialogue box as shown below right. This asks the user to browse and find the backup file to restore.

Once the file is chosen it is uploaded replacing the current configuration of the devices.

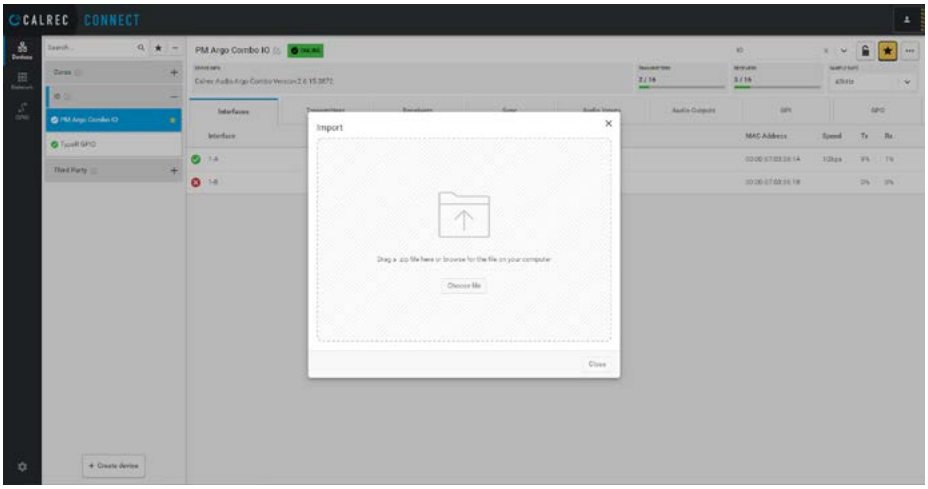
OPTIONS-DOWNLOAD BACKUP FOR CONNECT DEVICES



OPTIONS-RESTORE BACKUP FOR CONNECT DEVICES



OPTIONS-RESTORE CHOOSE FILE



Options-Forget Devices
This option becomes available when the user selects a device that is offline.

If the device is no longer required or no longer available, it may be removed from the device list by clicking on '**Forget Device'**.

A confirmation box will appear stating that "**This will remove the selected device from Connect.**" and that "Reconnecting the device will add it again."

Options-Delete Third Party Devices
This option is available when the user selects a third Party device that has been created. If these devices are no longer required or no longer required or no longer available, they may be removed from the device list by clicking on '**Delete Device'**.

A confirmation box will appear stating that **"This device and its configuration will be removed. You can't undo this."**

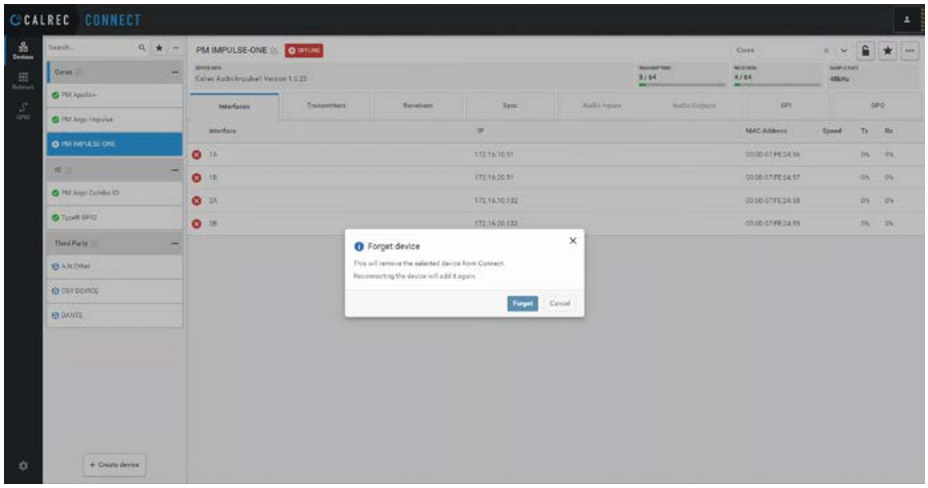
Options-Advanced-NMOS
The first advanced option '**NMOS**' allows the user to configure a HTTP port, Node port and Connection port to allow Media Nodes (devices on the IP network) to be discovered, registered and advertised based on the NMOS IS-04 specification.

It can be enabled using the associated slider switch as shown above right.

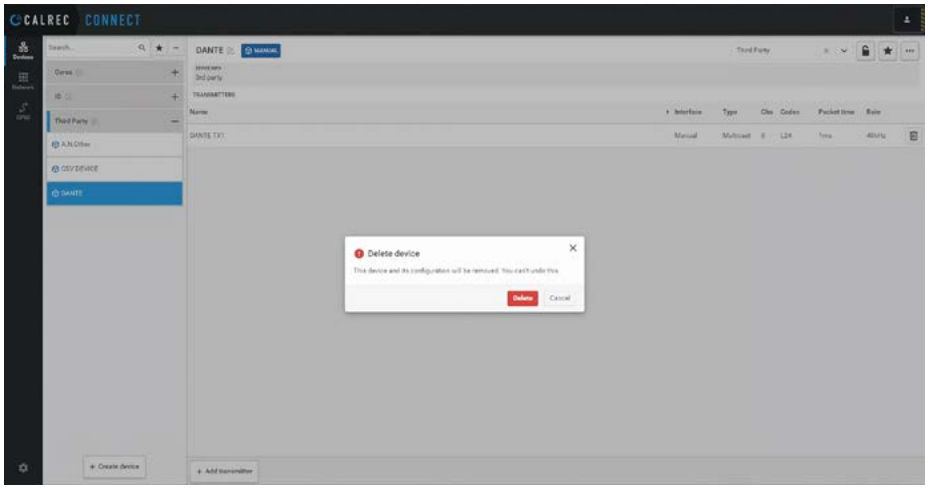
In addition there is a separate slider switch which enables the use of IS-05 based connection management to allow Media Nodes (devices on the IP network) to be configured to create connections between send and receive IP streams automatically passing SDP parameters between devices where appropriate.

This eliminates the requirement for SDP data to be manually copied from either endpoint to form a connection between devices.

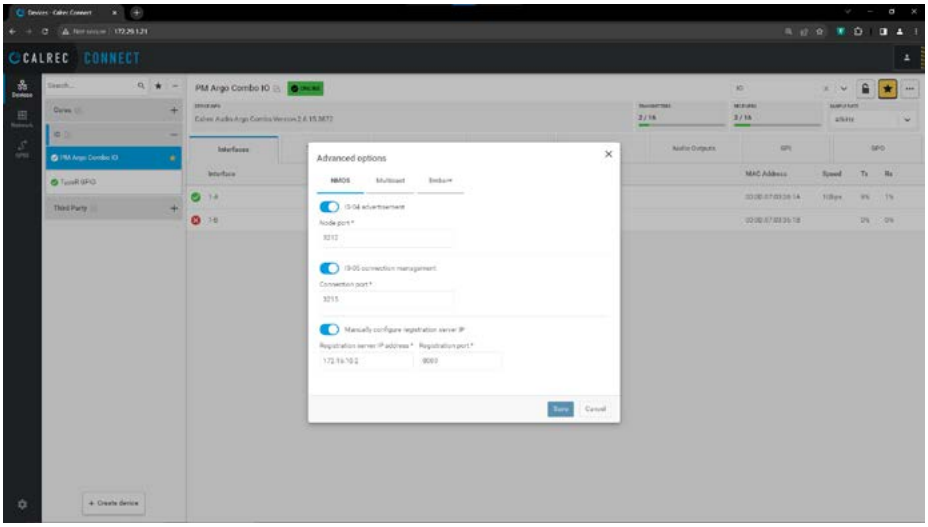
OPTIONS-FORGET DEVICE



OPTIONS-DELETE THIRD PARTY DEVICES



OPTIONS-ADVANCED NMOS IS-04 , IS-05 & MANUAL CONFIGURATION



The bottom half of the dialogue box allows the user to manually configure a registration server by entering its Server's Static IP address and Registration port. This has its own associated enable switch to the right as shown in the image at the bottom of the previous page..

Clicking on '**Save**', stores these NMOS settings, ready to access the available Media Nodes (devices on the IP network).

Options-Advanced-Multicast
The second advanced option '**Multicast**' is only displayed if the selected device is advertising it has the options available. The '**Use Source Specific Multicast by default**' setting is stored by the device, and is off by factory default.

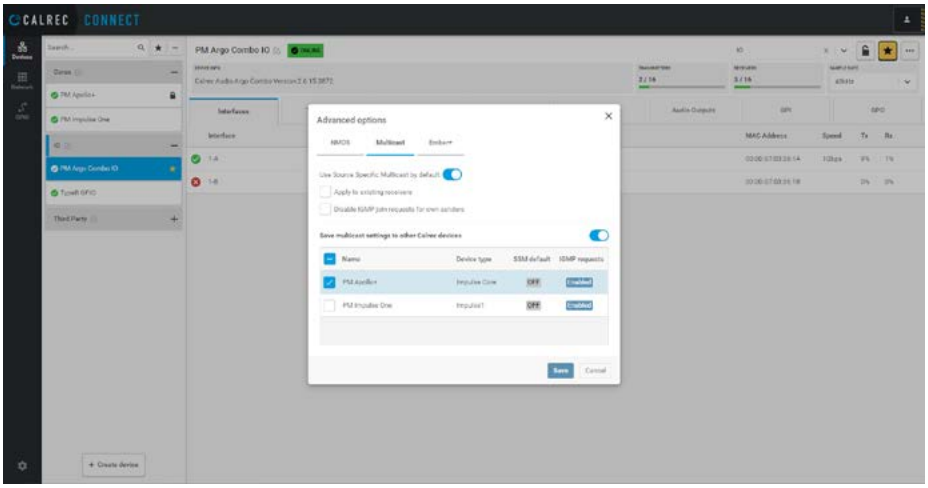
When this setting is off, the new "Source Specific Multicast" option within the receiver config menu when creating new receivers will be off, but users can choose to enable it when creating receivers.

When the setting is on, new receiver config will default to having SSM enabled, but again, users can override that. This advanced setting determines what the new receivers default to (SSM on or off), but users can change that on an individual receiver basis when creating each receiver. Users can also change individual receivers SSM state after they have been created using the stream config menu.

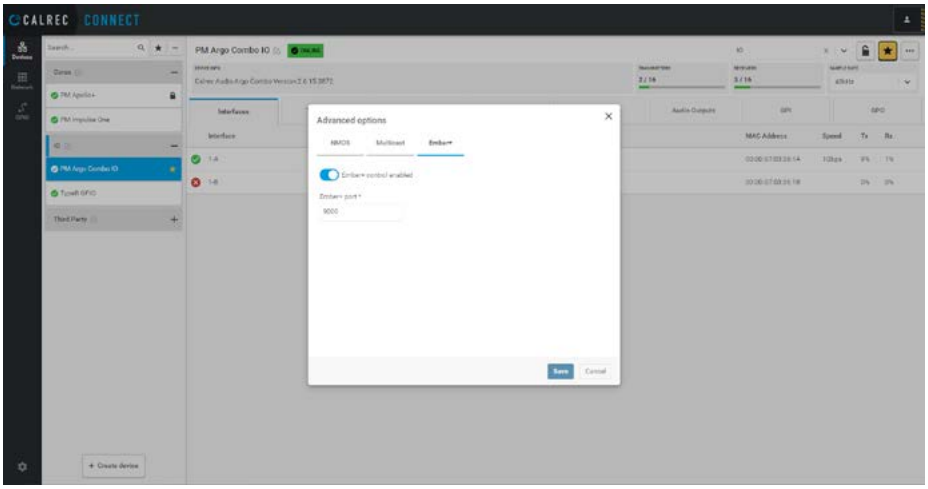
If '**Use Source Specific Multicast by default**' is enabled, and the user selects '**Apply to existing receivers**' and clicks save, then all of the existing receivers on that device should switch to having SSM enabled (as well as it making SSM enabled by default for future new receiver creation).

If '**Use Source Specific Multicast by default**' is set to off and '**Apply to existing receivers**' is selected, then all of the existing receivers already on that device should have SSM turned off (as well as making it default to being off on new receiver creation).

OPTIONS-ADVANCED MULTICAST



OPTIONS-ADVANCED EMBER+



The '**Disable IGMP join requests for own senders**' state is also stored by the device and should be defaulted to not selected (Devices should by default send join requests for their own senders to be AES67 compliant).

Changing the state of this option affects all current and new senders on the device (the device will de-register for all of its existing senders).

If the user selects '**Save Multicast settings to other Calrec devices**' from the device listing and clicks save, it should set all of the selected devices to be the same as the current one - applying the same SSM-by-default on/off, applying the SSM on/off to existing senders, as well as the enable/disable of sender joins, regardless of which, if any of the options on the current device have been toggled selecting other devices will make them behave the same as the current one.

Options-Advanced-Ember+
The third advanced option '**Ember+**' allows the user to configure a Connection port and enable an Ember+ server access to each I/O box which needs to support its own direct interface via 3rd party control based on the Ember+ protocol.

SETUP CORE ROUTERS & AOIP INTERFACES IN CONFIGURE / CONNECT

Although this is partially covered in the Configure application, it is easier to include the relevant section instead of cross referencing. In order to connect the Cores to the AoIP devices, it is first required to setup the AoIP Core Router Interfaces in Configure, and the AoIP device Interfaces in Connect, so that they can be networked together to allow Audio streams to be passed between the Routers in the Core and the various AoIP devices which provide Audio & GPIO resources. The first step is to setup the Core Router IP addresses in the core using the Configure application. To do this, access the Configure application by opening Google Chrome, on your PC type **172.16.255.19** into the address bar and press enter to navigate to the '**Configure**' landing page, then go to the Configure>Networks> AoIP Interfaces page as shown below.

AoIP Core Router Interfaces
This page can either be accessed from the Configure>Networks Menu and appears as shown above right.

Each Impulse core can have up to 4 router modules & each router module has 8 x 1 Gbps Ethernet ports which are arranged as 4 pairs to provide seamless packet switching redundancy. Alternatively a core router can operate at 10Gbps in which case only 1 pair of ports are available per router.

Each Type R core has 1 router module with 4 x 1 Gbps Ethernet ports which are arranged as 2 pairs to provide seamless packet switching redundancy.

Each port for either core is capable of passing 512 channels of audio in each direction, however there are communications overheads including mic gain control and GPIO switching functions amongst others, so Calrec suggests a practical limit of 256 bi-directional channels per router connection.

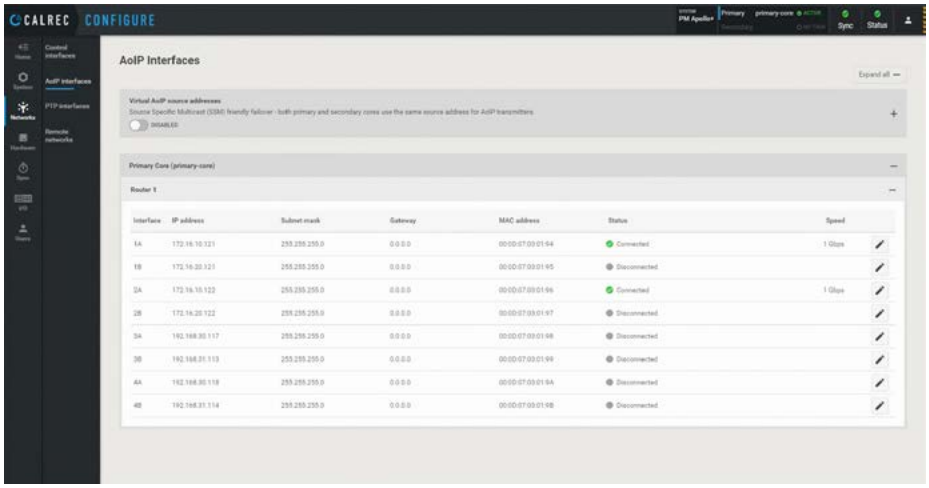
The page shown below right provides controls to view and configure the following parameters for each of the router interface ports:-

IP address
Subnet mask
Gateway (optional)

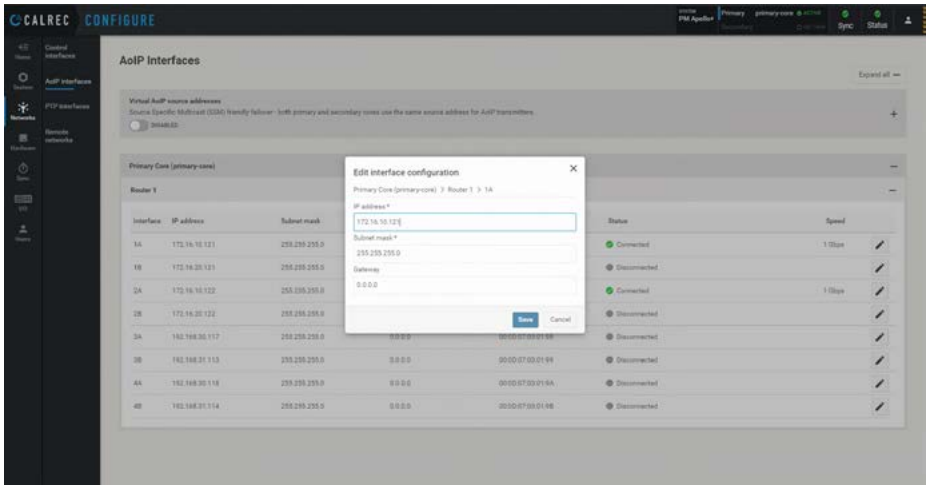
Note: the IP address and Subnet mask must be set with valid values. The Gateway is optional.

The current connectivity status of each port is displayed as either Connected or Disconnected. The images show that just the 1A router port is currently connected, for seamless packet switching redundancy the 1B port should also be connected.

CONFIGURE>NETWORKS>AOIP CORE ROUTER INTERFACES



CONFIGURE>NETWORKS>AOIP CORE ROUTER INTERFACES-EDIT



The MAC address of each port is displayed but is not editable. The speed of each port is displayed (in Gbps).

Note: all Impulse/Type R ports operate at either 1Gbps or 10Gbps.

Users can edit these by clicking on the pencil icon for each port to open an 'Edit interface configuration' dialogue box as shown below right.

Once the IP address, Subnet mask and optional Gateway has been edited the user can click on the 'Save' button to update the port configuration and return to the AoIP interfaces page.

The page entries can be 'folded-up' in order to show just the AoIP core Primary and Secondary Router Interfaces that the user is interested in.

Virtual AoIP Interfaces (SSM)
In a multi-vendor Source Specific Multicast environment, if a Calrec core failover happens we are reliant on customers having an NMOS controller that will dynamically react to source address changes in our IS-04 advertisements, and for them to update all non-Calrec receivers accordingly.

However, there are some customers that do not wish to change every receiver in the event of our failover.

The solution to this is to provide an optional failover system where both primary & secondary cores can use the same Virtual AoIP source address so there is no source address change required.

To use this optional addressing method an Enable switch is provided as highlighted above right.

This allows Virtual Routers to be displayed and configured with additional Virtual /Aliased IP addresses for the transmitters.

The common IP addresses are applied to both the Primary and Secondary cores of a redundant pair, but only operated on the currently active core.

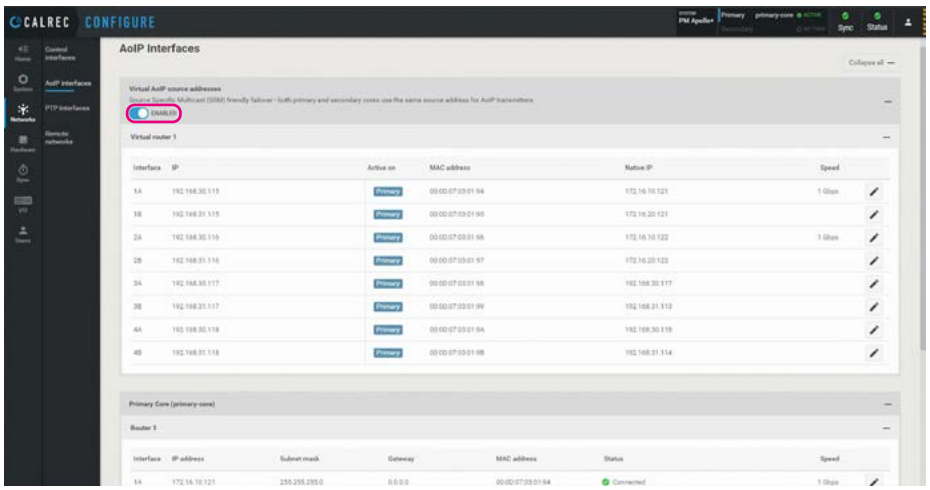
The image below right shows a Virtual IP address being edited for interface 1A on Virtual router 1 by tapping on the Edit icon on the end of each row.

Each row on the Virtual router also shows its MAC and Native IP addresses for reference.

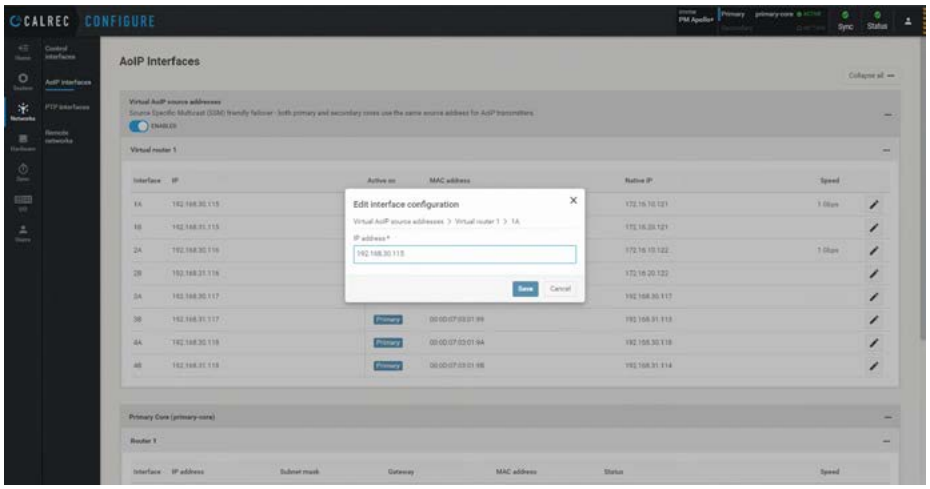
The tables at the end of this document offers suggested settings for the Primary and Secondary Router Interfaces of each of the Impulse or Type R Cores.

See:- **"Primary Core - Router Interfaces - Suggested Settings" on page 69** and **"Secondary Core - Router Interfaces - Suggested Settings" on page 70**

CONFIGURE>NETWORKS>VIRTUAL ROUTER SSM ADDRESSING



CONFIGURE>NETWORKS>VIRTUAL ROUTER SSM ADDRESSING EDIT



Also refer to the **Type R Configure Guide (926-285).pdf** or the **Impulse Configure Guide (926-290). pdf** for further information.

Once the AoIP Routers are setup in Configure the next step is to setup the AoIP devices for both the ores and the I/O.

To do this, access the **Connect** Application by opening Google Chrome, on your PC type **172.16.255.60** into the address bar and press enter.

This allows the user to setup the Transmitters/Receivers , Synchronisation, Audio Inputs & Outputs and the GPIO Triggers & Tallies.

AoIP Core Interfaces
This page is accessed from the Devices Menu in Connect and appears as shown above right. In this case, the '**PM Apollo+'** which is the chosen name for the core has been selected.

Each row on the interface sub-page table represents an AoIP interface connection, on an Impulse core there are a number of AoIP interfaces having both a Primary and Secondary connection. The columns in the table provide information about:-
- AoIP interface's status:- Green is active, Red is not active.
- AoIP interface name.
- AoIP interface IP address.
- AoIP interface's MAC address.
- AoIP interface connection speed.
- AoIP interface TX use percentage.
- AoIP interface RX use percentage.

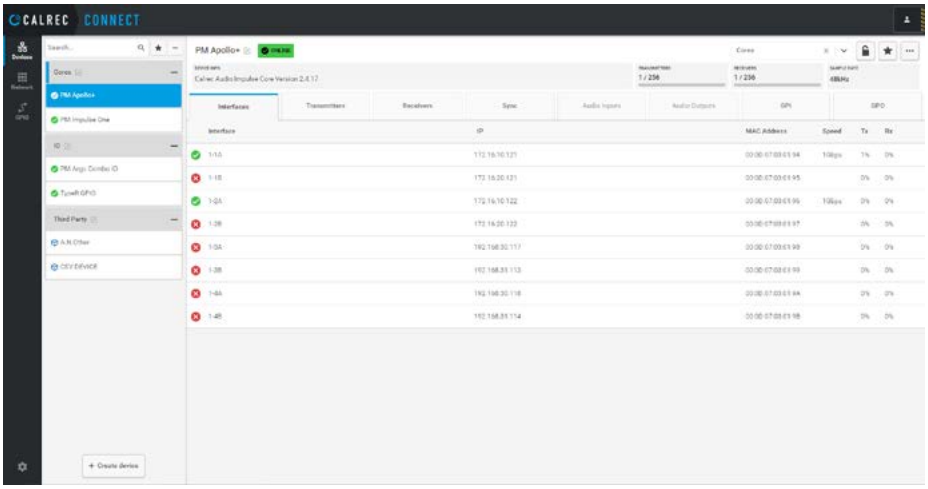
If a Virtual router has been configured for SSM addressing then the Connect> Devices page will appear as shown middle right, where the Virtual interface ports with their aliased IP addresses are also shown.

Note: toggling the Virtual AoIP source addresses 'Enable' switch on the relevant Configure page will hide as shown in the image above right or hide as shown middle right the Virtual router interfaces.

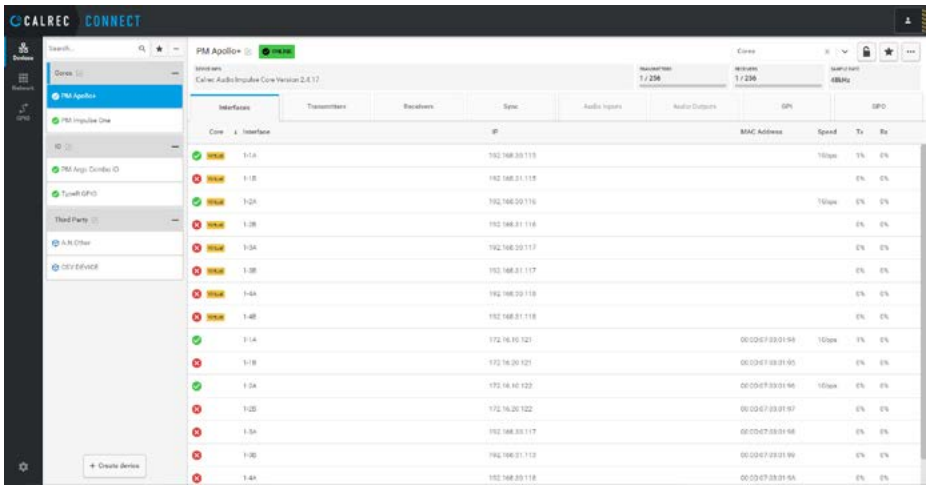
Configure IP Settings for AoIP Core Interfaces
Clicking on any of the AoIP interface rows on AoIP Core devices, opens the Edit interface dialogue for the IP settings for the selected AoIP interface as shown below right for the 1-1A primary router interface port for the PM Apollo+.

The user sets up the IP address by modifying the following 4 fields:-
Mode:- Type of IP address, i.e. if the IP address type is a static manually configured address or if it is a DHCP dynamically allocated address.
IP Address:- This example shows that the device & router are in the same subnet.
Subnet Mask:- Typically set to 255.255.255.0 or /24.
Gateway:- Optional

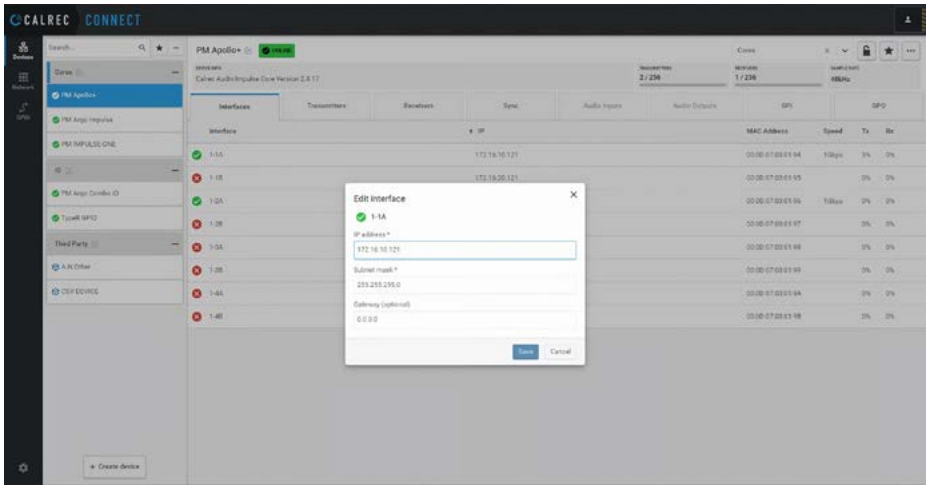
DEVICES >AOIP CORE INTERFACES



DEVICES>AOIP CORE INTERFACES WITH SSM ADDRESSING ENABLED



DEVICES>AOIP DEVICE INTERFACES-EDIT IP ADDRESS



AoIP Device Interfaces
This page is accessed from the Devices Menu in Connect and appears as shown above right. The '**PM Argo Combo IO'** which is the chosen name for the Argo AoIP device has been selected.

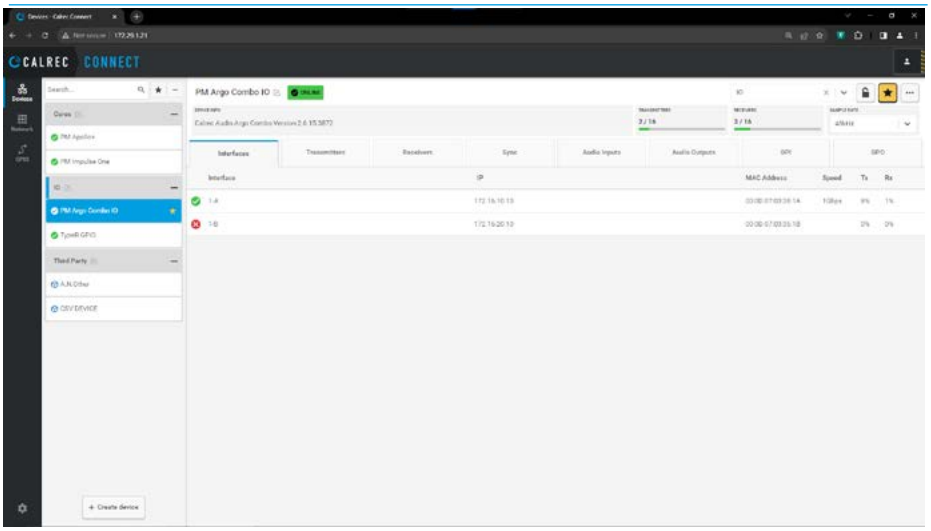
Each row on the interface sub-page table represents an AoIP interface connection, on an Argo AoIP device there is 1 set of AoIP interfaces having both a Primary and Secondary connection. The columns in the table provide information about:-
- AoIP interface's status:- Green is active, Red is not active.
- AoIP interface name.
- AoIP interface IP address.
- AoIP interface's MAC address.
- AoIP interface connection speed.
- AoIP interface TX use percentage.
- AoIP interface RX use percentage.

Configure IP Settings for AoIP Device Interfaces
Clicking on any of the AoIP interface rows on AoIP IO devices, opens the Edit interface dialogue for the IP settings for the selected AoIP interface as shown below right for the 1-A primary interface port for the PM Argo Combo IO.

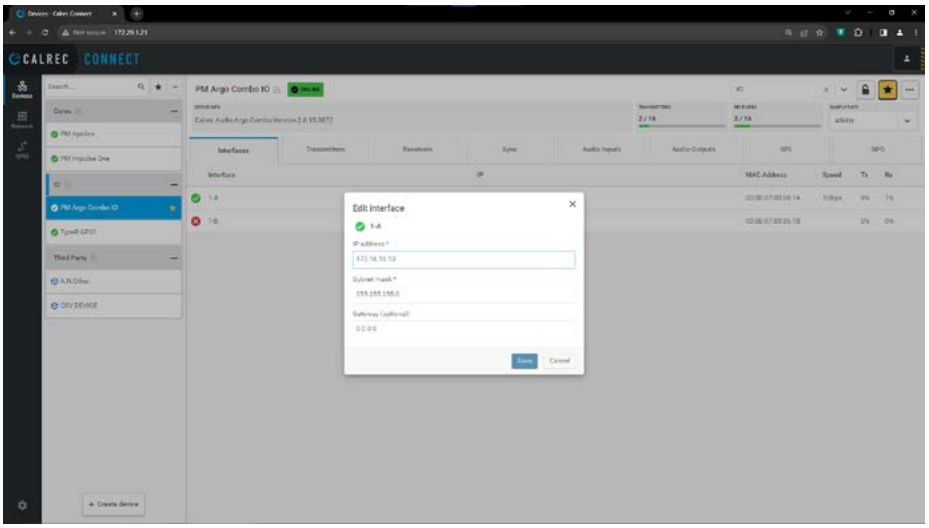
The user sets up the IP address by modifying the following 4 fields:-
Mode:- Type of IP address, i.e. if the IP address type is a static manually configured address or if it is a DHCP dynamically allocated address.
IP Address:- This example shows that the device & router are in the same subnet.
Subnet Mask:- Typically set to 255.255.255.0 or /24.
Gateway:- Optional

A table at the end of this document offers suggested settings for the IP addresses of the AoIP Device Primary & Secondary Network Interfaces. See:-
"AoIP Devices - Network Interfaces - Suggested Settings" on page 71

DEVICES >AOIP DEVICE INTERFACES



DEVICES>AOIP DEVICE INTERFACES-EDIT IP ADDRESS



Transmitters - AoIP Core
This page is accessed from the Devices menu when an AoIP Core's 'Transmitters' tab is selected and appears as shown above right.

In order to stream DSP audio from the core to an AoIP device, the user has to patch the audio to a core transmitter and connect it to an AoIP device receiver which, is then subsequently patched to audio output ports. This page is used to add/remove and configure transmitters.

The number of available transmitters shown here for the Impulse Core is 256.

This is based on having a one router installation. Each router has 4 AoIP Primary /Secondary interfaces, each of which carries 256 channels i.e. a total of 2048 channels. If each stream is carrying 8 channels then this provides 2048/8=256 available transmitters.

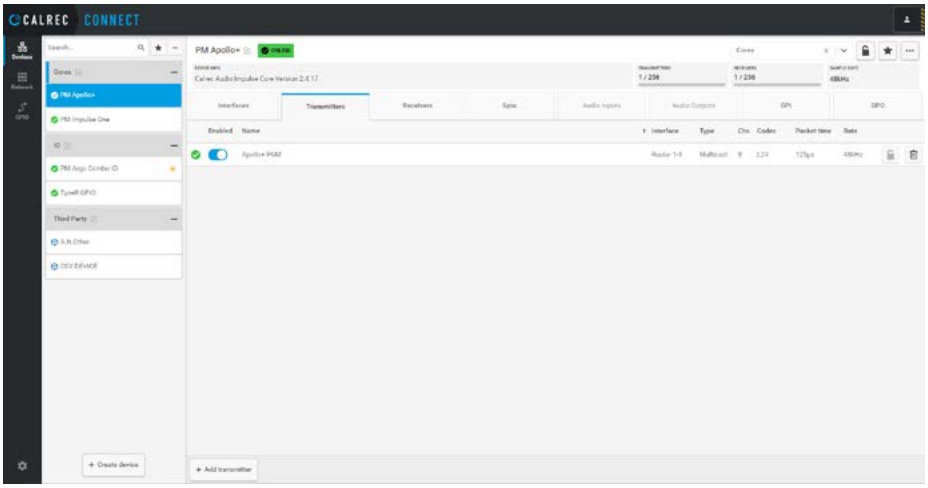
The number of transmitters for a Type-R Core is 64, this is based on having 2 AoIP interface Pri/Sec sets each of which carries 256 channels i.e. a total of 512 channels with each stream carrying 8 channels thus providing 512/8=64 available transmitters.

Each row on the transmitters sub-page table represents a transmitter as shown above right. The columns in the table provide the following information about:-
- Transmitter name e.g. 'Apollo+ PGM'.
- Transmitter interface e.g. Router 1-1.
- Type of transmitter, normally defined as a Multicast one to many type rather that a Unicast one to one type.
- Number of channels in each transmitter e.g. 8.
- Packet time e.g. 125us.
- Codec used e.g. L24.
- Sample Rate e.g. 48000Hz.

