

Audio over IP

AES67, Dante, RAVENNA, Livewire+

Quick Guide

Version 1.310 (23 February 2021)

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Audio over IP

Overview:

AES67

SAP

- AES67 is an open standard for Audio over IP interoperability, developed by the Audio Engineering Society (AES).
- AES67 sets a minimum standard for audio streaming and synchronisation in Audio over IP networks such as Dante, RAVENNA and Livewire+.
- AES67 is available as a software option for AVT MAGIC telephone hybrids, audio codecs and DAB encoders and features:
 - One AES67 stream in send direction with up to 8 audio channels (depending on the device).
 - Two AES67 streams in receive direction with up to 8 audio channels each.
- For the MAGIC THipPro, a Dante module is available, which also supports AES67 with up to 32 audio channels.
 - The Dante module cannot be combined with the AES67 software option.
 - Together with Audinate's Dante Domain Manager, redundant AES67 streams are also supported via SMPTE.

AES67

- SAP (Session Announcement Protocol) uses Multicast to distribute the description of each stream in the network.
- Each SAP message contains the stream description using the SDP (Session Description Protocol) format.
- Each sender of AES67 streams transmits SAP messages periodically to the multicast IP address: 239.255.255.255.
- AES67 receivers collect the stream information coming via SAP and present it to the user to choose from.
- In large networks it may be desirable to minimize periodic broadcast traffic. Therefore it is possible to exchange stream information between devices via SDP files instead of SAP.
- SDP files are text files which store the stream information in the SDP format.
- On AVT MAGIC devices you can switch off SAP on the AES67 configuration page.
- The TX stream information can be exported to an SDP file.
- To subscribe to a stream, SDP files can be imported.

SAP

Audio over IP

AES67 Software Option: General Configuration

- The AES67 functionality is activated on the corresponding configuration page.
- LAN INTERFACE: The network interface of the device for AES67.
- PTP DOMAIN: The network for clock synchronization.
- QUALITY OF SERVICE (DSCP): Classification of data to prioritize network traffic.
 - PTP: DSCP classification of the clock synchronization protocol.
 - RTP: DSCP classification of audio streams.
 - SET DEFAULT QOS VALUES: Default values for PTP and RTP.

Configuration

Local MAGIC THipPro

AES67

☒ Activate AES67 streaming

LAN Interface: LAN 2 : 192.168.96.28

PTP Domain: 0 0.127

Quality of Service (DSCP):

PTP: 56 (CS 7) (0.63) DiffServ: 224dec

RTP: 46 (EF) (0.63) DiffServ: 184dec

Transmission:

Channels: 8

SAP Stream Name: AVT THipPro

RTP UDP Port: 5300

Audio Mode: L24

Sampling Rate: 48 kHz

Address Mode: Auto IP Address: 239 0 96 28

Reception:

Stream 1: Mixer Studio A; 8 channels

Stream 2: Mixer Studio B; 8 channels

Buttons: Set Default QoS values, Export SDP File, Update Rx Streams, Import SDP File

Client ID: 5 Studio: 1

Buttons: OK, Abbrechen, Apply Now

AES67 Configuration (1)

- TRANSMISSION: Definition of the AES67 audio stream in send direction. The stream can contain up to 8 channels and is made accessible in the network by SAP (Session Announcement Protocol).

- CHANNELS: 1-8 mono audio channels.
- SAP STREAM NAME: Identifier of the AES67 stream in the network.
- RTP UDP PORT: Port of the audio stream.
- AUDIO MODE: Algorithm for audio coding:
 - L16: Linear PCM 16 bit
 - L24: Linear PCM 24 bit
- SAMPLING RATE: Sampling rate of the audio signal:
 - 32 kHz
 - 48 kHz

The screenshot shows the 'Configuration' window for AVT THipPro, specifically the 'AES67' tab. The sidebar on the left lists various configuration categories, with 'AES67' highlighted under 'System Settings'. The main configuration area includes the following settings:

- Activate AES67 streaming:** Checked.
- LAN Interface:** LAN 2: 192.168.96.28
- PTP Domain:** 0 (0.127)
- Quality of Service (DSCP):**
 - PTP:** 56 (CS 7) (0.63) DiffServ: 224dec
 - RTP:** 46 (EF) (0.63) DiffServ: 184dec
- Transmission:**
 - Channels:** 8
 - SAP Stream Name:** AVT THipPro
 - RTP UDP Port:** 5300
 - Audio Mode:** L24
 - Sampling Rate:** 48 kHz
 - Address Mode:** Auto
 - IP Address:** 239.0.96.28
- Reception:**
 - Stream 1:** Mixer Studio A; 8 channels
 - Stream 2:** Mixer Studio B; 8 channels

Buttons for 'Set Default QoS values', 'Export SDP File', 'Update Rx Streams', and 'Import SDP File' are available. At the bottom, there are 'OK', 'Abbrechen', and 'Apply Now' buttons. The status bar at the bottom left shows 'Client ID: 5' and 'Studio: 1'.

AES67 Configuration (2)

- TRANSMISSION:

- ADDRESS MODE:

- MANUAL: Free entry of the multicast address of the audio stream.
 - AUTO: The multicast address is derived from the IP address of the device. Only a multicast subnet can be entered.

- IP ADDRESS: Multicast IP address of the audio stream.

- EXPORT SDP FILE: Not all manufacturers support SAP to automatically discover AES67 streams in the network. In this case, the definition can be exported to a file in SDP format. Recipients must be able to import this file.

Configuration

Local MAGIC THipPro AES67

☒ Activate AES67 streaming

LAN Interface: LAN 2: 192.168.96.28

PTP Domain: 0 0.127

Quality of Service (DSCP):

PTP: 56 (CS 7) (0.63) DiffServ: 224dec

RTP: 46 (EF) (0.63) DiffServ: 184dec

Set Default QoS values

Export SDP File

Transmission:

Channels: 8

SAP Stream Name: AVT THipPro

RTP UDP Port: 5300

Audio Mode: L24

Sampling Rate: 48 kHz

Address Mode: Auto IP Address: 239 0 96 28

Reception:

Stream 1: Mixer Studio A; 8 channels

Stream 2: Mixer Studio B; 8 channels

Update Rx Streams

Import SDP File

Import SDP File

Client ID: 5 Studio: 1

OK Abbrechen Apply Now

AES67 Configuration (3)

- **RECEPTION:** If AES67 is activated, the device searches the network for AES67 streams. It may take up to 5 minutes for all available streams to be listed. AVT devices can subscribe to one or two AES67 streams.
 - **UPDATE RX STREAMS:** Restarts the search for AES67 streams.
 - **STREAM 1 / 2:** All AES67 streams published in the network via SAP are offered for selection.
 - If the definition of an AES67 stream is available as a file in SDP format, it can be subscribed to using **IMPORT SDP FILE**.

The screenshot shows the 'Configuration' window for a 'MAGIC THipPro' device, specifically the 'AES67' tab. The left sidebar lists various settings categories, with 'AES67' highlighted under 'System Settings'. The main panel contains the following configuration options:

- ☒ **Activate AES67 streaming**
- LAN Interface:** LAN 2: 192.168.96.28
- PTP Domain:** 0 (0.127)
- Quality of Service (DSCP):**
 - PTP:** 56 (CS 7) (0.63) DiffServ: 224dec
 - RTP:** 46 (EF) (0.63) DiffServ: 184dec
- Transmission:**
 - Channels:** 8
 - SAP Stream Name:** AVT THipPro
 - RTP UDP Port:** 5300
 - Audio Mode:** L24
 - Sampling Rate:** 48 kHz
 - Address Mode:** Auto
 - IP Address:** 239 0 96 28
- Reception:**
 - Stream 1:** Mixer Studio A; 8 channels
 - Stream 2:** Mixer Studio B; 8 channels

Buttons for 'Set Default QoS values', 'Export SDP File', 'Update Rx Streams', and 'Import SDP File' are available. At the bottom, there are 'OK', 'Abbrechen', and 'Apply Now' buttons. The status bar shows 'Client ID: 5' and 'Studio: 1'.

