AoIP User Manual

AES67 / ST 2110-30 / ST 2022-7
IP Streaming Configuration

Software Version: 3.08.xx
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1 AES67 Introduction

About AES67

AES67 is a technical standard developed by the Audio Engineering Society (AES) with interoperability guidelines for Audio-over-IP (AoIP). It is supported by major AoIP vendors like Wheatstone, Telos, Ravenna, Dante and others, and supports the streaming of audio between these different systems. Tieline codecs are used to transport audio between AES67 compatible networks and studios and using AES67 inputs and outputs facilitates inter-studio routing of high quality AoIP.

The AES67 standard defines requirements for synchronizing clocks using IEEE 1588-2008 Precision Time Protocol (PTPv2). It also defines setting of QoS media traffic priorities and initiating media streams using standard internet protocols. The standard also defines the audio sample format and sample rate, the number of supported channels, plus latency, buffering and the size of IP data packets. It does not define device discovery, connection management and media configuration protocols.

Introduction

Tieline Gateway and Gateway codecs are AES67 compliant and therefore capable of connecting to AoIP networks such as:

1. Wheatstone WheatNet-IP using optional WheatNet-IP card installed at purchase.
2. Wheatstone in AES67 compatibility mode.
3. Axia Livewire+
4. RAVENNA
5. Dante using an optional Dante card installed at purchase.
6. Dante in AES67 compatibility mode.

Other G6 codecs such as the MPX I, MPX II, Bridge-IT II and Bridge-IT XTRA II natively support RAVENNA and Livewire+, but do not provide options to install WheatNet-IP or Dante cards, and therefore must connect to these AoIP networks using AES67 compatibility mode if required.

AES67 compatibility makes it simple for different devices to use common configuration settings and stream audio between them. Similar to how different brands of codecs are configured to connect using SIP for interoperability, AES67 compliant audio products can stream audio between each other using common streaming configuration parameters. The codec is also compliant with ST 2110-30 for system timing/synchronization and PCM digital audio transport. This allows the codec to operate in live production, playout and other media applications, where real-time audio, video and ancillary data needs to remain independent yet co-timed, allowing independent streams to be synchronized. The SMPTE ST 2110 standards are based on AES67 so can work together. The configurable elements include:

1. Audio Sampling Rate
2. Audio Sampling Depth.
3. Packet Time.
4. Number of Channels streamed.

In addition, the SDP values defined in ST 2110-30 provide standard order for individual channels within the IP packets. E.g. for stereo and surround sound channels. Plus, ST 2110-30 also defines a set of six compliance levels for devices receiving audio streams. The codec supports A, AX, B and BX.

Level	1 millisecond Packet Time		125 microseconds Packet Time	
	48 kHz	96 kHz	48 kHz	96 kHz
A	1 to 8			
AX	1 to 8	1 to 4		
B	1 to 8		1 to 8	
BX	1 to 8	1 to 4	1 to 8	1 to 8
C	1 to 8		1 to 64	
CX	1 to 8	1 to 4	1 to 64	1 to 32

Note: Only the **AoIP 1** port on the Gateway and Gateway 4 codec rear panel can be used to connect to WheatNet-IP networks. All AoIP configuration settings are configured in the **AoIP** menu in the HTML5 Toolbox Web-GUI.

1.1 Glossary of Terms

AES67	A technical standard for audio over IP and audio over Ethernet (AoE) interoperability developed by the AES. It is a layer 3 protocol suite facilitating interoperability between IP-based audio networking systems. Proprietary AoIP systems like WheatNet-IP, RAVENNA, Livewire, and Dante offer compatibility modes for interoperability using AES67 standards.
API	An Application Programming Interface (API) is a software intermediary allowing two applications to talk to each other.
Dante	Dante is a proprietary AoIP system developed by Audinate which also supports AES67 streaming.
Destination	An AoIP Destination is an endpoint for receiving an AES67 or ST 2110-30 stream.
DSCP	The Differentiated Services Code Point or Diffserv Value is a field in an IP packet header for prioritizing data when traversing IP networks. This is often used in AES67 streaming.
Ember+	An open standard control protocol developed by Lawo which allows a third party application to gain access to device parameters.
Flow	The term Dante uses to describe an audio stream over an AoIP network. E.g. Source flow.
Follower	Devices that are synced to an elected PTP Leader Clock over an AoIP network.
GPIO	A General Purpose Input/Output is a customizable pin on an integrated circuit that can be controlled by software. A GPI is a General Purpose Input and a GPO is a General Purpose Output.
IGMP	A communications protocol used by hosts and adjacent routers on IPv4 networks to establish multicast group mships to IPv4 routers.
IGMP snooping	The process of listening to IGMP network traffic for delivery of IP multicasts. Network switches with IGMP snooping maintain a map of which links need which IP multicast transmission. Multicasts may be filtered to conserve bandwidth on links.
Livewire+ AES67	Livewire+ is a proprietary protocol from the Telos Alliance which is the second-generation of Livewire. It facilitates AES67 devices connecting directly to Livewire+ AES67 networks to exchange audio streams.
Multicast	Efficient one to many streaming of IP audio using multicast IP addressing.
NMOS	Networked Media Open Specifications (NMOS) delivering Discovery and Registration to ensure that parts of a networked media system can find each other. NMOS also provides connection management and audio channel mapping to device I/O channels.
Primary Leader Clock	The primary source of synchronization for clock distribution via PTP.
PTP	The general class clock distribution protocol standardized in IEEE 1588-2002, IEEE 1588-2008 and IEEE 802.1AS-2011. PTP syncing requires a leader clock source (often an installed PTP primary leader clocking device) with clocking replicated on synced devices.
RAVENNA	RAVENNA is an open technology developed by ALC NetworX for discovery and control. It is AES67 and SMPTE ST 2110 compliant.
Redundant Streaming (SMPTE ST2022-7)	Redundant AoIP streaming, also known as hitless merge or Seamless Protection Switching, is a redundancy mechanism whereby two streams of identical RTP packets are streamed over diverse paths. The objective is to seamlessly reconstruct a single perfectly reconstructed output stream based on the receipt of two potentially network impaired input streams.
SAP	SAP (Session Announcement Protocol) is a discovery protocol used by Dante to distribute SDP descriptions to receivers, enabling simplified

	connection management for multicast streaming. Settings are also available in other devices such as Livewire xNodes.
Signal	In WheatNet-IP a Signal is what defines a Source or Destination
Source	An AoIP Source in the codec is an AES67 stream sent by the codec to an AES67 LAN.
Time To Live (TTL)	TTL is the setting used in muliticast servers to ensure data packets have a finite life and don't cause congestion over networks. Each time a packet passes through a router it reduces by 1 until it reaches zero, at which point a router will no longer pass the packet.
WheatNet-IP	Proprietary IP protocol and networking system enabling AoIP audio streams to be intelligently distributed to devices across scalable networks. Compatibility mode also available for AES67 device interoperability.

1.2 About AES67 / ST 2110

About AES67 Streams

The AES67 interoperability standards allow IP audio streaming to multiple devices simultaneously and efficiently. The standard specifies essential configuration parameters and each manufacturer has it's own mechanism to manage and configure AES67 settings. For example, Tieline uses the Toolbox Web-GUI and Wheatstone uses Navigator software to configure WheatNet-IP AES67 settings.

Session Data Information

To connect an AES67 audio stream a node requires session information for stream connection management. The AES67 and ST 2110-30 standards support Session Description Protocol (SDP) for configuring the number of audio channels per stream, encoding format, bits per sample, sampling frequency and number of samples in a packet. Session discovery is not included in the standard.

The AES67 standard provides a way for devices transmitting streams to encapsulate all the multicast address, IP address, port, packet timing and format information about a particular stream using an SDP file. This file provides information in a standardized way and is supported by Tieline codecs. SDP information can be generated by WheatNet-IP Blades and Telos xNodes and simplifies the process of configuring AoIP streams by copying data from one device into another.

Multicasting

A unique multicast destination address is used to identify a stream over an AoIP network. Multicast routers replicate packets and send them to network devices that want to receive them. There is a large number of multicast IP addresses available and Tieline 'Source' multicast addresses use the range 239.3.3.0 to 239.3.3.7 to send 8 streams by default.

Normally the devices streaming audio to a Tieline codec will provide a destination multicast address to enter into the codec for each stream. By default, Tieline codec destination streams are populated with multicast addresses in the range 239.33.1.0 to 239.33.1.15. It is also possible to receive a stream on multiple Destinations using the same multicast IP address.

It is important to create a multicast address plan to allow simple identification and management of multiple streams. In a studio there will likely be many AES67 devices and it is very important to document which devices are transmitting and receiving streams from which addresses. Different manufacturers and devices use different auto-assigned multicast addresses. For example, Wheatnet uses a second octet of .192 in multicast addresses, so they appear as 239.192.xxx.xxx.

Important Note: The default destination multicast address will normally be configured by the device streaming audio to the Tieline codec. Verify this address with the network IT administrator and then place the correct address into the **Address** text box in the **AES67 Destinations** panel.

Unicasting

In the AES67 standard unicasting is supported using SIP for connection management as defined in RFC3261. SDP is used to configure information about the stream, including the encoding format, bits per sample, sampling frequency and number of samples in a packet. Note: Not supported in current firmware release.

IGMP

IGMP is a communications protocol used by codecs and other hosts and adjacent routers on IPv4 networks to establish multicast group memberships to IPv4 routers. IGMP support allows network filtering of undesired multicasts. The codec supports both IGMPv2 and v3. The default is IGMPv3. Select IGMPv2 if an AES67 network requires this configuration. IGMPv3 should be selected for ST 2110-30 network compatibility. IGMP snooping is the process of listening to IGMP network traffic for delivery of IP multicasts. Network switches supporting IGMP snooping maintain a map of which nodes require IP multicast transmissions. In this way multicasts can be filtered across networks to conserve bandwidth on links between nodes.

Ports

The AES67 standard recommends using port 5004 as the default port for RTP data streams. Port 5005 is recommended as the default RTCP port. Devices sending and receiving data may use other ports from 1024 to 65535 as well.

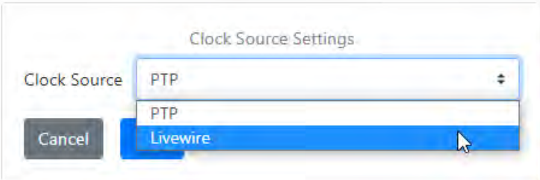
Routers

Managed switches are normally used and recommended for AES67 network routing of audio streams. These switches allow QoS DSCP/DiffServ packet prioritization and can usually be configured using a web interface. DiffServ is a framework for classification and differentiated treatment of network traffic. DiffServ operates on classes of traffic rather than on individual traffic flows. Per-hop behavior (PHB) is fundamental to and at each router or switch, traffic is classified and retransmitted according to that classification. When network congestion occurs, higher priority traffic is retransmitted promptly while lower priority traffic waits in buffers inside network equipment and may be discarded altogether.

1.3 Clocking

AoIP Clocking

AES67 and ST 2110-30 AoIP networks require devices to be synchronized to a common external clock source using IEEE 1588-2008 Precision Time Protocol (PTPv2). One device on a network will need to function as the 'Primary Leader' clock and all other devices will sync to this across the network. Most of the time a specialized PTP leader clock, synced to GPS for timing reference, is used to enable precise timing accuracy lower than 1 microsecond in many instances. Before enabling AES67 streaming it is necessary to ensure the codec is synced to a PTP leader clock source which is running. Most of the time the default settings will work fine. The codec can also act as the Primary Leader PTP clock source. Tieline recommends no more than 10 PTP followers are used in this mode. It is also possible to select Livewire as the AoIP clock source.

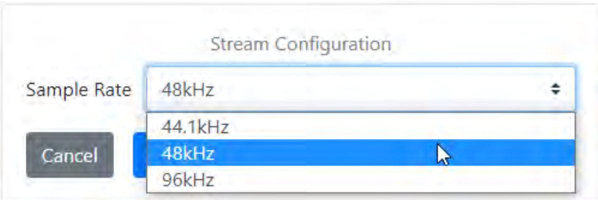


Tieline codecs support packet timing from 125 microseconds to 5 milliseconds. WheatNet-IP uses 250 microsecond packet timing by default and this is also supported by Telos xNodes and many other AES67 devices. At the time of writing Dante does not support this packet size and neither do Comrex AES67 codecs. The 1.25 millisecond (surround streams) frame size is unique to Livewire.

Sample Rates

The AES67 standard supports 44.1kHz, 48kHz and 96kHz sampling frequencies and most of the time AoIP systems run using 48kHz, 24 bit PCM digital audio. The rate of the media clock should also be the same as the sampling frequency. AES67 compatible devices should support:

- 16 bit, 44.1kHz sampling.
- 16 bit and 24 bit, 48kHz sampling.
- 24 bit, 96kHz sampling.



Important Note: It is not possible to enable Livewire as a service if the sample rate is not configured as 48kHz. If Livewire is running the sample rate cannot be changed from 48kHz.

PTP Settings

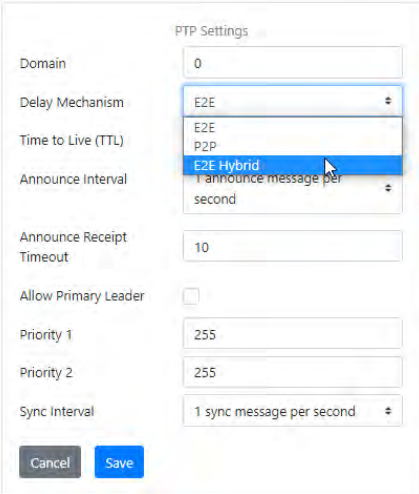
PTP Domain Number

A domain is an interacting set of clocks that synchronize to each other using PTP. Clocks are assigned to a domain using the **Domain** number and the default domain is **0** which is usually ok unless there is a requirement for multiple clock domains in a network. Adjust as required when on a network with multiple leader clocks in different domains.

PTP Delay Mechanism

PTP provides three different methods of calculating delay within a network: End-to-End (E2E), End-to-End Hybrid (E2E Hybrid), and Peer-to-Peer (P2P).

The default setting is usually E2E and this is automatically configured in the codec. If connected to a large PTP network without boundary clocks, End-to-End Hybrid mode may sometimes be preferred to support other AoIP devices that do not support delay request loads. This mode is more efficient with regard to PTP messages. It includes a mix of multicast and unicast PTP messages, to avoid follower clocks being bombarded by delay requests and responses.



Time to Live (TTL)

To avoid network congestion **Time to Live** configures the number of times a packet is allowed to pass through a multicast router before it will no longer pass the packet.

Announce Interval

The **Announce Interval** is configured by a PTP Leader and sets the announce message rate for a Precision Time Protocol (PTP) follower**.**

Announce Receipt Timeout

Use the **Announce Receipt Timeout** setting to configure the timeout value for receiving announce messages.

Allow Primary Leader

Select the **Allow Leader** check-box to set the codec as a PTP Leader clock across an AES67 AoIP network.

Priority 1 and 2

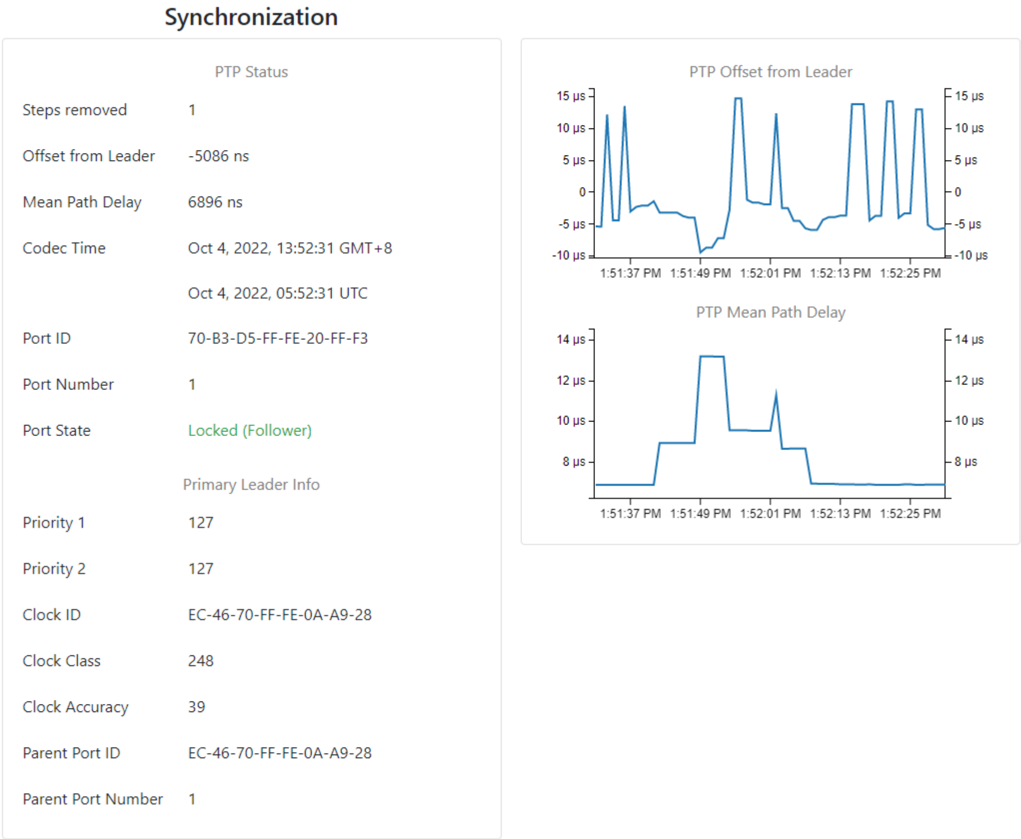
Priority 1 and **Priority 2** configure the user-assigned priority, ranging from 0-255, given to each PTP clock. The default setting is 255 which is appropriate for the codec as a PTP follower and this can be adjusted as required.

Sync Interval

The **Sync Interval** sets the mean interval to send synchronization messages across the AoIP network.

PTP status

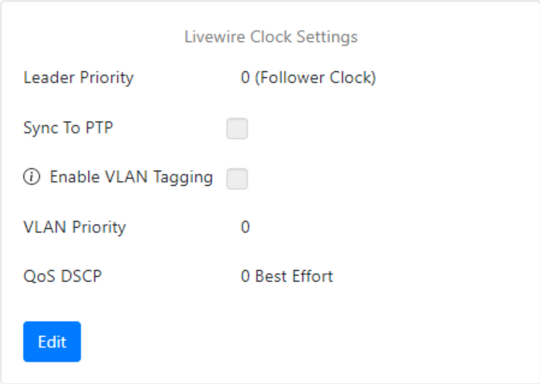
The PTP clocking status is displayed in the **Synchronization panel**.



If a Livewire Clock is in use then status is displayed in the **Livewire Clock Status** pane.

Livewire Clock Settings and Status

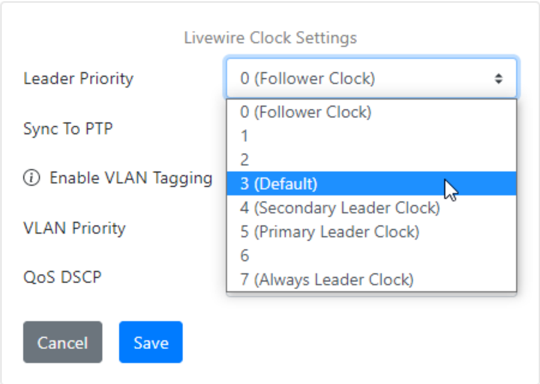
When streaming using Livewire+ the codec can be synced to a PTP clock or a Livewire device acting as the clock leader. Livewire systems can automatically change to a different clock leader if the current leader clock is disconnected or otherwise inoperable. This can be managed using the **Leader Priority** setting. **Livewire Clock Settings** are displayed for a follower codec.



Leader Priority

With Livewire each node has a clock leader configuration setting ranging from 0 to 7. Livewire clock **Leader Priority** 3 is the default suggested for a typical Livewire network. When a higher priority number is given to a node this will define which device becomes the network clock leader. A value of 7 insists a device becomes the clock leader. In summary:

- '0' means never a leader - follower only
- '7' means always - forced leader (note, do not use multiple forced leaders in a system)
- Factory default is 3. So all units have equal priority out of the box, and the following is used to break ties (in descending order): lowest Livewire+ audio transmit base channel, then lowest IP address, then lowest Ethernet address.



If no clock is present then devices negotiate a clock leader using **Leader Priority** settings and the lowest MAC address.

Sync to PTP

Sync to PTP is usually selected if your network uses PTP for synchronization. All devices should enable this setting if PTP is in use, unless the **Leader Priority** setting is **0**. When **Sync to PTP** is configured in **Livewire Clock Settings** it synchronizes the Livewire clock to the PTP clock when the codec is the Livewire clock master.

Enable VLAN Tagging

Select **Enable VLAN Tagging** to configure VLAN tagging for packet prioritization across the network.

VLAN Priority

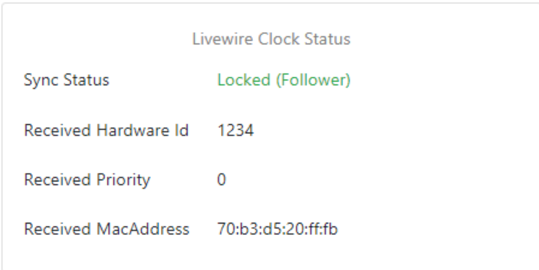
The recommended setting for AES67 is to set the **VLAN Priority** level at 6. Note: When operating in AVB networks the recommended setting is 5.

QoS DSCP

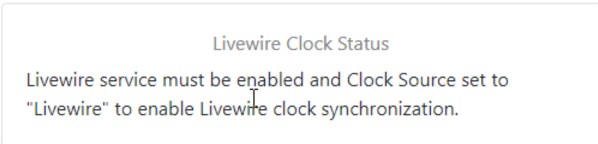
The QoS DSCP field in an IP packet header allows the network to easily recognize packets that need to be treated preferentially when traversing IP networks. AF41 (34) is the default setting and recommended for AES67. EF (46) is used for expedited forwarding of PTP clock packets.

Livewire Clock Status

Livewire Clock Status is displayed for leader and follower codecs when Livewire is in use.



When Livewire is not in use the **Livewire Clock Status panel** displays as follows.



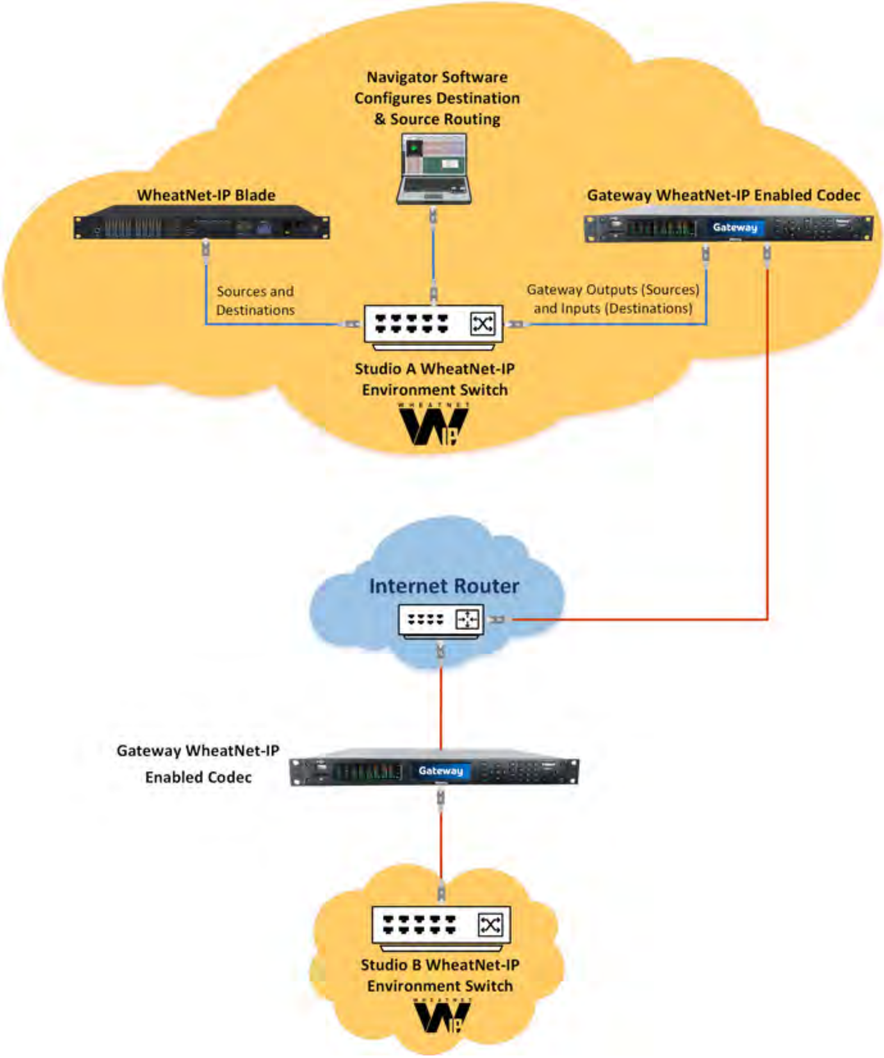
1.4 Discovery, Advertisement and Control

Stream Discovery and Advertisement

Only stream content parameters are specified in the AES67 standard and nothing is mandated regarding discovery and control. WheatNet-IP, Livewire and Dante use their own proprietary protocols for discovery and advertisement. RAVENNA is an open technology which also provides discovery and advertisement. Unless devices from vendors use or support the same stream discovery mechanism, e.g. Tieline WheatNet-IP codecs and Wheatstone WheatNet-IP equipment, it is necessary to configure streams manually using AES67. Therefore, to stream between the codec and other devices using AES67 it is necessary to configure the codec to send and receive streams using preset Source and Destination parameters.

WheatNet-IP Discovery and Control

Codecs with an optional WheatNet-IP card installed are capable of integrating with Wheatstone's WheatNet-IP Environment. This mean they can take advantage of proprietary WheatNet-IP discovery and control mechanisms. For example, Tieline WheatNet-IP enabled codecs support WheatNet-IP logic I/Os. The WheatNet-IP discovery protocol operates over a system-wide dedicated IP multicast "announce channel". Each device on a WheatNet-IP AoIP network, having subscribed to the multicast group, can hear the announcements of all other devices.



Livewire+

The Axia Livewire+ discovery protocol is natively supported and codec Sources and Destinations can be configured for use over Axia Livewire+ AoIP networks. Livewire+ operates over a system-wide dedicated multicast channel. Every device within the delivery scope for these multicasts, and having subscribed to the multicast group, can hear the announcements of all other devices. All devices periodically generate short presence announcements, and at longer interval, description advertisements. The description advertisements include attributes of the device, as well as a list and attributes of the streams they are able to transmit. Announcement data allows each participant to build a list of the other participants and streams available on the network, to assist with connection management. Livewire GPIOs are also supported.

SAP (Session Announcement Protocol)

SAP can be used to distribute SDP descriptions to multicast receivers, enabling simplified connection management for multicast streaming. Tieline uses SAP version 2 as defined in RFC 2974 for multicast AoIP streaming. SAP is used by Dante AoIP systems for audio stream discovery. When Tieline codec AES67 Sources are configured with SAP enabled they are discovered automatically in Dante Controller. This facilitates simple routing of Source audio streams within Dante systems. SAP Discovery can also be enabled in a Tieline codec to find and configure Tieline Destinations.

NMOS

NMOS stands for Networked Media Open Specifications and delivers Discovery, Registration and Control for the SMPTE ST 2110 suite of standards. Discovery and Registration ensure that parts of a networked media system can find each other. NMOS also provides connection management and audio channel mapping to device I/O channels. This is required because essential components for controlling and managing network devices are not included in the SMPTE standards. NMOS delivers specifications in the form of APIs. In an NMOS system each node exposes one or more NMOS APIs in order to find, register and control resources on each node. The codec uses TCP port 8081 by default for NMOS communications.

RAVENNA

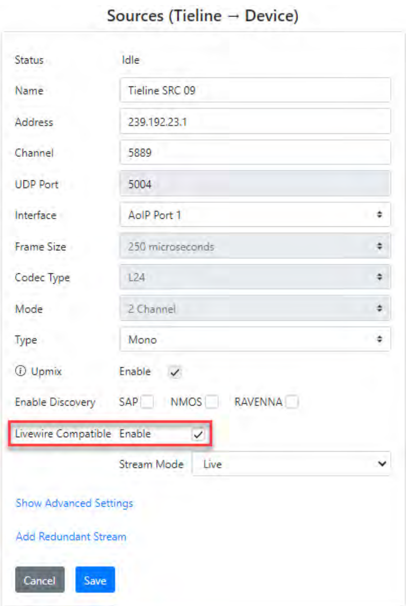
RAVENNA is an open technology that is AES67 and SMPTE ST 2110 compliant and it is used in many broadcast studios around the world. RAVENNA uses Bonjour for discovery and devices from different vendors that are compliant with RAVENNA can easily interoperate over AoIP networks.

Enabling Discovery and Advertisement Services

The codec supports Livewire, SAP, NMOS and RAVENNA for discovery and advertisement of AoIP streams.

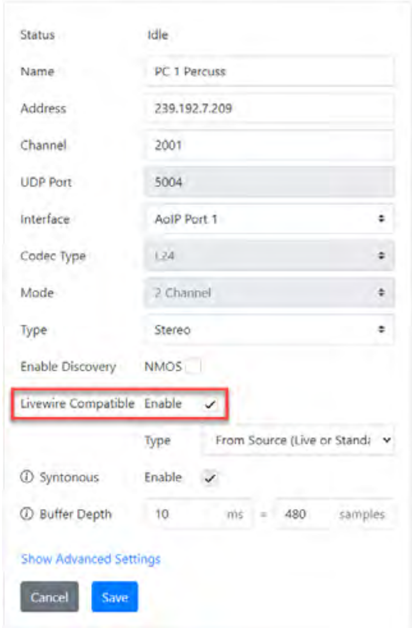
Configure Livewire Compatible Settings in Tieline Sources

Select the **Enable** check-box to enable **Livewire Compatible** streaming, which fixes some parameters to values compatible with Livewire Sources. The **Stream Mode** can be adjusted independently.