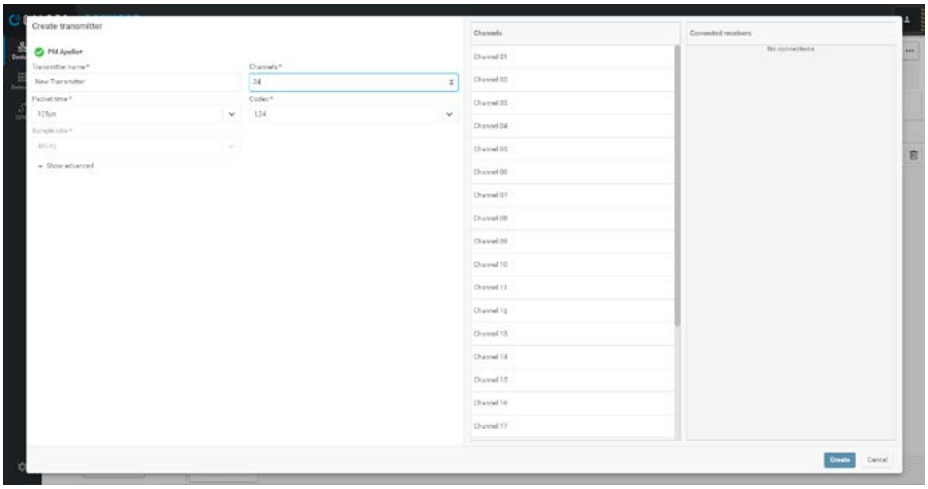
At the end of each transmitter row are two further options:- Lock & Delete.

Lock: Applying a Lock on the transmitter prevents any of the settings from being changed and appends a padlock icon when locked.

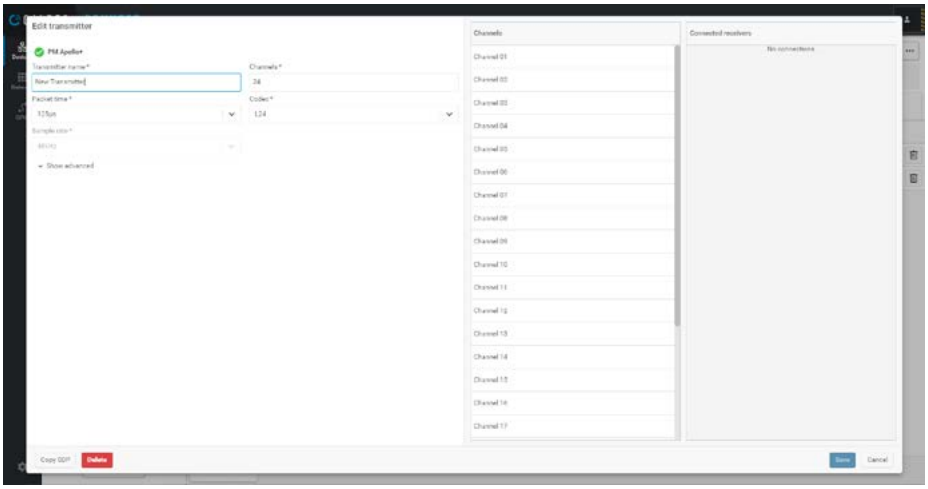
DEVICES>AOIP CORE TRANSMITTERS



DEVICES>AOIP CORE TRANSMITTERS-ADD TRANSMITTER



DEVICES>AOIP CORE TRANSMITTERS-EDIT/DELETE TRANSMITTER



When an existing locked transmitter is selected this becomes an 'Unlock' button which removes the lock restrictions and padlock icon.

Delete: This removes the transmitter from the device releasing the channels it used back into the available transmitter resource pool.

Transmitter Settings - AoIP Core
When the user selects one of the available transmitters to edit or clicks on '**Add Transmitter**' in the footer of the page.
A dialogue box appears allowing the user to configure the transmitter as shown middle right on the previous page. It shows information for **VIEWING ONLY** in **Grey** and also settings that can be **ADJUSTED** in **Black** as shown above right.

Transmitter Name: The user can edit the transmitter name by clicking in the Name field and saving it. These options can be edited at a later time or if the transmitter is no longer required it can be removed by pressing the Delete button in the Edit Receiver dialogue box shown below right on the previous page.

No of Channels in the Transmitter:
The maximum number of channels that can exist in a stream is 80, however a channel count of either 8, 16, 24 or 32 channels is more common for passing groups of related channels to a destination. As can be seen above right, the user is configuring a new transmitter to carry 32 channels.

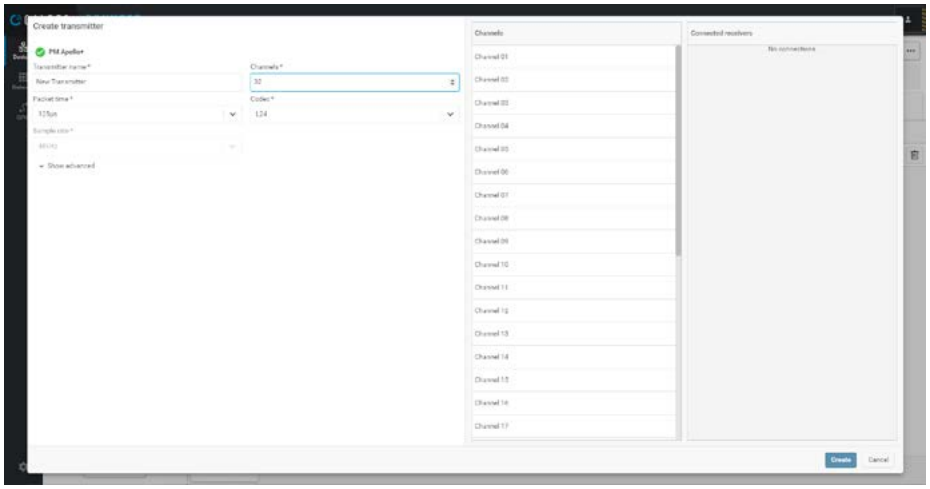
Packet Time:
The Packet Time can be set for the transmitter from the drop down list shown right. The default Packet time is 125us.

Sample Rate:
This is set from the Sample Rate Dropdown in the Device header see:-
"Change Sample Rate" on page 10

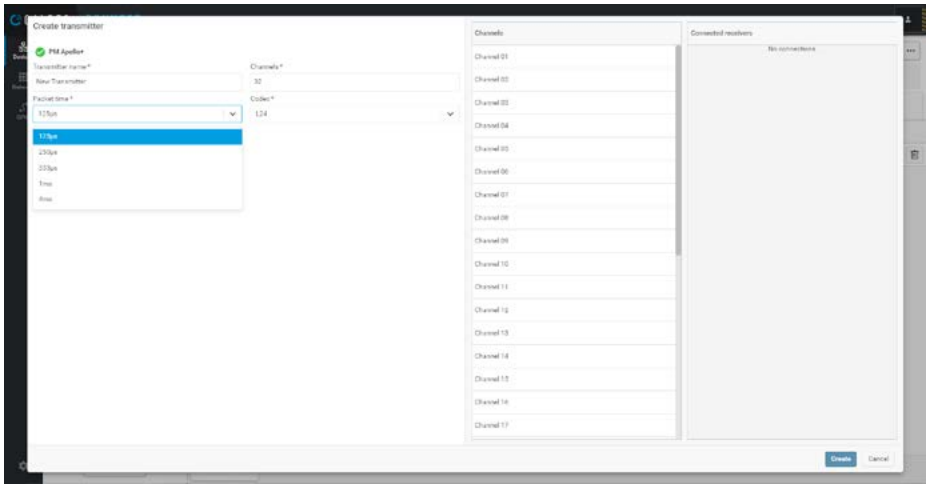
Codec Options-Transmitter:
As can be seen from the image below right the Codec type/size can be selected from the dropdown box. The most commonly used Codec for AoIP /AES67 broadcast is the L24 which carries 24 bit audio samples in two's complement format.
The other options are L16 (16 bit audio) and Calrec32, a 32-bit format which allows for greater bandwidth between Calrec AoIP devices such as in a Type-R system, but 3rd party devices may not be able to handle this Codec.

Note: with this Codec the number of channels may be restricted if using a larger packet size such as 1ms.

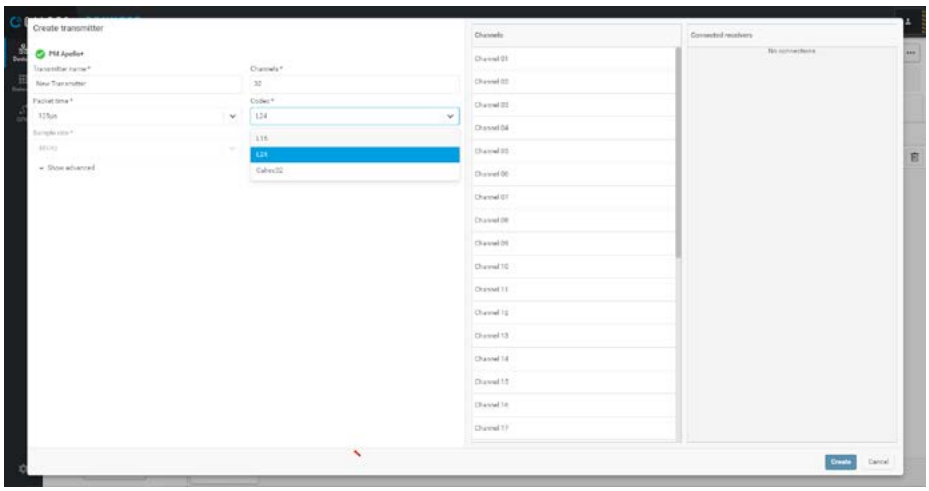
DEVICES>AOIP CORE TRANSMITTERS- NUMBER OF CHANNELS



DEVICES>AOIP CORE TRANSMITTERS-PACKET TIME OPTIONS



DEVICES>AOIP CORE TRANSMITTERS-CODEC OPTIONS



Advanced Settings - AoIP Core TX
As can be seen from the images above and below right there is an advanced settings section in the dialogue box that shows advanced information for **VIEWING ONLY** in **Grey** and also settings that can be **ADJUSTED** in **Black** as shown above right.

ID: The stream identifier session ID. This is shown at the top of the advanced section.

DSCP Level: "Differentiated services code point" this defines the traffic priority and packet drop probability. DSCP=34 can also be listed as "AF41" on other devices. This value should be used for AES67 audio streams.

TTL: "Time to live" value defines the number of network hops a packet can travel over before the packet is dropped.

Payload type: This is a 7-bit numeric identifier that identifies a payload format. Payload type 97 represents a Dynamic Audio payload format.

Interface: This dropdown selects which router interface this stream will be transmitted on. Each router has 4 pairs of Primary & Secondary interfaces for redundancy and the user has selected the **Router 1: 1A - Router 1:1B** interface pair as shown below right.

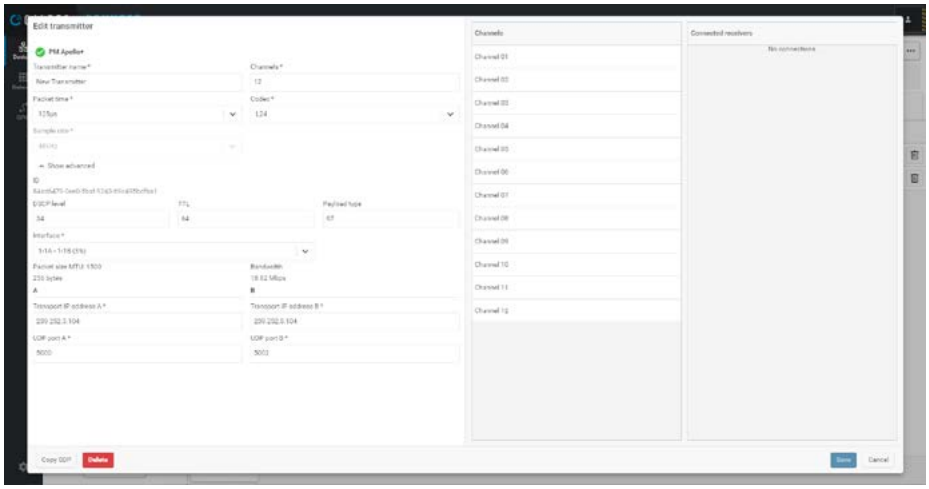
Packet Size: Number of bytes in Packet.

Bandwidth: Stream Bandwidth in Mb/s.

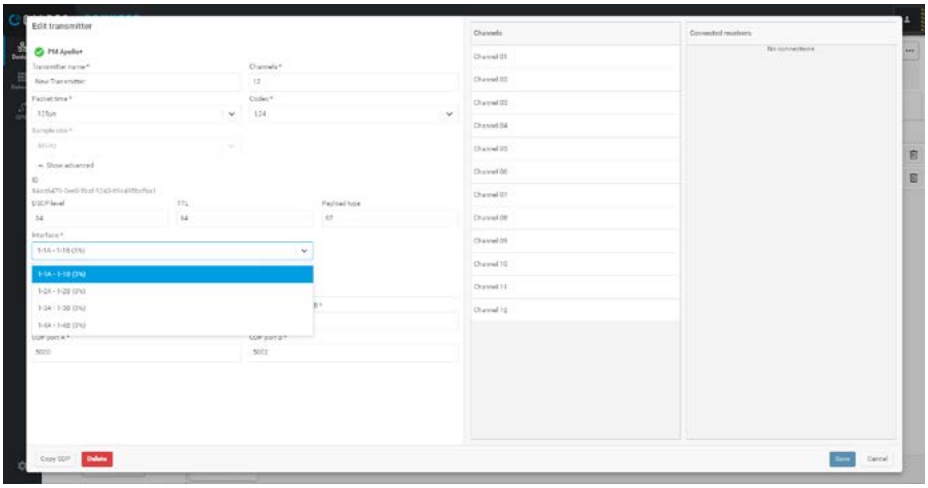
Transport IP: for A & B router port connections,the transport IP or Multicast addresses are the addresses on the switch that source audio is streamed to.

UDP Port: Specific port number on an IP address for use on streams on Router Ports A &B for real time audio transfer.

DEVICES>AOIP CORE TRANSMITTERS-ADVANCE SETTINGS 1



DEVICES>AOIP CORE TRANSMITTERS-ADVANCE SETTINGS 2



Copy SDP: This copies the SDP to the users PC for pasting session data into a 3rd party application and vice-versa with the Paste option as shown above right in the Edit Transmitter dialogue box.

It is used to describe multimedia communication sessions for the purposes of session announcement, session invitation, and parameter negotiation.

SDP does not deliver media itself but is used between end points for negotiation of media type, format, and all associated properties.

The set of properties and parameters are often called a session profile.

Transmitters - AoIP Devices
This page is accessed from the Devices menu when an AoIP device's 'Transmitters' tab is selected and appears as shown above right.

The number of transmitters shown here for an Argo AoIP device is 16, this is based on there being 1 AoIP Primary / Secondary Interface which carries 128 channels. If each stream carries 8 channels then this provides 128/8=16 available transmitters, however it should be noted that one of the I/O streams is carrying 64 channels i.e. ARGO MADI IN. Each row on the transmitters sub-page table represents a transmitter.

Transmitter Settings - AoIP Devices
When the user selects one of the available transmitters or clicks on 'Add Transmitter' in the footer of the page, a dialogue box appears allowing the user to configure the transmitter. It shows information for **VIEWING ONLY** in **Grey** and also settings that can be **ADJUSTED** in **Black.**

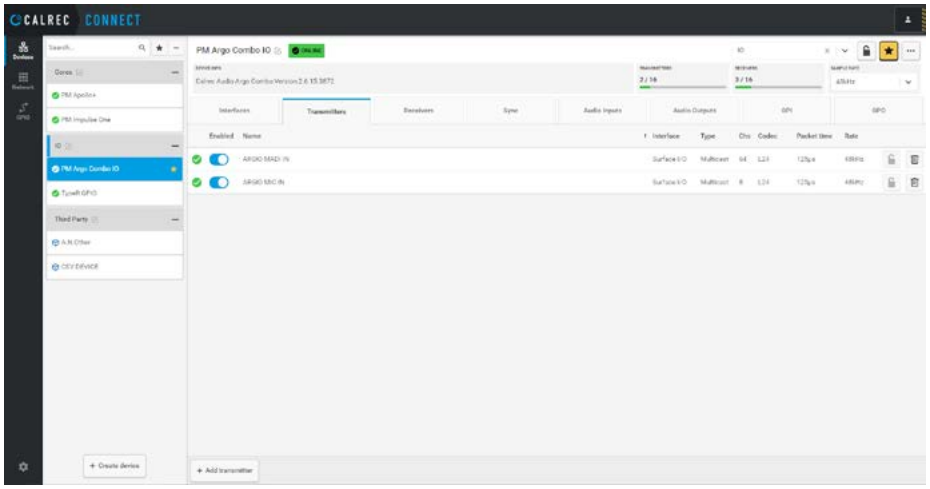
The dialogue box for the AoIP device transmitter settings as shown right has the same structure as those for the Transmitter Settings - AoIP Core. See **"Transmitters - AoIP Core" on page 22**

Advanced Settings - AoIP Device TX
There is an advanced settings section in the dialogue box, that shows advanced information for **VIEWING ONLY** in **Grey** and also settings that can be **ADJUSTED** in **Black**.

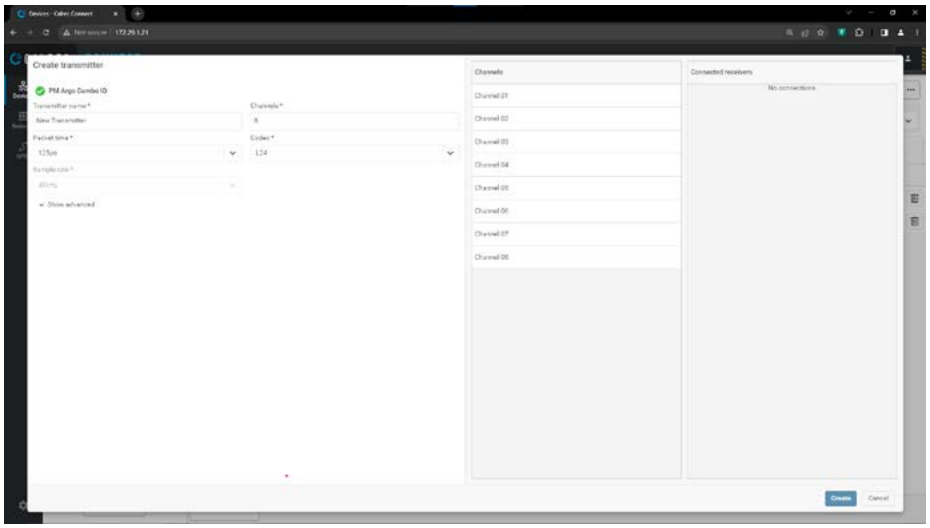
Interface: As can be seen from the image below right this dropdown selects which device interface this transmitter stream will be carried on, for Argo Combo AoIP devices there is only 1 pair of Primary & Secondary interfaces for redundancy, **1-A & 1-B**.

The rest of the Advanced AoIP device transmitter settings has the same structure as those for the Advanced Settings - Core transmitter. See **"Advanced Settings - AoIP Core TX" on page 24**

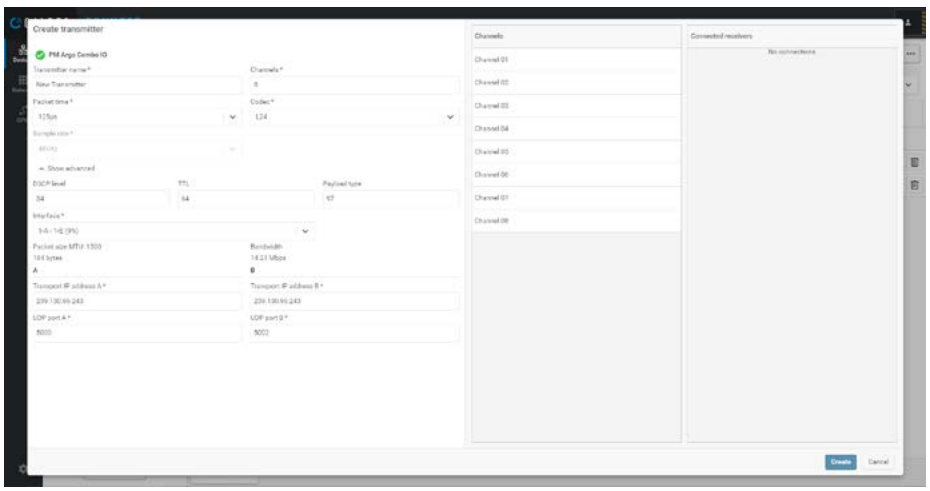
DEVICES>AOIP DEVICE TRANSMITTERS



DEVICES>AOIP DEVICE TRANSMITTER-SETTINGS



DEVICES>AOIP DEVICE TRANSMITTER-ADVANCED SETTINGS



Receivers - AoIP Core

This page is accessed from the Devices menu when an AoIP Core's 'Receivers' tab is selected and appears as shown above right. In order to stream audio data from an AoIP device to the core, the user has to patch audio input ports in the AoIP device to a transmitter and connect it to a core receiver which is then subsequently patched to DSP inputs in the core.
This page used to add/remove and configure receivers.

The number of available receivers shown here for this Impulse 1 Core is 64 and has up to 2 AoIP A/B router pairs. Each router has 1 AoIP Primary /Secondary interfaces, which carries 256 channels at 1Gbps. If each stream is carrying 8 channels then this provides 512/8=64 available receivers, however it should be noted that one of the I/O ports is carrying 64 channels i.e. 'IMP1 64 ch in'.

Each row on the receiver sub-page table represents a receiver.
The columns in the table provide the following information about:-
- Receiver name e.g. 'Imp1 Inputs 1-8'.
- Receiver interface e.g. Impulse 1-1
- Number of channels in each receiver e.g. 8.
-Link Offset e.g. 2000us or 2ms.
At the end of each receiver row are two further options:- Lock & Delete

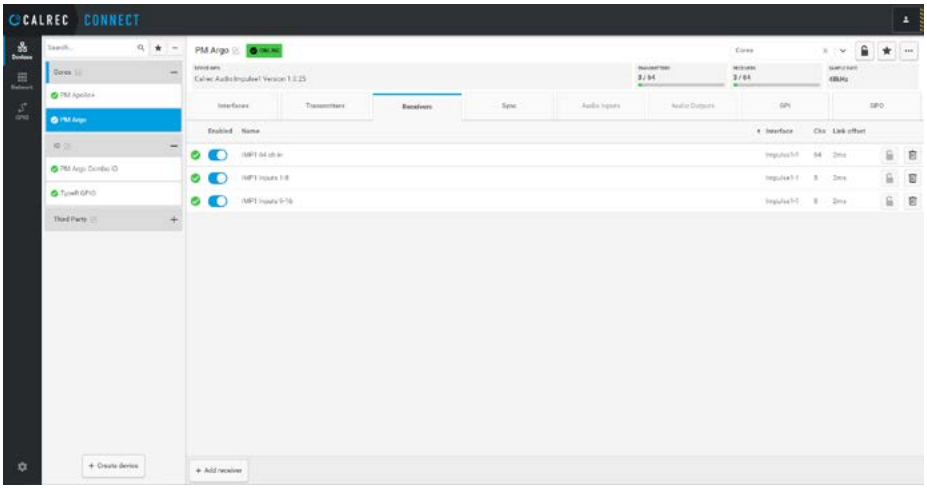
Lock: Applying a Lock on the receiver prevents any of the settings from being changed and appends a padlock icon when locked.
When an existing locked receiver is selected this becomes an 'Unlock' button which removes the lock restrictions and padlock icon.

Delete: This removes the receiver from the device releasing the channels it used back into the available receiver resource pool.

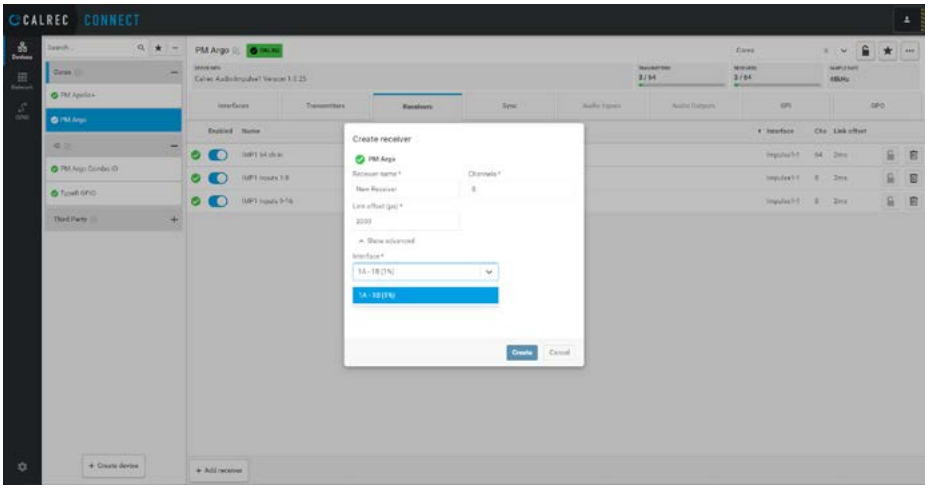
Receiver Settings - AoIP Core

When the user selects one of the available receivers or clicks on 'Add Receiver' in the footer of the page, a dialogue box appears allowing the user to configure the receiver.

DEVICES>AOIP CORE RECEIVERS



DEVICES>AOIP CORE RECEIVER-SETTINGS AND ADVANCED SETTINGS



Receiver Name:
From the image below right, the user can edit the receiver name by clicking in the Name field and saving it. The receiver options can also be edited at a later time or if the receiver is no longer required it can be removed by pressing the Delete button in the Edit Receiver dialogue box.

No of Channels in the Receiver:
The maximum number of channels that can exist in a stream is 80, typically however a channel count of either 8 or 16 channels is more common for passing groups of related channels to a destination. As can be seen from the image below right, the receiver is set to accept 8 channels.

Link Offset: This is the duration in microseconds of the network latency, at a packet time of 125us, the minimum theoretical setting being two times the packet time i.e. 250us and the default setting is 2000us which is sixteen times the packet size i.e. 2ms.

Advanced Settings - AoIP Core RX
There is an advanced settings section which in this case has been opened.

Interface: This dropdown selects which router interface this stream will be received on. Each router has 4 pairs of Primary & Secondary interfaces for redundancy and the user has selected the **Router 1: 1A - Router 1:1B** interface pair.

Receivers - AoIP Devices
This page is accessed from the Devices menu when an AoIP device's 'Receivers' tab is selected and appears as shown above right.

The number of receivers shown here for an Argo AoIP device is 16, this is based on there being 1 AoIP Primary /Secondary Interface which carries 128 channels.

If each stream carries 8 channels then this provides 128/8=16 available receivers, however it should be noted that one of the I/O streams is carrying 64 channels i.e. ARGO MADI OUT.

Lock: Applying a Lock on the receiver prevents any of the settings from being changed and appends a padlock icon when locked.

When an existing locked receiver is selected this becomes an 'Unlock' button which removes the lock restrictions and padlock icon.

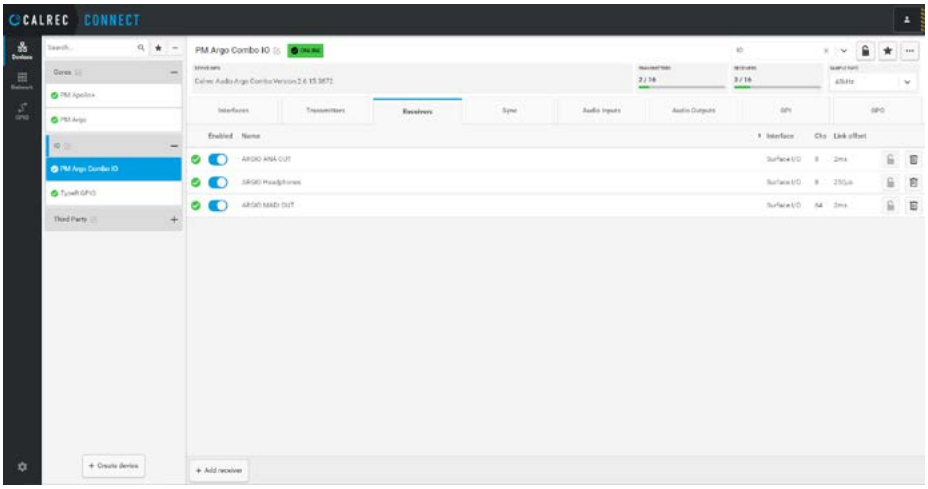
Delete: This removes the receiver from the device releasing the channels it used back into the available receiver resource pool.

Receiver Settings - AoIP Devices
When the user selects one of the available receivers or clicks on 'Add Receiver' in the footer of the page, a dialogue box appears allowing the user to configure the receiver.

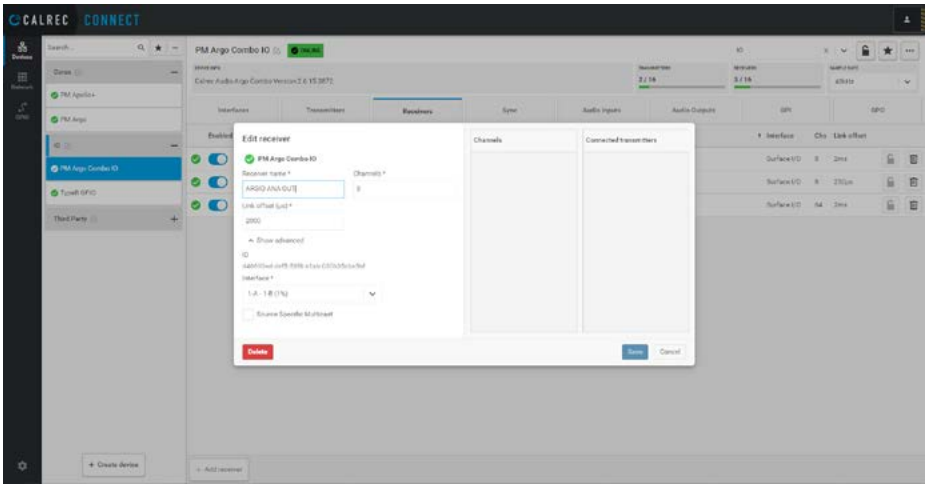
The dialogue box for editing the AoIP device receiver settings as shown below right has the same structure as those for the Receiver Settings - Mixer section.
See **"Receiver Settings - AoIP Core" on page 26.**

Advanced Settings - AoIP Device RX
There is an advanced settings section which in this case has been opened.

DEVICES>AOIP DEVICE RECEIVERS



DEVICES>AOIP DEVICE RECEIVER-SETTINGS & ADVANCED SETTINGS



Interface: This dropdown selects which device interface this stream will be received on, for Argo AoIP devices there is only 1 pair of Primary & Secondary interfaces for redundancy, **1-A & 1-B**.

The Advanced AoIP device receiver settings has the same structure as those for the Advanced Settings - Core receiver.
See **"Advanced Settings - AoIP Core RX" on page 26.**

UNDERSTANDING SYNCHRONISATION

PTP & Synchronisation
Precision Timing Protocol, specifically version 2 is also known as IEEE-1588-2008. PTP is a method of synchronising clocks across a network very accurately. For Media networks of a larger scale, it is important to understand the PTP configuration and to ensure all switches and end-points are correctly configured for PTP. Getting this wrong will result in unstable audio.

All AoIP end-points contain their own clock, and PTP is used to synchronise those clocks. One device will act as the master, this could be one of the AoIP end-points, or a PTP capable switch on the network, but having a dedicated Grand-Master device is recommended for a broadcast media network.

The Master sends out time-of-day messages to all the slaved devices (devices that are referencing a master). It takes a finite amount of time for a message to travel across a network, so by the time a slave receives a time-of-day message, it is already out-of-date as time has moved on since it was originally time-stamped and sent out by the master. Slaves exchange "receipt request/response" messages with the master, and by timing how long it takes to get a response, they are able to adjust the timestamps to factor in the network latency and therefore maintain their clock in relation to the master with a very high degree of accuracy.

On a small network with a few end-points connected to a single switch, cost effective non PTP aware switches like Cisco SG350 series can be used as long as they are is correctly configured for IGMP and QoS. As networks scale up, PTPv2 aware switches become very important as PTP aware switches can adjust the time-stamps of PTP messages that they are forwarding on, to factor in the delay caused by the switch by the amount of time it takes to receive and forward on the message.

Synchronising with PTPv2-IEEE1588
The following PTP Terms are used when deciding which device is Grandmaster.

Best Master Clock Algorithm:
This is used by all devices within the same domain to determine which device is most suitable to become the Grandmaster.

The following attributes are used to determine this (in order of priority):-

Priority 1 – the user can assign a specific static-designed priority to each clock, preemptively defining a priority among them.

Class – each clock is a member of a given class, each class getting its own priority.

Accuracy – precision between clock and UTC, in nanoseconds (ns).

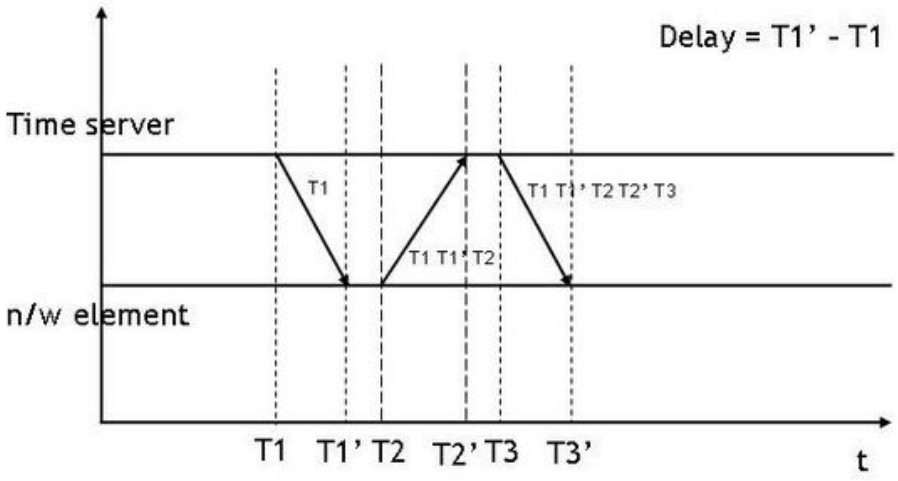
Variance – variability of the clock.

Priority 2 – final-defined priority, defining backup order in case the other criteria were not sufficient.

Source Port ID - If the BMCA has failed to choose a master from the above criteria, it resorts to using the clock with lowest port ID (usually its MAC address)

BMCA makes these checks in order, and stops as soon as it has identified a master at any stage (it does not check the next step unless there is no clear choice based on the current step).

A PTP CYCLE



Once the PTP GMC has been determined, all clocks will begin to sync to the newly elected clock.

- Master will send "sync_message" to all slaves (packet timestamped from GMC)
- Slave will timestamp the packet on reception
- Slave replies with "delay_request" to the GMC
- GMC will timestamp the packet on reception
- GMC sends "delay_response" to the slave.
- This packet will include the GMC receipt timestamp of the "delay_request"

By calculating the offset between the timestamps of these message being sent and subsequently received. A slave device can correctly adjust its clock to match that of the GMC to less than 1ms.

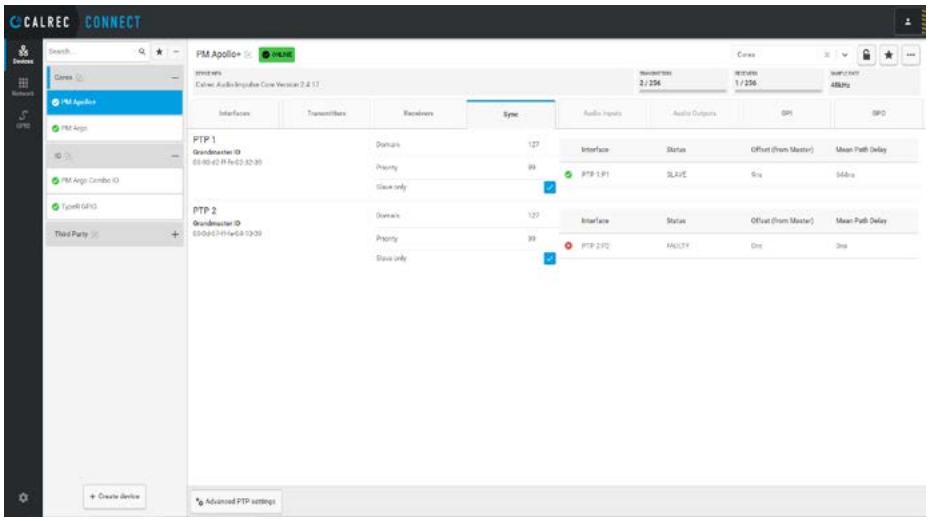
Synchronisation - Impulse Core
This page is accessed from the Devices menu when a Core device's 'Sync' tab is selected and appears as shown above right for '**PM Apollo+**'. It provides the ability to synchronise the network clocks of the devices by setting up PTP clocks and determine how the AoIP interfaces respond on the selected core. The two PTP interfaces on the Impulse Core are located on the front of the Control Module and are described as PTP 1 and PTP 2 on the synchronisation screens.