AES67 Configuration (4)

Audio over IP

AES67 Software Option:

AES67 Audio Channel Assignment for

TH2_{plus} / TH6

THipPro

ACip3

- On the MODE & AUDIO LINE page, an AES67 channel can be assigned to any audio line.
 - AUDIO INTERFACE:** Select an AES67 channel.
 - AES67 RX:** Select a subscribed AES67 channel.
- All other audio interfaces of the *MAGIC TH2_{plus} / TH6* remain available.

Configuration

Mode & Audio Line

Mode

Operation Mode : **Two Faders**

☒ PRE TALK Conference ☐ Use only 1 VoIP Line ☐ Anonymous Calling ☐ Voice Disguise

Audio Line Assignment

Name	Audio Line	Audio Interface	AES67 Rx	ON AIR Access	Custom Label	Chat Name
PRE 1	PRE TALK Keypad 1	not used		<input checked="" type="checkbox"/>		
PRE 2	PRE TALK Keypad 2	not used		<input checked="" type="checkbox"/>		
PRE 3	PRE TALK Keypad TH2plus	not used		<input checked="" type="checkbox"/>		
PRE 4	PRE TALK PC 1	Handset 1		<input checked="" type="checkbox"/>		PC1
PRE 5	PRE TALK PC 2	IP Audio Stream 1		<input checked="" type="checkbox"/>		PC2
PRE 6	PRE TALK PC 3	AES67 Channel 1	Channel 1	<input checked="" type="checkbox"/>		PC3
AIR 1	ON AIR 1	AES67 Channel 2	Channel 2			
AIR 2	ON AIR 2	AES67 Channel 3	Channel 3			
HLD	HOLD/Monitoring	Audio 1/AES Left				

Caution: Invalid settings are red! Settings for this client have dark gray background colour.

☐ Default Audio Line on Drop for Keypad TH2plus

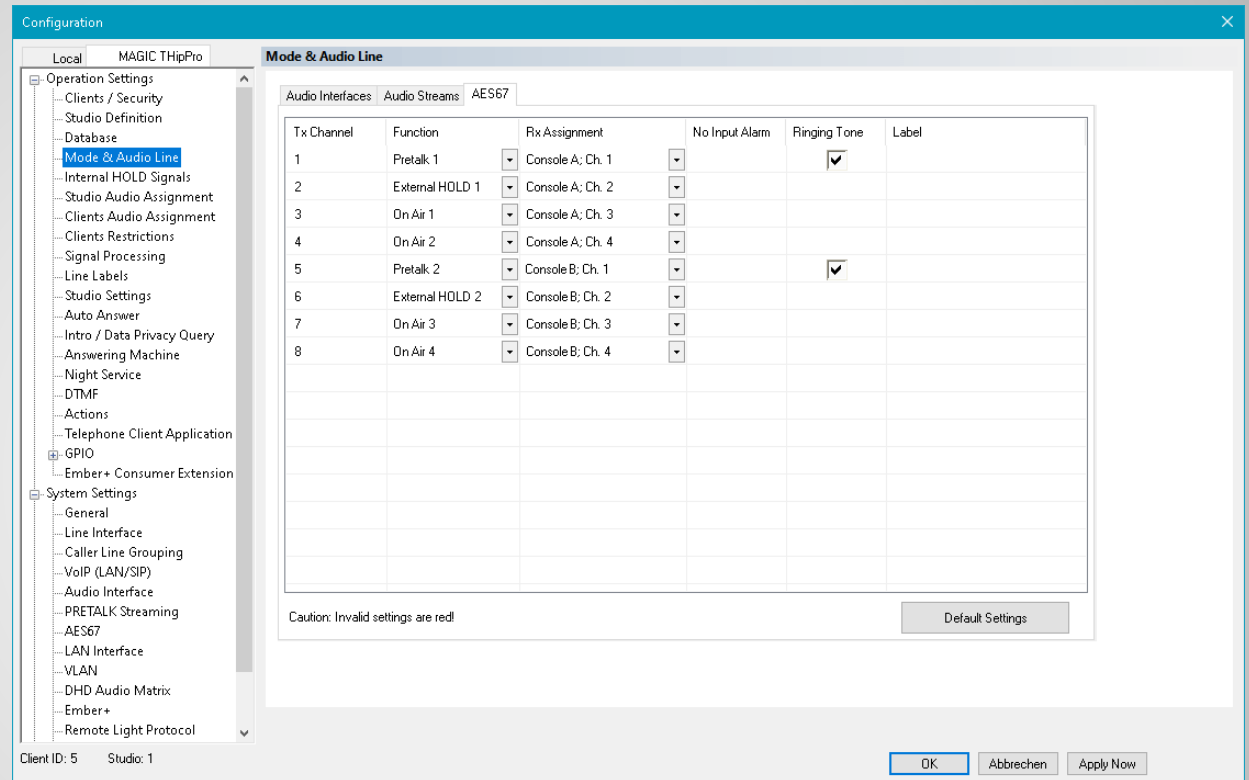
HOLD

Default Settings

OK Abbrechen Apply Now

AES67 Audio Assignment TH2_{plus} / TH6

- On the MODE & AUDIO LINE page, an audio line (function) is assigned to each of the AES67 channels.
 - TX CHANNEL:** Each line represents one channel of the outgoing AES67 stream.
 - FUNCTION:** Select an audio line.
 - RX ASSIGNMENT:** Assign channels of the subscribed AES67 streams to the audio lines.
- All other audio interfaces of the *MAGIC THipPro*, except for the Dante module, remain active.