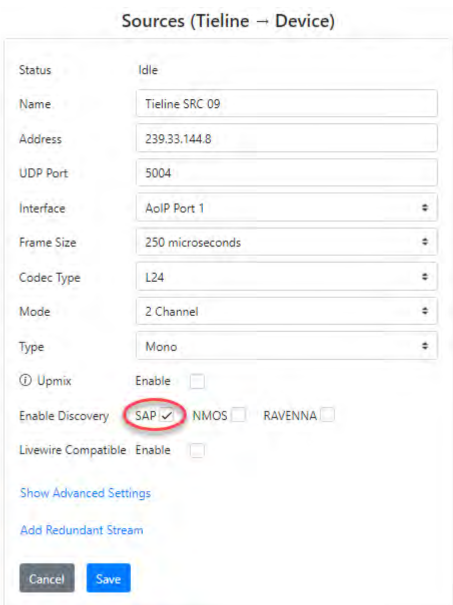
Configure Livewire Compatible Settings in Tieline Destinations

Select the **Enable** check-box to enable **Livewire Compatible** streaming, which fixes some parameters to values compatible with Livewire Destinations. Stream **Type** can be adjusted independently.



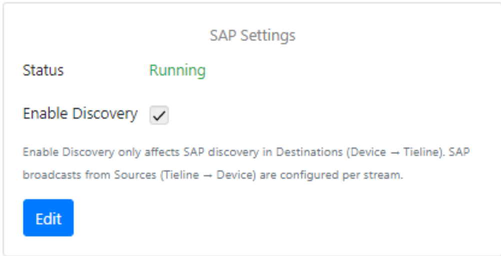
Enabling SAP Advertisement in Tieline Sources

Select the **SAP** check-box when creating a Tieline Source to enable SAP and allow other devices to easily discover a Source stream. SAP broadcasts from Sources are configured individually in each stream.



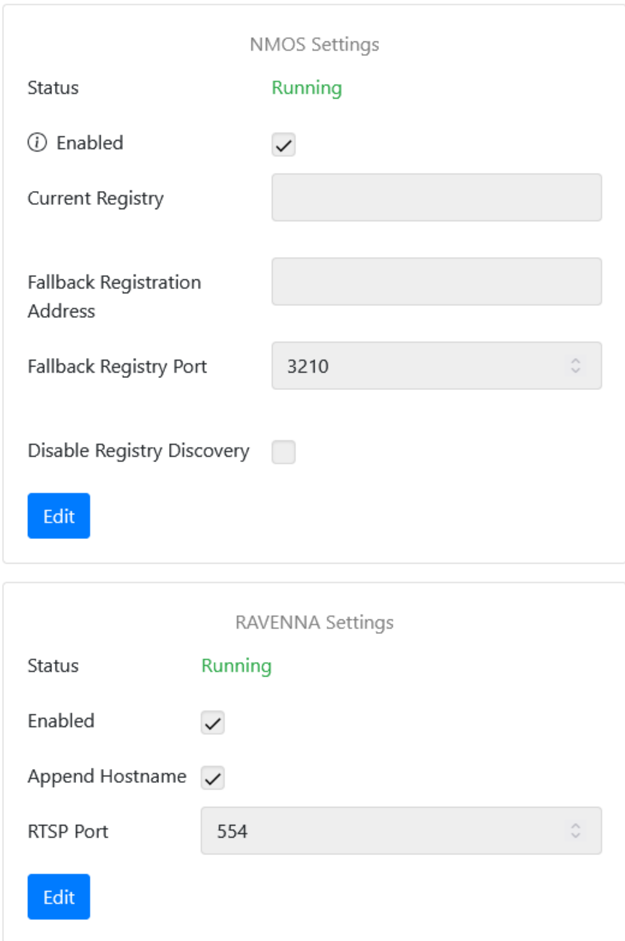
Enabling SAP Discovery for Tieline Destinations

Select **Enable Discovery** in the **Services panel** in the HTML5 Toolbox Web-GUI or AoIP Web-GUI to receive Destination audio streams from devices supporting SAP. Note: This setting only affects SAP discovery in Tieline codec Destinations.



Enabling NMOS and RAVENNA

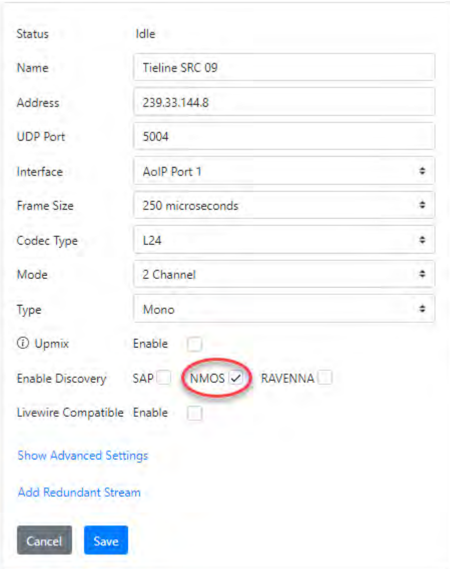
To allow use of NMOS and RAVENNA these services need to be enabled in the **Services panel** in the HTML5 Toolbox Web-GUI or AoIP Web-GUI.



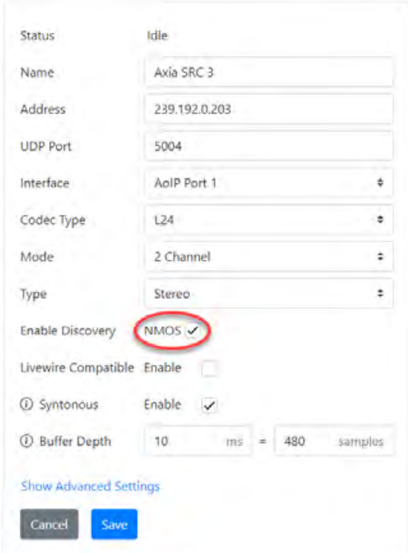
Services panel in the AES67 Web-GUI

If **Disable Registry Discovery** is not selected, the codec will try to discover a registry automatically using mDNS and it will be displayed as the **Current Registry**. If **Disable Registry Discovery** is selected, the codec will use the **Fallback Registration** (IP) **Address** to specify the registry manually. This may be necessary if more than one registry is present across a network.

After enabling NMOS in the **Services panel**, it is also necessary to select the check-boxes for NMOS in the **Destination panel** and **Sources panel** in the HTML5 Toolbox Web-GUI or AoIP Web-GUI for these streams to be discoverable.

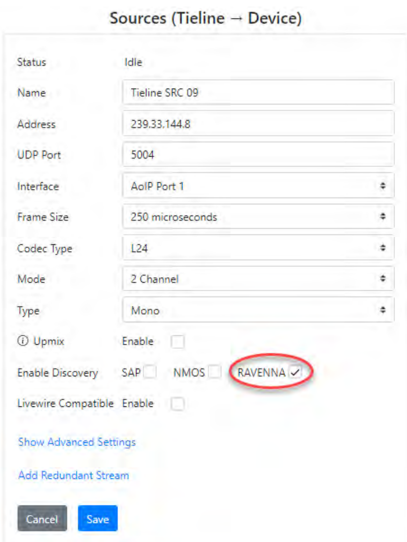


AoIP Web-GUI Sources panel displays NMOS enabled



AoIP Web-GUI Destinations panel with NMOS enabled

After enabling RAVENNA in the **Services panel**, it is also necessary to select the check-box for RAVENNA in the **Sources panel** in the HTML5 Toolbox Web-GUI or AoIP Web-GUI for each stream to be advertised.



AoIP Web-GUI Sources panel displays RAVENNA enabled

For more information about using RAVENNA for discovery and control see RAVENNA Streaming.

1.4.1 NMOS Stream Management

Gateway and other Tieline G6 codecs are Networked Media Open Specifications (NMOS) compliant and support NMOS IS-04 and IS-05 for discovery, registration and connection management. NMOS discovery and registration ensures that different elements of a networked media system can find each other easily using a common software management tool.

An NMOS server in a codec can integrate with software tools such as Riedel NMOS Explorer, which can discover, manage and connect IP Media Devices following AMWA IS-04 and IS-05. Some of the options that NMOS offers and that are currently supported include:

1. Discovery of nodes, devices and streams.
2. The ability to view and configure stream parameters - including displaying and copying SDP.
3. Starting and stopping streams.

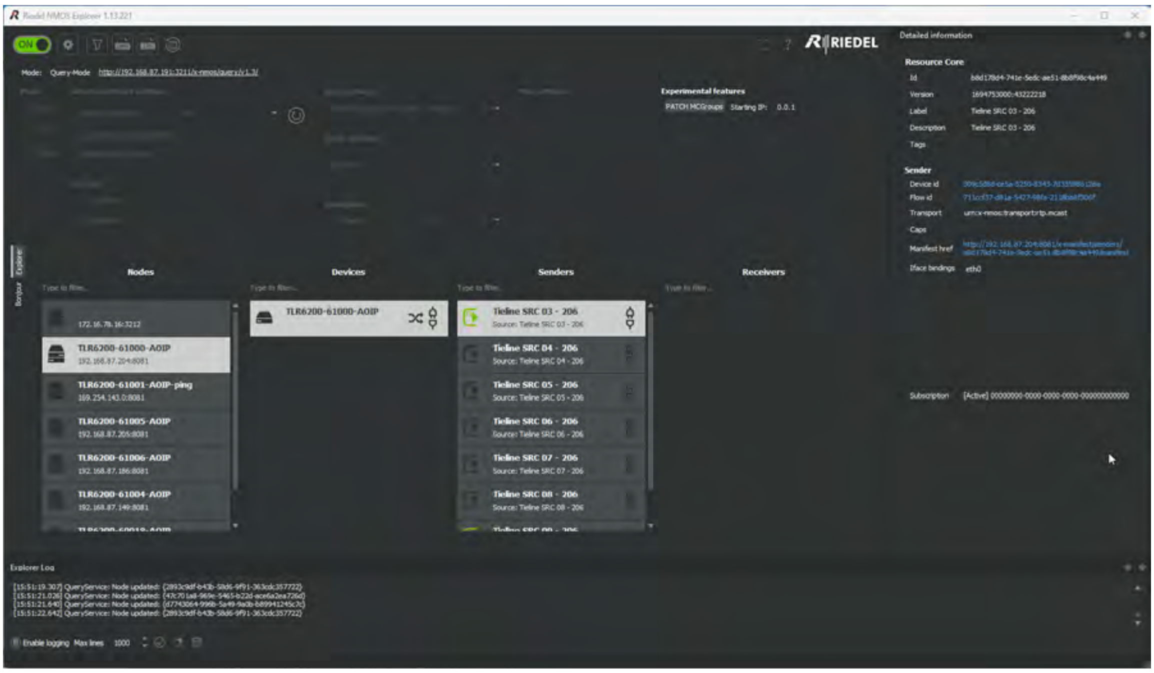
Important Notes:
- TCP port 8081 is used for NMOS communications by default.
- Codecs do not currently support managing redundant streams using NMOS**.**
- Append the NMOS TCP port number to the IP address of a codec to confirm the NMOS server is running in the codec, e.g. http://192.168.87.179:8081
- mDNS is used along with NMOS for device discovery.

Using NMOS to Manage Codec Streams

To use NMOS features it is necessary to:

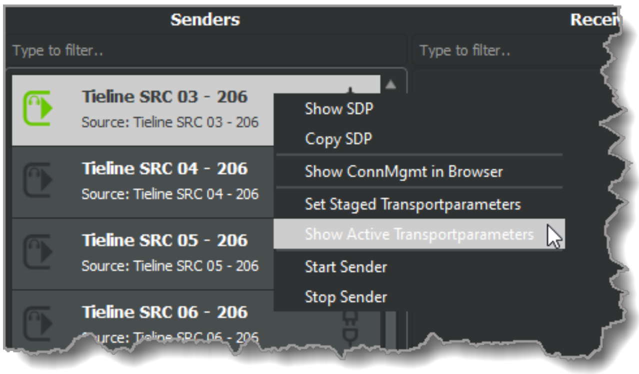
1. Enable NMOS in the **Services panel** in the HTML5 Toolbox Web-GUI or AoIP Web-GUI. See Discovery, Advertisement and Control for more info.
2. Select the **Enable Discovery** check-box for individual streams in the **Sources panel** and **Destinations panel**.

This allows codecs and streams to be discovered and managed in software tools like Riedel NMOS Explorer.



Riedel NMOS Explorer displaying available codecs and streams

It is simple to view and manage stream parameters, connect and disconnect streams, or copy SDP.



1.5 Codec AES67/ST2110-30 Setup

How to Configure AoIP Settings in Tieline Codecs

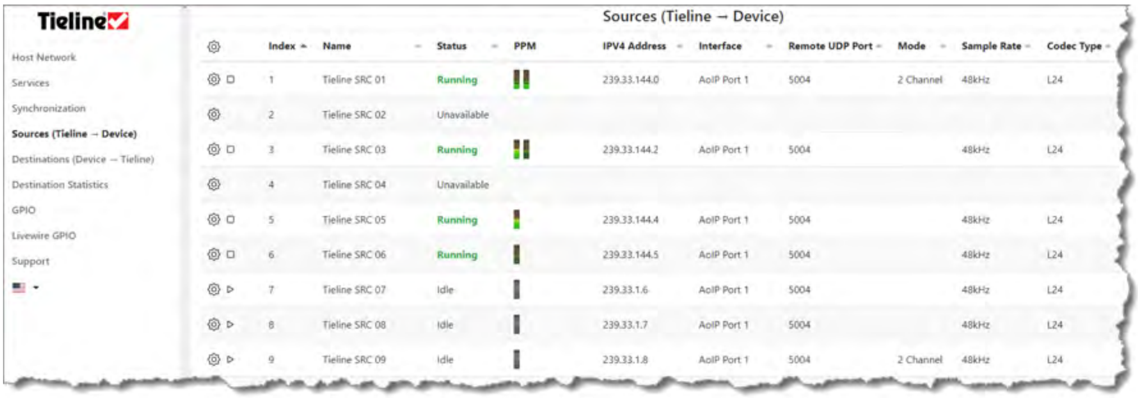
There are two ways to configure AoIP Sources and Destinations: Tieline's HTML5 Toolbox Web-GUI, or the AoIP Web-GUI.

1. Tieline's browser-based AoIP Web-GUI configuration tool configures codec AoIP streams when a computer is connected to an AES67 LAN. The AoIP Web-GUI has a several screens that allow:

- Configuration of source and destination audio streams.
- Monitoring and configuration of Synchronization settings.
- Monitoring and configuration of Host Network settings.
- Viewing Destination Statistics.
- GPIO monitoring and configuration.

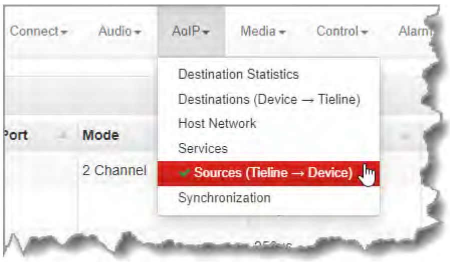
To access the AoIP Web-GUI type the IP address of the AoIP port interface into a browser, or type the codec hostname using the following format: **http://TLR6200-61086-AOIP**. Hostname and IP address details can be found in the codec via the **AoIP Host Network panel** accessed via the HTML5 Toolbox Web-GUI **AoIP** menu. Alternatively, from the codec **Home** screen select **Settings > Unit Details > AoIP1/2** to view the AoIP network IP address.

The AoIP Web-GUI looks a bit different to the HTML5 Toolbox Web-GUI but the functionality in the available AoIP panels is the same.



Gateway-16 AoIP Web-GUI for Configuring Sources and Destinations

2. The easiest way to configure settings remotely when away from the studio is with the Toolbox HTML5 Web-GUI connected to a codec via a WAN/LAN connection. All AoIP settings can be configured in the panels available within the **AoIP** menu at the top of the Web-GUI screen. The **Audio Options panel** can also be used to configure digital I/O settings, like selecting AES3 or AoIP as codec inputs or outputs. The Cloud Codec Controller can also be used to configure and monitor connections and audio streams.

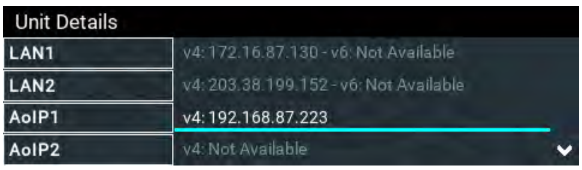


Important Note: The AoIP Web-GUI displays all the panel settings available within the **AoIP** menu in the Toolbox HTML5 Web-GUI.

Configure AoIP Host Network Settings

To discover the IP address of an AES67 host network using the codec front panel:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Unit Details** and press the button.
3. IP address details are displayed under the LAN address details and the address can be used to launch the AoIP Web-GUI.



It's simple to configure host network settings using the **AoIP Host Network panel** which is accessed from the HTML5 Toolbox Web-GUI **AoIP** menu.

1. Click **Edit** to adjust the **Hostname** or **IP Mode** and IP address details etc.

2. Click **Save** to store new settings.

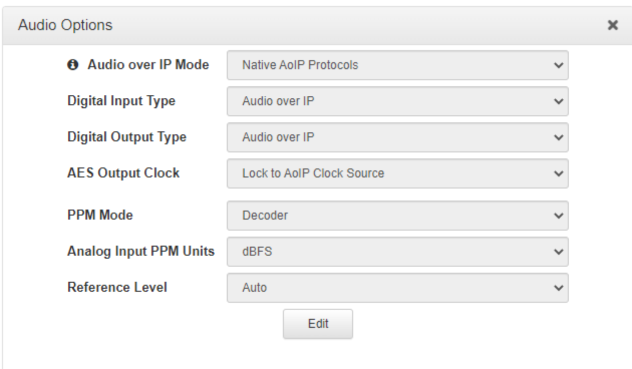
Configuring Codec Inputs and Outputs for AoIP Streaming

Some global AoIP-related settings reside in the **Audio Options panel** in the HTML5 Toolbox web-GUI. These include:

1. Audio over IP Mode (Note: this setting is only visible if a codec has WheatNet-IP card or Dante card (future release) installed):
   - **Native AoIP Protocols** (AES67, ST2110-30, Livewire+, RAVENNA), or
   - **WheatNet-IP**, or.
   - **Dante**.
2. Digital Input Type: AES3 or Audio over IP (AES67, ST 2110-30, Livewire+, RAVENNA, WheatNet-IP and Dante).
3. Digital Output Type: AES3 or Audio over IP (AES67, ST 2110-30, Livewire+, RAVENNA, WheatNet-IP and Dante).

To adjust these settings:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** at the top of the screen, then click **Audio Options** to display the **Audio Options panel**.
2. Click **Edit** to adjust settings.
3. Click **Save** to save settings.



Important Notes:

- AES3 audio will be output and also AES67 regardless of the **Digital Output Type** setting. However, there is an important point to note. When both the **Digital Input Type** and **Digital Output Type** on the codec is set to AES3, the AES Output Clock can be changed. However, when either the **Digital Input Type** or **Digital Output Type** is set to AoIP, the AES Output Clock is locked to the AoIP clock (i.e. PTP in AES67, WNIP clock in WNIP mode etc). Since the AES Output Clock also sets the synchronization of the AES67 sources, it is important that the AoIP clock is selected when using AES67 output. If it is not set to use the AoIP Clock, then the AES67 output will usually appear to be ok initially, but the timestamps will eventually deviate from the master clock reference (e.g. PTP) and some devices may stop receiving the streams when the timestamp deviates too far. So, when using AES67 outputs, it's always best to set the Digital Output Type to AoIP. If AES3 outputs are also being used, then the AES3 output sampling clock will be locked to the AoIP reference clock.
- When a Gateway codec is configured for WheatNet-IP the Tieline AoIP Web-GUI in the codec is inactive because the device AoIP interface is controlled by Wheatstone's Navigator software.

Audio over IP Mode

If an optional WheatNet-IP card is supported and installed in the codec select **WheatNet-IP** as the AoIP mode for used by the codec. This mode supports connecting to WheatNet-IP LANs and allows codec stream configuration using Navigator software. If an optional Dante card is supported and installed in the codec select **Dante** as the AoIP mode used by the codec. Then use Dante Controller software for stream configuration.

Select **Native AoIP Protocols** as the AoIP mode when connecting with natively supported AES67, ST2110-30, Livewire+ and RAVENNA protocols.

Digital Input Type

Select either **AES3** or **Audio over IP** as the **Digital Input Type** for the codec. This setting is global for all digital inputs.

Important Note: PPMs on the AoIP **Destinations panel** are inactive if the **Digital Input Type** is not set to **Audio over IP** and the corresponding input **Type** is not set as **Digital**.

Digital Output Type

Select either **AES3** or **Audio over IP** as the **Digital Output Type** for the codec. This setting is global for all digital outputs.

Sources and Destinations

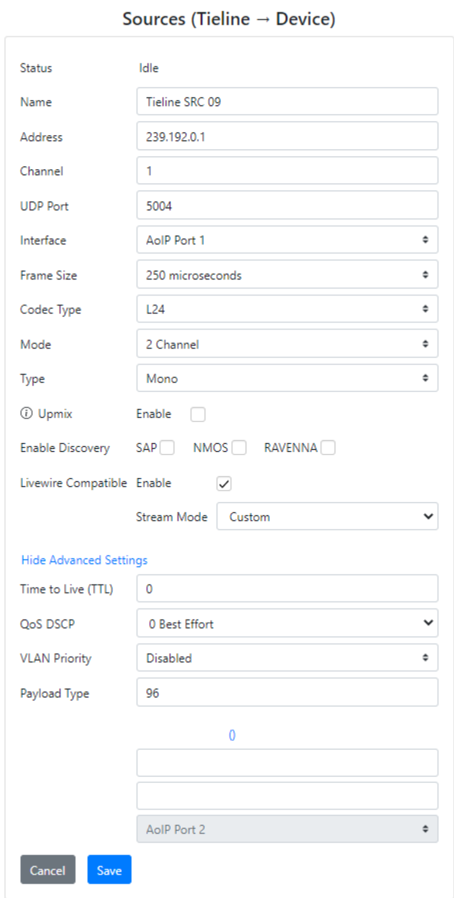
Sources and destinations identify outgoing and incoming AoIP streams and are configured separately. When configuring streams, a codec 'source' is an endpoint from which a stream is sent by the codec to an AoIP network, and a 'destination' is an endpoint at which a stream is received from an AoIP network by the codec.

One way to remember this is to think of the codec as an 'edge' device, bridging the divide between WANs like the internet, and a studio AoIP network. Most devices are configured with a studio-centric approach, so streams sent to the codec from the studio AoIP network, e.g. a WheatNet AoIP network, route audio to codec destinations. Therefore the codec receives audio as a 'destination' from the studio's perspective. Accordingly, the studio receives audio from the codec as an AoIP network 'source'.

Product	Feature	Notification
Gateway 4	Sources and Destinations	Configure up to 4 mono, or 2 stereo, or 1 x 8 channel (with up to 4 active channels) Source/Destination stream.
Gateway 8/16	Sources and Destinations	Configure up to 16 mono, or 8 stereo, or 2 x 8 channel AES67 Source/Destination streams in a Gateway 16; configure up to 8 mono, or 4 stereo, or 1 x 8 channel Source/Destination stream in the Gateway 8 codec.
MPX I	Sources and Destinations	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.
MPX II	Sources and Destinations	Configure up to 4 mono or 2 stereo Source/Destination streams.
Bridge-IT II and Bridge-IT XTRA II	Sources and Destinations	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.

Source Configuration Settings

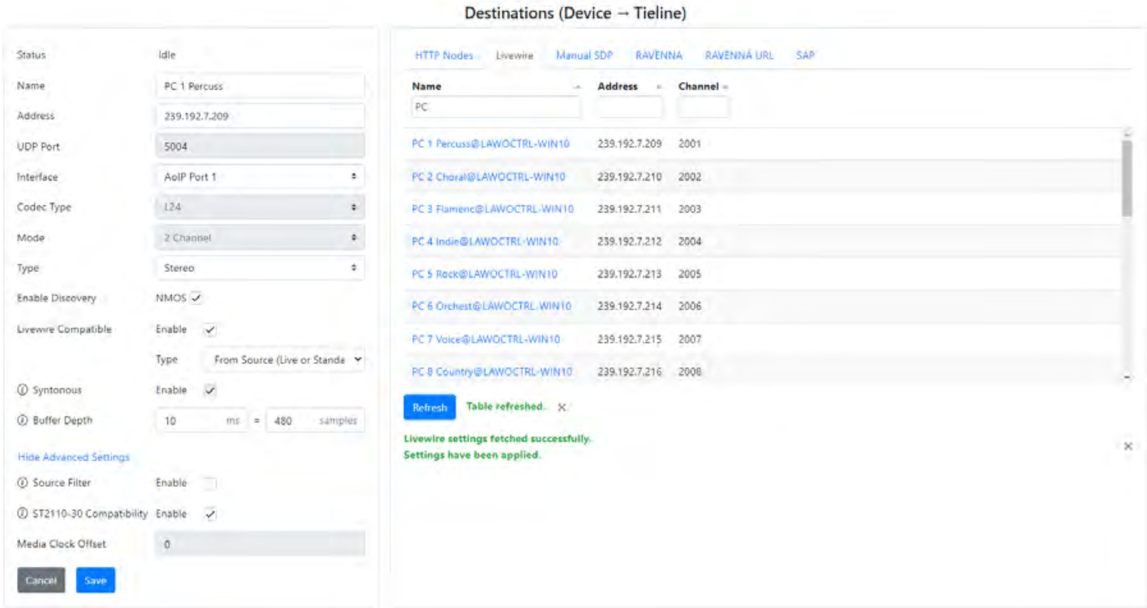
Configurable Source settings for audio streams sent by the Tieline codec over the AES67 LAN are listed in the following table.



Parameter	Settings
Name	Source stream name for identification purposes only; no effect on stream.
Address	Multicast IP address details; the default address for 16 source streams is 239.33.1.0 to 239.33.1.15.
UDP Port	The default UDP streaming port for RTP packets is usually 5004
Interface	Select the primary AoIP interface to use for the source stream.
Frame Size	Various settings from 125 microseconds to 4 milliseconds
Codec Type	Choose from L16 (16 bit) or L24 (24 bit) sampling; note the sampling frequency is configured in the **Synchronization panel** and is usually 48 kHz. All streams us a common sampling rate.
Mode	Select from 2 channel or 8 channel streaming in streams 1 and 9; this setting is not available for other streams and 2 channel is the default. Note: streams become unavailable when **8 Channel** mode is selected, based on how many channels out of 8 are selected. Note: The number of channels available depends on the model of codec purchased and channel licenses installed. Between 8 and 16 channels may be available.
Type	Configure mono or stereo streams; 3 - 8 channels also available for streams 1 and 9.
Upmix	Select the **Enable** check-box to duplicate a mono channel for a stereo stream.
Enable Discovery	Select a discovery protocol for simplified stream connection management, e.g. SAP, NMOS and/or RAVENNA. Note: No SAP announcements are sent if PTP is not locked.
Livewire Compatible	Select the **Enable** check-box to fix several parameters to values compatible with Livewire Sources.
Stream Mode	Select the preferred Livewire+ **Stream Mode** to suit the type of Source being configured.
Time To Live (TTL)	Setting used in multicast routers to ensure data packets have a finite life and don't cause congestion over networks. Each time a packet passes through a router it reduces by 1 until it reaches zero, at which point a router will no longer pass the packet.
QoS DSCP	Quality of Service setting using the DiffServ method as described in RFC 2474. Uses the DSCP field to mark packets according to their packet class. 34 is the default setting and recommended for AES67. 46 is used for expedited forwarding.
VLAN Priority	Represents a QoS prioritization scheme for Ethernet frames.
Payload Type	Some equipment requires a payload setting and the Tieline default setting is 96
Add Redundant Stream	Add a redundant IP stream and transmit an identical IP packet stream over a different AoIP interface. Packets are seamlessly reconstructed by the receiver to ensure flawless audio is received.

Destination Configuration Settings

Configurable Destination settings for audio streams received by the codec over the AES67 LAN are listed in the following table.



Parameter	Settings
Name	Destination stream name
Address	Multicast IP address details; the default address for 16 destination streams is 239.46.1.0 to 239.46.1.15. Note: The default destination multicast address will normally be configured by the device streaming audio to the Tieline codec. Verify this address with the network IT administrator and then place the correct address into the **Address** text box in the **Destinations** panel. Note: The codec can receive a stream on multiple Destinations using the same multicast IP address. Configuration settings that need to be the same include the destination address, UDP port, AoIP Port, source filter and Type. Other settings are automatically configured as per the initial Destination stream configured.
Channel	The unique channel number used by Livewire to identify streams. Note: Only displayed for Livewire streams.
UDP Port	The default UDP streaming port for RTP packets is usually 5004
Interface	Select the primary AoIP interface to use for the destination stream.
Codec Type	Choose either L16 (16 bit) or L24 (24 bit) samples. All streams us a common sampling rate.
Mode	Select from 2 channel or 8 channel streaming in streams 1 and 9; this setting is not available for other streams and 2 channel is the default. Note: streams become unavailable when **8 Channel** mode is selected, based on how many channels out of 8 are selected.
Type	Configure mono or stereo streams; 3 to 8 channels also available for streams 1 and 9.
Downmix	By default -6dB of gain reduction is applied to the left and right channels of the received 2 channel stereo stream and downmixes audio into mono; selectable configuration options include **Left + Right**, **Left**

	only, or **Right only**. Note: When configuring a mono Destination stream, if a stereo RAVENNA, SAP or Livewire stream is parsed, then Downmix will be automatically selected for compatibility. RAVENNA and SAP include SDP info on the number of channels in a stream.
Enable Discovery	Select the check-box to enable NMOS discovery and configuration/control of a Destination audio stream.
Livewire Compatible	Select the check-box to Enable Livewire+ compatibility in the codec. This feature fixes some parameters to values compatible with Livewire Destinations. The Livewire tab in the **Destinations panel** displays all advertised Livewire+ Sources across the network.
Type	The **Livewire Compatible** stream **Type** is typically configured when a Livewire stream is parsed using the **Livewire tab**. It can also be manually configured using the **Type** drop-down menu. Options define elements like whether a specific Livewire IP address range is configured, or a custom stream configuration can also be defined.
Syntonous	Syntonous receive mode is used by default in Livewire Destinations and is supported by some other vendors. In Syntonous mode the sampling frequency is locked, but the absolute time reference at the sender and receiver may be different. Redundant streaming is not supported in this mode.
Buffer Depth	When operating in Syntonous mode, the storage buffer used to capture incoming data packets ensures the continuity of audio streams by smoothing out packet arrival times during periods of network congestion.
Link Offset	The Link Offset is defined as the difference in time between when the audio enters the transmitter and exits the receiver. The link delay setting must be greater than the worst case link delay, taking into account network jitter. This setting is not available in Syntonous mode.
Source Filter Enable (Advanced Settings)	Select the check-box to enable and disable Source filtering.
Source Filter Address (Advanced Settings)	Only accept data from the origin address that is pushing to the multicast address. Leave empty to accept any source address.
ST 2110-30 Compatibility	Select the **Enable** check-box to configure the **Media Clock Offset** setting as 0. This is required for ST2110-30 compliant streaming.
Media Clock Offset	Configurable **Media Clock Offset** setting. Note: when **ST 2110-30 Compatibility** is enabled this setting is greyed out and set to 0.
Add Redundant Stream	Add a redundant IP stream and stream an identical IP packet stream over a different AoIP interface. Packets are seamlessly reconstructed by the receiver to ensure flawless audio is received.
HTTP Nodes	RAVENNA allows advertisement of web management interfaces controlling devices to which you are connecting. The **HTTP Nodes tab** displays **Hostname** and **Address** links to launch a device management interface directly.
Livewire	Displays available Livewire+ Sources. When a Source is selected, the SDP is fetched, parsed and populated in the fields within the **Destinations panel**.
Manual SDP	Copy and paste SDP text from an AES67 compatible device and click the **Parse SDP** button to configure a destination stream. This will configure the codec to receive an audio stream from a device.

