Dante Interoperability AES67/ST2110

Technical Dive

Interoperability Context

AES67

SMPTE | ST 2110

Interoperability: A Word Processing Analogy

Microsoft® Word

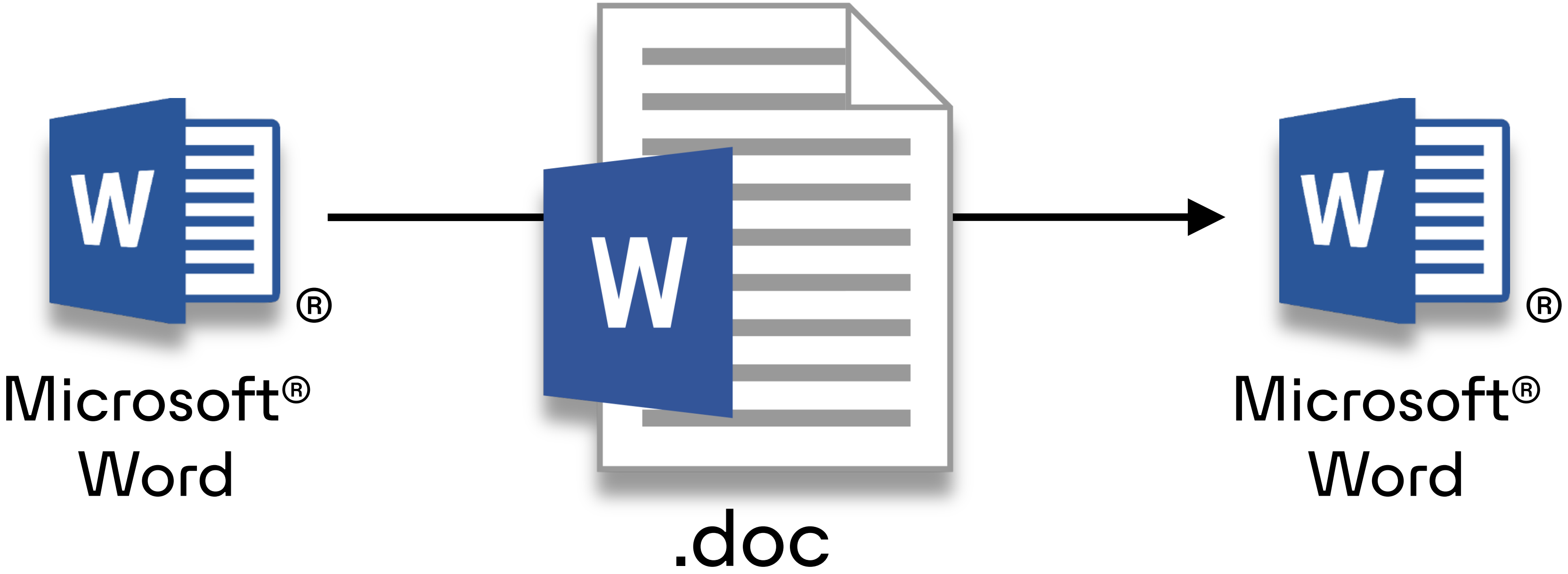
Corel® WordPerfect®

Google Documents

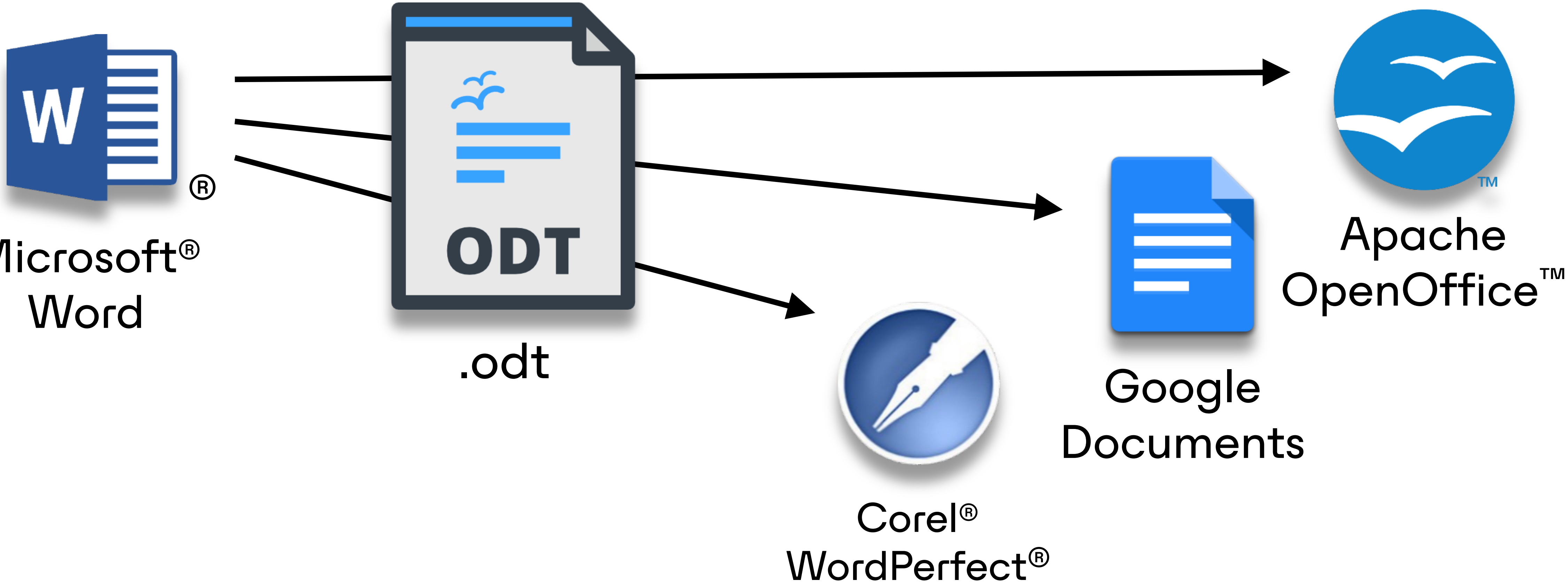
Apache OpenOffice™

Interoperability: A Word Processing Analogy

Dante®



Interoperability: A Word Processing Analogy



The Goal of AES67

Best Practices for Audio Networks

AES AESTD1003V1, Published June 6, 2009

http://www.aes.org/technical/documents/AESTD1003V1.pdf


AES WHITE PAPER

Best Practices in Network Audio

AESTD1003V1 2009-06-04

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EXECUTIVE SUMMARY

Analog audio needs a separate physical circuit for each channel. Each microphone in a studio or on a stage, for example, must have its own circuit back to the mixer. Routing of the signals is inflexible. Digital audio is frequently wired in a similar way to analog. Although several channels can share a single physical circuit (e.g., up to 64 with AES10), thus reducing the number of cores needed in a cable. Routing of signals is still inflexible and any change to the equipment in a location is liable to require new cabling.

Networks allow much more flexibility. Any piece of equipment plugged into the network is able to communicate with any other. However, installers of audio networks need to be aware of a number of issues that affect audio signals but are not important for data networks and are not addressed by current IT networking technologies such as IP. This white paper examines these issues and provides guidance to installers and users that can help them build successful networked systems.

1 BACKGROUND

Network audio technologies find application across a wide number of domains, including studio recording and production activities, archiving and storage, musical performance in theater or concert, and broadcasting. These range in scale from desktop audio production systems to 500-acre theme-park installations.

Each domain imposes its own, sometimes conflicting, technical requirements. For example, the low-latency demands of live sound can be ...

... 1394. Differences between the various implementations include issues of audio format, number of channels supported, and end-to-end transport latency. In addition to the large number of low-fidelity consumer applications for network audio streaming, various software architectures have been developed to support professional grade audio transport over the Internet, albeit without the higher performance that characterizes local area network solutions.

1.1 Marketplace

Outside the telecommunications industry, audio networking was first conceived as a means of transferring work units within large studios and postproduction facilities. In the 1990s when the digital audio workstation became the dominant audio production platform, standards-based Token Ring (IEEE 802.5) and Ethernet (IEEE 802.3) networks were employed to transfer audio files from one workstation to another. Initially these were non-real-time transfers, but facility operators, technology providers and networking standards grew to allow real-time playback over the network for the purpose of media-storage consolidation, final mix-down, and production.

True real-time audio networking was first introduced in installed sound reinforcement applications. By the mid 1990s, digital signal processing was in widespread use in this market segment [12][19]. Digital signal processing improved the flexibility and scalability of sound reinforcement systems. It became possible to infuse facilities such as theme parks and stadiums with hundreds of individually processed audio signals, and thus, the need arose for a flexible, optimized distribution system for these signals. Since audio

2009 September 729

Dante® Livewire WHEATNET IP CobraNet™ Ether Sound OPTOCORE

Commonality:

"Audio networking systems are characterized by the transport of synchronized uncompressed audio in PCM format, which in principle could be reformatted as requested."

Differences:

"In practice, there are several issues for compatibility between formats that should be addressed and solved with specific implementations."

AES67 Ratified 2013, Updated 2014, 2015, 2018

FAQ: What is AES67?

Media Networking Alliance
Promoting the Adoption of AES67

AES67 enables interoperability between audio networking [solutions] currently available, such as Dante, Livewire+, Q-LAN, Ravenna [and WheatNet IP].

AES67 is not a new technology, but a bridging compliance mode.

AES67

Media Networking Alliance Web Page, FAQ Section. https://medianetworkingalliance.com/faq/

FAQ: What is AES67?

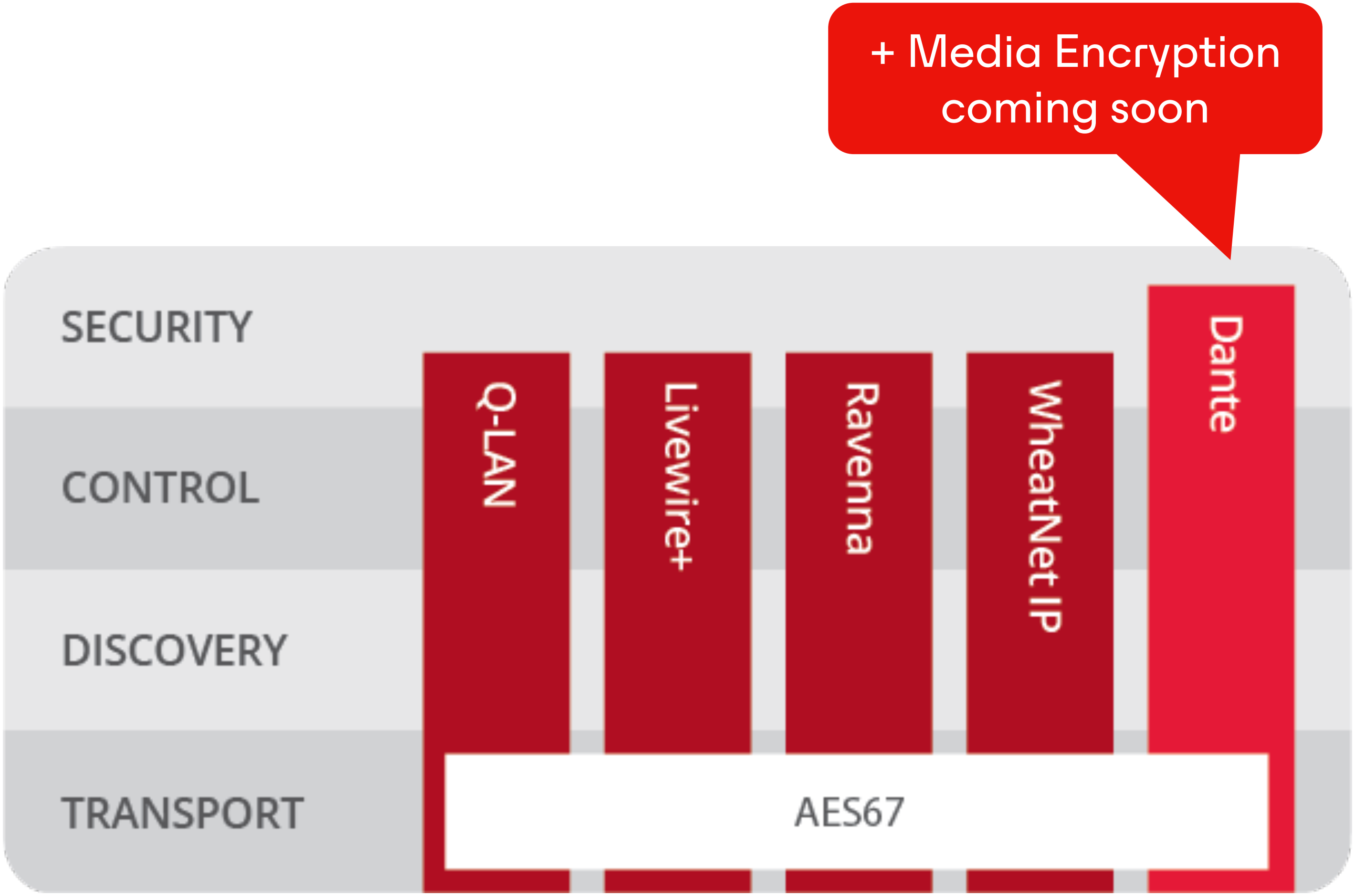
Interoperability Standard:

AES67

Network Solutions:

Dante, Livewire+, Ravenna,

Q-LAN, WheatNet IP



SSL: *IP Technology for Broadcast Audio Routing Systems.*, https://s3.us-west-2.amazonaws.com/sslweb.solidstatelogic.com/content/system-t/SSL%20AoIP%20Information.pdf

AES67 Scope

AES67 defines some common building blocks:

- Synchronization: PTPv2 Multicast (Domain 0)
- Session Description: Stream Description Protocol (SDP)
- Encoding & Streaming: Real-time Transport Protocol (RTP)
    - Pivot format: 48Khz / 16-24bit / 1ms packet time / 1~8 channels
- Connection Management: Multicast (IGMPv2/3)
    - Audio flows need to be Multicast
- Quality of Service (QoS): DiffServ
    - PTP EF (46) / RTP AF41 (34)

AES67 doesn't reinforce:

- Discovery: Manufacturer specific
    - mDNS, Session Announcement Protocol (SAP), SDP copy&paste
- Redundancy

What is SMPTE ST 2110?

