

Cisco Collaboration Devices – Audio Interoperability over AES67

Setup guide

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Revision history

Version	Date	Comment
D15516.01	Sept 4, 2023	
D15516.02	April 19, 2024	Added path for setting the gain on Codec Pro, in the examples.
D15516.03	April 25, 2024	Codec Pro based devices now support 8 input streams
D15516.04	July 8, 2024	Support added for Board Pro and Board Pro G2
D15516.05	March 4, 2025	Corrected instances of AEC_REFERENCE Removed references to Developer API on pages 16 and 23

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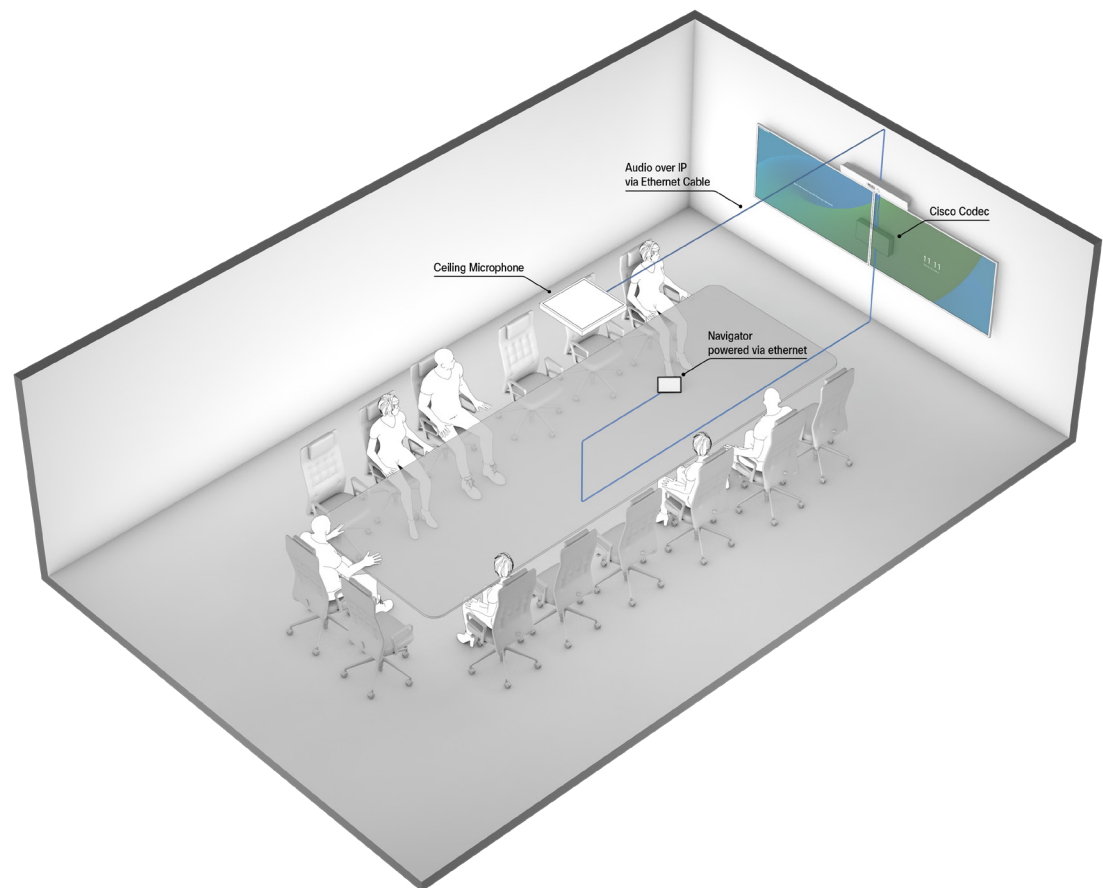
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Overview

This document is a setup guide covering how to connect third-party microphones and loudspeakers to Cisco collaboration devices using Ethernet connectivity. To enable Ethernet for audio transport, the supported Cisco devices implement the AES67 standard, developed by the Audio Engineering Society. It is designed to provide interoperability between previously competing networked audio systems. Among adopters are Dante¹ and Ravenna².

The current solution for connecting networked audio to Cisco collaboration devices is using a separate device as a bridge to either a USB or analog connection. Using Ethernet connectivity directly simplifies cabling and can remove the need for extra equipment such as external DSPs or converters.

A typical use case is a ceiling microphone installed in a medium to large meeting room. If the microphone supports audio transport using AES67, it can now be connected directly to the Cisco device.



¹ https://dev.audinate.com/GA/dante-controller/userguide/webhelp/content/aes67_config.htm

² <https://www.ravenna-network.com/aes67-practical-guide/>

Scope

This guide covers how to connect third-party audio peripherals that support the AES67 standard and how to subscribe to and set up AES67 streams on a Cisco collaboration device.

Except for the examples, it does not cover how to set up the peripherals other than a high-level description of the necessary steps, as this will vary between vendors and implementations. Please refer to the documentation from the vendor for details on how to execute these steps.

Supported devices

- Cisco Codec EQ (with the AVIntegrator option key)
- Cisco Room Bar Pro (with the AVIntegrator option key)
- Cisco Codec Pro³
- Cisco Board Pro (with the AVIntegrator option key)
- Cisco Board Pro G2 (with the AVIntegrator option key)

Device	Supported number of streams	
	Input	Output ⁴
Codec EQ	8	4
Room Bar Pro	8	4
Codec Pro	8	4
Board Pro	8	4
Board Pro G2	8	4

Input streams support up to 8 channels. If echo cancellation is performed on the Cisco device, there is a limit on 8 channels total.

Output streams support up to 2 channels.

Codec EQ, Codec Pro, and Room Bar Pro:

- The minimum required software version is RoomOS 11.8.

Board Pro and Board Pro G2:

- The minimum required software version is RoomOS 11.18.

³ Due to a hardware limitation, Cisco Codec Pro does not support the role of PTP timeReceiver. Thus, if a Codec Pro is to be used, it must be the PTP timeTransmitter on the network.

⁴ The first output stream is reserved for Cisco peripherals for the time being. For devices with a Cisco DNAM IV amplifier, the second stream is also reserved.

Network setup

The collaboration device supports AES67 only on its local network. This is both a security measure and to ensure reliability. A limitation of this approach is that physical access to the device is required to connect to the local network.

Another limitation is that the Cisco collaboration device will not provision IP addresses for arbitrary devices on its local network. For this reason, all peripherals on this network needs a static IP address configured before connection. The available addresses are 169.254.1.[225-254].

If needed, the number of local network ports of the Cisco device can be extended with a network switch. Cisco recommends the use of CBS350, CBS250, and C1200 switch series as these will be recognized and assigned an IP address by the Cisco device. Other switches may be used, but need to be assigned a static address and should support the following features:

- Quality of Service (QoS)
- Possibility to disable 802.3 Energy Efficiency Ethernet (EEE)
- PoE capability if required by your peripherals

Some peripherals, including most Dante devices, have two (or more) network interfaces where one is reserved for media and one for management and control. In that case, the two interfaces may be connected to different (V)LANs to allow for remote management and control, while keeping the media traffic on the local network. The practical and security implications of this approach are up to the installer to evaluate.