RAVENNA tab	Displays available RAVENNA Sources. When a Source is selected, the SDP is fetched, parsed and populated in the fields within the **Destinations panel**.
RAVENNA URL tab	Copy a known RTSP URL address for a device and then paste into the **RAVENNA URL** tab to populate data in the **Destinations panel**.
SAP tab	Displays sources advertised using SAP. When a Source is selected, the SDP is fetched (and can be viewed), then it can be parsed and populated in the fields within the **Destinations panel**.

Use the following sections to configure Source and Destination streams for the various proprietary AoIP systems.

1.6 Configuring AES67 Sources and Destinations

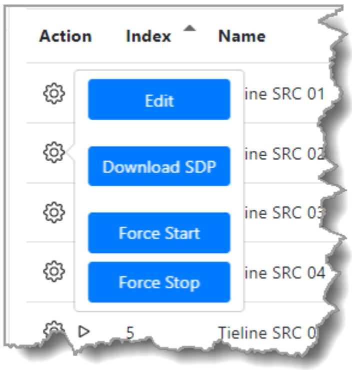
Product	Feature	Notification
Gateway 4	Sources and Destinations	Configure up to 4 mono, or 2 stereo, or 1 x 8 channel (with up to 4 active channels) Source/Destination streams.
Gateway 8/16	Sources and Destinations	Configure up to 16 mono, or 8 stereo, or 2 x 8 channel AES67 Source/Destination streams in a Gateway 16; configure up to 8 mono, or 4 stereo, or 1 x 8 channel Source/Destination stream in the Gateway 8 codec.
MPX I	Sources and Destinations	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.
MPX II	Sources and Destinations	Configure up to 4 mono or 2 stereo Source/Destination streams.
Bridge-IT II and Bridge-IT XTRA II	Sources and Destinations	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.

More information about individual settings in this section are available in the AES67 Setup section within this manual.

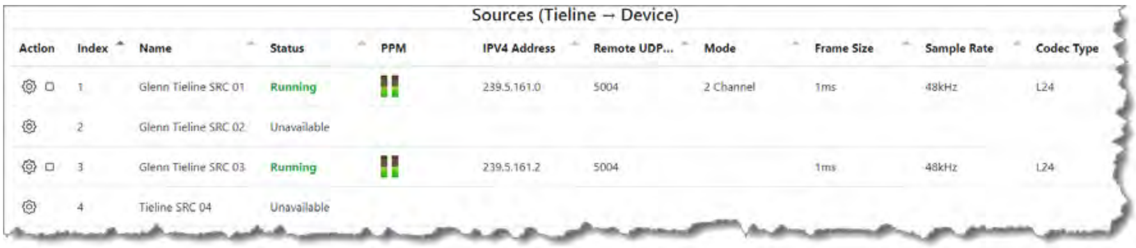
Creating an AES67 Source (Outgoing audio stream from Tieline codec)

1. To create an outgoing Source stream open the Tieline AoIP Web-GUI. Note: if you don't know the IP address of the codec on the AES67 LAN, open Toolbox HTML5 Web-GUI for the codec and then open the **AoIP Host Network panel** in the AoIP menu to view the address.
2. In the AoIP Web-GUI select **Sources** (or in the HTML5 Toolbox Web-GUI click **AoIP** and then **Sources** to open the **Sources panel**).

3. Choose a stream to configure and select the **Configuration symbol** , then click **Edit**. Note: **Matrix Editor** hard coded Source output channel assignments for streams depend on the stream configuration mode. This is outlined in the table that follows later in this section.



4. Edit the **Name** of the stream if required. Note: this is for identification purposes only and has no effect on the stream.
5. The multicast **Address** is entered into the AoIP device that will receive a client multicast stream and the **UDP Port** should also be configured correctly. Note: use the default destination address or edit as appropriate. Take care to not use one of the reserved addresses (see https://en.wikipedia.org/wiki/Multicast_address).
6. Select the preferred primary AoIP **Interface** used to stream audio.
7. Select the **Frame Size** (packet time) from 125 microseconds to 4 milliseconds.
8. Configure the **Codec Type**. In most situations this will be **L24** (24 bit).
9. Streams 1 and 9 can be configured in either **2 Channel** or **8 Channel** Mode. This mode needs to be configured in both devices for streaming to succeed. Streams become unavailable when **8 Channel** mode is selected, based on how many channels out of 8 are selected. Note: The number of channels available depends on the model of codec purchased and channel licenses installed.
10. Configure the **Type** of connection. For most connections this will be mono or stereo; streams 1 and 9 can also be configured to stream between 1 and 8 channels of audio. Up to 8 PPMs are displayed for a single stream with up to 8 channels of audio.
11. Click **Save** to save all configured settings.
12. Verify audio is metering on the PPMs and click the **Start symbol** ▷ to start streaming. **Running** appears in green when streams are successfully streaming.



Action	Index	Name	Status	PPM	IPV4 Address	Remote UDP...	Mode	Frame Size	Sample Rate	Codec Type
	1	Glenn Tieline SRC 01	Running		239.5.161.0	5004	2 Channel	1ms	48kHz	L24
	2	Glenn Tieline SRC 02	Unavailable							
	3	Glenn Tieline SRC 03	Running		239.5.161.2	5004		1ms	48kHz	L24
	4	Tieline SRC 04	Unavailable							

Gateway codec Sources panel

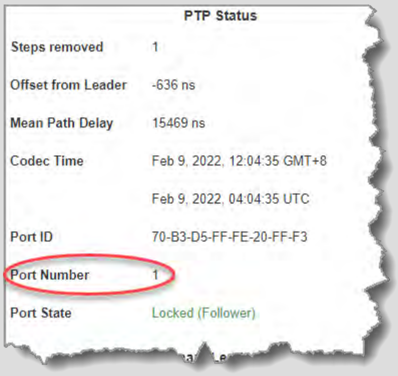
Redundant Streaming

SMPTE ST 2022-7 compliant redundant streaming (Seamless Protection Switching) can be configured for a Source or Destination stream. When configured, the codec transmits two packet streams, one on each interface, containing identical copies of the payload in each packet. This allows hitless merge, whereby a single reconstructed output stream is created through seamless protection switching at the RTP datagram level. For each stream it is possible to select whether port **AoIP 1** or **AoIP 2** will be used to transmit or receive packets.

Important Notes:

- Two different AoIP interfaces must be configured for redundant streaming. **AoIP 1** is the default interface for the primary stream.

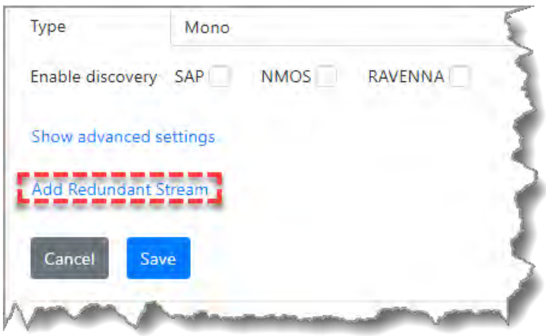
- While both AoIP ports can be connected to the same switch and use the same subnet, it is recommended that they are connected to separate switches on separate subnets for network resilience.
- When a Source employs redundant streaming using the same multicast address then each AoIP interface must be configured with a different subnet. E.g. 192.168.87.x and 192.168.88.x.
- If both AoIP ports are on the same subnet then each redundant source stream leg must use a different Destination multicast address.
- Multiple PTP Leaders are required to configure redundancy. We recommend using at least two Primary Leader PTP clocks. When configuring PTP in a SMPTE ST2022-7 redundant network setup, each Primary Leader should be connected to both networks in order to avoid different synchronization times in the two networks. E.g. In case only one clock loses its GPS reference signal. PTP Primary Leader synchronization on each interface must have very closely aligned clocks, otherwise glitches may occur when failing over from one Primary Leader to the other. For example, in the case of a link failure, if a PTP Leader fails over to a redundant PTP Leader clock, PTP sync will be lost momentarily until it locks to the redundant Leader clock. The AoIP **Port Number** change will be displayed in the **Synchronization panel**.



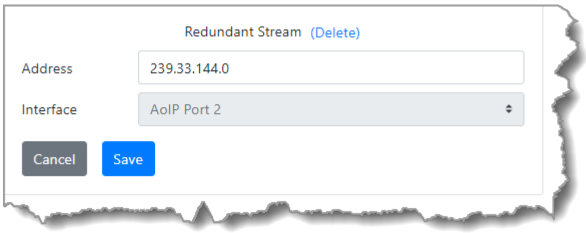
- **DO NOT** engage loopback of AES67/ST2022-7 Source and Destination streams on the same codec as this is not supported.

Configuring Tieline Source Redundant Streaming

1. In the AoIP Web-GUI select **Sources**.
2. Select **Add Redundant Stream**.



3. Insert the multicast address and adjust the redundant streaming **Interface** if required. Please note: the same multicast address can be used for the primary and redundant streams when each leg is on a different subnet.



4. Click **Save** to store the new settings.

Matrix Editor Source Output Assignments

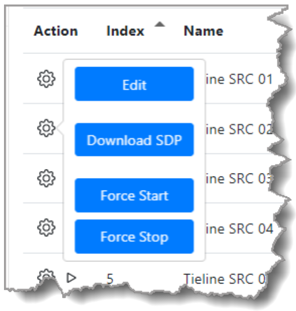
When streaming in AES67 mode, the **Matrix Editor** hard coded output channel assignments depend on the stream configuration **Mode**. Verify audio is displayed on the PPMs in the **Sources panel** before commencing streaming.

AoIP Audio Stream	Mode	Mix Outputs Used
Source 1	Mono	1
	Stereo	1 and 2
	8 Channel	1 to 8
Source 2	Mono	2
Source 3	Mono	3
	Stereo	3 and 4
Source 4	Mono	4
Source 5	Mono	5
	Stereo	5 and 6
Source 6	Mono	6
Source 7	Mono	7
	Stereo	7 and 8
Source 8	Mono	8
Source 9	Mono	9
	Stereo	9 and 10
	8 Channel	9-16
Source 10	Mono	10
Source 11	Mono	11
	Stereo	11 and 12
Source 12	Mono	12
Source 13	Mono	13
	Stereo	13 and 14
Source 14	Mono	14
Source 15	Mono	15
	Stereo	15 and 16
Source 16	Mono	16

Source SDP File Download

SDP is a format for describing RTP sessions and operating parameters, including network addressing, encoding format and other metadata. When configuring Destination streams on other devices, it is possible to download an SDP file for each Tieline codec AES67 audio stream and copy the text file contents into other devices to configure the stream on that device. To copy Tieline codec Source stream information onto other devices supporting this feature:

1. Choose a stream to configure and select the **Configuration symbol**, then click **Download SDP**.



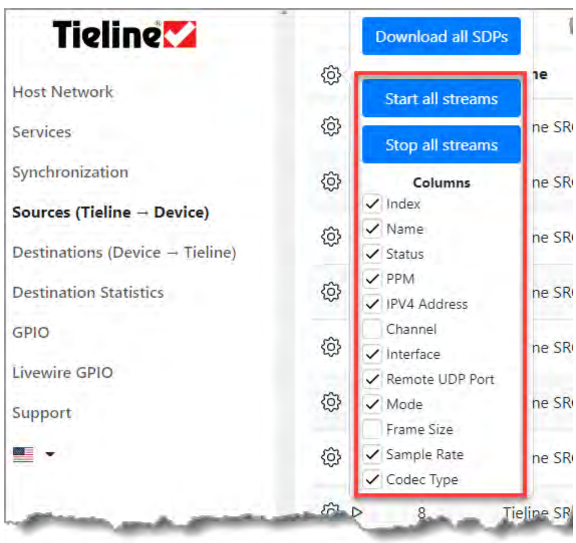
2. Save the file on a computer and using a text file editor copy the text from within the file and paste it into the Destination stream on the other device supporting this feature. Note: not all devices support this feature.

Download all Source SDP Files and Select Columns to Display

To download all Source SDP files simultaneously select the **Configuration symbol** at the top of the screen and click **Download all SDPs**. Note: It may be necessary to "allow" the download of multiple files in your browser.

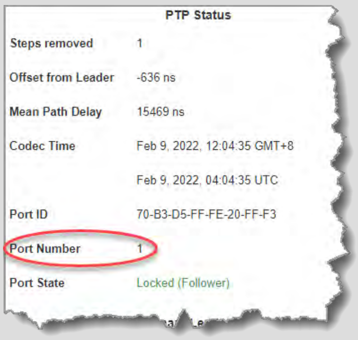
Helpful Tip:

Select the **Configuration symbol** at the top of the screen and click **Start all streams** or **Stop all streams** to simultaneously control all Sources. Click to select and deselect **Columns** to show / hide columns in the panel.



Important Notes:

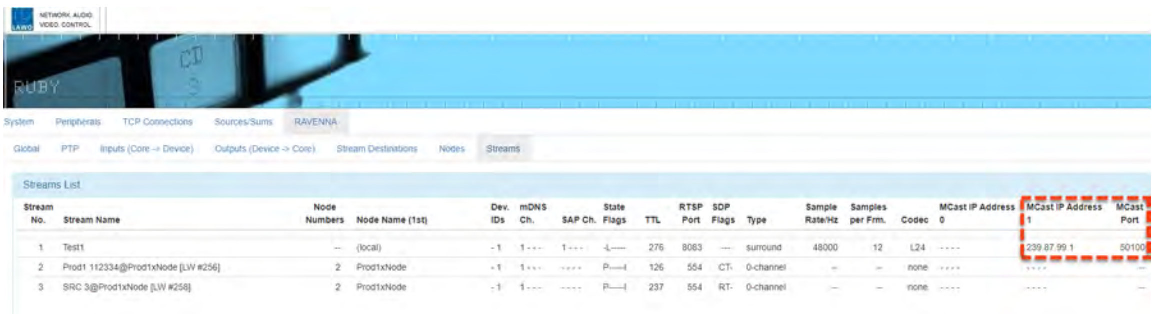
- Two different AoIP interfaces must be configured for redundant streaming. **AoIP 1** is the default interface for the primary stream.
- While both AoIP ports can be connected to the same switch and use the same subnet, it is recommended that they are connected to separate switches on separate subnets for network resilience.
- When a Source employs redundant streaming using the same multicast address then each AoIP interface must be configured with a different subnet. E.g. 192.168.87.x and 192.168.88.x.
- If both AoIP ports are on the same subnet then each redundant source stream leg must use a different Destination multicast address.
- Multiple PTP Leaders are required to configure redundancy. We recommend using at least two Primary Leader PTP clocks. When configuring PTP in a SMPTE ST2022-7 redundant network setup, each Primary Leader should be connected to both networks in order to avoid different synchronization times in the two networks. E.g. In case only one clock loses its GPS reference signal. PTP Primary Leader synchronization on each interface must have very closely aligned clocks, otherwise glitches may occur when failing over from one Primary Leader to the other. For example, in the case of a link failure, if a PTP Leader fails over to a redundant PTP Leader clock, PTP sync will be lost momentarily until it locks to the redundant Leader clock. The AoIP **Port Number** change will be displayed in the **Synchronization panel**.



- **DO NOT** engage loopback of AES67/ST2022-7 Source and Destination streams on the same codec as this is not supported.

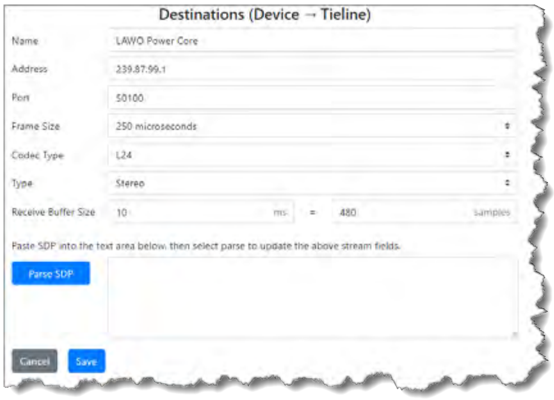
Creating an AES67 Destination (Incoming audio stream to Tieline codec)

1. To create an incoming Destination stream it is necessary to find out the multicast address and port used by the device sending the stream to the codec over the AES67 LAN.

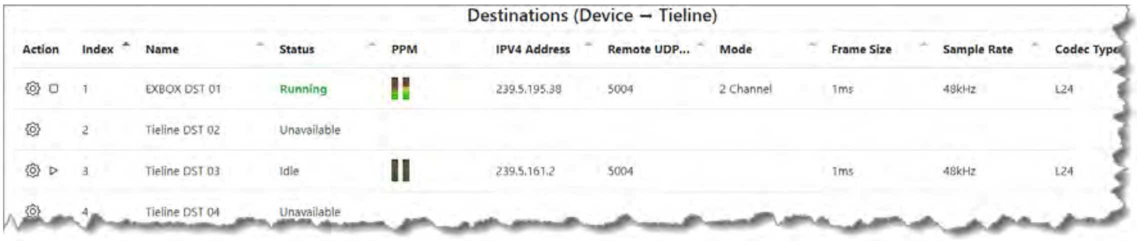


Example of LAWO device displaying its Multicast Source IP address and port

2. This information can be used to create a Destination audio stream in the codec as follows. Note: **Matrix Editor** input channel assignments for Destination streams depend on the stream configuration **Mode**. This is outlined in the table that follows later in this section.



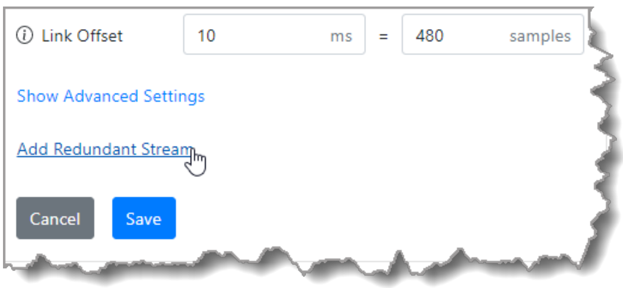
3. Click **Save** to save all configured settings.
4. Click the **Start symbol** ▷ to commence streaming. **Running** appears in green when successfully streaming and incoming audio should appear on the PPMs.



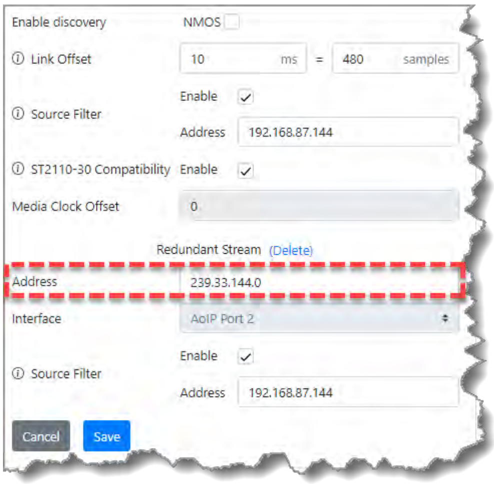
Configuring Destination Redundant Streaming

SMPTE ST 2022-7 compliant redundant streaming can be configured for an incoming destination stream.

1. In the AoIP Web-GUI select **Destinations**.
2. Select **Add Redundant Stream**.



5. Insert the multicast address and adjust the redundant streaming **Interface** if required. Please note: the same multicast address can be used for the primary and redundant streams.



6. Click **Save** to store the new settings.

Important Note:
- The destination Link Offset for redundant streams must be configured for the leg which has the highest link latency/jitter.
- The **Link Offset** includes link delay and receiver jitter buffer delay. To calculate the **Link Offset** setting determine network delay, including time in transmitter jitter buffers, and then add receiver jitter buffer latency. Ensure the **Link Offset** is higher than the total of network and jitter latencies.

Matrix Editor Destination Input Assignments

When streaming in AES67 mode, the **Matrix Editor** input channel assignments depend on the stream configuration **Mode** as displayed in the following table.

AoIP Audio Stream	Mode	Mix Inputs Used
Destination 1	Mono	1
	Stereo	1 and 2
	8 Channel	1 to 8
Destination 2	Mono	2
Destination 3	Mono	3
	Stereo	3 and 4
Destination 4	Mono	4
Destination 5	Mono	5
	Stereo	5 and 6
Destination 6	Mono	6
Destination 7	Mono	7
	Stereo	7 and 8
Destination 8	Mono	8
Destination 9	Mono	9
	Stereo	9 and 10
	8 Channel	9-16
Destination 10	Mono	10
Destination 11	Mono	11
	Stereo	11 and 12
Destination 12	Mono	12
Destination 13	Mono	13
	Stereo	13 and 14

Destination 14	Mono	14
Destination 15	Mono	15
	Stereo	15 and 16
Destination 16	Mono	16

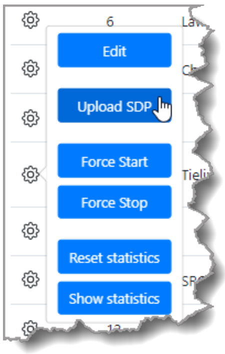
SDP Destination File Upload, or SDP Copy and Paste Function

When configuring Destination streams on a Tieline codec it is possible to upload an SDP file, or copy and paste the contents of an SDP text file, for each audio stream configured. To facilitate this feature, the other device must support SDP text file creation, or copying and pasting SDP info. Note: WheatNet-IP Blades and Telos xNodes support this feature.

Important Note: Ensure a stream is not running when uploading an SDP file or pasting SDP data into the web-GUI or it will fail.

To upload a saved SDP text file with Destination information:

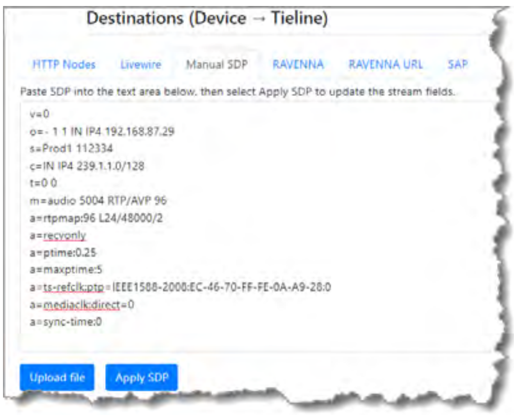
1. Select the **Configuration symbol** on a stream and then click **Upload SDP**.



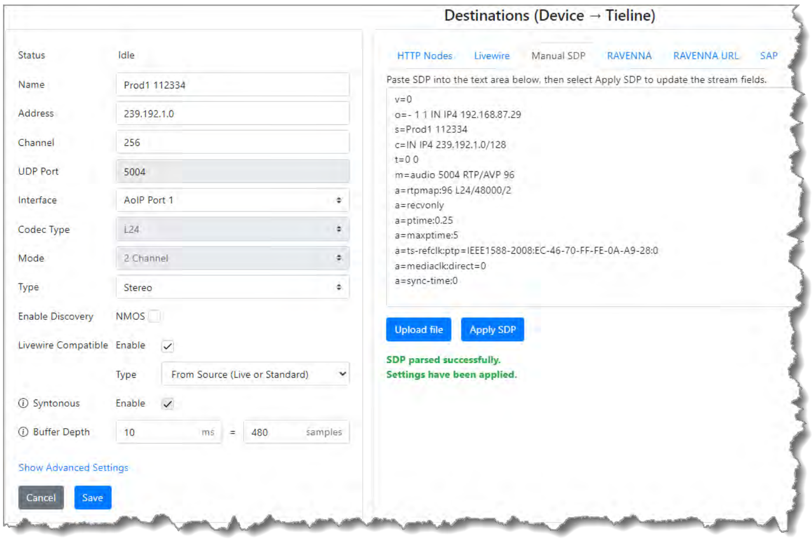
2. Select a saved SDP file and upload it.

To copy and paste SDP information from a text file into a Destination:

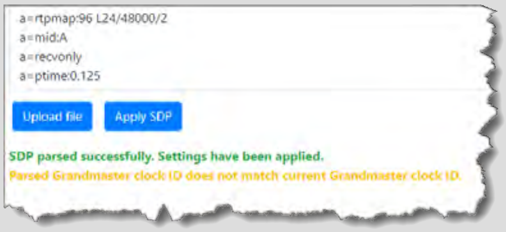
1. Select the **Configuration symbol** on a stream and then click **Edit**.
2. Copy the SDP information from the text file extracted from the other device into the text box adjacent to the **Apply SDP** button.



3. Click the **Apply SDP** button.
4. Ensure Destination settings have been populated correctly and click **Save.**

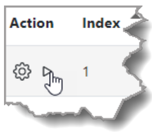


Important Note: If there is a PTP mismatch, a warning will display in the **Destinations panel** stating **Parsed Grandmaster clock ID does not match current Grandmaster clock ID**. This may occur when there is a change in a PTP Primary Leader and an SDP file has been generated using info from a previous PTP clock.



To avoid this message, download a new Source SDP file and apply this SDP file in the Destinations panel.

5. Click the **Start symbol** ▷ to commence streaming. **Running** appears in green when streams are successfully streaming.



Source and Destination Status Indications

Following are the **Status** indications for a Source and Destination.

Source Status Displayed	Source Status Description
Idle	The stream is configured and not running
Running	The stream is active and running
Unavailable	The stream is unavailable because it is allocated to another stream
Unlicensed	The stream requires a license upgrade to be available for streaming
Error	The stream has detected an error. This will also appear when an IP interface is removed. Stop and restart a stream to remove this error message.

Destination Statistics

There are two ways to monitor Destination packet and jitter statistics.

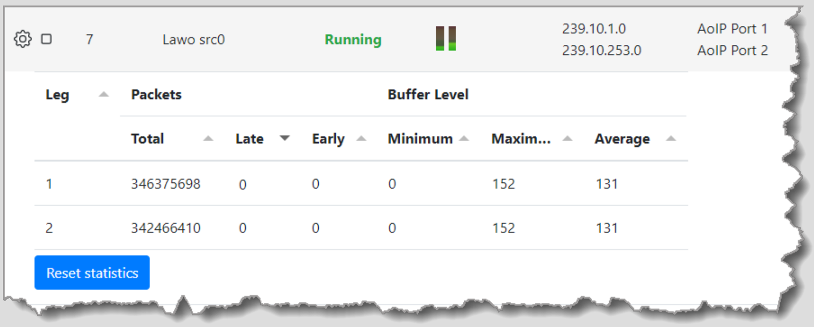
Statistics in Destinations Panel

1. Select the **Configuration symbol** ⚙ for a **Destination**, click **Edit**, and then **Show statistics**.

Packet and jitter stats are displayed for each Destination **Interface**.

Important Note:

- If the **Minimum** displays **0** this indicates the buffer has emptied and it is recommended that the **Link Offset** be increased to avoid artifacts and loss of audio.

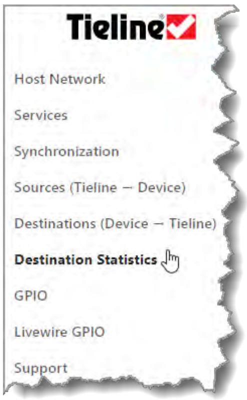


- A red **Caution symbol** is displayed when a Destination stream issue may require attention. Examples of when this will appear include:
  - o Late packets: If packets have been received late and have been lost the **Link Offset** may need to be increased. If the **Minimum** for received packets is a number close to zero (or zero) the **Link Offset** should be increased.
  - o Early packets: If packets arrive early and the buffer is already full packets will be dropped. Decreasing the **Link Offset** may resolve this issue. Also check that the sender and receiver are properly synchronized to the same PTP Primary Leader.
  - o Zero Minimum Buffer Level: A warning is displayed if the stream is running and the buffer has previously emptied.

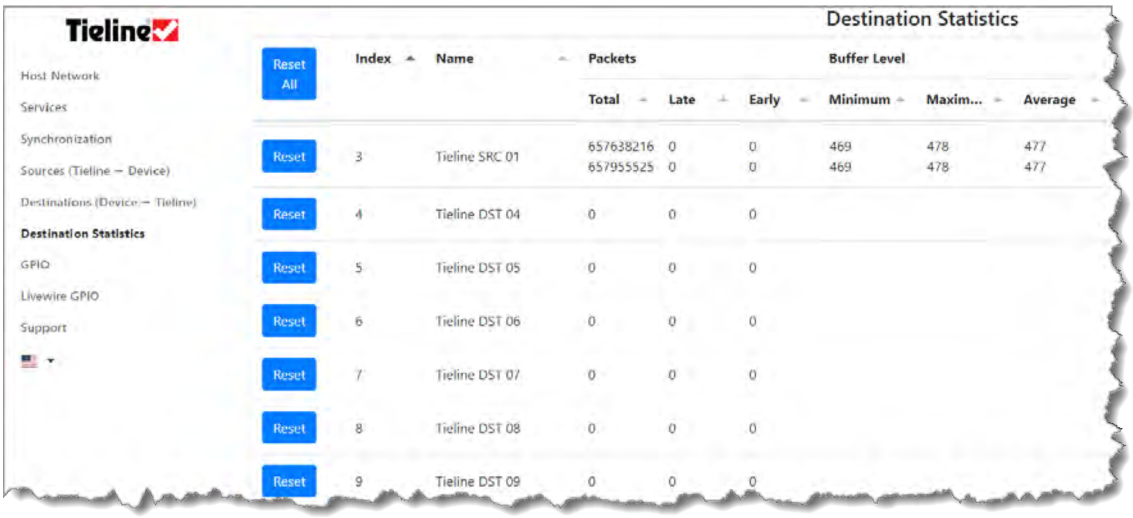
- o High Maximum Buffer Level: If the **Maximum** latency is high, e.g. over 1000, causes of high network latency should be investigated.

