

Interfacing Comrex NX Rack and MultiRack with Dante AoIP Systems

Dante is an Audio-over-IP (AoIP) system developed by Audinate. Dante is used in many studio and live audio environments worldwide. While the Dante protocol is proprietary, it also supports a mode that allows inter-operation with devices that support the AES67 AoIP standard available in Comrex NX Rack and MultiRack audio codecs. This document provides a guide on how to set up a Comrex and Dante system to talk to each other.

Unlike most AES67 compatible products, Comrex codecs utilize both transmit and receive AoIP drivers. Each of these operate independently and must be set up separately.

DANTE CONTROLLER

Dante settings are configured using the Dante Controller software. Dante network users should be familiar with this software, and it is required to establish routing between devices. To support AES67, the Dante Controller software used must be v3.6.2.4 or later.

MORE ABOUT DANTE

To inter-work with Dante devices, the device that will send audio to the Comrex codec and the device to receive audio from the Comrex codec must have firmware to support AES67 (not all do). Check the manual and the firmware level of your devices to make sure AES67 is supported. If your device does not support AES67, these settings will not appear when your device is configured using Dante Controller.

Dante supports Unicast and Multicast delivery of streams. *Comrex codecs support only Multicast operation, so this is the only mode possible when inter-working with Dante.*

CLOCKING

All AoIP systems require a network clock to synchronize audio within their systems. Generally, one device on the network will function as the Master clock, and all others operate as Slaves to it.

Many Dante capable products (but not all) incorporate the ability to deliver a Master clock. Selection of the device to deliver Master clock is normally done automatically within the network, but there are settings available within the Dante Controller software to favor one device over others.

Dante natively delivers a Master clock that is not compatible with AES67. By enabling AES67 mode, you are instructing your Dante system to deliver both the Dante Master clock (PTPv1) and an AES67 Master clock (PTPv2).

DISCOVERY

AES67 does not define a method of advertising and discovering AoIP streams (Dante calls these “flows”). This is the way AoIP streams are found and routed within the network. Dante uses SAP for this function, which is also supported by Comrex codecs.

Unlike some AoIP systems, Dante requires the use of SAP discovery and doesn't support a way to manually define streams entering the Dante system.

SETTING UP A DANTE DEVICE FOR AES67

For the purposes of this guide, we've connected to a Focusrite Rednet 8AR Dante analog node.

Open Dante Controller and select the device you'd like to send and/or receive AES67 streams. If your device supports AES67, a tab with that label will appear. Select it.

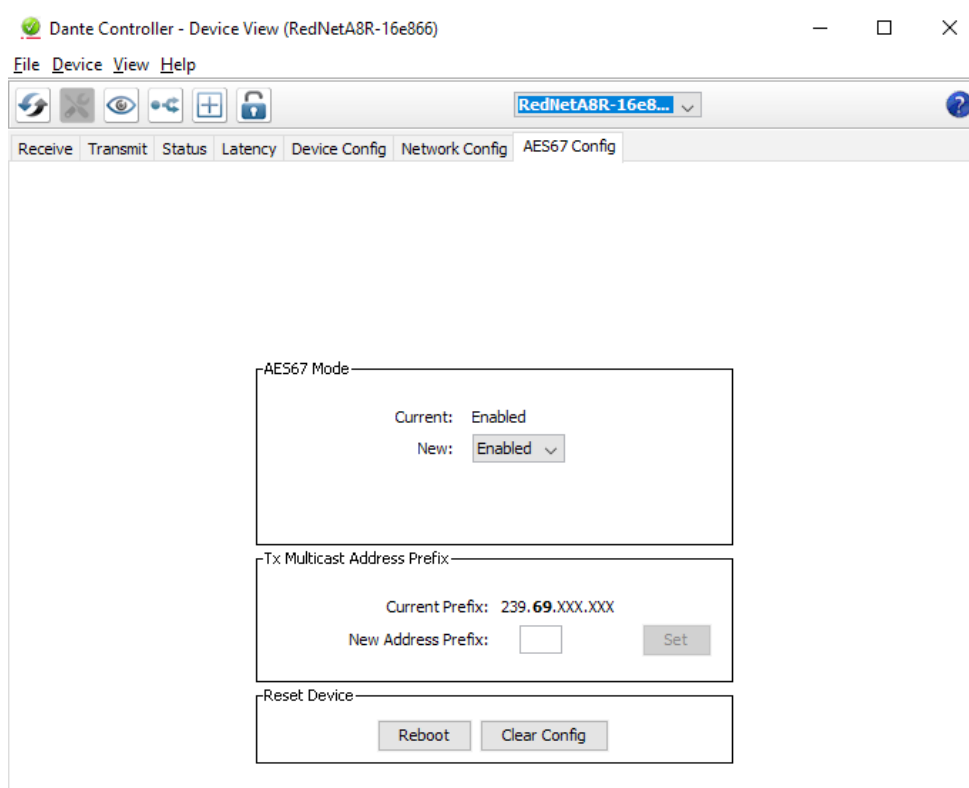


FIGURE 1 DANTE AES67 DEVICE VIEW TAB

Select "Clear Config", as some previous settings in the Dante device may interfere with AES67.