SMPTE ST 2110 is a suite of standards and specifications primarily focused on the transport of video on professional production environments.

- Flexibility: Evolution from a point-to-point solution like SDI to a network-based solution.
- More bandwidth: Allowing uncompressed video streams
- Efficiency: Separating the audio, video and ancillary data streams.

SMPTE ST 2110 Audio Scope

ST 2110 audio steams are essentially a carbon copy of AES67*

- 2110-10: Defines the timing relationships between all components in the system: Media, RTP and PTP clocks
    - Common reference clock using IEEE 1588-2008 Precision Time Protocol (PTPv2)
- 2110-30: RTP Audio transport, like AES67 (Multicast!)
  Conformance levels:
    - A – 48KHz streams, 1 to 8 audio channels, 1 ms packet times
    - B – Level A plus: 1 to 8 audio channels, 125 μs packet times
    - C – Level A plus: 1 to 64 audio channels, 125 μs packet times
- 2059-2: PTP Media Profile (IEEE 1588-2008)
- **2022-7: Redundancy**
    - RTP headers and payloads need to be identical between the two streams

with a few parameters tightened down.

SDP Anatomy

Dante

```
v=0
o=- 123456 11 IN IP4 192.168.100.2
s=Example of a SMPTE ST2110-20 signal
i=this example is for 720p video at 59.94
t=0 0
a=recvonly
a=group:DUP primary secondary
m=video 50000 RTP/AVP 112
c=IN IP4 239.100.9.10/32
a=source-filter:incl IN IP4 239.100.9.10 192.168.100.2
a=rtpmap:112 raw/90000
a=fmtp:112 sampling=YCbCr-4:2:2; width=1280; height=720;
exactframerate=60000/1001; depth=10; TCS=SDR; colorimetry=BT709;
PM=2110GPM; SSN=ST2110-20:2017;
a=ts-refclk:ptp=IEEE1588-2008:39-A7-94-FF-FE-07-CB-D0:37
a=mediaclk:direct=0
a=mid:primary
m=video 50020 RTP/AVP 112
c=IN IP4 239.101.9.10/32
a=source-filter:incl IN IP4 239.101.9.10 192.168.101.2
a=rtpmap:112 raw/90000
a=fmtp:112 sampling=YCbCr-4:2:2; width=1280; height=720;
exactframerate=60000/1001; depth=10; TCS=SDR; colorimetry=BT709;
PM=2110GPM; SSN=ST2110-20:2017;
a=ts-refclk:ptp=IEEE1588-2008:39-A7-94-FF-FE-07-CB-D0:37
a=mediaclk:direct=0
a=mid:secondary
```

Session ID

Version

Session Name

Session Info

Port

Sender (Primary)

Directly references to the clock in *ts-refclk*

Sender (Secondary)

Group Address

SDP Anatomy Clock focus

- Reference Clock

“a=ts-refclk:localmac=<Ethernet MAC address of sender> ”

- PTP Form

“a=ts-refclk:ptp=IEEE1588-2008:39-A7-94-FF-FE-07-CB-D0:37

a=ts-refclk:ptp=traceable

a=ts-refclk:localmac=7C-E9-D3-1B-9A-AF”

Domain

Traceable time source

Sender

Grand Leader

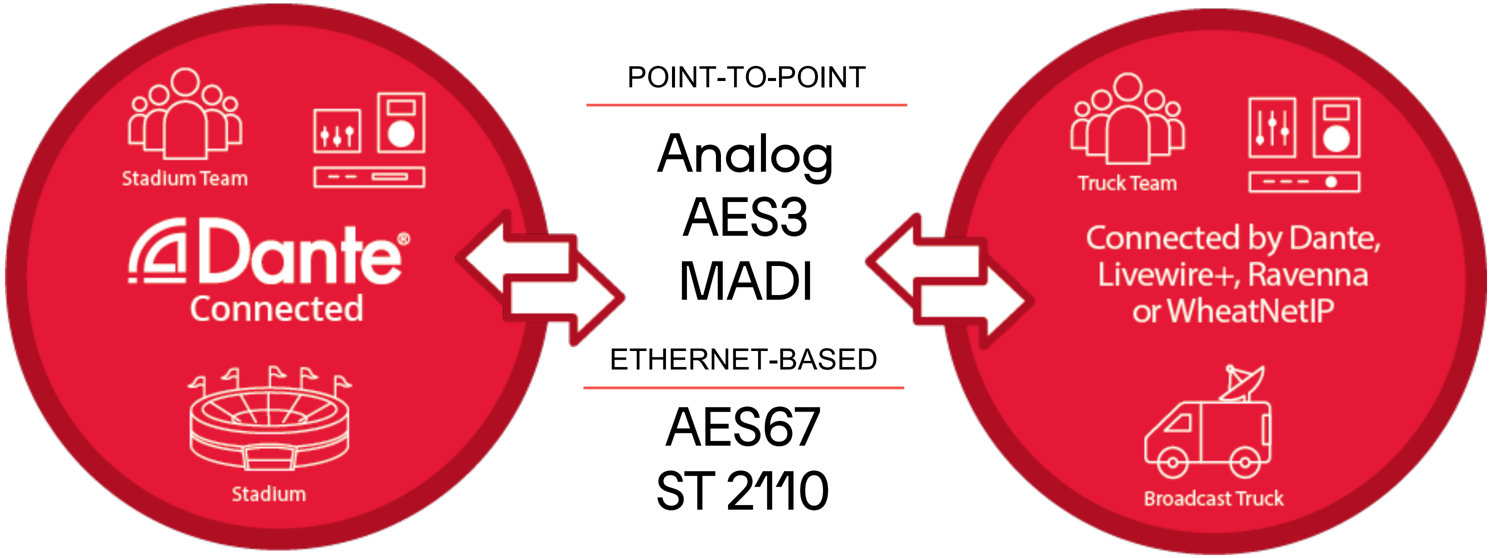
The Goals of AES67 and ST 2110 in Audio

In the audio space,

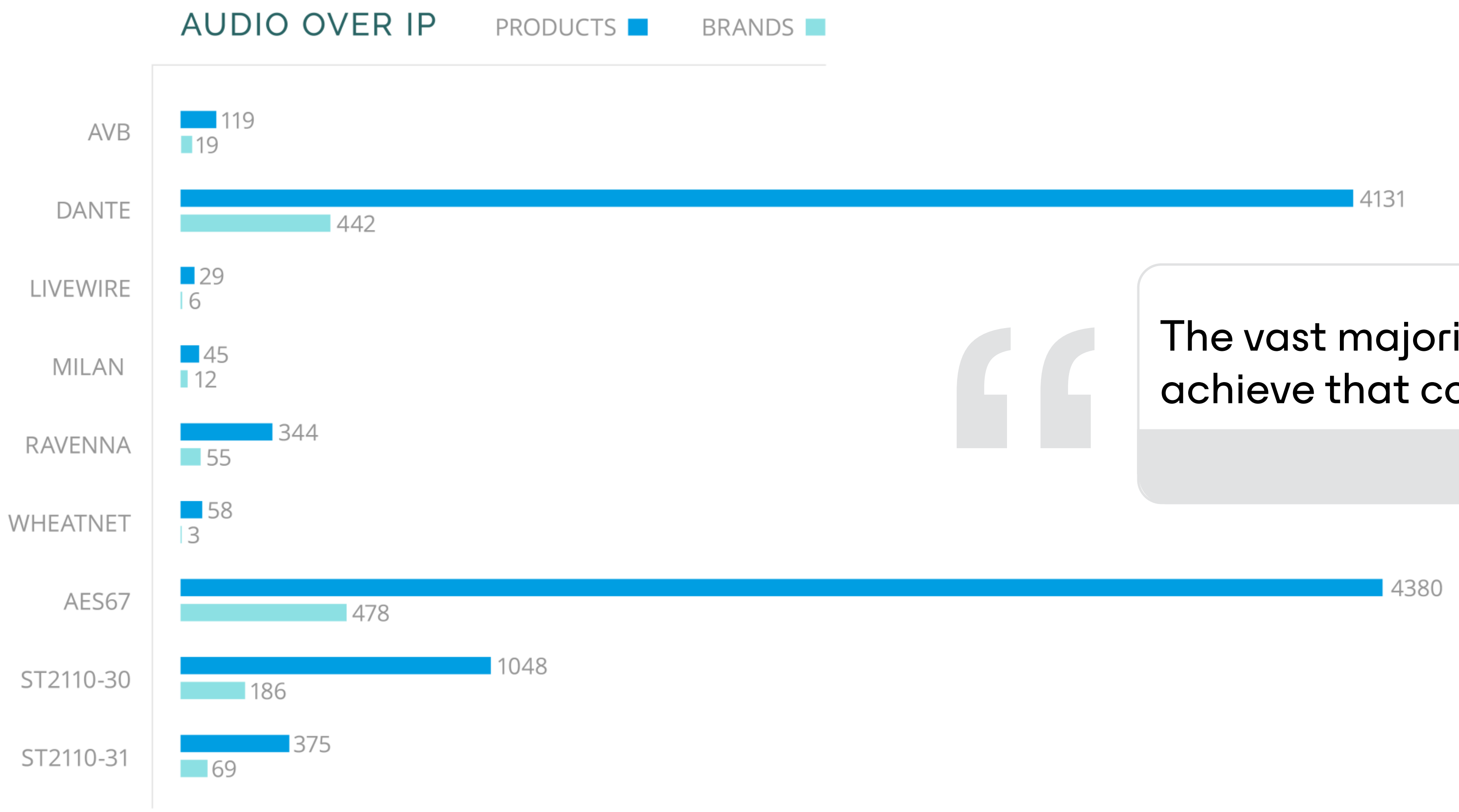
AES67 and ST 2110 replace:

- Video: SDI Embed/De-Embed

- Audio: MADI, AES3 and Analog



Market Presence



> The vast majority of AES67 products achieve that compatibility using Dante.
>
> - Roland Hemming

https://rhconsulting.uk/blog/networked-audio-products-2024/

Interoperability responsibility

Audinate maintains active relationships with our manufacturer partners to enable the seameless interoperability between the wide range of Dante enabled products available in the market. By positioning ourselves as the Dante experts in the partnership, we free up manufacturers to focus on their strengths.

We acknowledge the role played by standards such AES67 and SMPTE 2110 in promoting vendor agnostic device interoperability. Manufacturers implementing these standards are faced with the responsibility of not only creating standards compliant devices but are also tasked on working diligently to ensure device interoperability.