Destination Statistics Panel

1. In the AoIP Web-GUI select **Destination Statistics**.



2. This provides an overview of packet and jitter statistics for all Destinations in one screen.



Reset Statistics

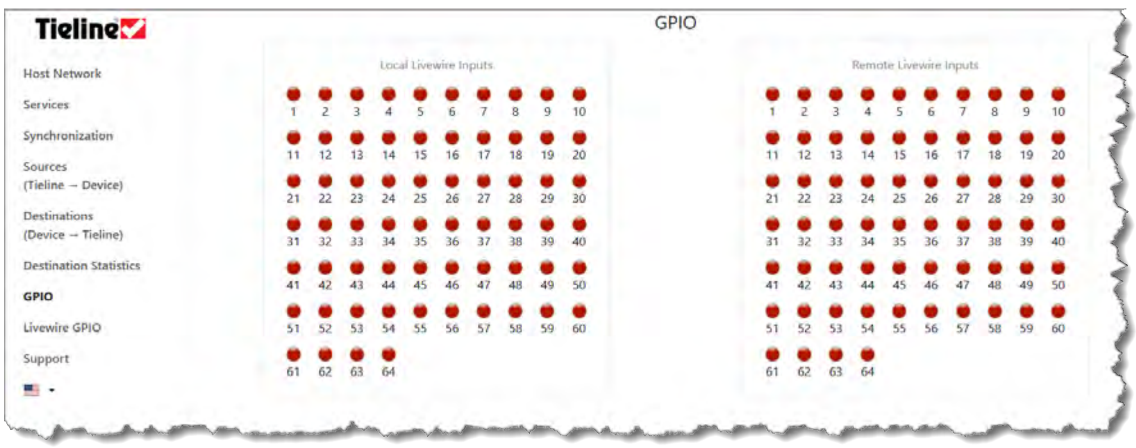
Click the **Reset** button to reset the statistics for a Destination or click **Reset All** to reset the statistics for all streams in the **Destinations Statistics panel**. It is also possible to show, hide, and reset Destination statistics from within the **Destinations panel**.

1. In the AoIP Web-GUI select **Destinations**.
2. Select the **Configuration symbol** at the top of the screen and click to show and hide statistics for all Destination streams within the panel; click **Reset all statistics** to reset the display.

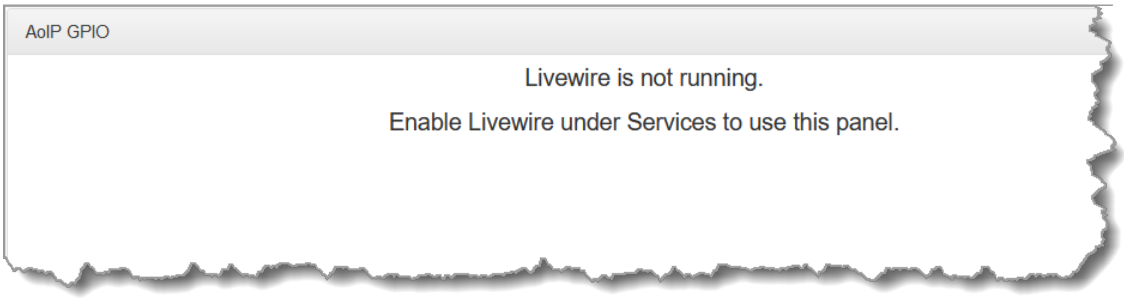
Helpful Tip:
Select the **Configuration symbol** ⚙ at the top of the screen and click **Start all streams** or **Stop all streams** to simultaneously control all Destinations. Click to select and deselect **Columns** to show / hide columns in the panel.

1.7 GPIO Panel

The **GPIO panel** in the AoIP Web-GUI displays the current active state of logical Livewire GPIOs in the codec. A maximum of 64 are available and displayed.



Note: Livewire must be enabled in the **Services panel** or the panel is disabled


AoIP GPIO
Livewire is not running.
Enable Livewire under Services to use this panel.

1.8 WheatNet-IP Streaming

Configuring Codec WheatNet-IP Cards for Use with Navigator

Wheatstone Corporation uses the WheatNet-IP protocol to transport and manage IP streams throughout broadcast plants. A Tieline Gateway with an optional WheatNet-IP card on-board can interface directly and seamlessly with these systems. When a WheatNet card is installed inside the Tieline Gateway and Gateway 4 codecs it does not require configuration using Wheatstone's Razor setup tool like Genie and Merlin WheatNet-IP codecs. Audio stream source and destination routing can be directly configured using WheatNet-IP Navigator software.

Important Notes:
- Navigator v3.8.798 was used in testing and configuration.
- If all blades are Blade-1 or Blade-2 blades with the green single-line display on the front panel, then it is not possible to use PTP/AES67.
- Ensure your network switch supports full-duplex 1000BASE-T and verify the link speed by checking that only the green WheatNet Ethernet port LED is on or flashing.

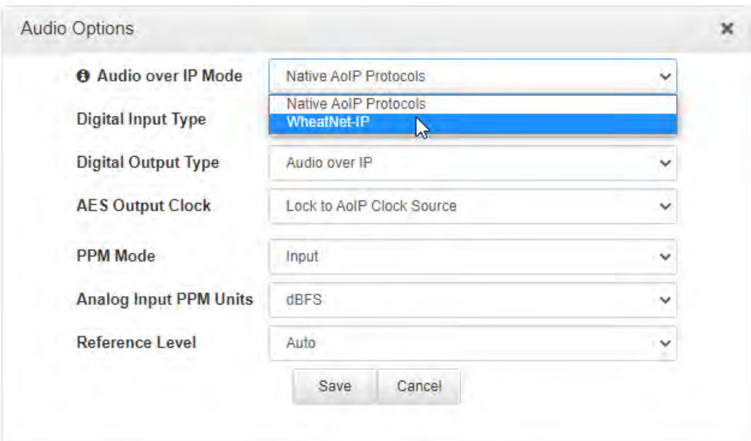
Prerequisites

1. The Tieline codec must have a WheatNet-IP card installed.

Configure the Codec for WheatNet-IP Streaming

Before streaming audio from the codec using WheatNet-IP it is necessary to configure the codec for WheatNet-IP.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Audio** to display the **Audio Options panel**.
2. Click **Edit** to adjust settings.
3. Click the drop-down menu for **Audio over IP Mode** and select **WheatNet-IP**.

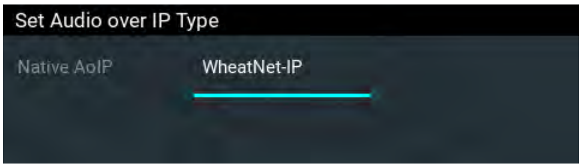


4. Both the **Digital Input Type** and **Digital Output Type** in the **Audio Options panel** need to be configured for **Audio over IP**.
5. Click **Save** to save settings.

The Audio over IP Mode can also be configured using the front panel of the codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Audio Options** and press the button.
3. Select **AoIP Mode** and press the button.

4. Select **WheatNet-IP** and press the button.



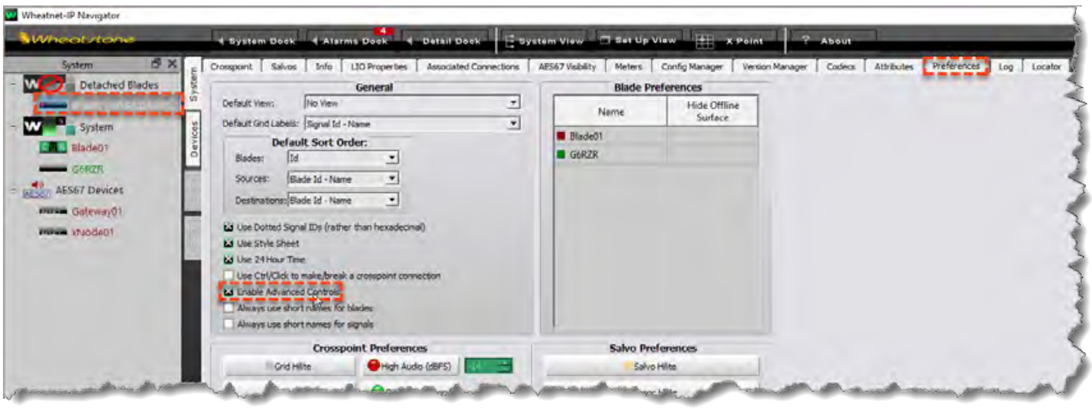
Important Note:

- When the **AoIP Mode** is changed we strongly recommend rebooting the codec using either the **Reset / Backup panel** in the Toolbox Web-GUI, or by navigating to **Settings > Reset & Backup > Reboot Codec** via the codec front panel.
- It is a good idea to close and relaunch Navigator if you change the **Audio over IP Mode** setting from **Native AoIP Protocols** to **WheatNet-IP**. Otherwise the codec may not be initialized successfully in the following procedure.

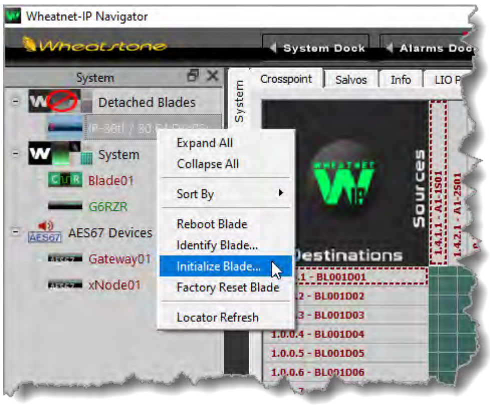
Configure a Gateway Codec with a WheatNet Card

After attaching a Gateway or Gateway 4 codec to a WheatNet-IP network and configuring the codec for WheatNet-IP streaming it is first necessary to configure a few settings. Note: the codec will appear as a Blade in Navigator and can be configured like a Wheatstone Blade.

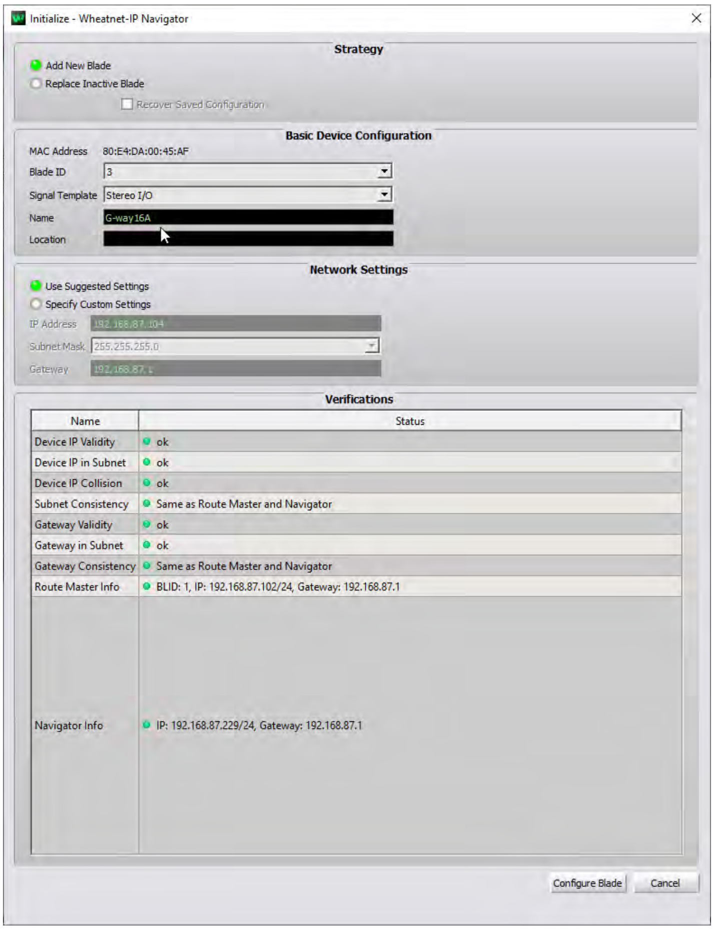
1. Launch Navigator and click to select the codec (Blade) within the **System pane** in the top-left corner of Navigator, then select the **Preferences tab**. Click to select the **Enable Advanced Controls** check-box.



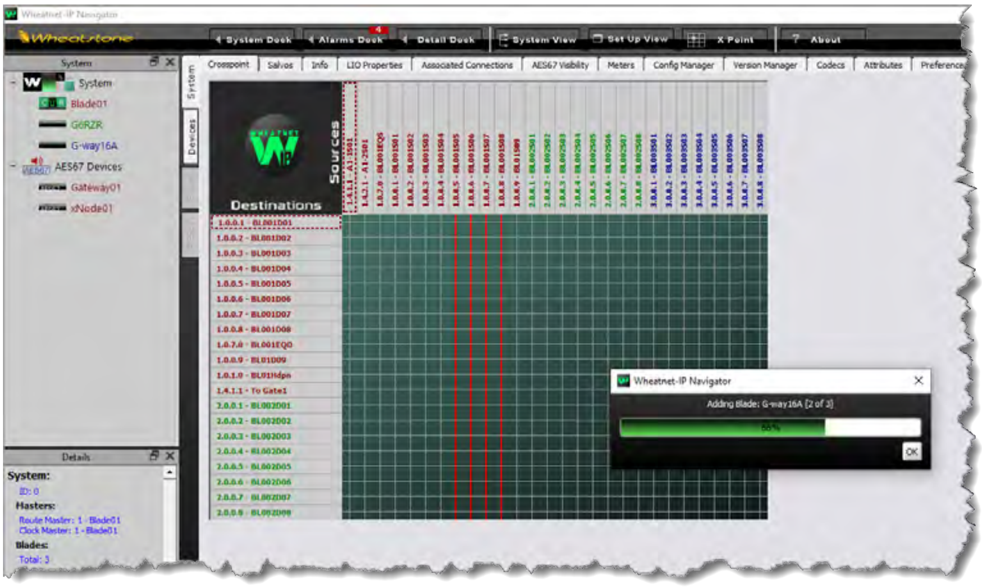
2. Right-click the codec under **System** and select **Initialize Blade**. Note: if you are connecting multiple codecs simultaneously, the MAC address can be used to identify each codec. The MAC address displayed on codecs in the **System** pane in Navigator can be matched to the WheatNet card MAC address displayed on the sticker on the chassis of each codec.



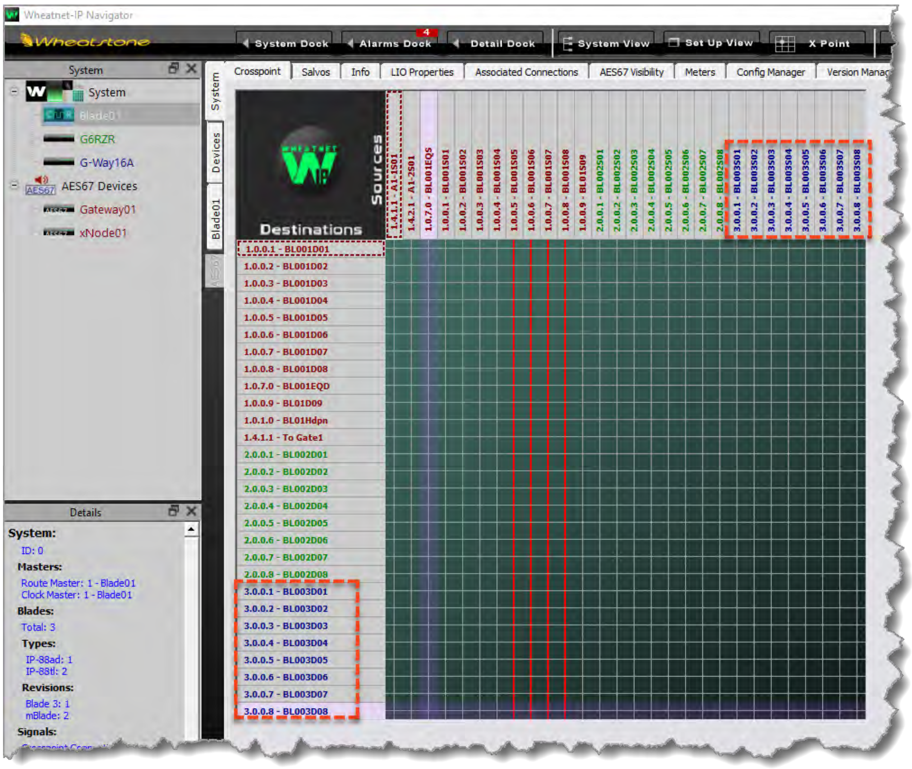
3. The **Initialize** dialog should open allowing you to rename the codec and if necessary configure network settings. Click the **Configure Blade** button to save new settings.



4. Navigator may take several minutes to configure the codec into the WheatNet system. Initially the codec may disappear from the **System** pane in Navigator and then reappear after a minute or two and be renamed with the new name given to the device.

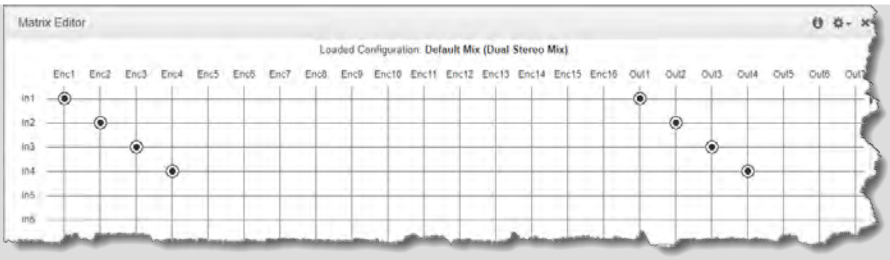


When configuration is complete, click to select the Blade in the **System pane** and select the **System tab** on the left and the **Crosspoint tab** at the top to verify all Sources and Destinations for the newly added codec are visible and selectable.

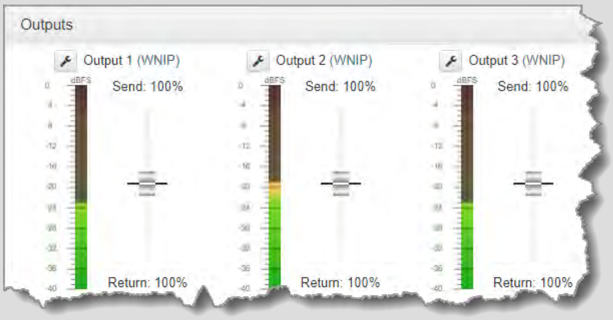


Important Note:

- Ensure audio is routed in the codec **Matrix Editor** from inputs to outputs if the codec is feeding audio into the WheatNet-IP network from codec input sources. I.e. route audio from inputs to outputs to create a WheatNet source in this way.



- In addition, if input audio is routed to outputs to create a WheatNet source, ensure the **Outputs panel** sliders used to output **Send/Return** audio from the output are mixing **Send 100%** audio to the outputs. Otherwise audio will not be outputted to the Blade Source. By default, the outputs are configured as **Return 100%** without Send audio mixed in. If **Return** audio is not required this can be removed from the mix.

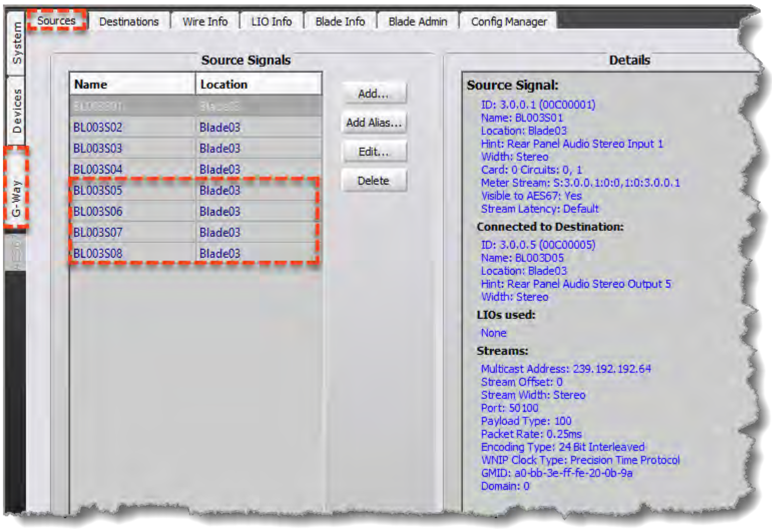


- When the codec is configured for WheatNet-IP the Tieline AoIP Web-GUI in the codec is inactive because the device is controlled by Wheatstone's Navigator software.

Managing Available Gateway Sources and Destinations

By default, when a Gateway or Gateway 4 codec is added to Navigator as a Blade it is configured with 16 available channels, or **Signals** in Navigator. If you have purchased a Gateway with fewer than 16 channels, e.g. Gateway 8, or have purchased a Gateway 4 codec supporting 4 channels, it may be necessary to delete the unavailable **Source Signals** and **Destination Signals** in Navigator. This will avoid any confusion with unavailable **Signals**, whereby Sources and Destinations within Navigator appear as available, even though they are not.

For example, if the codec Blade in the following image was a Gateway 8 with 8 available channels/**Signals**, simply select the **Sources tab** in Navigator for the device and delete the 4 unavailable stereo **Source Signals**. Then do the same in the **Destinations tab** for 4 unavailable stereo **Destination Signals**.

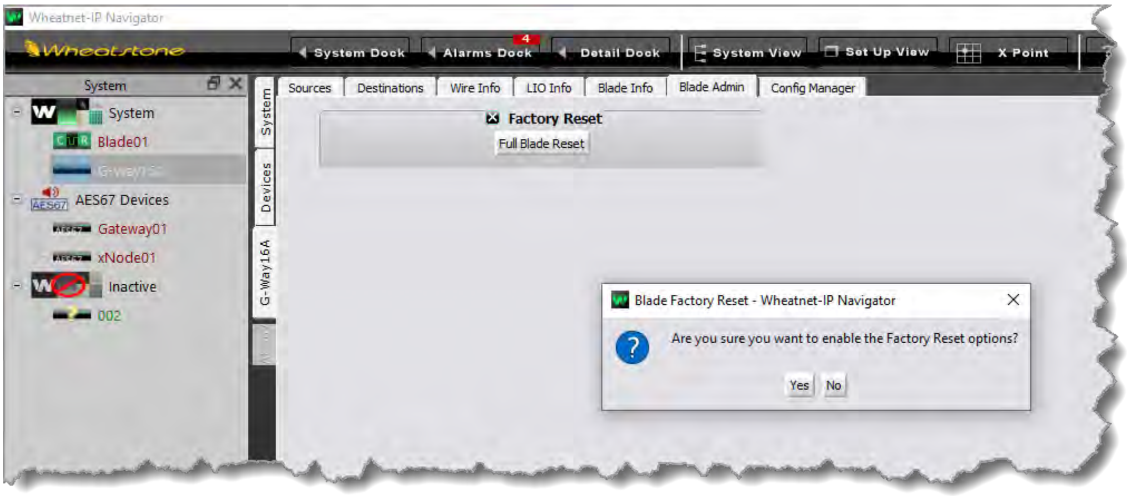


The same principle would apply for mono **Source Signals** and **Destination Signals** in Navigator, i.e. delete 8 the unavailable mono **Source Signals** and **Destination Signals**. If a Gateway is upgraded to increase channel density, e.g. from 8 channels to 16, it is relatively simple in Navigator to add **Source Signals** and **Destination Signals** as required.

Reset the WheatNet Card to Factory Defaults

To reset the WheatNet card back to factory default settings:

1. Select the **Blade Admin** tab and then select the **Factory Reset** check-box. Select **Yes** in the confirmation dialog that is launched.



2. Click the **Full Blade Reset** button to reset the codec's WheatNet card to factory defaults.

1.8.1 Stream to WheatNet using AES67

Tieline Codecs Streaming without a WheatNet-IP Card

It is possible to configure a Tieline codec to stream audio using AES67 without a WheatNet-IP card. For example, a Gateway codec may not have a WheatNet card installed or you may be streaming using an MPX or Bridge-IT II codec. Following is the configuration procedure.

Setting Up a Blade for Receiving AES67 Audio

To configure AES67 streaming it is necessary to:

1. Ensure the Blade is licensed for AES67 streaming.
2. Add the codec as an AES67 Device in Navigator for connectivity.
3. Verify the Blade's AES67-PTP timing status.

Important Notes:

- A Blade-3 blade must be set as the system's leader clock to use PTP/AES67, or a separate PTP clock must be designated as the network's primary leader clock.
- If all blades are Blade-1 or Blade-2 blades with the green single-line display on the front panel, then it is not possible to use PTP/AES67.

Ensure the Blade is Licensed for AES67 Streaming

Check that Navigator has both licenses by clicking the **? About** button in navigator, then click the **Show License** button to view Navigator's license.txt file, which is saved in: C:\ProgramData\Wheatstone\Navigator\. The text for both licenses must be listed. The top one is your Navigator license, which has the product entry line: **PROD WNIPNavigator**. Below that is the PTP clock license, which has this product entry line: **PROD PTPClocking**.

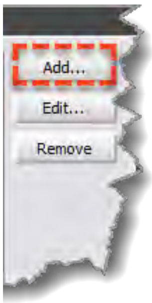
If the **PTPClocking** license is not displayed, contact Wheatstone support to request an AES67-PTP License.

Add a Tieline Codec to Navigator

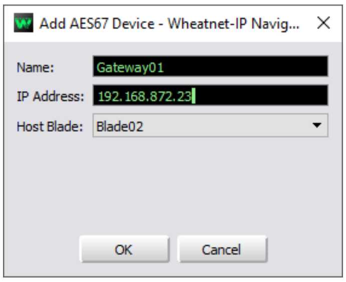
The codec needs to be added to navigator as an AES67 device so the WheatNet-IP Blade can see the codec and stream AES67 audio between the devices.

1. Select **Devices** and then select the **AES67 Devices** tab.

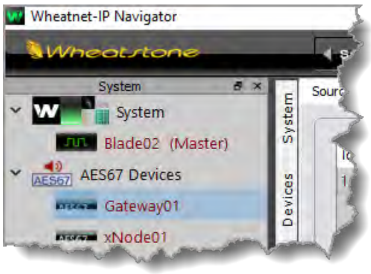
2. Select **Add** to add a new device.



3. Enter the **Name** and the AES67 LAN **IP Address** details for the codec, then click **OK**. Note: AES67 LAN IP Address details for the codec are displayed in the **AoIP Host Network panel** in the Toolbox HTML5 Web-GUI, or the AoIP Web-GUI.



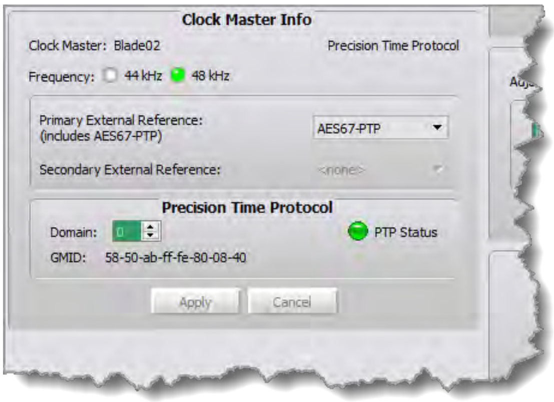
4. The codec is now added as a device in Navigator under **AES67 Devices**. Complete this for multiple codecs if required.



Verify the Blade's PTP Timing Status

1. Ensure the PTP Leader Clock is running on the AES67 LAN to which the codec is connecting.
2. Select the Blade and **System** and then select the **Info** tab.

3. In the **Clock Master Info** section ensure the **Frequency** is set as **48kHz** and change the **Primary External Reference** to **AES67-PTP**. The **PTP Status** is green and locked when the clock is detected.

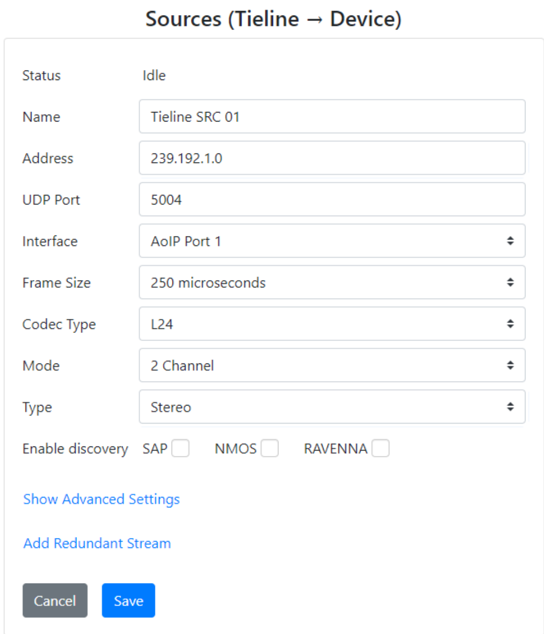


4. The Tieline codec supports low latency AES67 audio streams down to 125 microseconds and in testing Wheatnet Blades default to 250 microseconds, which is also supported by Tieline. To use an AES67 device in WheatNet systems using the 1 millisecond AES67 latency setting, it will be necessary to also configure the default AES67 packet rate to 1 millisecond. To view or adjust this setting select the **Blade** and then select the **Blade Info** tab. WheatNet-IP will successfully stream between Tieline codecs using 250 microsecond packet timing, as well as other devices using 1 millisecond timing, using a process they call "packet timing translation."