The sync page is split into 2 areas:-
The right hand area shows the Port Status of each PTP interface with respect to the master clock. The columns in this Port Status table provide information about:-
- Port status indicators:- Green is active, Red is not active.
- Primary and Secondary port identifiers for each device i.e. PTP 1:P1, PTP 2:P2.
- Port statuses:- MASTER provides a clock source for it's downstream connections, SLAVE follows the clock source that is in the Master state or it is identified as FAULTY or passive which is neither the master or a slave of a master, or when no connection has been made.
- Offset (from Master):- This is a measure of how accurately a slave synchronises with a master clock.
- Mean Path Delay is the average time taken for PTP frames to travel between master and slave, a large mean path delay is indicative of jitter and latency.

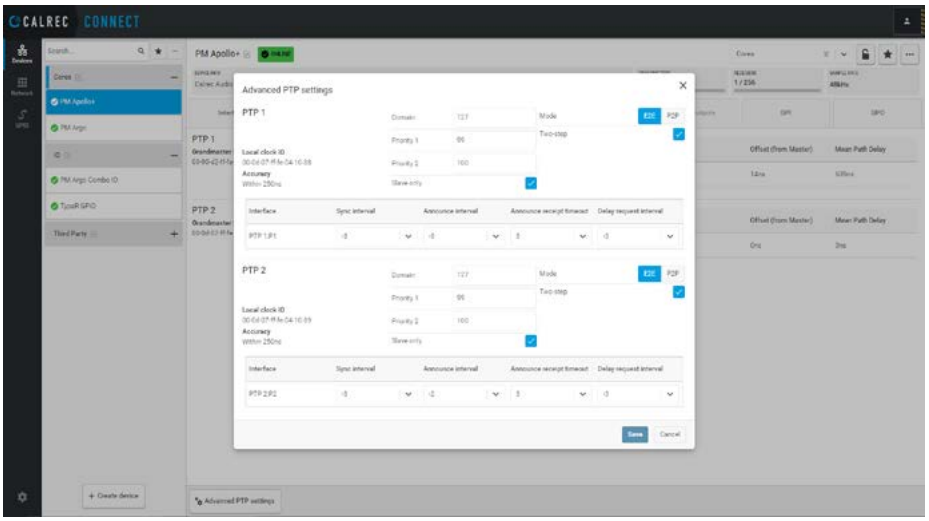
The left hand area displays the PTP Grandmaster ID of the current clock source. The settings to the right of this allow the user to modify how each of the interfaces will interact with the clock. There are basic configuration settings and advanced configuration settings which are accessed from the 'Advanced settings' button in the footer of the page.

Basic Synchronisation Settings
As shown above and below right:-
Domain: Multiple PTP clocks can exist on the same network these are kept separate by placing them in different domains. The domain number (0-255) selects the domain to be used.

DEVICES>SYNC>BASIC IMPULSE CORE SYNCHRONISATION



DEVICES>SYNC>ADVANCED IMPULSE CORE SYNCHRONISATION



Priority: This value (0-255) is used in the Best Master Clock Algorithm to determine which AES67 device will become the PTP Grand Master Clock in that domain.
Slave Only: This tick box is used to prevent a device from establishing itself as a Master and should be selected when locking to an external PTP clock.

Advanced Synchronisation Settings
As shown below right:-
Priority 1: Same as basic Priority setting.
Priority 2: Backup Priority setting.
Modes: These transparent clock modes calculate the required time it takes to send traffic either from E2E (end to end) or from P2P (peer to peer) and updates the PTP time correction field as required.
Two Step: This improves the flexibility of the synchronisation system by first sending the sync message followed by a separate time stamp.
Sync Interval: Time period between sync messages which can vary from -7 to +1.
Announce Interval: Time period between PTP announcement messages which can vary from -3 to +4.
Announce Receipt Timeout: Number of PTP intervals a device can miss before timing out which can vary from 2 to 10.
Delay Request Interval: Time period between PTP delay request messages which can vary from -1 to +6.

Note:- in the images above the Secondary PTP port is not connected.

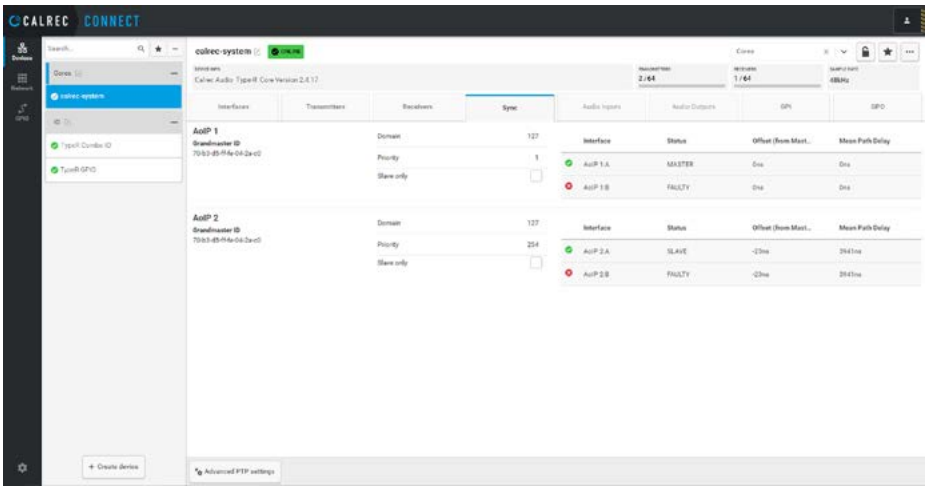
Synchronisation - Type R Core
This page is accessed from the Devices menu when a Core device's 'Sync' tab is selected and appears as shown above right for '**calrec-system**'. It provides the ability to synchronise the network clocks of the devices by setting up PTP clocks and determine how the AoIP interfaces respond on the selected core. The two AoIP interfaces on the Type R Core are located on the back of the core and are described as AoIP 1 and AoIP 2 on the synchronisation screens.

The sync page is split into 2 areas:-
The right hand area shows the Port Status of each AoIP interface with respect to the master clock. The columns in this Port Status table provide information about:-
- Port status indicators:- Green is active, Red is not active.
- Primary and Secondary port identifiers for each device i.e. AoIP 1:A, AoIP 1:B AoIP 2:A and AoIP 2:B.
- Port statuses:- MASTER provides a clock source for it's downstream connections, SLAVE follows the clock source that is in the Master state or it is identified as FAULTY or passive which is neither the master or a slave of a master, or when no connection has been made.
- Offset (from Master):- This is a measure of how accurately a slave synchronises with a master clock.
- Mean Path Delay is the average time taken for PTP frames to travel between master and slave, a large mean path delay is indicative of jitter and latency.

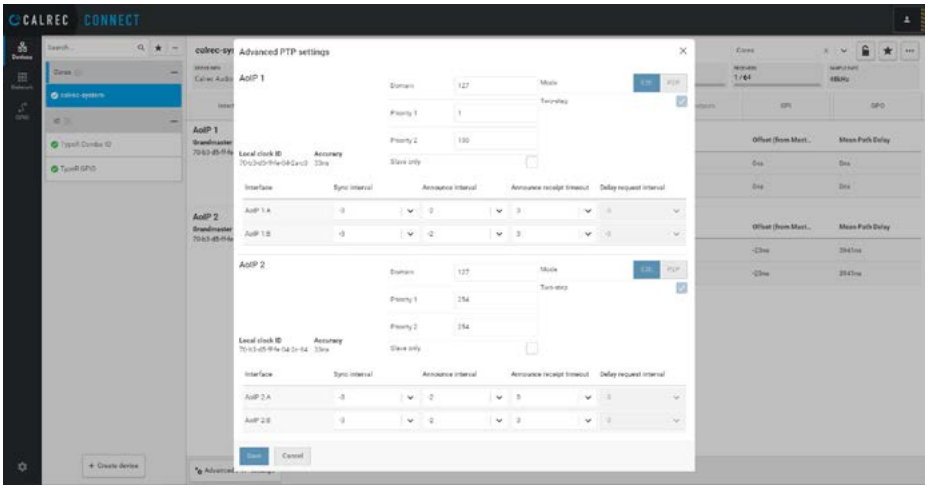
The left hand area displays the PTP Grandmaster ID of the current clock source. The settings to the right of this allow the user to modify how each of the interfaces will interact with the clock. There are basic configuration settings and advanced configuration settings which are accessed from the 'Advanced settings' button in the footer of the page.

Basic Synchronisation Settings
As shown above and below right:-
Domain: Multiple PTP clocks can exist on the same network these are kept separate by placing them in different domains. The domain number (0-255) selects the domain to be used.

DEVICES>SYNC>BASIC TYPE R CORE SYNCHRONISATION



DEVICES>SYNC>ADVANCED TYPE R CORE SYNCHRONISATION



Priority: This value (0-255) is used in the Best Master Clock Algorithm to determine which AES67 device will become the PTP Grand Master Clock in that domain.
Slave Only: This tick box is used to prevent a device from establishing itself as a Master and should be selected when locking to an external PTP clock.

Advanced Synchronisation Settings
As shown below right:-
Priority 1: Same as basic Priority setting.
Priority 2: Backup Priority setting.
Modes: These transparent clock modes calculate the required time it takes to send traffic either from E2E (end to end) or from P2P (peer to peer) and updates the PTP time correction field as required.
Two Step: This improves the flexibility of the synchronisation system by first sending the sync message followed by a separate time stamp.
Sync Interval: Time period between sync messages which can vary from -7 to +1.
Announce Interval: Time period between PTP announcement messages which can vary from -3 to +4.
Announce Receipt Timeout: Number of PTP intervals a device can miss before timing out which can vary from 2 to 10.
Delay Request Interval: Time period between PTP delay request messages which can vary from -1 to +6.

Note:- in the images above the Secondary AoIP ports are not connected.

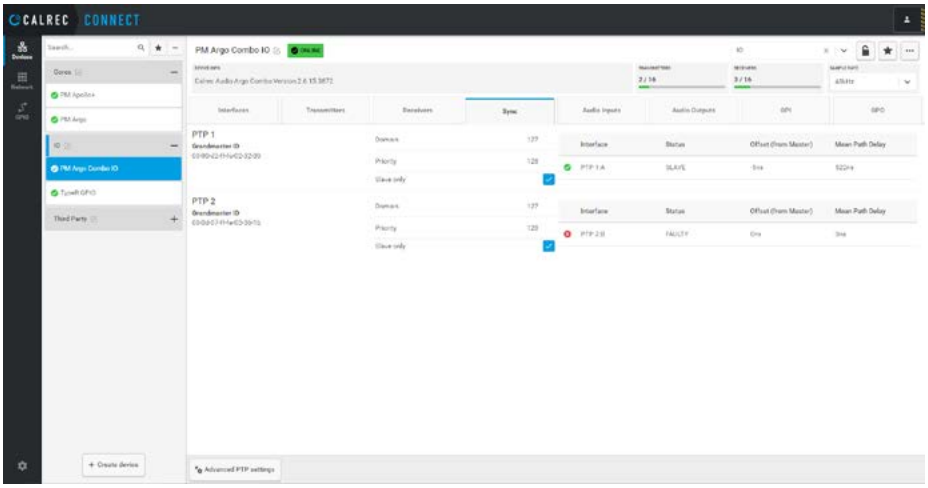
Synchronisation - AoIP Devices
This page is accessed from the Devices menu when an AoIP device's 'Sync' tab is selected and appears as shown above right for '**PM ARGO Combo IO**'. It provides the ability to synchronise the network clocks of the devices. The network clocks are carried in the ethernet data packets so do not need separate synchronisation inputs. This page is used to set the domain and priority of the selected device to determine its response.

The sync page is split into 2 areas:-
The right hand area shows the Port Status of the AoIP device interface primary and secondary ports with respect to the master clock. The columns in this Port Status table provide information about:-
- Port status indicators, Green is active, Red is not active.
- Primary and Secondary port identifiers for each device i.e. 1A & 1B.
- Port statuses:- MASTER provides a clock source for it's downstream connections, SLAVE follows the clock source that is in the Master state or it is identified as FAULTY or passive which is neither the master or a slave of a master, or when no connection has been made.
- Offset (from Master):- This is a measure of how accurately a slave synchronises with a master clock.
- Mean Path Delay is the average time taken for PTP frames to travel between master and slave, a large mean path delay is indicative of jitter and latency.

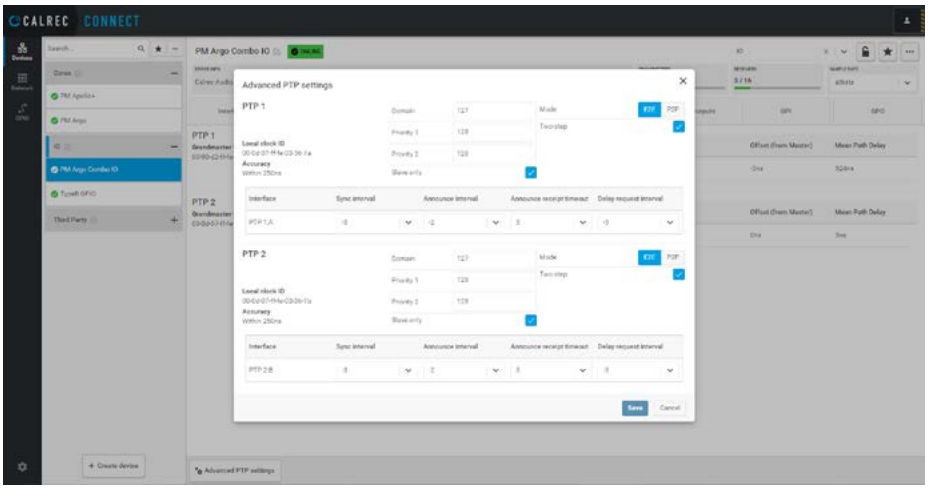
The left hand area displays the PTP Grandmaster ID of the current clock source. The settings to the right of this allow the user to modify how the interface will interact with the clock.
There are basic configuration settings and advanced configuration settings which are accessed from the 'Advanced settings' button in the footer of the page.

Basic Synchronisation Settings
As shown above and below right:-
Domain: Multiple PTP clocks can exist on the same network these are kept separate by placing them in different domains. The domain number (0-255) selects the domain to be used.

DEVICES>SYNC>BASIC AOIP DEVICE SYNCHRONISATION



DEVICES>SYNC>ADVANCED AOIP DEVICE SYNCHRONISATION



Priority: This value (0-255) is used in the Best Master Clock Algorithm to determine which AES67 device will become the PTP Grand Master Clock in that domain.
Slave Only: This tick box is used to prevent a device from establishing itself as a Master and should be selected when locking to an external PTP clock.

Advanced Synchronisation Settings
As shown below right:-
Priority 1: Same as basic Priority setting.
Priority 2: Backup Priority setting.
Modes: These transparent clock modes calculate the required time it takes to send traffic either from E2E (end to end) or from P2P (peer to peer) and updates the PTP time correction field as required.
Two Step: This improves the flexibility of the synchronisation system by first sending the sync message followed by a separate time stamp.
Sync Interval: Time period between sync messages which can vary from -7 to +1.
Announce Interval: Time period between PTP announcement messages which can vary from -3 to +4.
Announce Receipt Timeout: Number of PTP intervals a device can miss before timing out which can vary from 2 to 10.
Delay Request Interval: Time period between PTP delay request messages which can vary from -1 to +6.

Note:- in the images above the Secondary AoIP port is not connected.

CONFIGURE AUDIO INPUTS AND AUDIO OUTPUTS

Audio Inputs configuration
This page is accessed from the Devices menu when an AoIP device's '**Audio Inputs**' tab is selected and appears as shown above right for Mic/AES audio inputs and below right for MADI inputs.

Along the top are shown 2 transmitter streams which can be folded open + or closed- to make the streams easier to manage.

The second transmitter stream '**ARGO MIC IN**' is expanded to show the stream channels.

Down the left side are shown the physical input ports available in the device and the user simply clicks on the crosspoint between the input ports and transmitter stream channels to patch them together.

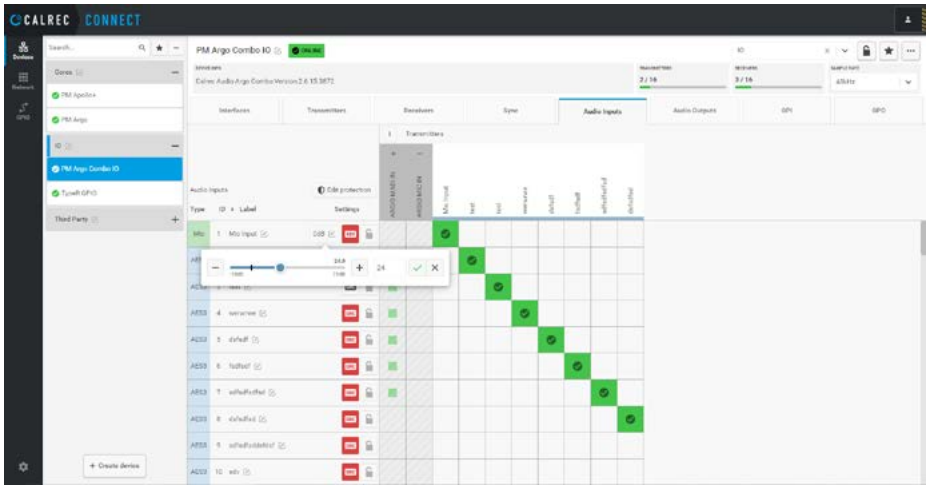
In addition to allowing the user to connect audio input ports to transmitter channels, the user can give each input port a Label, a Gain value and switch the 48v phantom power setting if it is a Mic/Line input or, the SRC if it is an AES input.

It also allows the user to Lock out the port allocation to prevent any parameter changes which are described below, from being made.

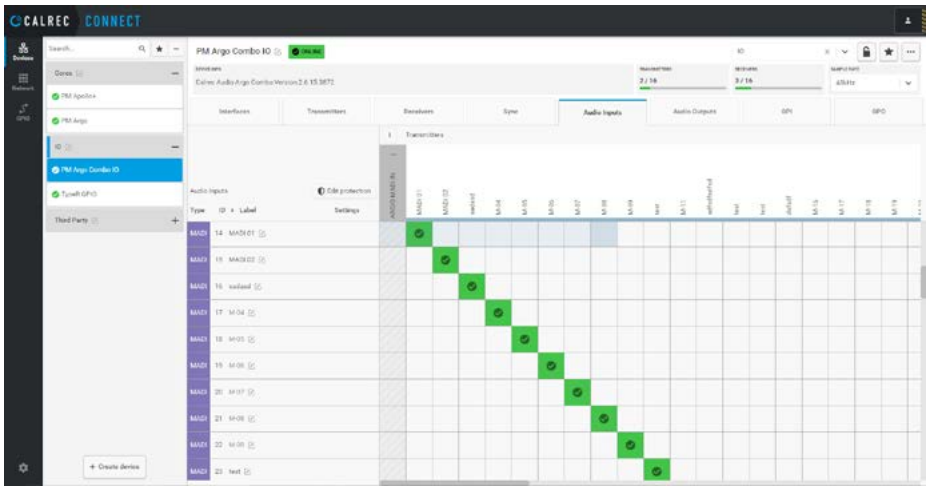
Gain Selection (Mic/Line Only)
The user clicks on the existing gain setting and either enters a gain value or moves the slider to the desired setting and clicks on the '✓' symbol.

48v or SRC Selection (Mic or AES)
The user clicks on the 48v symbol or SRC symbol next to the gain setting this will toggle the state of that function and will appear in Red when active.

DEVICES>MIC/AES AUDIO INPUTS TO AOIP DEVICE TRANSMITTER CHANNELS



DEVICES>MADI AUDIO INPUTS TO AOIP DEVICE TRANSMITTER CHANNELS



The image below right shows the connection of MADI Audio input ports on the left to the top where there are shown transmitter streams which can be folded open + or closed- to make the streams easier to manage.

In this image the first transmitter stream '**ARGO MADI IN**' is expanded to show the stream channels which are typically connected in the one to one diagonal pattern shown, however the MADI transmitter could be replaced with a number of smaller transmitter streams as required to arrange the inputs into logical groups if necessary.

Audio Outputs configuration
This page is accessed from the Devices menu when an AoIP device's 'Audio Outputs' tab is selected and appears as shown above right for Line and AES outputs and below right for MADI outputs.

Down the left side are shown 3 receivers which can be folded open + or closed- to make the receiver channels easier to manage.

The second receiver '**ARGIO Line Out**' & the third receiver '**ARGIO Headphones**' are both expanded to show the receiver channels.

Along the top are shown the physical output ports available in the device and the user simply clicks on the crosspoint between the receiver channels and output ports to patch them together.

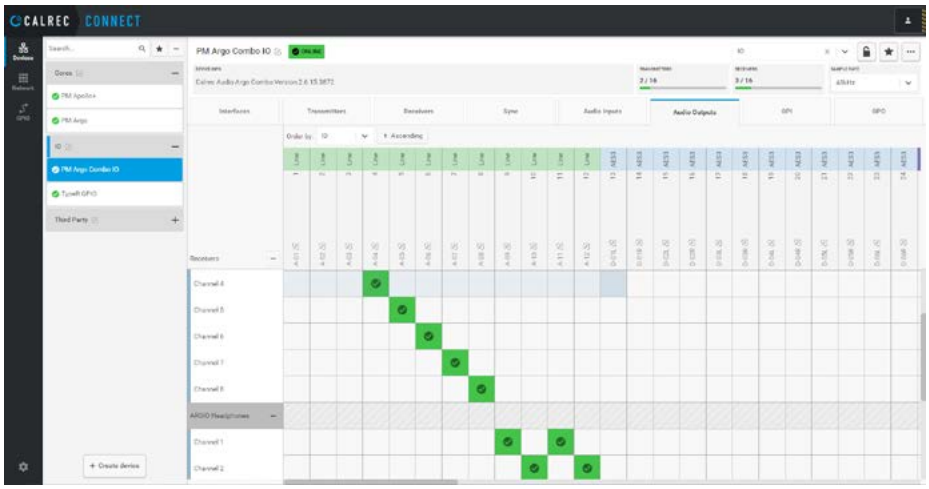
In addition to allowing the user to connect receiver channels to the audio output ports, the user can give each output port a label.

It also allows the user to Lock out the port allocation to prevent any parameter changes from being made.

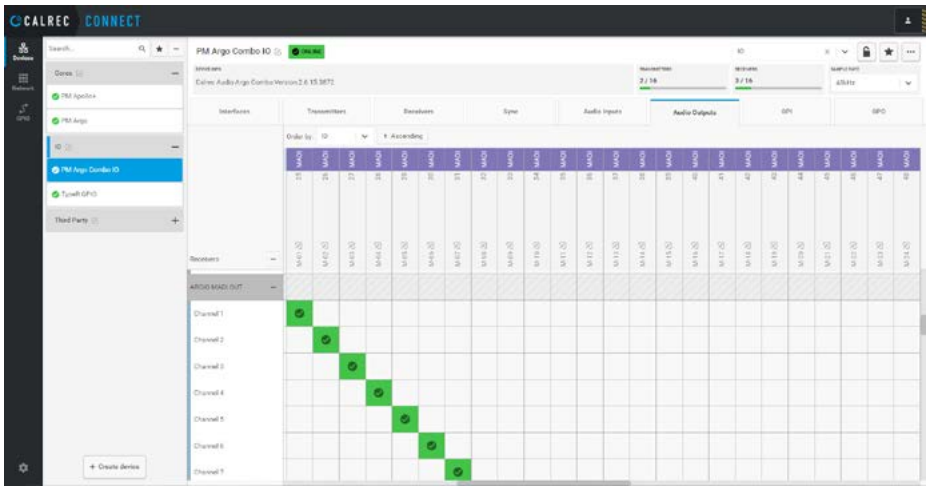
The image below right shows the connection of receiver streams on the left to the top where there is shown the MADI Audio outputs which can be folded open + or closed- to make the streams easier to manage.

In this image the first receiver stream '**ARGIO MADI OUT**' is expanded to show the stream channels which are typically connected in the one to one diagonal pattern shown, however the MADI receiver could be replaced with a number of smaller receiver streams as required to arrange the outputs into logical groups if necessary.

DEVICES>AOIP DEVICE RECEIVER CHANNELS TO MIC/AES AUDIO OUTPUTS



DEVICES>AOIP DEVICE RECEIVER CHANNELS TO MADI AUDIO OUTPUTS



Label Selection (Audio I/P & O/P)
The user clicks on the existing label edit symbol and the label entry box appears. The user edits the label as required and clicks on the '✓' symbol.

Lock Selection (Audio I/P & O/P)
The user clicks on the padlock symbol this will toggle the state of that function and will appear as a yellow padlock when active.

INPUT PORT PROTECTION

Calrec Impulse, Type-R and their various companion AoIP based I/O products provide a scheme to protect against changes to physical audio input port settings & patches to destinations that already have sources patched by other users.

Input Port Protection States
Audio input sources can be shared by all consoles connected on the same audio network. All operators can control a shared input using their own console's input controls, but no operator has direct control of the shared input's Mic Gain, phantom power (48v) or SRC. To help avoid unwanted or accidental changes, these critical input controls can be placed in 1 of 3 states which are set using the 'Connect' application:-

Unlocked:- When an Input port is unlocked it's settings can be changed by any network user and operates on the basis of the last control change received.

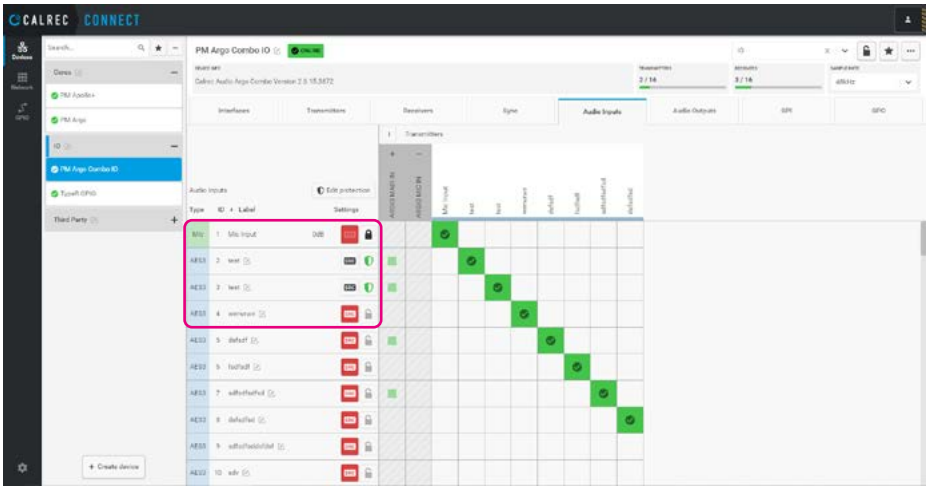
Protected:- When an Input port is protected, users can change its settings, but they have to make this a conscious operation, they are made aware that the port is marked as protected and have to make an extra step to change settings.

Locked:- When an Input port is locked, its settings (Input gain & phantom power for Mic/Line inputs, SRC for AES3 inputs) cannot be changed by any network user.

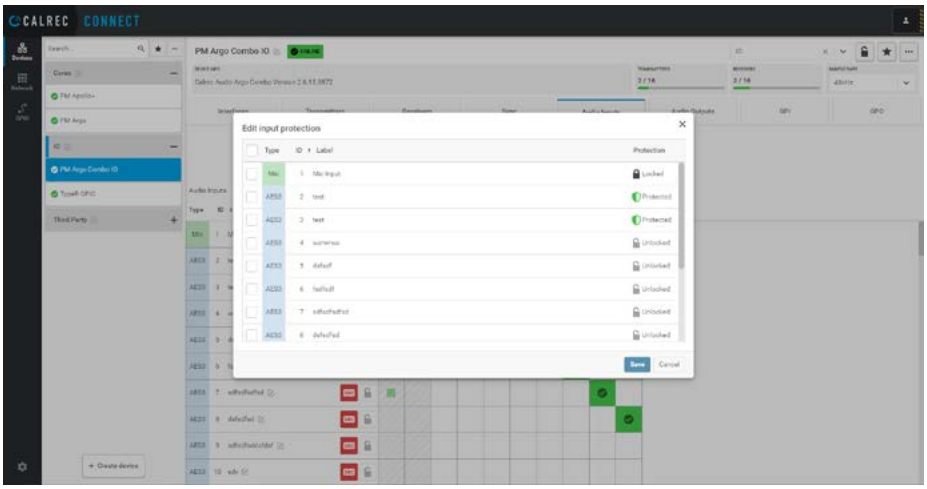
The image above right shows these three highlighted states as set in the 'Connect' application. AES 4 is Unlocked as shown by the open padlock, AES 3 & AES 2 are Protected as shown by the shield and Mic 1 is Locked as shown by the closed padlock.

Note: tapping on the 'Edit Protection' button opens the 'Edit input protection' dialogue box which allows the user to quickly select some or all of the ports using the tickboxes in the left column and then apply the Lock, Unlock or Protect state, to that selection as shown below right.

SETTING INPUT PORT PROTECTION IN CONNECT



EDITING INPUT PORT PROTECTION IN CONNECT



GPI & GPO LABELLING AND GPO MODES OF OPERATION

Remote GPI configuration
This page is accessed from the Devices menu and appears as shown above right when the user selects a Core and clicks on GPI.

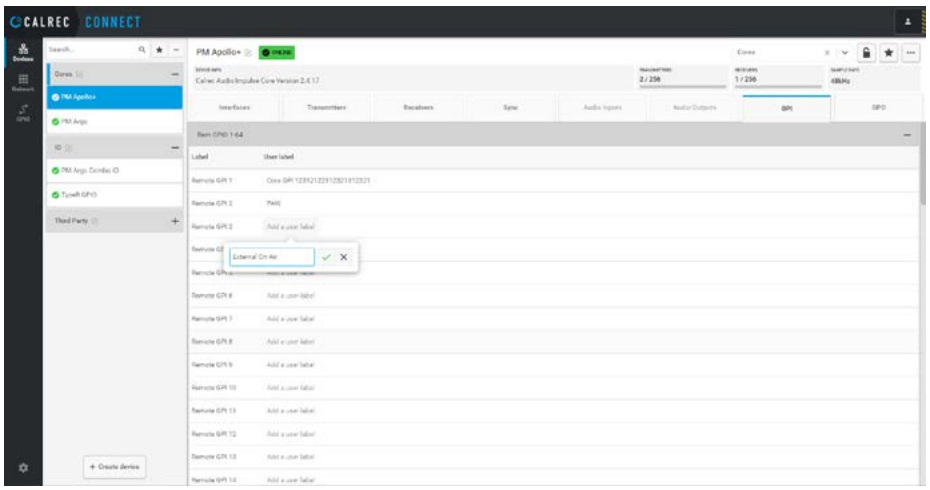
This allows the user to access each of the 256 Remote GPIs in the selected core and edit their user labels which also appear in the Console PC and the various Connect pages associated with GPI functionality.

The 256 Remote GPIs shown in the '**PM Apollo+**' core are arranged in 4 sets of 64 GPIs labelled:-

Rem GPIO 1-64,
Rem GPIO 65-128,
Rem GPIO 129-192 &
Rem GPIO 193-256.

These Remote GPIO pages allows the user to open (expand) or close (collapse) the 4 sets as required. Clicking on Remote GPI entries in the User Label column allows the user to enter their own label.

DEVICES>GPI>AOIP CORE>REMOTE GPI LABELS



Remote GPO configuration
This page is accessed from the Devices menu and appears as shown below right when the user selects a Core and clicks on GPO.

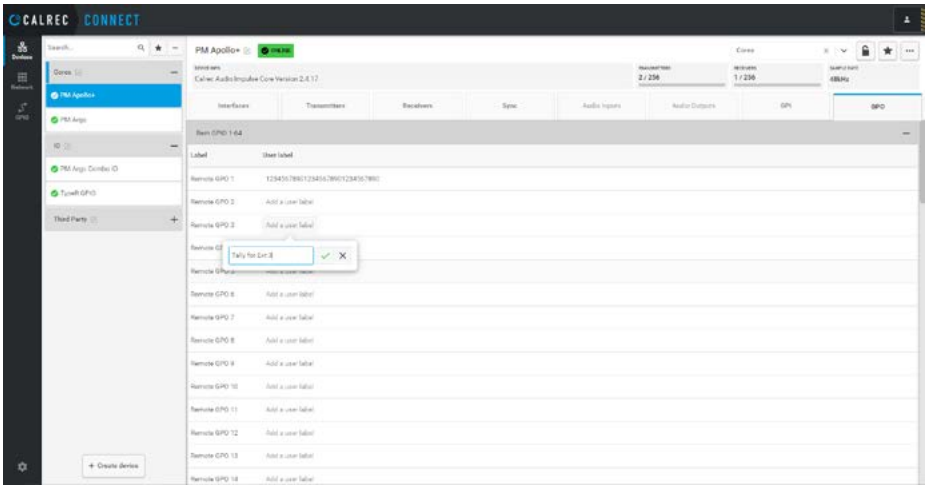
This allows the user to access each of the 256 Remote GPOs in the selected core and edit their user labels which also appear in the Console PC and the various Connect pages associated with GPO functionality.

The 256 Remote GPOs shown in the '**PM Apollo+**' core are arranged in 4 sets of 16 GPOs labelled:-

Rem GPIO 1-64,
Rem GPIO 65-128,
Rem GPIO 129-192 &
Rem GPIO 193-256.

These Remote GPIO pages allows the user to open (expand) or close (collapse) the 4 sets as required. Clicking on Remote GPO entries in the User Label column allows the user to enter their own label.

DEVICES>GPO>AOIP CORE>REMOTE GPO LABELS



GPI configuration

This page is accessed from the Devices menu and appears as shown above right when the user selects an AoIP device and clicks on GPI.

This allows the user to access each of the GPIs on that device and edit their user labels which appear on the various Connect pages associated with GPI functionality.

The AoIP device shown above right is a Type R GPIO device which has 32 GPIs labelled:- GPI 1 through to GPI 32.

Clicking on GPI entries in the User Label column allows the user to enter their own label.

In addition the user can click on the invert Icon at the end of each row and change the sense of the trigger from a normal state, where the state is true when active i.e. ON, to an inverted state where the state is false when active, i.e. OFF.

GPO configuration

This page is accessed from the Devices menu and appears as shown below right when the user selects an AoIP device and clicks on GPO.

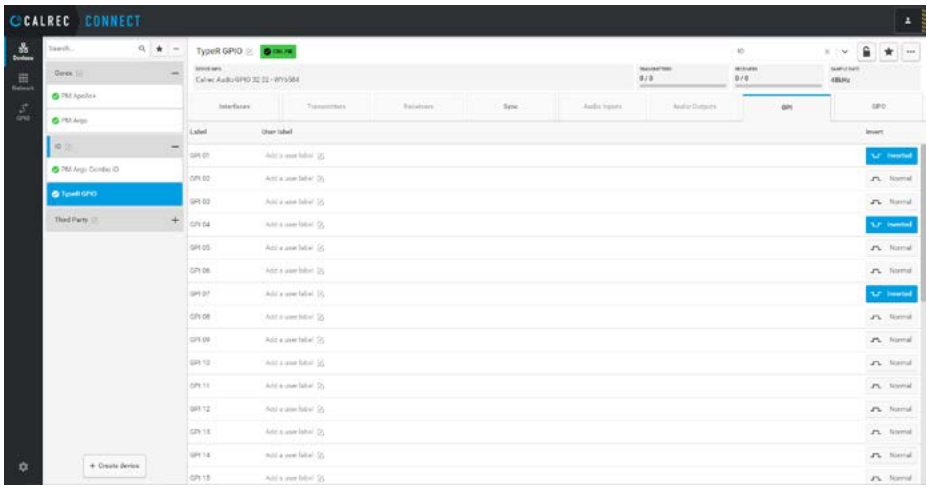
This allows the user to access each of the GPOs on that device and edit their user labels which appear on the various Connect pages associated with GPO functionality.

The AoIP device shown below right is a Type R GPIO device which has 32 GPOs labelled:- GPO 1 through to GPO 32.