Dante Design & Implementation

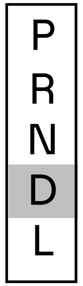
AES67 vs ST 2110 Modes
Managed vs Unmanaged Dante
PTPv1 and PTPv2
QoS DSCP Markings

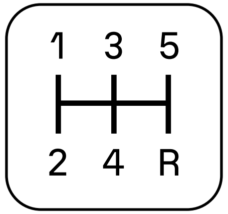
Dante Interoperability “modes”

Dante

Dante with AES67

Managed
Unmanaged





Dante with SMPTE ST 2110

Managed

Dante Software & Firmware Requirements

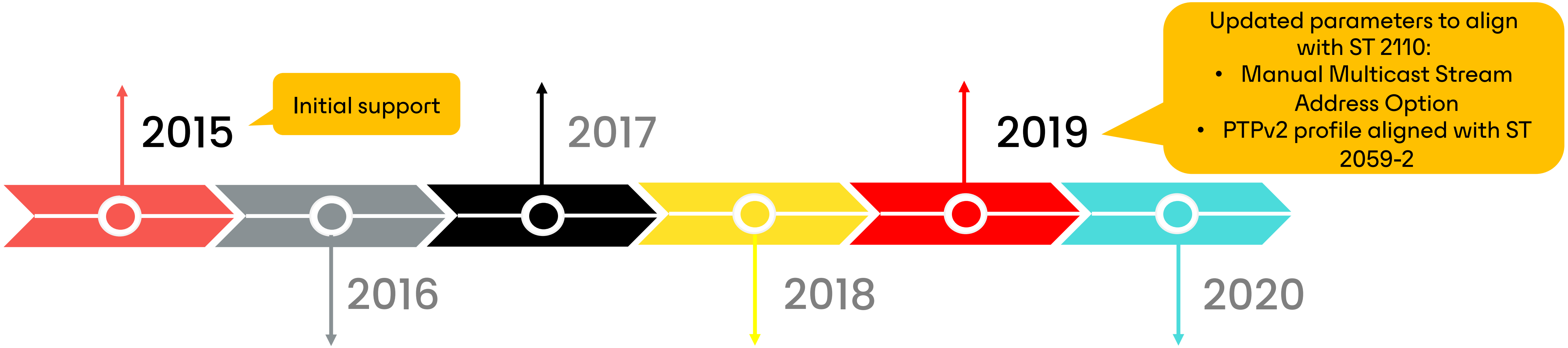
Dante Device	Unmanaged AES67 Dante Controller (v4.2)	Managed AES67 or ST2110 Dante Domain Manager (v1.1)
Brooklyn 2/3 Broadway HC PCIe IP Core	Supported (v3.9)	Supported (v4.2)
UltimoX AVIO	Supported (v4.1)	Unsupported
Dante Embedded Platform	Supported (v1.2.1.1)	Unsupported
Dante AV-H Dante AV-A	Supported (v1.0.7)	Unsupported
Dante AV Ultra Dante Virtual Soundcard Dante VIA Dante Studio Dante Application Library	Unsupported	Unsupported

Dante AES67

Dante with AES67

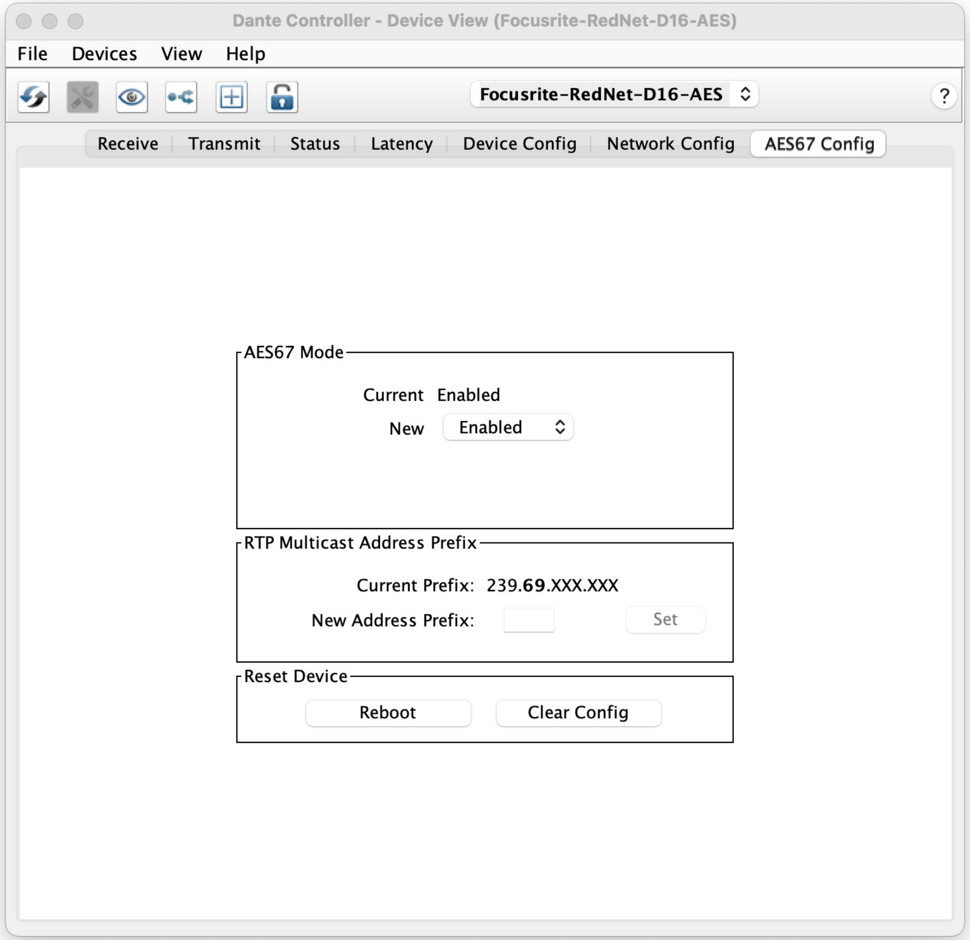
P
R
N
D
L

- “Simple” system configuration.
- Works Unmanaged or Managed (extra settings)
- Hardcoded to PTPv2 Domain 0 (for simplicity)
  - DDM gives extra PTP options
- Requires SAP Discovery Announcements
  - DDM allows manual SDP import



Unmanaged AES67: Enable

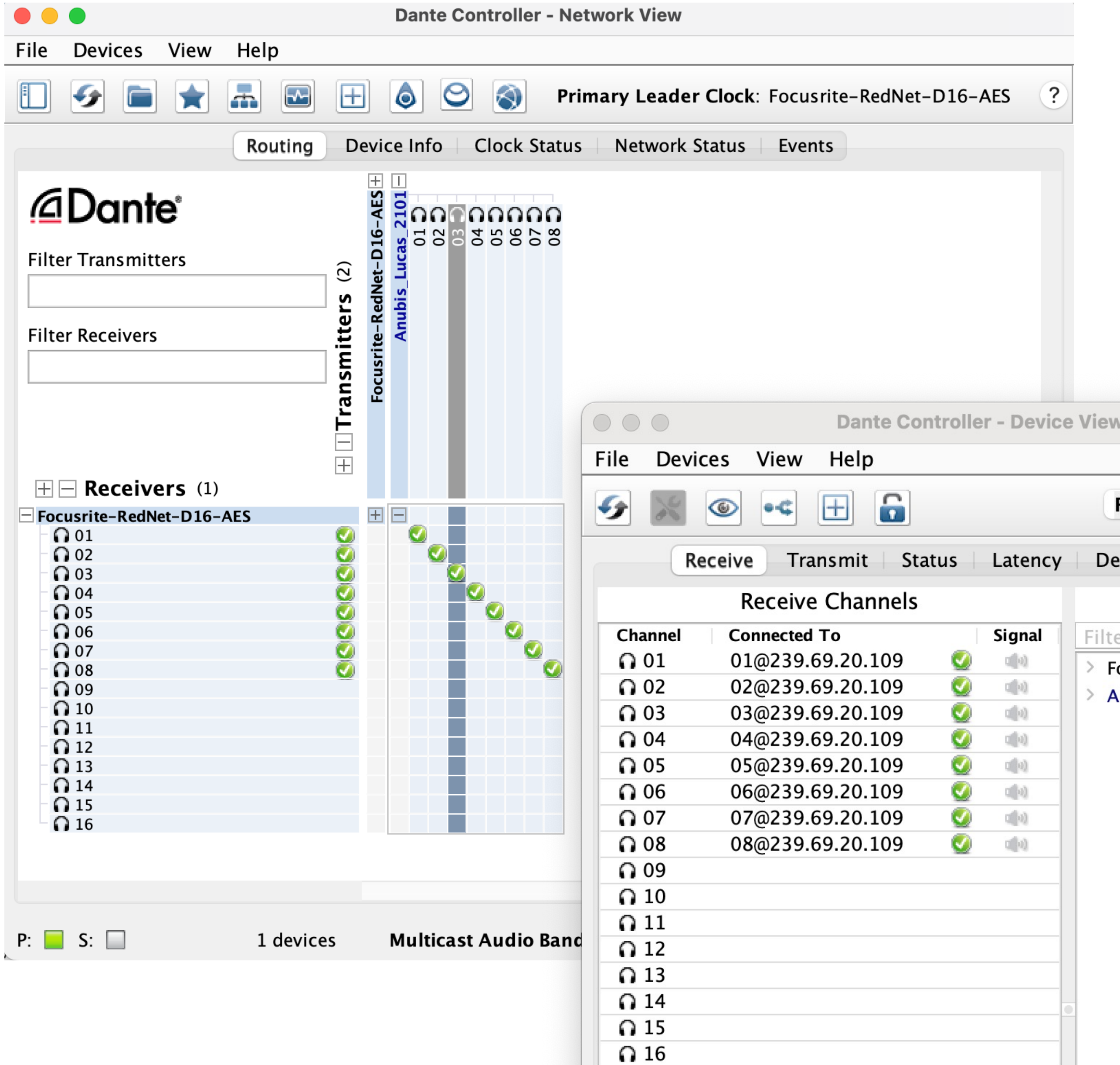
1. AES67 Mode needs to be Enabled in Dante Controller for each Device
2. RTP Multicast Address Prefix can be modified to:
   1. Match 3rd party incoming RTP flows*
   2. Set default Dante outcoming RTP flows range (*can be manually override*)



**Due to Hardware Limitations*

Unmanaged AES67: Subscribe to 3rd party

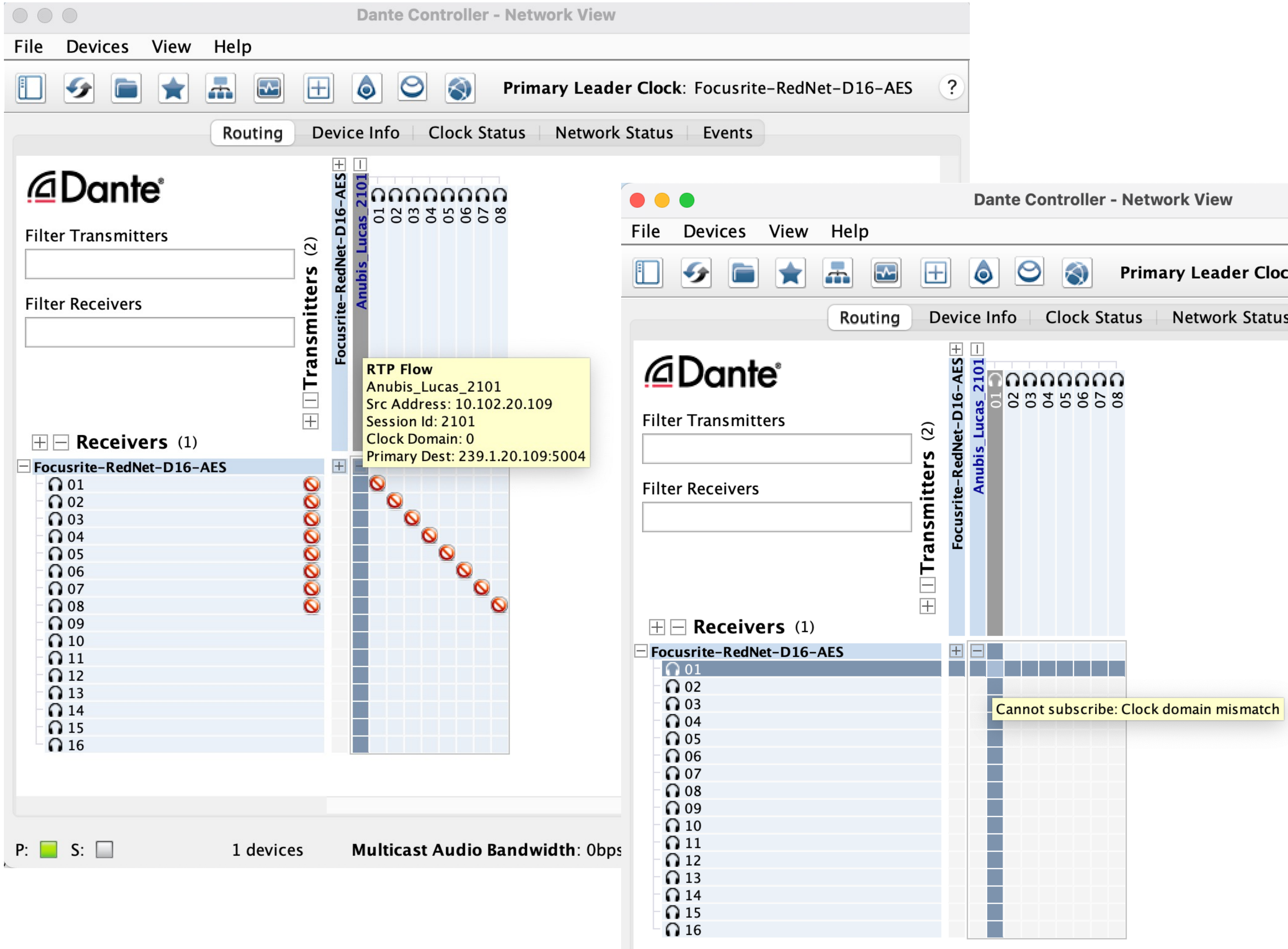
1. 3rd party devices need to announce their AES67 streams using the Session Announcement Protocol (SAP)
   1. Simple protocol for distributing the SDP content over multicast (239.255.255.255)
   2. Dante Controller will show the streams in Blue
2. Subscription is then possible
   1. RX Latency is fixed to 2ms