Setup of peripherals

Note: A static IP address must be assigned before connecting the peripheral to a collaboration device.

Transmitting peripheral setup

This section covers the required configuration of an AES67 enabled peripheral that transmits audio, such as a microphone.

Setting up a stream

Make sure that your peripheral is set up for AES67. For Dante devices this is done in the device menu in the Dante Controller. You also need to set up an AES67 stream with the desired channels.

To identify the streams from a peripheral, they should be named. Many peripherals allow for naming the unit itself and will name its streams thereafter. This name will be announced in the SDP and will be available to the Cisco device upon discovery.

Static IP addresses

Give the peripheral a static IP address in the range 169.254.1.[225-254]. The gateway address is 169.254.1.1, and the subnet mask should be 255.255.0.0. If the peripheral has multiple network interfaces, all interfaces that will be connected to the Cisco device's local network needs a separate static IP address.

Receiving peripheral setup

A static IP address is also required for receiving peripherals. Other than that, follow the steps below to set up an outgoing stream from the collaboration device. Then follow the peripheral vendor's instructions on how to subscribe to an AES67 stream.

Setup of Cisco collaboration device

At this point the peripheral may be connected to the Cisco device.

Terminology

Ethernet connector: This should not be confused with the physical Ethernet ports on the Cisco device or switch. The Cisco device supports a given number of incoming and outgoing audio streams. In the API these have pre-allocated slots which are dubbed “connectors”. They have a type, direction, and index. For multichannel streams they also have sub-indices. For AES67 streams, the type is “ethernet”, and the index can be assigned using the *Register* commands described below.

Allowing unencrypted media traffic

The video device only allows encrypted media by default. This is not supported by the AES67 standard, so unencrypted audio streams must be explicitly allowed. To do this, set:

xConfiguration Audio Ethernet Encryption: Optional

Discovering announced streams

Cisco collaboration supports discovery using SAP. To discover streams announced by peripherals, set:

xConfiguration Audio Ethernet SAPDiscovery Mode: On

The default address is 239.255.255.255 as 239.xxx.xxx.xxx is the range used for multicast in the AES67 standard. If your device announces on a different address, it can be changed by setting:

xConfiguration Audio Ethernet SAPDiscovery Address

Any discovered streams will now be enumerated and are visible in:

xStatus Audio Input Ethernet DiscoveredStream

It will then report the discovered streams like this:

```
*s Audio Input Ethernet DiscoveredStream <DiscoveryIndex> Channels: <channels>
*s Audio Input Ethernet DiscoveredStream <DiscoveryIndex> MediaIP: "239.xxx.xxx.xxx"
*s Audio Input Ethernet DiscoveredStream <DiscoveryIndex> Name: <Stream name>
*s Audio Input Ethernet DiscoveredStream <DiscoveryIndex> OriginIP: "169.254.1.2xx"
*s Audio Input Ethernet DiscoveredStream <DiscoveryIndex> Status: "Valid offer"
```

Connecting to a discovered stream

To receive audio from a discovered stream it needs to be assigned to an available Ethernet connector. The current input connector usage can be inspected by:

`xStatus Audio Input Connectors Ethernet`

The assignment can be done with the command:

`xCommand Audio LocalInput Ethernet Register`

with parameters `ConnectorId` and `StreamName`:

`ConnectorId` is the index of the Ethernet connector you want to assign the incoming stream to (i.e. Ethernet 1). This argument is optional and if not supplied, the first available connector will be used.

`StreamName` is the name of the stream as reported by `xStatus Audio Input Ethernet DiscoveredStream` earlier.

If needed, this assignment can be undone using:

`xCommand Audio LocalInput Ethernet Deregister`

with parameters `ConnectorId`, `StreamName`, and `ResetConfig`:

`ConnectorId` is the index of the Ethernet connector you want to release.

`StreamName` is the name of the stream.

`ResetConfig` takes `True` or `False` as arguments. If `True`, all configurations for this connector are reset. This is an optional parameter and defaults to `False`.

Note: Only `ConnectorId` or `StreamName` is required.

Configuration of input processing

Some peripherals have DSP capabilities built in, which can overlap or conflict with the Cisco device's processing. You need to decide which to use. Usually, the peripheral's processing is recommended as it has been tailored to the use case the peripheral was meant for.

Two common processing steps to configure are gain and echo cancellation.

Gain

Depending on the output level of the connected peripheral, the input gain of the Cisco device may need to be adjusted. The target should be an average input level of -26dBFS for speech. This can be checked using the "Microphone check" on the Cisco device where -26dBFS translates to a VU meter reading just above the middle of the scale. The gain can be adjusted by setting

`xConfiguration Audio Input Ethernet <Connector Index> Gain`

If using a Codec Pro, this configuration is named

xConfiguration Audio Input Ethernet <Connector Index> Level

Echo cancellation

If your peripheral provides echo cancellation, it is recommended to use that as it could be required for other features, such as automatic speech tracking. If you want to use the peripheral's echo cancellation, echo cancellation in the Cisco device must be turned off. This is achieved by setting

xConfiguration Audio Input Ethernet <Connector Index> EchoControl Mode

Set up an outgoing stream

To start transmission of audio over AES67 from the video device, an outgoing audio stream must be configured and assigned to an output connector using

xCommand Audio LocalOutput Ethernet Register

This command takes the following parameters:

StreamName is the name of the stream which will be visible to other AES67 devices on the network.

Channels is the number of channels to transmit in the stream. Either one or two channels is supported.

ConnectorId is the index of Ethernet output connector to use. This parameter is optional. If it is not present, the first available connector is used.

Note: Ethernet 1 is a reserved connector. On devices with a Cisco DNAM IV amplifier, Ethernet 2 is also reserved.

MediaIp is the address to use for the audio transmission. This should be a valid multicast address.

Note: Some devices have a specified range of multicast addresses they consider valid. For example, at the time of writing, Dante devices require addresses in the range 239.69.xxx.xxx

If needed, this assignment can be removed using:

xCommand Audio LocalOutput Ethernet Deregister

The stream is then announced using SAP every 30 seconds on the address set in:

xConfiguration Audio Ethernet SAPDiscovery Address

Currently active streams and their parameters are listed in:

xStatus Audio Output Connectors Ethernet

Troubleshooting

General

A more complex network architecture has more potential failure points. If experiencing issues in setups using a switch, try connecting the peripheral directly and see if the problem persists. In setups with more than one peripheral, try connecting them one by one and see when the problem occurs.

If the peripheral was connected to the Cisco device before setting a static IP address

Disconnect the peripheral from the Cisco device, reboot the Cisco device, assign a static IP address to the peripheral and reconnect.

If a peripheral stream does not show up on the Cisco device

If the streams from a transmitting peripheral does not show in `xStatus Audio Input Ethernet DiscoveredStream`, make sure that the peripheral is configured to transmit AES67 streams. Also make sure that `xConfiguration Audio Ethernet SAPDiscovery Mode` is set to **On** and that `xConfiguration Audio Ethernet Encryption` is set to **Optional**.

If the stream still does not show, check the peripheral vendor's documentation, and confirm that it announces its streams using SAP and that the IP address used matches that of `xConfiguration Audio Ethernet SAPDiscovery Address`.