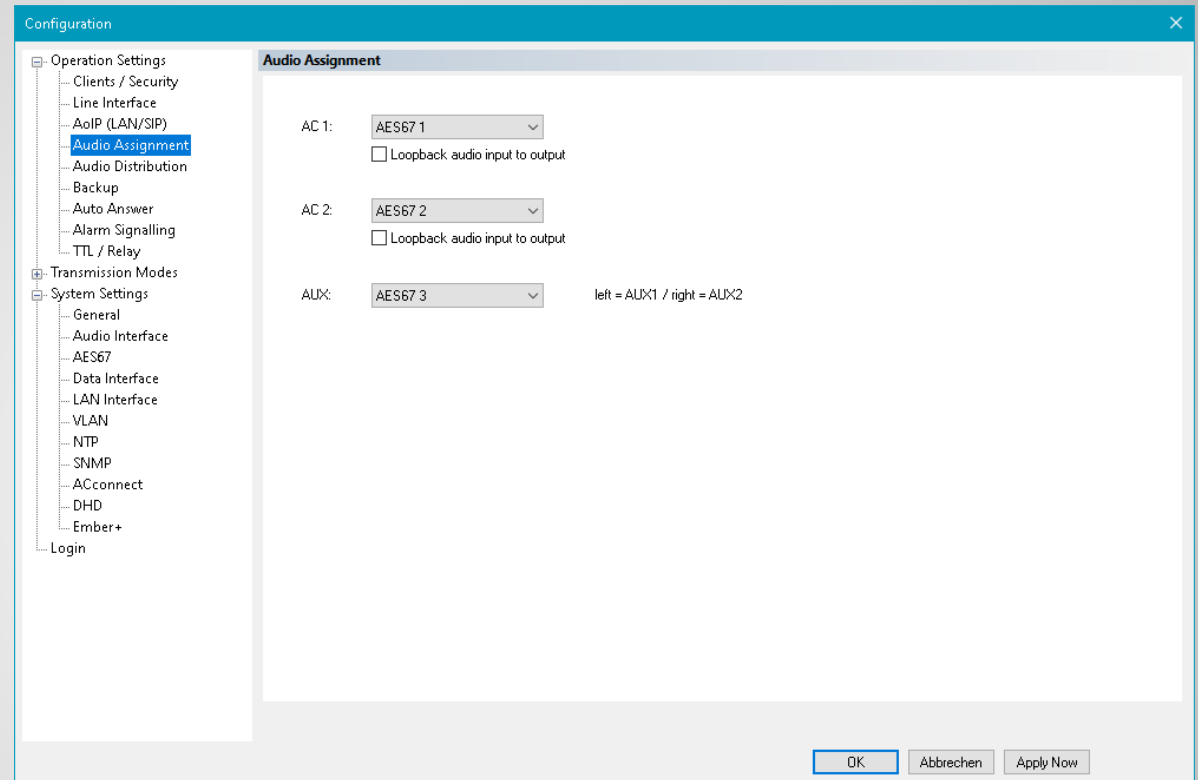


AES67 Audio Assignment *THipPro*

- First, on the AES67 page of the *MAGIC ACip3*, logical audio interfaces must be defined. These consist of AES67 channels of the transmit and receive audio streams.
 - AUDIO INTERFACE:** Each logical audio interface consists of two mono AES67 channels.
 - OUTPUT CHANNELS:** The AES67 channels in the send direction are automatically assigned continuously. The number of logical audio interfaces is limited by the number of channels defined under TRANSMISSION → CHANNELS.
 - INPUT CHANNELS:** The channels of the subscribed AES67 streams in RX direction can be freely assigned to the logical audio interfaces.

AES67 Audio Assignment ACip3 (1)

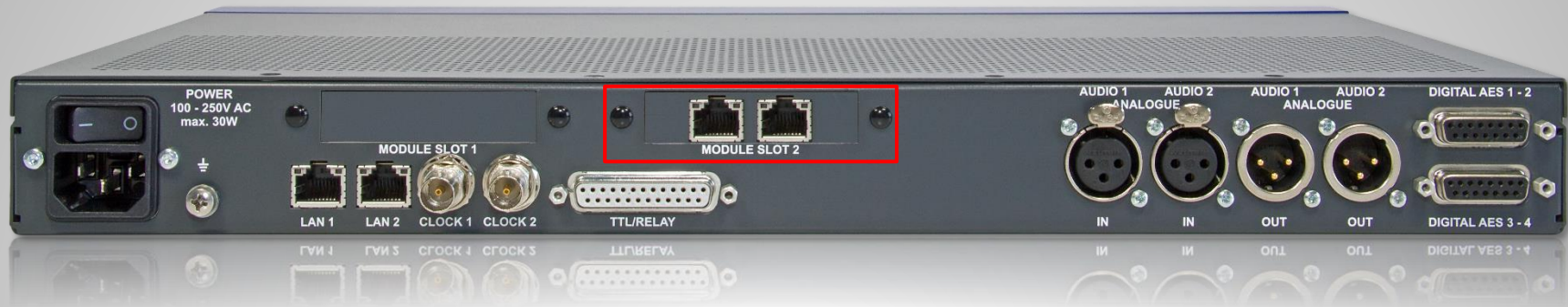
- On the AUDIO ASSIGNMENT page, one of the logical AES67 interfaces can be assigned to each channel.
 - AC1: Codec 1
 - AC2: Codec 2
 - AUX: Mono command channels 1 and 2. The first channel of the logical AES67 interface is assigned to AUX1.
- All other audio interfaces of the *MAGIC ACip3* remain available.



AES67 Audio Assignment ACip3 (2)

Audio over IP

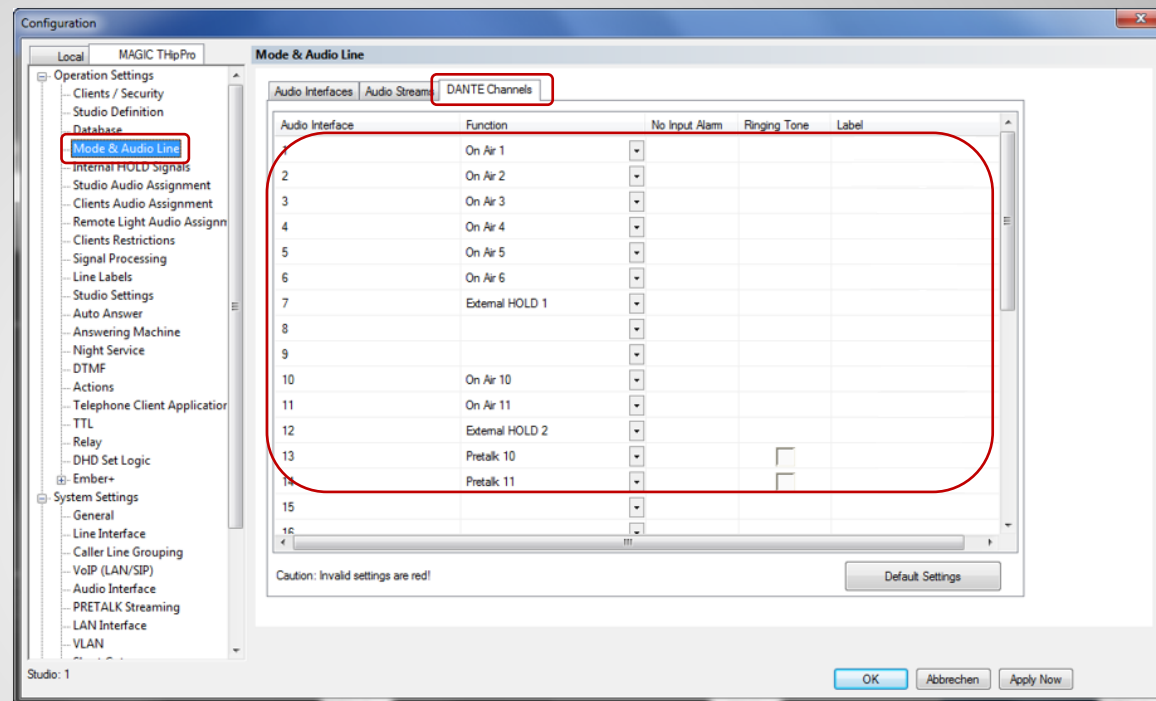
MAGIC Dante module



- Optional Dante module
- Available for *MAGIC THipPro* VoIP
- Two Ethernet interfaces
 - Option to select between redundancy and switch.
- 32 audio inputs
- 32 audio outputs

Dante/AES67 module for *MAGIC THipPro*

- The Configuration page *OPERATION SETTINGS* → *MODE & AUDIO LINES* shows the tab *Dante CHANNELS* if a Dante-module is equipped and *AES67* is deactivated under *SYSTEM SETTINGS – AES67*.
- Audio lines can be assigned to 32 Dante-channels.
- Standard audio interfaces remain operational without limitations.