Unmanaged AES67: Subscribe to 3rd party

Troubleshooting

1. **No audio data** can be caused by
   1. Wrong IGMP settings
   2. Incorrect 3rd party RTP Prefix
2. Impossible subscription
   1. Wrong 3rd party PTP settings



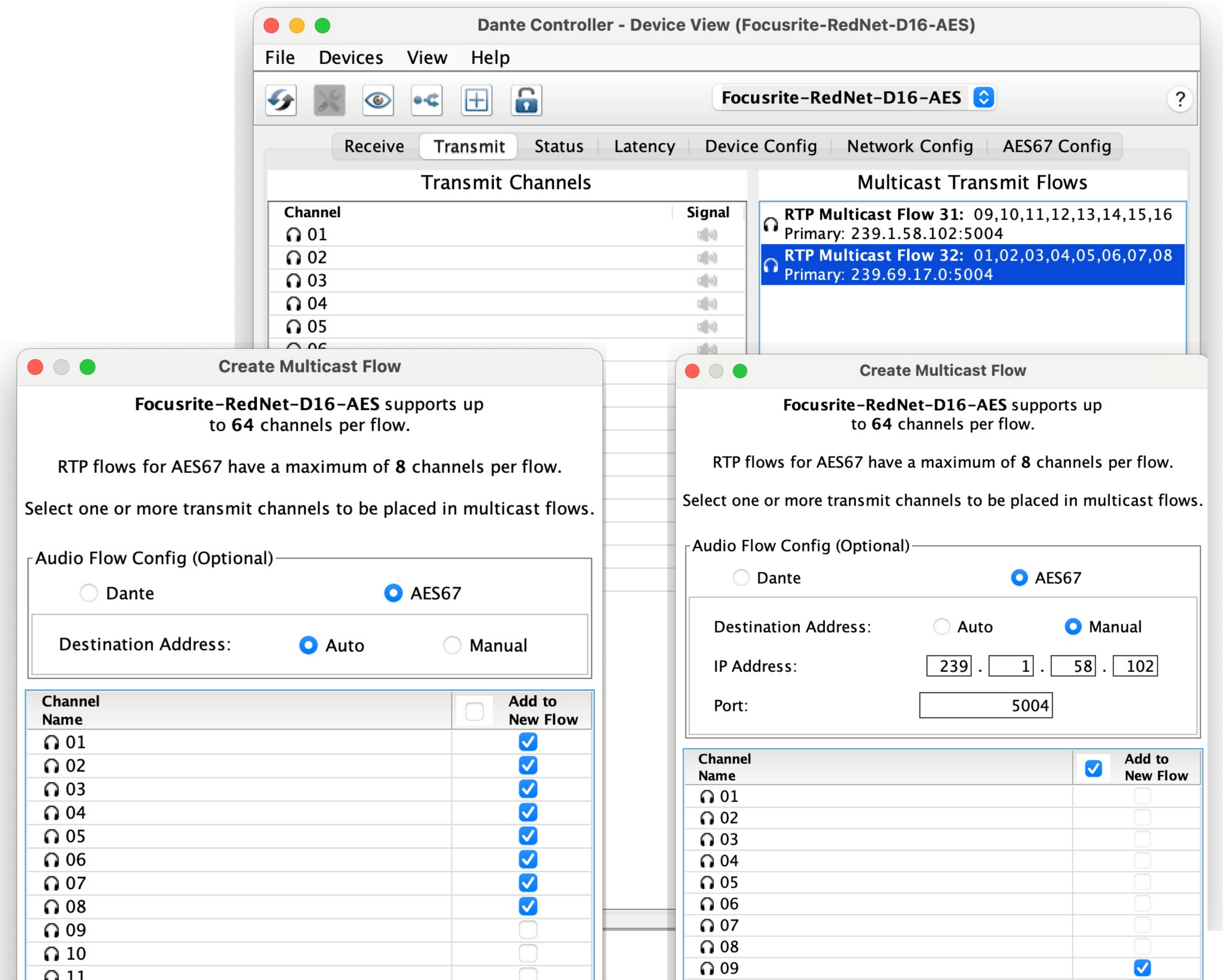
Unmanaged AES67: Create RTP Flow

Dante

1. AES67 Multicast flows can be created from Dante Controller
   1. Using automatic destination addresses (based on the device RTP Prefix)
   2. Manually specifying the Destination IP + Port

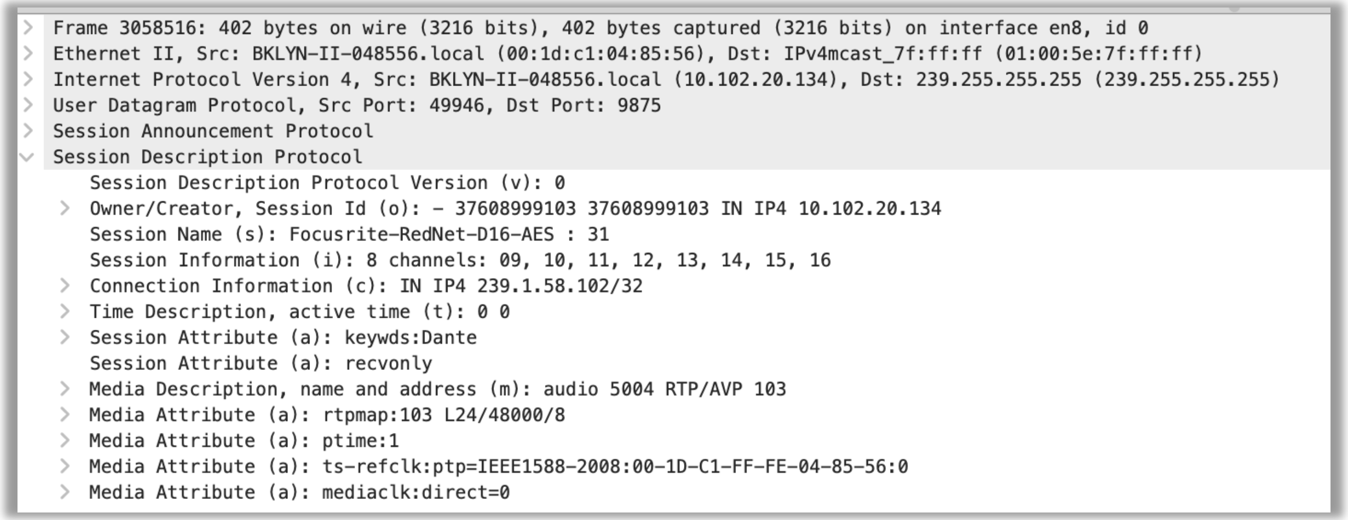
Audio between Dante devices will not use AES67 flows

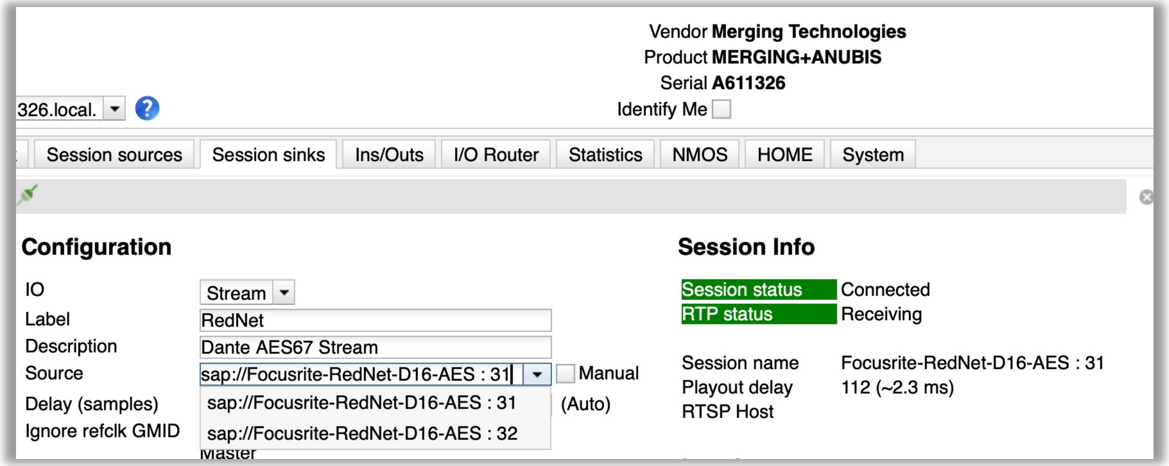
Bandwidth = Dante flows + AES67 flows



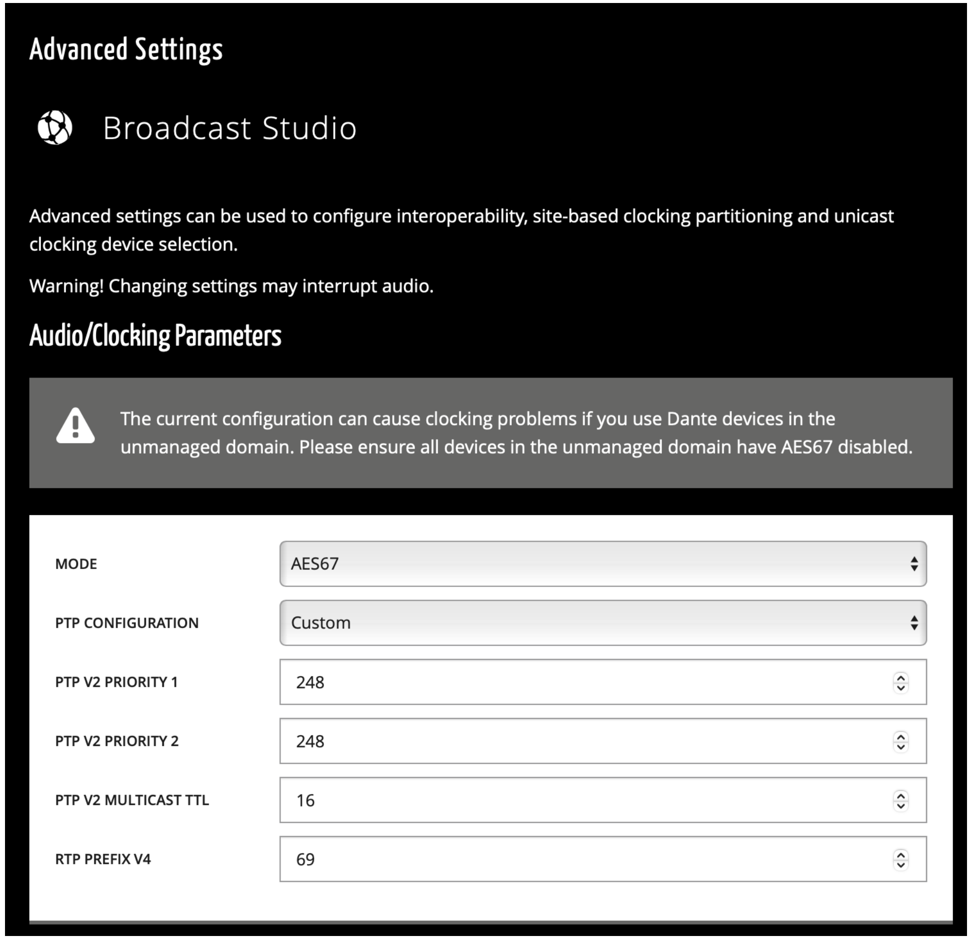
Unmanaged AES67: Create RTP Flow

- Dante devices will automatically announce the SDP session over SAP
  - *Won't be visible in Dante Controller*
- Will be automatically displayed on 3rd party devices with SAP support