If there is no audio from a peripheral stream

First, make sure that the configured Ethernet connector is not muted. Also, if the audio routing on the Cisco device has been customized, make sure that the Ethernet connector receiving the stream is routed where it should. You can also verify that the Cisco device is receiving audio packets, using `xStatus Audio Input Connectors Ethernet <Connector index> PacketsReceived`.

There is also `xStatus Audio Input Connectors Ethernet <Connector index> PacketsLost`. These counters can be reset by the command `xCommand Audio LocalInput Ethernet PacketStatisticsReset`.

If the peripheral is unable to subscribe to a stream from the Cisco device

If the peripheral uses SAP to discover incoming streams, make sure that it is listening for announcement on the IP address set in `xConfiguration Audio Ethernet SAPDiscovery Address`. If that is the

case, verify that the SAP is not being blocked by for example a switch on the network. This can be done with a tool like Wireshark.

If the stream is discovered by the peripheral but does not contain any sound, make sure that the corresponding Ethernet connector is not muted on the Cisco device. Also, if the audio routing of the Cisco device has been customized, make sure that there is audio routed to the Ethernet connector associated with the stream.

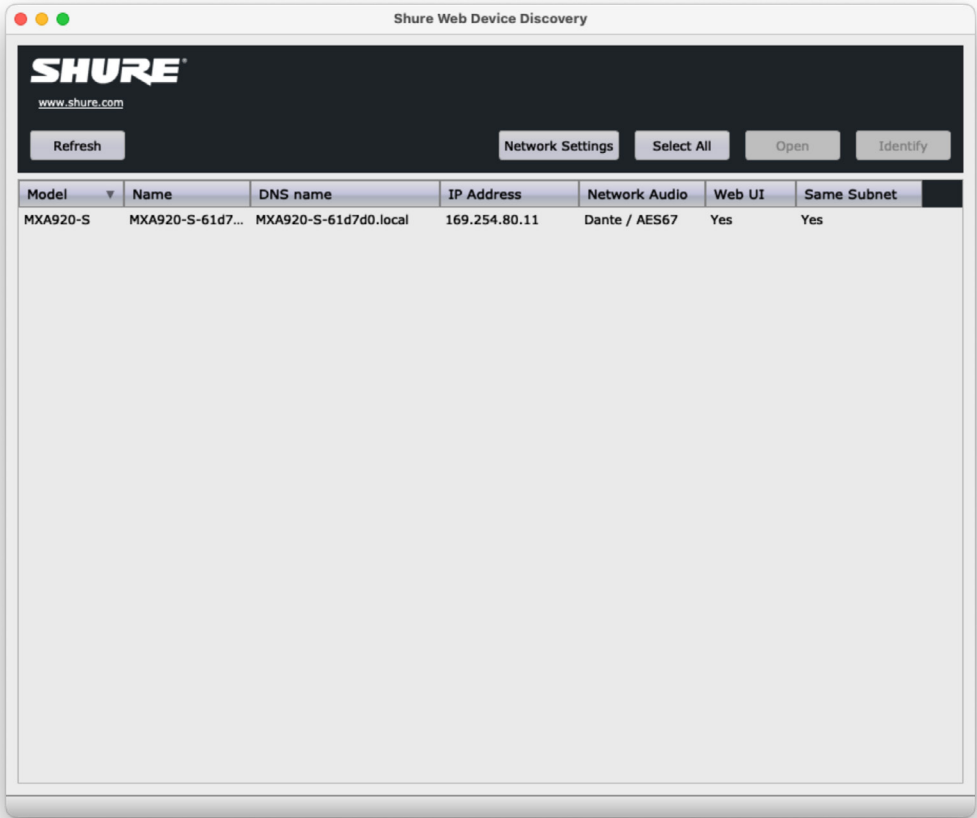
Examples

Disclaimer: The setup procedures for peripherals described here are meant for guidance only and may be subject to change. Please consult the documentation from the peripheral vendor for correct, up to date instructions.

Shure MXA920

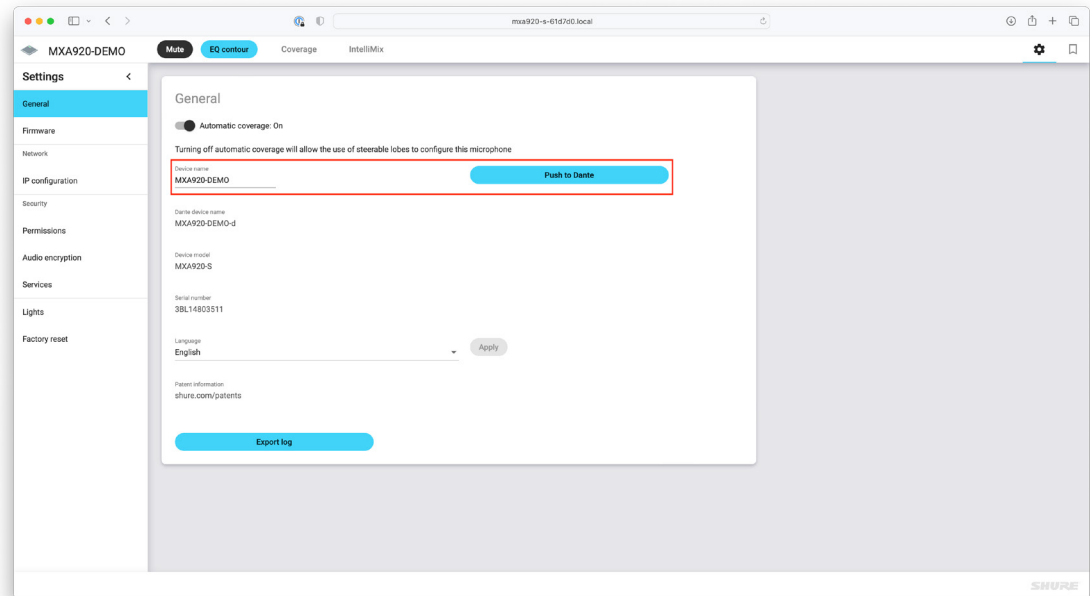
Start by connecting the microphone’s control interface to a PoE output of a switch or a PoE injector for power. **Do not connect to the Cisco device’s local network yet.** Also connect a computer to this network to access the microphone.

Find it using the Shure Web Device Discovery tool and access its interface by double clicking on it.

