



RAVENNA

AES67 *now!*

Configure Merging and AES67 devices

AES67 interoperability with Non-Merging interface

When the Audio Engineering Society ratified the X-192 groups findings, the outcome was the AES67 proposal for interoperability of devices transmitting real-time audio streams on networks.

AES67 is completely compatible with RAVENNA networks and manufacturers announcing that they are delivering AES67-ready devices means that they are in fact delivering RAVENNA compatible ones.

The RAVENNA protocol is able to automatically provide the source description (SDP) to the target, but in AES67 with non-Merging interfaces that are not compatible with ANEMAN, it needs to be done manually, as it is not standardized, all manufacturers tend to do it in their own ways.

For details on ANEMAN devices compatibility, see [ANEMAN user manual](#) – Plugins section.

Notes:

- *Merging Products are supporting AES67 in multicast*
- *MassCore and VAD (CoreAudio) supports AES67 and the RAVENNA ASIO driver supports AES67 as of v11.0 for Horus and Hapi and v12.0 for Anubis.*

1. Configuring devices in AES67 mode:

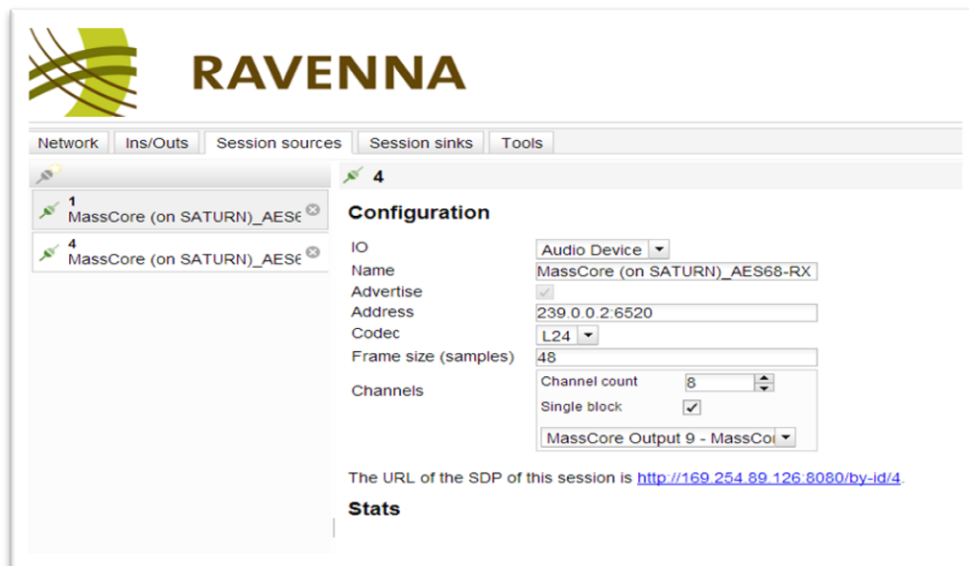
- Horus/Hapi/Anubis:
 - Launch MT Discovery to open the Horus/Hapi/Anubis Web Access or go on device screen
 - Go to Setup>System page for Horus and Hapi users. Anubis users go to Settings>General
 - Set "Latency" to AES67 (48 smpl)
- MassCore:
 - From VS3 Control Panel, set "Latency" to AES67 (4ms)
- RAVENNA Virtual Audio device (Mac CoreAudio):
 - From RAVENNA CoreAudio Panel, set "Latency" to 48 smpl (AES67)
- RAVENNA ASIO Driver (PC):
 - From RAVENNA ASIO Panel, set "Latency" to 48 smpl (AES67)

2. Creating sources (on the transmitter):

- From the MT Discovery go to the advanced WebAccess pages of Horus/Hapi/Anubis (open advanced) MassCore/CoreAudio/ASIO (open) e.g. http://Horus_80001/advanced/index.html
- Select the "Session sources" tab
- Create a Source by clicking on the little connection cable icon (top left), fill your parameters (description on how to create a Source manually is available in the Merging Advanced settings Guide)
- Download the SDP file describing this source by clicking on the blue link at bottom right of the page. This will be important in order to configure the non-Merging AES67 device