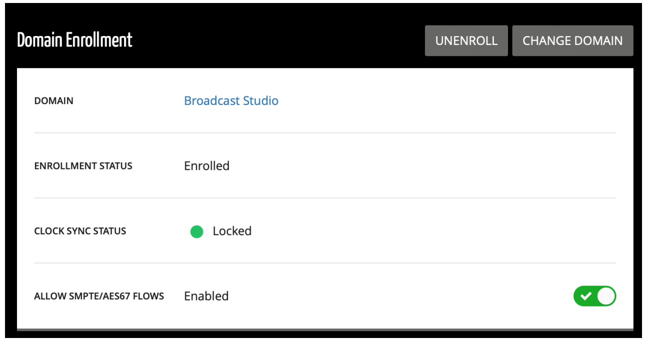




Managed AES67 with DDM

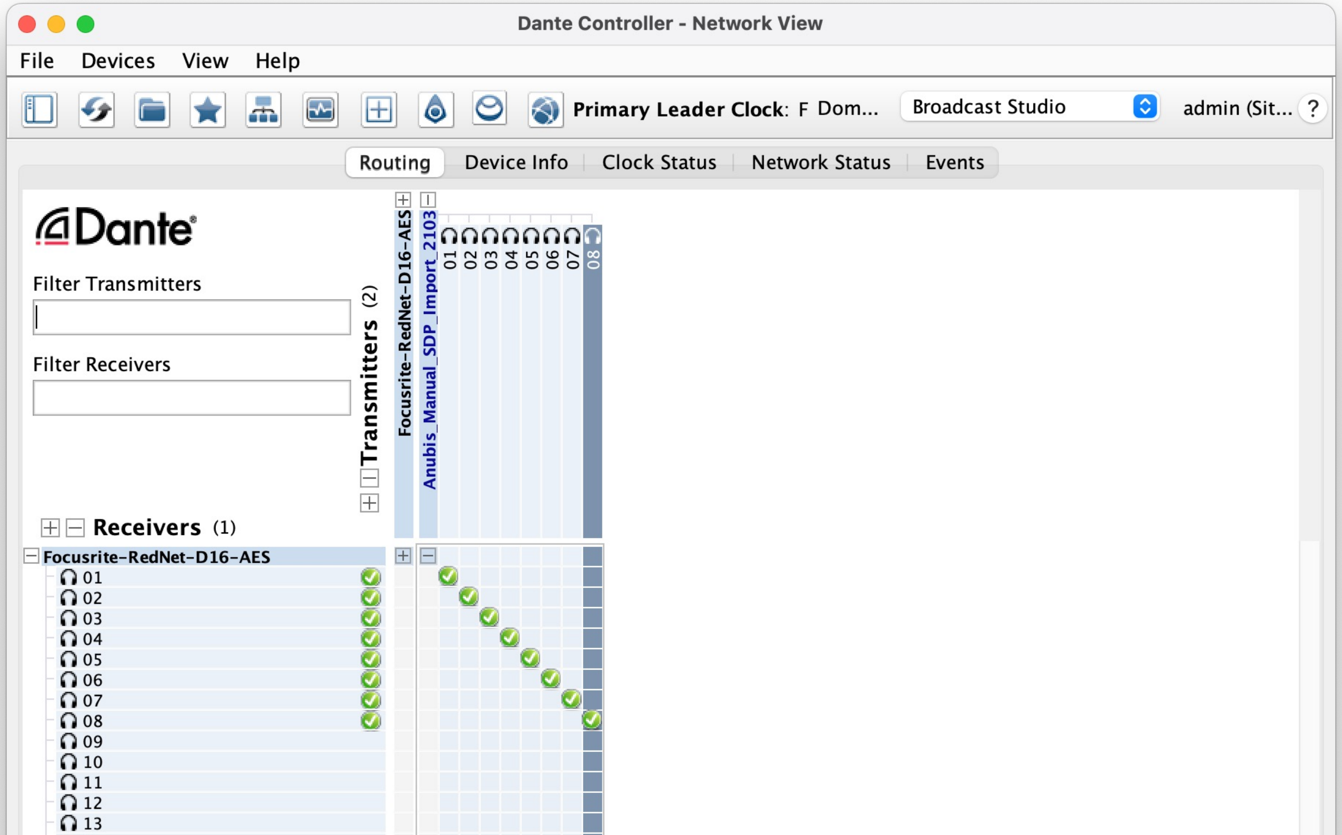


- Enable AES67 mode and RTP Prefix at the Domain Level
    - Only on one Domain or Shared Audio Group
    - Can be restricted by Device
- Allow fine tuning of PTP settings
    - **PTPv2 Priorities 1 & 2**: Determines which devices in a PTPv2 clock domain will be automatically elected as clock leader
    - **Time To Live (TTL):** The range over which a PTPv2 multicast packet is propagated in your network
    - **RTP Prefix v4**: The IP address prefix for RTP flows (second octet)



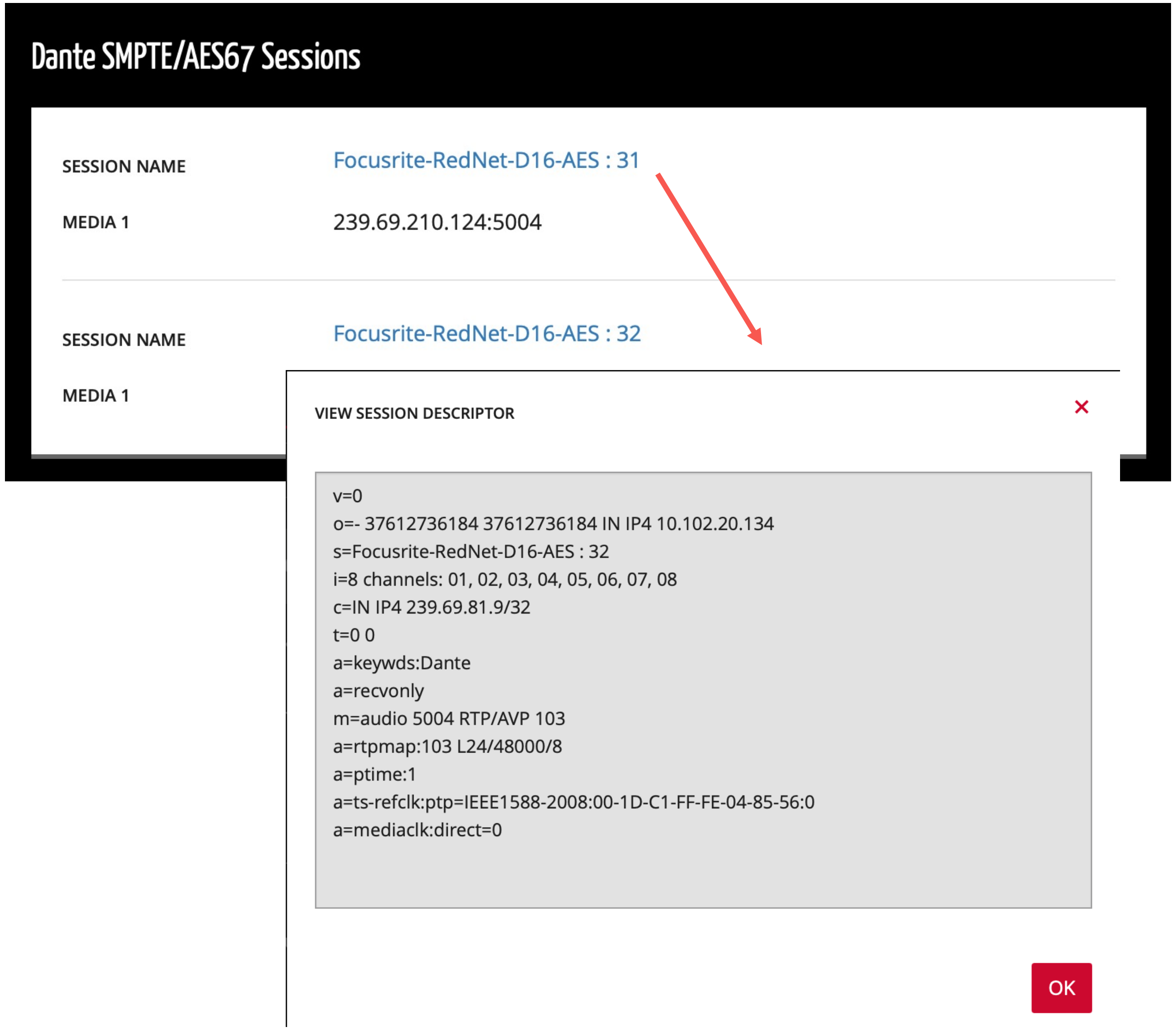
Managed AES67: Import 3rd party SDP

1. Session Description Protocol (SDP) details from 3rd party devices can be imported manually at the Domain Level
    1. 3rd party devices with SAP support will still be shown in Dante Controller (*if DC is in the same subnet as the device*)
2. DDM will display them as a Blue Virtual Device inside the Domain



Managed AES67: Export Dante SDP

1. Session Description Protocol (SDP) details from Dante devices can be exported manually at the Domain Level
    1. Flow still needs to be created in Dante Controller
    2. Dante devices will continue to announce the Flows over SAP



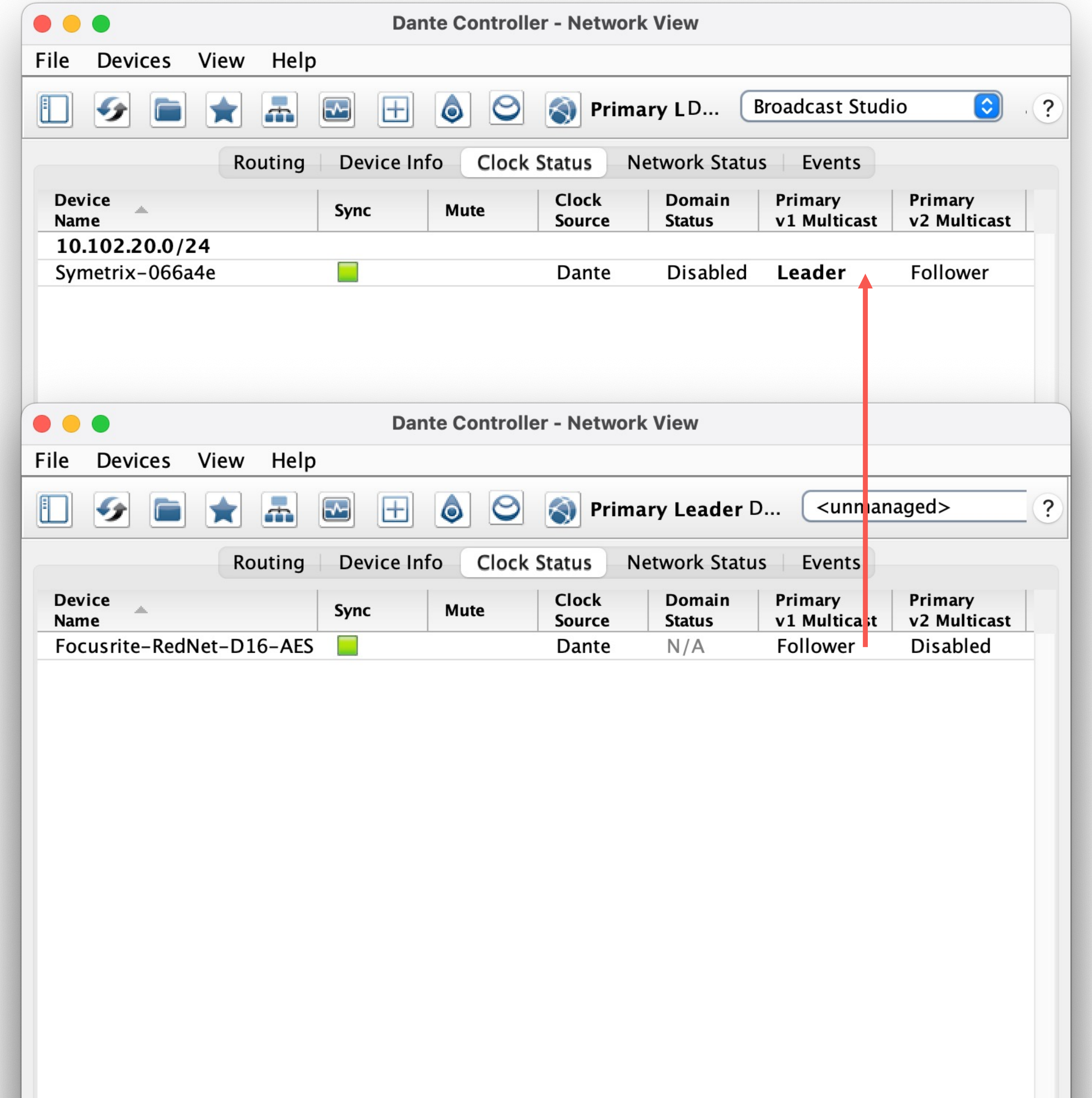
Managed AES67 and Unmanaged Devices

Managed AES67 Modes (or ST2110 or Dante PTPv2 using the PTPv2 Domain 0):

- Use the same PTPv1 Subdomain as Unmanaged Dante devices
- Use the same PTPv2 settings as Unmanaged AES67

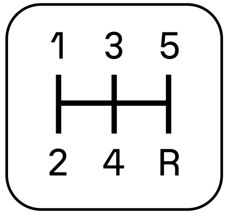
This can result in clocking interference between Managed and Unmanaged Devices