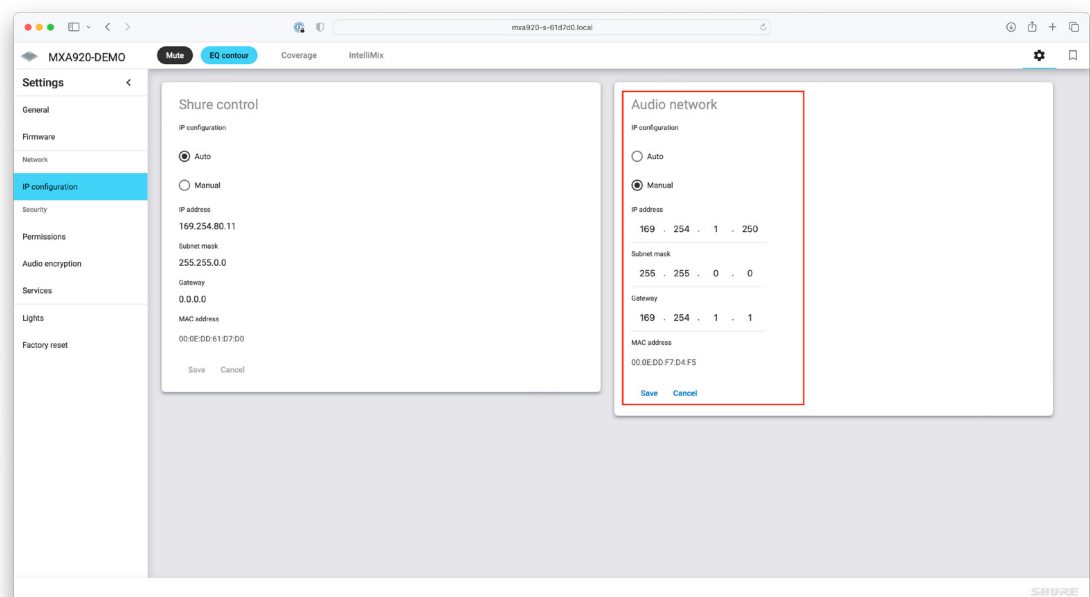


As a general rule, we recommend upgrading the firmware of the microphone before starting. The firmware version used in this example was 1.5.22.0.

In the microphone's interface, click the cogwheel icon in the top right to access settings. If you want, change the name, and click "Push to Dante". This will name the streams you set up later in Dante Controller accordingly.

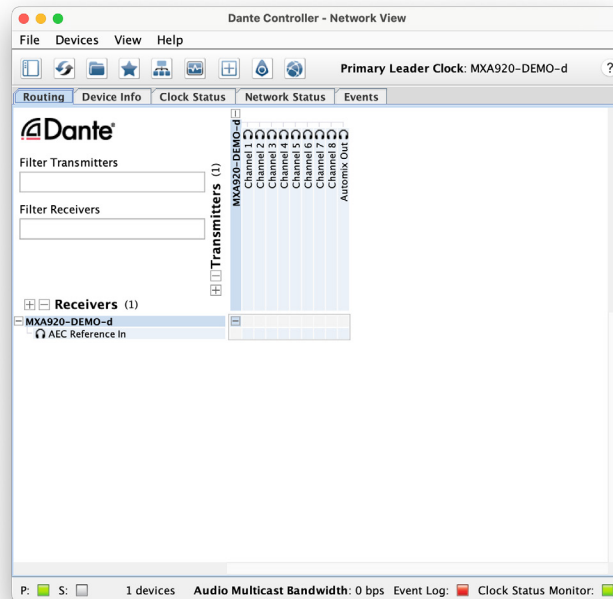


Under Network, find IP configuration in the left-hand sidebar and set a static IP address for the audio network interface. If you want the Shure control interface on the Cisco device's local network as well, this needs a static IP address too.

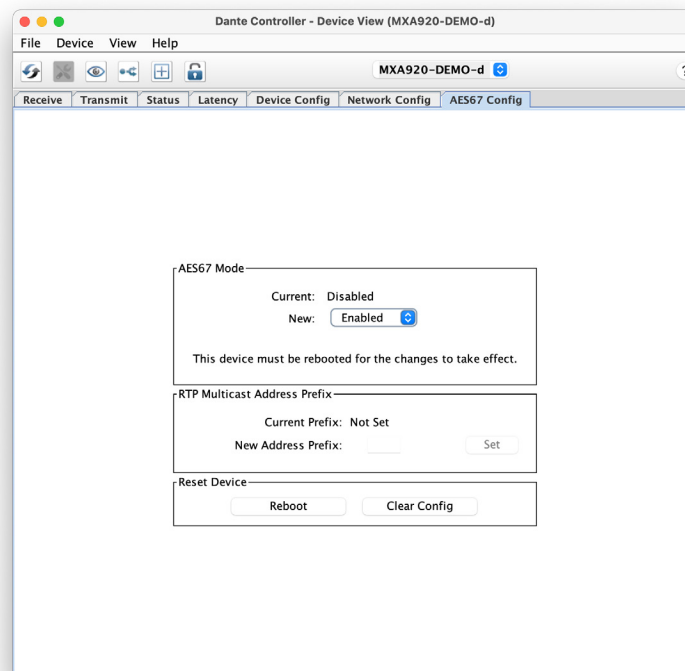


At this point you can connect the MXA920 to the Cisco device's local network.

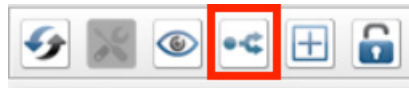
Open Dante Controller and find the microphone in the routing table.



Double click the microphone's name to access its settings. First enable AES67 under the “AES67 Config” tab.



Reboot the microphone. When it is back up, click the button to create a new multicast flow.



Create an AES67 flow and select the channels you want to include. We want to leverage the MXA920's built in DSP features, so we include only Automix Out.

Create Multicast Flow

MXA920-DEMO-d supports up to **64** channels per flow.

RTP flows for AES67 have a maximum of **8** channels per flow.

Select one or more transmit channels to be placed in multicast flows.

Audio Flow Config (Optional)

☐ Dante ☒ AES67

Destination Address: ☒ Auto ☐ Manual

Channel Name	☐	Add to New Flow
🔊 Channel 1	☐	☐
🔊 Channel 2	☐	☐
🔊 Channel 3	☐	☐
🔊 Channel 4	☐	☐
🔊 Channel 5	☐	☐
🔊 Channel 6	☐	☐
🔊 Channel 7	☐	☐
🔊 Channel 8	☐	☐
🔊 Automix Out	☒	☒

Create Cancel

Now, access the Cisco device's command line interface (SSH) using a terminal tool like Tera term or Putty. (Hint: If your computer is connected to the device's local network, it will have the IP address 169.254.1.1).

Allow unencrypted audio traffic, and enable SAP discovery:

```
$ xConfiguration Audio Ethernet Encryption: Optional
$ xConfiguration Audio Ethernet SAPDiscovery Mode: On
```