If you will be sending audio to the Comrex codec from this device, make sure the Tx Multicast Address prefix is the default 69 (The second decimal digit bolded). If not, set it to 69.

Finally, set the AES67 mode to "enabled". Then select "Reboot".

Go back to the Network View and make sure this device (or some device on your network) shows up as "Grand Master Clock" on the top banner.

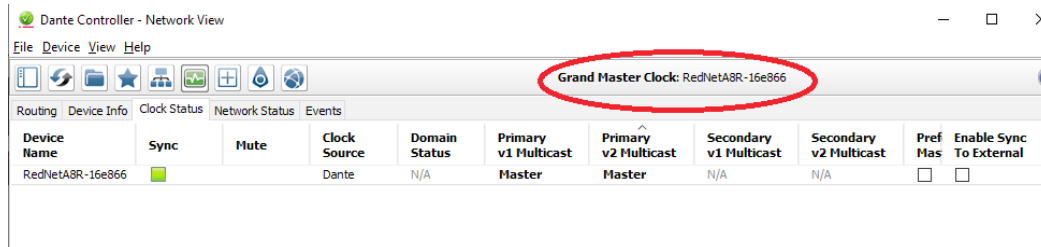


FIGURE 2 NETWORK VIEW-CLOCK STATUS TAB

Next you will Open the Comrex Codec Web-based user interface page, and navigate to the Network Manger section. You must be sure the Ethernet interface used (typically the secondary) has a valid IP address on your AES67 network.

The default settings of these ports are to use a DHCP server. If you require static addressing (or you have no DHCP server on your network) you must apply static settings here; using a reserved IP address on your AES67 LAN. **Note:** **When configuring the AES67 port for static operation, the DNS server field must not remain blank. A DNS server address (such as 0.0.0.0) must be entered here, although it does not need to be a valid address.**

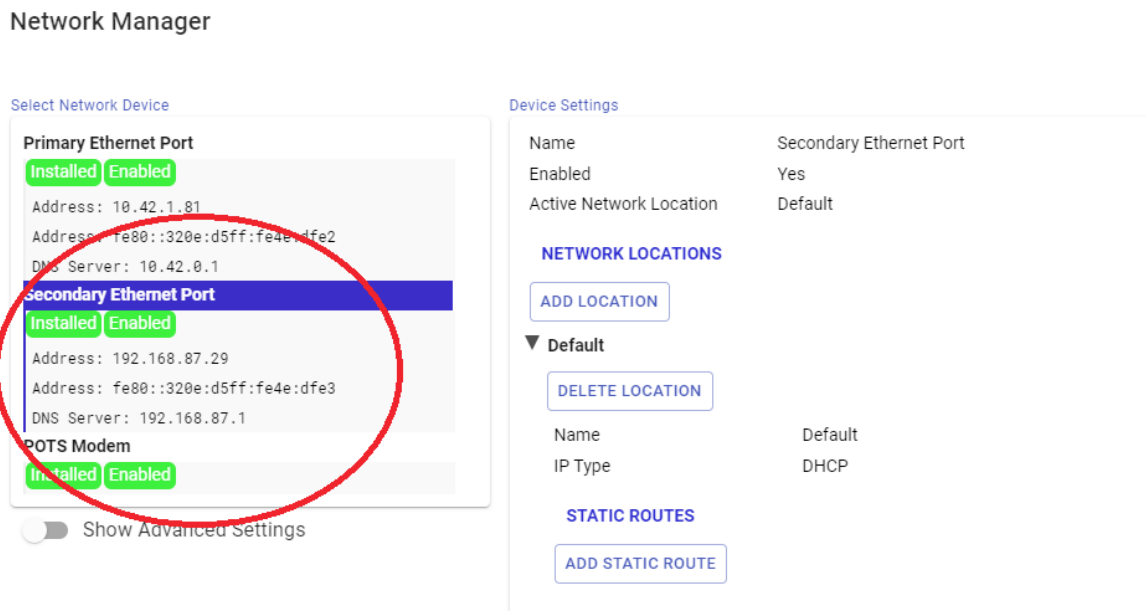


FIGURE 3 ETHERNET PORT SETTINGS

You now have an AES67 compatible Master clock on your Dante network. Continue with the Web-based control page on your Comrex codec and select the “System Settings” Tab. If not done already, enable AES67 operation under that section, as described in the manual. Leave all other AES67 settings at their defaults, as shown. Look for the PTP clocking status to show “locked”.