- In any case this will result in shared audio between Managed and Unmanaged

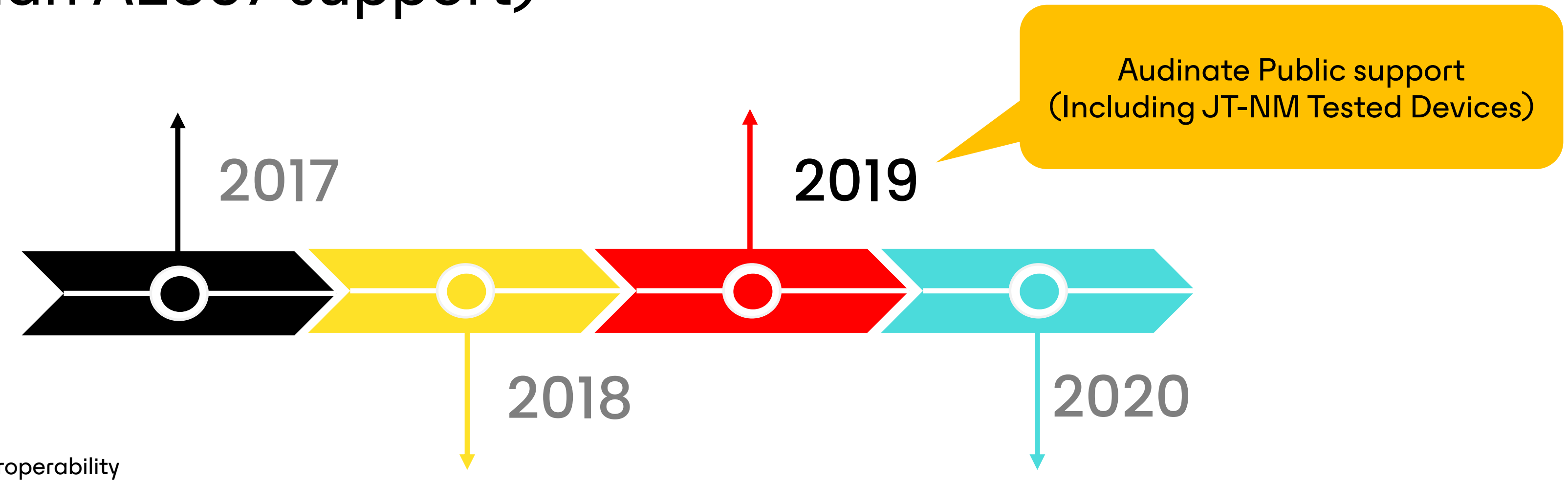


Dante ST2110 Audio

- Requires DDM which Centrally Adds:
- Consolidated Control for Whole Deployment
- Layer 3 Network Spanning, Access Control
- Device Monitoring and Logging
- Support needs to be enabled by the OEM (different setting than AES67 support)
- No NMOS support

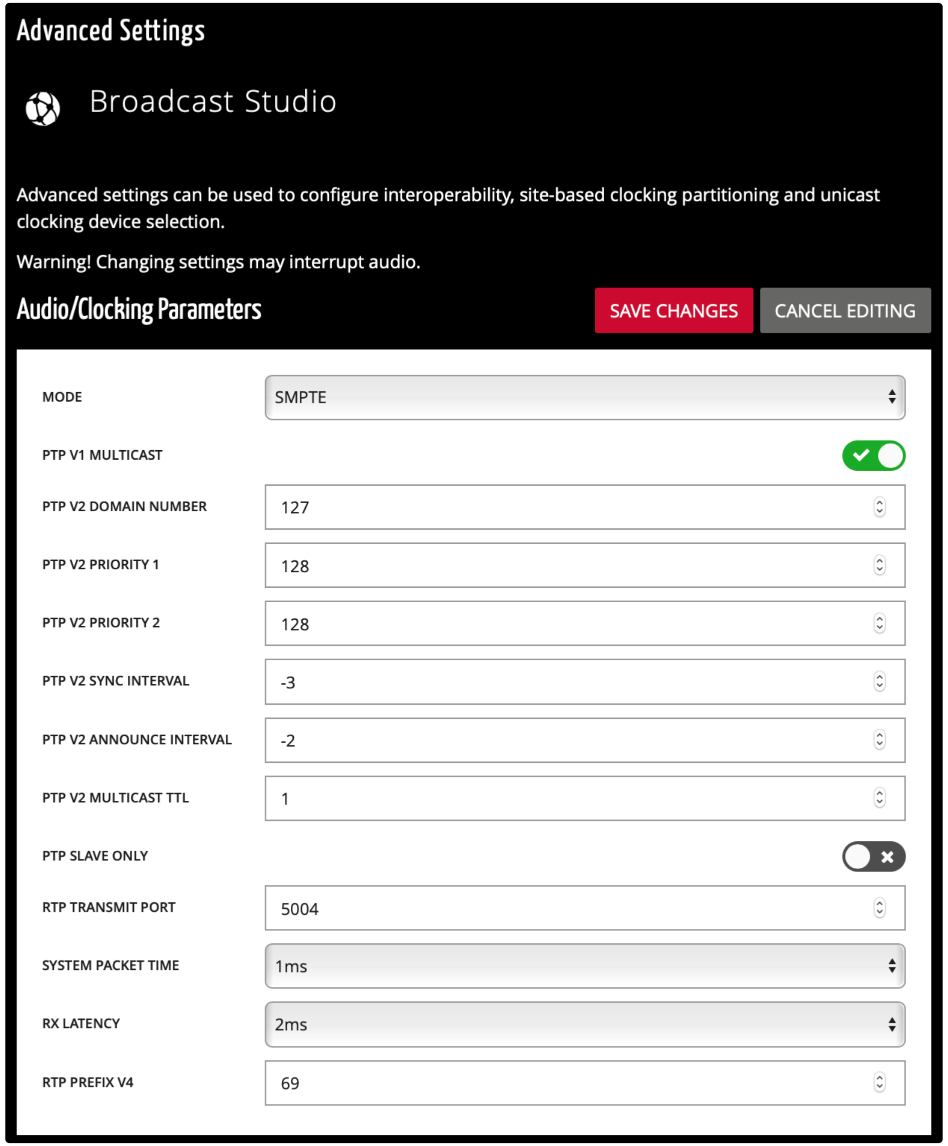


Dante with SMPTE ST 2110



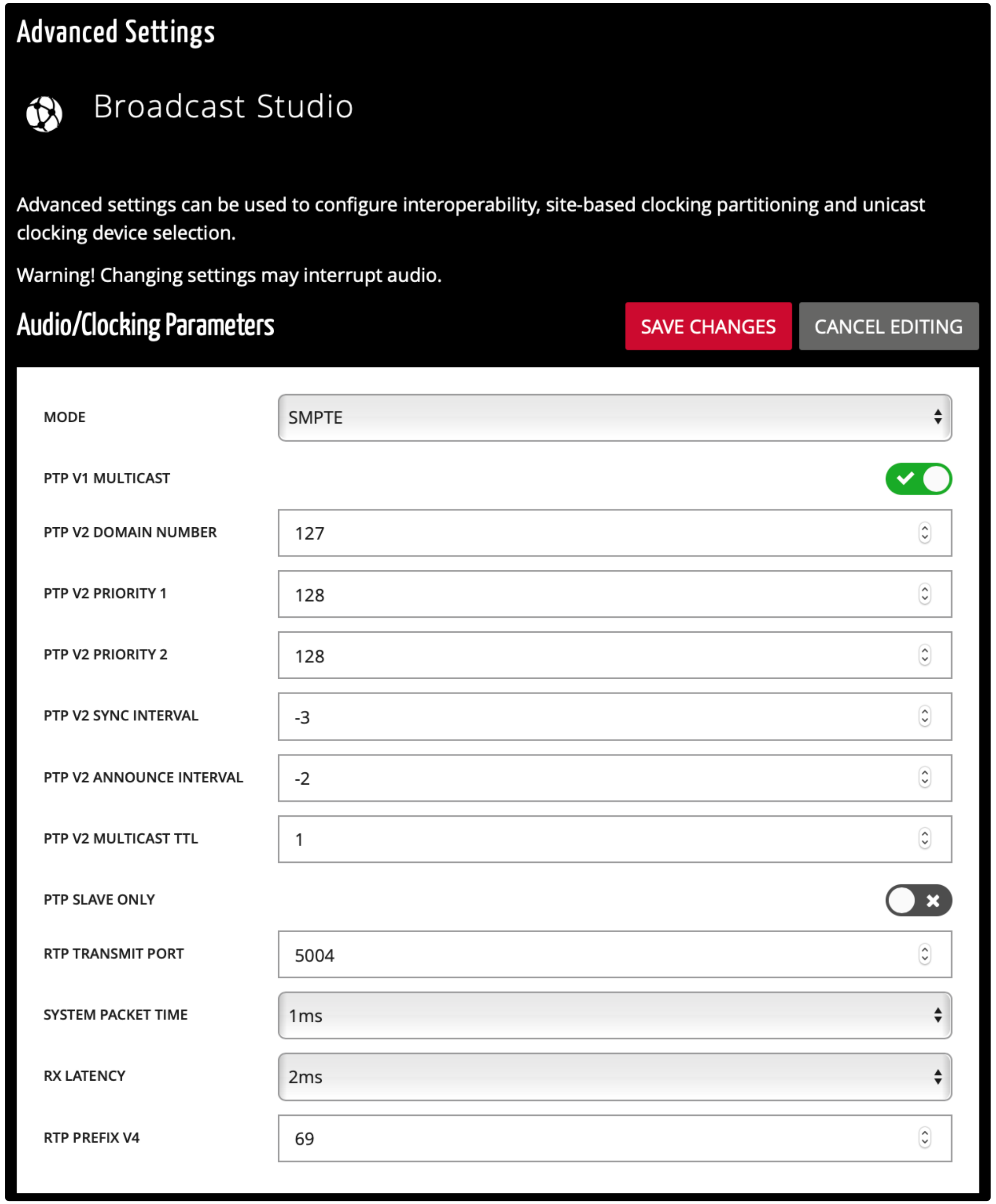
Managed ST2110

- Enable SMPTE mode at the Domain Level
- PTP related settings:
  - **PTP v1 Multicast**: Disabling PTP v1 multicast can prevent instability in non-Dante SMPTE devices
    - Requires that all Dante devices must be PTPv2 compatible.
  - **PTP v2 Domain Number**
  - **PTP v2 Priority 1 / 2**
  - **PTP v2 Sync / Announce Interval**
  - **PTP v2 Multicast TTL**
  - **PTP follower Only**: Devices in domain will not be elected as clock leader



Managed ST2110

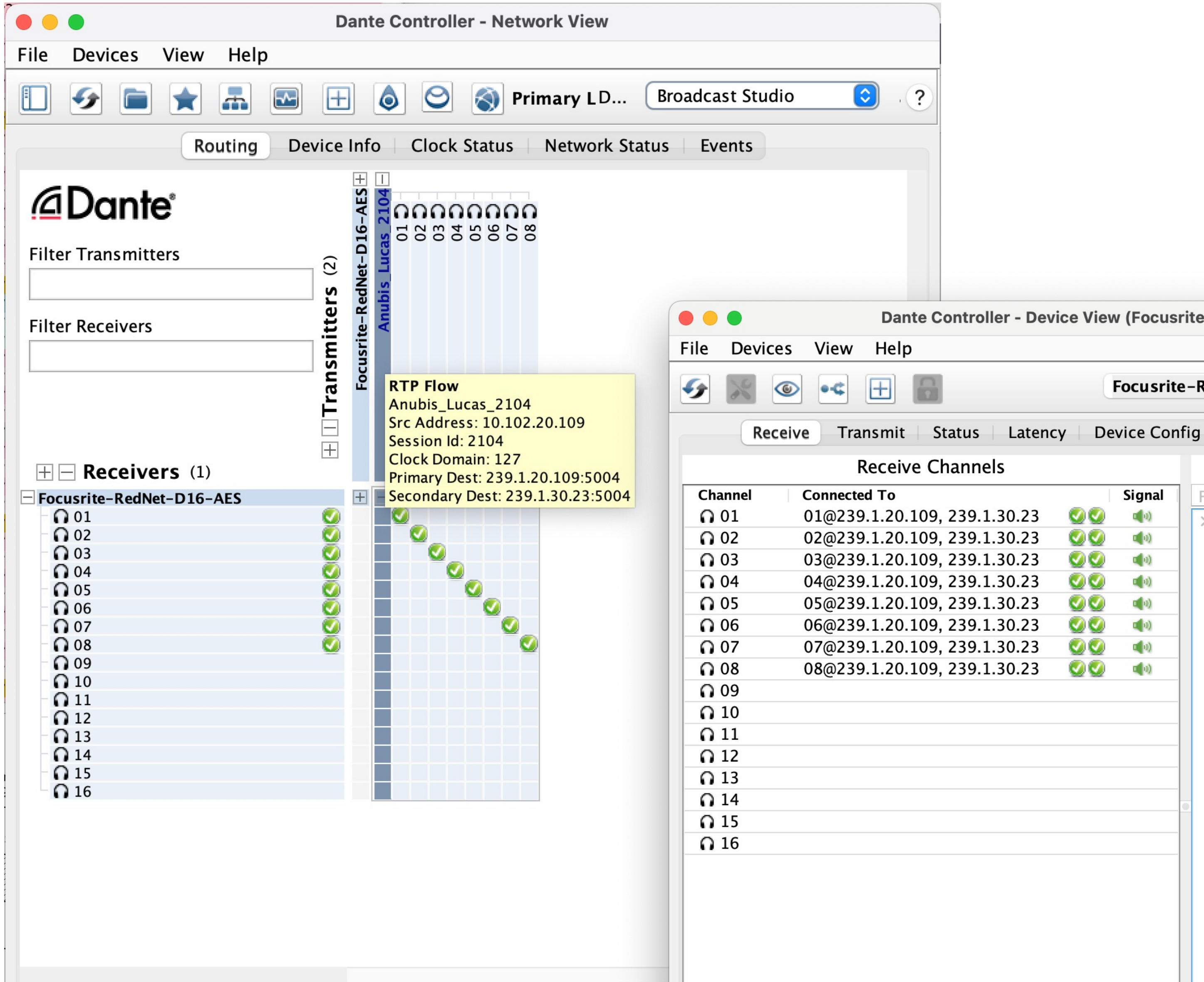
- RTP related settings:
  - **RTP Transport Port:** Transmit port number for RTP packets.
  - **System Packet Time:** Transmit time of the RTP stream expressed as the number of samples of each channel in one packet (1ms or 125us)
  - **Rx Latency:** Receive latency for SMPTE flows in the domain (2ms or 3ms)
  - **RTP Prefix v4:** The IP address prefix for RTP flows (second octet)