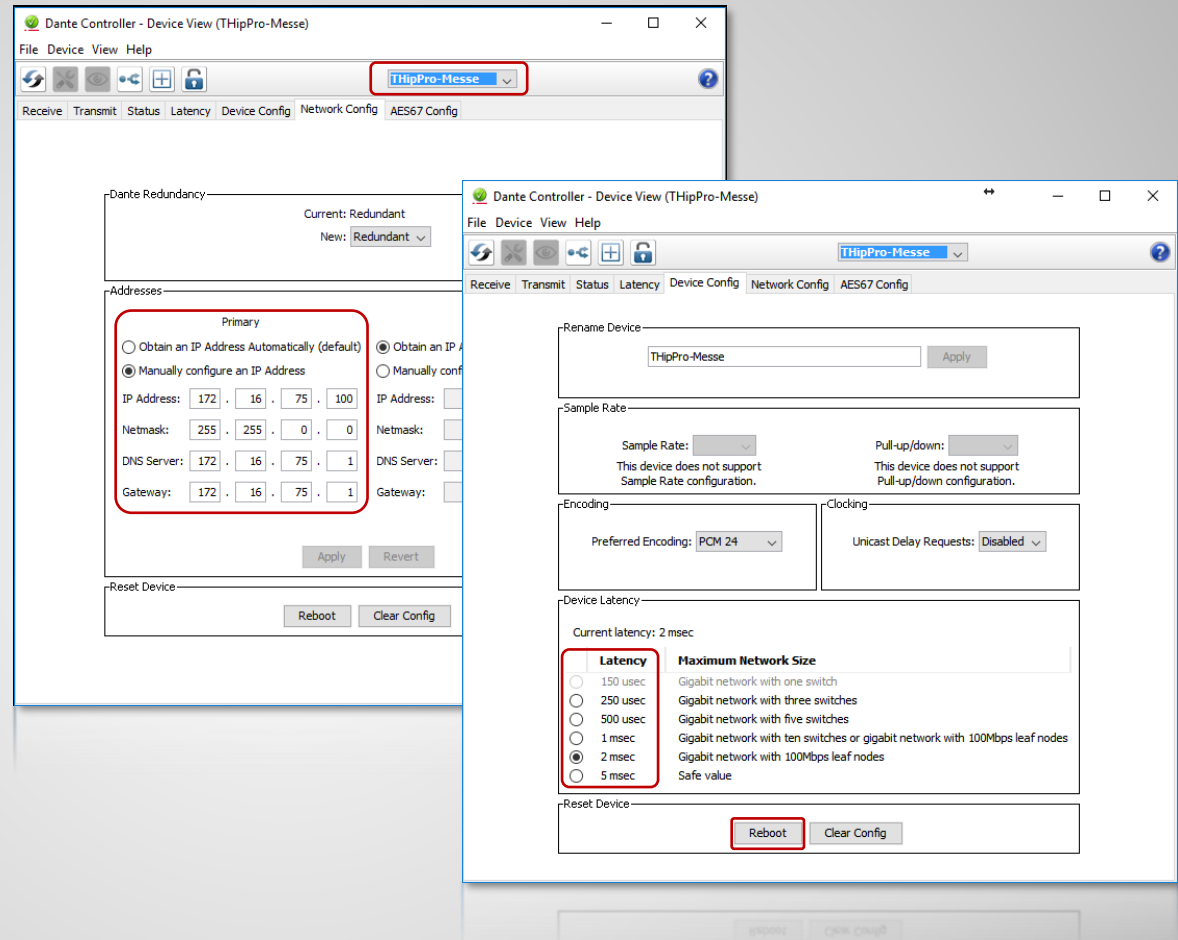


Configuration *MAGIC THipPro*

Audio over IP

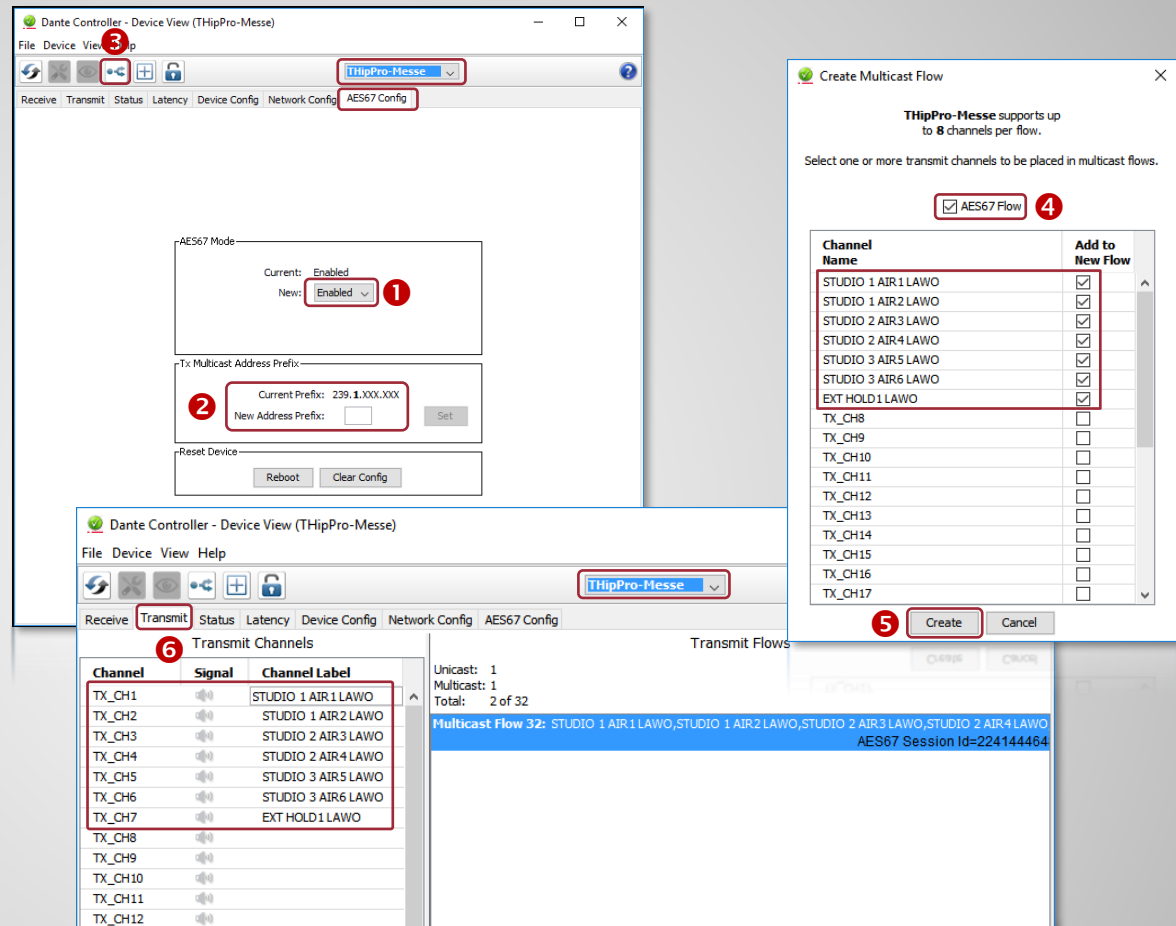
Configuration Dante Module

- Start the Dante Controller software and select DEVICE → DEVICE VIEW from the menu bar.
- Select the desired Dante device and open the tab NETWORK CONFIG.
- Set the IP-address as desired.
- Set the latency on the tab DEVICE CONFIG to the same value on all Dante devices in the network.
- Click REBOOT to apply the changes.
- Also switch the *MAGIC THipPro* off and on again if the front display signals an alarm.



Configuration Dante module (1)

- Select the tab AES67 CONFIG and enable AES67 Mode (❶) .
- Set the MULTICAST ADDRESS PREFIX to the **exact same value** for all units in the audio network. (Note: Avoid using "0" since it is not supported by every device). (❷) .
- Press button ❸ to create a MULTICAST FLOW.
- Activate the check box AES67 FLOW (❸).
- Select up to 8 channels in the column ADD TO NEW FLOW and press the button CREATE (❹).
- Create another flow if you need more than 8 channels.
- Channels added to the flow are displayed on tab TRANSMIT (❺).