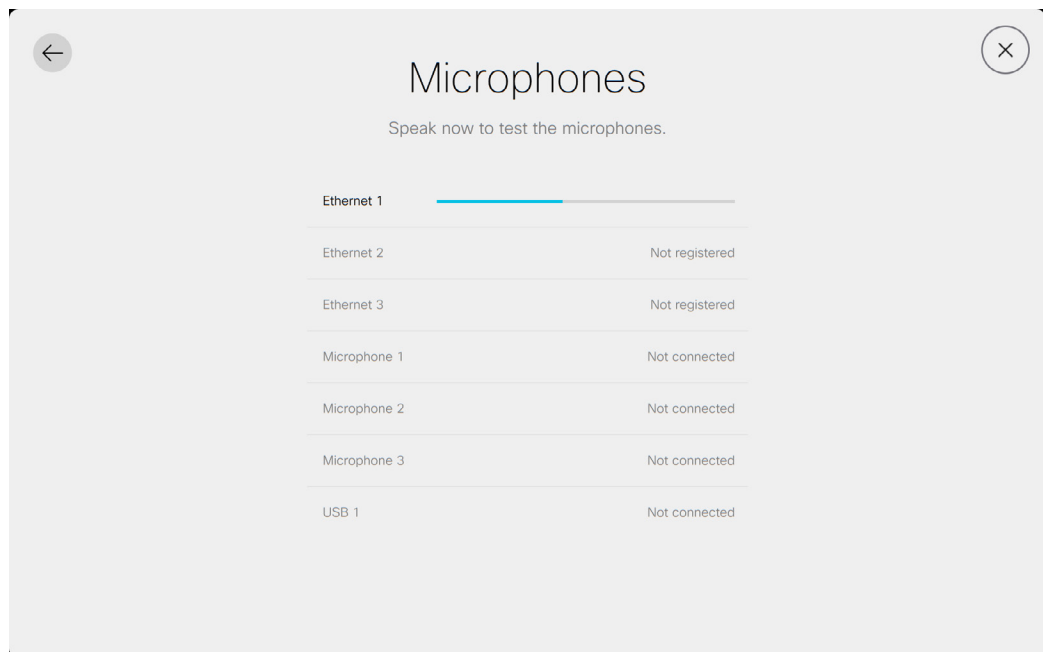
Verify that the stream set up in the previous step is discovered by the Cisco device:

```
$ xStatus Audio Input Ethernet DiscoveredStream
*s Audio Input Ethernet DiscoveredStream 1 Channels: 1
*s Audio Input Ethernet DiscoveredStream 1 MediaIP: "239.69.105.96"
*s Audio Input Ethernet DiscoveredStream 1 Name: "mxa920-demo-d : 16"
*s Audio Input Ethernet DiscoveredStream 1 OriginIP: "169.254.1.250"
*s Audio Input Ethernet DiscoveredStream 1 Status: "Valid offer"
```

Now register the stream to an available Ethernet connector:

```
$ xCommand Audio LocalInput Ethernet Register StreamName: "mxa920-demo-d : 16"
```

We can now check the VU meter to see that we have audio and then adjust the gain.



In our case we found a gain of 15 well suited for this microphone:

```
$ xConfiguration Audio Input Ethernet 1 Channel 1 Gain: 15
```

or for Codec Pro:

```
$ xConfiguration Audio Input Ethernet 1 Channel 1 Level: 15
```

As we will be using the MXA920's own echo processing, turn off echo cancellation on the Cisco device:

```
$ xConfiguration Audio Input Ethernet 1 EchoControl Mode: Off
```

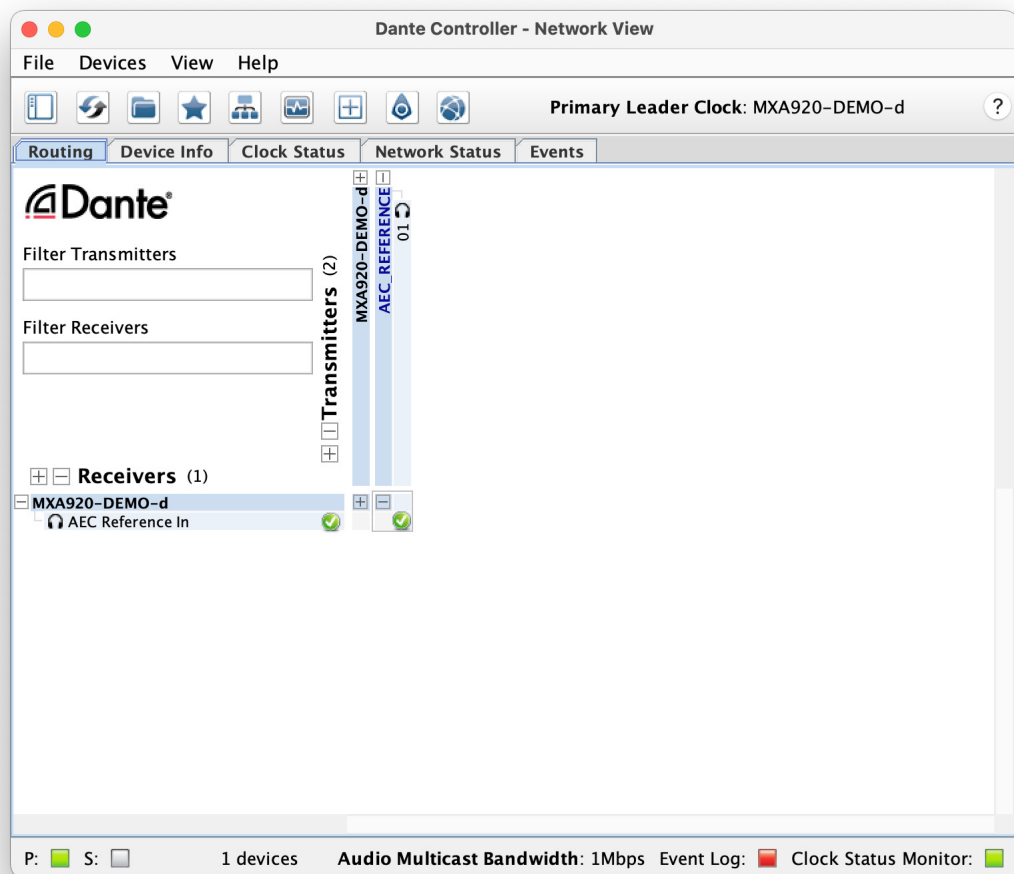
We also need to set up a reference for the echo cancellation on the microphone. This should have one channel, and we need to pick a suitable name and a multicast address:

```
$ xCommand Audio LocalOutput Ethernet Register Channels: 1 Medialp: 239.69.45.45 StreamName: "AEC_REFERENCE"
```

This stream is now listed in an xStatus:

```
$ xStatus Audio Output Connectors Ethernet
*s Audio Output Connectors Ethernet 2 Channels: 1
*s Audio Output Connectors Ethernet 2 Medialp: "239.69.45.45"
*s Audio Output Connectors Ethernet 2 Name: "AEC_REFERENCE"
```

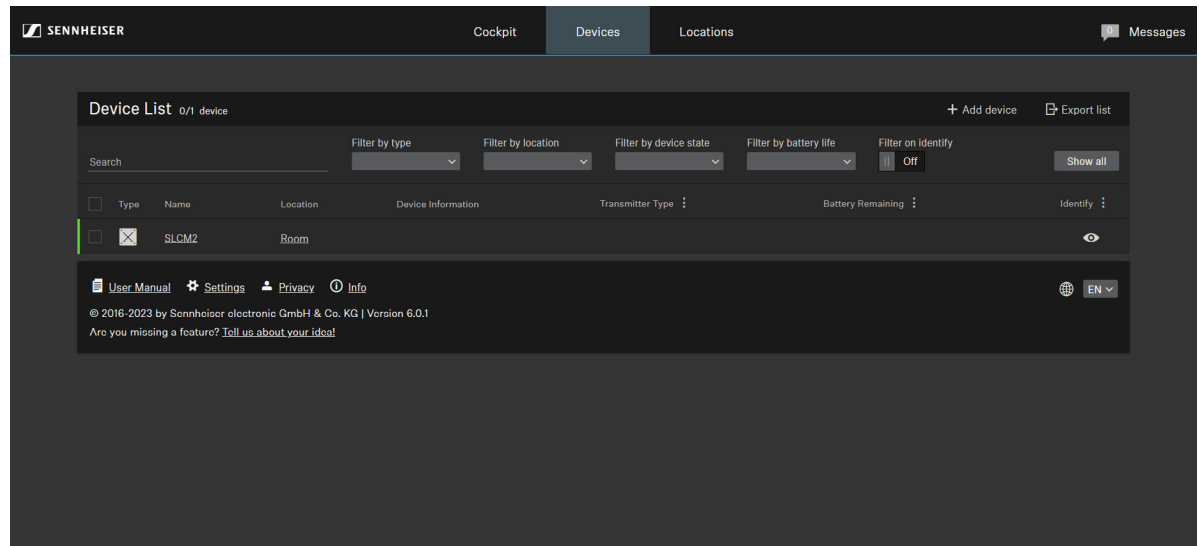
Going back to Dante Controller, this stream should now be visible in the routing view. Route it to the “AEC Reference In” input channel of the MXA920.