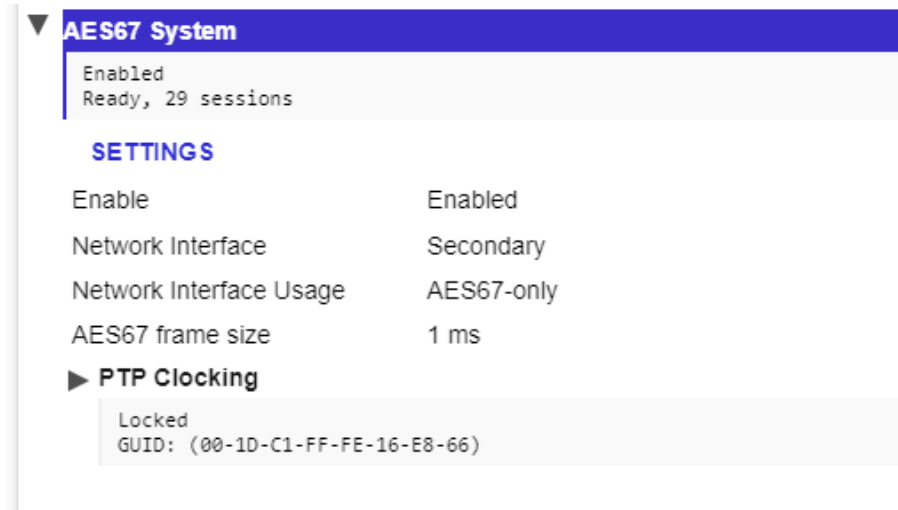


FIGURE 4 AES67 SYSTEM SETTINGS

Dante offers a PTPv2 Master clock domain of 0. This is the default of Comrex codecs. If not locking, check that the clock domain is set to 0 under the PTP clocking options on the Comrex codec.

CREATING AN OUTGOING AES67 STREAM FROM A COMREX CODEC

Once AES67 is enabled on the codec and the clock is “locked”, an AES67 outgoing stream is automatically created to the default Multicast node address. It carries a label taken from the Unit Name setting under the System Settings -> Connections section of the user interface.

You’ll need to change the outgoing Multicast node address to one within the Dante accepted range. Still under AES67 settings, navigate to the “Outputs” section and select the outgoing stream you wish to change. (NX Rack has only one output, MultiRack has five).

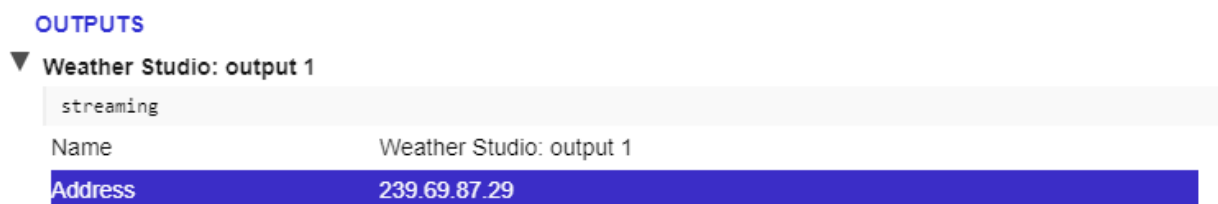


FIGURE 5 OUTPUTS

Set the “Address” field to an address within the 239.69.x.x range. Click “apply” then disable and re-enable AES67 mode on the codec to initiate the stream to the new address.

Note: the default outgoing address is likely not in the Dante Multicast range. You must set this field correctly.

The stream should now be discoverable and routable within the Dante controller. Comrex AES67 streams will always appear as 48KHz sampled 24-bit two channel streams. Codec connections that are received as mono will deliver identical audio to both AES67 channels.

If your Dante device does not receive the Comrex stream reliably, you may need to increase the latency setting of the receiving device under Dante Controller. In the Device View of the receiving device, choose the Latency Tab. The 2mS setting is recommended for most networks, but some devices may require an increase to 5mS for stability.