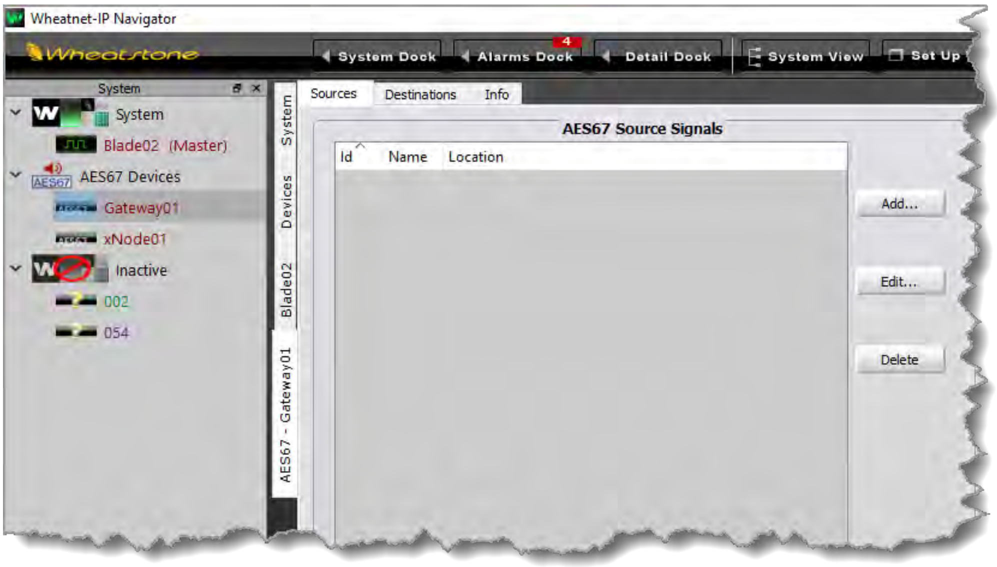
Configure a Tieline AES67 Source and Stream to a Blade

Note: In testing the multicast IP address range 239.192.1.xxx was used successfully.

1. Configure a Source in the Tieline codec.

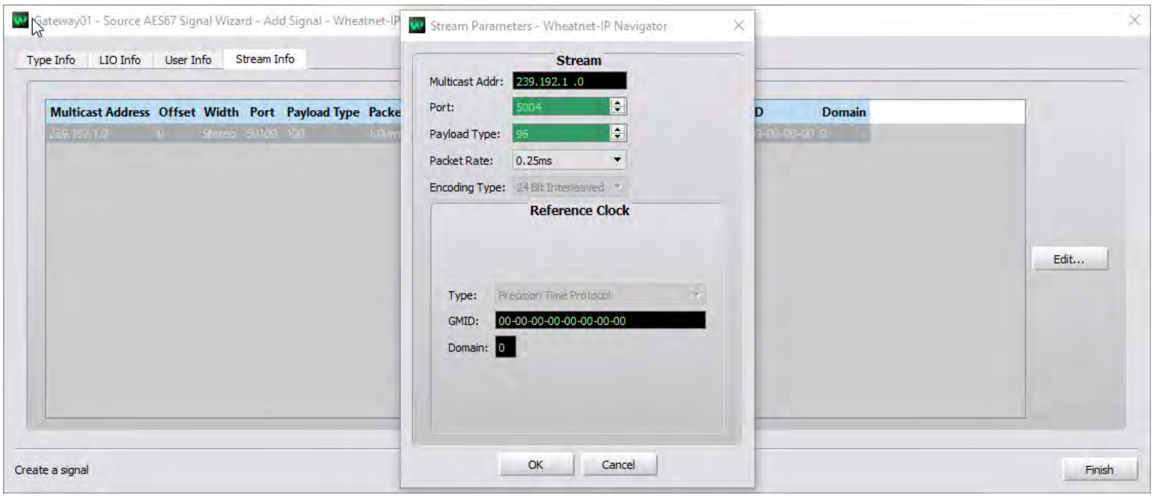


2. In Navigator select the Tieline codec under **AES67 Devices** and select the **Sources** tab. Then click the **Add** button.

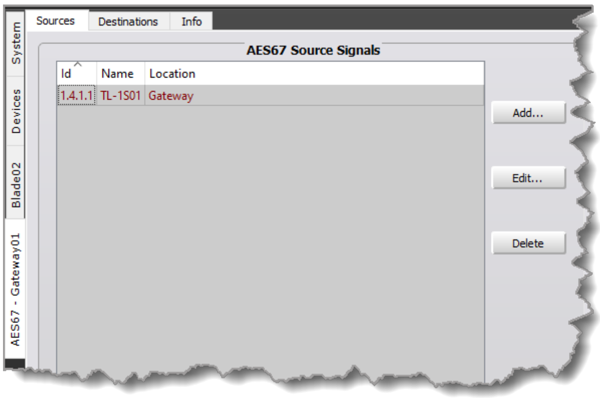


3. Name the Source stream and then select the **Stream Info** tab.

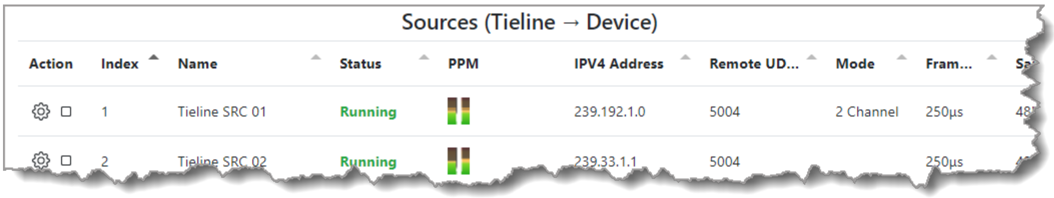
4. Add the multicast address and port details and click **OK** to create the Source stream, then click **Finish**.



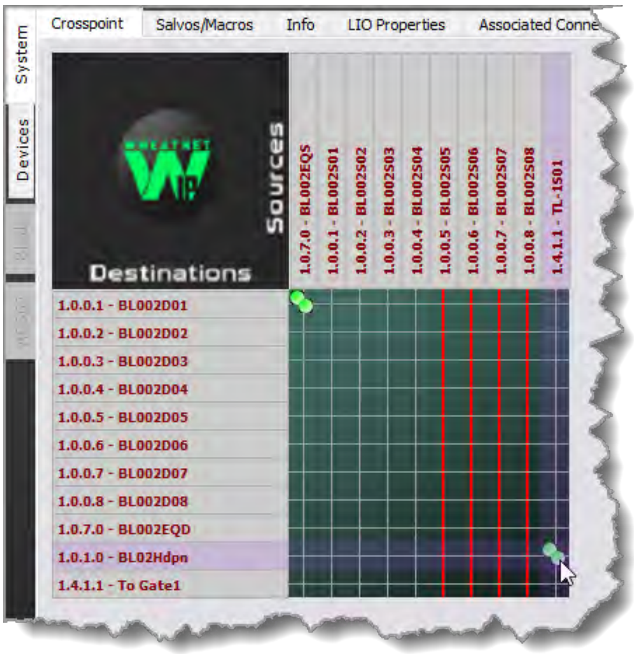
5. The Tieline codec AES67 Source stream is now configured in Navigator.



6. Click the **Start symbol** ▷ to start streaming.



7. The Tieline Source stream can now be monitored and routed as required using Navigator.



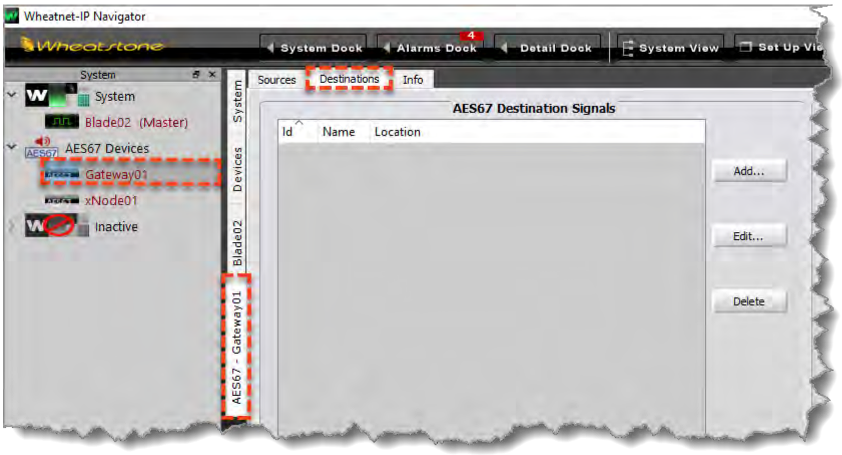
Configure a Tieline AES67 Destination and Receive a WheatNet-IP Source Stream

To configure a Blade Source signal to stream to a codec Destination it is necessary to:

1. Create an AES67 Source stream from the Blade.
2. Route it to the Tieline codec in Navigator.
3. Configure the Destination stream settings on the Tieline codec. Note: the WheatNet-IP multicast IP addresses in testing used addresses in the range 239.192.192.xxx.

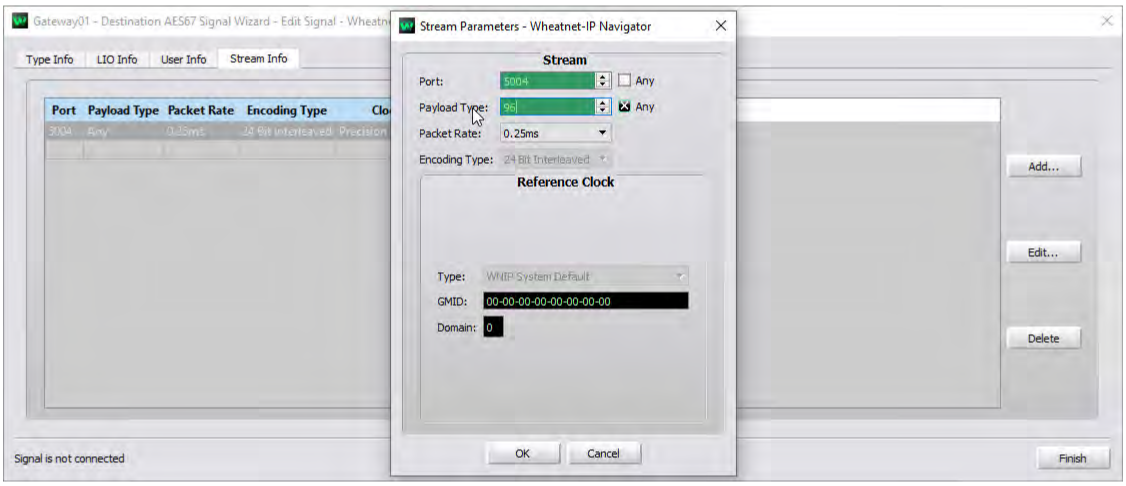
Create an AES67 Stream from a Blade

1. Select **AES67 Devices** and then select the **Destinations** tab.



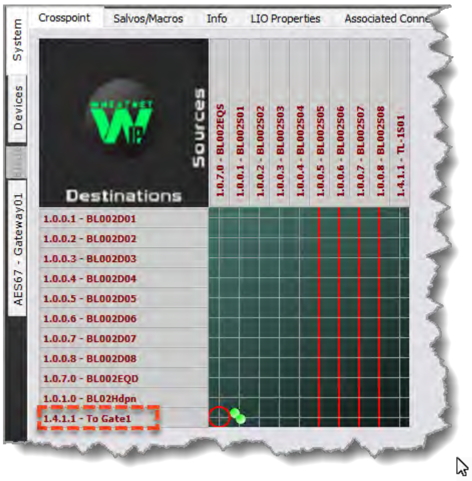
2. Click **Add** to open the **Destination AES67 Signal Wizard** and enter a name for the Destination.

3. Select the **Stream Info** tab and then **Edit**. Most settings can be left as the default configuration. The following settings were successfully configured and used to stream audio. Click **OK** to store the stream parameters, then click **Finish** to close the **Destination AES67 Signal Wizard** dialog.

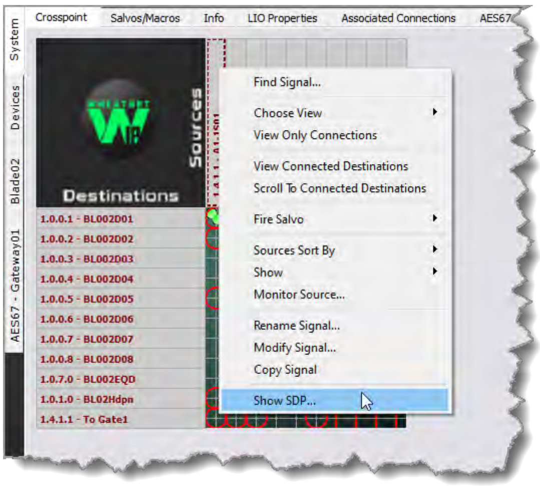


Note: By default WheatNet-IP uses port 50100 but this can be any supported port e.g. 5004, as long as the ports match on both devices connecting. The Tieline codec supports 250 microsecond packet timing.

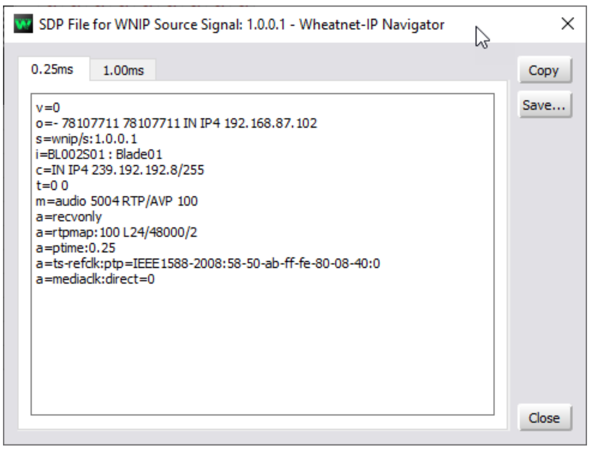
4. Select **System** and then the **Crosspoint** tab to view the newly created Tieline Destination just created displayed at the bottom of the **Destinations** list. Note: WheatNet-IP sources are not active until they are routed somewhere in Navigator.



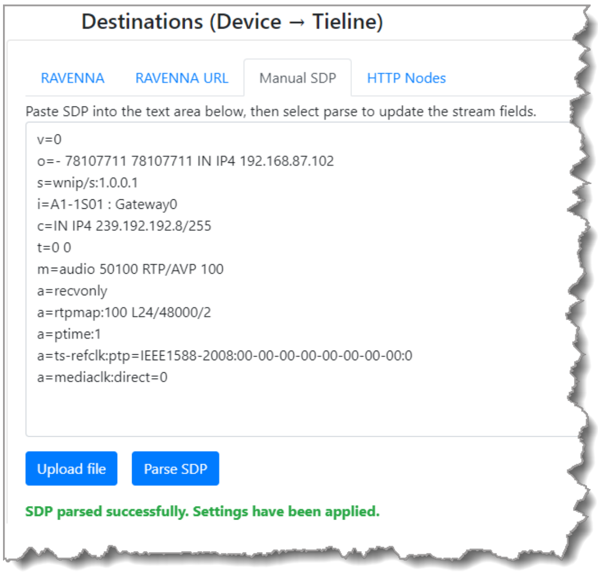
5. Right-click the source to be sent to the Tieline codec Destination and select **Show SDP**.



6. Copy the SDP file info.



7. Copy the SDP text into the **Parse SDP** text box in the Tieline codec **Destination panel** for a Destination, then click **Parse SDP**. Next click **Save**.



8. Click the **Start symbol** ▷ to start streaming.

AES67 Audio Streaming

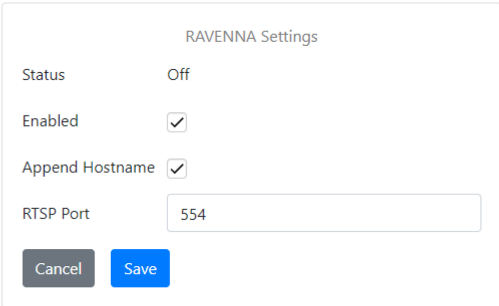
1.9 RAVENNA Streaming

RAVENNA is an AoIP technology developed by ALC NetworX. It is built upon open standards and extends AES67 and ST2110 by specifying a mechanism for stream discovery and advertisement. RAVENNA is used in many broadcast studios around the world.

Enabling RAVENNA in the Codec

To enable the codec to use RAVENNA:

1. Open the AoIP Web-GUI and select **Services** to display the **Services panel**.
2. Click **Edit** for **RAVENNA Settings** to adjust panel settings.
3. Select the **Enabled** check-box and then select the **Append Hostname** check-box to add the device hostname to RAVENNA Source names. Adjust the RTSP port used to pass SDP data as required.



Important Notes:

- Advertisement and discovery for RAVENNA streams is only available using the **AoIP 1** port. Streaming is available on both **AoIP 1** and **AoIP 2** ports.
- If identical Source names may exist on multiple devices, it is best to use the **Append Hostname** feature to add the device hostname, e.g. TLR6200-60007-AOIP to a Source. This will create unique RAVENNA names which is important. If this feature is not used and the same advertised RAVENNA names exist on different devices, some may either not be displayed, or the panel may display randomly appended RAVENNA names. This makes it very difficult to find and identify the correct stream.
- The codec hostname can be changed in the **AoIP Host Network panel** accessed from the **AoIP** menu in either the HTML5 Toolbox Web-GUI or the AoIP Web-GUI.

4. Click **Save** to store settings.
5. The **Services panel** displays when RAVENNA is running successfully.

Configure the Codec for RAVENNA Streaming

Before streaming audio from the codec using RAVENNA it is necessary to configure the AoIP mode correctly.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Audio** to display the **Audio Options panel**.
2. Click **Edit** to adjust settings.
3. Click the drop-down menu for **Audio over IP Mode** and select **Native AoIP Protocols**.
4. Click **Save**.

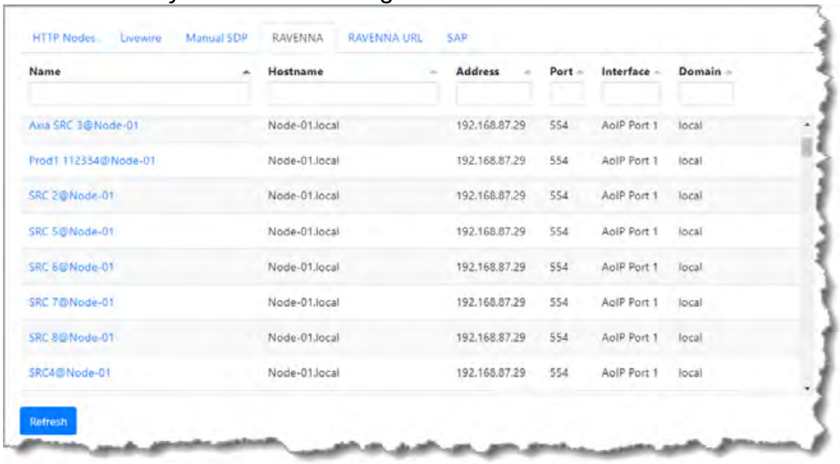
Configure a Tieline Destination to Receive a RAVENNA AoIP Stream

After RAVENNA has been enabled as a service in the codec it is simple to discover RAVENNA Sources from other devices over an AoIP network. Tabs are displayed in the **Destinations panel** for RAVENNA.

Configure a Destination using the RAVENNA Tab

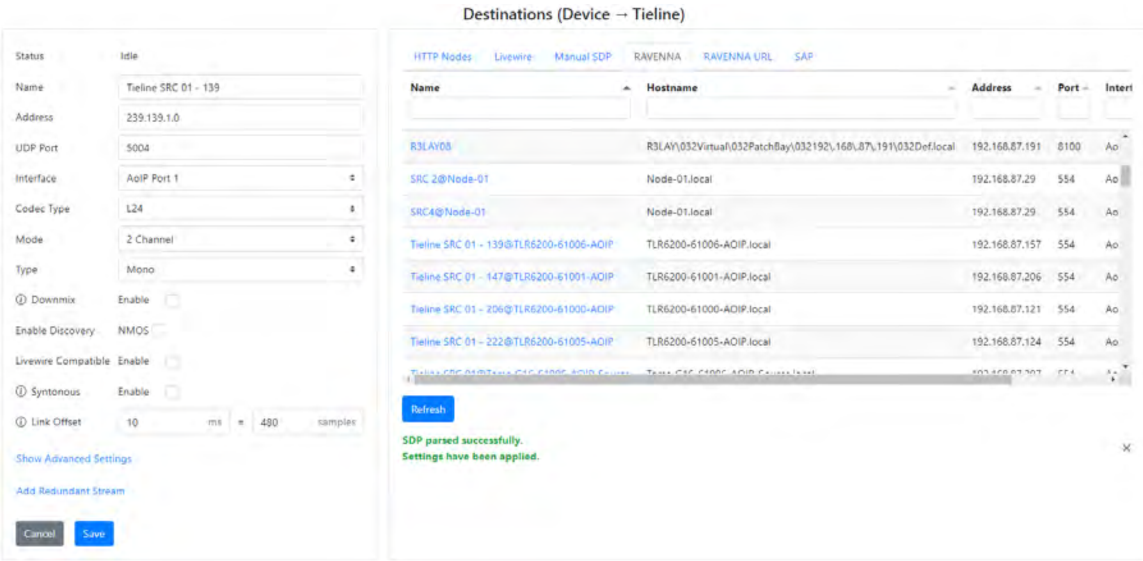
The **RAVENNA tab** displays available RAVENNA Sources. Also note:

- The **Hostname** is the Hostname of the actual device/box, e.g. codec.
- The **Address** is the IP address of the device.
- The **Port** is the RTSP port used to fetch SDP data.
- The **Interface** is the **AoIP Port** on the codec used to connect to the external device.
- The **Domain** is always local when using mDNS.



To configure a Destination using the **RAVENNA tab**:

1. Click **Refresh** to display all available RAVENNA Sources advertised on the AoIP network. At the top of the tab there are search fields to filter available Sources. Note: A progress bar may be displayed if a very large number of network streams need to be populated in the **RAVENNA tab**.
2. Select the **Name** of the preferred RAVENNA Source. Note: When a Source is selected, SDP data is fetched, parsed and populated in the fields within the **Destinations panel**. Confirmation that SDP has been successfully parsed is also displayed at the bottom of the **RAVENNA tab**.

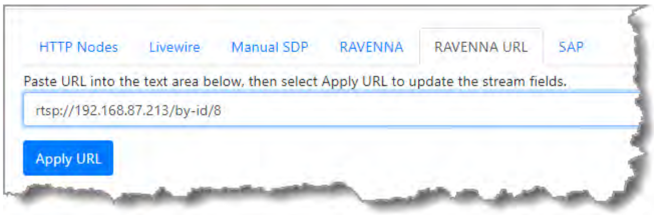


3. Click **Save** to store the new Destination settings.

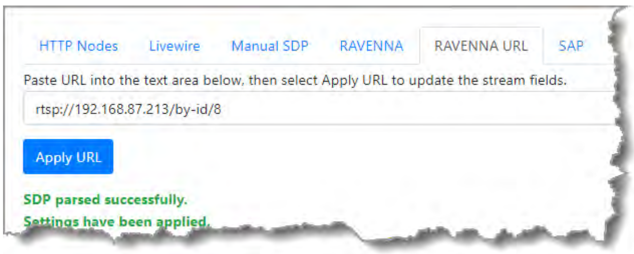
Configure a Destination using the RAVENNA URL Tab

Use the **RAVENNA URL** tab to copy or create a known RTSP URL address for a device and then paste it into the **RAVENNA URL** tab to populate data in the **Destinations panel**.

For example, a Tieline codec RTSP URL by stream name would use the format "rtsp://192.168.87.213/by-name/Tieline%20SRC%2009". The "%20" is used to replace the spaces in the codec name in this example, but is not essential. It is also possible to create an RTSP URL using the device stream ID, which is the index number of the stream, e.g. "rtsp://192.168.87.213/by-id/8". This example populates data for the 8th stream on the codec.



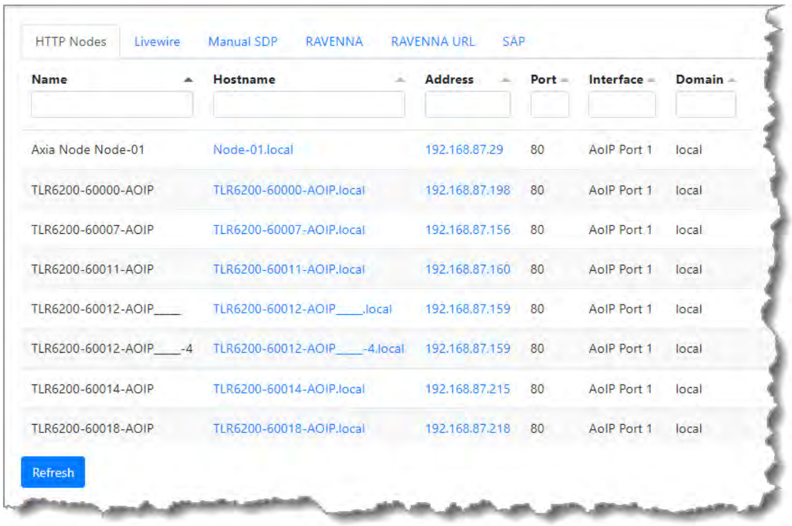
Click **Apply URL** to fetch, parse and populate SDP data in the **Destination panel** fields.



Configure Devices using the HTTP Nodes Tab

RAVENNA allows advertisement of web management interfaces controlling devices to which you are connecting. The **HTTP Nodes tab** displays **Hostname** and **Address** links to directly launch the device management interface for advertised RAVENNA devices on your AoIP network. This facilitates easy control of advertised devices across a network.

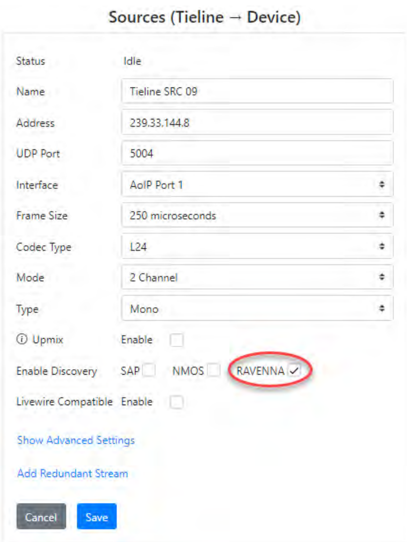
1. Click **Refresh** to display all available nodes advertised on the AoIP network. At the top of the tab there are search fields to filter available Sources.
2. Click **Hostname** or **Address** links to launch a device's browser-based interface.



Configure a Tieline Source as a RAVENNA AoIP Stream

To advertise a Tieline codec Source as a RAVENNA AoIP stream across an AES67 network it is necessary to:

1. Ensure RAVENNA as a service is **Enabled** and **Running** in the **Services panel**.
2. Configure Source settings and then select the RAVENNA check-box in the **Sources panel** and save these settings.



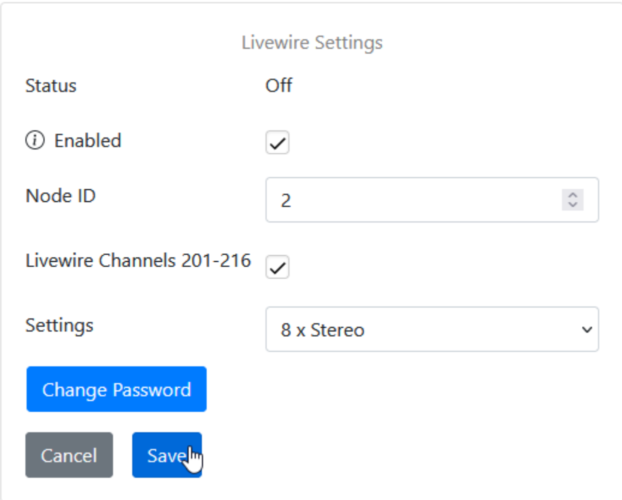
1.10 Enable Livewire and Use Pathfinder

Livewire is a proprietary AoIP system developed by the Telos Alliance and Livewire+ AES67 is the second generation of Livewire which is AES67-compliant. Tieline G6 codecs from firmware release v3.06.xx can interface natively with Livewire and Livewire+ AES67 systems over an AES67 LAN. This facilitates codecs supporting Livewire+ stream discovery and advertisement, as well as GPIO and Program associated data. For details on Livewire clock settings see the Clocking section of this manual.

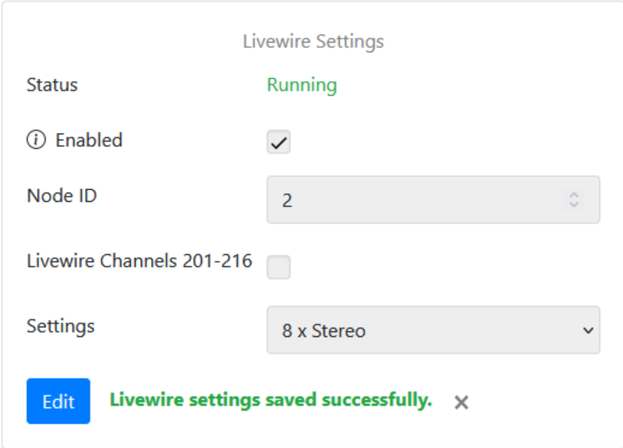
Enabling Livewire+ in the Codec

To configure the codec to use Livewire+:

1. Open the AoIP Web-GUI and select **Services** to display the **Services panel**.
2. Click **Edit** in the **Livewire Settings** panel to adjust settings. Note: Click **Change Password** to enter a Livewire password for Pathfinder software.
3. Livewire devices provide a fast setup routine using a Node ID to assign I/O channel numbers quickly. Enter a unique Node ID with a value within the range 1 - 319. Note: 16 sequential Livewire Channels are assigned from the entered **Node ID**.
4. If the check-box for **Livewire Channels** is selected, the Source channel numbers and addresses are automatically assigned within the codec.



5. Use the Settings drop-down menu to configure Source streams. Selections include:
   a. **Custom** only changes the IP addresses of existing Livewire Channels.
   b. The other options configure the codec for the preferred number of inputs per Livewire Channel (stream), e.g. **8 x Stereo**, **16 x Mono**, or **2 x Surround**. Note: Livewire Channels available depend on how many inputs/outputs are supported in the codec, e.g. 4, 8 or 16.
6. Select the **Enabled** check-box and then click **Save**. Note: It can take up to a minute for Livewire to start running.