Now, the only thing left is to verify the setup. We recommend either placing a test call or listening to the microphone locally using “Test microphone” in the microphone menu of the Cisco device.

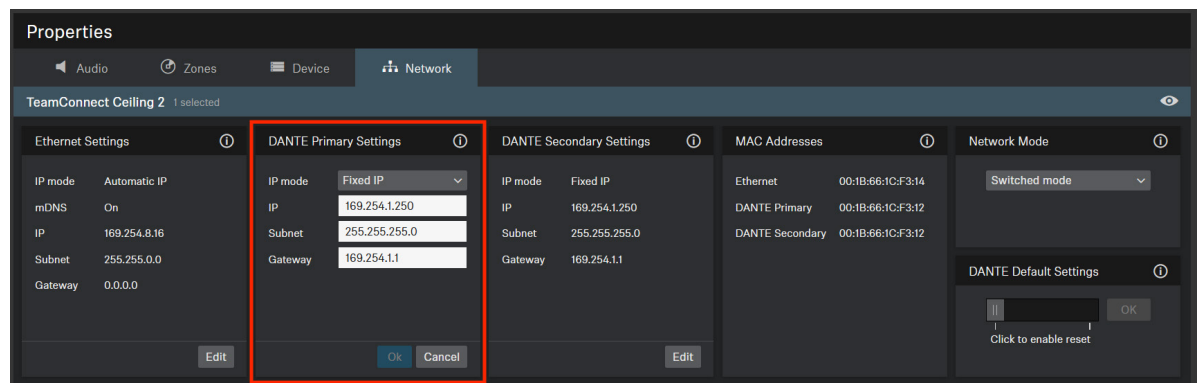
Sennheiser TeamConnect 2

Start by connecting the microphone's control interface to a PoE output of a switch or a PoE injector for power. Also connect a computer to this network to access the microphone in Sennheiser's Control Cockpit software. **Do not connect to the Cisco device's local network yet.**



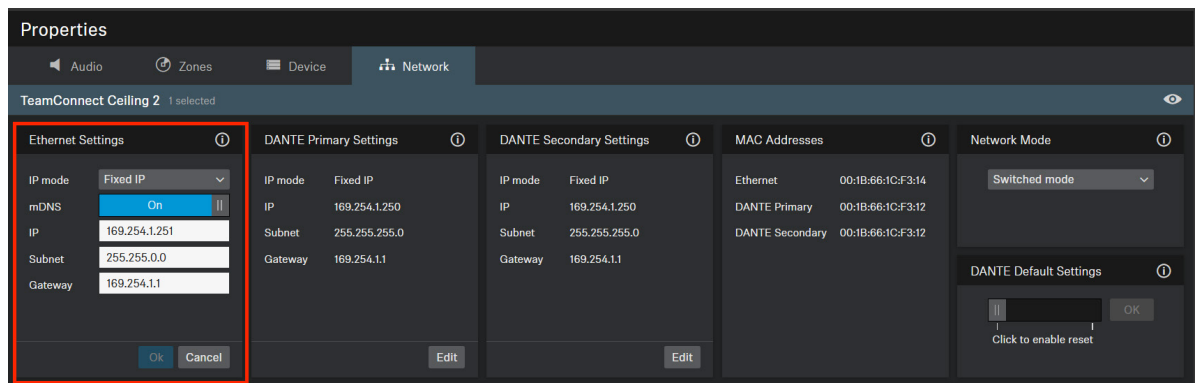
As a general rule, we recommend upgrading the firmware of the microphone before starting. The firmware version used in this example was 1.8.4.

Click the name of the device to access its settings and move to the *Network* tab. Set a static IP address for the Dante interface, here we have chosen 169.254.1.250. Also make sure to set the gateway to 169.254.1.1.



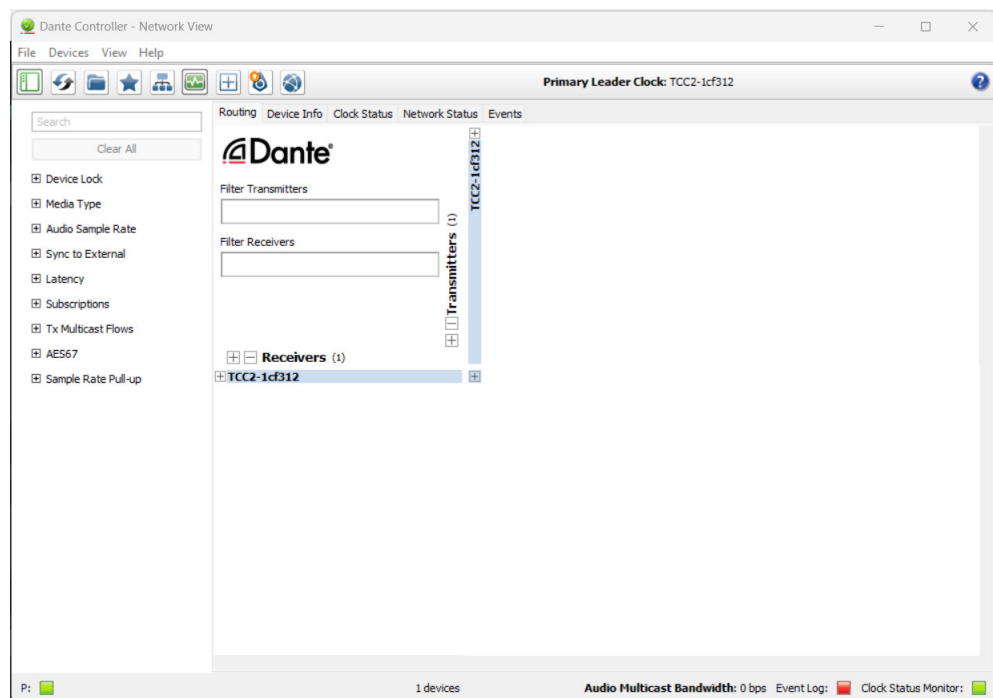
The Dante Ethernet interface can now be connected to the Cisco device's local network.

If you want the Control Ethernet interface connected to the device local network as well, this also needs a static IP address *before* connecting.