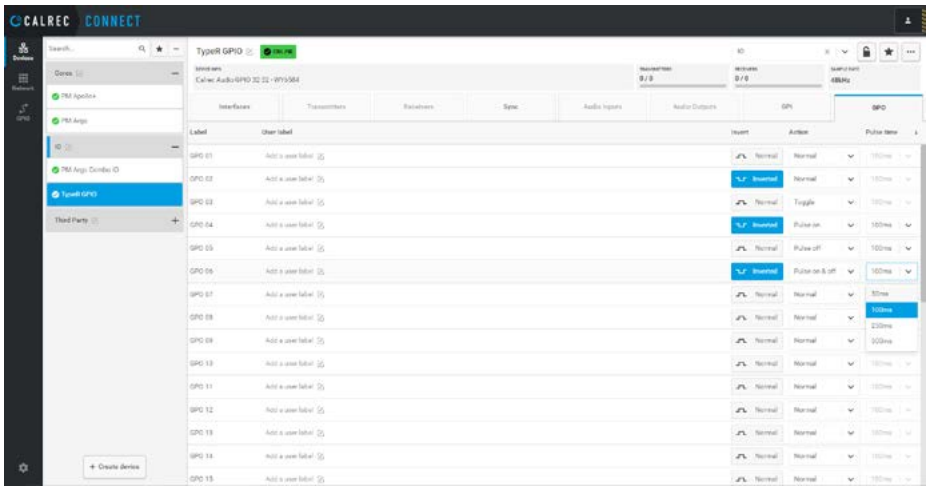
Clicking on GPO entries in the User Label column allows the user to enter their own label.

In addition the user can click on the invert Icon at the end of each row and change the sense of the trigger from a normal state, where the state is true when active i.e. ON, to an inverted state where the state is false when active, i.e. OFF.

DEVICES>GPI>AOIP DEVICE>GPI LABELS & INVERTS



DEVICES>GPO>AOIP DEVICE>REMOTE GPO LABELS, INVERTS & MODES



GPO Modes

The image below right also shows the available modes on the GPO page.
The user clicks on each GPO in the action column and a dropdown appears with the following modes:-

'Normal' - GPO held closed whilst console function is active.

'Toggle' - useful if controlled by a user button where a press to activate and a subsequent press to deactivate is required.

'Pulse On' - when function activates,
'Pulse Off' - when function deactivates.
'Pulse On & Off' - where a pulse is sent on both activation and deactivation of a function.

Pulse Time

When the GPO Action is set to one of the three Pulse modes, tapping the entry in the Pulse time column opens another dropdown offering four different pulse times to select from:- 50ms, 100ms, 250ms & 500ms, this sets the required duration of the pulse(s).

NETWORK

Network configuration
This page is accessed from the Network menu and appears as shown above right. This allows the user to connect audio streams between Cores and AoIP devices across a Network. These streams are set up in the Devices section allowing audio inputs from AoIP devices to be placed on transmitter streams to be connected to receivers on the core for processing or to be passed on to other AoIP devices.

Similarly audio outputs from the core can be placed on transmitter streams to be connected to receivers on AoIP devices.

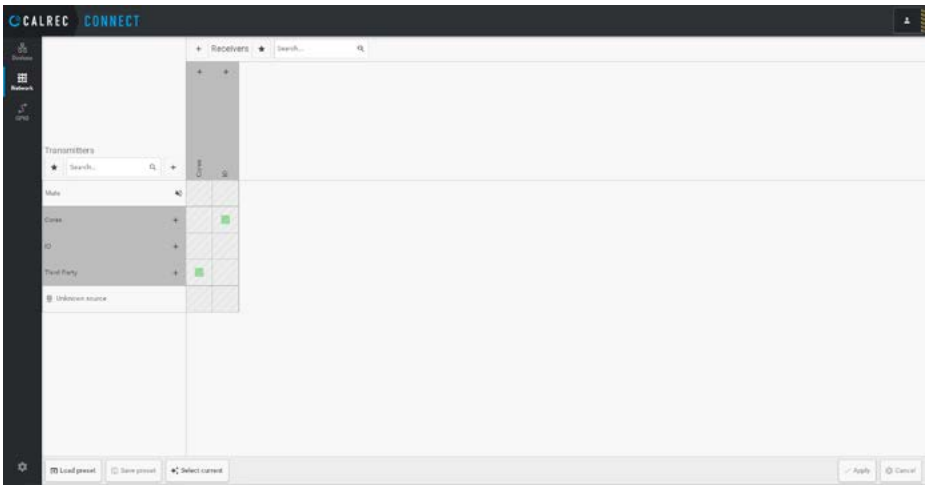
AoIP Stream Interfaces
The Network page is arranged as an X-Y matrix with the transmitters in rows and the receivers in columns.
This page allows the user to collapse or expand Groups such as '**Cores**' & '**IO**', or Devices such as '**PM Apollo+**' and '**PM Argo Combo IO**' so that the matrix is easier to navigate, see above right & middle right.

There is also a Collapse All/Expand All option for both transmitters and receivers. At the top of the transmitter rows is an extra row called 'Mute', this is where a crosspoint is placed when no connection is required and it applies a muted audio signal to that stream. The matrix shows where valid connections can be made in white. The streams are shown as members of Devices placed in the Groups.

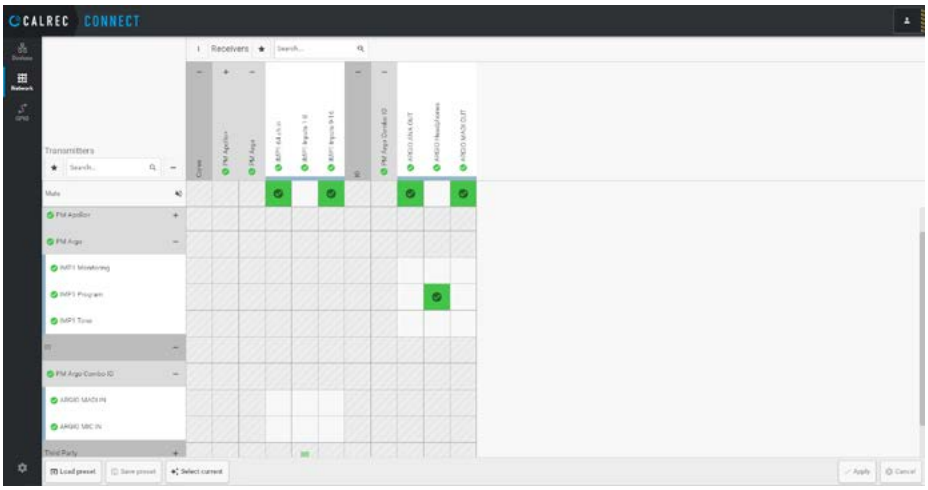
For example the transmitter stream labelled 'IMP1 Program' from the core in Control room is an 8-wide output stream which is connected to a receiver labelled 'ARGO Headphones' providing 4 pairs of headphone feeds. Also note that the transmitter stream can be directly connected to a receiver stream in another AoIP device without passing through the core allowing the same source to be presented to a number of devices.

Similarly the transmitter stream labelled 'ARGIO MADI IN' (shown in the image above right) from 'PM Argo Combo IO' is a 64-wide input stream which is connected to a receiver labelled 'Imp1 64 ch in' in the 'PM Argo' core.

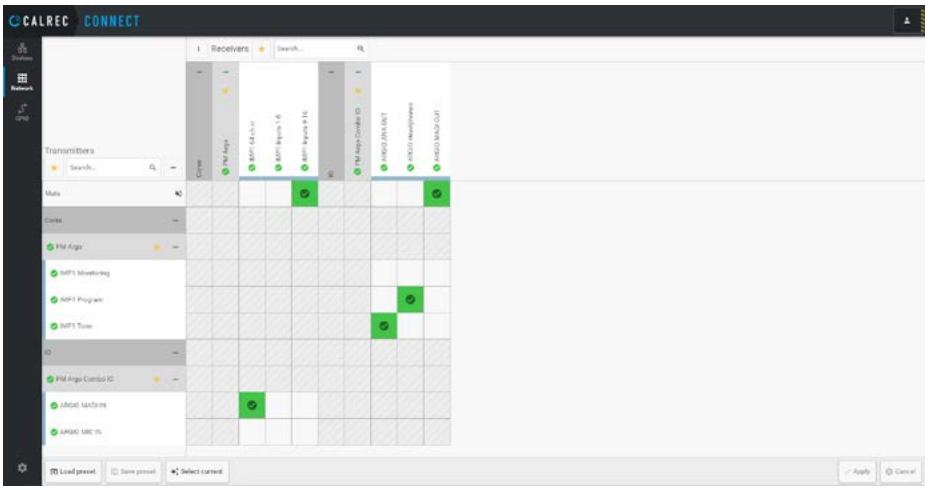
NETWORK>CONNECTED AOIP STREAM INTERFACES DISPLAY COLLAPSED



NETWORK>CONNECTED AOIP STREAM INTERFACES DISPLAY EXPANDED



NETWORK>CONNECTED AOIP STREAMS>CORE TO AOIP DEVICES-FILTERED



This has been setup to allow 64 tracks from an external MADI device to be connected to 64 channels in the core as shown above right.

Each crosspoint selection to be made appears as a red open square as shown. When all the required selections are made the user then clicks on the 'Apply' button on the bottom right of the footer at which point the pending connections become green closed squares. The selected crosspoint is being used to 'Mute' the feed to the ARGIO headphones.

Favourite selection on Network Page
In the same way as the Devices and GPIO pages, the Network pages have the ability to filter connect rows and columns based on 'Favourite' selections which were made from the Devices pages. These are applied by clicking on the 'Star' symbol at the head of the Transmitter and Receiver areas as shown above right.

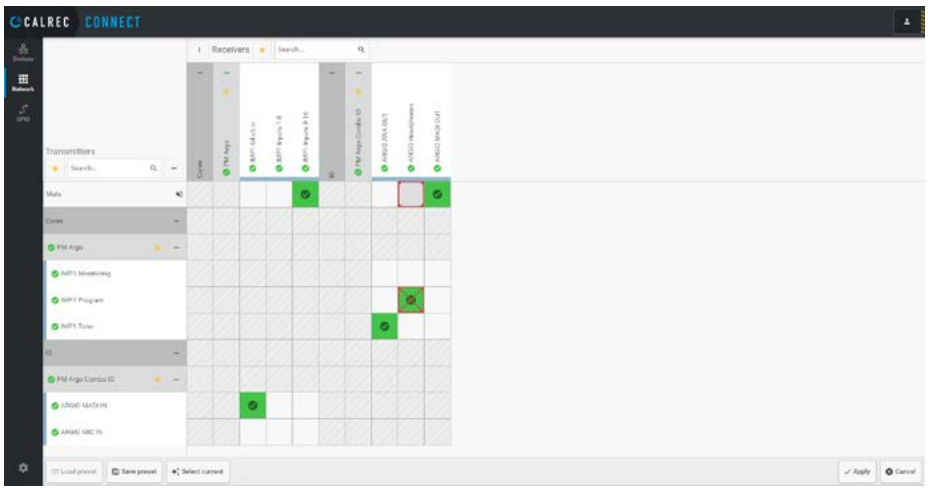
Select and Save Network Presets
In the left side of the footer are 3 buttons labelled:- Load preset, Save preset and Select current. Pressing the 'Select current' button selects all the current crosspoints marking them with a Red border for inclusion in saving a preset.

Alternatively the user can click on individual crosspoints they want to save, see middle right. The user then clicks on the 'Save preset' button and a dialogue box appears asking the user to enter a name for the preset. On pressing 'Save' the connections are saved into a file shown in the footer along with a 'Preset exported' message as shown as an inset middle right.

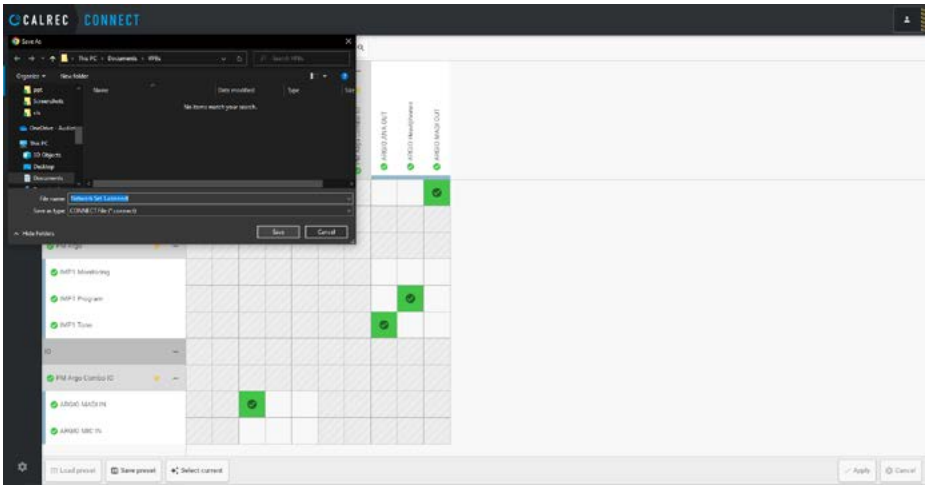
Load Network Presets
Selecting the 'Load preset' button opens the dialogue box shown below right. This asks the user to browse for the Preset file.

Once the file is chosen, it is uploaded replacing the current configuration of selected routes and a 'Success' message appears as shown as an inset below right.

NETWORK>CONNECTED AOIP STREAMS>AOIP DEVICES TO CORE-FILTERED

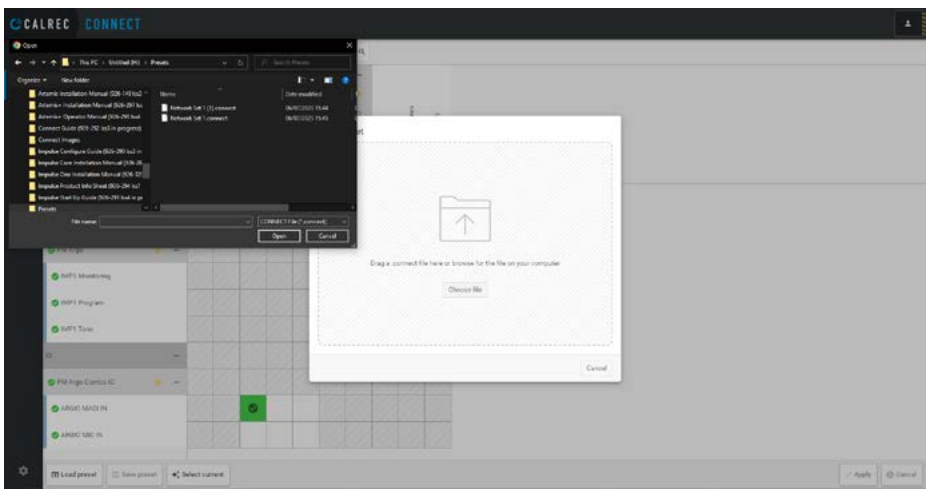


NETWORK>SELECT AND SAVE PRESETS



<-SAVE PRESET CONFIRMATION AND FILE ATTACHMENT

NETWORK>LOAD PRESETS



<-LOAD PRESET CONFIRMATION

CONNECT GPI TO REMOTE GPI & REMOTE GPO TO GPO

GPIO configuration
This page is accessed from the GPIO menu and allows the user to access the GPI and GPO of AoIP devices, by connecting them to Remote GPI and Remote GPO connections in the Core.

Note: this example is a Type-R core which has 64 Remote GPIs / GPOs in 4 sets of 16 rather than the 256 Remote GPIs / GPOs in 4 sets of 64 in an Impulse Core.

GPI connection to Remote GPI
The pages shown above right and below right are arranged as an X-Y matrix with the Sources in rows and the Destinations in columns.

It allows the user to open (expand) or close (collapse) Groups such as 'Control room' & 'Studio', or Devices such as 'calrec-system' & 'Combo 1', so that the matrix is easier to navigate.

There is also a Collapse All / Expand All option for both Sources and Destinations.

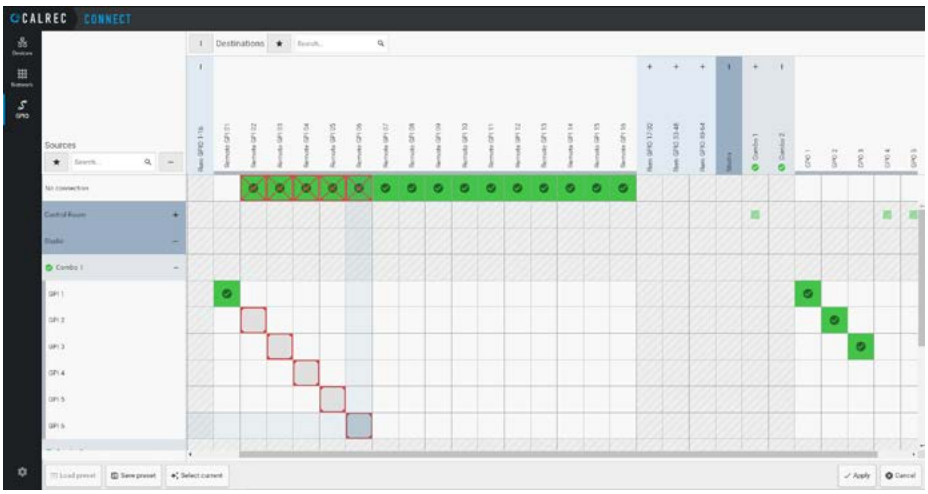
At the top of the Source rows is an extra row called 'No Connection', this is where a crosspoint is placed when no connection is required.

In order to connect a GPI of an AoIP device to a core function, the user clicks on the required crosspoint of a GPI with a Remote GPI on the core which has a function patched to it.

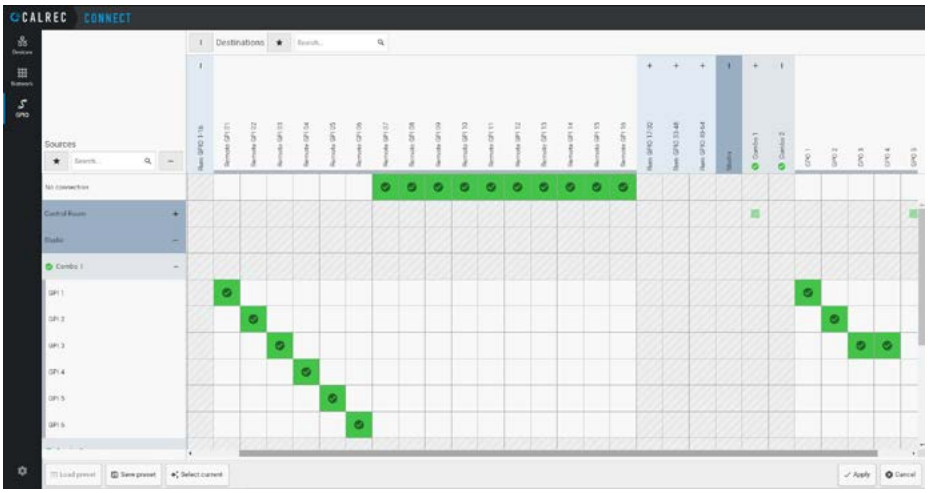
Each crosspoint selection to be made appears as a Red open square as shown above right.

When all the selections are made, the user clicks on the 'Apply' button in the bottom right of the footer at which point the connections become green closed squares, as shown middle right.

GPIO>GPI IN AOIP DEVICE READY TO CONNECT TO REM GPI IN CORE



GPIO>GPI IN AOIP DEVICE CONNECTED TO REM GPI IN CORE



The selected crosspoints in this case are the 6 GPIs in Combo 1, which have been connected to 6 Remote GPIs belonging to the core.

Also note that GPIs 1, 2 & 3 are also directly connected to GPOs in Combo 2 .

Two examples of Remote GPI connection are shown on the next page:-

The first shows GPI connections from an AoIP Combo unit being used to trigger general functions via remote GPI in the Impulse core.

The second shows GPI connections from an AoIP Combo unit being used to trigger Fader Cut functions via remote GPI in the Type R core.

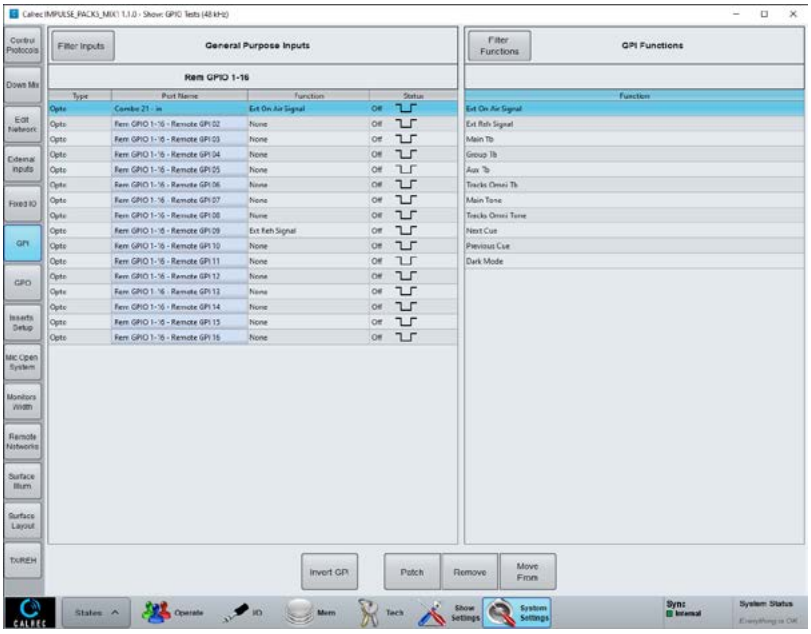
Remote GPI connection to Function

Once the AoIP device GPIs have been connected to the Remote GPIs in the core, the user can connect those GPIs via the Remote GPIs to functions in the system using the Console PC Application for an Apollo+ or Artemis+ Impulse based system.

The image above right shows that the GPI labelled '**Combo 21-in**' has been patched to the 'External On-Air Signal' via Remote GPI 01 whose label is now hidden.

So when GPI 1 on Combo 21 is triggered, then the hidden Remote GPI 01 changes the state of the External On-Air Signal.

CONSOLE PC>SYSTEM SETTINGS>GPI>REMOTE GPI TO FUNCTION



Connect Remote GPI to Fader Cut

Once the AoIP device GPIs have been connected to the Remote GPIs in the core, the user can connect those GPIs via the Remote GPIs to functions in the system using the Assist application for a Type R based system.

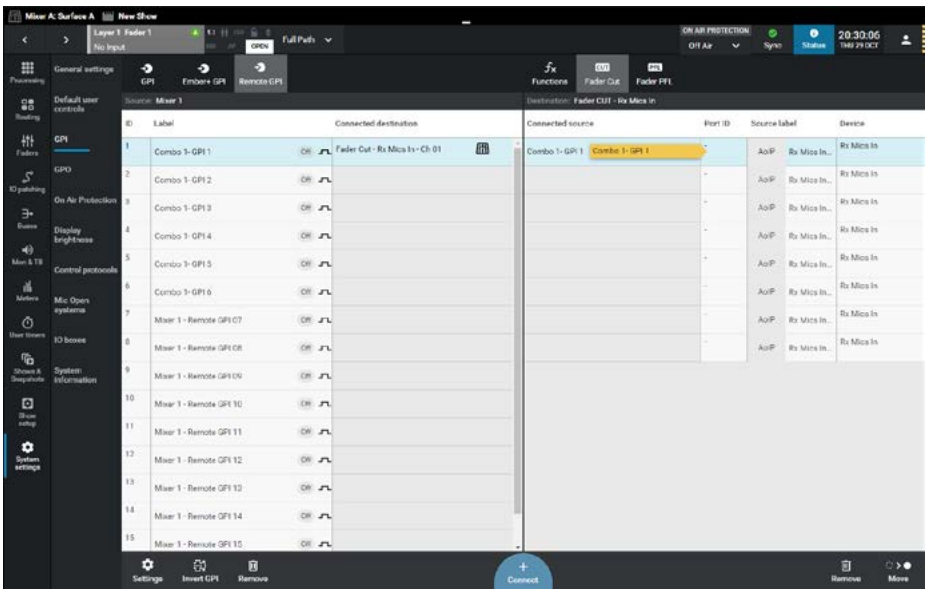
In order to externally control a fader's cut state, the user selects a **'Remote GPI'** from the source list on the left side of the screen, the '**Fader Cut**' function button from the function header, then the required audio receiver channel on the right side of the screen and clicks on connect as shown below right.

The Remote GPI, when activated from its connected GPI in the AoIP Combo unit, switches that Fader's Cut on/off as required.

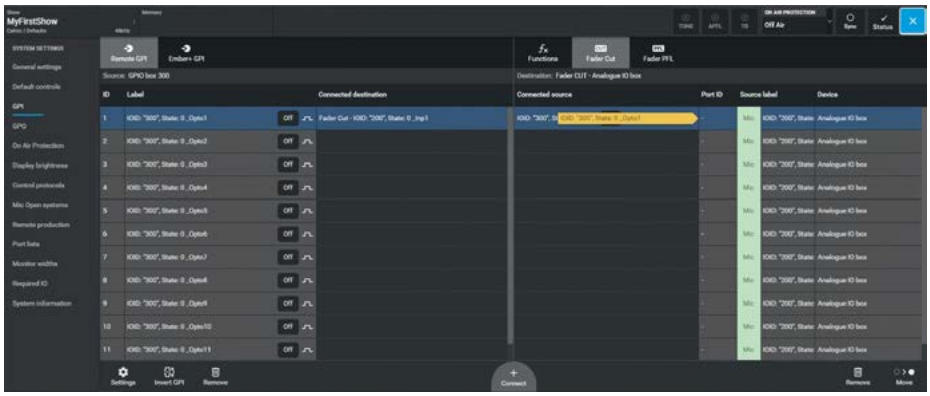
'Fader Cut' functions are audio receiver specific, for example, if a user connects a GPI to an audio receiver 'Fader Cut', that GPI will stay connected to that audio receiver's Fader Cut even if the audio receiver is moved to a different fader.

The image below, shows the same action of connecting a remote GPI to a Fader Cut on the Argo Touchscreen in **System Settings>GPI**.

ASSIST>SYSTEM SETTINGS>GPI>CONNECT REMOTE GPI SOURCE TO FADER CUT



ARGO>SYSTEM SETTINGS>GPI>CONNECT REMOTE GPI SOURCE TO FADER CUT



Favourites selection on GPIO Pages
In the same way as the Devices and Network pages, the GPIO pages in Connect have the ability to filter connect rows and columns based on 'Favourite' selections, which were created in the Devices page.

These are applied by clicking on the 'Star' symbol in the headers of the Sources and Destinations areas.

Remote GPO connection to GPO
In order to connect a core function to a GPO of an AoIP device, the user clicks on the required crosspoint from a Remote GPO on the core which has the function patched to it, to the GPO in the AoIP device.

Each crosspoint selection to be made appears as a red open square, as shown above right.

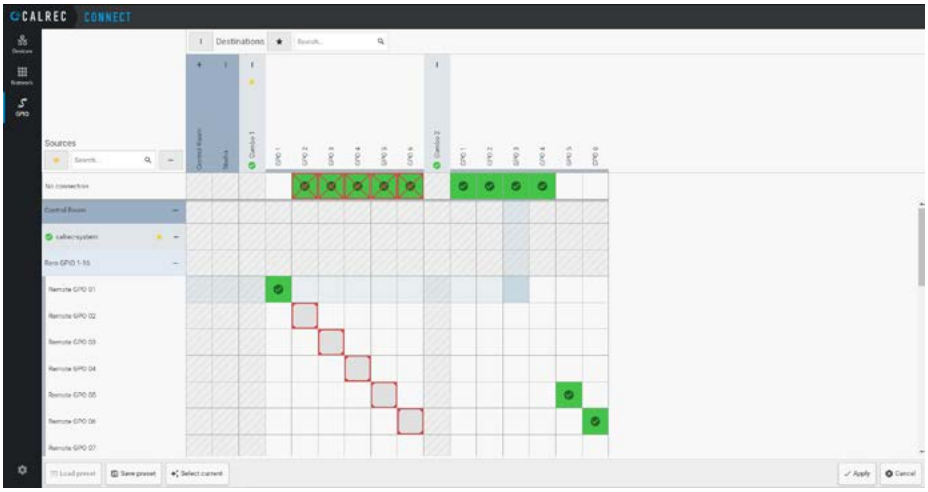
When all the selections are made the user then clicks on the 'Apply' button in the bottom right of the footer at which point the connections become green closed squares, as shown below right.

The selected crosspoints in this case are the 6 Remote GPOs belonging to the core which have been connected to the 6 GPOs in Combo 1.

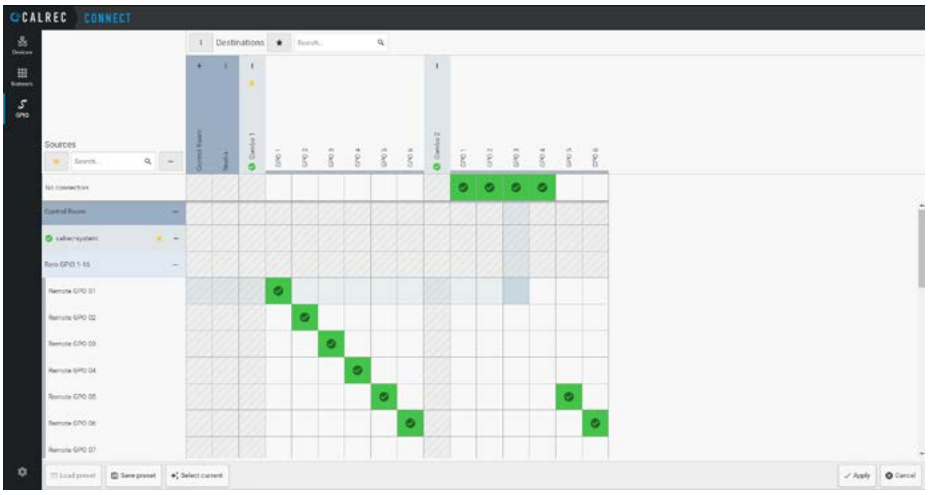
Note: Remote GPOs 5 & 6 are also connected to GPOs in Combo 2 showing that signals can be sent to multiple AoIP devices.

Remote GPO to Remote GPI connection between Impulse cores
When there are 2 or more core systems connected together, it is possible to connect Remote GPOs from one core system directly to the Remote GPIs of another core system to provide virtual signalling between them.

GPIO> REM GPO IN CORE READY TO CONNECT TO GPO IN AOIP DEVICE



GPIO> REM GPO IN CORE CONNECTED TO GPO IN AOIP DEVICE



Two examples of Remote GPO connection are shown on the next page:-

The first shows an 'On-Air' general function tally connected to a GPO connection in an AoIP Combo unit via a remote GPO in the Impulse core.

The second shows a 'Fader On' function tally connected to a GPO connection in an AoIP Combo unit via a remote GPO in the Type R core.

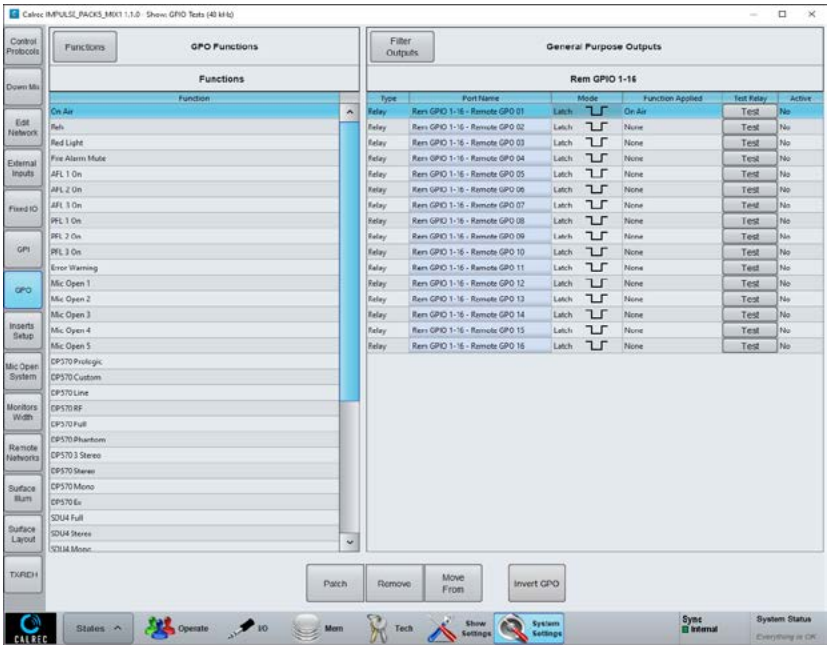
Function connection to Remote GPO

Once the Remote GPOs in the core have been connected to the AoIP device GPOs, the user can connect to those GPOs via the Remote GPOs from functions in the system using the Console PC Application for an Apollo+ or Artemis+ Impulse based system.

The image below right shows that the 'On-Air' function has been patched from the Remote GPO 01 in the core.

This means that when the On-Air function is triggered the Remote GPO 01 changes state which in turn changes the state of the unlabelled GPO 1 on Combo 1.

CONSOLE PC>SYSTEM SETTINGS>GPO>FUNCTION TO REMOTE GPO



Connect Fader On to Remote GPO

Once the Remote GPOs in the core have been connected to the AoIP device GPOs, the user can connect to those GPOs via the Remote GPOs from functions in the system using the Assist application for a Type R based system.

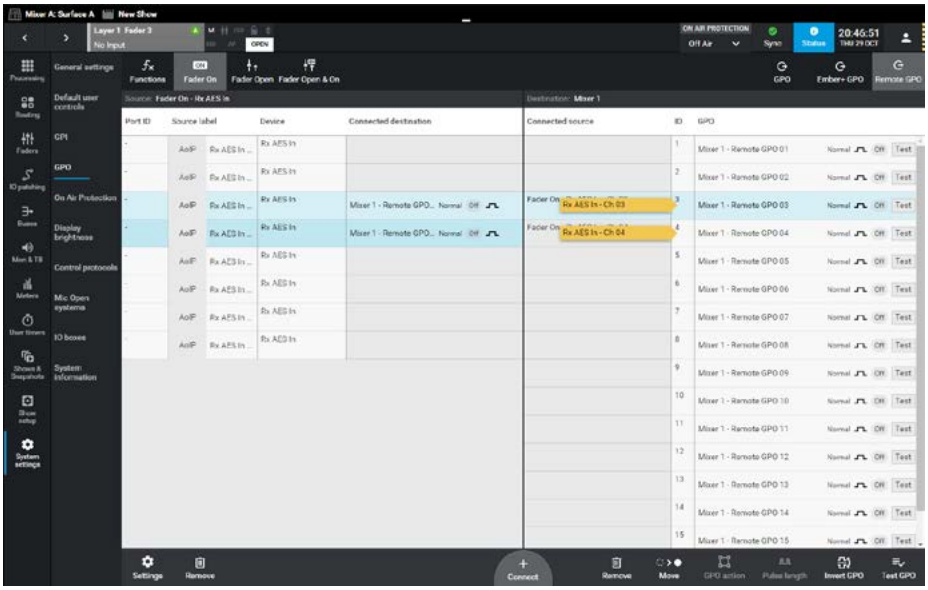
In order to control a **'Remote GPO'** from a **'Fader On'** state, the user selects a Remote GPO from the destination list on the right side of the screen, the required audio receiver channel on the left side of the screen and clicks on **'Connect'** as shown middle right.

The Fader On state, when activated, switches the GPO On/Off as required via the Remote GPO in the Type R core.

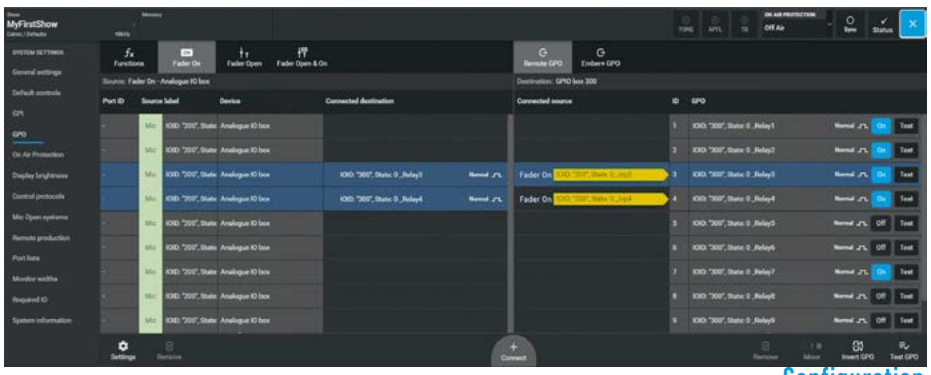
'Fader On' functions are audio receiver specific, for example, if the user connects a GPO to an audio receiver 'Fader On', that GPO will stay connected to that audio receiver's 'Fader On' even if the audio receiver is moved to a different fader.

The image below, shows the same action of connecting a Fader On function to a remote GPO on the Argo Touchscreen in **System Settings>GPO**.