Important Notes:

- It is not possible to enable Livewire as a service if the sample rate is not configured as 48kHz.
- Assign a Node ID and Livewire channels in the codec first before creating any RAVENNA, SAP, or AES67 streams.
- When a Node ID is configured in the codec any existing Livewire channels and streams are reconfigured.

- Livewire Channel Numbers are assigned in blocks of 8.
- 16 sequential Livewire Channels are assigned from the entered **Node ID**, even if a codec supports fewer inputs and outputs.

Configure the Codec for Livewire Streaming

Before streaming audio from the codec using Livewire+ it is necessary to configure the AoIP mode correctly.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Audio** to display the **Audio Options panel**.
2. Click **Edit** to adjust settings.
3. Click the drop-down menu for **Audio over IP Mode** and select **Native AoIP Protocols**.
4. Click **Save**.

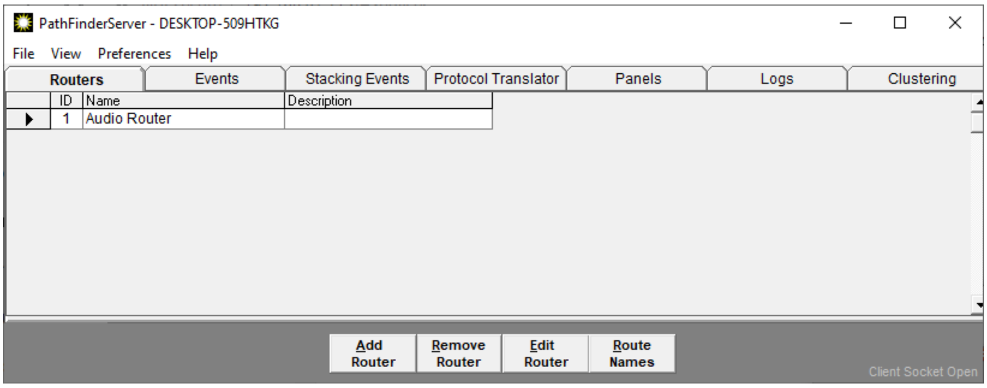
Configuring a Codec for Use with Pathfinder

Pathfinder Server communicates with all Livewire system nodes and supports virtual routing, audio metering, silence detection, and Livewire GPIOs. Livewire systems can also be controlled using Pathfinder PC.

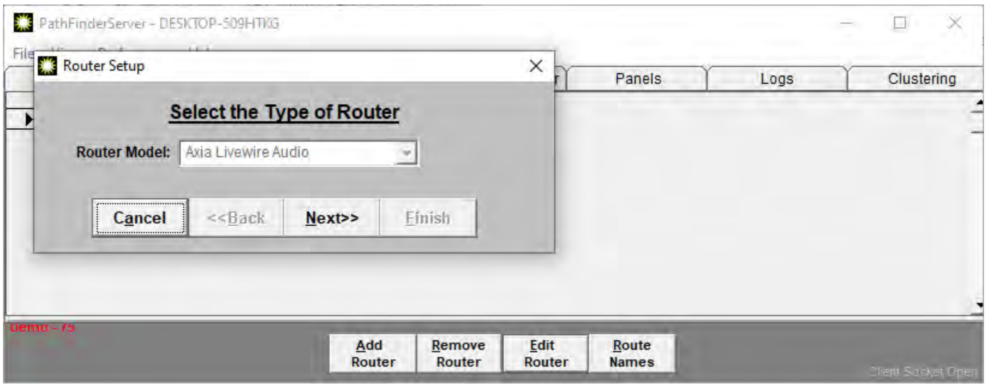
Adding a Tieline Codec to Pathfinder Server

When you attach a new codec to your network you need to add it using Pathfinder Server. Ensure that it has been added to Pathfinder Server.

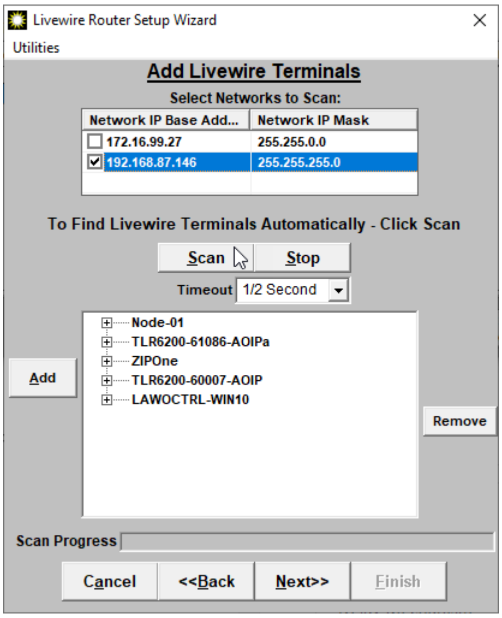
1. Launch Pathfinder Server software.



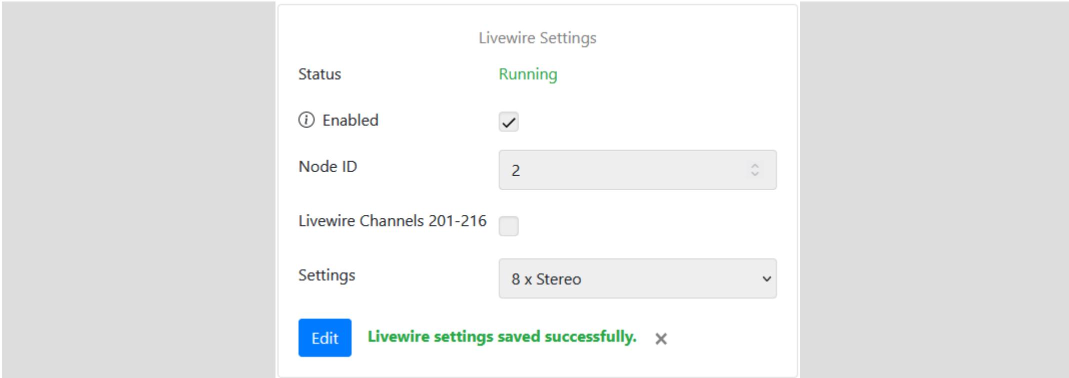
2. Click **Edit Router** and select **Next**.



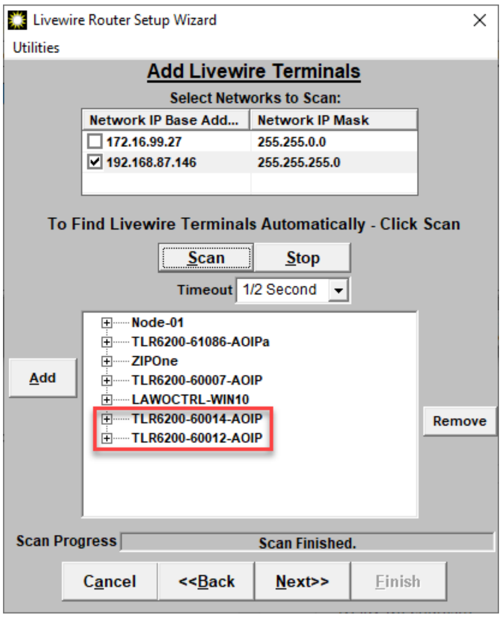
3. Click **Next** again and select the **Network to Scan**, then click **Scan**.



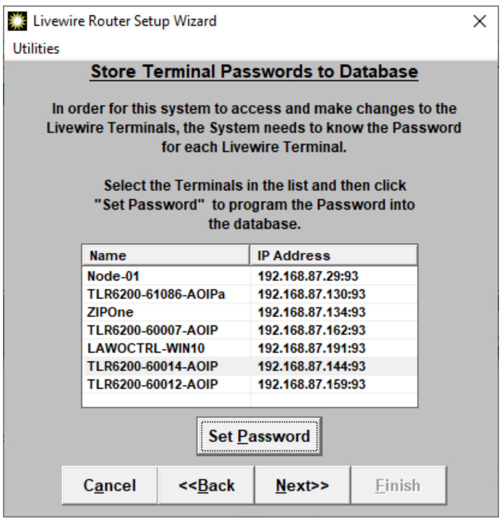
Important Note: Ensure Livewire is enabled in the Services panel of each codec before performing a network scan.

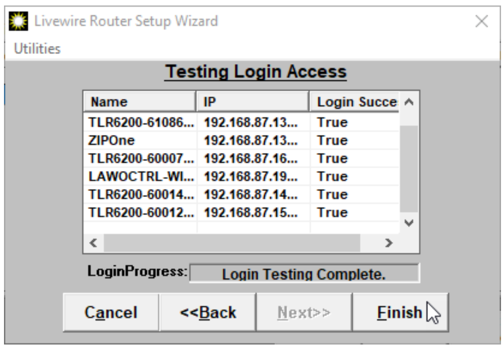


4. Pathfinder Server will scan the AoIP network for new devices and any detected will be displayed (codec model and serial number), then click **Next**. Note: If devices are not automatically detected click **Add** to manually add them into Pathfinder Server.

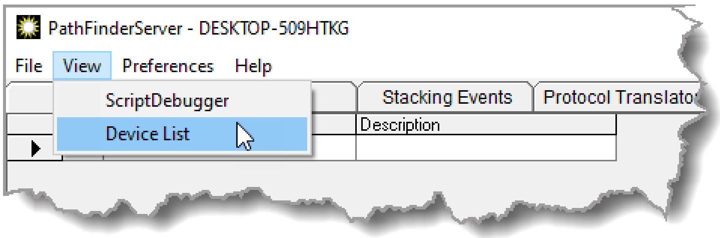


5. Set the passwords for each new device, then click **Next**. and then **Finish** in the **Testing Login Access** dialog.

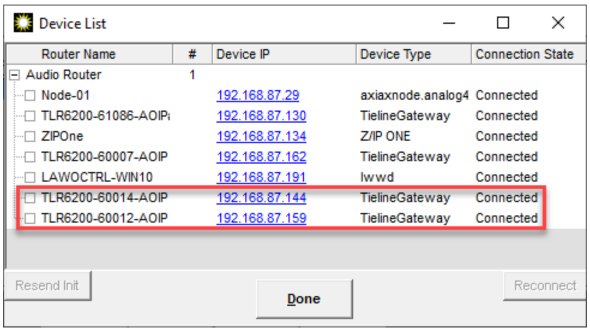




6. To double-check that the codec has been added to Pathfinder Server select **View > Device List**.

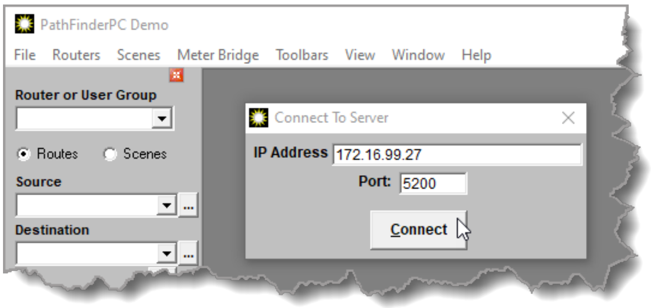


7. In the following image the newly added codecs display as "TielineGateway".

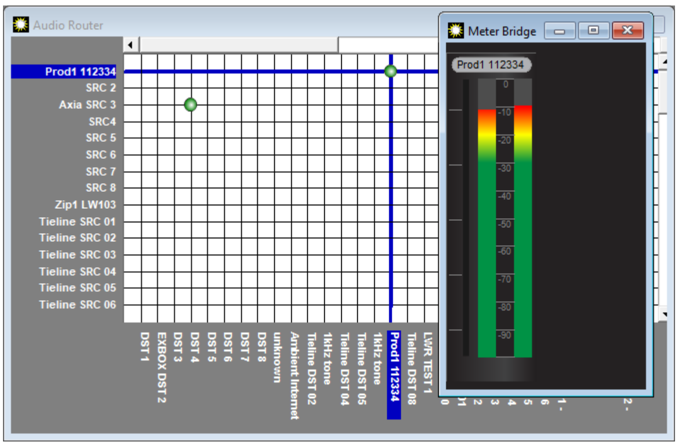


Using Pathfinder PC

After adding your codec to Pathfinder Server, launch Pathfinder PC and select **File > Connect to Server** and click **Connect**.



It should now be possible to route Sources and Destinations through the codec using PathfinderPC.



1.10.1 Livewire+ Streaming

Product	Feature	Notification
Gateway 4	Livewire streams	Supports up to 4 mono or two stereo streams.
Gateway 8/16	Livewire streams	16 channel Gateway supports up to 16 mono or 8 stereo streams; 8 channel Gateway supports up to 8 mono or 4 stereo streams.
MPX I	Livewire streams	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.
MPX II	Livewire streams	Configure up to 4 mono or 2 stereo Source/Destination streams.
Bridge-IT II and Bridge-IT XTRA II	Livewire streams	Configure two mono Source/Destination streams, or a single stereo Source/Destination stream.

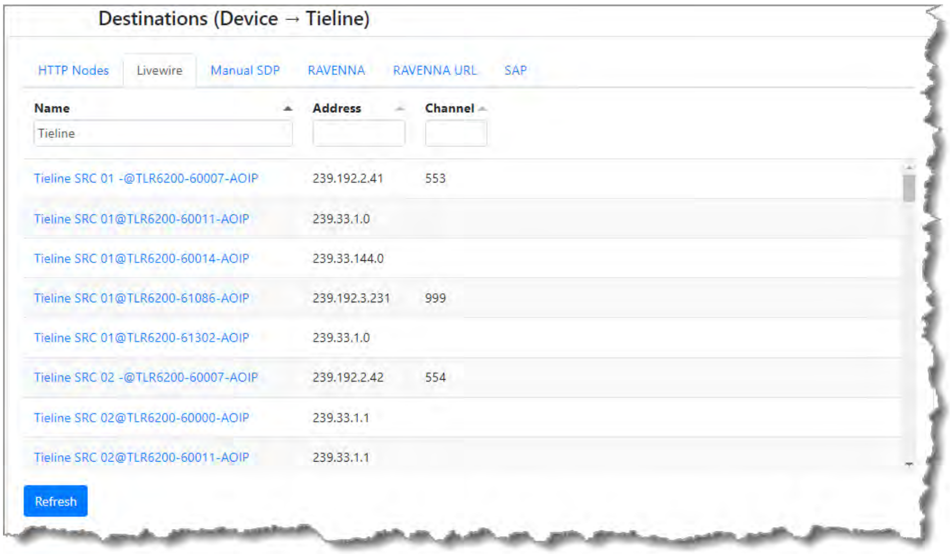
Livewire Compatible Destinations: Receiving a Livewire+ Stream

The **Livewire tab** in the **Destinations panel** lists all available Livewire+ Sources.

Configure a Destination using the Livewire Tab

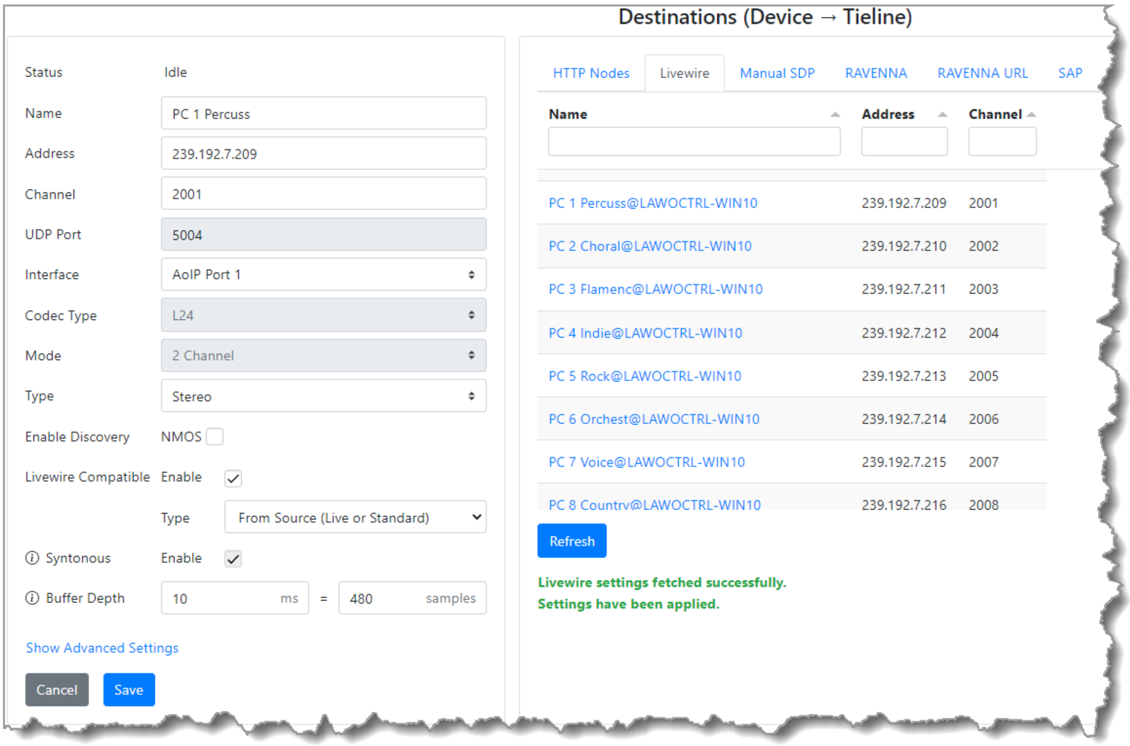
The **Livewire tab** displays available Livewire+ Sources alphabetically. At the top of the tab there are search fields to filter available Sources. Note: A progress bar may be displayed if a very large number of network streams need to be populated in the tab.

- The **Name** is the name of the stream.
- The **Address** is the multicast IP address used to create the Destination stream.
- The **Channel** is the unique channel number used by Livewire to identify streams.



To configure a Destination using the **Livewire tab**:

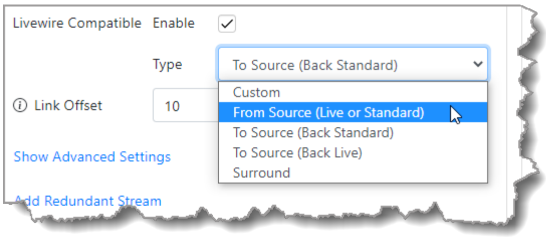
1. Click **Refresh** to display all available Livewire Sources advertised on the AoIP network.
2. Select the **Name** of the preferred Livewire Source. Note: When a Source is selected, SDP data is fetched, parsed and populated in the fields within the **Destinations panel**. A message at the bottom of the **Livewire tab** confirms that SDP has been successfully applied.



3. Click **Save** to store the new Destination settings.

Livewire Compatible Stream Type

The **Livewire Compatible** stream **Type** is typically configured when a Livewire stream is parsed using the **Livewire tab**. It can also be manually configured using the **Type** drop-down menu.



Type drop-down menu

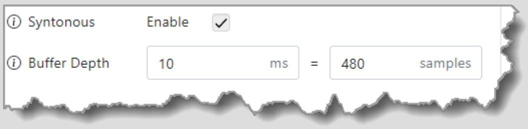
A stream type can be defined as:

1. Custom: This is an AES67 mode that allows you to override and modify Livewire settings which are non-standard.
2. From Source: Also known as Livestream or Standard streams using the Livewire address range 239.192.000.0/15
3. To Source: Also known as Back Standard Stereo Streams using the Livewire address range 239.193.000.0/15
4. To Source: Also known as Back Live streams using the Livewire address range 239.193.000.0/15
5. Surround: Surround streams using the Livewire address range 239.196.128.0/15

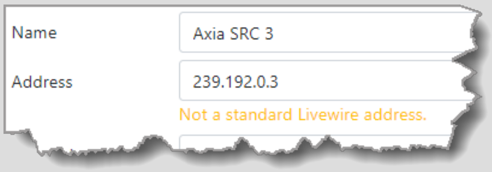
Stream type is explained in more detail in the following topic about creating a Livewire+ Source.

Important Notes:

- When a **Livewire Compatible** stream is configured the **Downmix** option is only selectable for a **Custom** stream type. Other stream types are automatically configured.
- When configuring a mono Destination stream, if a stereo RAVENNA, SAP or Livewire stream is parsed, then Downmix will be automatically selected for compatibility.
- Syntonous receive mode is enabled by default in Livewire Destinations. In Syntonous mode the sampling frequency is locked, but the absolute time reference at the sender and receiver may be different. If you experience streaming issues adjust the buffer depth to suit network latency conditions.



- If a non-standard Livewire address is configured then a message will be displayed below the **Address** field in the **Destinations panel**.



Livewire Channels

Livewire uses multicast addresses for AES67 and unique addressing Channels for Livewire, which are automatically calculated. Livewire channels range from 0 to 32767.

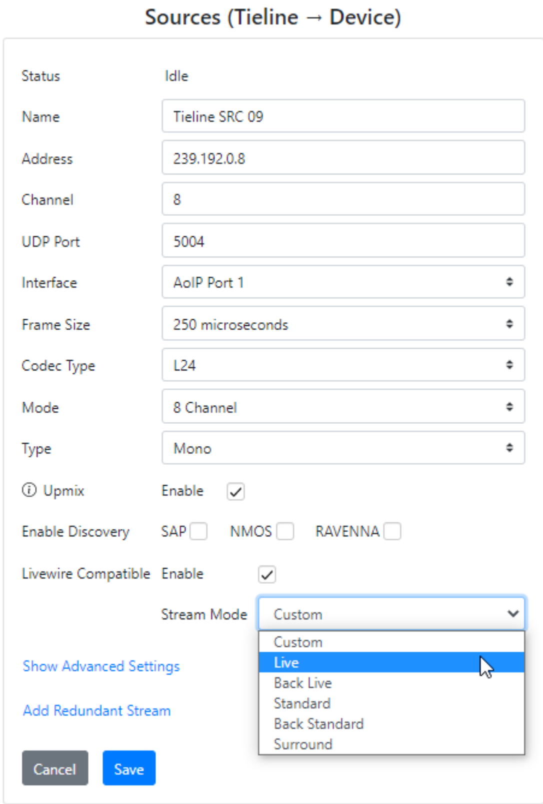
If we assume a multicast IP address is 239.192.35.48 then the Livewire Channel number is calculated as follows:

IP address = 239.192.35.48
To calculate multiply 35 x 256 + 48 = Channel number **9008**

Configure a Tieline Source as a Livewire+ Stream

To advertise a Tieline codec Source as a Livewire+ AoIP stream across an AES67 network it is necessary to:

1. Ensure Livewire as a service is **Enabled** and **Running** in the **Services panel**.
2. Configure Source settings including the **Channel** number and then enable the **Livewire Compatible** check-box in the **Sources panel**.
3. Select the preferred Livewire+ **Stream Mode** to suit the type of Source being configured.
4. Click **Save** when all settings are configured.



When the stream mode is selected it configures the Livewire address range and packet structures. The settings include:

1. Custom: This is an AES67 mode that allows you to override and modify Livewire settings which are non-standard.
2. Live (also known as Livestream) streams use the Livewire address range 239.192.0.0/15: Total bytes per packet = 72; Core delay = .25ms.
3. Back Live streams use the Livewire address range 239.193.0.0/15: Total bytes per packet = 72; Core delay = .25ms.
4. Standard streams use the Livewire address range 239.192.0.0/15; Total bytes per packet = 1440.; Core delay = 5ms.
5. Back Standard Stereo Streams use the Livewire address range 239.193.0.0/15; Total bytes per packet = 1440.; Core delay = 5ms.
6. Surround streams use the Livewire address range 239.196.128.0/15: Total bytes per packet = 1440; Core delay = 1.25ms.

Important Notes: When a **Livewire Compatible** stream is configured the **Upmix** option is only selectable with the **Custom** Stream Mode. Other options are automatically configured.

1.10.2 Livewire GPIOs

64 Livewire GPIO ports are supported in the codec. Livewire uses ports to activate GPIOs and ports 1-8 are configured with 5 GPIs and GPOs. Port 9-16 are configured with 3 GPIs and GPOs. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs as displayed in the **Livewire GPIO panel** in the AoIP Web-GUI.

Livewire GPIO panel displaying the first 10 GPIOs

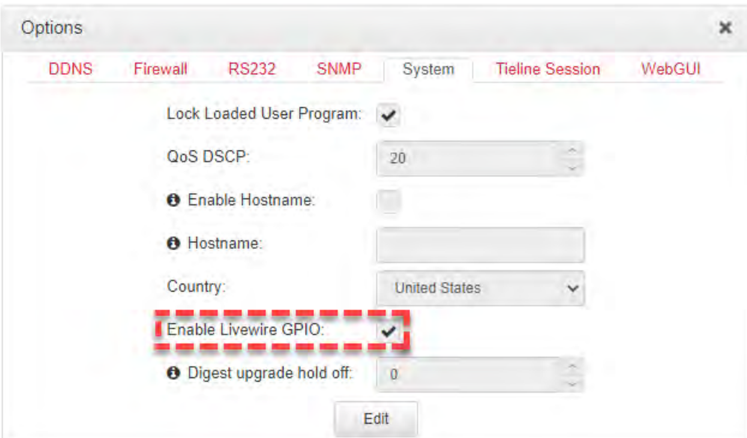
In general Livewire devices think of GPIs as signals going to the network and GPOs as signals coming from the network. A Livewire port can be configured in 3 ways:

1. Port routing with Tieline's browser-based AoIP Web-GUI: GPI activity on a port will trigger the GPO of the port being configured. To do this enter the IP address of the device, followed by a forward slash and the port number.
2. Channel assignment with Tieline's browser-based AoIP Web-GUI: The GPIO port will follow the logic defined by a control surface (or the codec) through the use of the unique channel number loaded to a fader or other console functionality.
3. Pathfinder PC Server software can monitor and control GPIO pins on any of these ports using its event system. It can also route closures using the IP address and port number.

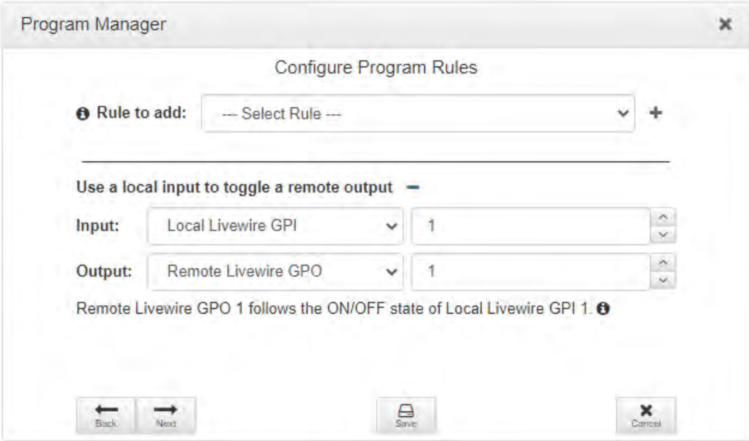
In the IP Address/Port example, the GPIO device opens a TCP connection to the routed GPIO device and monitors the GPI pins tripping its own GPO pins. In the Livewire Channel number example, GPIO messages are sent as multicast messages in the same way as multicast audio traverses the network. A device monitors the GPIO multicast channel for closures tagged with the correct Livewire Channel number.

Configuring a Livewire GPIO with the Codec Web-GUI

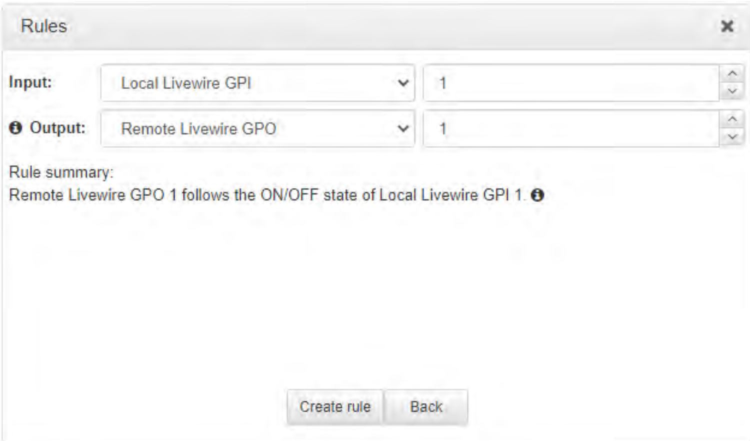
Livewire GPIOs can be configured using the Tieline HTML5 web-GUI. First, select the **Enable Livewire GPIO** checkbox in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs.



Program and stream level Livewire GPIO rules can be configured using the **Program Manager panel** when creating new programs.



The **Rules panel** can also be used to create Livewire GPIO rules for hardware or virtual IOs.



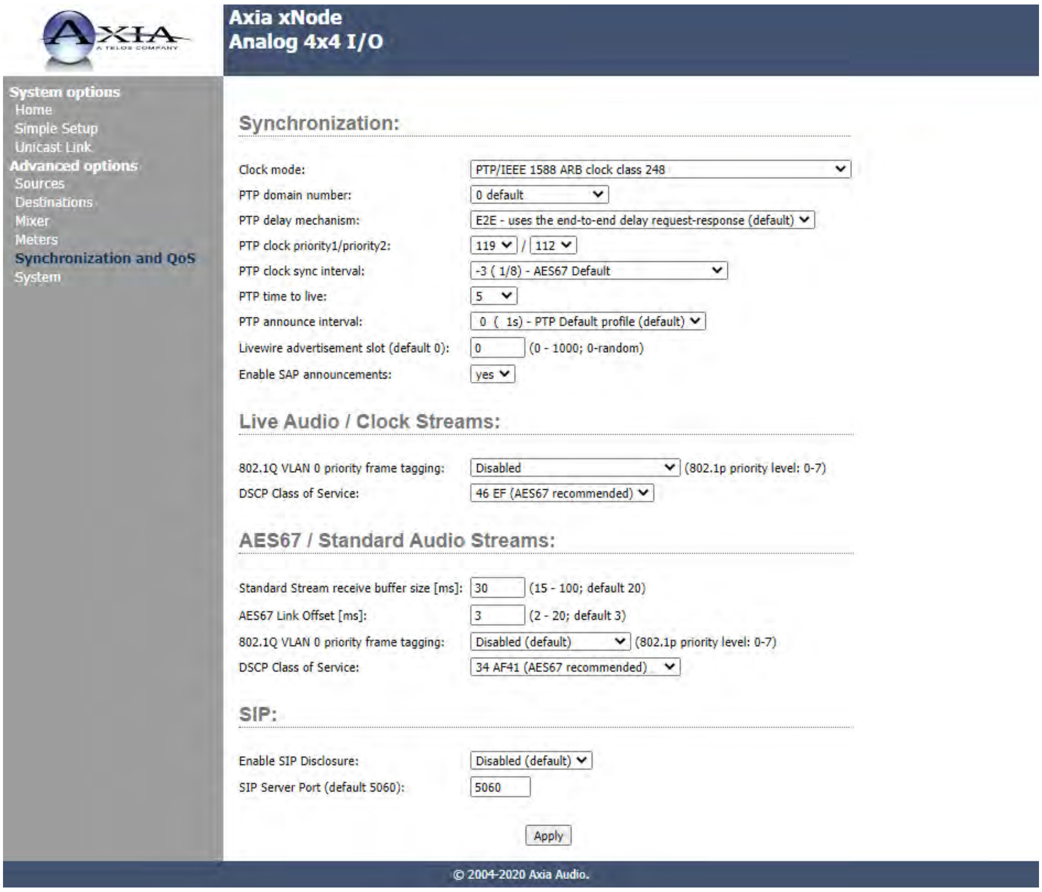
See the "Creating Rules" section in the codec user manual for more detailed information on configuring rules.