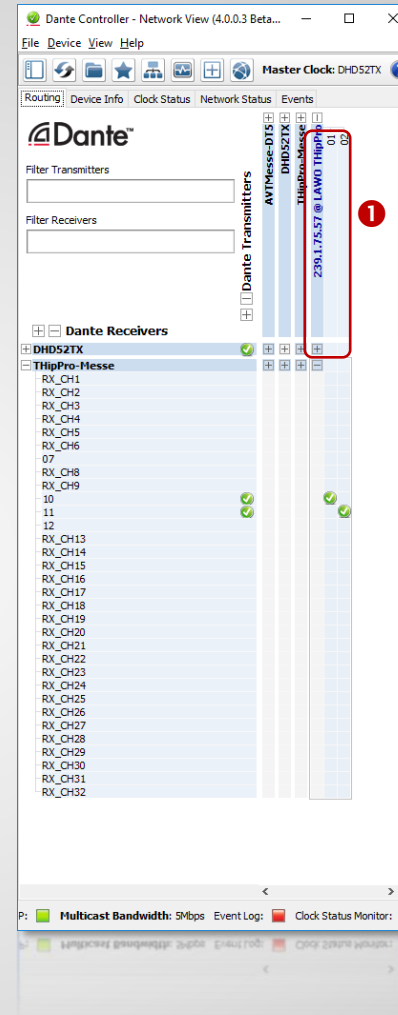


Configuration Dante module (2)

Audio over IP

Dante Audio Routing

- Start the Dante Controller software and select the tab ROUTING to see all devices in the network supporting the Dante-protocol.
- The Dante, AES67 und RAVENNA TX STREAMS show up under DANTE TRANSMITTERS (❶).
- A click with the left mouse button sets cross points in the matrix connecting the streams to DANTE RECEIVERS (like the *THipPro* Dante module).



Configuration Dante audio routing

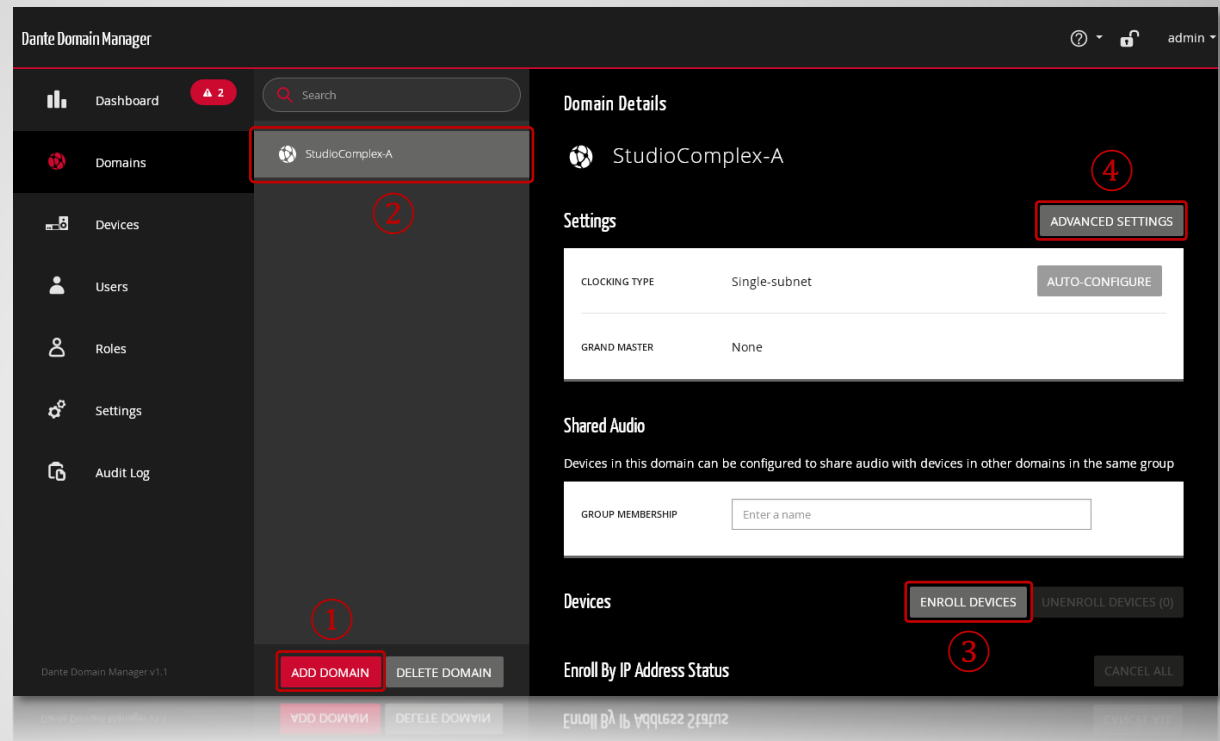
Audio over IP

Dante Domain Manager
SMPTE / Redundant AES67

- Dante Domain Manager brings user authentication, role-based access and network management to Dante Audio-over-IP networks.
- Dante Domain Manager also enables the SMPTE mode in Dante networks.
- Redundant AES67 audio streams are only available in SMPTE mode.
- The AVT MAGIC Dante module supports SMPTE from software version 1.0.4.
- Any Dante Domain Manager Edition (Silver, Gold, Platinum) supports SMPTE.
- Dante Domain Manager contains its own OS and comes as an ISO image which must be installed on bare metal hardware or a hypervisor (VMware, VirtualBox, ...).
- Instructions for installing and starting up the Dante Domain Manager can be found on the Audinate support website (registration required).

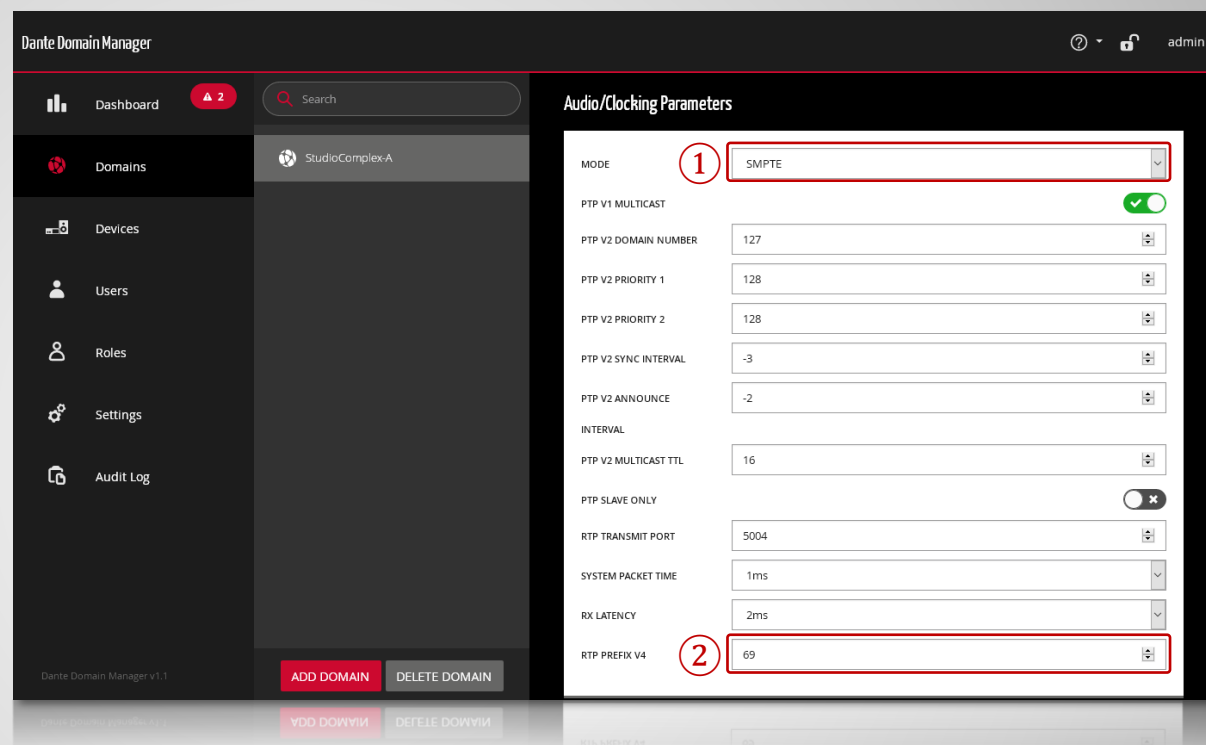
Overview

- Log in to the Dante Domain Manager via a web browser.
- Click ADD DOMAIN ① on the DOMAINS page and enter a name for the new domain.
- Select the new domain ② and click ENROLL DEVICES ③ to add the MAGIC Dante modules to the domain.
- Click ADVANCED SETTINGS ④ to configure the domain.