ASSIST>SYSTEM SETTINGS>GPO>CONNECT FADER ON TO REMOTE GPO DEST'N



ARGO>SYSTEM SETTINGS>GPO>CONNECT FADER ON TO REMOTE GPO DEST'N



CONNECTING 3RD PARTY DEVICES

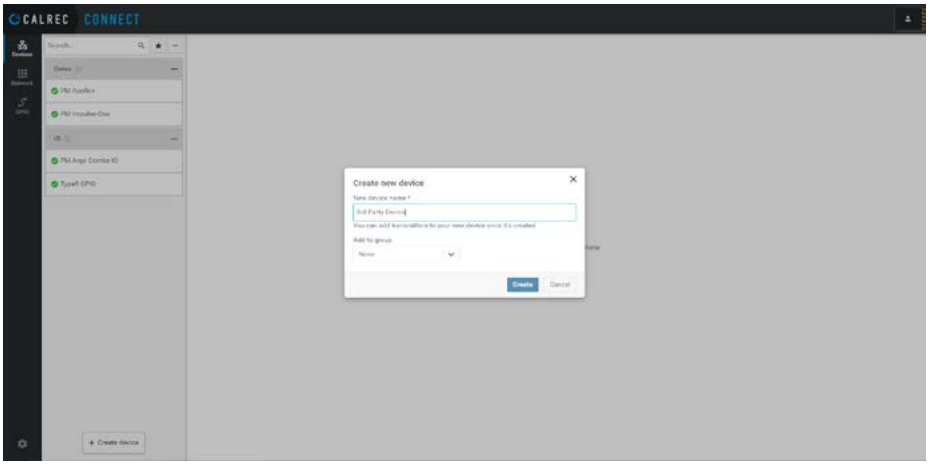
It is recommended that the console and AoIP network is set up to establish that the user can create Transmitters and Receivers in Connect and pass audio between a Calrec IO box and the Core before setting up the 3rd party devices. This will confirm the AoIP network is running and importantly that PTP clocking has been established.

Manual Creation - 3rd Party Devices
The Type R and Impulse Cores will automatically recognise Calrec AoIP devices in Connect however some 3rd party devices will not be recognised. NMOS IS-04 & IS-05 support will detect and manage connections with 3rd party devices, but there are many vendors that do not support NMOS. Connect will also support Ravenna and SAP for wider coverage, but we need to provide a method that will work with any AES67/ ST2110 stream regardless of the protocols supported. In order to work with these unrecognised 3rd party AoIP devices/streams, the devices have to be manually configured so that the 3rd party transmitters can be received by the core and that transmitters from the core are received by the 3rd party device receivers.

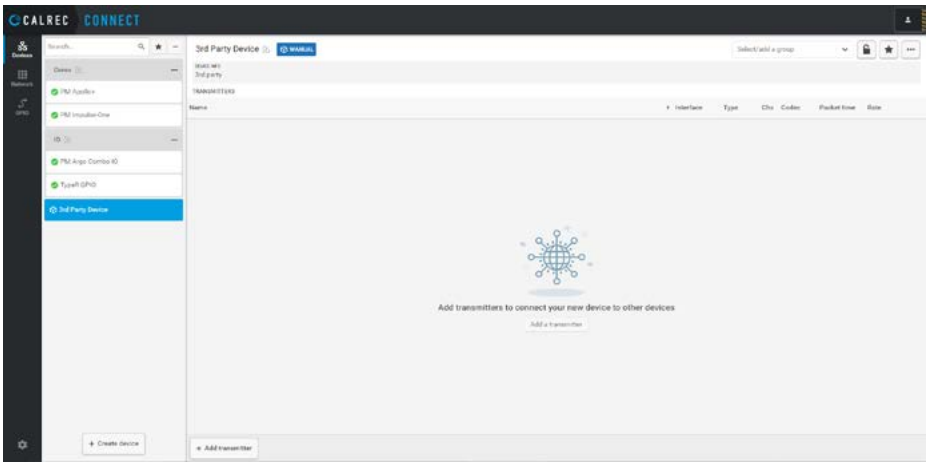
Create Devices
The process to manually create devices are shown here, the first step is to click on the 'Create device' button in the footer of the Connect>Devices page as shown above right. This opens the 'Create new device' dialogue box and the user enters the device name and clicks on the 'Create' button which adds the device to the devices list and when selected allows the user to add transmitters to connect the new device to other devices as shown right.

Create 3rd party Transmitters in Connect
Clicking on 'Add a transmitter' button for the first transmitter or the 'Add transmitter' button in the footer of the devices page opens the 'Create transmitter' dialogue box shown below right. The parameters of the 3rd party transmitter have to be manually copied from the 3rd party device using the Session Description Protocol.

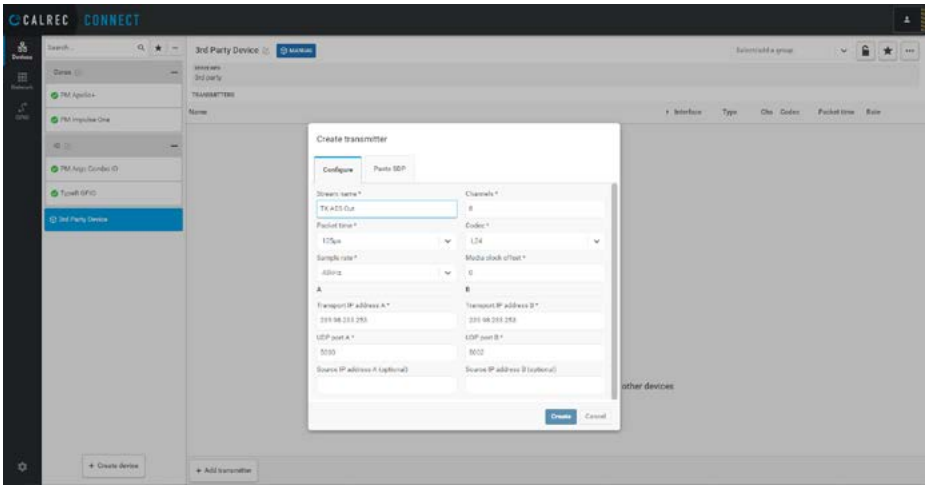
DEVICES>MANUAL CREATE DEVICE NAME



DEVICES>ADD TRANSMITTERS TO NEW DEVICE



DEVICES>CREATE TRANSMITTER PAGE



SDP Requirements
It is a mandatory requirement that all AES67 devices provide an SDP for each of their transmitter streams. The SDP contains all of the information about a transmitter required in order to be able to receive it.

SDPs are exchanged automatically between devices using advertisement protocols such as IS-04, Ravenna/mDNS & SAP, but they can also be manually copied and pasted between different vendors' UI to create connections between devices that do not use the same advertisement method as each other.

Copying and Pasting SDP data
The image shown above right is the Sources (Transmitters) page for a 'Bach' AoIP module. Transmitters can be created on this page and the SDP data is displayed for each transmitter, in this case Tx AES Out in the 'Advanced' section shown on the right of the page.

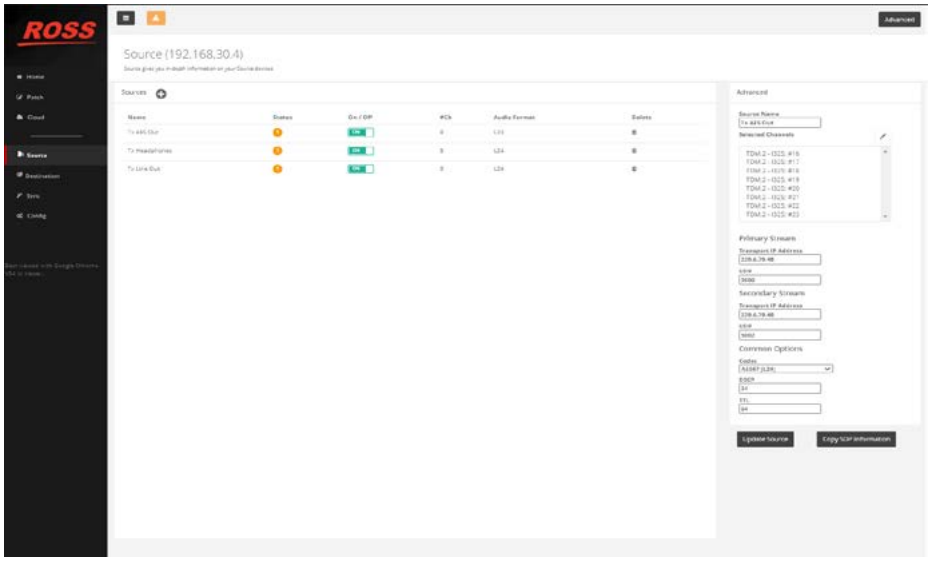
A button at the bottom of this section allows the user to 'Copy SDP Information'.

Clicking on this button, opens the SDP information box shown right. In order to transfer this information to connect the user highlights the SDP information and copies it with the usual CTRL C key.

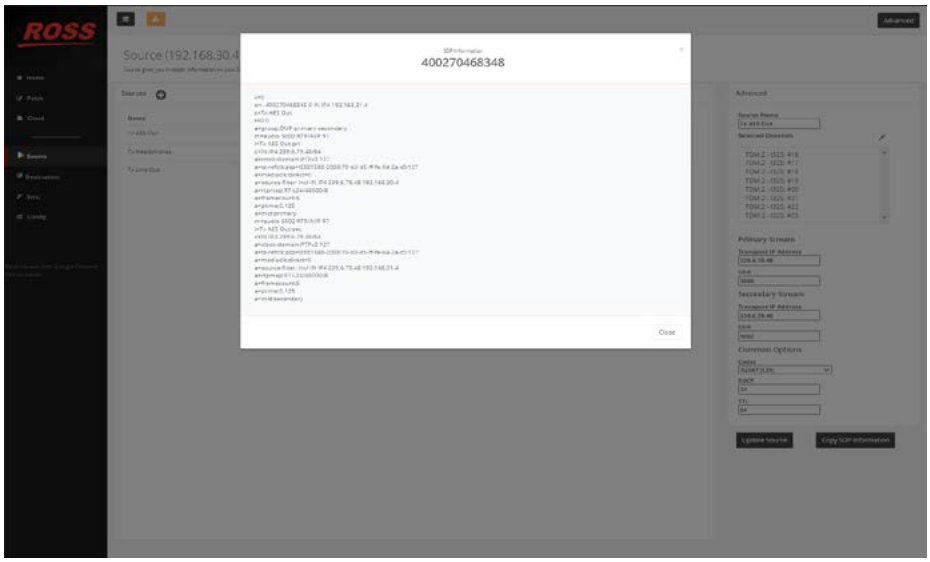
The user then returns to the Connect Devices page and either selects the transmitter to edit its parameters or in this case, creates a new transmitter which opens the 'Create transmitter' dialogue box as described earlier. In this dialogue box the user selects the 'Paste SDP' tab.

The SDP data copied from the 3rd Party transmitter is then pasted into the SDP screen area with the usual CTRL V key as shown below right. Clicking on the 'Create' button will create a transmitter based on the SDP information that was pasted.

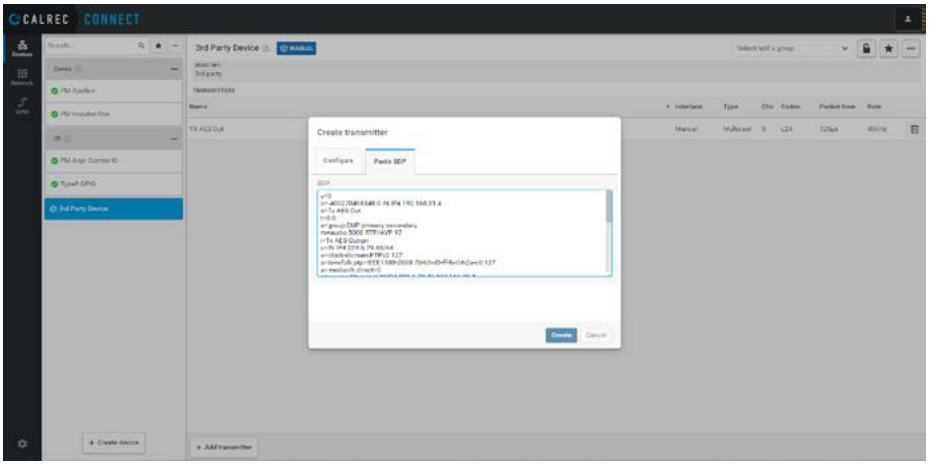
3RD PARTY>TRANSMITTER PAGE SHOWS SDP INFORMATION



3RD PARTY>SDP INFORMATION CREATED AND COPIED



DEVICES>CREATE 3RD PARTY TRANSMITTER SDP PASTED

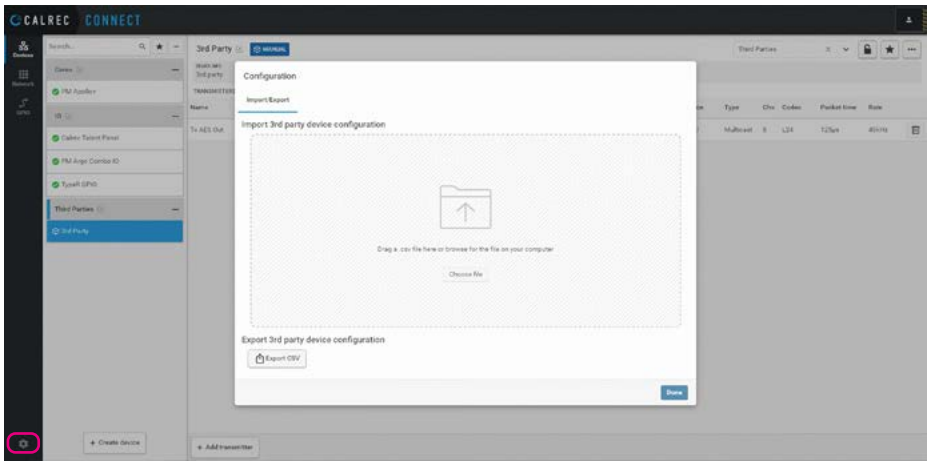


Exporting 3rd Party Devices via CSV
Importing and Exporting 3rd Party Device configurations save a lot of time instead of creating SDP files or editing the stream parameters directly from 3rd party devices repeatedly.

Once the 3rd party transmitters and receivers have been created, the user can export the 3rd party configuration file to a PC using a CSV format.

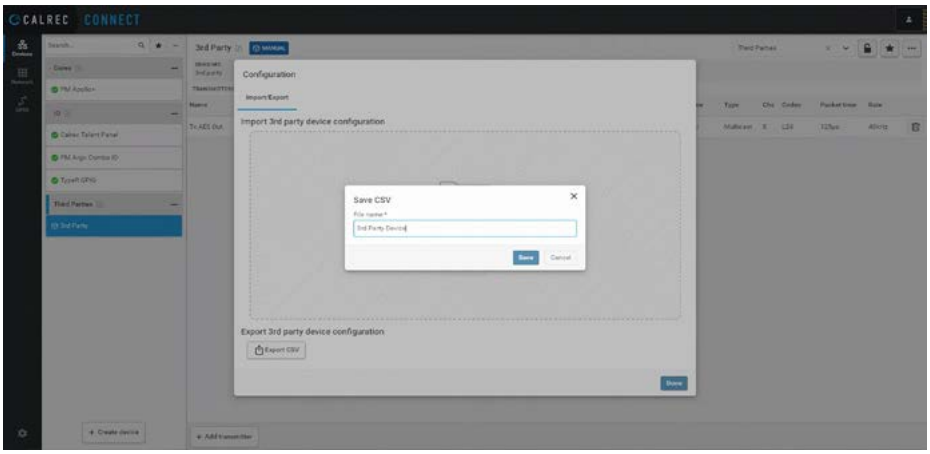
The image shown above right is accessed by tapping on the configuration gearwheel highlighted. In order to export the CSV file the user taps on the Export 3rd party device configuration button labelled 'Export CSV'.

DEVICES>SELECT EXPORT 3RD PARTY DEVICE CONFIGURATION



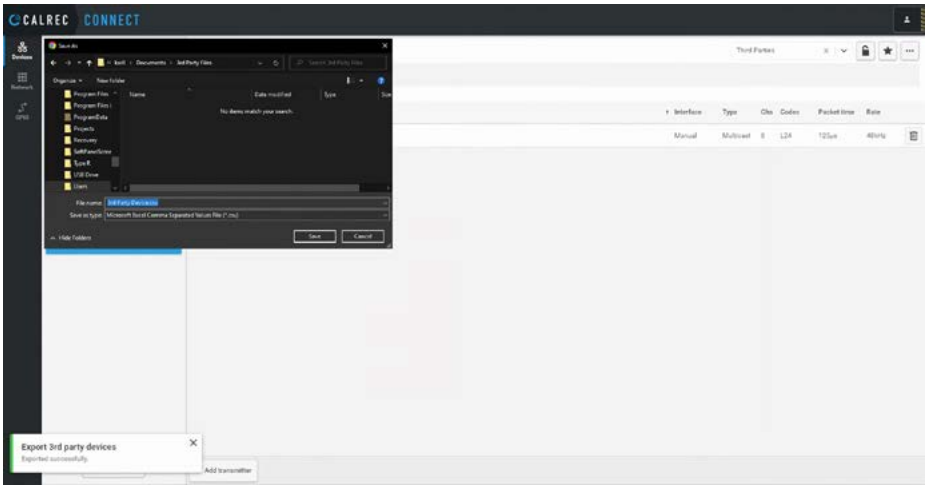
This opens the Save CSV dialogue shown in the image middle right and after creating a File name for the CSV file the user taps on 'Save'.

DEVICES>NAME & SAVE EXPORT 3RD PARTY DEVICE CONFIGURATION



This opens an Explorer window on the PC allowing the user to save the CSV file to an appropriate directory as shown below right and once saved a message appears briefly to show that the 3rd party device has been successfully exported. See bottom left of page.

DEVICES>SAVE 3RD PARTY DEVICE CSV FILE TO COMPUTER

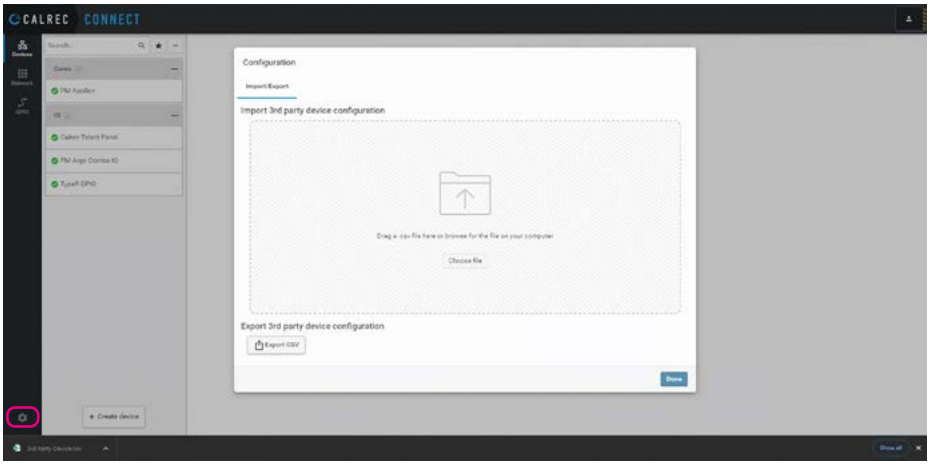


Importing 3rd Party Devices via CSV
Importing and Exporting 3rd Party Device configurations save a lot of time instead of creating SDP files or editing the stream parameters directly from 3rd party devices repeatedly. .

The image shown above right is accessed by tapping on the configuration gearwheel highlighted. The left column area shows that there are currently no 3rd party devices shown in the devices list.

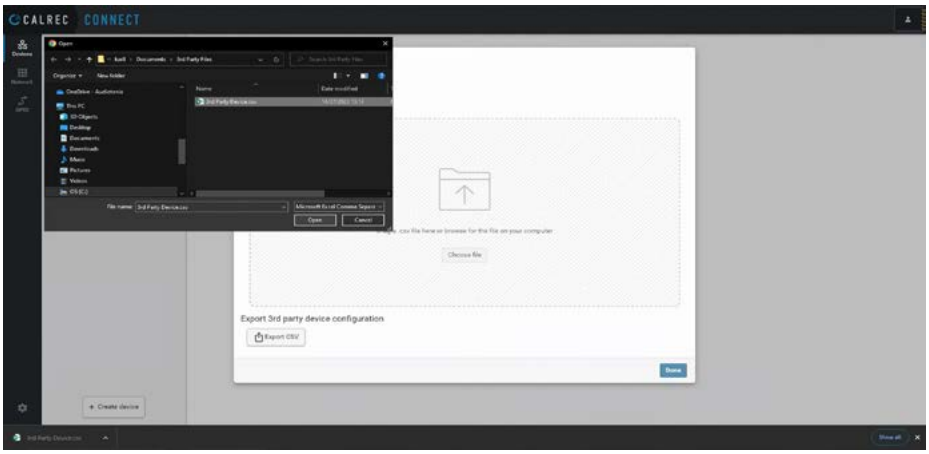
In order to import the CSV file the user can either drag a CSV file or in this case taps on 'Choose file'.

DEVICES>CHOOSE FILE FOR IMPORTING 3RD PARTY DEVICE CONFIGURATION



This opens an Explorer window on the PC allowing the user to select a CSV file from an appropriate directory as shown in the image middle right and once selected the user taps on 'Open'

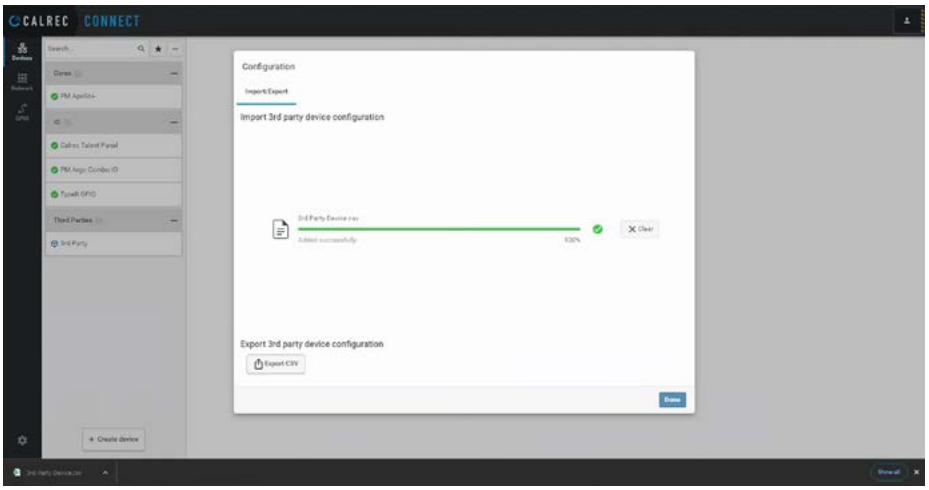
NETWORK>OPEN 3RD PARTY DEVICE CSV CONFIGURATION FILE



A progress bar appears showing that a 3rd party device CSV configuration file named '3rd Party device.csv' has been added to the devices list as shown in the devices column on the Devices screen.

Note: when the CSV file was created it appears in the bottom left of the footer, which can then be accessed for later recall when required. The footer can be cleared of this message by clicking on the X at the bottom right of the footer.

DEVICES>CONFIRM 3RD PARTY CSV CONFIGURATION FILE ADDED



Editing Stream parameters directly
Often engineers are given a list of stream parameters to configure rather than SDP files so, in addition to pasting and editing raw SDP data, transmitters can be manually configured by inputting the stream parameters into separate fields with meaningful labels, as shown above right.

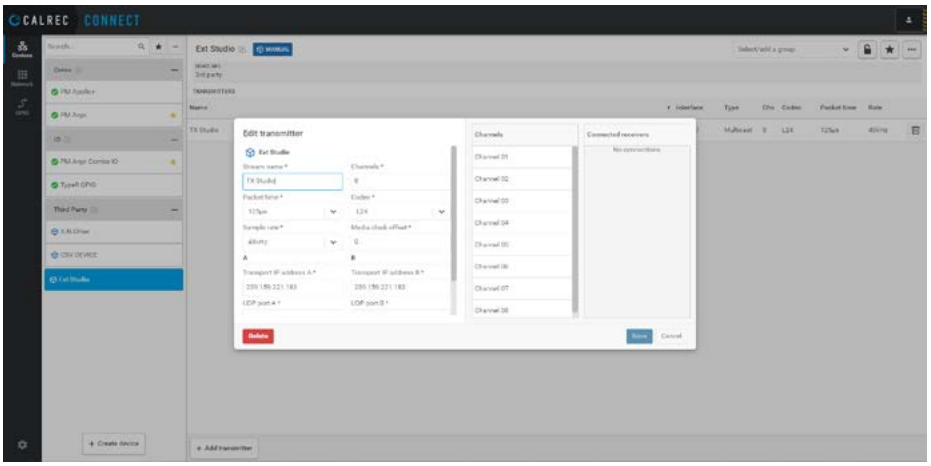
These fields are:-
Name: User friendly name for transmitter.
Channels: Quantity of audio channels, 1-80 depends on packet time, sample rate & codec (packet size<MTU). Default=8.
Packet Time: Drop-down offering 125us, 250us, 333.3us, 1ms, 4ms. Default=125us.
Codec: Drop-down offering L16, L24. Default=L24
Sample Rate: 44.1kHz, 48kHz, 88.2kHz, 96kHz. Default 48kHz
Media Clock Offset: Default=0
Transport IP Address A: IP address, check its entered in the correct format but don't error check range if already used.
Transport IP Address B: As above but can be blank if no redundant stream.
UDP Port A: Numeric input, use Ravenna default for primary.
UDP Port B: Numeric input, use Ravenna default for secondary, can be blank if no redundant stream.

Connect from 3rd Party Transmitters
Once configured the 3rd party transmitters can be connected to other devices in the usual way by accessing the Network page in 'Connect' as shown right. This allows the user to connect the 3rd party audio streams being transmitted to be connected to a receiver in the core or to other receivers in the network.

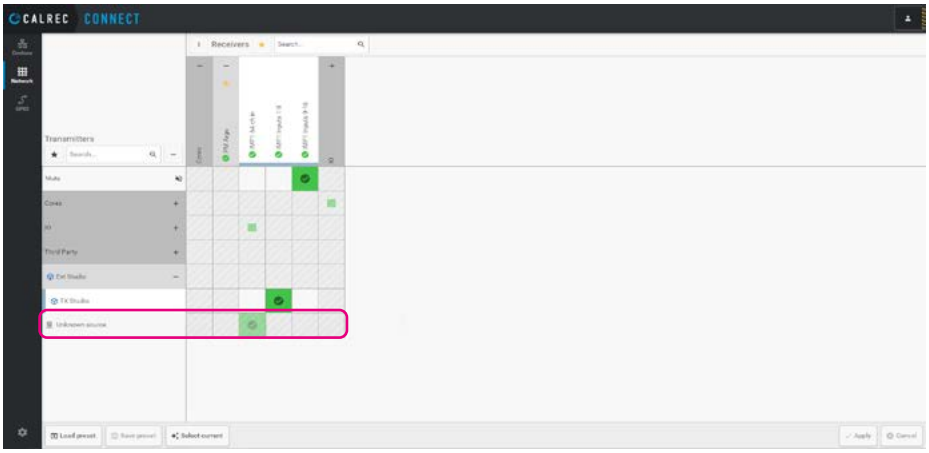
Note: sometimes streams are created outside of 'Connect' these are shown as an 'Unknown' source' to show that they are present as highlighted middle right.

Connect to 3rd Party Receivers
Similarly audio outputs from the core can be placed on the core's transmitter streams to be connected to receivers on 3rd party AoIP devices. In order to do this the core transmitter's SDP can be copied as shown in the image below right for the 'Imp TX2' transmitter. This would then be pasted into the 3rd party Destination (Receivers) page such as on a 'Bach' AoIP module.

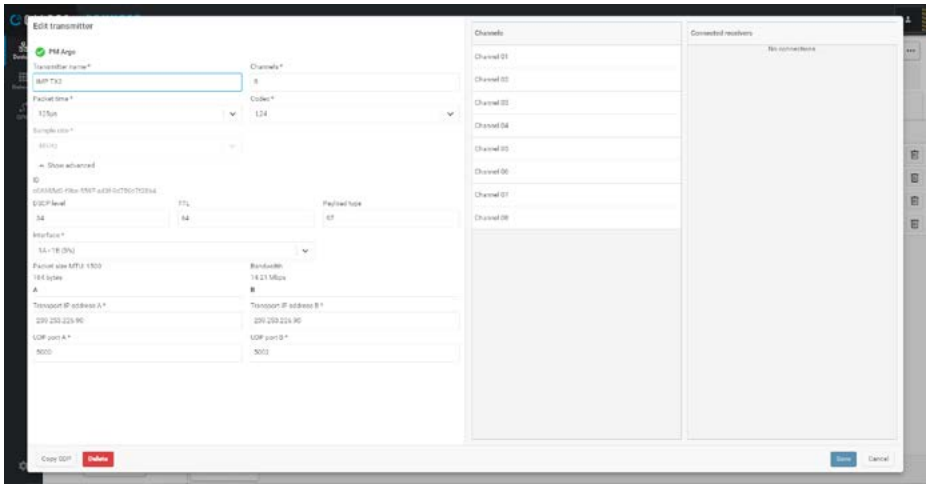
DEVICES>TRANSMITTER CREATED THEN EDITED IN CONNECT



NETWORK>3RD PARTY TRANSMITTER CONNECTED TO CORE RECEIVER



DEVICES>AOIP CORE TRANSMITTERS-COPY SDP FOR 3RD PARTY RECEIVER

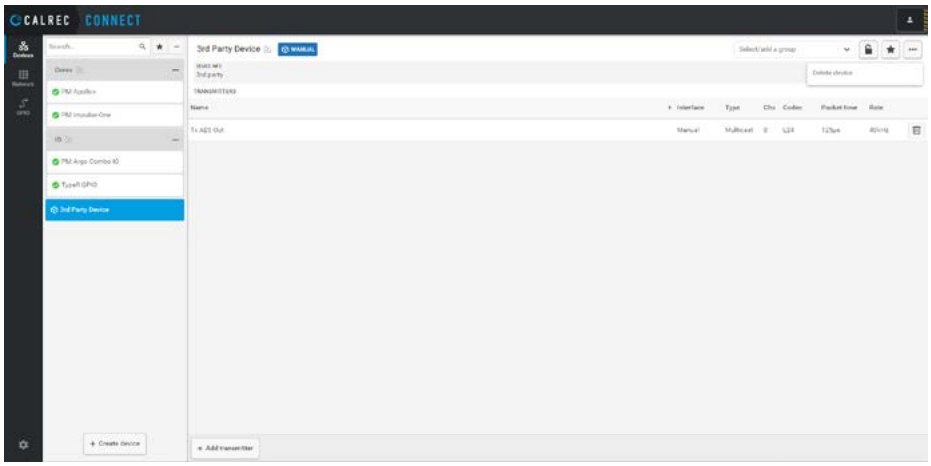


Receivers can be created on the 3rd party device page and the SDP data is entered for each receiver in its 'Advanced' section. An important point to note with manual connection is that without NMOS or other advertising protocols, 3rd party receivers will not appear in the connect networking page and that the connection between Core transmitters and 3rd party receivers have to be manually added in a stream by stream basis.

Deleting 3rd Party Devices
3rd party devices can be removed from the devices list when no longer required.

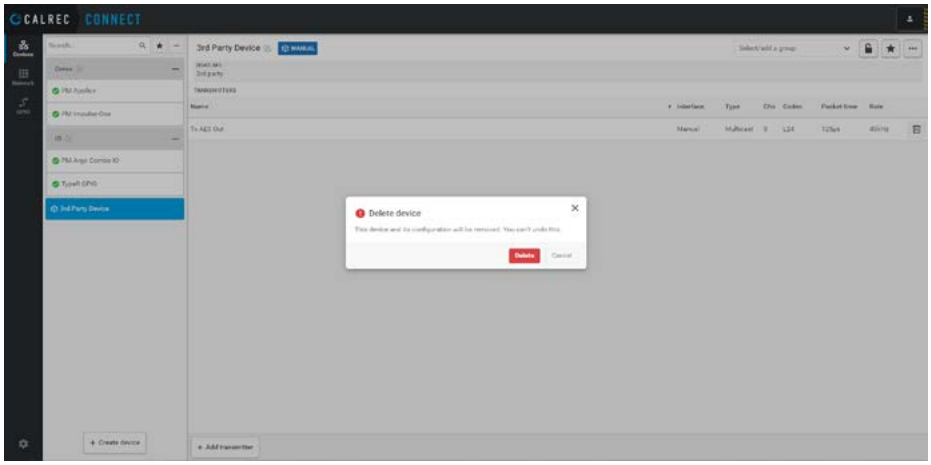
Clicking on the '▪▪▪' in the top right of the devices page open the options menu and appears as shown above right.

DEVICES>DELETE 3RD PARTY DEVICE



The user then clicks on the 'Delete device' button, which opens the confirmation dialogue box shown right.

DEVICES>CONFIRM DELETE 3RD PARTY DEVICE



Clicking on the 'Delete' button will remove the device including its configuration and a 'Device Forgotten' message appears confirming that the device has been forgotten and as shown below right, the device has been removed from the list.

DEVICES>3RD PARTY DEVICE FORGOTTEN AND REMOVED FROM LIST

3RD PARTY CONNECT CASES

Case 1: Merging Technologies HAPI
The HAPI unit has been designed so that MADI, AES, SPDIF/ADAT and Analog modules can be fitted to the unit and all feed into or take their sources from the network over RAVENNA streams, providing up to 88 channels of I/O @ 1FS over a single CAT5e or CAT6 cable to any other devices on the network. The RAVENNA connection on the HAPI allows audio, control and sync information and remote control access to the configuration and routing of the HAPI unit itself.

Create the HAPI device in Connect
As shown earlier in **"Create Devices" on page 43** the first step to connect to the HAPI unit is to create the HAPI device. This has been done as shown above right.

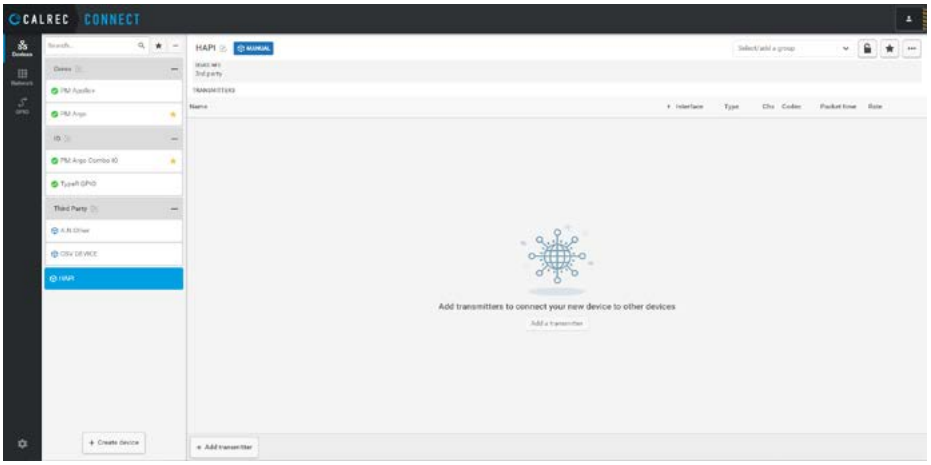
Configuring the HAPI device
To familiarise the user with the HAPI unit its user manual can be found at https://www.merging.com/uploads/assets/Installers/Firmware/Hapi%20User%20Manual.pdf

Using the Merging Technologies ANEMAN audio network manager application which can be downloaded from https://www.merging.com/support/downloads#aneman, the user can access and configure the various parameters so that it can be connected to the Core.

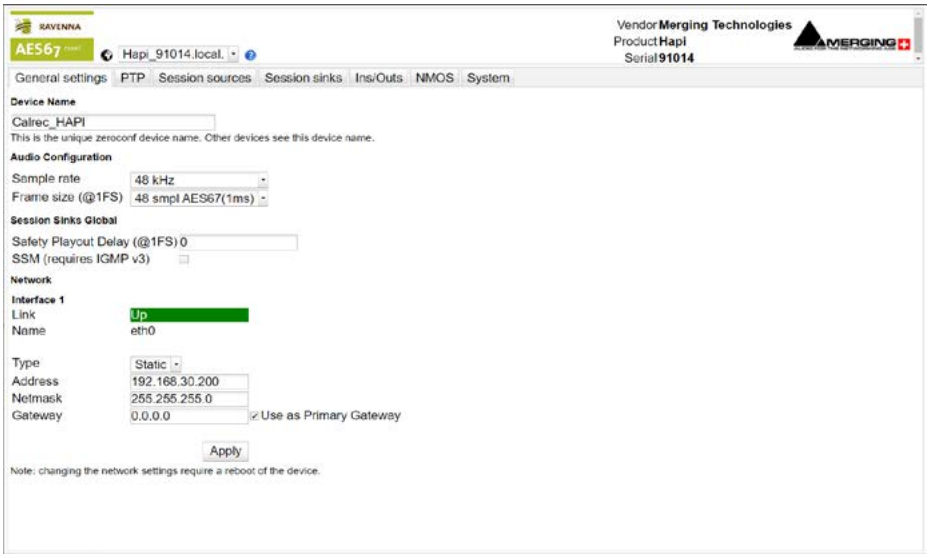
HAPI General settings
The page on the right shows that the IP Address and subnet of the HAPI unit has been changed to be in the same range as the Impulse/Type R AoIP network. The other important points to note are that the Sample Rate is set to 48kHz and the Frame size has to be set to support AES67 at a Packet Size of 1ms. In order to connect into the Calrec core the Packet size of the Calrec core's receivers have to be set to match the HAPI's transmitters

HAPI PTP settings
The page below right shows that the PTP domain has been set to '127' and is set to Slave only which allows it to Lock to the Core's PTP clock. This is confirmed in the status area of the page. The other important setting here is the DSCP class of service which should be set to 46(EF).