1.10.3 AES67 Streams and Livewire

Tieline recommends sending and receiving Livewire+ streams in natively supported **Livewire Compatibility** mode. It is also possible to send and receive AES67 streams between the codec and Axia Livewire systems. The following AES67 configuration information was tested using an Axia xNode with firmware version 2.4.7.

xNode Synchronization Settings and SAP

During testing the following settings were applied to the xNode and SAP Announcements (in the Synchronization menu) were enabled to allow discovery.

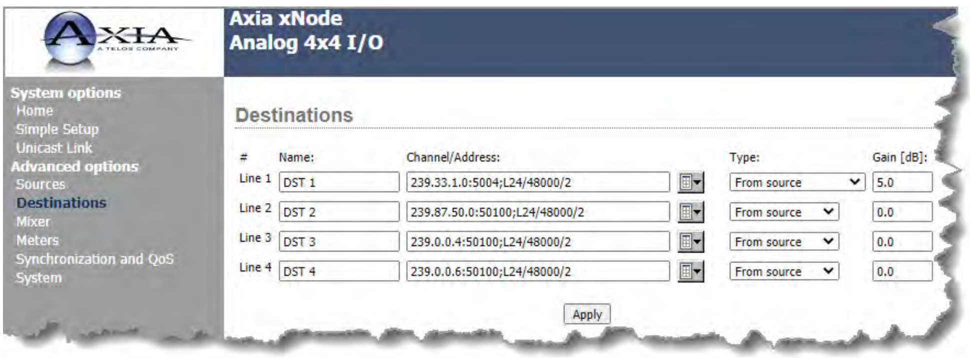


Configure a Tieline Source and an xNode Destination

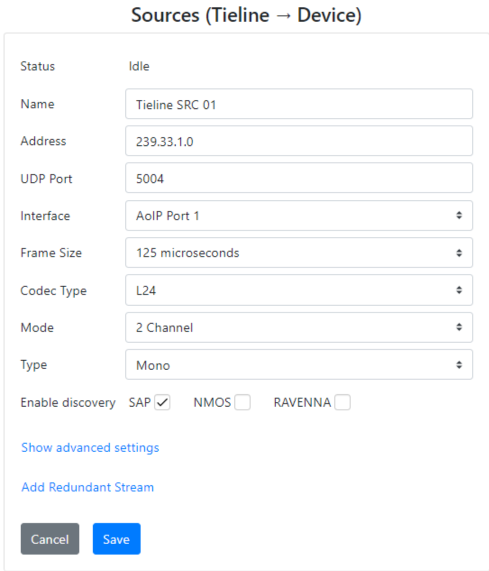
The Destinations page in the Axia xNode web-GUI defines the audio outputs of an xNode. There are 4 stereo output ports and these outputs may be the physical outputs of the unit, or network stream inputs, such as an AES67 Source stream from a Tieline codec. To receive the Source audio from the Tieline codec the xNode destination **DST 1** in the following image is configured to connect to:

- Multicast address 239.33.1.0.
- UDP port 5004.
- L24 (24 bit) sampling at 48kHz in stereo.

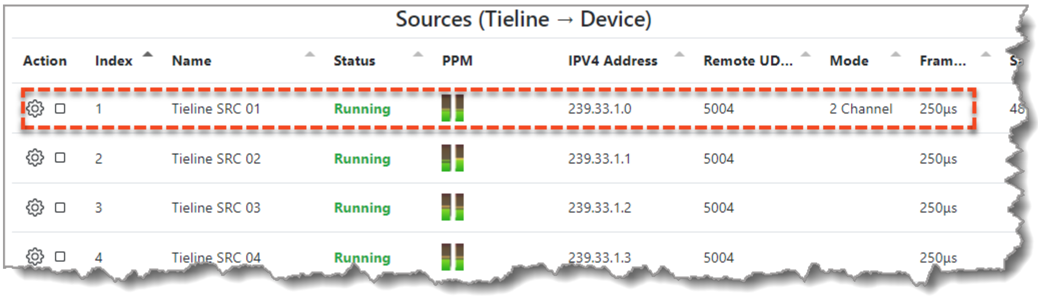
After configuring the xNode click **Apply**.



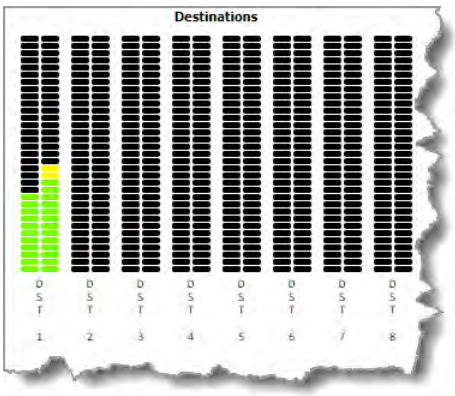
The Tieline codec Source configuration using the Tieline AoIP Web-GUI for **Tieline SRC 01** is as follows:



Once configured, click the **Start symbol** ▷ to commence streaming. In the following configuration example, the Tieline codec is streaming Source audio to the xNode over multicast address 239.33.1.0.

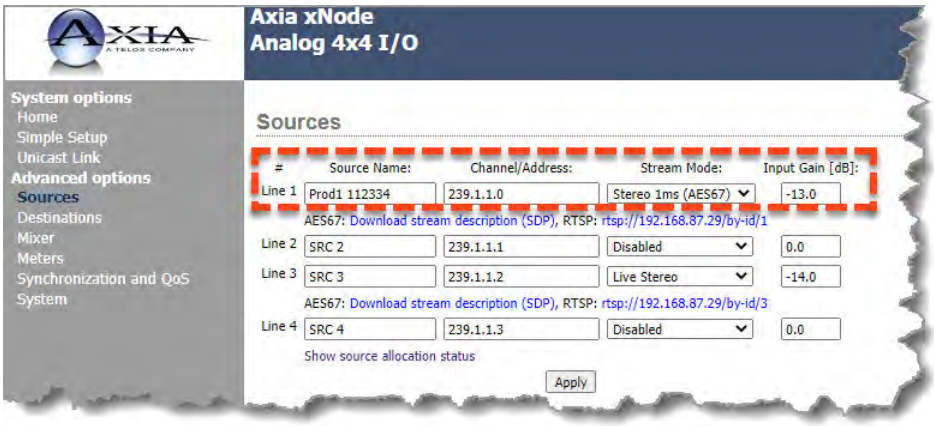


In the xNode audio is received and can be monitored in the xNode web-GUI **Meters** page via **DST 1** in this example.

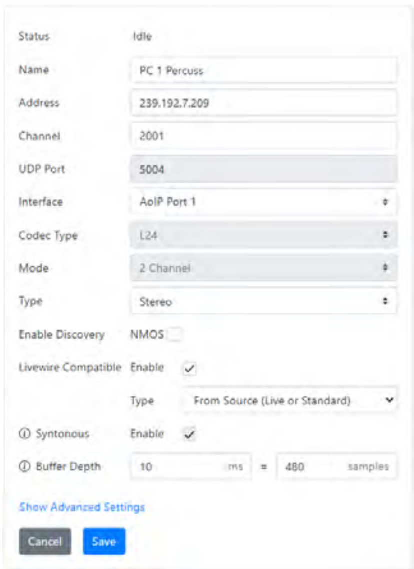


Configure a Tieline Destination and an xNode Source

The xNode can transmit a multicast stream to a Tieline codec once a Source has been configured with a multicast IP address and the Stream Mode is set to a value other than disabled, e.g. **Stereo 1ms (AES67)**, or **Live Stereo** which supports 250 microsecond packet timing. After configuring the xNode click **Apply**.



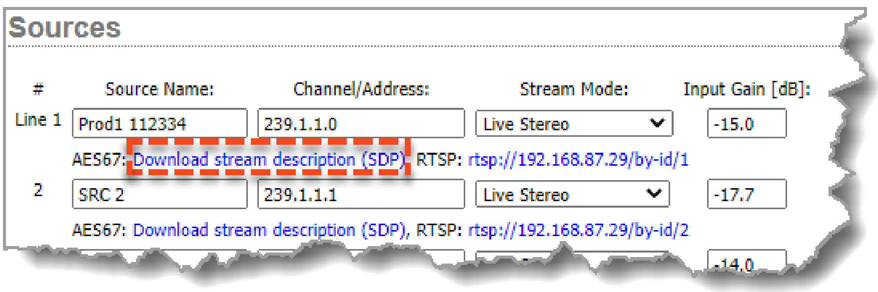
An example of the Tieline codec Destination configuration using the Tieline AoIP Web-GUI is as follows:



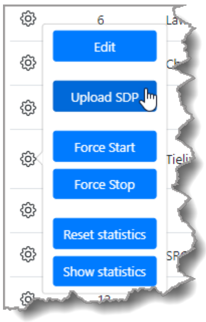
Download an SDP File from an xNode

To download the configuration info for an audio stream from an xNode:

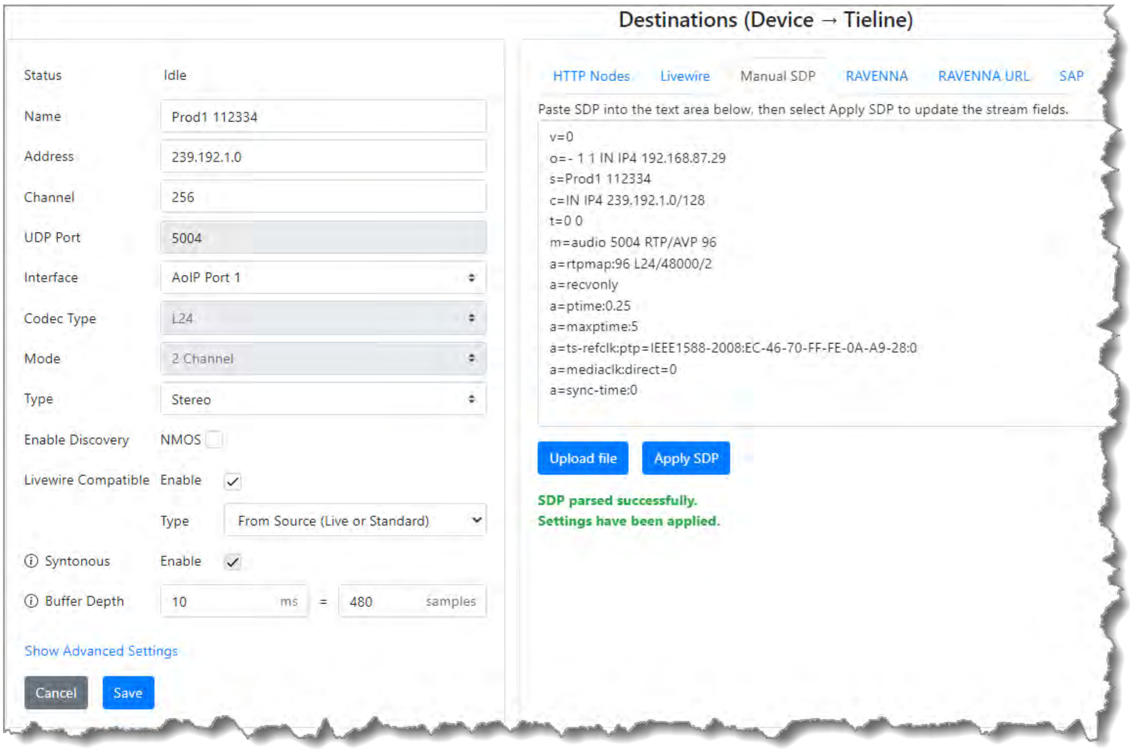
1. Click **Download stream description (SDP)** for a Source audio stream on the xNode and save it to a PC.



2. On the Tieline codec open the **Destinations panel** and ensure the stream being configured is not active. Click the **Configuration symbol** ⚙ for a Tieline Destination audio stream and select **Upload SDP** and then select the xNode SDP saved to the PC.



3. The SDP settings within the file are parsed and applied to the Tieline codec Destination audio stream as displayed.



4. Click **Save** to store these settings.
5. Click the **Start symbol** ▷ to receive stream audio. In the following configuration example, the Tieline codec is receiving audio from the xNode Source via multicast address 239.1.1.0.

Troubleshooting

During testing, in 8 channel streaming mode the xNode required an RTP **Payload Type** of 99 and testing was successful with a 1ms frame size. The **Payload Type** setting is available in the **Advanced Settings** menu in the **Sources panel**. I think they only support 1ms packets in 8 channel mode.

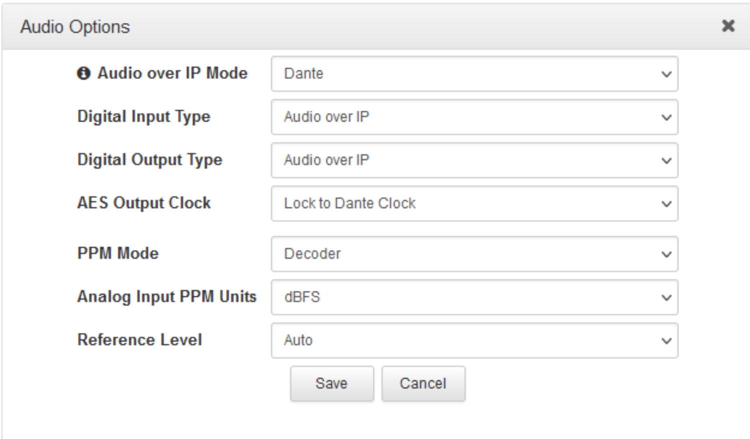
1.11 Dante Streaming

A Tieline Gateway with an optional Dante card on-board can interface directly and seamlessly with other Dante compatible devices. Audio stream source and destination routing can be directly configured using Dante Controller software.

Configuring the Codec for Dante

Before streaming audio from the codec using Dante it is necessary to configure the codec **Audio over IP Mode** correctly.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Audio** to display the **Audio Options panel**.
2. Click **Edit** to adjust settings.
3. Click the drop-down menu for **Audio over IP Mode** and select **Dante**.



Important Notes:

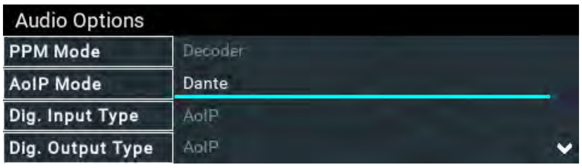
- If Dante is not displayed as an **Audio over IP Mode** option then a Dante license may not be installed in the codec. There are two ways to update this:
  - Check the **Licensing panel** in the Toolbox HTML5 Web-GUI to verify if the Dante license is listed. If it is not listed, and the codec is connected to the internet, click **Get license file from TieServer** to install the Dante license.
  - Using the codec front panel screen navigate to **Settings > System > Licenses** and press the **OK** button for **Update From TieServer**.
- If the codec is not connected to the internet, contact Tieline and notify us of the serial number of your codec to request a Dante license file. Then use the **Upload a selected file** option in the **Licensing panel** to upload the license file provided by Tieline.
- If Dante is displayed as an **Audio over IP Mode** option, but not selectable, then the Dante card is likely not fitted, or not fitted correctly.

4. Both the **Digital Input Type** and **Digital Output Type** in the **Audio Options panel** can be configured for **Audio over IP**. Note: It is possible to configure the codec to use Dante AoIP inputs and AES3 outputs or visa versa. In these situations the **AES Output Clock** should be set to **Lock to Dante Clock**, unless the selected clock is definitely synchronized to the Dante clock. In most cases, the **AES Output Clock** should be set to **Lock to Dante Clock**.
5. Click **Save** to save settings.

The Audio over IP Mode can also be configured using the front panel of the codec:

1. Press the **SETTINGS** button.

2. Use the navigation buttons to select **Audio Options** and press the button.
3. Select **AoIP Mode** and press the button.
4. Select **Dante** and press the button.



Important Notes:

- When the **AoIP Mode** is changed we strongly recommend rebooting the codec using either the **Reset / Backup panel** in the Toolbox Web-GUI, or by navigating to **Settings > Reset & Backup > Reboot Codec** via the codec front panel.
- Advertisement, discovery and streaming is only available using the **AoIP 1** port when using Dante. Redundant AoIP streams are not supported when using Dante.
- Native AES67 will not run when the **Audio over IP Mode** is set to **Dante**.

Configure Flows with Dante Controller

1. Launch Dante Controller software on your PC after configuring **Dante** as the **AoIP Mode**.
2. The codec should appear in the Device Info tab.

3. Double-click the device to open the Device View dialog and configure Dante flows.

Important Notes:

- The **AoIP Sources panel** and **AoIP Destinations panel** are disabled in the AoIP Web-GUI and HTML5 Toolbox Web-GUI when the codec is configured for Dante AoIP Mode.
- The **Synchronization panel** is disabled in the AoIP Web-GUI and HTML5 Toolbox Web-GUI when the codec is configured for Dante AoIP Mode. Dante handles clocking automatically via election and the codec will automatically sync to the Dante clock. Over a Dante network a "Primary Leader Clock" device is automatically assigned, even if more than one "Preferred Leader" is configured.

1.11.1 Dante AES67 Streaming

Dante is a proprietary AoIP system developed by Audinate which also supports AES67 streaming. Tieline AES67 enabled codecs support streaming AES67 audio between Dante enabled devices. Dante Controller is the software interface used to manage Dante devices and to establish audio stream routing between devices. Dante Controller is a free download and AES67 and SMPTE ST 2110-30 RTP streaming is supported from v4.2.x of this software.

For a Dante device to connect using AES67 it needs to be AES67 compatible and not all devices are. Sometimes a firmware upgrade may be possible to add AES67 support, so verify if this is possible if AES67 is not available in a Dante device.

Sources, Destinations and Flows

Tieline and other AES67 compatible devices usually refer to multicast audio streams as Sources and Destinations, whereas Dante calls these streams "flows." Dante devices in AES67 mode can transmit and receive AES67 multicast flows to and from non-Dante AES67-enabled devices. However, between Dante devices Dante's native audio transport protocol is used, even when AES67 is enabled on the Dante devices.

Dante Clocking

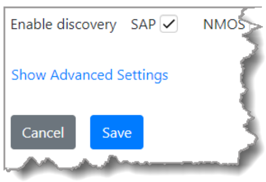
Dante handles clocking automatically via election and each Dante device includes a clock using IEEE1588 PTP. Over a Dante network a "Primary Leader Clock" device is automatically assigned, even if more than one "Preferred Leader" is configured. More information on clocking is available in the Clocking section of this manual.

Discovery: Displaying AES67 Sources within Dante Controller

To display Tieline codec Sources within Dante Controller:

1. In the AoIP Web-GUI select **Sources** (or in the HTML5 Toolbox Web-GUI click **AoIP** and then **Sources** to open the **Sources panel**).

2. Select a source, then select **Settings** and select the **Enable Discovery** checkbox for **SAP** in the **Sources panel**.

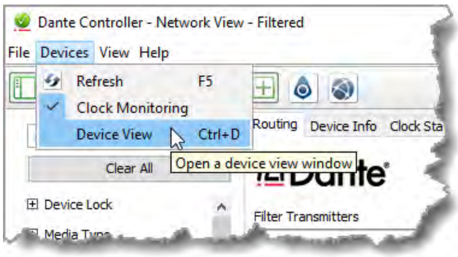


3. Save the setting, then start the Source stream and ensure it is **Running**.

Setting up a Dante Device for AES67 Operation

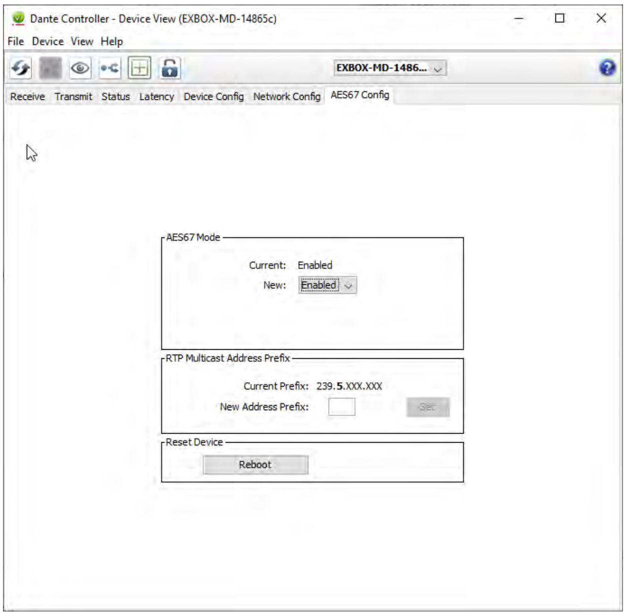
It is necessary to configure an AES67 compatible Dante device to stream in AES67 mode. In the following example the device connected to Dante Controller is an EXBOX.MD AoIP to MADI converter. It is assumed that the user has some knowledge of how to use Dante Controller. Note: The following procedures were created using Dante Controller version 4.3.3.15.

1. In Dante Controller select **Devices > Device View**.



2. Select the device to which the codec will send or receive AES67 streams.

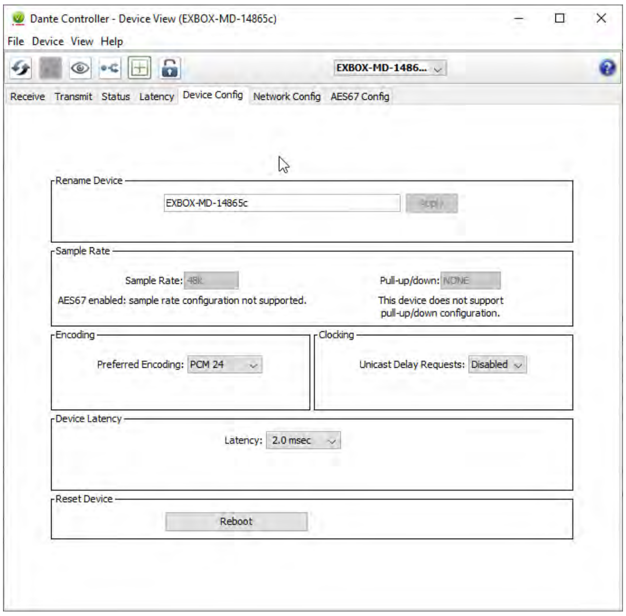
3. Select the **AES67 Config** tab in the **Device View** and ensure the **AES67 Mode** is **Enabled**. From v4.2.x of Dante Controller it is not necessary to reboot a device to reconfigure this setting. It is also possible to reconfigure the **RTP Multicast Address Prefix** for situations where it is necessary to transmit or receive streams from a particular address range.



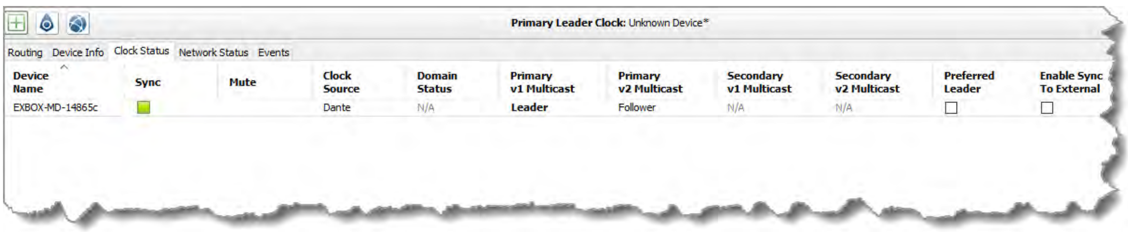
Important Note:

- If the **AES67 Config** tab is not displayed for a device it is probably either not able to stream in AES67 mode, or will require a firmware update to upgrade it for AES67 streaming if this is available.
- The multicast address prefix configured at the Dante device level (239.5.xxx.xxx in the preceding example), does not affect which AES67 flows (streams) from non-Dante devices are displayed in Dante Controller. However, when subscribing a Dante device to a non-Dante Source stream (Tx flow), the multicast address prefix set for the receiver must match that of the Source stream. In other words, a Tieline Source stream should use the multicast address prefix configured in the Dante device. Otherwise, if they do not match, the multicast Source subscription will appear to succeed, but audio will not actually flow/stream to the Dante device.

4. Select the **Device Config** tab in the **Device View**. Verify settings like the **Sample Rate** of **48k** and **PCM 24** bit are selected to match the Tieline Source and Destination streams. The **Latency** setting of **2.0 msec** is recommended for most networks, but **5 msec** can be configured for stability on Dante devices if required.

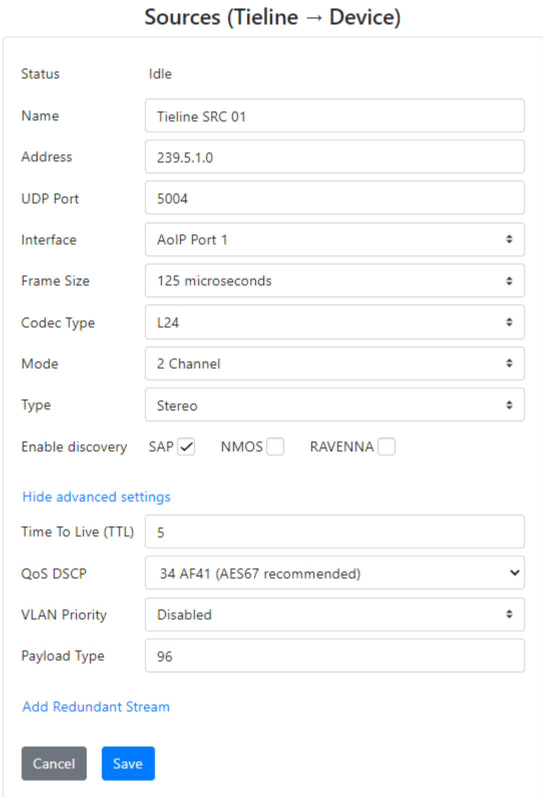


5. In Dante Controller confirm the Dante device is synced to the Primary Leader Clock on the AES67 network using the **Clock Status** tab.

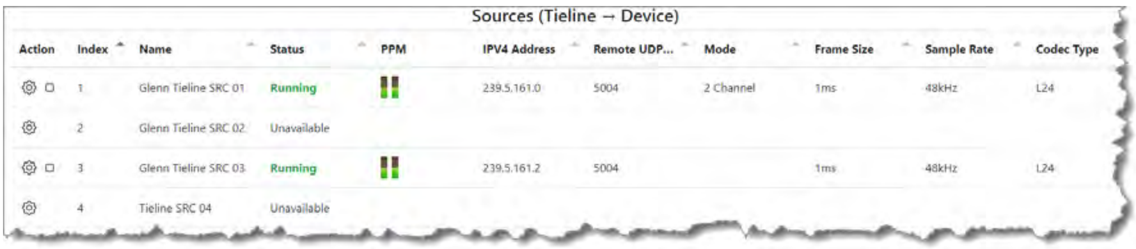


Create a Tieline Source Stream and Configure a Dante Flow to Receive Audio

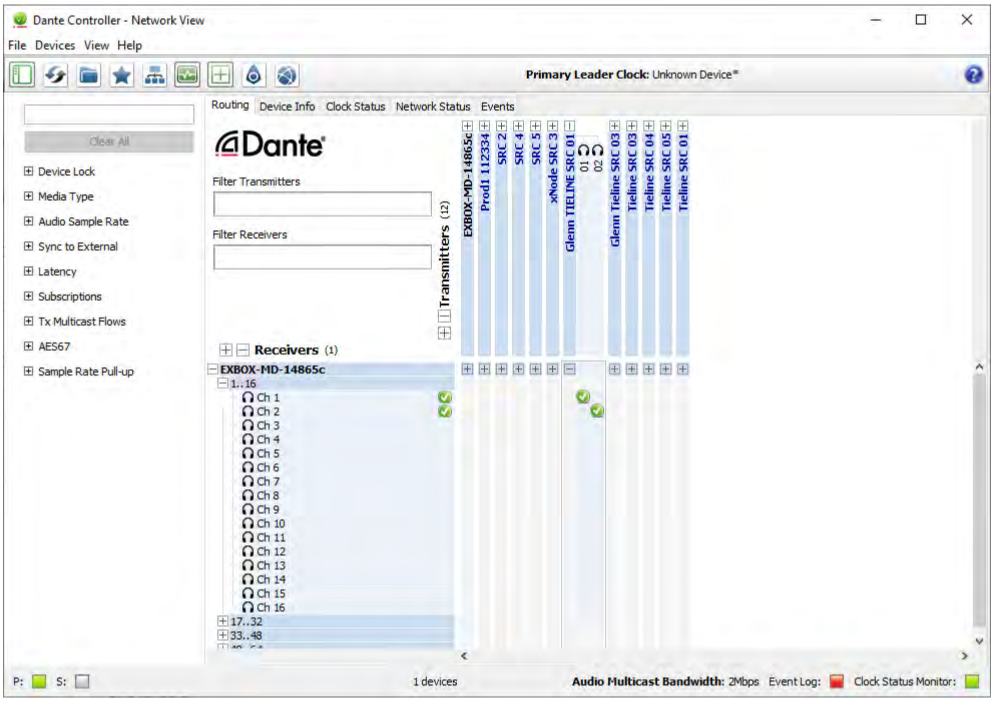
1. Configure a Source in the Tieline codec using similar settings to the following image. Notes: Ensure the Enable Discovery SAP check-box is selected. The multicast address prefix configured must match prefix in the Dante device receiving the stream. E.g. 239.5.xxx.xxx



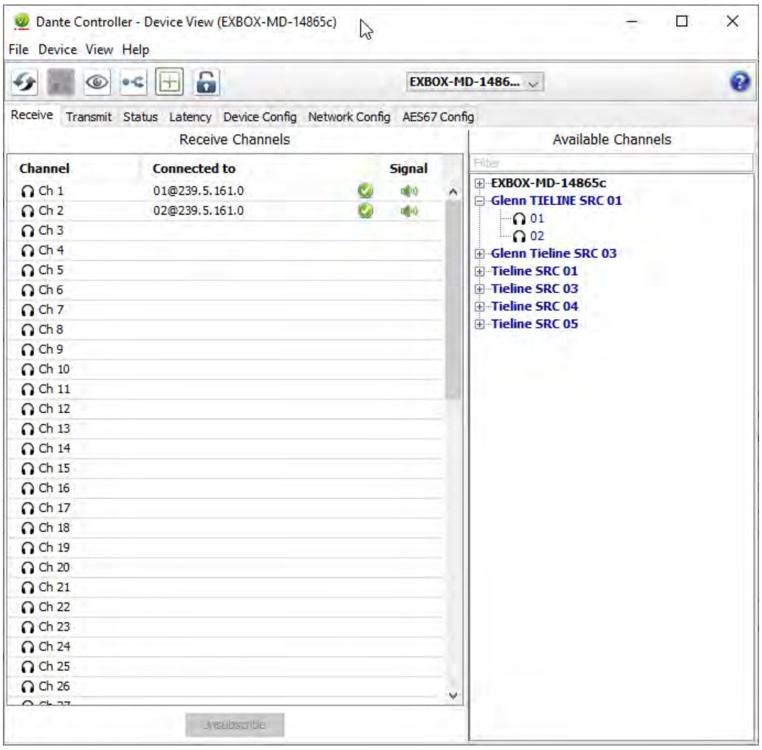
2. Verify audio is metering on the Source PPMs and click the **Start symbol** ▷ to start streaming. **Running** appears in green when streams are successfully streaming.