Notes:

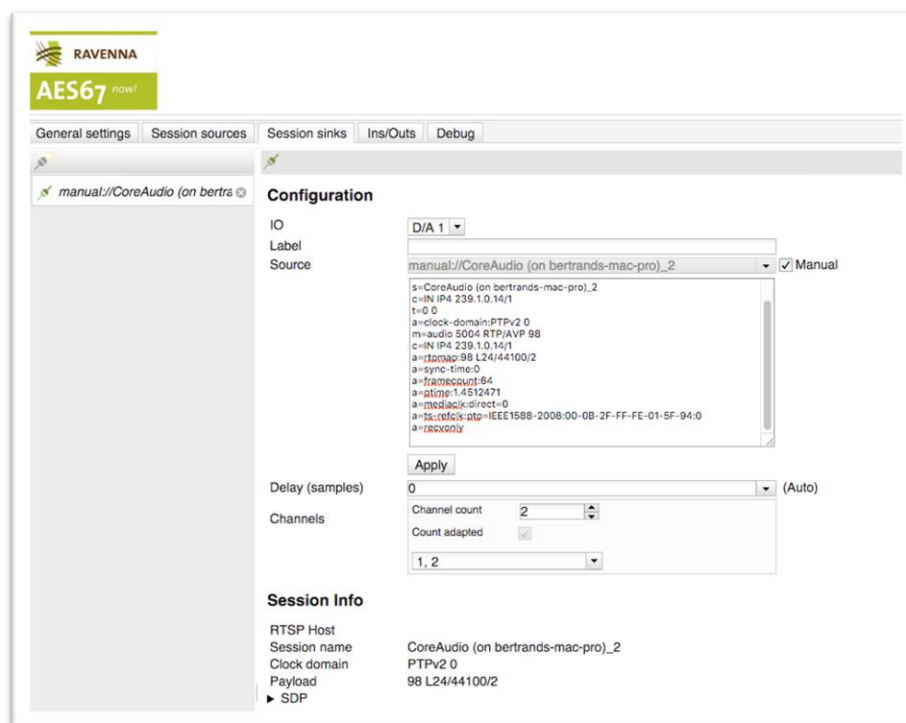
- *If the target needs a specific port it is possible to specify the port for the address accordingly (e.g. 239.1.40.12:6520).*
- *The port end number must be even (0, 2, 4, 6, 8)*



c)

Creating sink (receiver):

- From the MT Discovery Browser go to the advanced WebAccess pages of Horus/Hapi/Anubis (open advanced) MassCore/CoreAudio/ASIO (open) e.g. http://Horus_80001/advanced/index.html
- Select Session sinks tab
- Create a Sink by clicking on the little connection cable icon (top left)
- Configure the sink:
 - Select the source:
 - RAVENNA: select the source from the combo box
 - For AES67: you have to click on Manual and provide the SDP description



SDP file Specification Information:

```
v=0                                ; protocol version
o=- 1 0 IN IP4 169.254.89.126    ; device(session creator/owner) IP
s=Test1                           ; session name
t=0                               ; time the session is active
a=clock-domain:PTPv2 0
m=audio 5678 RTP/AVP 98          ; port:5678 payload type:98
c=IN IP4 239.1.40.64             ; stream dest IP
a=rtpmap:98 L24/48000/8          ; payload type:98 , codec:L24, SR:48000, 8 channels
; RAVENNA
a=sync-time:0
a=framecount:1-48                ; samples per frame [1..48]
; AES67
a=ptime 1.0                      ; packet time
a=mediaclk:direct=0
a=ts-refclk:ptp=IEEE1588-2008:00-0B-2F-FF-FE-01-38-83:0
a=recvonly
a=maxptime 1.0                   ; max packet time
```

Note: when a=rtcp:xxxx is present and xxxx != 0 then MassCore and RAVENNA CoreAudio driver send RR and SR.