Managed ST2110: Subscribe to 3rd party

Dante

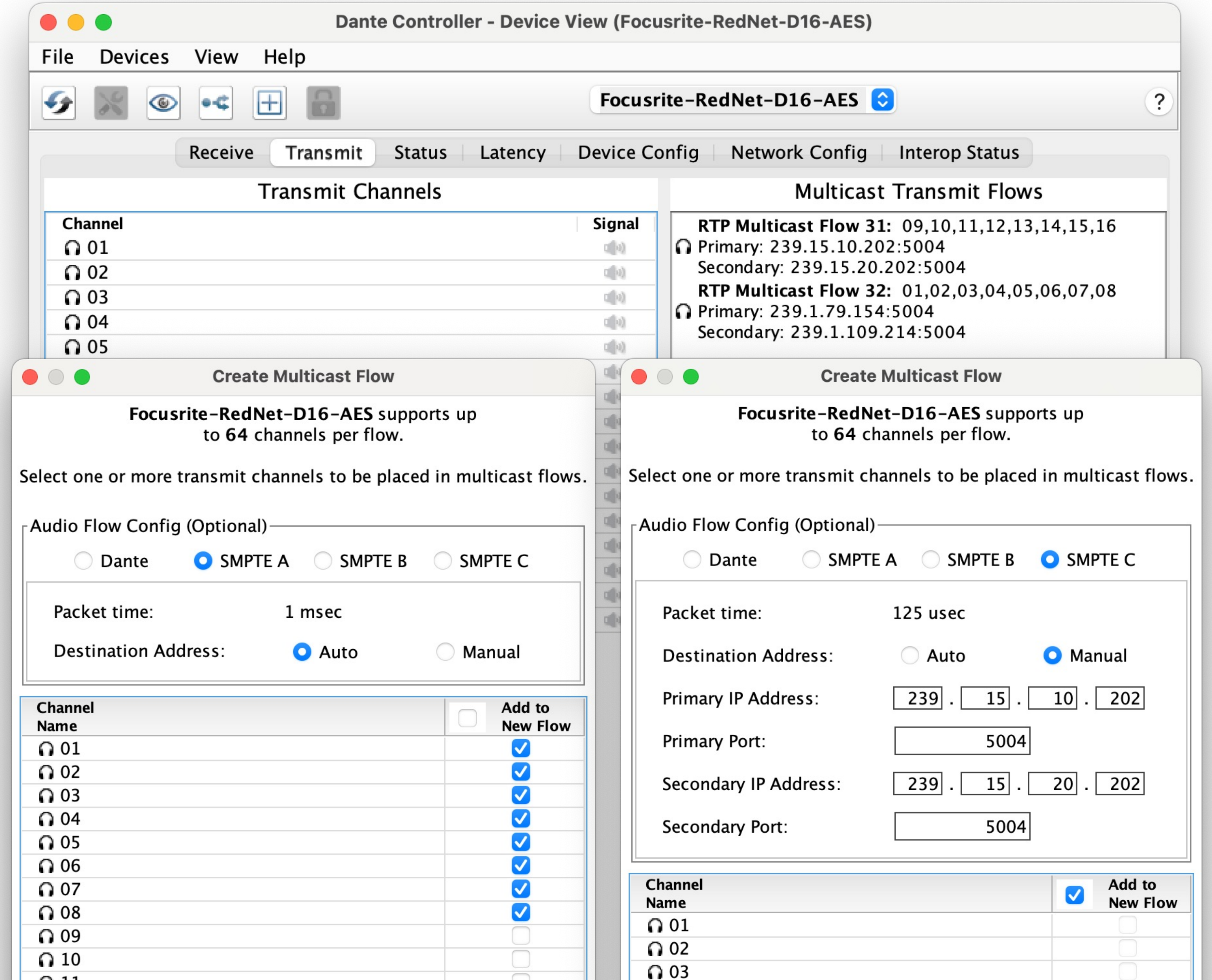
1. 3rd party devices flows will be shown in Blue in Dante Controller
    1. If the 3rd party device supports SAP (*and DC is in the same subnet*)
    2. If the SDP has been manually imported to DDM
2. Subscription is then possible
    1. Be sure RTP Prefix Match!
3. Flow redundancy (ST2022-7) will be used if available



Managed ST2110: Create Redundant Flow

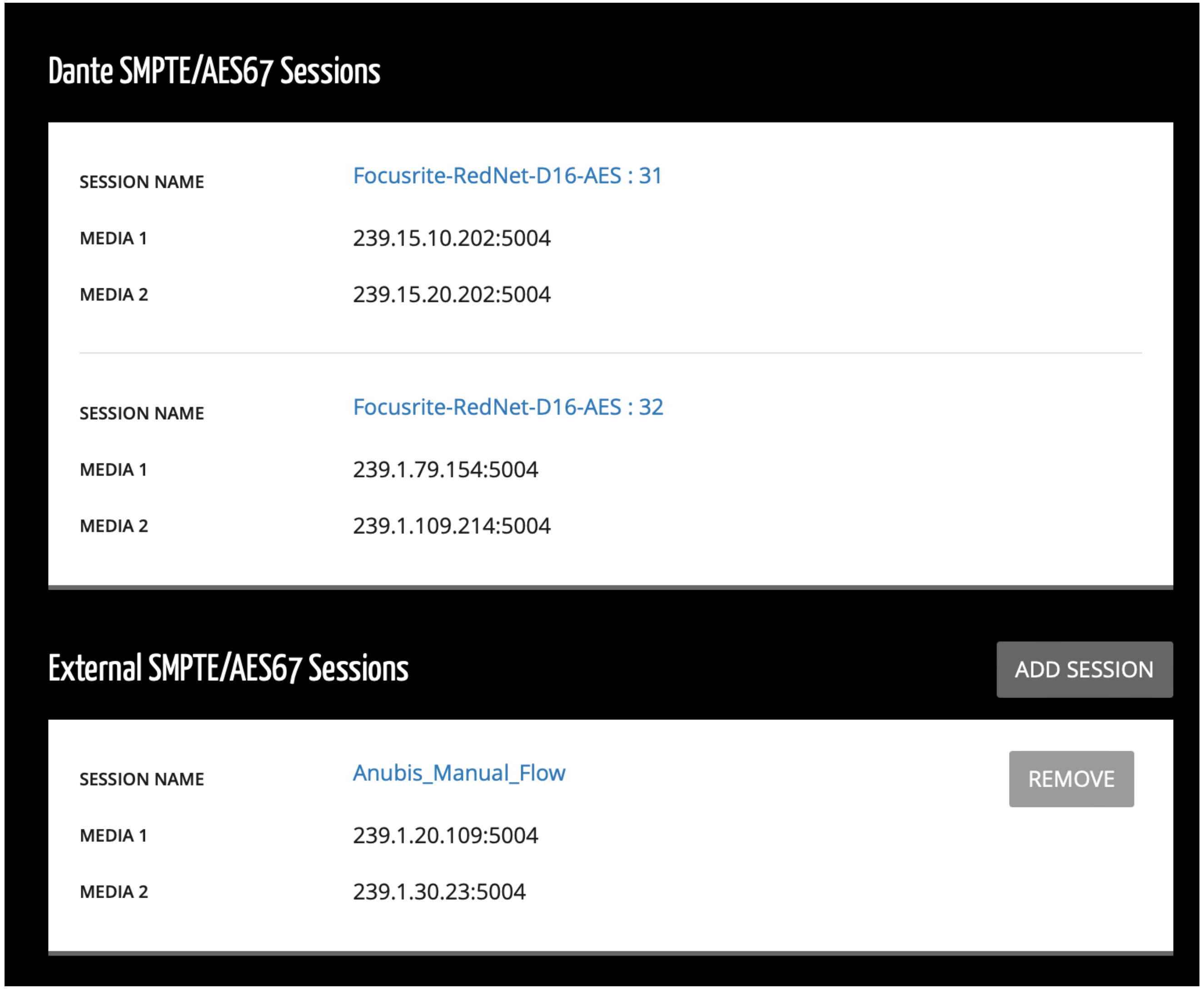
- SMPTE 2110 Multicast flows can be created from Dante Controller
  - With the desired Conformance Level
  - Using automatic destination addresses (based on the device RTP Prefix)
  - Manually specifying the Destination IP + Port

Conformance Level	Sample Rate	Audio Channels	Packet Times
SMPTE A	48 KHz	1~8	1 ms
SMPTE B	48 KHz	1~8	125 us
SMPTE C	48 KHz	1~64	125 us



Managed ST2110: Import/Export SDP

- Dante Controller will still show 3rd party flows using SAP
- Dante Devices will still announce their 2110 flows using SAP
- Dante Domain Manager will also allow to:
  - Import 3rd party SDP files manually
  - Export Dante SDP flows content

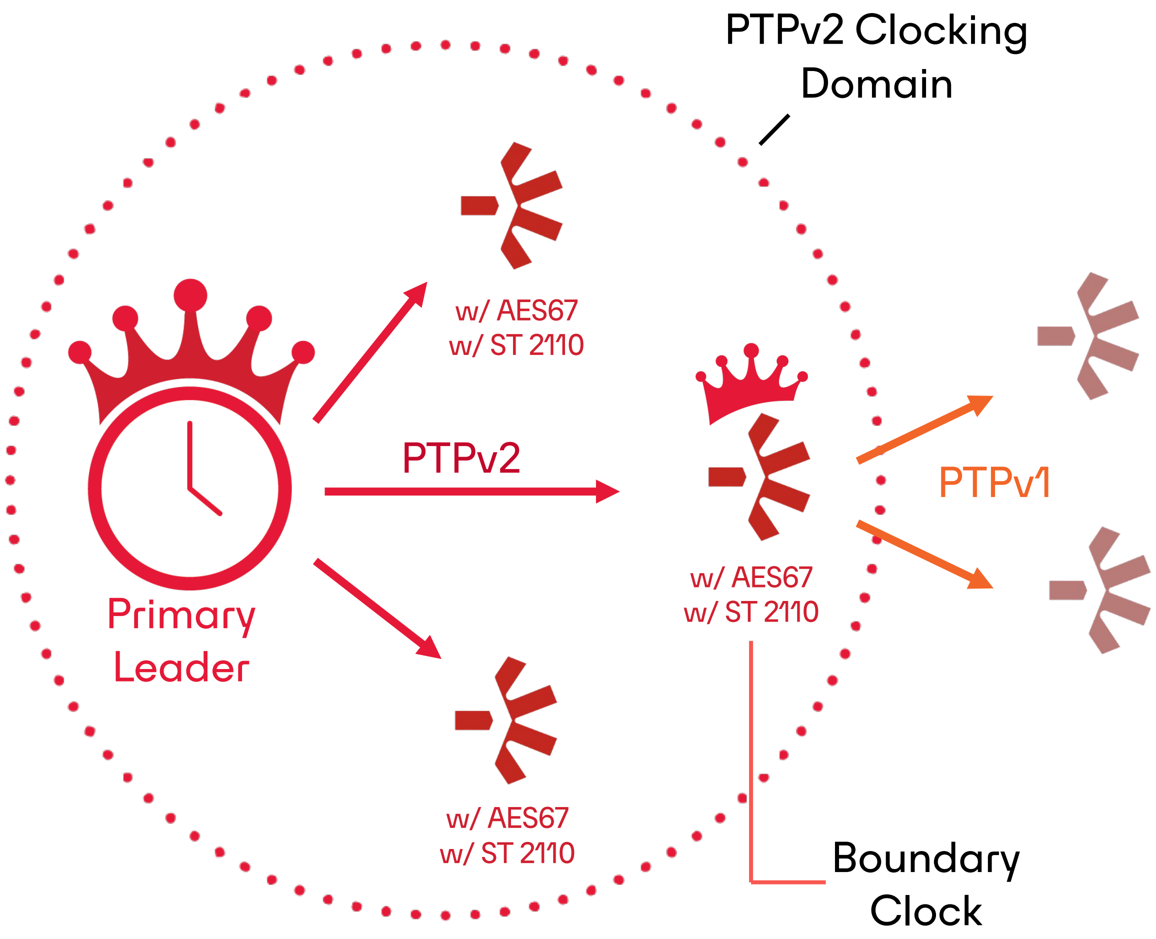


Precision Time Protocol (PTP): v1 & v2

Dante devices with AES67 or ST 2110 engaged can chase PTPv2 master clocks.