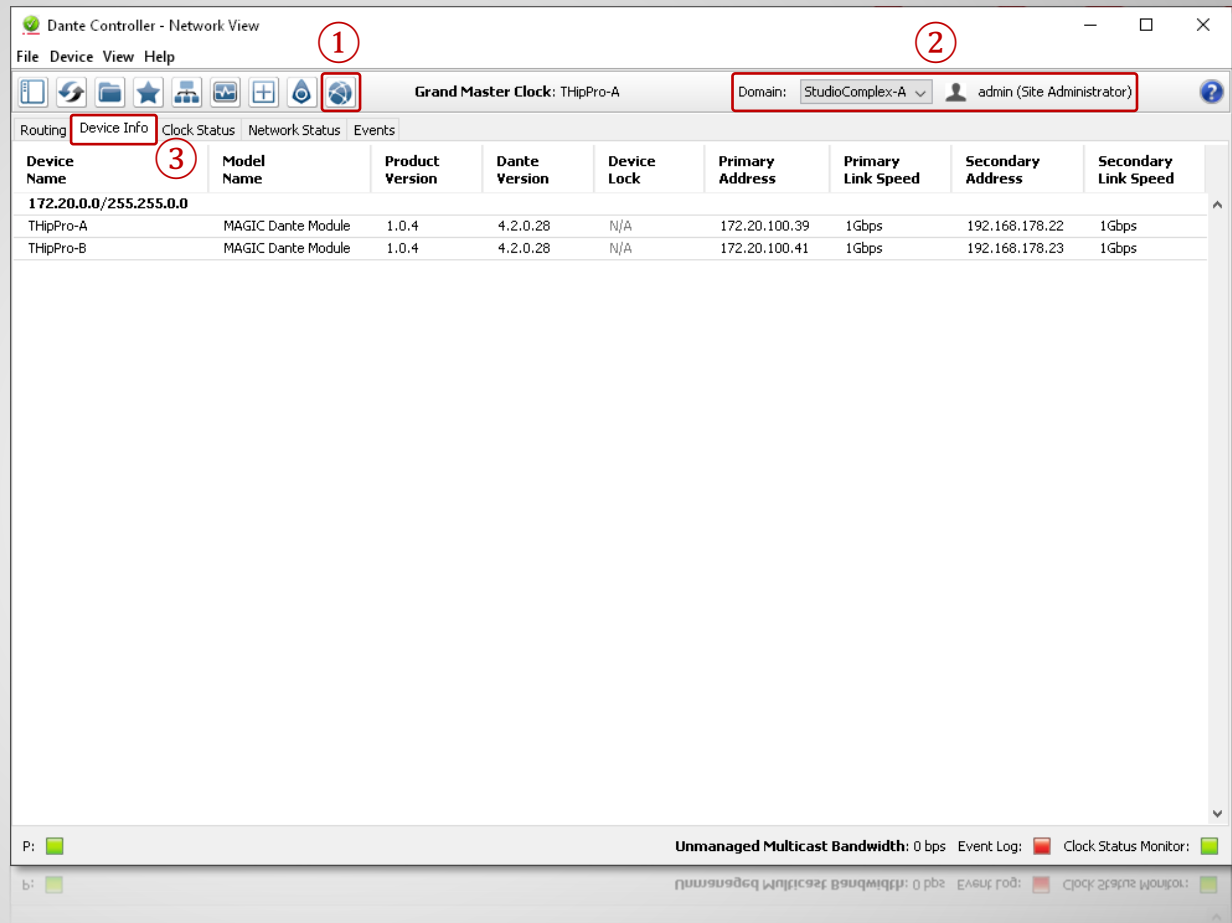
Create a Domain

- Enter ADVANCED SETTINGS and set the MODE to SMPTE ①.
- Set the RTP PREFIX V4 ② to the multicast address prefix used throughout the rest of the AES67 network. Otherwise audio will not come through. (e.g. if set to 69 the RTP streams will use multicast addresses in the 239.69.xxx.xxx range).



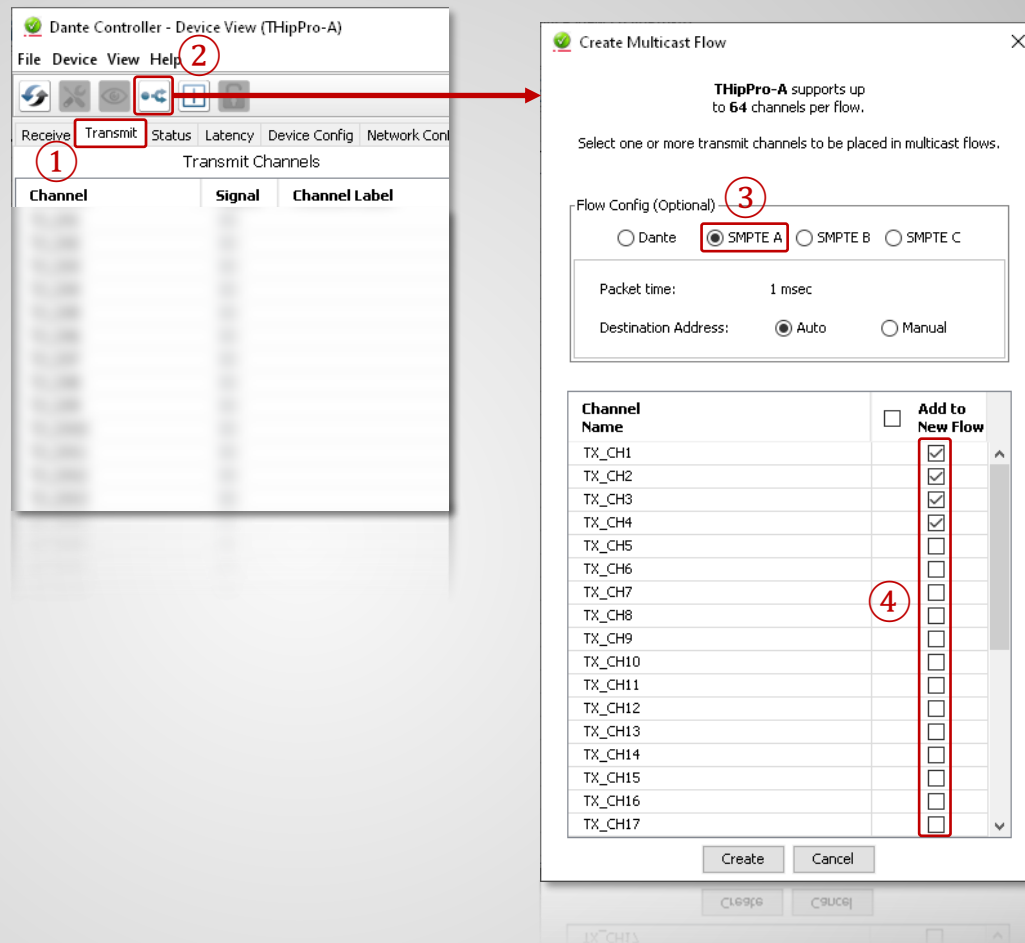
Enable SMPTE for the Domain

- Open Dante Controller and Login ① to the Domain configuration.
- Domain and logged in user are displayed on the menu bar ②.
- Select the DEVICE INFO tab ③.
- Double click on a Device to open the Device View.