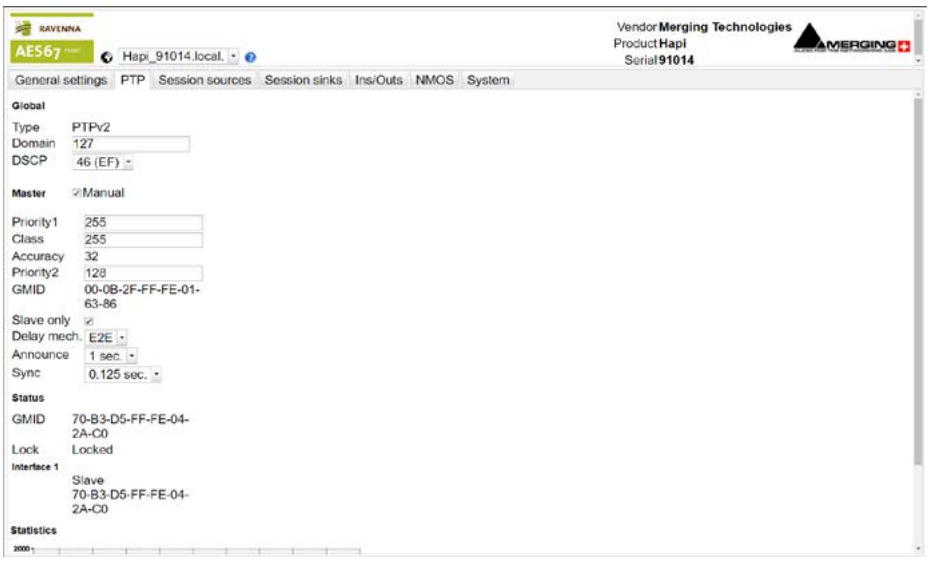
DEVICES>CREATE HAPI DEVICE



HAPI>RAVENNA>GENERAL SETTINGS



HAPI>RAVENNA>PTP



Creating a HAPI Transmitter
Select the 'Session sources' tab and create a source (Transmitter) by clicking on the little connection cable Icon (top left), this opens the configuration page shown above right. The user then selects and fills in the various parameters to create an SDP:-
IO: this defines the physical interface module from which the audio to be transmitted comes from, e.g. 'AES'.
Name: this is the session name e.g 'Calrec_HAPI1'.
Description: optional description info.
Output Interface: which AoIP interface port will be used e.g. 'Interface 1'
Stream Destination IP: typically an IP4 based multicast address e.g. '239.1.152.1'.
TTL: Time To Live duration/number of Hops before packet is dropped e.g. '15'.
Payload Type: For Audio encodings , this is typically 97 or 98 and forms part of the information provided in the SDP.
Codec: 'L24' is the default codec.
Frame Size: in this case '48 samples' are defined in the AES67 1ms Packet Size.
DSCP: is set to 34 (AF41).such as
Channels: Using 1ms Packet times provides a maximum of 8 channels/stream.

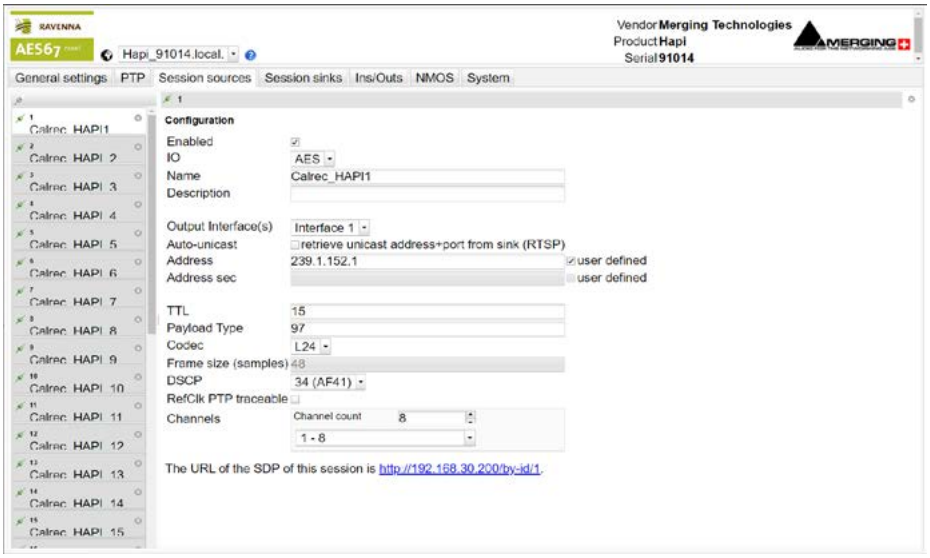
Once all the parameters are defined the SDP file can be copied by clicking on the blue link at the bottom of the page. This can then be saved as a notepad file, copied using 'CTRL C ', and pasted into the Create Transmitter> Paste SDP tab in Connect as shown right, using 'CTRL V'.

Clicking on the 'Create' button will create a transmitter based on the SDP information that was pasted.

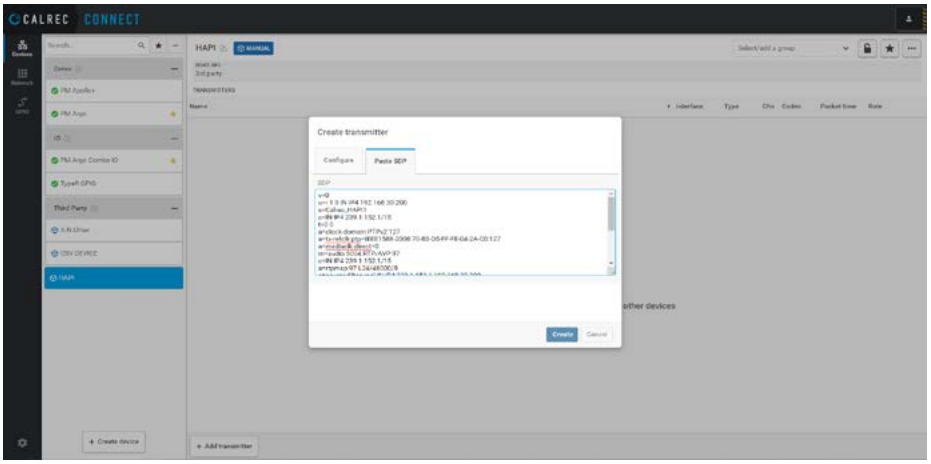
Connect from 3rd Party Transmitters
Once configured the HAPI transmitter can be connected to other devices in the usual way by accessing the Network page in 'Connect' as shown below right.
This allows the user to connect the HAPI audio stream being transmitted as 'Calrec_HAPI1' to be connected to a receiver in the Calrec core which has been created called 'HAPI Rx1'.

Note: the Calrec receiver has to be configured with a Packet size of 1ms to match the HAPI transmitter in order for it to connect.

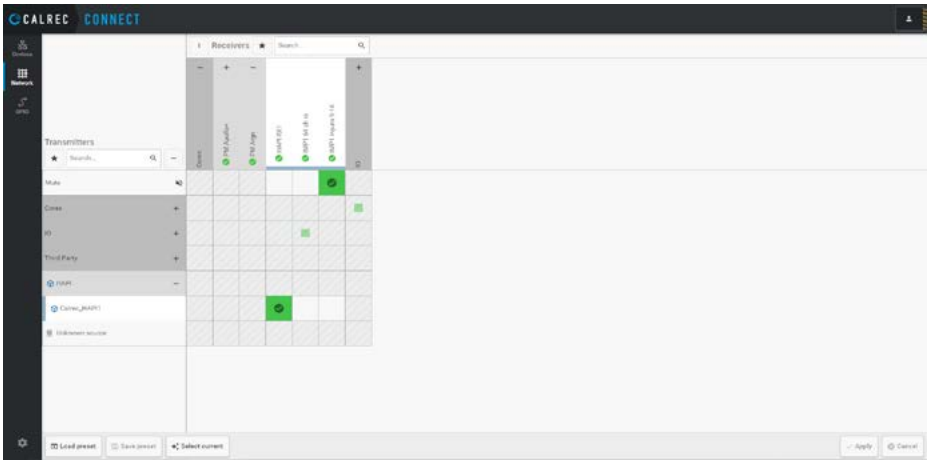
HAPI>RAVENNA>CREATE HAPI TRANSMITTER AND COPY SDP



DEVICES>CREATE HAPI TRANSMITTER AND PASTE SDP



NETWORK>HAPI TRANSMITTER CONNECT TO CORE RECEIVER



Creating a HAPI Receiver
The first step is to create a suitable Transmitter from Connect that can be connected to a receiver in the HAPI device.

The image above right shows that an appropriate transmitter labelled 'HAPI Tx1' has been created in Connect.

An important point to note is that it has been created with a Packet time of 1ms in order to successfully connect to the receiver in the HAPI device.

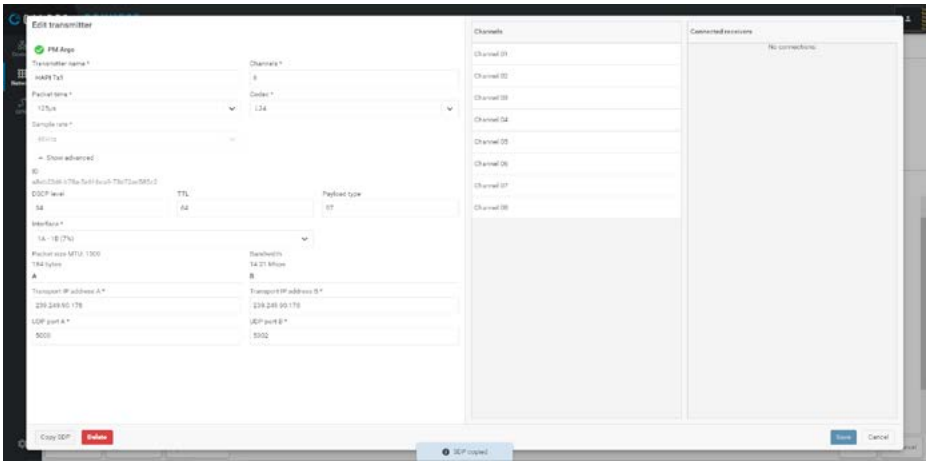
The SDP for this transmitter has been saved as a file and copied ready to be transferred to the HAPI device as shown.

The user then selects the 'Session sinks' tab in the ANEMAN application and creates a sink (Receiver) by clicking on the Source dropdown as shown right.

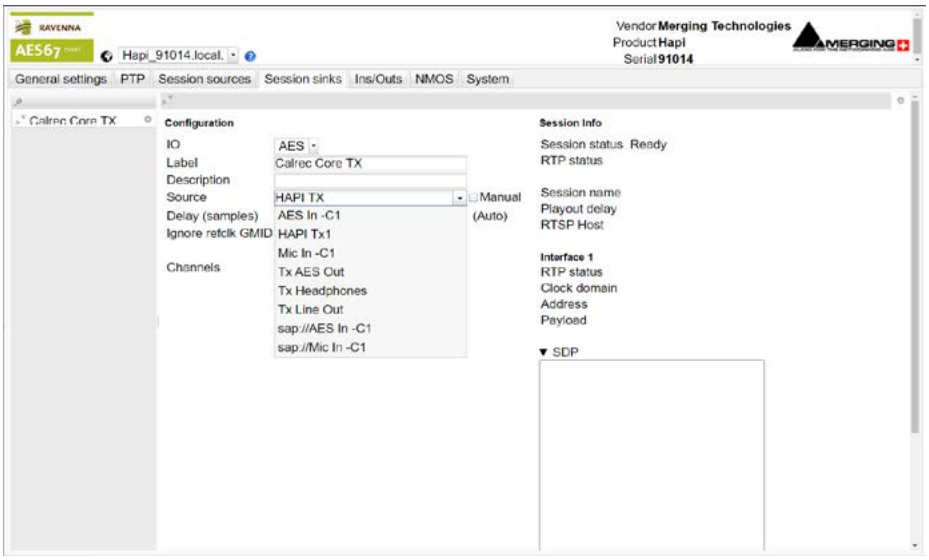
This opens a list of transmitters to connect to and the user selects the HAPI Tx1 entry created from the Connect page 's SDP file.

Once selected all the Session Info and SDP data appears on the right side of the Session sinks page to show that the HAPI device is connected to and receiving RTP packets from the address defined in the SDP as shown below right.

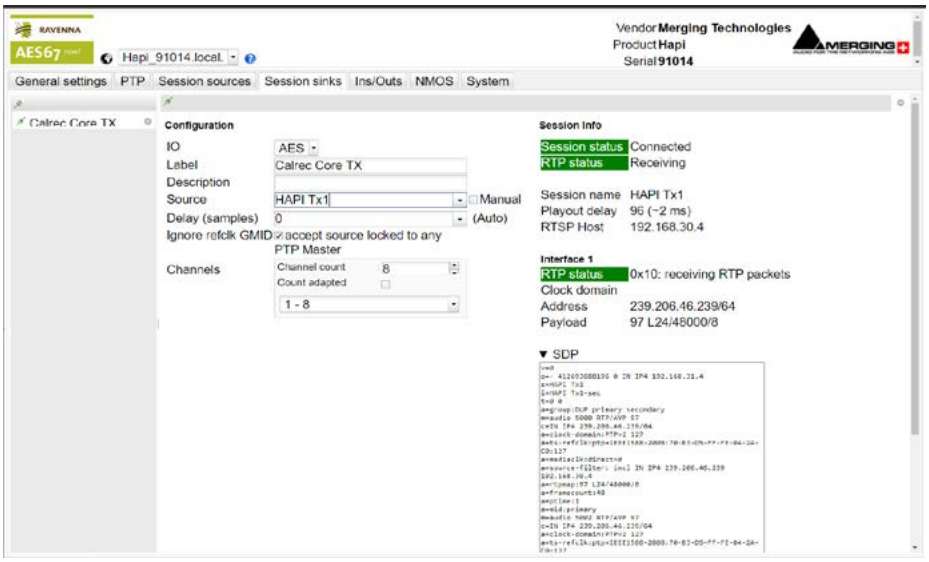
HAPI>RAVENNA>CREATE HAPI TRANSMITTER AND COPY SDP



DEVICES>CREATE HAPI RECEIVER AND SELECT ITS SDP FILE



NETWORK>HAPI TRANSMITTER CONNECT TO CORE RECEIVER



Case 2: AXIA Livewire+ IP Driver
The Axia IP-Audio Driver is the software interface between PC audio applications and the AoIP network and provides the following functions:
• Sends audio "sources" to the AoIP network from PC/Windows audio applications such as multichannel delivery systems and other audio players.
• Receives audio from the AoIP network to "destinations" on the PC/Windows system such as audio recording apps.

Versions of the IP-Audio driver appear to Windows as standard audio input/output devices. There are a wide variety of applications where the Axia IP driver and the Windows software are used together, and the Driver provides all software components needed to record and play AoIP streams using standard recording software for Windows.

The Axia IP driver also provides a software interface ASIO driver that can interface audio to and from DAW workstations without the use of a Hardware Audio Interface and arranged as a number of AES67 input and output streams for use with other AoIP devices such as Type R or Impulse cores.

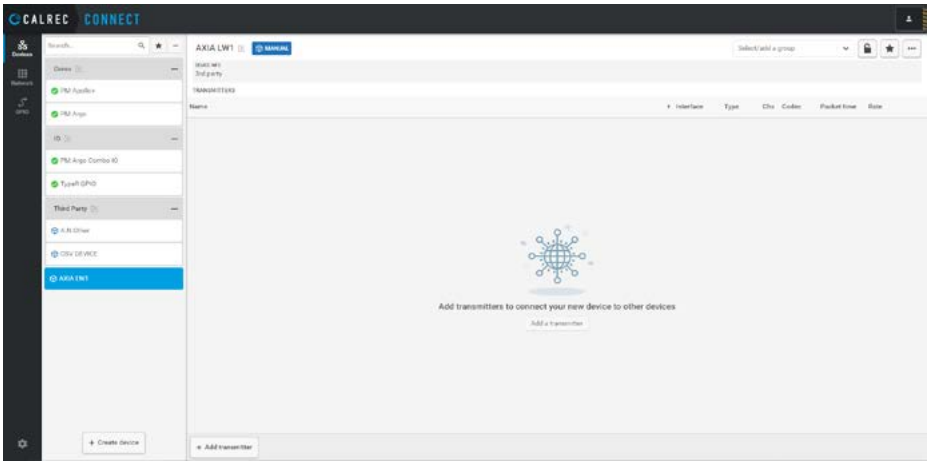
Create the AXIA device in Connect
As shown earlier in **"Create Devices" on page 43** the first step to connect to the AXIA IP driver is to create the AXIA device. This has been done as shown above right.

Installing the AXIA IP Audio Driver
In order to install the Axia IP – Audio Driver you will need both a License Code and a License Key which you will receive from Axia when you purchase a license for the Axia IP – Audio Diver.

Note: Calrec does not sell the licenses, these are purchased from Axia directly.

To install the AXIA IP- audio driver interface, the current Axia IP - Driver version 2.11.0.10 release and install information can be found here:
https://support.telosalliance.com/article/cm6rtgztlb-axia-ip-driver-version-2-11-0-10-release-notes-and-install-instructions

DEVICES>CREATE AXIA DEVICE



This version of the driver adds the 1ms packet time required in AES67, the following information is useful when setting up the streams from the Axia IP - Driver, which supports AES67 streams.

This is shown as the "Low Latency Stereo (AES67)" Mode and the default configuration is:
Packet Time: 1ms
Depth: 24 bit (L24)
Sample Rate: 48kHz
Channels per stream: 2
UDP Port number: 5004

Axia IP Channel numbers use the multicast transport IP range 239.192.0.0 where
Channel 1 = 239.192.0.1
Channel 2 = 239.192.0.2
etc.

Details of how Axia Channel numbers are defined can be found here:
https://www.telosalliance.com/images/Axia%20Products/Support%20Documents/Tech%20Tips/AxiaLivewireChannelNumbering.pdf

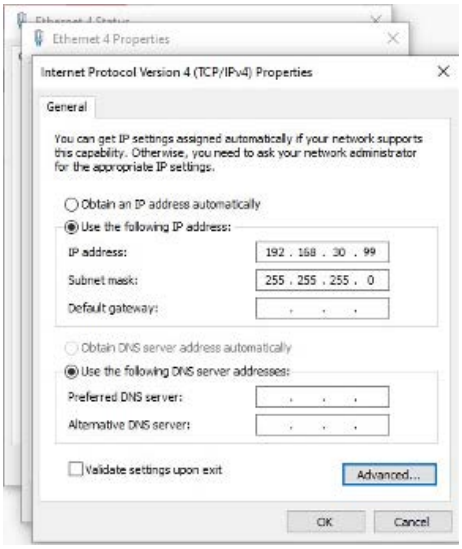
Configuring the Network Interface on the PC for the Axia IP-Driver.
The PC Network Adapter for the Axia IP - Audio Driver will need a unique address on the AoIP network.

The user will need to configure the Ethernet adapter in the PC with an IP address of 192.168.30.xxx and a subnet mask of 255.255.255.0 (where xxx avoids conflict with other IP addresses on the AoIP network).

In this case the PC adapter has been configured with 192.168.30.99 as shown below right to match the existing AoIP network addressing range.

Note: please refer to the PC's operating instructions, if unsure how to configure ethernet adapters.

WINDOWS ETHERNET SETTINGS



Configuring the Axia IP-Driver.
To configure the Axia IP – Audio Driver;
1. First, select the PC Network Adapter to be used with the Axia IP – Audio Driver, this will be the network adapter which you have just set up giving the PC adapter access to the AoIP network.

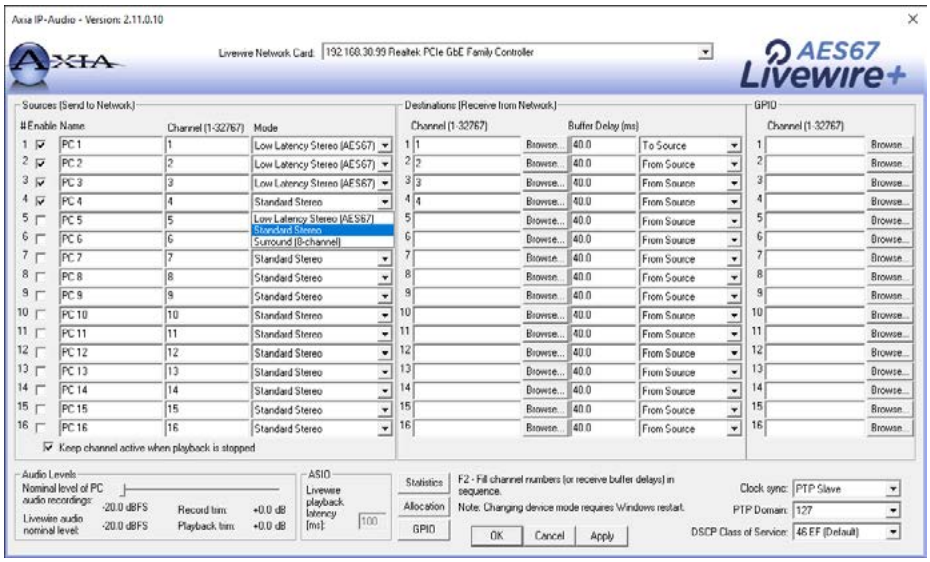
2. Open the Axia IP-Audio Driver Configuration application as shown above right and configure the PTP and DSCP settings:
-Set the Axia IP – Audio Driver Clock Sync to “PTP Slave”
-Set the PTP Domain to “127”

This needs to be set to the same PTP Domain for the PTP clock, Impulse or Type R Core and IO Boxes on the same AoIP network. The “SMPTE:2059-2” PTP profile utilises a domain value of 127, this is the default for Type R/Impulse units.

Note: on Type R units, setting a PTP priority of 0 has been seen to cause the PTP domain to switch to 127 (SMPTE default). Do not set Type-R devices to have a priority of 0 unless using the domain 127.

DSCP Class of Service should be set to the default value of “46 EF (Default)

AXIA>LIVEWIRE + CONFIGURATION WINDOW FOR THE IP DRIVER.



3. To create a stereo Source (Transmitter) from the Axia IP – Audio Driver (Send to Network):
-Tick the “Enable” box by the first source (labelled “PC 1”)
-In the Mode column, use the drop down to select “Low Latency Stereo (AES67)” this is shown in the image above right.
-The Channel number may be changed but to make things simple, keep channel number as “1” as this translates to a transport multicast IP address of 239.192.0.1.

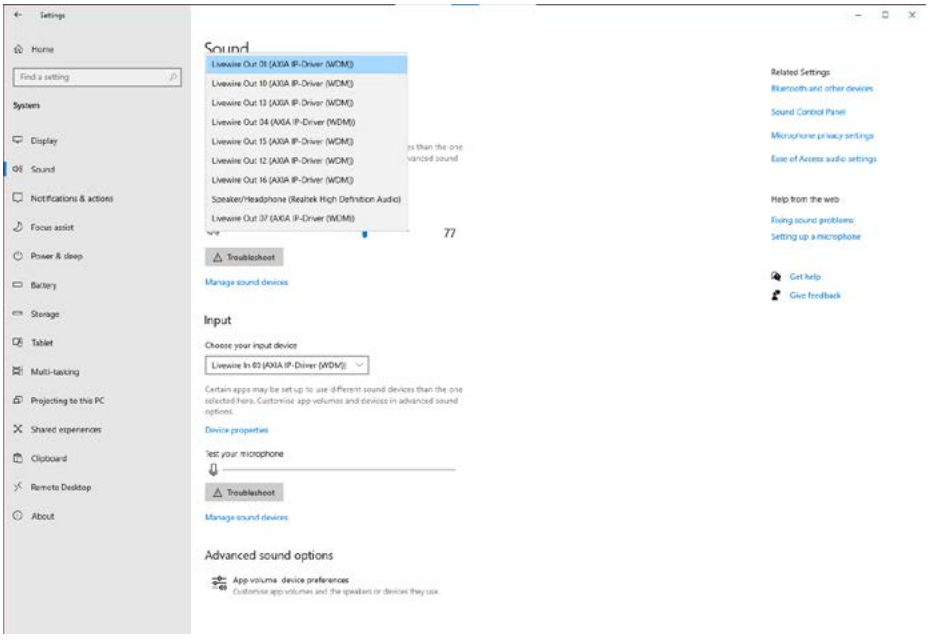
Note: any changes made to the Axia IP Audio Driver UI will not take effect until the “Apply” button is clicked.

Clicking on the Statistics button provides useful information on the PTP clock, Tx/Rx stream activity and Jitter.
Further information on the Axia IP-Audio driver can be found here:
https://www.telosalliance.com/images/Axia%20Products/IP%20Audio%20Driver/Support%20Files/Axia_IP_Audio_Driver_Software_Interface.pdf

Configuring Windows Input and Output Sound settings.
In order to use the Axia IP- Audio driver on the PC, the user goes to the Windows Settings>Sound page and in the ‘Choose your output device’, field the user selects ‘Livewire Out 01 (AXIA IP-Driver(WDM))’ as shown in the image above to route the PC audio output to the first Axia IP-Audio Driver Source (transmitter).

Similarly, if the user wants to use the Axia IP- Audio driver on the PC as an input source, the user goes to the Windows Settings>Sound page and in the ‘Choose your input device’ field the user selects ‘Livewire In 01 (AXIA IP-Driver(WDM))’ as shown in the image above to route the first Axia IP-Audio Driver destination (receiver) to the PC audio input.

WINDOWS SOUND SETTINGS AXIA LIVEWIRE+ SELECTION



Creating an Axia AES67 Transmitter
Sending audio from LiveWire+ (e.g. a DAW via ASIO, or QuickTime via WDM) to Type R or Impulse as a transmitter:

To generate a stream in the LiveWire GUI, set the top left Source (PC1) to Ch1 and select Mode = "Low Latency Stereo (AES67)" See **"AXIA>Livewire + configuration Window for the IP Driver." on page 53.**

Connect does NOT see incoming LiveWire streams, so a dummy device transmitter must be created in Connect see above right to make it visible to the core, the transmission stream should be named and configured as a 2ch/48k/L24 with a 1ms Packet time.

The PC 1 source is sent using the 239.192.0.x multicast address where x represents the channel stream number so in this case the transport IP address is 239.192.0.1 and the UDP ports A & B should be set to 5004 and 5006 respectively as shown right.

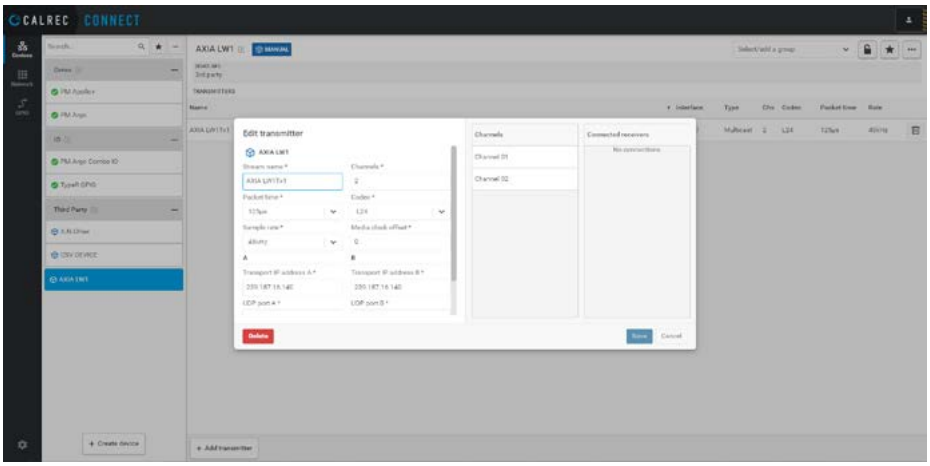
Connect from Axia Transmitter
Once configured the Axia transmitter can be connected to other devices in the usual way by accessing the Network page in 'Connect' as shown below right.

This allows the user to connect the Axia audio stream being transmitted as 'AXIA LW1Tx1' to be connected to a receiver in the Calrec core which has been created called 'AXIA LW1Rx1' shown right.

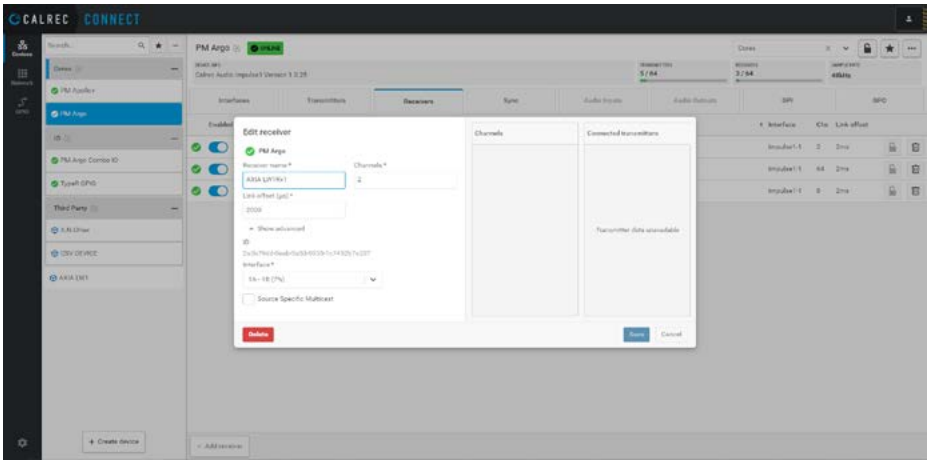
It is important to note that the Calrec receiver has to be configured with a Packet size of 1ms to match the AXIA transmitter in order for it to connect.

The image below right shows that some of the receivers are set to a different packet time i.e 125us and as such will not connect. As many 3rd party devices only work at 1ms packet times, then the other devices in the system would need to be set to the same packet time for it to all work together.

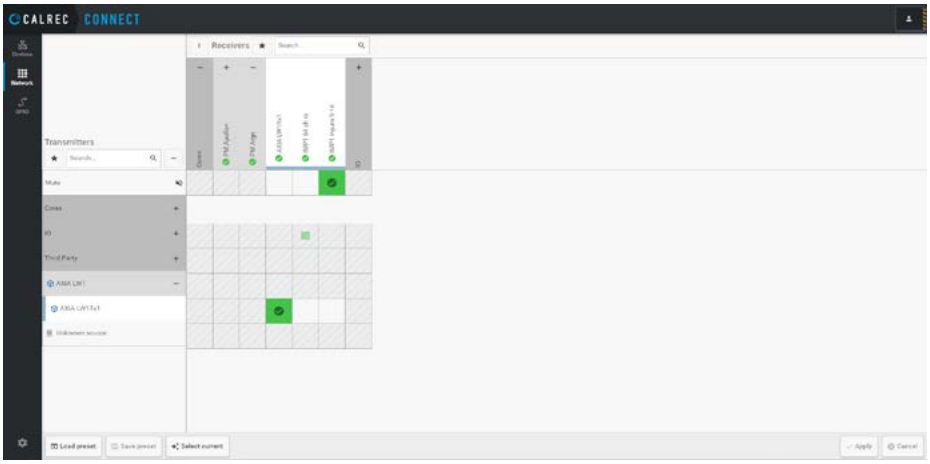
DEVICES>CREATE AXIA TRANSMITTER



DEVICES>CREATE CORE RECEIVER



NETWORK>AXIA TRANSMITTER CONNECT TO CORE RECEIVER



Creating an Axia AES67 Receiver
Receiving audio from Type R or Impulse into LiveWire+ (e.g. a DAW via ASIO, or PC's Sound control panel Recording Device via WDM) as a receiver:

Create a Transmitter stream in Connect as shown above right. Set it up as 2ch/48k/L24 and set the transmit ports to 5004/5006 to avoid having to use Putty to change Livewire's expected defaults. In this case we are using Axia Livewire channel 3. In this case the Multicast address is 239.192.0.3

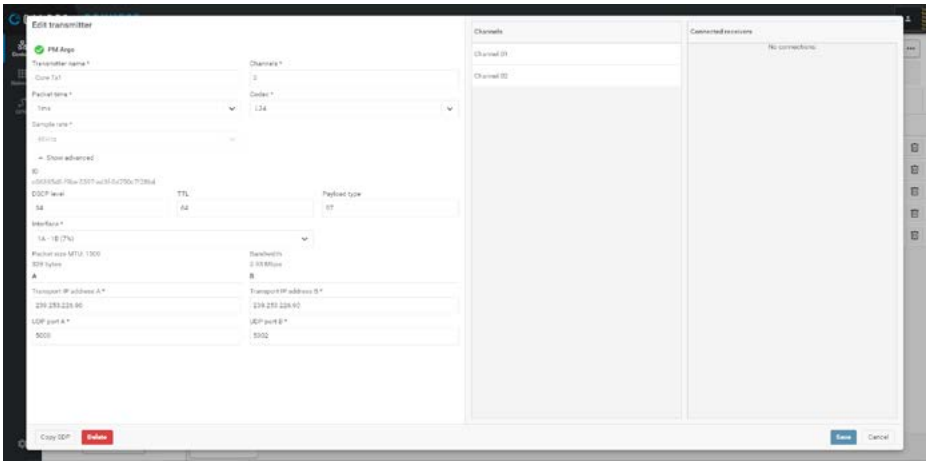
Select the correct IP interface from the driver's Livewire Network Card pull-down menu (top centre) as shown right.

Type in the Transmitter's multicast audio transport IP address directly in to the LiveWire driver's 'Receive from Network'.

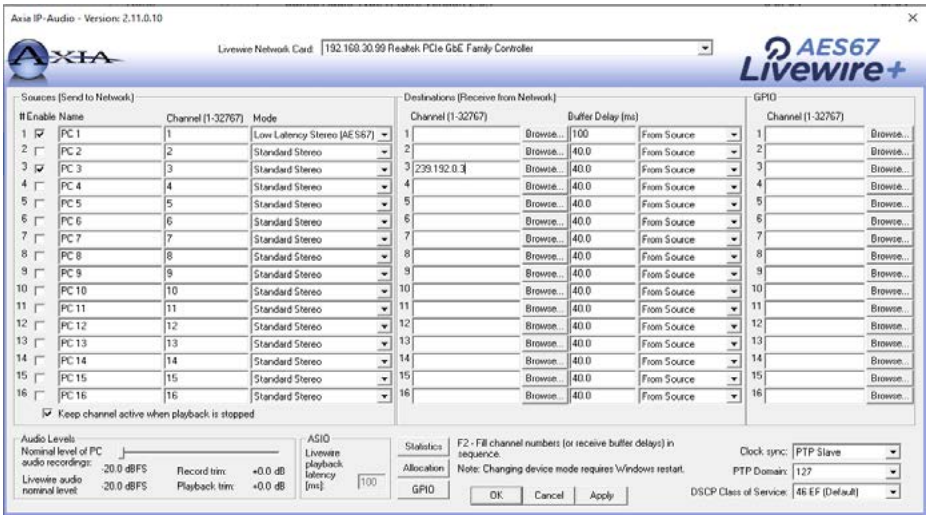
In this case the entry field for channel 3 enter 239.192.0.3 and click on Apply. When this is applied the field will change to 3. This is because Axia uses the range of channels from 1 to 32767 and this translates to the multicast address range of 239.192.0.1 to 239.192.127.255

There is an Axia Livewire SDP Generator which can be downloaded from the Axia ftp site at
ftp.zephyr.com › pub › axia › Tools › sdpgen
Entering the IP interface for the network card and the Livewire channel number as shown below produces:-
Source node IP address: 192.168.30.99
Livewire channel number: 3
v=0
o=Node 1 1 IN IP4 192.168.30.99
s=TestSine
t=0 0
a=type:multicast
c=IN IP4 239.192.0.3
m=audio 5004 RTP/AVP 97
a=rtpmap:97 L24/48000/2 .

DEVICES>CREATE CORE TRANSMITTER FOR AXIA RECEIVER



AXIA CONFIGURATION WINDOW CONNECT CORE TRANSMITTER TO RECEIVER



The Axia LiveWire+ driver DOES NOT see the Transmitter generated by the Type R or Impulse it only sees its own transmitters and receivers. **T**o use these in the PC as output and inputs they need to be setup using the Windows Sound settings, see **"Configuring Windows Input and Output Sound settings." on page 53**

When everything is set right...
- Axia LiveWire+ locks to PTP
- Type R or Impulse receives stereo 48k L24 streams from the PC.
- The Livewire+ driver receives stereo 48k L24 streams from the Type R or Impulse.