Configure MERGING and third party DANTE devices in AES67 mode

We will demonstrate how to configure a Merging Hapi and third party Dante device for both inputs and outputs. The same procedure can be applied to any Merging RAVENNA/AES67 device.

1. You can transmit audio from the Dante device to Merging Hapi
2. You can transmit audio from Merging Hapi to the Dante device

Notes

- Dante device must be AES67 capable, please consult Audinate website for details. Dante Virtual Sound card is not AES67 capable.
- Dante devices, in AES67 mode, only support 48 kHz.
- Maximum number of streams for Dante in AES67 mode is 32.
- Merging products AES67 compatibility details can be found [on this page](#).

Warning: Users with Cisco SG350 must refer to the end of the Setup section at page 7.

Setup

We connect all following network devices on the same network switch:

- One Merging Hapi - firmware V3 or above required.
- One Dante device - AES67 capable.
- A computer with Dante Controller and ANEMAN installed.
- Prerequisite: Some devices require that you remotely enable their AES67 mode, either from Dante Controller or from their remote access application

Audio transmission from a Dante supporting AES67 to a Merging device Hapi, Horus or Anubis.

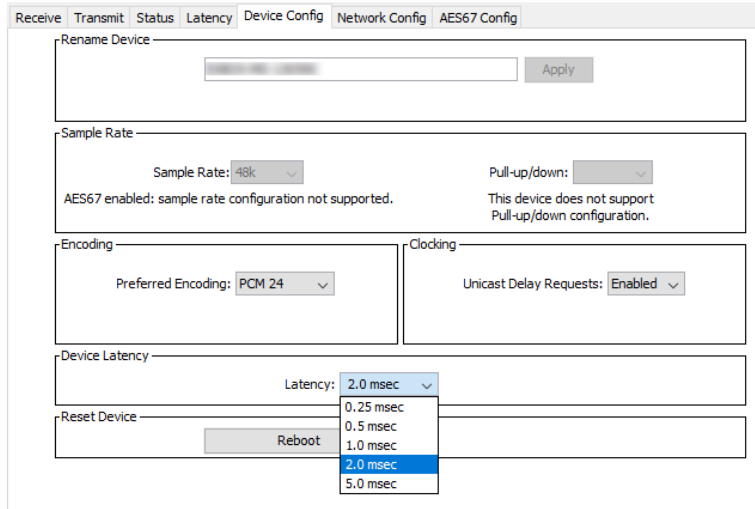
1. Create AES67 Flow

- a. Go in the *Device > Device View* page in Dante Controller
- b. Select your Dante device.
- c. Go in the AES67 Config tab
- d. Set the AES67 mode to Enabled. **A reboot is required if it was disabled.**

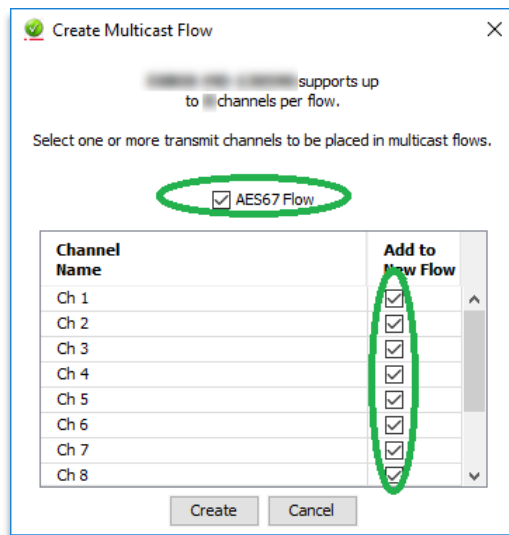
Make sure the *Multicast Address Prefix* is set to 239.69.xxx.xxx

The screenshot shows the 'AES67 Config' tab in the Dante Controller interface. It has three main sections: 'AES67 Mode', 'Tx Multicast Address Prefix', and 'Reset Device'. In the 'AES67 Mode' section, both 'Current' and 'New' are set to 'Enabled', with a green box highlighting the 'New' dropdown. The 'Tx Multicast Address Prefix' section shows the 'Current Prefix' as '239.69.xxx.xxx', with a green box around the '239.69' part, and a 'New Address Prefix' input field with a 'Set' button. The 'Reset Device' section contains a 'Reboot' button.