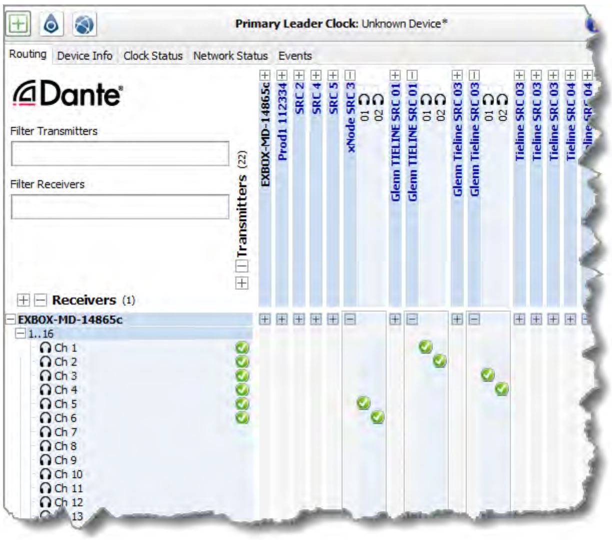
3. Launch Dante Controller and the Tieline codec Source streams should be populated at the top. In the example below, **Glenn TIELINE SRC 01** has been successfully routed via the crosspoints to the **EXBOX Receiver** (Destination) Channels 1 and 2.



4. In Dante Controller select **Devices > Device View** and then select the Dante device and view the **Receive** tab. The green ticks and green speaker icons on each channel confirm Tieline Source audio is being received as a Dante 'flow'.

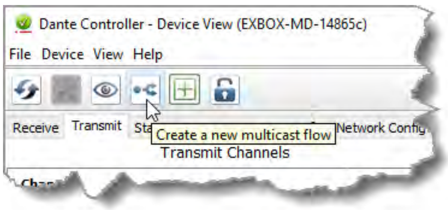


5. Other Tieline and non-Tieline Source streams can also be routed as required.

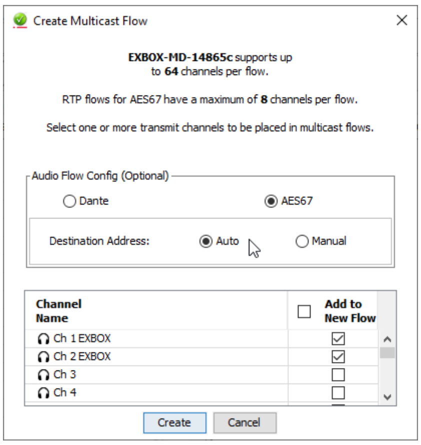


Create a Dante Tx Flow and a Tieline Destination Stream to Receive Dante AES67 Audio

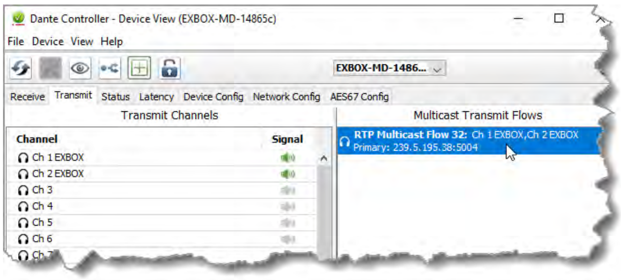
1. In the Dante Device Manager menu select **Device > Create Multicast Flow**, or click the **Create a new multicast flow** symbol.



2. Select **AES67** as the **Audio Flow Config** and leave the **Destination Address** setting as **Auto**. Next, select the channels to add to the new Dante flow (which will be the destination Tieline stream). Two channels are configured in the following example, however Tieline supports connecting up to 8 channels in AES67 streams between Dante devices. Finally, click **Create**.

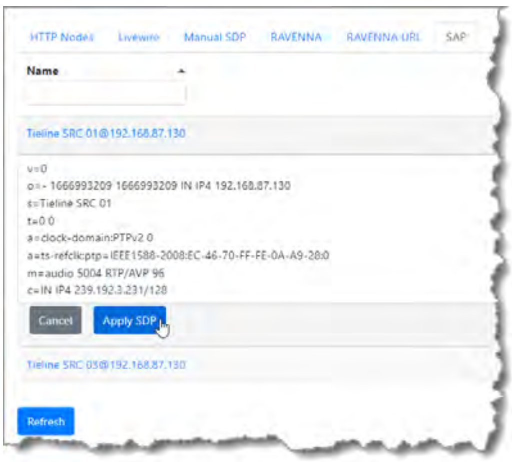


3. The new multicast flow is presented in the Transmit tab of the Device Manager and the green speaker symbols confirm audio is present.

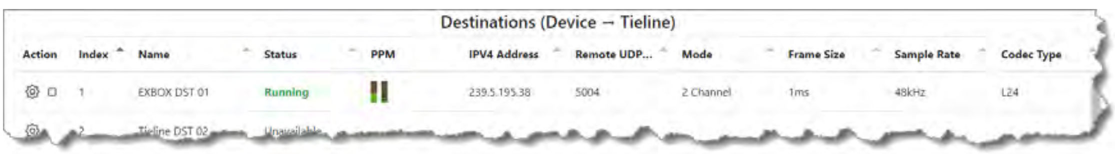


4. Next, configure a Tieline Destination stream with the multicast IP address displayed in the Device Manager and the same port, frame size, sample rate and codec settings.

5. Alternatively, Dante devices support SAP so it is possible to use the **SAP tab** in the **Destinations panel** to discover streams. Select the **SAP tab** and then select a stream to view stream details. Select **Apply SDP** to configure the Destination stream automatically using the displayed SAP data.



6. Save the settings and click the **Start symbol** ▷ to stream audio from the Dante device. **Running** displays when streaming and PPMs should meter audio.



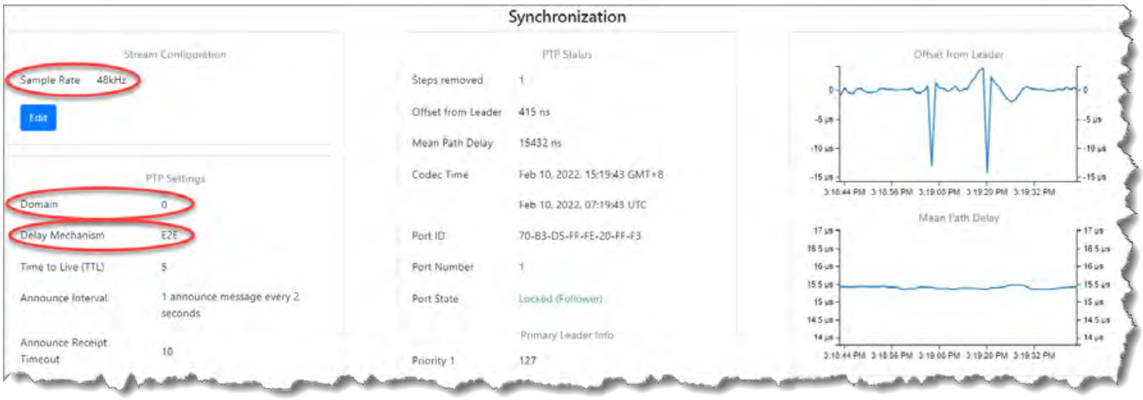
1.12 LAWO AES67 Streaming

Connectivity tests with RAVENNA AES67 interoperability were conducted using a LAWO Power Core using firmware v.6.4.1565.

Getting Started with LAWO over AoIP

In testing the following settings were successfully configured in the **Synchronization panel** in the Tieline AoIP Web-GUI. Note: Verify the LAWO settings are the same as they should match on the Tieline and LAWO devices.

1. Global Sample Rate is 48kHz.
2. Domain is set to 0.
3. Delay Mechanism is set to E2E.



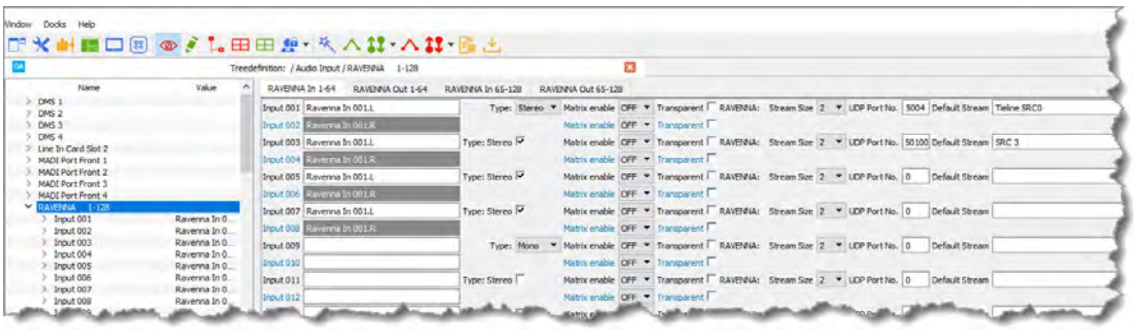
Check that PTP synchronization is running correctly on the device to which the Tieline codec will connect.

Important Notes:
- PPMs for AoIP streams are visible when viewing Sources and Destinations using the Tieline AoIP Web-GUI if a computer is connected to the AES67 LAN at the studio.
- The AoIP Web-GUI also displays all the panel settings available within the **AoIP** menu in the Toolbox Web-GUI, however it looks quite different.

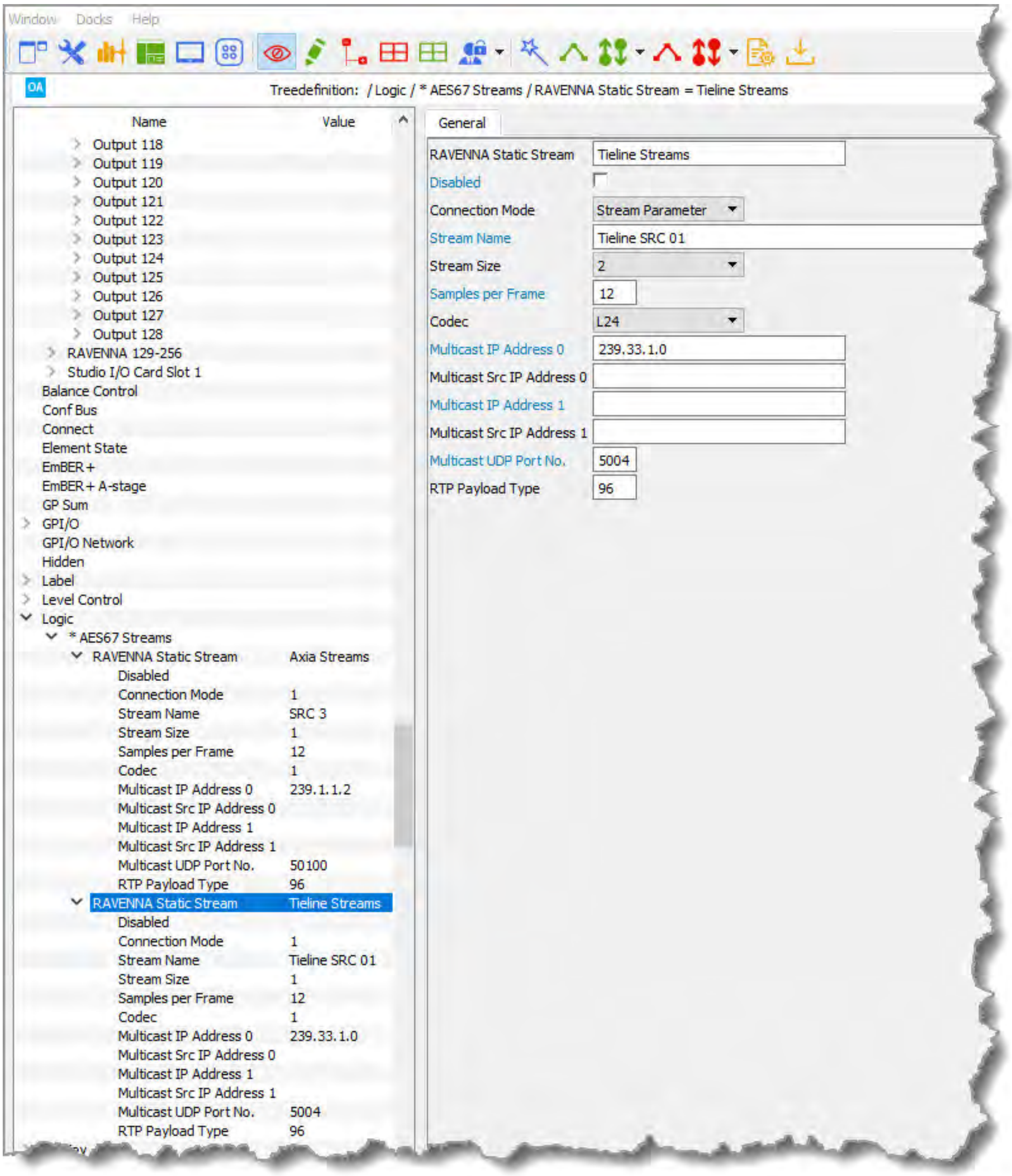
Set up a LAWO Power Core to Receive a Tieline 'Source' Stream

Configuration of the LAWO Power Core over an AES67 LAN in testing required the setup of installed LAWO ON-AIR Designer software as follows:

1. Configure the RAVENNA input to receive the Tieline codec source. This includes the **Stream Size**, **UDP Port No.** and the **Default Stream** name.



2. Configure the AoIP AES67 stream, including the **Stream Name**, **Stream Size**, **Samples Per Frame**, **Codec** and **Multicast IP Address**, **Multicast UDP Port No.** and the **RTP Payload Type** is **96**.

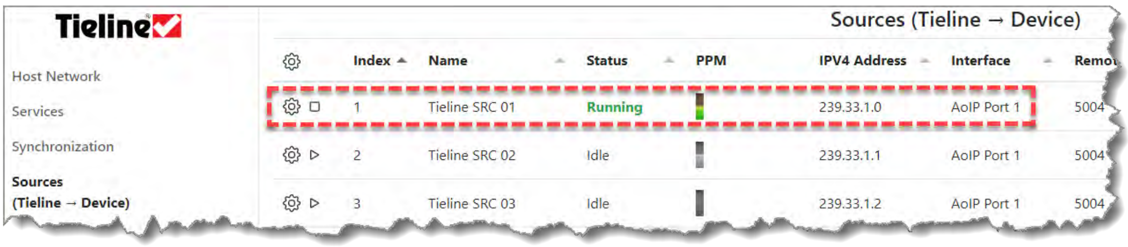


Note: When configuring Tieline codecs please note that **Frame Size** is displayed in microseconds or milliseconds. Some products like LAWO use **Samples Per Frame** instead. It is possible to calculate the relevant setting using the following table:

Packet Time	Packet samples (48 kHz)	Packet Samples (96kHz)	Packet Samples (44.1 kHz)	Notes
125 microseconds	6	12	6	Compatible with class A AVB transport

250 microseconds	12	24	12	High-performance, low-latency operation. Interoperable with class A and compatible with class B AVB transport
333 microseconds	16	32	16	Efficient low-latency operation
1 millisecond	48	96	48	Required common packet time for all devices adhering to AES67 standard
4 milliseconds	192	NA	192	For applications desiring interoperability with EBU Tech 3326 or transport over wider areas or on networks with limited QoS capability

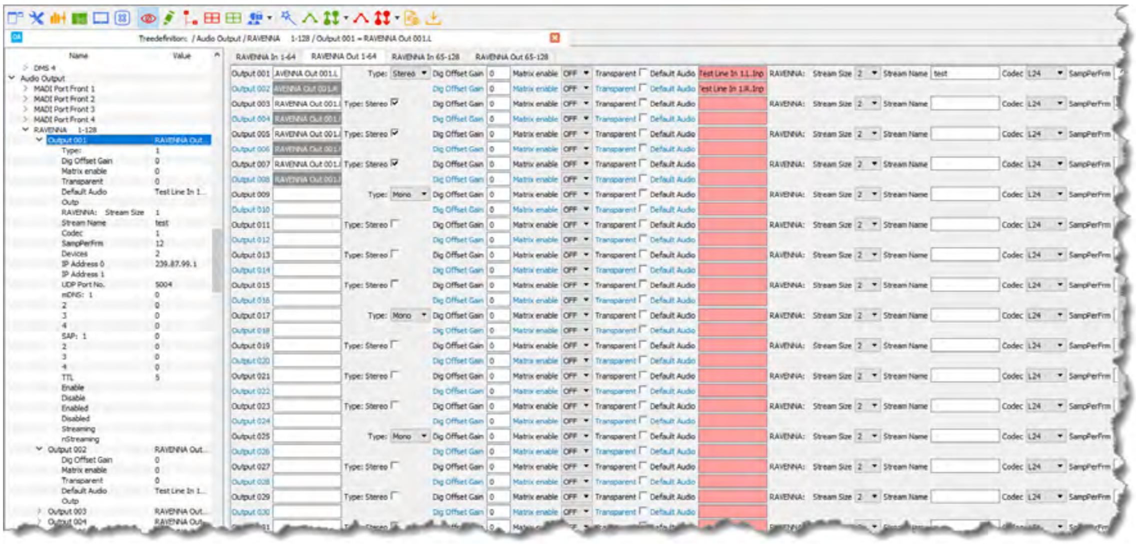
3. Using ON-AIR Designer save the configuration and select **Transfer > Config to Unit** and wait for the configuration to be uploaded.
4. Navigate to the **RAVENNA** screen and select the **Outputs (Device to Core) tab**, then click **Connect** to start activate the AES67 source stream from the Tieline codec.
5. In the **Source** screen within the Tieline AoIP Web-GUI click the **Configuration symbol** and then select **Start** to commence streaming. The configured source should display outgoing audio to the LAWO device on the PPMs.



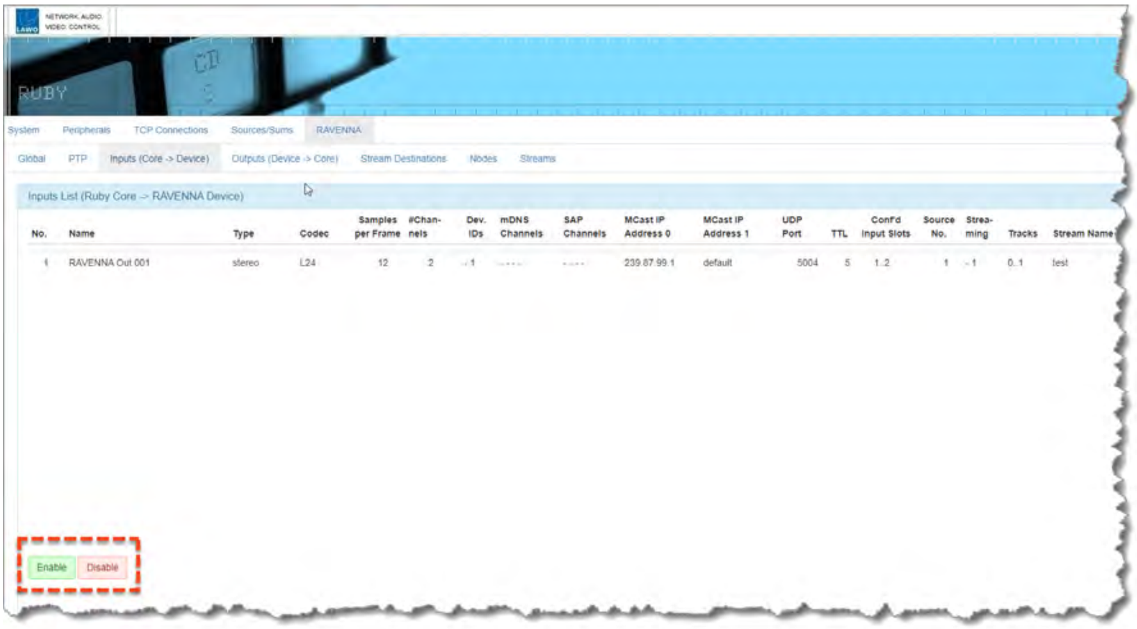
Set up a LAWO Power Core to Send a 'Destination' Stream to a Tieline Codec

A device streaming AES67 audio to a Tieline codec will normally be configured with a destination multicast IP address. This IP address is then entered into the Destination screen on the Tieline AoIP Web-GUI to receive the stream. Configuration of the LAWO Power Core over an AES67 LAN in testing required the setup of installed LAWO ON-AIR Designer software as follows:

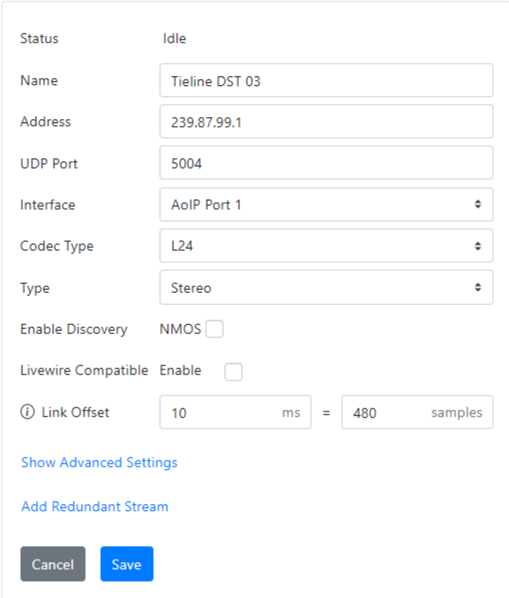
1. Configure the RAVENNA output to send the Tieline codec destination audio. This includes the **Stream Size**, **UDP Port No.**, **Stream Name**, **Codec** and **IP Address** as displayed.



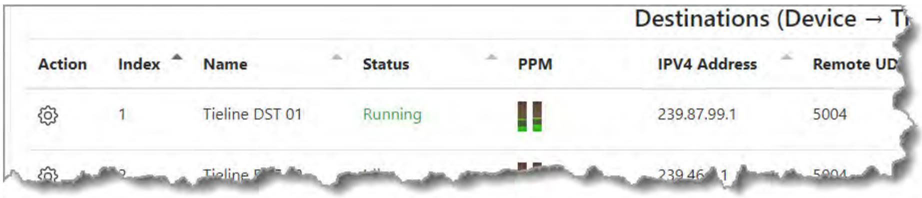
2. Using ON-AIR Designer save the configuration and select **Transfer > Config to Unit** and wait for the configuration to be uploaded.
3. In the Ruby Power Core Web-GUI click **Enable** to start the Destination stream.



4. In the **Destination** screen within the Tieline AoIP Web-GUI click the **Configuration symbol** and then select **Edit** to enter the Destination multicast node IP address, then click **Save**.



5. Next click the **Configuration symbol** and then **Start** to commence receiving the AES67 audio stream from the LAWO device. The configured Destination should display incoming audio from the LAWO device on the PPMs. Verify input audio is displayed on the codec to confirm audio is being received ok.



1.13 Troubleshooting AES67 and WheatNet

Some things to check if AES67 streaming is not working include:

1. Check the PTP clock synchronization in the Tieline codec and the device to which it is streaming and confirm they are locked and synchronized. This is verified in the **AES67 Synchronization panel** in the Tieline codec's HTML5 Toolbox web-GUI, or the **Synchronization panel** in the AoIP Web-GUI if this is accessible.
2. During testing, in 8 channel streaming mode the xNode required an RTP **Payload Type** of 99 and testing was successful with a 1ms frame size. The **Payload Type** setting is available in the **Advanced Settings** menu in the **Sources panel**.
3. WheatNet-IP only stream audio using the AoIP 1 port.

Destination Troubleshooting (Device to Tieline)

1. Ensure the global digital input setting is configured for AoIP and not AES3 via the **Audio Options panel > Digital Input Type [Audio Over IP]**. To configure this via the codec front panel press the **SETTINGS** > **Audio Options** > **Dig. Input Type > AoIP**.
2. If audio is still not being received from another device double-check that the input itself is configured for digital audio and not analog (only for inputs 1-8) in the **Inputs panel**. This can also be checked via **SETTINGS** > **Audio Inputs [Select input] > Type > Digital**.
3. Confirm all settings match the device streaming audio to the Tieline codec. E.g. Audio Sampling Rate, Audio Sampling Depth, Packet Time, Number of Channels streamed.
4. Confirm the stream **Status** is **Running**.

Important Note: PPMs on the AoIP **Destinations panel** are inactive if the **Digital Input Type** is not set to **AES67** and the corresponding input **Type** is not set as **Digital**.

Source Troubleshooting (Tieline to Device)

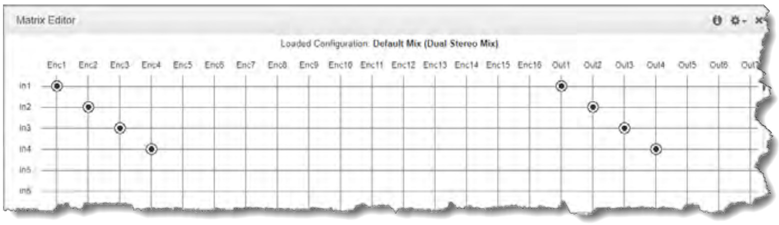
1. Ensure audio is metering on the outputs streaming to the other device. Note: When streaming in AES67 mode, the **Matrix Editor** hard coded output channel assignments used to feed AoIP audio are dependent on the stream configuration mode selected in the Tieline codec **Sources panel**. See Configuring AES67 Sources and Destinations for more details.
2. Confirm all settings match the device to which audio is being streamed by the Tieline codec. E.g. Audio Sampling Rate, Audio Sampling Depth, Packet Time, Number of Channels streamed.
3. Confirm the stream **Status** is **Running**.

Discovery Troubleshooting

1. No SAP announcements are sent if PTP is not locked.
2. Advertisement and discovery for RAVENNA streams is only available using the **AoIP 1** port. Streaming is available on both **AoIP 1** and **AoIP 2** ports.

WheatNet-IP Navigator Troubleshooting

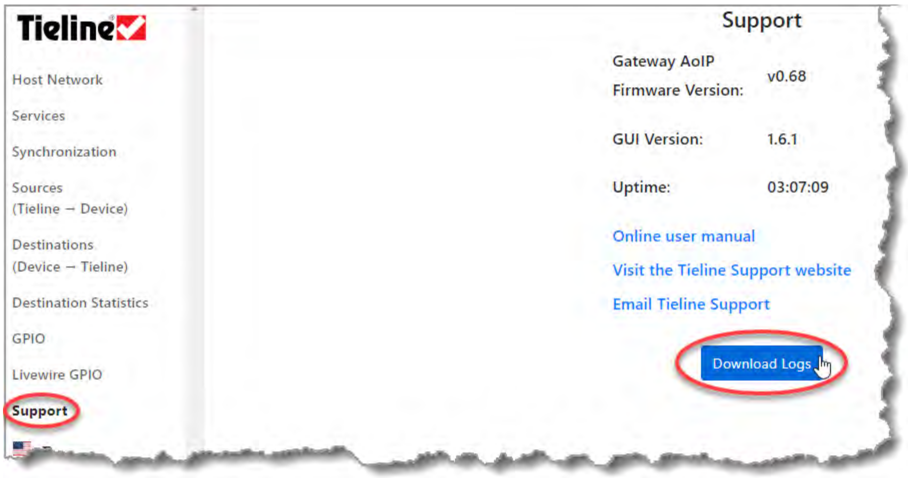
1. Ensure audio is routed in the codec **Matrix Editor** from inputs to outputs if the codec is feeding audio into the WheatNet-IP network from codec input sources. I.e. route audio from inputs to outputs to create a WheatNet Source from the codec.



Download Logs for Support

Download AoIP network related logs to troubleshoot any support issues:

1. In the AoIP Web-GUI select **Support** and then click the **Download logs** button.



2. Save the file and send it to Tieline.

Important Note: The logs generated using the AoIP Web-GUI are not the same as the logs generated using the HTML5 Toolbox Web-GUI.

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