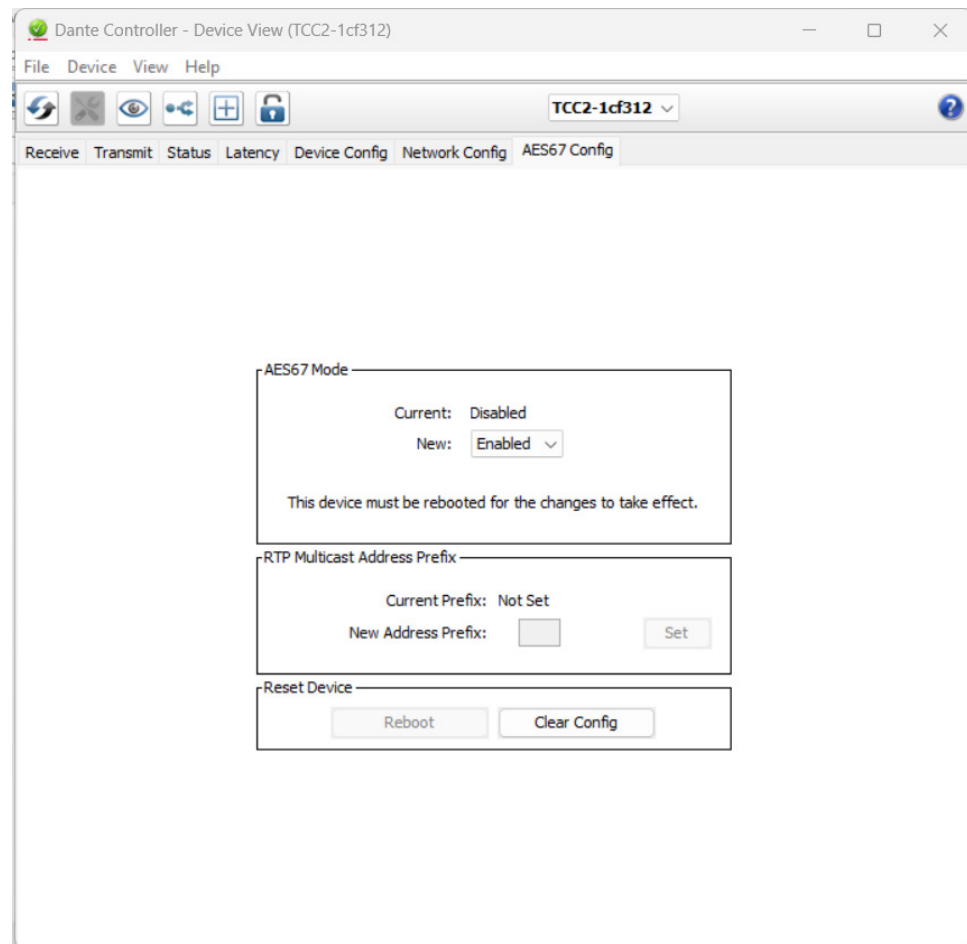


Now, connect your computer to the local network of the Cisco device and open Dante Controller.

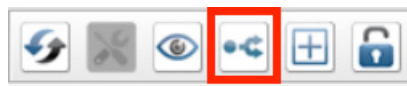
You should see the microphone appear.



Open the device by double clicking on its name in the routing view. Now move to the *AES67 Config* tab to enable AES67 mode.



Reboot the device in the Sennheiser Control Cockpit. When the device is back up, click the button to create a new multicast flow.



Create an AES67 flow and include the *MainOut* channel.

Create Multicast Flow

TCC2-1cf312 supports up to 16 channels per flow.

RTP flows for AES67 have a maximum of 8 channels per flow.

Select one or more transmit channels to be placed in multicast flows.

Audio Flow Config (Optional)

Dante

AES67

Destination Address:

Auto

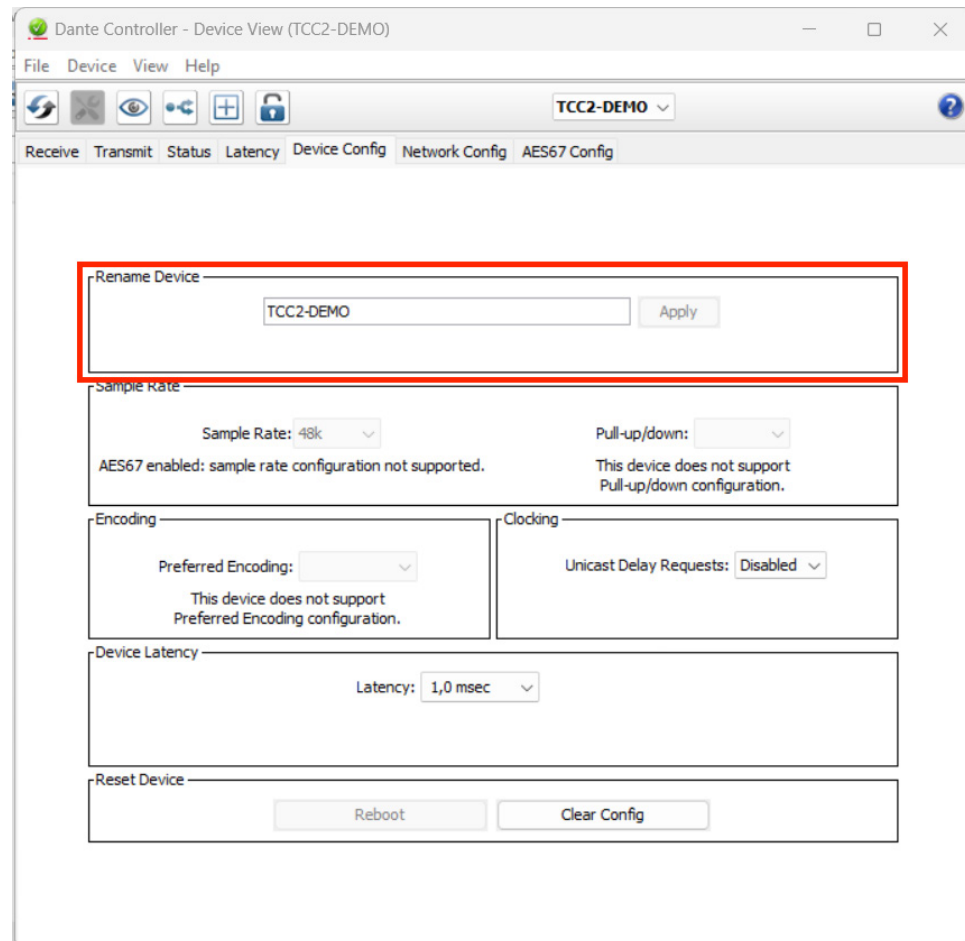
Manual

Channel Name	Add to New Flow
MainOut	☒
FarEndOut	☐

Create

Cancel

Dante automatically chose the name TCC2-1cf312 for this microphone, and will derive stream names from this. If you want a more descriptive name, you can change it in the device config tab.



Now, access the Cisco device's command line interface (SSH) using a terminal tool like Tera term or Putty. (Hint: If your computer is connected to the device's local network, it will have the IP address 169.254.1.1).