- e. Go now in the *Device Config* tab.
Set the *Latency* to 2 or 5msec (AES67 recommended value is 3 msec)

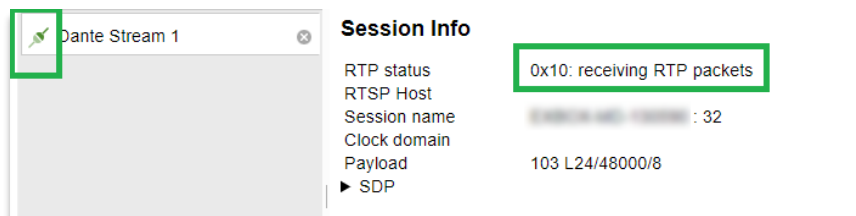
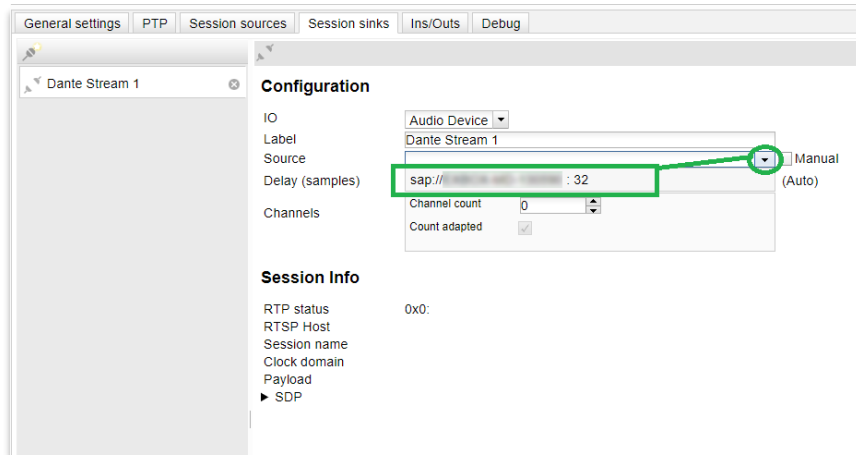


- f. In the *Device* drop down menu, select *Create Multicast Flow*.
- g. Tick the *AES67 Flow* box, and select the channels you need to transmit. Click on *Create* once done.



2. Connect Merging Hapi (Horus or Anubis) to Dante AES67 flow

- a. Open Merging device advanced page. Either via ANEMAN / MTDiscovery -> *Open Advanced* or directly using device name. For the device **hapi_90007** the url is http://hapi_90007.local/advanced/index.html
- b. On the *Session sinks* tab, create a session sink (receiver) using the icon at the upper left corner
- c. Select the desired IO module.
- d. Click on the arrow next to the "Source" field, you will get a list of the currently available SAP sources.
- e. Select your Dante stream.
The stream will now connect, the connection icon in the left column will turn green.
Depending on the device version, a *RTP Status* will be displayed, it will change to *receiving RTP packets*.



Cisco SG350 Users

For users wanting to connect AES67-Dante devices to their Cisco SG350 switches, please be aware that an additional Multicast Group configuration must be done for the ports where these devices are to be connected.

Dante AES67 users have to add IP Multicast Groups :

Once the configuration file has been applied, and the switch has been rebooted.

- Connect to the Cisco Administration page (default address with Merging configuration file is 169.254.1.254)
- Make sure the Display Mode of the Cisco Administration page is set to Advanced
- Browse to *Multicast > IP Multicast Group Address* and click on *Add*.
- Enter VLAN ID 1 (assuming you have 1 VLAN) and enter 224.0.0.230 as IP Multicast Group Address.
Click on *Apply*.
- Now select the 224.0.0.230 Group and click on *Details*.
Set the ports connected to Dante devices to Static and click on *Apply*.
- Repeat the same operation for addresses 224.0.0.231, 224.0.0.232 and 224.0.0.233

(Example with Dante devices on port 9-10).