Options:

1. Prefer PTPv2 (BMCA)
2. PTPv2 Follower Only (ST 2110 Compliant)



Dante Interoperability PTPv2 options

	AES67	SMPTE
Domain	**0**	0 ~ **127**
Priority 1 & 2	Device Specific 0 ~ **248** ~ 244 *(Custom Managed AES67)*	0 ~ **128** ~ 255
Announce Interval	**0** (1 sec)	-3 ~ **-2** ~ 1 (0,125 ~ **0,25** ~ 2 sec)
Announce Timeout Interval	**3** (8 sec)	
Sync Interval	**-2** (0,25 sec)	-7 ~ **-3** ~ -1 (0,0078125 ~ **0,125** ~ 0,5 sec)
Delay-Req Interval	**0** (1 sec)	
Timestamp Mechanism	One or Two Step	
Delay Mechanism	End to End (E2E)	
Time To Live (TTL)	16	1 ~ **16** ~ 63

*Default values in **BOLD***

QoS DSCP Queues

Audinate's markings when sending

AES67 & SMPTE ST 2110

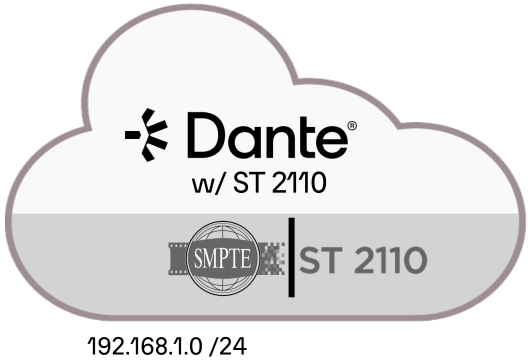
QoS Priority	Data Type	Recommended DSCP Values
High	PTPv2	46 (EF)
Medium	RTP Media	34 (AF41)

Audinate implements AES67-recommended DSCP values
for AES67 and ST 2110 clocking packets and media streams as of v4.2 in 2019.

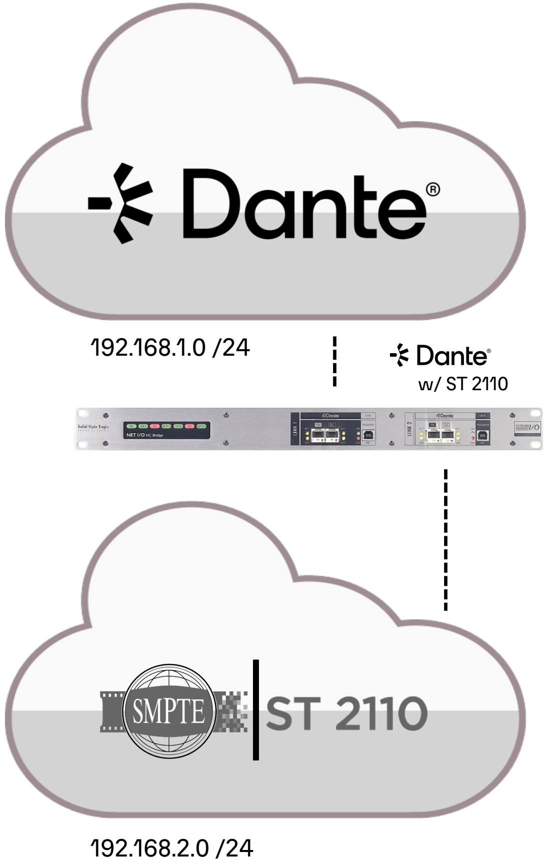
Design Topology

Topology Options

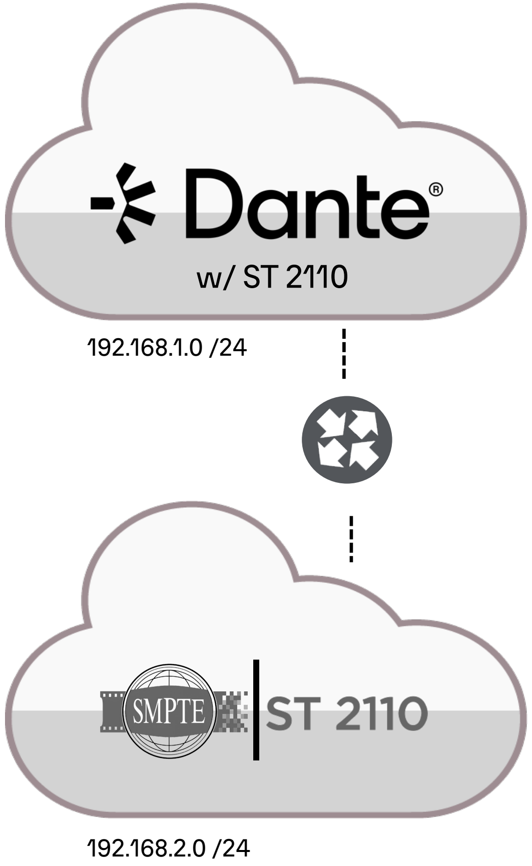
Option 1: **Complete Coexistence**



Option 2: **Bridge Device**

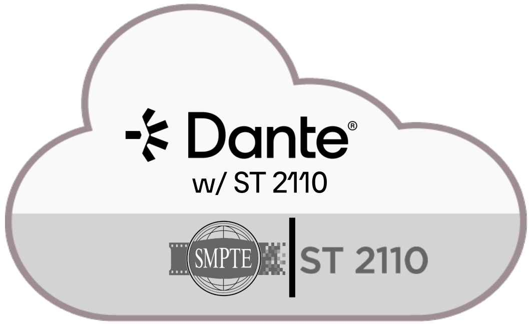


Option 3: **Multicast Routing (PIM)**



Topology Options: Coexistence

Option 1:
Complete Coexistence

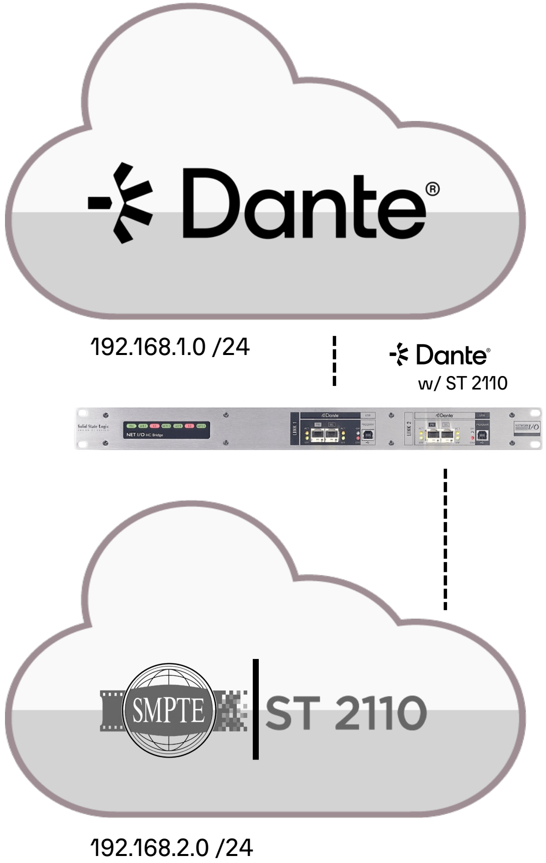


192.168.1.0 /24

- Basic Principles:
  - Dante devices will be managed by Dante Domain Manager
- Special notes if QoS is engaged
  - PTPv1 uses DSCP 56 (CS7) - Higher than PTPv2
  - PTPv1 Multicast can be Disabled in DDM
    - Requires all Dante devices to support ST 2110

Topology Options: Bridge Device

Option 2:
Bridge Device



- Basic Principles:
  - Dante Domain Manager only required for the Dante devices in ST 2110 mode
  - Separate VLANs simplifies separate Dante traffic optimization
- Special notes:
  - Default Dante and PTPv1 are completely isolated from the ST2110 Network
  - Bridge device can manage sample rate conversion if production and network sample rates are different.

192.168.2.0 /24

Topology Options: Bridge Device

Dante

Dante
w/ ST 2110

Pri / Sec
Slot #1

Pri/ Sec
Slot #2

Dante

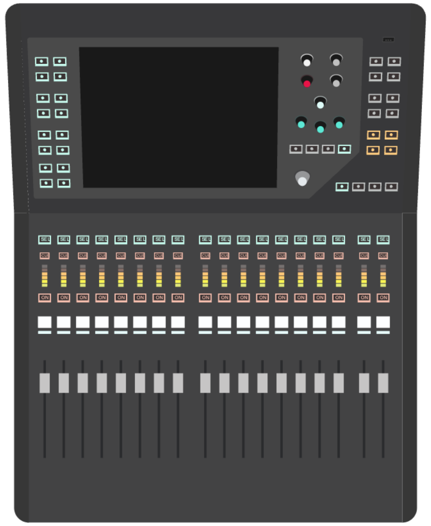
192.168.1.0 /24

SMPTE ST 2110

192.168.2.0 /24

You don't necessarily need to have a dedicated bridge device on the network
You can also do it with a single device (like a mixer).

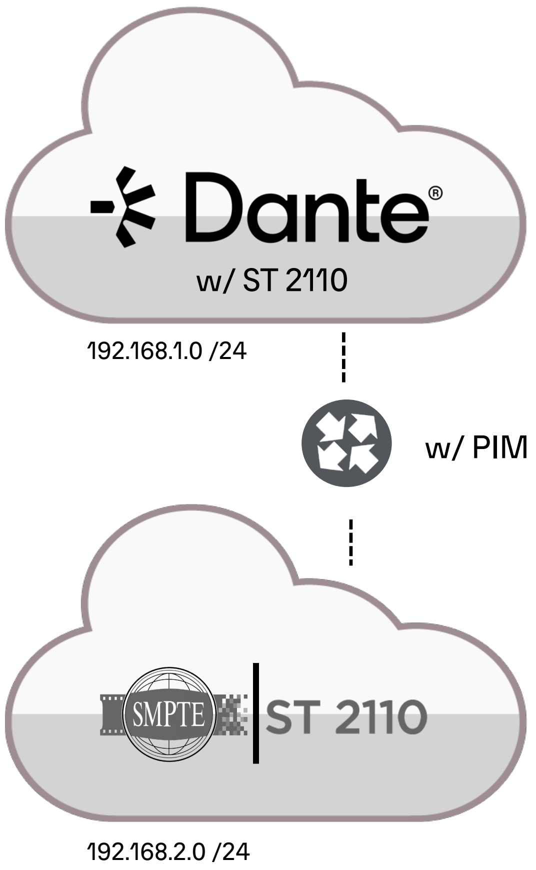
Dante
Slot #1



Dante
w/ ST 2110
Slot #2

Topology Options: Multicast Routing

Option 3:
Multicast Routing (PIM)



- Basic Principles:
    - Dante devices will be managed by Dante Domain Manager
    - Network infrastructure needs to be PIM capable
        - To be implemented by a network specialist
- Special notes:
    - RTP Multicast flows from/to Dante/ST2110 devices need to be declared on the Network
    - Dante does NOT support Source Specific Multicast (SSM)
    - PIM allows range forwarding. Ex: 239.**69**.xx.yy/16
        - Need to be configured in DDM

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“Audio and video quality is exceptional, and the flexibility to route audio and video around the facility on just a few cables was very liberating”.

Dennis Cham, Founding Partner and Chief Technology Officer of HIT Productions