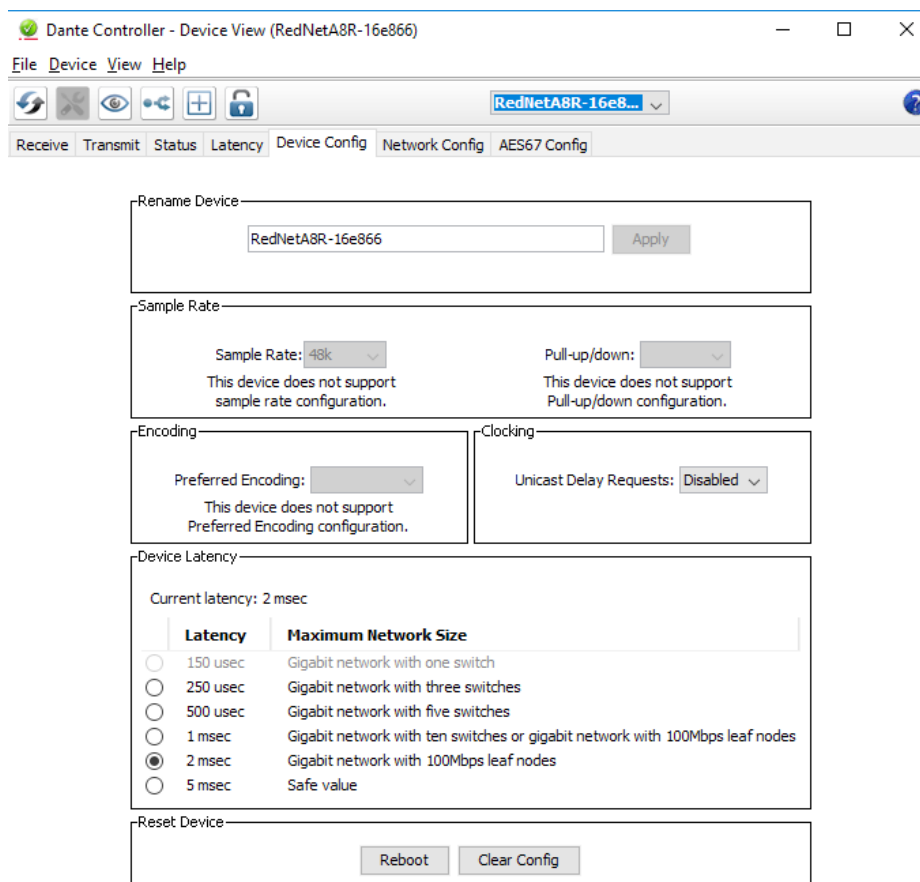


FIGURE 6 DEVICE VIEW-DEVICE CONFIG TAB

RECEIVING AN INCOMING AES67 STREAM TO A COMREX CODEC

Unlike codec outgoing streams, incoming AES67 streams cannot be routed via Dante Controller. They will be created and configured on the Dante device via the Dante Controller Device view, but they will be selected on the Comrex Codec’s configuration page.

On the Device View from the Dante device that will create the stream to the codec, choose the “Create New Multicast Flow” option from the top menu.

Channel Name	<input type="checkbox"/> Add to New Flow
01	<input type="checkbox"/>
02	<input type="checkbox"/>
03	<input type="checkbox"/>
04	<input type="checkbox"/>
05	<input type="checkbox"/>
06	<input type="checkbox"/>

FIGURE 7 AES67 MULTICAST FLOW

First select the “AES67 Flow” option before selecting any channels. Leave the Destination Address as “Auto”. Choose two channels from this device to be added to the outgoing stream. Comrex codecs can only accept two-channel AES67 streams. Choose “Create”.

Log into the Comrex web-based interface. Under the System Settings Tab, expand the AES67 options, and find the input you wish to assign (NX Rack has one, MultiRack has five). Find and select the “Source” entry under your desired input.

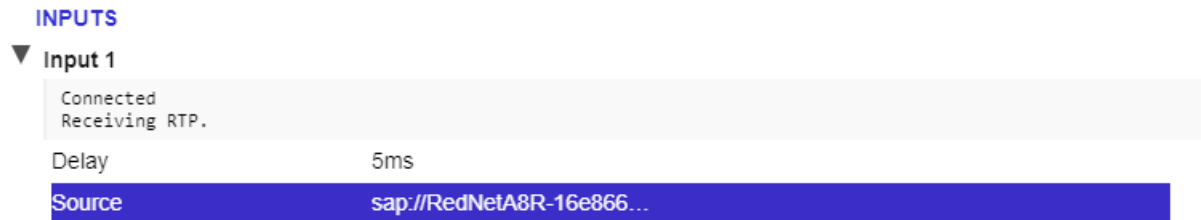


FIGURE 8 INPUTS

On the right side of the page, a pull-down menu will appear. This will display all detected streams on your network. Note that Dante (non-AES67) streams may be listed, but will not be received correctly. Choose the name of the AES67 stream you've created in the previous step, and choose "Apply Settings". The codec will now be locked to that incoming stream, and the status message under the selected input should read "Receiving RTP".

If the status does not read "Receiving RTP", or if the incoming stream has distortion or drop-outs, you may need to adjust the "delay" setting under the affected input. This adjusts the amount of buffer used by the AES67 receiver. The default is 5ms, and you can increase or decrease it to fixed values in the pull-down menu.