VLAN ID: 1
 IP Version: Version 4
 IP Multicast Group Address: 224.0.0.230
 Source IP Address: *

Filter: Interface Type equals to Port Go

Interface	GE1	GE2	GE3	GE4	GE5	GE6	GE7	GE8	GE9	GE10
Static	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>
Dynamic	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>
Forbidden	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>	<input type="radio"/>
None	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input checked="" type="radio"/>	<input type="radio"/>	<input type="radio"/>

Apply Close

Audio transmission from Merging Hapi (Horus or Anubis) to Dante device

1. Create Merging Hapi (Horus or Anubis) AES67 flow

- Open Merging device advanced page.
- On the Session sources tab, create a session source (transmitter) using the icon at the upper left corner.
- Select the desired IO module.
- Make sure the multicast address ("Address" field) of Merging device is using the Dante multicast prefix. Default is **239.69.xxx.xx**.
- Set the required number of channels (Channel count)
(Codec must be L24 and Frame size 48)

3

Configuration

IO: AES 1

Name: Hapi_90003_1

Advertise: ☒

Address: 239.69.194.33 ☒ user defined

TTL: 15

Payload Type: 98

Codec: L24

Frame size (samples): 48

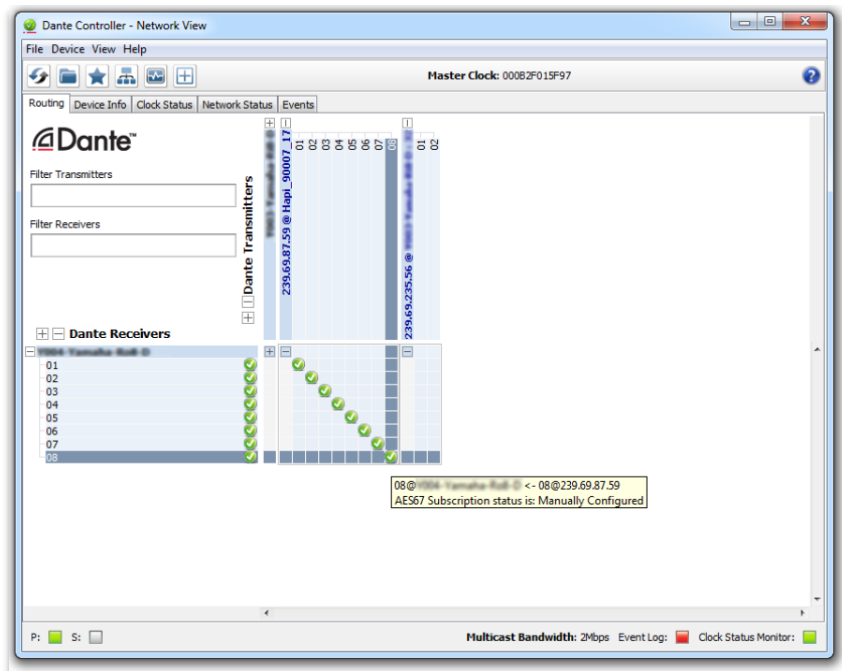
DSCP: 34 (AF41)

Channels: Channel count: 8
1 - 8

The URL of the SDP of this session is <http://169.254.194.33:8080/by-id/3>.

2. Connect Dante to AES67 flow

- Open Dante Controller
- Merging Device should appear in the Dante transmitter list
- Connect inputs and outputs as desired



The screenshot shows the Dante Controller - Device View window. The top menu bar includes File, Device, View, and Help. Below the menu is a toolbar with icons for Refresh, Add, Remove, and other functions. The main area displays a table of Receive Channels and Available Channels.