Login to the Domain in Dante Controller

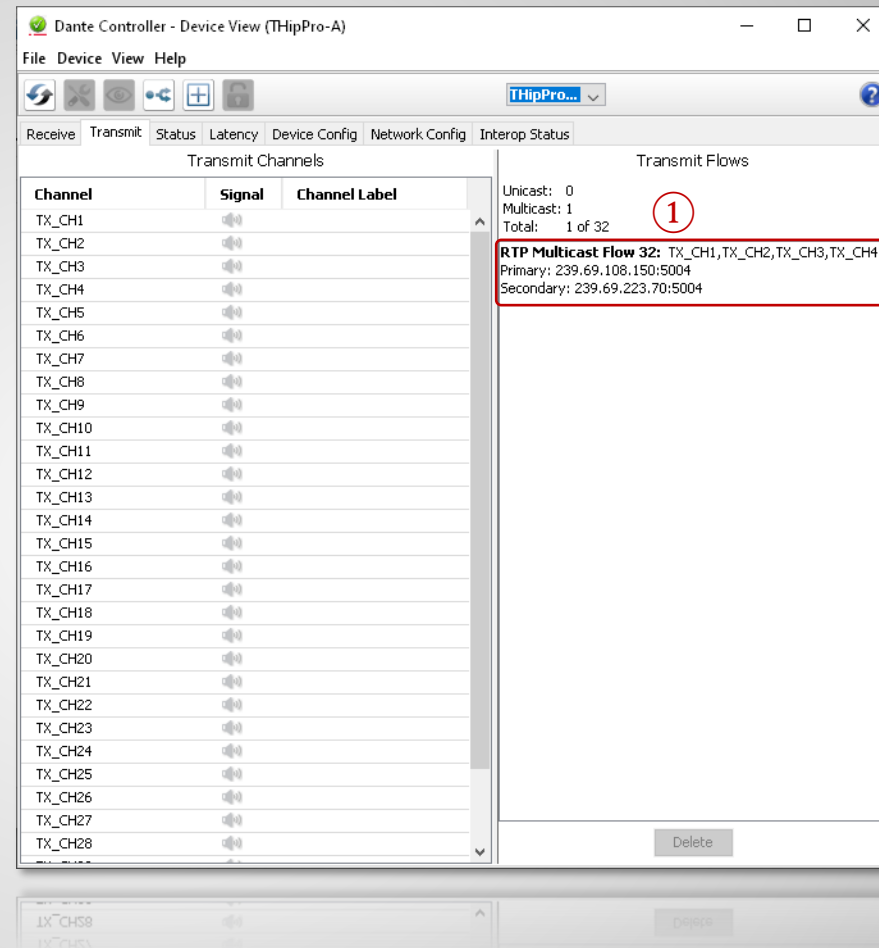
- Select the TRANSMIT tab ① in the Device View.
- Click ADD FLOW ② to open the CREATE MULTICAST FLOW window.
- Select SMPTE A ③ as stream type.
- Select the channels ④ to be included in the new audio stream.



Create SMPTE TX Flows

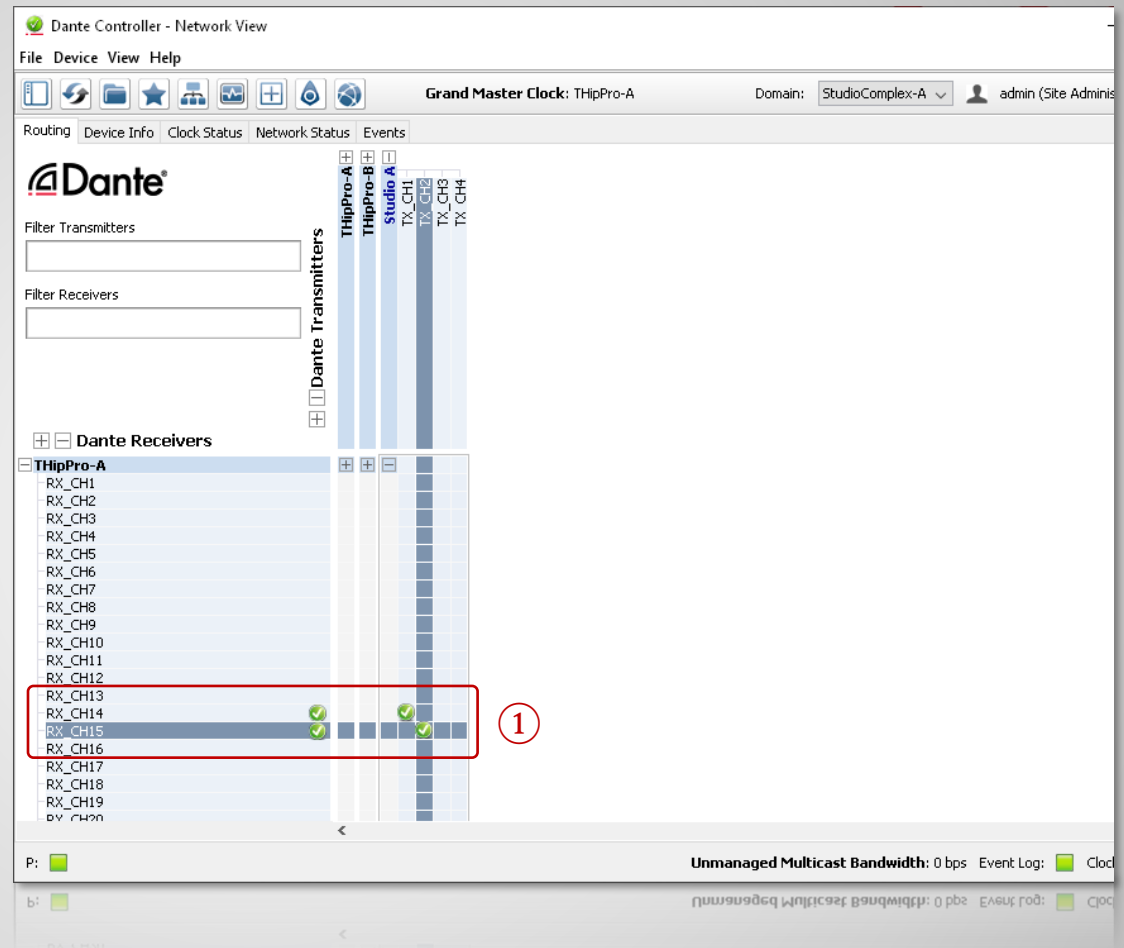
- Active audio streams are displayed on the TRANSMIT tab ① in the Device View:

- Name of the audio stream.
- Channels contained in the audio stream.
- Multicast addresses of the audio streams in the primary and the secondary network.



Check SMPTE TX Flows

- The ROUTING tab of the Dante Controller shows SMPTE flows available in the network in blue.
- Set a cross point ① to subscribe to a channel of the SMPTE flow.



Subscribe to SMPTE Flows

Audio over IP

Configuration RAVENNA

- Open the RAVENNA-module configuration in a web browser and enable EXPERT SETTINGS (❶).
- Use the button CONNECT (❷) under RX STREAMS to subscribe to a AES67 stream.
- Enter a meaningful LABEL (❸) and press APPLY.
- The subscribed channels show up under RX STREAMS. They turn green (❹) if the RAVENNA module receives the AES67 streams from the network.

The image displays three screenshots of the RAVENNA web interface, illustrating the steps to configure RX streams:

- Step 1:** The main configuration page for 'AVT Lawo bare 946/41-60'. The 'Expert Settings' tab is selected (marked with ❶). The 'Rx Streams' section is visible, showing a 'Connect' button (marked with ❷).
- Step 2:** A modal window titled 'Rx Stream Properties' is open. The 'Label' field is set to 'THipPro LAW0' (marked with ❸). The 'Stream Source' is set to 'sap:172.16.75.100#38236 THipPro-Messe : 32'. The 'Receiver Settings' section shows 'Delay (samples)' set to 512 and 'Channel count' set to 7.
- Step 3:** A close-up of the 'Rx Streams' section. The 'THipPro LAW0' stream is listed with its channels (1L, 1R, 2L, 2R, 3L, 3R, 4L) highlighted in green (marked with ❹), indicating it is successfully received.