Case 3: Audinate DANTE
Dante is developed by Audinate and is available to manufacturers of audio and video products in several formats that suit different product types. Dante audio is available in the form of hardware modules, chips, reference designs and software that may be incorporated into products.

The Dante hardware or software in a device takes digital audio & 'packetises' it. The audio is segmented and wrapped it in IP (Internet Protocol) packets suitable for transmission across a standard IP network.

The packets contain timing information and source and destination network addresses, allowing them to be efficiently routed through the network to the correct destination. This connect case provides guidance on using the Dante Controller to connect AES67 Dante end points with the Calrec Type R or Impulse system cores. In order to proceed the user needs to obtain copies of the Dante Domain Manager and Dante Controller.

Obtaining DDM and Dante Controller
The Dante Controller application can be downloaded from the Audinate website:
Windows version https://my.audinate.com/content/dante-controller-v4231-windows

Mac OS version https://my.audinate.com/content/dante-controller-v4253-macos

Note: The user must be logged in with an Audinate.com account to see the download link,

The Dante Controller v4.2.3.1 for Windows introduces Dante Updater, support for SMPTE 2110-30 RTP audio flows* for DDM-enrolled supporting devices (requires v4.2 firmware), and a range of bug fixes and performance improvements.

* Requires Dante Domain Manager (DDM) v1.1 and v4.2 device firmware. Please talk to the manufacturer regarding firmware availability.

In order to install Dante Domain Manager the user will need a license for either the Silver, Gold or Platinum edition.

Note Calrec does not sell the licenses, these are purchased from Audinate directly.

These case notes are based upon Dante Controller v4.2.3.1, Dante Domain Manager (DDM) v1.1 and v4.2 device firmware.

Procedure Outline for Type R
1. Set up your Type R console and AoIP Network.
2. Install Dante Controller and DDM.
3. Configure the Network Interface on the PC on which the user has installed the Dante Controller and DDM .
4. Set Up the Dante device and create a source transmitter to the network.
5. Use Connect to set up Type R Devices to work with the Dante card.
Create a Receiver for the Type R Core and Network the DANTE transmitter to the Core Receiver.
6. Use Connect and the Dante Controller tor receive channels from the core transmitter to the DANTE receiver.

Note: the setup for an Impulse core follows the same procedure.

Procedure Detail for Type R
1. Set up your Type R console and AoIP Network
It is recommended that the user sets up the Type R console and AoIP network establishing that Transmitters and Receivers can be created in Connect and pass audio between a Type R IO box and the Type R Core before proceeding to set up Dante. This will confirm that the AoIP network is running and importantly that PTP clocking has been established.

More information on how to Set up the Type R Console is provided in the **Type R Start Up Guide (926-282).pdf'** which can be requested from your Calrec representative or downloaded from the Calrec Web site.

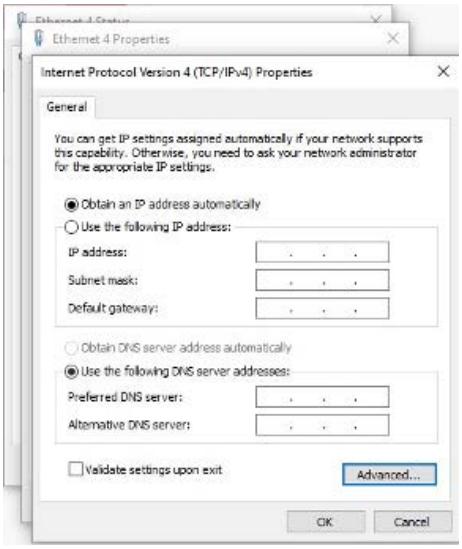
2. Install Dante Controller
The Dante Controller v4.2.3.1 for Windows release and install information can be found here: https://my.audinate.com/content/dante-controller-v4231-windows

A user guide is available here: https://dev.audinate.com/GA/dante-controller/userguide/webhelp/content/front_page.htm

The Dante Controller supports AES67 streams. By default, they are configured as follows:
Multicast address: 239.69.xxx.xxx
Packet Time: 1mS
Depth: 24 bit (L24)
Sample Rate: 48kHz
Channels per stream: 8
UDP Port number: 5005
PTP Domain: 0
PTP Priority 1: 128
For DDM networks, supporting devices that are enrolled in a SMPTE-enabled domain can transmit and subscribe to SMPTE ST 2110-30 flows.

3. Configure the Network Interface on the PC on which you have installed Dante Controller The Dante device IP address should be set to be within the same subnet of the media network as the AoIP devices. Initially, configure the Ethernet adapter in the PC to obtain an IP address automatically. The Dante device IP address should be set to be within the same subnet of the media network as the AoIP devices. Initially, configure the Ethernet adapter in the PC to obtain an IP address automatically so that this can later be set to the 192.168.30.xxx range as shown below.

WINDOWS ETHERNET SETTINGS



RECONFIGURING THE DANTE INTERFACE ADDRESS METHOD FOR TYPE R USE

Using a Cat5e cable , plug directly from the PC Ethernet adapter to the Dante device Ethernet port. In Dante Controller, select the icon shown above right in red to go to the network view.

In the drop-down menu select the correct Ethernet interface. In this case, it's called "Ethernet". In this setup, disable "Use shared Dante interface". See image middle right

Each Dante device will need a unique address on this network. This can be set using Dante Controller.

From the Device Info tab, double click the Device Name row. Go to Device View, Network Config tab. Select "Manually configure an IP Address". See image far right.

In this example, the device has been configured with 192.168.30.6. Hit "Apply", then Reboot the device. A pop-up box will appear asking for confirmation.

AES67 mode needs enabling as well. This is done from Device View, AES67 Config tab as shown lower far right.

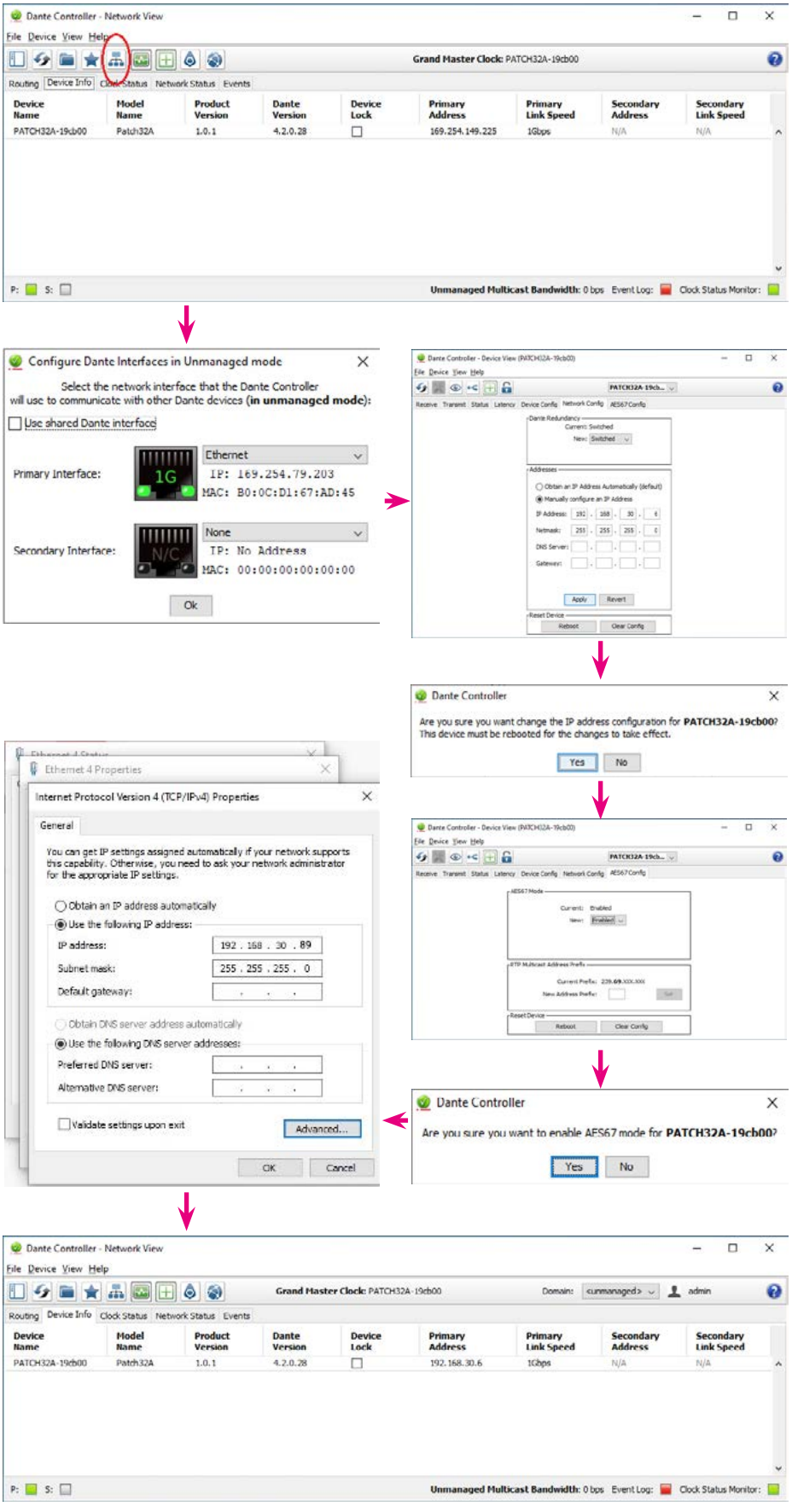
A reboot isn't required to enable AES67 mode. A pop-up box will appear asking for confirmation.

Configure the Ethernet adapter in the PC with IP address 192.168.30.xxx and subnet mask 255.255.255.0 (where xxx avoids conflict with other IP addresses on the AoIP network).

Note: please refer to the PC's operating instructions, if unsure how to configure its ethernet adapter.

In this case, the PC adapter has been configured with 192.168.30.89 as shown lower middle right.

Go to Device Info and verify the Dante device is on the 192.168.30.xx network as shown below right.



4. Set Up Dante Domain Manager (DDM), Dante Controller and create a Multicast Flow

It is beyond the scope of this document to detail setting up a Virtual Machine and licensing Dante Domain Manager. Assuming that part is complete, start the Virtual Machine and launch DDM.

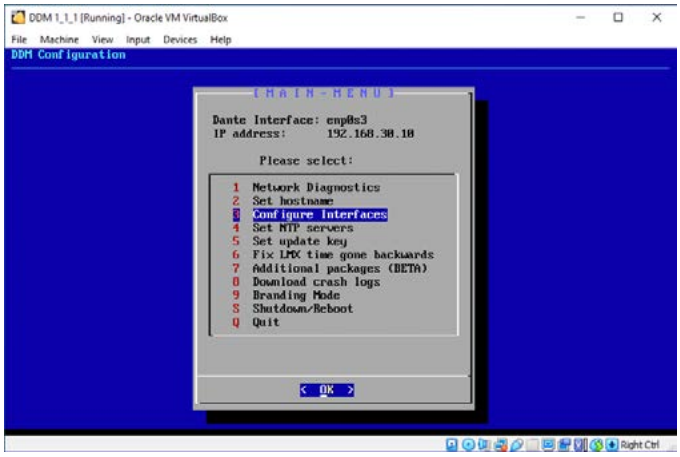
From the Main Menu, select item 3 to configure the IP address of DDM.

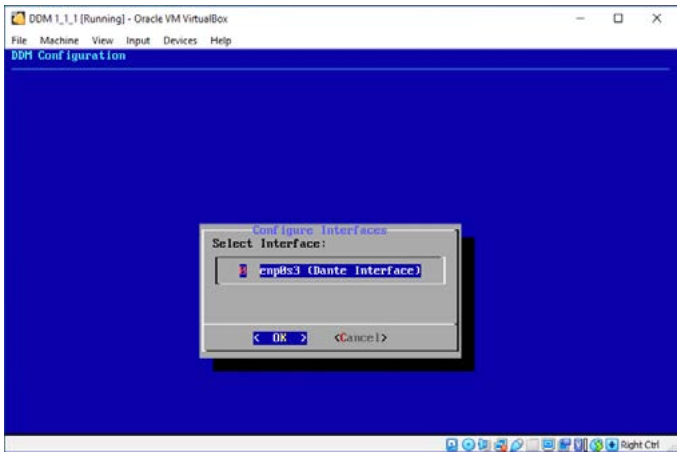
Select the virtual adapter:

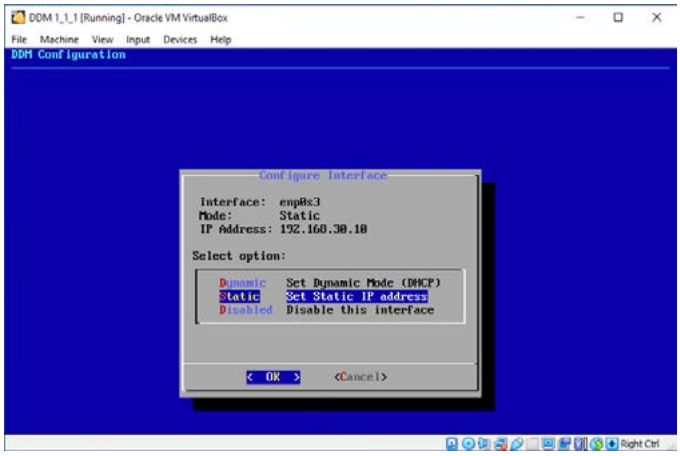
Select “Set Static IP Address” option:

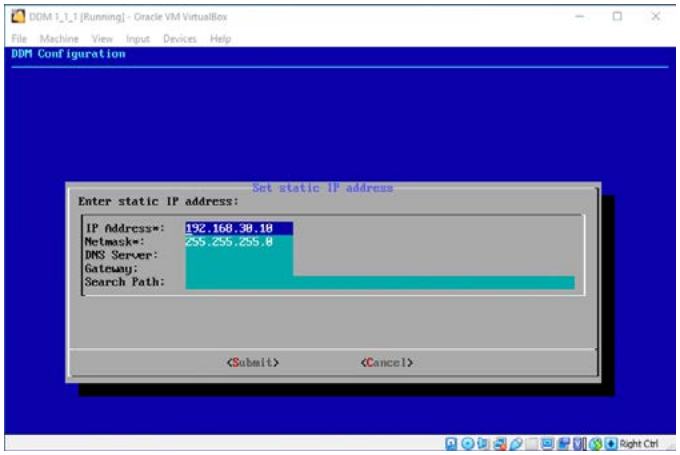
Enter the address, mask and other pertinent info as required. In this case, using 192.168.30.10

CONFIGURING THE IP ADDRESS FOR THE DDM



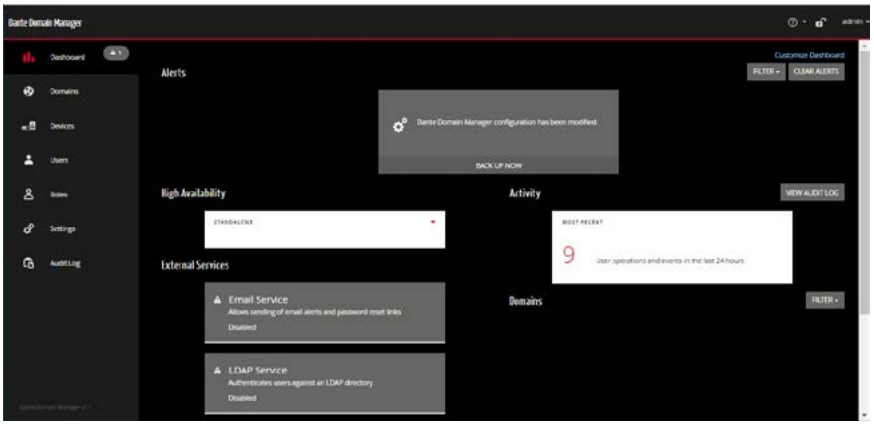




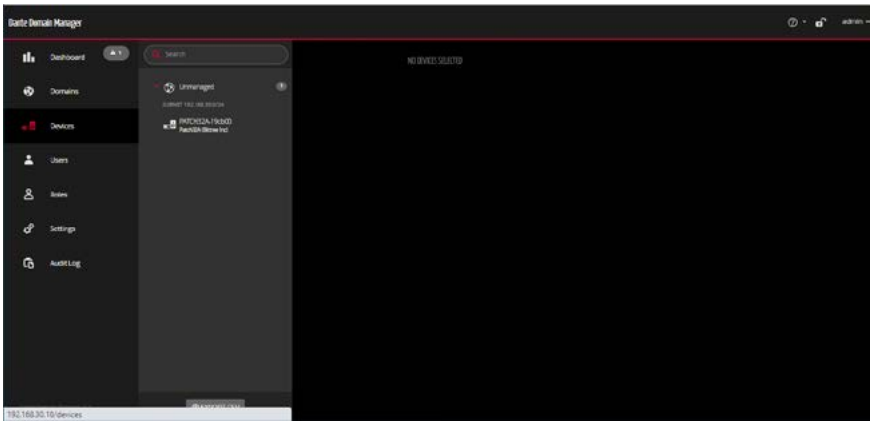


CREATE A DOMAIN

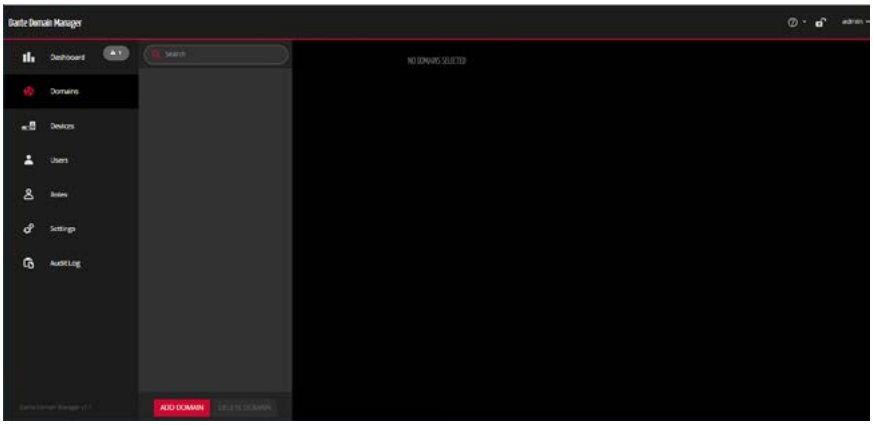
Once the DDM has an IP address, open Chrome and go to the address just configured (192.168.30.10)



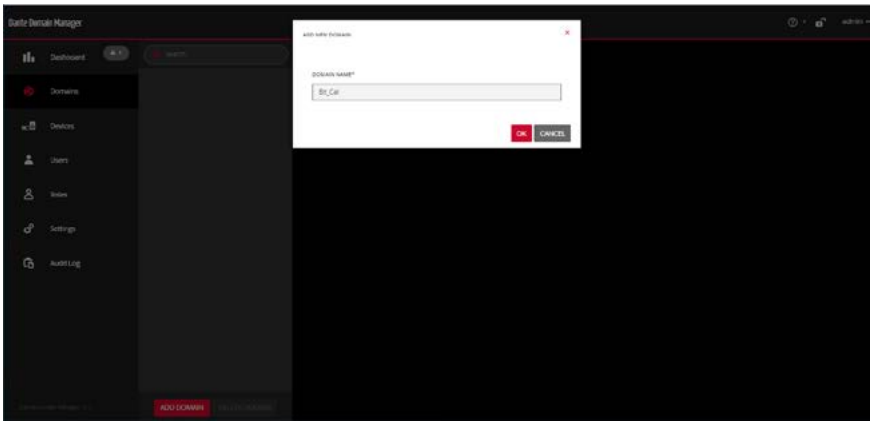
In the left window pane, select “Devices”. Confirm the Dante device appears under “Unmanaged”



Select “Domains” in the left windowpane.

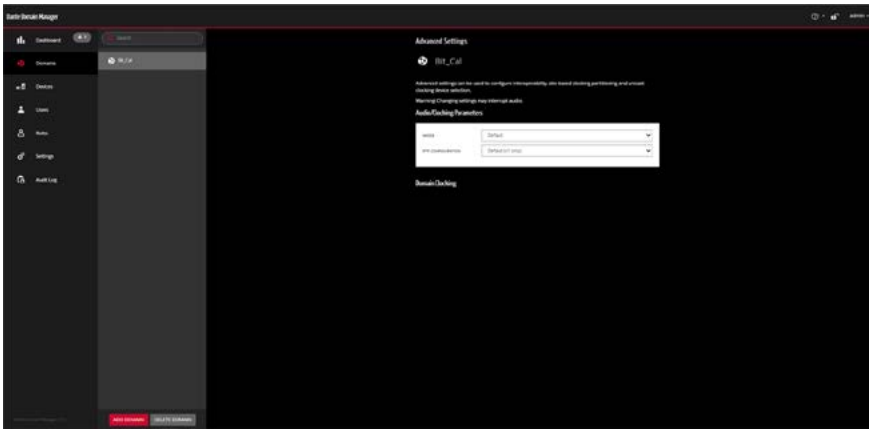


Select “Add Domain” and enter a name when prompted:

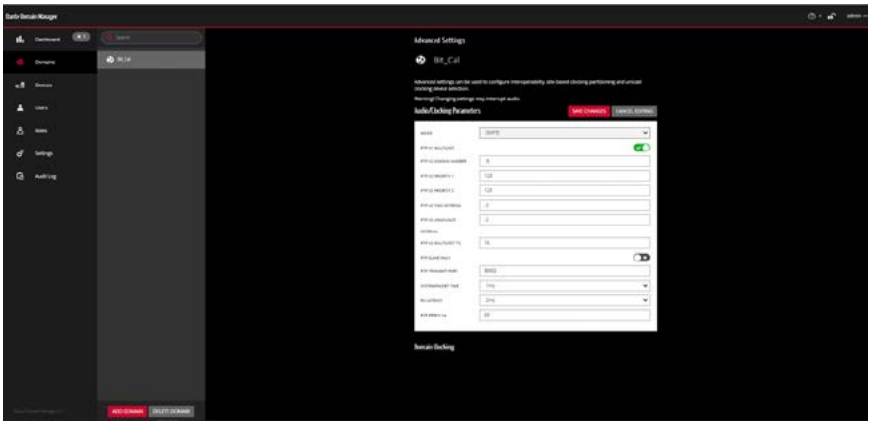


CONFIGURE PARAMETERS

Go to “Advanced Settings” as shown above right.

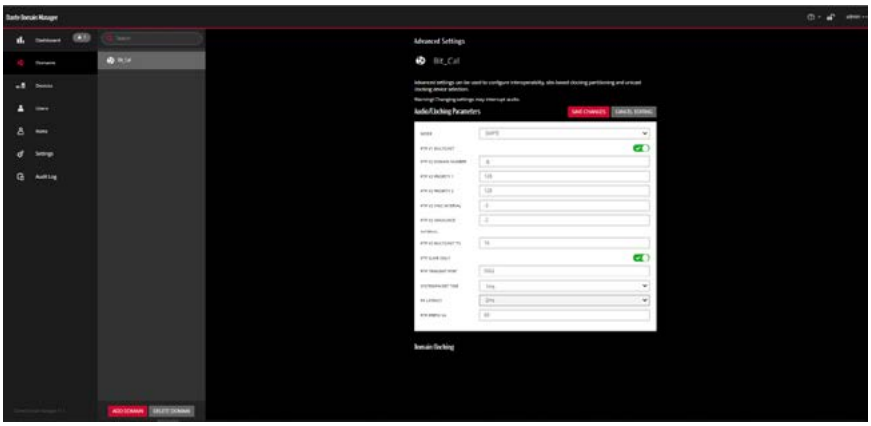


Under “Mode”, select SMPTE as shown right.

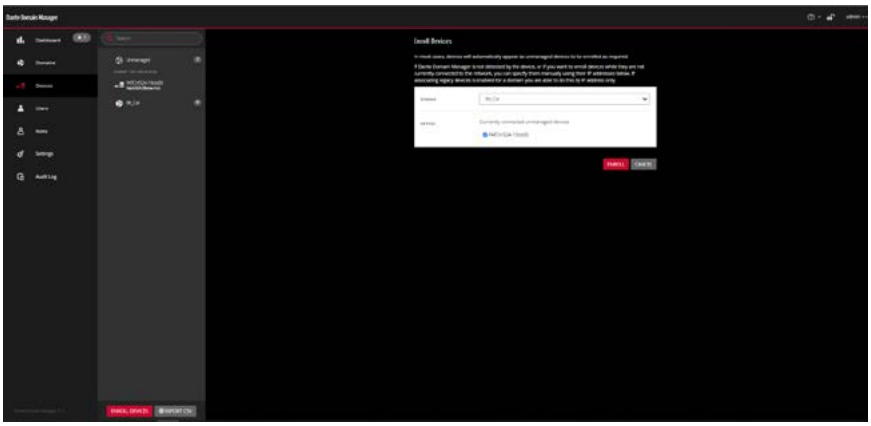


Change the PTP v2 Domain Number to 0 (default = 127) and Priority 1 & 2 to desired values. The PTP v2 Sync and Announce Interval will be set to SMPTE default values of -3 and -2 respectively. Optionally, change the PTP v2 TTL value from 16 to the desired value. Be sure to change “PTP Slave Only” to “Enabled”.

The RTP Transmit Port is normally set to a value between 5000 and 5005. In this example 5002 is used.
System Packet Time set to 1ms
RX Latency left at 2ms and RTP Prefix V4 at 69. All settings shown lower right.

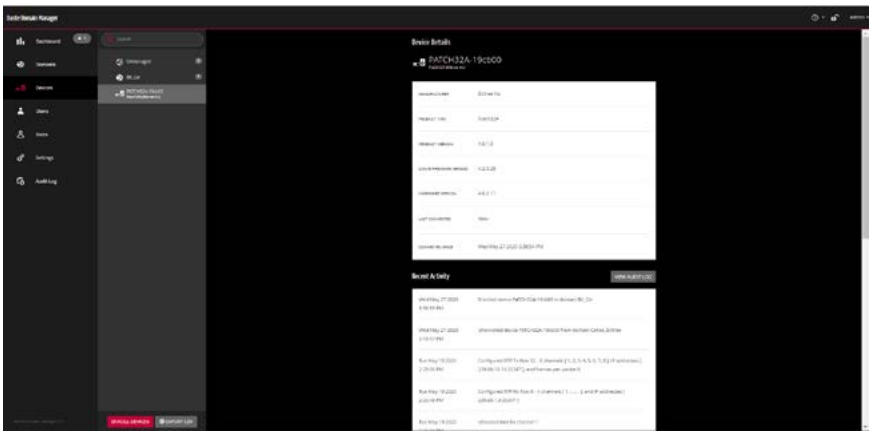


Press SAVE CHANGES when finished. Go back to the Devices menu> Enrol Devices and Add the Dante Device, tick the selection box for the device, press Enrol. Press Ok for “Confirm Device Enrolment”. See below right.

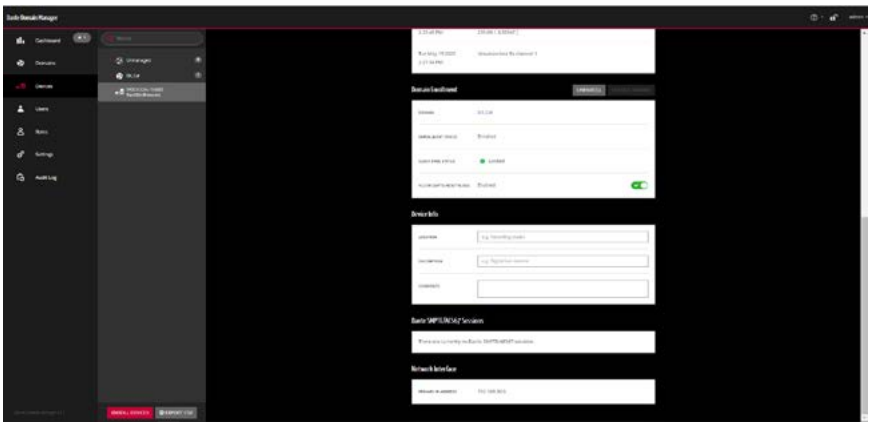


CONFIGURE PARAMETERS CONTINUED

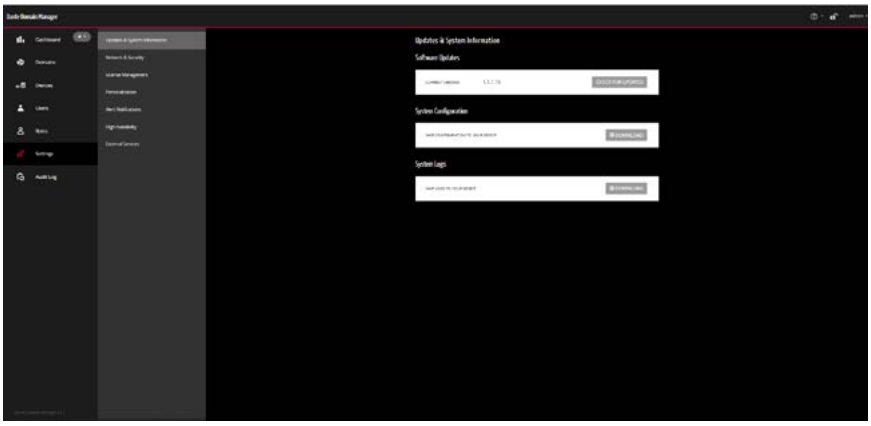
A page showing “Device Details” will appear.



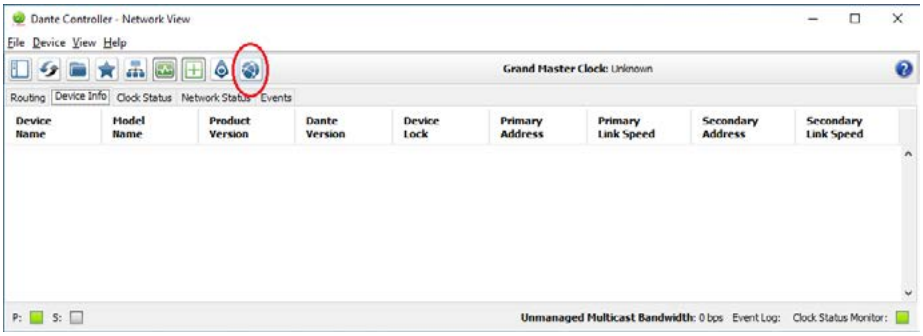
Scroll down for more information.



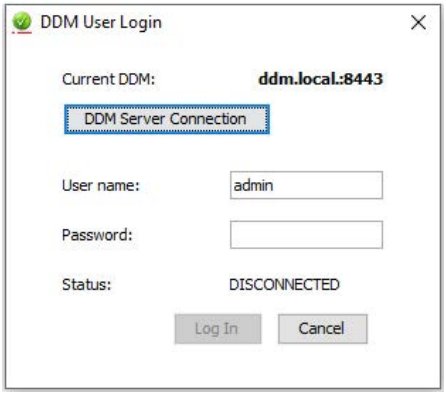
In the left windowpane, Settings, Updates & System Information, the System Config can be saved.



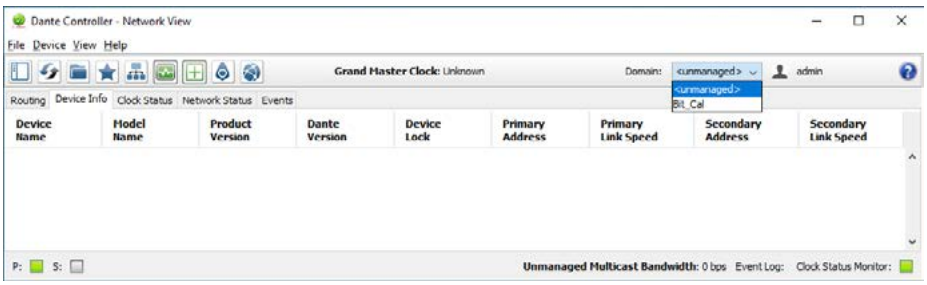
Go back to the Dante Controller. Select Device Info and press on the World Icon.



A login screen for DDM will appear as shown above right. Fill in the pertinent info and press Log In.



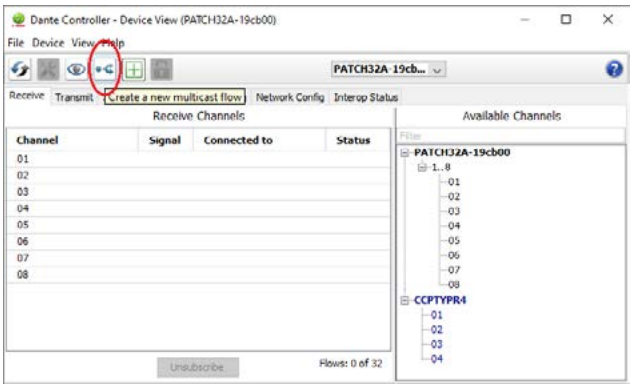
In the Dante Controller-Network view window a Domain drop-down menu appears in the top right corner of the window. Select the required domain as shown right.



For DDM networks, supporting devices that are enrolled in a SMPTE-enabled domain can transmit and subscribe to SMPTE ST 2110-30 flows. SMPTE flows created in Dante Controller are advertised on the network via SAP/SDP.

To create a stream from the Dante device to the Type R, proceed as follows:

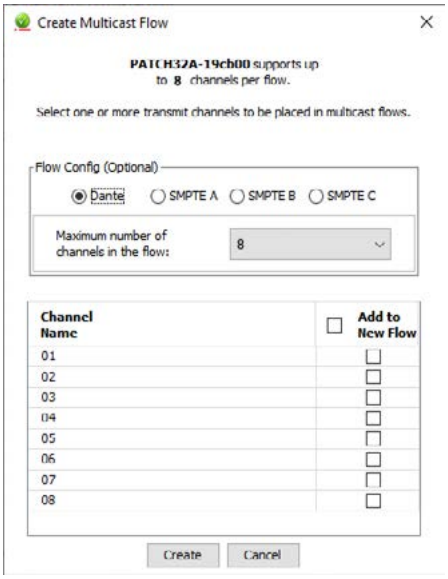
From Device View window, select Create New Multicast Flow as shown lower right.



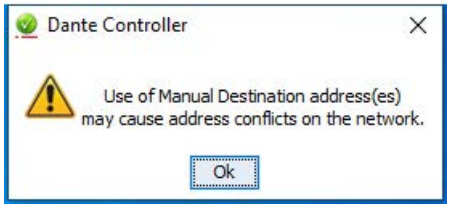
SMPTE mode supports 3 flow configurations:

- SMPTE A: 48kHz flow with up to 8 channels at packet times of 1ms
- SMPTE B: 48kHz flow with up to 8 channels at packet times of 125µs
- SMPTE C: 48kHz flow with 64 channels at packet times of 125µs

Select 'SMPTE A' and 'Manual' to specify the address port and enter the number of channel as shown below right.



Dismiss the warning popup concerning manual destinations as shown below.

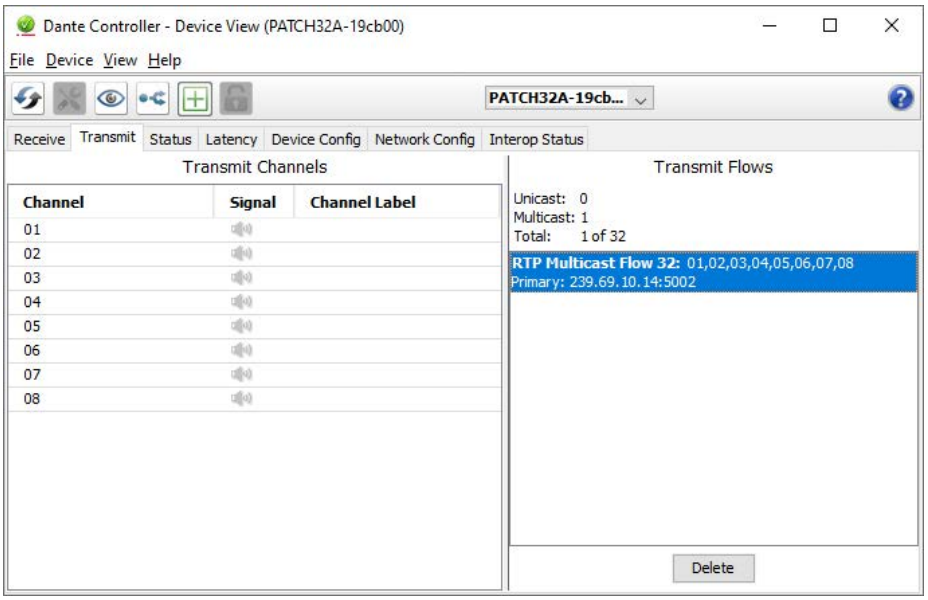


The IP address must be in the range 239.0.0.0 to 239.254.254.254.