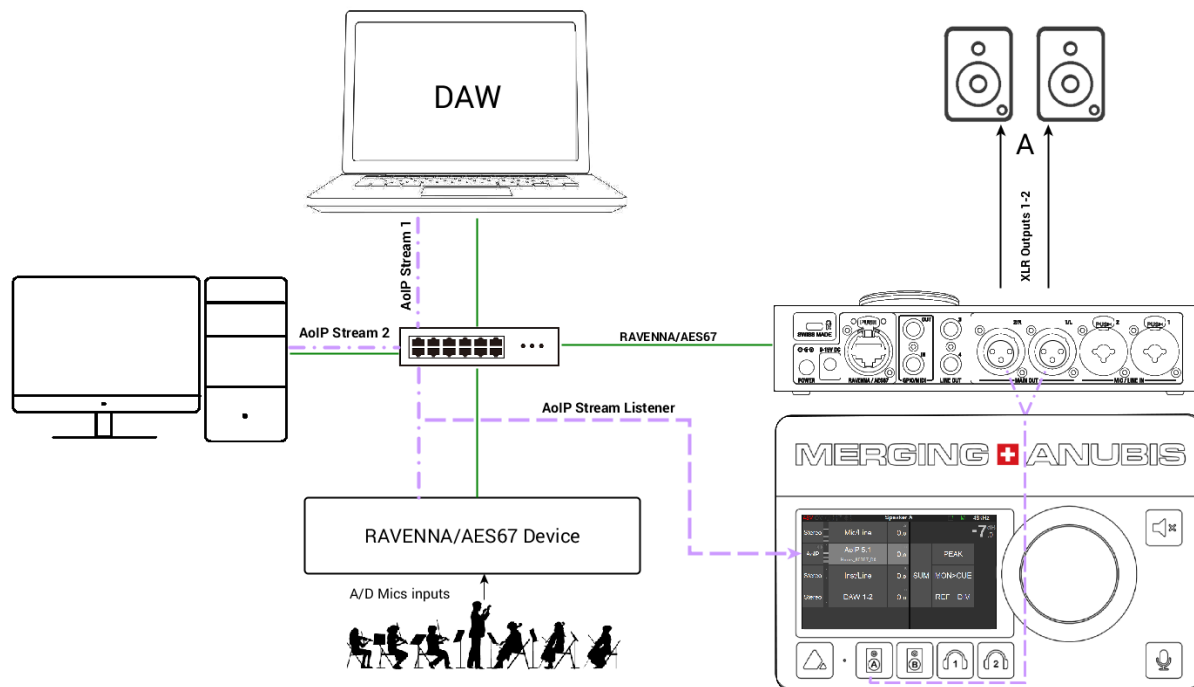
Channel	Signal	Connected to	Status
01	01@239.69.87.59	01@239.69.87.59	✓
02	02@239.69.87.59	02@239.69.87.59	✓
03	03@239.69.87.59	03@239.69.87.59	✓
04	04@239.69.87.59	04@239.69.87.59	✓
05	05@239.69.87.59	05@239.69.87.59	✓
06	06@239.69.87.59	06@239.69.87.59	✓
07	07@239.69.87.59	07@239.69.87.59	✓
08	08@239.69.87.59	08@239.69.87.59	✓

Available Channels:

- Yamaha 08 @
- 239.69.87.59 @ Hapi_90007_17
- 01
- 02
- 03
- 04
- 05
- 06
- 07
- 08
- 239.69.235.56 @ Yamaha 08 @ : 32

MERGING+ANUBIS AoIP STREAM LISTENER SETUP

Exclusive to Anubis



SetUp:

Anubis can monitor RAVENNA/AES67 streams that are available over a same network. Whether they are from Merging's RAVENNA/AES67 Devices or not.

Such Setup could be applicable in Broadcast where an Anubis station could monitor streams from various production rooms or in an educational classroom where a teacher could monitor each student station from his local Anubis and as well in Live where a sound engineer could monitor each musician Anubis stream.