Configuration RAVENNA streams RX

- Press the button **CREATE** (❶) under **TX STREAMS** to stream audio to the network.
- Enter a meaningful **NAME**.
- Set **PAYLOAD** to **AES67 STANDARD STEREO STREAM** (❷).
- Press the arrow button (❸) and enable **SAP**.
- Apart from that keep the default settings as shown.
- Press **APPLY**.
- The new audio streams show up under **TX STREAMS** (❹).

The image displays the RAVENNA web interface for configuring audio streams. The main window shows the 'Tx Streams' section with a 'Create' button (❶). A modal window titled 'Tx Stream Properties' is open, showing the configuration for a new stream named 'LAWO THipPro' (❸). The 'Payload' is set to 'AES67 Standard Stereo Stream' (❷). The 'Address' is set to 'auto'. The 'Medium' is set to 'RAVENNA Audio-ra0'. The 'Recording tracks' are set to '1L, 1R'. The 'Link' section shows the RTSP URI: 'rtsp://172.16.75.57:8081/by-id/15874928097930248195'. The 'Apply' button is visible at the bottom right of the modal. A red arrow points from the 'SAP' button in the 'Advertise this session as' section to the 'SAP' button in the 'Tx Stream Properties' modal.

Configuration RAVENNA streams TX

- If SAP (Session Announcement Protocol) should not be used, Ravenna can exchange the stream information as text in SDP format.
- Open the RAVENNA-module configuration in a web browser and enable EXPERT SETTINGS.
- Subscribe to an existing AES67 stream:
 - Press CONNECT under RX STREAMS and choose CUSTOM URL.
 - Enable SHOW RAW SDP.
 - Paste the content of an SDP file.
- Create a TX an SDP file for a TX stream:
 - Click on an AES67 stream under TX STREAMS.
 - Click on SDP.
 - Copy the content and paste it to a text file with *.sdp extension.

RAVENNA streams without SAP

Audio over IP

Configuration Livewire+

- Open the Axia xNode configuration page in the web browser.
- Under SOURCES, audio streams in send direction are assigned to the audio inputs.
- Two modes are available for AES67:
 - Stereo 1ms (AES67) generates an AES67 stream with 2 channels.
 - 8ch 1ms (AES67) generates an AES67 stream with eight channels.
- Via the link DOWNLOAD STREAM DESCRIPTION (SDP) the definition of the AES67 stream can be downloaded to your PC as a file in SDP format.

Axia xNode
AES/EBU 4x4 I/O

System options
Home
Simple Setup
Unicast Link
Advanced options
Sources
Destinations
Mixer
Meters
Synchronization and QoS
System

Sources

#	Source Name:	Channel/Address:	Stream Mode:	Input Gain [dB]:
1	Axia01_8ch	239.0.100.14	8ch 1ms (AES67)	0.0
AES67: Download stream description (SDP), RTSP: rtsp://172.20.100.14/by-id/1				
AES 1-L			Surround: Center, LFE	0.0
AES 1-L			Surround: Back L, R	0.0
AES 1-L			Stereo L, R	0.0
5	SRC 5	105	Disabled	0.0
6	SRC 6	106	Disabled	0.0
7	SRC 7	107	Disabled	0.0
8	SRC 8	108	Disabled	0.0

[Show source allocation status](#)

Channel/Address empty Access using AES67/SIP or RTSP. IP unicast will be used as a transport.
 Channel Number Unique channel number of Livewire multicast stream (any number from 1 to 32767)
 IP Address Destination multicast address of the stream if other range than Livewire is required

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Axia xNode Configuration (1)

- Under DESTINATIONS, audio streams are assigned to the audio outputs in the receiving direction.
- Two types are available:
 - FROM SOURCE to receive a 2-channel AES67 stream.
 - SURROUND to receive an 8-channel AES67 stream.
- The multicast address of the AES67 stream is entered under CHANNEL/ADDRESS. If the port differs from the Axia standard port 5004, it can be appended with ":".

Axia xNode
AES/EBU 4x4 I/O

System options
Home
Simple Setup
Unicast Link
Advanced options
Sources
Destinations
Mixer
Meters
Synchronization and QoS
System

Destinations

#	Name:	Channel/Address:	Type:	Gain [dB]:
1	DST 1	239.0.30.15:5300	Surround: Front L, R	0.0
2			Surround: Center, LFE	0.0
3			Surround: Back L, R	0.0
4			Stereo L, R	0.0
5	DST 5	5	From source	0.0
6	DST 6	6	From source	0.0
7	DST 7	7	From source	0.0
8	DST 8	8	From source	0.0

Apply

Channel Type: From/To/Surround Livewire channel number or stream address (stereo or 8-channel)
Address Type: From/Surround AES67 SIP URI
 Type: Ravenna Ravenna session name

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Axia xNode Configuration (2)

Audio over IP

Interworking AES67 – Dante

- Audio stream AES67 → Dante:
 - Create an AES67 TX audio stream using the AVT MAGIC PC software.
 - Select the second byte of the multicast address matching the AES67 MULTICAST ADDRESS PREFIX of the Dante network.
 - Connect these streams to the Dante receivers via the Dante Controller software.
- Audio stream Dante → AES67:
 - Create AES67 TX audio streams using the Dante Controller software.
 - Subscribe to these audio streams using the AVT MAGIC PC software.

Interworking AES67 – Dante

Audio over IP

Interworking AES67 – RAVENNA

- Audio stream AES67 → RAVENNA:
 - Create AES67 TX audio streams using the AVT MAGIC PC software.
 - Subscribe to these audio streams via the RAVENNA web interface.
- Audio Stream RAVENNA → AES67 :
 - Create AES67 SAP audio streams using the RAVENNA web interface.
 - Subscribe to these audio streams using the AVT MAGIC PC software.

Interworking AES67 – RAVENNA

Audio over IP

Interworking AES67 – Livewire+

- Audio stream AES67 → Livewire+:
 - Create AES67 TX audio streams using the AVT MAGIC PC software.
 - An Axia xNode only accepts AES67 streams with 2 or 8 channels and 48 kHz sampling rate.
 - Either L16 or L24 can be selected as bitrate. This is automatically detected by the Axia xNode.
 - Enter the streams information into the Axia xNode web interface.
 - If no audio signal is output although all parameters have been entered correctly, restarting the Axia xNode may help.
- Audio Stream Livewire+ → AES67 :
 - Create AES67 audio streams using the Axia xNode web interface.
 - Download the stream description as an SDP file from the Axia xNode web interface.
 - Change the file extension of the SDP file to ".sdp".
 - Import this SDP file using the AVT MAGIC PC software.

Tested with Axia xNode firmware version 2.2.2.

Interworking AES67 – Livewire+

Audio over IP

Interworking Dante – RAVENNA

- Audio stream Dante → RAVENNA:
 - Set the AES67 MULTICAST ADDRESS PREFIX in the Dante module via the Dante Controller software. Ravenna streams and AES67 streams must use the same multicast address prefix otherwise audio will not come through. The multicast address prefix must not be 0.
 - Create AES67 TX audio streams via Dante Controller software.
 - Subscribe to these audio streams via RAVENNA web interface.
- Audio stream RAVENNA → Dante:
 - Create AES67 SAP audio streams via RAVENNA web interface.
 - Connect these streams to Dante receivers via Dante Controller software.

Interworking Dante – RAVENNA

Audio over IP

Interworking Dante – Livewire+

- Audio stream Dante → Livewire+:
 - Create AES67 TX audio streams using the Dante Controller software.
 - Enter the streams details into the Axia xNode web interface.
- Audio stream Livewire+ → Dante:
 - Create AES67 audio streams with 2 channels (stereo 1ms (AES67)) via the Axia xNode web interface.
 - Switch the ENABLE SAP ANNOUNCEMENTS setting to YES in the web interface of the Axia xNode under SNYCHRONIZATION AND QOS.
 - Connect these streams to the Dante receivers using the Dante Controller software.

Tested with Axia xNode firmware version 2.2.2.

Interworking Dante – Livewire+

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