While in DDM earlier, we set the 2nd octet to be 69 and the port to 5002.

This example uses an address of 239.69.10.14, port 5002. Select how many channels desired. In this case 8. After successfully creating the flow, confirmation will appear in the right window of the page when looking at Device View, Transmit, Transmit Flows as shown above right..

DANTE CONTROLLER VICES>CREATE DANTE DEVICE



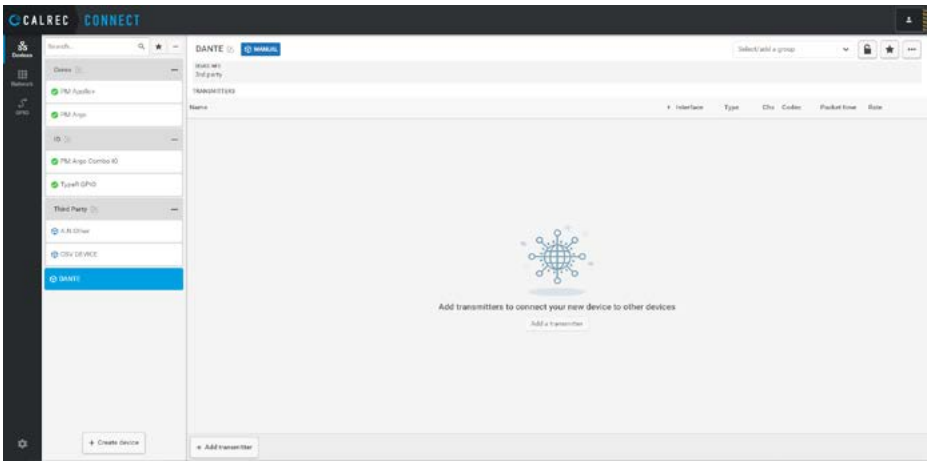
Create the DANTE device in Connect
As shown earlier in **"Create Devices" on page 43** the first step to connect to the DANTE interface is to create the DANTE device. This has been done as shown below right.

Modify Type R streams to match the DANTE device
Dante supports AES67 streams configured as follows:

Packet Time: 1mS
Depth: 24 bit (L24)
Sample Rate: 48kHz
Channels per stream: 1-8
Domain: 0
UDP Port: 5002

Type R supports SMPTE2110/AES67 streams as follows:

DEVICES>CREATE DANTE DEVICE



Packet Time: 125uS (default) or 250us/333us/1ms/4ms
Depth: 24 bit (L24)
Sample Rate: 48kHz
Channels per stream: 64/16/8
Domain: 127 (default)
UDP Port: 5000 or greater

The default Type R settings need to be changed to match the Dante Streams configuration as needed when creating transmitters and receivers.

5. Use Connect to set up Type R Devices to work with Dante devices

Create a DANTE AES67 Transmitter
Sending audio from DANTE to Type R or Impulse as a transmitter:

Connect does NOT see incoming DANTE streams, so a dummy device transmitter must be created in 'Connect' see above right to make it visible to the core, the transmission stream should be named and configured as an 8ch/48k/L24 with a 1ms Packet time.

The DANTE source is sent using the 239.69.xxx.yyy multicast address so in this case the transport IP addresses for A & B are 239.69.10.14, 239.69.11.14 and the UDP ports A & B should be set to 5002 and 5004 respectively as shown right.

Connect from DANTE Transmitter
Once configured the DANTE transmitter can be connected to other devices in the usual way by accessing the Network page in 'Connect' as shown below right.

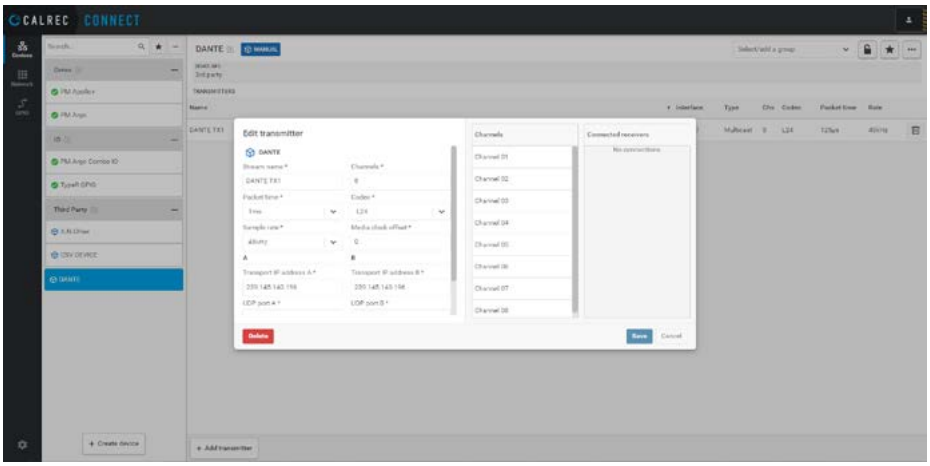
This allows the user to connect the DANTE audio stream being transmitted as
'DANTE TX1' to be connected to a receiver in the Calrec core which has been created called 'DANTE RX1' shown right.

It is important to note that the Calrec receiver has to be configured with a Packet size of 1ms to match the DANTE transmitter in order for it to connect.

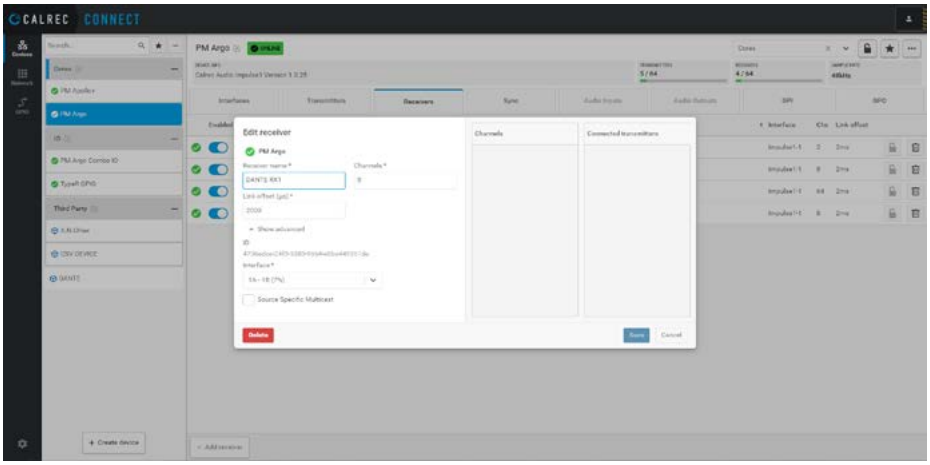
The image below right shows that some of the receivers are set to a different packet time i.e 125us and as such will not connect.

As many 3rd party devices only work at 1ms packet times, then the other devices in the system would need to be set to the same packet time for it to all work together.

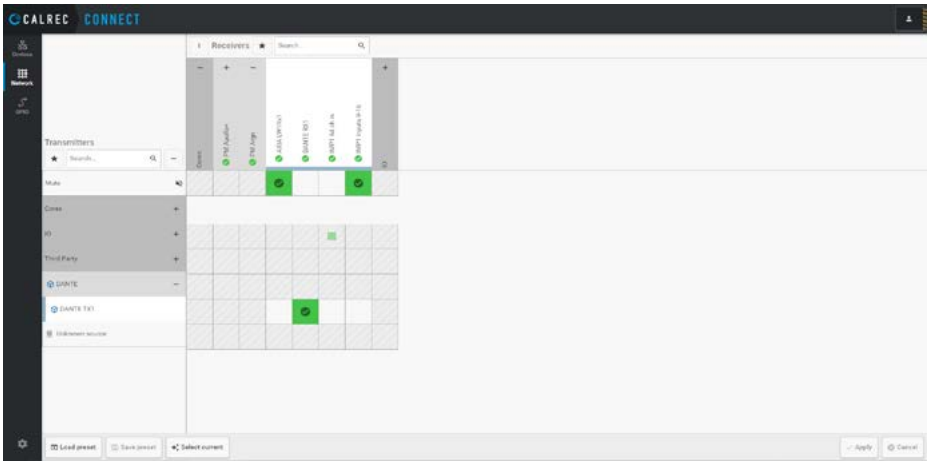
DEVICES>CREATE DANTE TRANSMITTER



DEVICES>CREATE CORE RECEIVER



NETWORK>DANTE TRANSMITTER CONNECT TO CORE RECEIVER

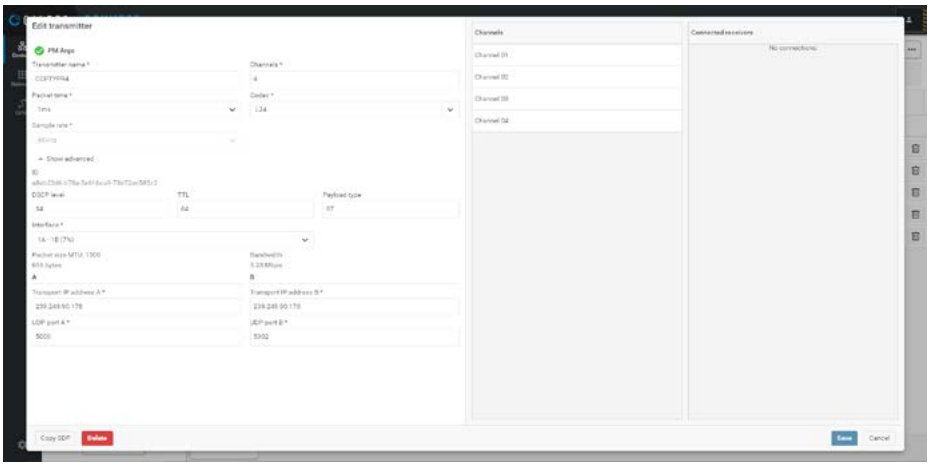


6. Use Connect and Dante Controller to receive channels from the core transmitter to the DANTE receiver.

Creating a DANTE AES67 Receiver
First Create a Transmitter stream from the Core in Connect, as shown above right with the following parameters:

Packet Time: 1mS
Depth: 24 bit (L24)
Sample Rate: 48kHz
Channels per stream: 1-8
Domain: 0
UDP Port: 5004/5006.
In this case the Multicast addresses are 239.69.10.20 and 239.69.11.20

DEVICES>CREATE CORE TRANSMITTER FOR DANTE RECEIVER



The Dante Controller DOES see the transmitters generated by the Type R as long as SAP is activated on the Covaloz BACH page on the AoIP modules in the Core.

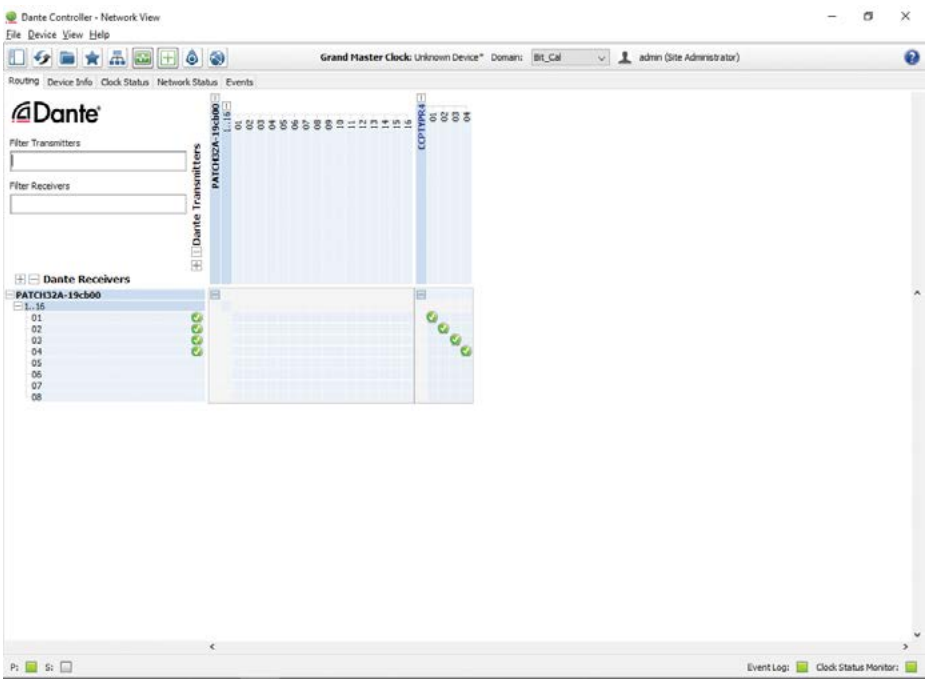
The Core Transmitter 'CCPTYPR4' appears in the Dante Controller Network view as shown below right and the four channels in that stream have been connected to a DANTE receiver labelled 'PATCH32A-19cb00' ready for use.

Using the Calrec Assist application, outputs from the Type R or Impulse Mixer can be patched to the 'CCPTYPR4' transmitter from the core sent to the DANTE device which receives those outputs and connects them to the DANTE receiver 'PATCH32A-19cb00' .

When everything is set right...

- The DANTE device and Type R both act as sync slaves to the PTPv2 clock from Type R (but the Dante device remains a PTP-Master in the PTPv1 domain.
- Type R or Impulse receives 8-channel 48k L24 streams from the Dante device.
- The Dante device receives 8-channel 48k L24 streams from the Type R or Impulse.

DANTE CONTROLLER>>PATCH CORE TRANSMITTERS TO DANTE RECEIVERS



GLOSSARY OF TERMINOLOGY

Audio stream
See RTP stream.

Audio Video Bridging
AVB
Describes enhanced Ethernet networks specified in IEEE 802.1BA, IEEE 802.1Q-2011 and IEEE 802.1AS.

Boundary Clock
A clock that has multiple Precision Time Protocol (PTP) ports in a domain and maintains the timescale used in the domain. It may serve as the source of time, that is, be a master clock; and may synchronize to another clock, that is, be a slave clock. See IEEE 1588-2008.

Byte
A unit comprising 8 bits of data. Over IP networks, data is transported in units of bytes.

Digital Audio Reference Signal
DARS
An audio clock signal defined in AES11.

CSRC
The contributing source (CSRC) is the source of a stream of RTP packets that has contributed to the combined stream produced by an RTP mixer.

DiffServ
Differentiated services (DiffServ) is a system for classifying traffic and providing quality of service (QoS) on an IP network.

DSCP
The differentiated services code point (DSCP) is a 6-bit field in the IP packet header that is used for classification purposes. DSCP is part of the differentiated services architecture.

End-to-end Transparent Clock
A transparent clock that supports the use of the end-to-end delay measurement mechanism between slave clocks and the master clock. See IEEE 1588-2008.

Ethernet
Ethernet is a physical & data link layer set of computer networking technologies for local area networks (LANs). Ethernet uses a bus or star topology and supports data transfer rates from 10 Mbps through 100 Mbps (Fast Ethernet) onto Gigabit Ethernet, supporting data rates of 1 gigabit per second or 1000 Mbps.

EUI-64
A 64-bit globally unique identifier formed by combining a registered 24 or 36-bit company identifier and a company unique device identifier. The EUI-64 is similar to the EUI-48 which is used to assign Ethernet media access control (MAC) addresses.

Grandmaster identifier
GMID
An EUI-64 used in IEEE 1588 and IEEE 802.1AS synchronization standards to uniquely identify the grandmaster serving a synchronization domain.

Grandmaster
The master source of synchronization for clock distribution via PTP. The Grandmaster is a network device and is identified by an EUI-64.

IEEE
Institute of Electrical and Electronics Engineers is a professional association dedicated to advancing technological innovation and excellence. The IEEE publishes communications standards.

IETF
Internet Engineering Task Force is the volunteer standards-developing organization responsible for the Internet Protocol suite.

IGMP
Internet Group Management Protocol (IGMP) is a communications protocol used by hosts to report their multicast group memberships to IPv4 routers.

Internet Protocol IP
The network layer protocol commonly used to transport data on networks built through interconnection of one or more local-area networks.

IPv4
Internet Protocol version 4 is the most widely deployed version of the Internet Protocol and is widely used on the Internet and on local area networks (LANs).

IPv6
Internet Protocol version 6 is the most recent revision of the Internet Protocol and is intended to replace IPv4 eventually.

Link offset
Link offset specifies the amount of time media spends on the network and in buffers at the sender and receiver. Link offset is also known as network latency or playout delay.

Multicast DNS (mDNS)
This protocol resolves hostnames to IP addresses within small networks that do not include a local name server. It is a zero-configuration service, using essentially the same programming interfaces, packet formats and operating semantics as the unicast Domain Name System

Media clock
The clock used by senders to sample and receivers to play digital media streams. The media clock for audio streams reads in units of samples.

Media packet
One of the data packets carrying media data as part of a media stream. A media packet contains one or more samples for one or more audio channels.

Media stream
See RTP stream.

Maximum transmission unit
MTU
The size of the IP packet, measured in bytes, that can be transferred using a specific data link connection. The MTU for an Ethernet data link is 1500 bytes.

Network clock
The time delivered by the network synchronization mechanism. The network clock reads in units of seconds.

Network layer
The network layer is layer 3 of the OSI model and is responsible for packet forwarding and routing of variable length data sequences from a source to a destination.

OSI model
The Open Systems Interconnect Model characterizes and standardizes the functions of a communications system in terms of abstraction layers.

Packet time
The real-time duration of the media data contained in a media packet. For example, a packet containing 12 samples of 48 kHz audio has a packet time of 12 ÷ 48 kHz = 250 microseconds.

Peer-to-peer Transparent Clock
A transparent clock that, in addition to providing Precision Time Protocol (PTP) event transit time information, also provides corrections for the propagation delay of the link connected to the port receiving the PTP event message. In the presence of peer-to-peer transparent clocks, delay measurements between slave clocks and the master clock are performed using the peer-to-peer delay measurement mechanism.

Precision time protocol
PTP
The general class clock distribution protocol standardized in IEEE 1588-2002, IEEE 1588-2008 & IEEE 802.1AS-2011.

Quality of service
QoS
Describes a system for classifying, marking and delivering traffic across a network in accordance with its performance requirements.

Receiver
A network device with ability to receive at least one media stream from the network.

Request for Comment
RFC
Request for Comments are documents published by the IETF relevant for the working of the Internet and Internet-connected systems. RFCs are referenced by number. RFC 791, for example, defines the Internet Protocol version 4 (IPv4).

RTCP
A companion protocol of the Real-time Transport Protocol (RTP), providing statistics and control information for RTP media packets.

Real-time Transport Protocol
RTP
Is defined in RFC 3550 and provides a means for applications to organize, mark and transport their media packets using UDP/IP networking.

RTP clock
Timestamps are carried in RTP packets containing stream data. Each stream has its own RTP clock. There is a constant offset between the media clock and the RTP clock.

RTP session
An RTP session is a media connection between sender and receiver. RTP sessions may be unicast or multicast. In teleconferencing RTP applications, multicast sessions may have multiple senders and receivers. However, under this standard, a session is allowed only one sender.

RTP stream
An RTP stream is a sequence of RTP packets with media data sent at regular interval. A stream may contain multiple channels. There may be multiple media streams per RTP session.

SAP
Session Announcement Protocol is a protocol for advertising multicast session information. SAP typically uses Session Description Protocol as the format for Real-time Transport Protocol session descriptions. Announcement data is sent using IP multicast and the User Datagram Protocol Session Description Protocol

SDP
A format for describing RTP sessions and their parameters including network addressing, encoding format and other metadata. SDP is defined in RFC 4566.

Sender
A network device with ability to source at least one media stream onto the network.

Session
See RTP session.

Session Initiation Protocol
SIP
A telecommunications connection management protocol defined in RFC 3261.

SIP URI
A SIP URI is a URI used by SIP to identify user agents. SIP URI take the form sip:<user>@<domain> or sips:<user>@<domain>.

Slave Clock
A clock that is synchronized to a master clock (the provider of time) within an environment that uses the Precision Time Protocol (PTP). A slave may, in turn, be a master to another clock and may simultaneously be a boundary clock.

Stream
See RTP stream.

Transmission Control Protocol/ Internet Protocol TCP/IP
See Internet Protocol.

Transport Layer Security
TLS
A cryptographic protocol for secure communication over IP networks.

Transparent clock
A device that measures the time taken for a Precision Time Protocol (PTP) event message to transit the device and provides this information to clocks receiving this PTP event message. See IEEE 1588-2008. See also: end-to-end transparent clock; peer-to-peer transparent clock.

Transport layer
The network layer is layer 4 of the OSI model and provides end-to-end communication services for network applications.

User datagram protocol
UDP
Constitutes a simple transport layer for the IP network layer. Defined in RFC 768.

Uniform resource identifier
URI
An identifier for a network resource. An identification URI enables interaction with the resource over a network.

User agent
A SIP endpoint device such as a VoIP telephone.

Virtual LAN
VLAN
A single layer-2 network may be partitioned to create multiple distinct broadcast domains, which are mutually isolated so that packets can only pass between them via one or more routers, such a domain is referred to as a Virtual Local Area Network.

PRIMARY CORE – ROUTER INTERFACES – SUGGESTED SETTINGS

The suggested default settings for the Primary Impulse Core Router Interfaces are shown below. Use the tables below to configure your AoIP Router ports and to record the serial number and label details of each router port for later reference. After configuring the AoIP Router interfaces, you may want to attach labels to them. A blank label sheet is provided with the Impulse core for this purpose. A printer template for this Label sheet is available to distributors on the Calrec Website.

Primary Core Router I/F Number	Primary Core Router Port Label	Primary Core Router Module Serial number	Primary interface IP Address	Subnet mask*	Secondary interface IP Address	Subnet mask**
1	Router #1-1A/1B		192.168.30.111	/24	192.168.31.111	/24
2	Router #1-2A/2B		192.168.30.112	/24	192.168.31.112	/24
3	Router #1-3A/3B		192.168.30.113	/24	192.168.31.113	/24
4	Router #1-4A/4B		192.168.30.114	/24	192.168.31.114	/24
5	Router #2-1A/1B		192.168.30.121	/24	192.168.31.121	/24
6	Router #2-2A/2B		192.168.30.122	/24	192.168.31.122	/24
7	Router #2-3A/3B		192.168.30.123	/24	192.168.31.123	/24
8	Router #2-4A/4B		192.168.30.124	/24	192.168.31.124	/24
9	Router #3-1A/1B		192.168.30.131	/24	192.168.31.131	/24
10	Router #3-2A/2B		192.168.30.132	/24	192.168.31.132	/24
11	Router #3-3A/3B		192.168.30.133	/24	192.168.31.133	/24
12	Router #3-4A/4B		192.168.30.134	/24	192.168.31.134	/24
13	Router #4-1A/1B		192.168.30.141	/24	192.168.31.141	/24
14	Router #4-2A/2B		192.168.30.142	/24	192.168.31.142	/24
15	Router #4-3A/3B		192.168.30.143	/24	192.168.31.143	/24
16	Router #4-4A/4B		192.168.30.144	/24	192.168.31.144	/24

**Subnet masks are displayed in CIDR notation

The entries in GREEN are the defaults for a Redundant Impulse System with the minimum of 1 Router Module Installed

SECONDARY CORE – ROUTER INTERFACES – SUGGESTED SETTINGS

The suggested default settings for the Secondary Impulse Core Router Interfaces are shown below. Use the tables below to configure your AoIP Router ports and to record the serial number and label details of each router port for later reference. After configuring the AoIP Router interfaces, you may want to attach labels to them. A blank label sheet is provided with the Impulse core for this purpose. A printer template for this Label sheet is available to distributors on the Calrec Website.

Secondary Core Router Interface Number	Secondary Core Router Port Label	Secondary Core Router Module Serial number	Primary interface IP Address	Subnet mask*	Secondary interface IP Address	Subnet mask**
1	Router #1-1A/1B		192.168.30.211	/24	192.168.31.211	/24
2	Router #1-2A/2B		192.168.30.212	/24	192.168.31.212	/24
3	Router #1-3A/3B		192.168.30.213	/24	192.168.31.213	/24
4	Router #1-4A/4B		192.168.30.214	/24	192.168.31.214	/24
5	Router #2-1A/1B		192.168.30.221	/24	192.168.31.221	/24
6	Router #2-2A/2B		192.168.30.222	/24	192.168.31.222	/24
7	Router #2-3A/3B		192.168.30.223	/24	192.168.31.223	/24
8	Router #2-4A/4B		192.168.30.224	/24	192.168.31.224	/24
9	Router #3-1A/1B		192.168.30.231	/24	192.168.31.231	/24
10	Router #3-2A/2B		192.168.30.232	/24	192.168.31.232	/24
11	Router #3-3A/3B		192.168.30.233	/24	192.168.31.233	/24
12	Router #3-4A/4B		192.168.30.234	/24	192.168.31.234	/24
13	Router #4-1A/1B		192.168.30.241	/24	192.168.31.241	/24
14	Router #4-2A/2B		192.168.30.242	/24	192.168.31.242	/24
15	Router #4-3A/3B		192.168.30.243	/24	192.168.31.243	/24
16	Router #4-4A/4B		192.168.30.244	/24	192.168.31.244	/24

**Subnet masks are displayed in CIDR notation

The entries in GREEN are the defaults for a Redundant Impulse System with the minimum of 1 Router Module Installed

AOIP DEVICES – NETWORK INTERFACES – SUGGESTED SETTINGS

The suggested default settings for the first 20 AoIP devices attached to a given Impulse or Type R system are shown below. Use the tables below to configure your AoIP devices and to record the serial number and label details of each device for later reference. Note that MOD I/O boxes can have 2 AoIP network interface connections which would take up 2 box entries. After configuring the AoIP network interfaces, you may want to attach labels to them. A blank label sheet is provided with the core for this purpose. A printer template for this Label sheet is available to distributors on the Calrec Website.

Box No	IO box label	Serial number	Primary interface IP Address	Subnet mask*	Secondary interface IP Address	Subnet mask**
1	Mod I/O #1-1A/1B		192.168.30.11	/24	192.168.31.11	/24
2	Mod I/O #1 2A/2B		192.168.30.12	/24	192.168.31.12	/24
3	Mod I/O #2-1A/1B		192.168.30.13	/24	192.168.31.13	/24
4	AoIP Box Label		192.168.30.14	/24	192.168.31.14	/24
5	Mod I/O #3-1A/1B		192.168.30.15	/24	192.168.31.15	/24
6	AoIP Box Label		192.168.30.16	/24	192.168.31.16	/24
7	AoIP Box Label		192.168.30.17	/24	192.168.31.17	/24
8	AoIP Box Label		192.168.30.18	/24	192.168.31.18	/24
9	AoIP Box Label		192.168.30.19	/24	192.168.31.19	/24
10	AoIP Box Label		192.168.30.20	/24	192.168.31.20	/24
11	AoIP Box Label		192.168.30.21	/24	192.168.31.21	/24
12	AoIP Box Label		192.168.30.22	/24	192.168.31.22	/24
13	AoIP Box Label		192.168.30.23	/24	192.168.31.23	/24
14	AoIP Box Label		192.168.30.24	/24	192.168.31.24	/24
15	AoIP Box Label		192.168.30.25	/24	192.168.31.25	/24
16	AoIP Box Label		192.168.30.26	/24	192.168.31.26	/24
17	AoIP Box Label		192.168.30.27	/24	192.168.31.27	/24
18	AoIP Box Label		192.168.30.28	/24	192.168.31.28	/24
19	AoIP Box Label		192.168.30.29	/24	192.168.31.29	/24
20	AoIP Box Label		192.168.30.30	/24	192.168.31.30	/24

**Subnet masks are displayed in CIDR notation

The Mod I/O entries above are examples from an Artemis+ Impulse System with 3 Mod I/O AoIP boxes

TYPE R CORE & IO BOX AOIP DEVICES – NETWORK INTERFACE SETTINGS

The suggested default settings below are for the Type R Core Interfaces and the basic IO Box AoIP devices attached to a given Type R system.

Box Type / Core	AoIP port	IP Address	Subnet mask*
Primary Core Connect Server	**Pri (I/F 5)**	**192.168.30.100**	**/24**
	Sec (I/F 6)	**192.168.31.100**	**/24**
Primary Core AoIP 1	**AoIP 1 - Pri**	**192.168.30.4**	**/24**
	AoIP 1 - Sec	**192.168.31.4**	**/24**
Primary Core AoIP 2	**AoIP 2 - Pri**	**192.168.30.64**	**/24**
	AoIP 2 - Sec	**192.168.31.64**	**/24**
Combo I/O box	**AoIP - Pri**	**192.168.30.1***	**/24**
	AoIP - Sec	**192.168.31.1***	**/24**
Analogue I/O Box	**AoIP - Pri**	**192.168.30.2***	**/24**
	AoIP - Sec	**192.168.31.2***	**/24**
AES I/O Box	**AoIP - Pri**	**192.168.30.3***	**/24**
	AoIP - Sec	**192.168.31.3***	**/24**

**Subnet masks are displayed in CIDR notation

* Note: whilst the IP Addresses for connect and the Core Router settings are the recommended system settings, the Combo, Analogue and AES boxes were set to these basic addresses for simplicity before Impulse was created.

It may be more appropriate to use the suggested setting shown on the previous page, the important thing is that whatever IP addresses are used they must be unique to each device. Type-R can now also use AoIP based Mod IO boxes, which can have 2 AoIP network interface connections, i.e. 2 Primary and 2 Secondary connections via the UJ6429 AoIP interface card.

FURTHER READING – TYPE R

Type R has a number of Manuals associated with it. This is the Connect Application Guide:-

1. Type R Product Info Sheet (926-272)
This information sheet shows how to collect information on Type R.

2. Type R Start Up Guide (926-282)
This guide shows how to Unpack, Power Up and access/configure the system core, Install a licence key, Configure the Core for the Surface panel layouts, Access Assist and create a New Show, Configure the network devices, Configure Panel ID's, Connect up a PC to the Surface switch, Upload control layouts to the Soft Panels, Create layouts using the Type R Soft Panel Designer, Configure and Connect an Audio Switch and AoIP Devices to the Core.

3. Type R Installation Manual (926-228)
This contains a number of chapters including: Control surface measurements, Mounting and Assembly instructions, Defining the system elements of a Type R system and describes the Core, Panels and IO Box layouts, Synchronisation, Audio & GPIO Connections, Core DSP packs, Surface panel layout examples, AoIP network examples, Software Updating and Technical specifications.

4. Type R Assist Manual (926-229)
This defines how a pre-configured Type R console is setup and controlled via Calrec Assist, which is Calrec's web-based configuration tool. It includes creating/managing shows, setting up shows in terms of configuring paths, displaying and controlling the fader surface, saving and loading snapshots and patching inputs and outputs to the channels and buses. There are then various sections about parameter access including:- processing, routing, configuring and controlling the buses & outputs and setting up the monitoring & metering. The show setup and system settings sections provide configuration tools for both show and system configuration.

5. Type R Assist and Console Panel Operation (Merged with Type R Assist Manual now both in 926-283)
This covers how the pre-configured Type R Console is operated via a combination of Fader Panels, Large and Small 'Soft' surface panels, incorporated with the Assist manual

6. Type R Panel Designer (926-284)
This defines how different controls can be configured to appear on the Console Panels using the Panel Designer Application which allows the user to design their own custom layouts for the Large and Small 'Soft' surface panels.

7. Type R Configure Guide (926-285)
This defines how the Type R Core(s) can be configured and partitioned into different mixing consoles with varying amounts of DSP processing channels available in different 'Pack' sizes under licence. It also is used to configure Control Surfaces from the available panels, setup synchronisation sources, configure Network AoIP interface, configure the Core I/O including Virtual Patchbays and provide User management permissions for Operators.

8. Connect (926-292)
This defines how the Impulse/Type R Core IP Input and Output streams are connected to AoIP based interfaces and how the AoIP streams are managed including GPIO devices. These can be connections to and from either Calrec AoIP Devices or other 3rd party AoIP streams.

9. AoIP I/O Manual (926-293)
This contains information about AoIP devices available for use with Impulse/Type R in terms of Control, Audio & GPIO Connections.

FURTHER READING – IMPULSE

Impulse has a number of Manuals associated with it. This is the Connect Application Guide:-

1. Impulse Product Info Sheet (926-294)
This information sheet shows how to collect information on Impulse.

2. Impulse Start Up Guide (926-291)
This guide shows how to Power Up and Access/Configure the Impulse core, Configure the Surface IP connections, Connect the Surface to the Impulse Cores, Power Up the Surface & Create a New Show, Configure Network Switches & Devices, Access the Configure/Connect/Software Updater/Assist* applications, Update the Core Software to the latest version (optional), Configure AoIP Router & AoIP Device IP addresses, Connect Audio Switches & AoIP Devices to the Core and Examine an example system.

3. Impulse Installation Manual (926-288) * Updated for Argo S & Argo Q
This contains a number of chapters including an overview of the Impulse system, Defining the system elements of an Impulse core, Core DSP pack options, Synchronisation, Surface Connections, AoIP network connections, Redundancy, AoIP network examples, External Control connections and Technical specifications.

4. Impulse Configure Application Guide (926-290) * Updated for Argo S & Argo Q
This defines how Impulse system Core(s) can be configured and partitioned into different mixing surfaces with varying amounts of DSP processing channels available in different 'Pack' sizes under licence. It provides guidance on updating the system software, backing up and restoring user data, setting the sample rate, controlling the application containers that run the system and provide maintenance logs. It's also used to configure the IP addresses of the Network Interface Controllers for the application containers, manage the Remote Network interfaces such as the RP1, AoIP interfaces for the Audio Routers, PTP interfaces for synchronisation, setting up synchronisation sources, Core I/O Virtual Patchbays and I/O Status.

5. Connect Application Guide (926-292)
This defines how the Impulse/Type R Core IP Input and Output streams are connected to AoIP based interfaces and how the AoIP streams are managed including GPIO devices. These can be connections to and from either Calrec AoIP Devices or other 3rd party AoIP streams.

6. AoIP I/O Manual (926-293)
This contains information about AoIP devices available for use with Impulse/Type R in terms of Control, Audio & GPIO Connections.

7. Apollo+ Installation Manual (926-295)
This contains information about the installation and setup of the Apollo+ surface for use with Impulse systems.

8. Apollo+ Operator Manual (926-296)
This defines how an installed Apollo+ console is configured and controlled via its surface and console PC. It includes creating/managing shows, setting up shows in terms of configuring paths, displaying and controlling the fader surface, saving and loading snapshots and patching inputs and outputs to the channels and buses. There are then various sections about parameter access including:- processing, routing, configuring and controlling the buses & outputs and setting up the monitoring & metering.
The show setup and system settings sections provide configuration tools for both show and system configuration.

9. Artemis+ Installation Manual (926-297)
This contains information about the installation and setup of the Artemis+ surface for use with Impulse systems.

10. Artemis+ Operator Manual (926-298)
This defines how an installed Artemis+ console is configured and controlled via its surface and console PC. It includes creating/managing shows, setting up shows in terms of configuring paths, displaying and controlling the fader surface, saving and loading snapshots and patching inputs and outputs to the channels and buses. There are then various sections about parameter access including:- processing, routing, configuring and controlling the buses & outputs and setting up the monitoring & metering. The show setup and system settings sections provide configuration tools for both show and system configuration.

11. Apollo+/Artemis+ Assist Manual (926-307)
This defines how an Apollo+ or Artemis+ with or without a physical console is setup and controlled via Calrec Assist, which is Calrec's web-based user operation tool.

New Argo Product Manual Additions

12. Impulse - Argo Product Info Sheet (926-320)
This information sheet shows how to collect information on Impulse - Argo.

13. Impulse - Argo Start Up Guide (926-321)
This guide shows how to Power Up and Access/Configure the Impulse core, Configure the Argo Surface IP connections, Connect the Argo Surface to the Impulse Cores, Power Up the Argo Surface & Create a New Show, Configure Network Switches & Devices, Access the Configure/Connect/Software Updater/Assist applications, Update the Core Software to the latest version (optional), Configure AoIP Router & AoIP Device IP addresses, Connect Audio Switches & AoIP Devices to the Core and Examine an example system.

14. Argo Installation Manual (926-312)
This contains technical information about the configuration, installation and setup of the Argo surface for use with Impulse systems.

15. Argo Operator Manual (926-313)
This defines how an installed Argo console is configured and controlled via its surface. It includes creating/managing shows, setting up shows in terms of configuring paths, displaying and controlling the fader surface, saving and loading snapshots and patching inputs and outputs to the channels and buses. There are then various sections about parameter access including:- processing, routing, configuring and controlling the buses & outputs and setting up the monitoring & metering. The show setup and system settings sections provide configuration tools for both show and system configuration.

16. Argo Assist Manual (926-317)
This defines how an Argo with or without a physical console is setup and controlled via Calrec Assist, which is Calrec's web-based user operation tool.

USER NOTES

Calrec Audio Ltd
Nutclough Mill
Hebden Bridge
West Yorkshire
England UK
HX7 8EZ

Tel +44 (0)1422 842159
Fax +44 (0)1422 845244
Email Enquiries@calrec.com

calrec.com

(926-292 Iss.11)