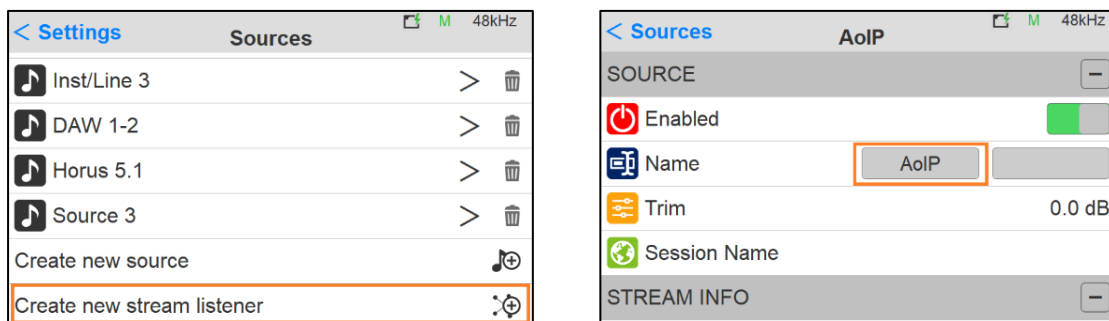
Prerequisites:

RAVENNA/AES67 streams have to be available over the same network to which Anubis is connected to.

Note: AoIP stream listener supports a maximum of 8 channels per stream.

Procedure:

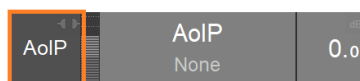
1. Anubis should be setup for monitoring, refer to the basic monitoring use case.
2. Have streams available over the network. Those streams would in most situations be connections between other RAVENNA/AES67 devices or could be between a Horus and a DAW, or between two RAVENNA/AES67 nodes. Nothing prevents a user to monitor streams between a single Anubis and a DAW as the AoIP stream Listener can monitor any source available from within the Listener source dialog.



3. Once you do have streams available create a New Source Listener from the Anubis Settings>Source. By factory one AoIP source listener should already be available.

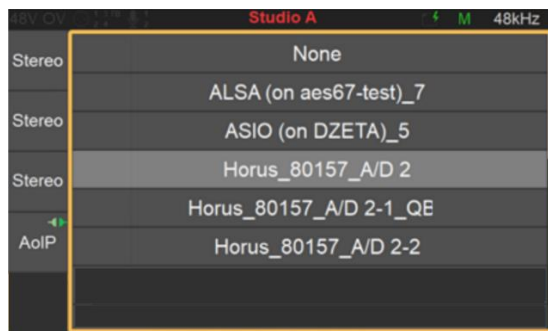
Note: a Source Listener has very few parameters to configure, only name and trim are applicable

4. From the Anubis Source Page open the AoIP Listener dialog by tapping the AoIP entry on the left



Tap to open the Listener Dialog

5. The Listener Dialog should open and be populated with the available RAVENNA/AES67 on your network.



6. Tap the Stream you wish to Monitor.
7. To close the Listener Dialog, tap the AoIP Entry on the left, or Tap anywhere outside the dialog

Note: AoIP Stream Listener does not provide selectable speaker control from the Monitor Page.

You are ready to monitor any stream RAVENNA/AES67 over your network from your Anubis locally.

TROUBLESHOOTING

1. **Merging Transmitter doesn't appear in the Dante transmitter list**
 - a. Make sure the devices are correctly connected to the same switch
 - b. Make sure Merging device multicast address has same prefix as the one in the Dante controller
2. **No Audio**
 - a. Make sure the switch is Gigabit
 - b. Make sure devices are operating using the same sampling rate. Merging device Sample rate is available under the general settings of the advanced page or directly in the device setup menu.
 - c. Make sure you have ticked the AES67 Flow when you created the Dante Multicast Stream.
 - d. Latency performance issue :
Go in *Dante Controller* > *Device view* > *Latency* tab.
The latency value displayed should be green all time. If that is not the case, go in Dante Device Config tab and increase the Late.
 - e. PTP Clocks :
Make sure you only have 1 PTP Master and the other devices are slaves and synced on it.
See *Dante Controller* > *Clock Status* and *ANEMAN PTP* tab.

Refer to the Merging Knowledge database for more troubleshooting solutions.

<https://merging.atlassian.net/wiki/spaces/PUBLICDOC/overview>