Allow unencrypted audio traffic, and enable SAP discovery:

```
$ xConfiguration Audio Ethernet Encryption: Optional
$ xConfiguration Audio Ethernet SAPDiscovery Mode: On
```

Verify that the stream set up in the previous step is discovered by the Cisco device:

```
$ xStatus Audio Input Ethernet DiscoveredStream
*s Audio Input Ethernet DiscoveredStream 1 Channels: 1
*s Audio Input Ethernet DiscoveredStream 1 MediaIP: "239.69.129.129"
*s Audio Input Ethernet DiscoveredStream 1 Name: "tcc2-demo : 16"
*s Audio Input Ethernet DiscoveredStream 1 OriginIP: "169.254.1.250"
*s Audio Input Ethernet DiscoveredStream 1 Status: "Valid offer"
```

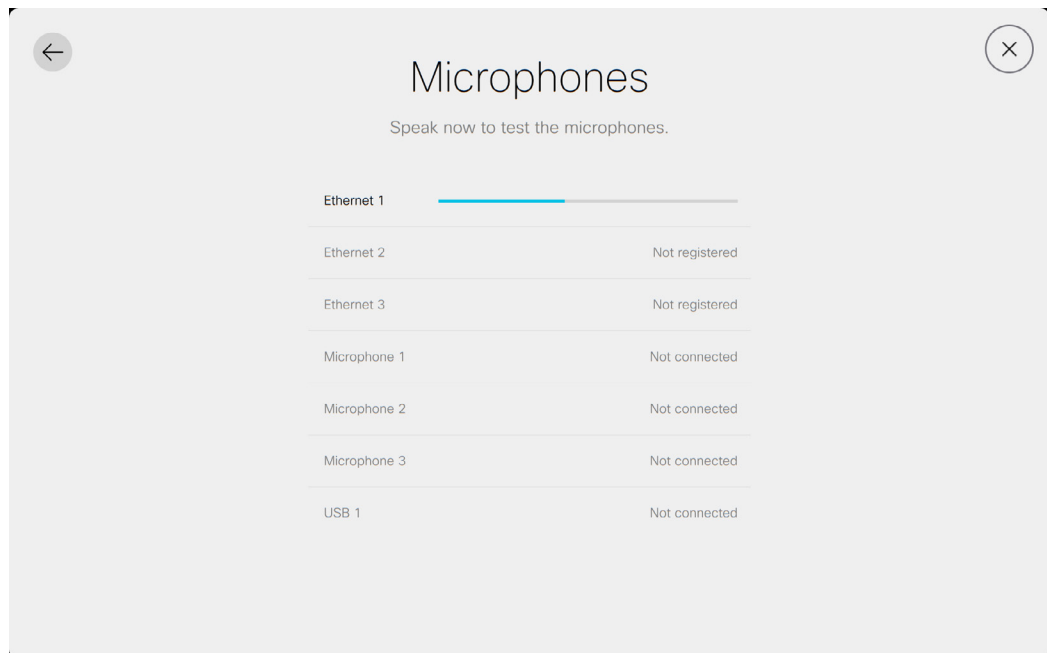
We see that the stream was indeed discovered. Now we need to register this stream to an available Ethernet connector:

```
$ xCommand Audio LocalInput Ethernet Register StreamName: "tcc2-demo : 16"
```

If we check the xStatus again, we see that the status of the discovered stream has changed:

```
$ xStatus Audio Input Ethernet DiscoveredStream
*s Audio Input Ethernet DiscoveredStream 1 Channels: 1
*s Audio Input Ethernet DiscoveredStream 1 MediaIP: "239.69.129.129"
*s Audio Input Ethernet DiscoveredStream 1 Name: "tcc2-demo : 16"
*s Audio Input Ethernet DiscoveredStream 1 OriginIP: "169.254.1.250"
*s Audio Input Ethernet DiscoveredStream 1 Status: "Active"
```

We can now check the VU meter to see that we have audio and then adjust the gain.



In our case we found a gain of 15 well suited for this microphone:

```
$ xConfiguration Audio Input Ethernet 1 Channel 1 Gain: 15
```

or for Codec Pro:

```
$ xConfiguration Audio Input Ethernet 1 Channel 1 Level: 15
```

This microphone can also receive an AEC reference⁵. To set that up, first register an outgoing stream from the Cisco device. We called ours **AEC_REFERENCE** and selected 239.69.45.45 as our media address:

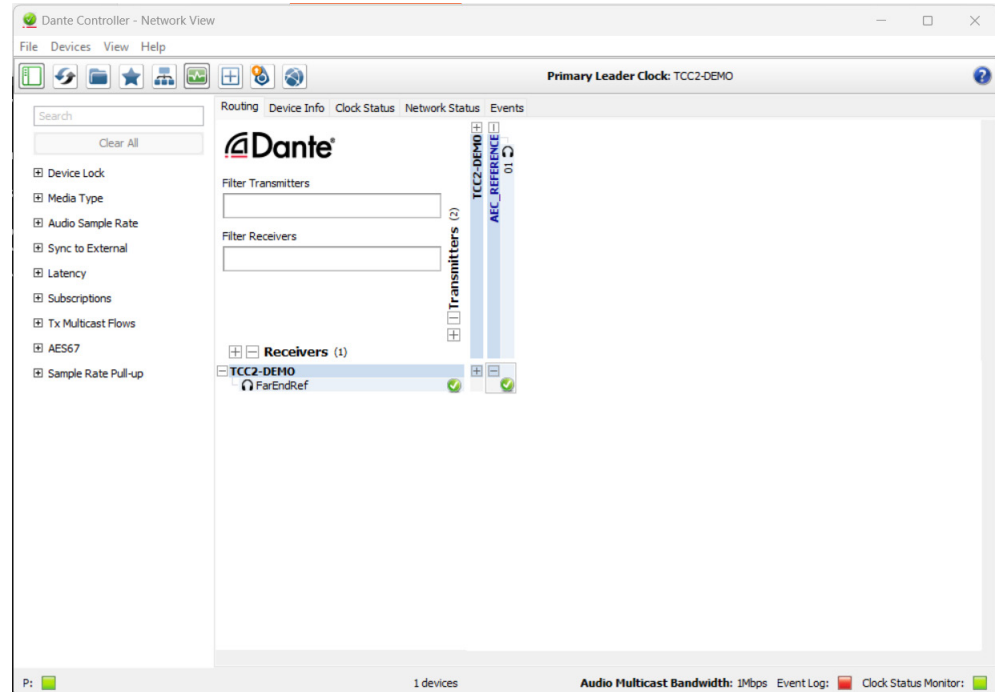
```
$ xCommand Audio LocalOutput Ethernet Register Channels: 1 ConnectorId: 2 MediaIp: 239.69.45.45 StreamName: "AEC_REFERENCE"
```

⁵ Note that the microphone does not do echo cancellation, but takes the reference to avoid tracking echo sources. Hence, we do not turn off echo cancellation on the Cisco device.

This will now appear in the list of active streams:

- \$ xStatus Audio Output Connectors Ethernet
- *s Audio Output Connectors Ethernet 2 Channels: 1
- *s Audio Output Connectors Ethernet 2 MediaIp: "239.69.45.45"
- *s Audio Output Connectors Ethernet 2 Name: "AEC_REFERENCE"

It should also appear in Dante Controller where it can be routed to the microphone's "FarEndRef".



Now, the only thing left is to verify the setup. We recommend either placing a test call, or listening to the microphone locally using "Test microphone" in the microphone menu of the Cisco device.

March 2025