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 A	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

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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)

	Supported nu	mber of streams
Device	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 RoomOS 11.8.

Board Pro and Board Pro G2:

• The minimum required software version 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 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.

e DNS name 20-S-61d7 MXA920-S-6	IP Addres.	s Network Audio	Web UI Sar	me Subnet
20-3-0107 MXA920-3-0	10700.10Cal 109.254.80	Dante / AECE7	Vec Vec	
		Dante / AES67	tes Yes	

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.

••• 🗆 - < >	0 mma920-s-61d7d0.local	د (🕲 🖞 + د
MXA920-DEMO	Mute EQ contour Coverage IntelliMix	۵ 🗅
Settings <		
General	General	
Firmware	Automatic coverage: On	
letwork	Turning off automatic coverage will allow the use of steerable lobes to configure this microphone	
P configuration	MXA920-DEMO Push to Dante	
Security	Durite divice same	
Permissions	MXA320-DEMO-3	
Audio encryption	Delde model MXA920-S	
Services	Serial number	
Jghts	38L14803511	
Factory reset	Language English	
	Pater Information shure.com/patents	
	Fronting	
	edbarrad	

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.

MXA920-DEMO Mail Coverage Intellificit Settings Audio network Poordgaration Firmmare @ Audo Audio network Poordgaration @ Audo Poordgaration @ Audo @ Manual Poordgaration Poordgaration @ Manual Podotase 169 - 254 - 11 - 250 Sodor mask Audo encryption 2552:55.00 255 - 0 - 0 Sodor mask	\$
Settings Settings Shure control Pointguation Pointguation Name O Audo Pointguation O Name O Name O Pointguation O Name O Pointguation O Name O Name O Pointguation O Name O Name O Name D Name </th <th></th>	
stand Shure control Audio network Pronfiguration Pronfiguration Pronfiguration Namad O Auto Pendiguration O Manual Storry Prodest 199.254.00.11 199.254.00.11 Tommark 255.255.0.0	
Immare Ponfgation Ponfgation ennot • Ando • Ando • Ando ponfgation • Manual • Manual • Manual • Manual secury • poderse • Poderse • Poderse • 199 254 • 1 255 255	
Auto O Auto Peomfguration Manual Image: Comparison of the state of the stateo	
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P advise P advise temstators 169/254.0.1 159/254.0.1 256/256.0 udio encrystion 255/255.0 0 0	
169/254.011 199/254.01 199/254.01 umisators but mask 100/200 udo encyption 255/255.00 0.00	
udio encryption 255.255.0.0 255 . 0 . 0	
arvices 0.0.0.0 Contemport	
109 254 1 . 1	
actory reset 00.06EDD 61.07.00 MAC address	
Save Cancel 00.0EDB.97.04.45	
Save Cancel	

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

🗋 😏		- E	± 💩 🚳	Primary Leader Clock: MXA920-DEMO-d	?
Routing	Device Info	Clock Status	Network Status	Events	
Da Cilter Trans	nte [®] mitters vers	Transmitters (1)	MX4920-DEMO Channel 1 Channel 3 Channel 3 Channel 5 Channel 6 Channel 6 Channel 6		
⊞ ⊟ Reco MXA920-D - ∩ AEC Re	eivers (1) EMO-d Iference In	E			

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.

Image: Contract of the second seco
Receive Transmit Status Latency Device Config Network Config AES67 Config AES67 Mode Current: Disabled
AES67 Mode Current: Disabled New: Enabled This device must be rebooted for the changes to take effect. RTP Multicast Address Prefix Current Prefix: Not Set New Address Prefix: Set Reset Device Reset Device Reboot Clear Config
New Address Prefix: Set Reset Device
Reboot Clear Config
Clear coning

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.

Dante O AES67 Destination Address: Auto Manual Channel Name O Channel 1 O Channel 2 O Channel 3 O Channel 4
Destination Address: Auto Manual Channel Name Add to New F Channel 1 Channel 2 Channel 3 Channel 4
Channel Add to Name Add to New F O Channel 1 O Channel 2 O Channel 3
Name New F ∩ Channel 1 ∩ Channel 2 ∩ Channel 3 ∩ Channel 4
Channel 1 Channel 2 Channel 3 Channel 4
Channel 3
O Channel 4
Channel 4
O Channel 5
∩ Channel 6
∩ Channel 7
Channel 8

Now, access the Cisco device's command line interface. If you prefer to use the web interface, configurations and statuses can be found under *Settings* and commands can be executed from the *Developer API* page. (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.

\leftarrow	Microp Speak now to test t	hones he microphones.	×
	Ethernet 1		
	Ethernet 2	Not registered	
	Ethernet 3	Not registered	
	Microphone 1	Not connected	
	Microphone 2	Not connected	
	Microphone 3	Not connected	
	USB 1	Not connected	

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 MediaIp: 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 MediaIp: "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.

🗾 SENN	IHEISER	Cockpit	Devices	Locations	ž.		μ α	Messages
	Device List 0/1 device					+ Add device	🕞 Export list	
		Filter by type Filter by loca	tion Filter b	y device state V	Filter by battery life	Filter on identify	Show all	
9								
	SLCM2 Room						0	
	 <u>■ User Manual</u>	D Info o. KG Vorsion 6.0.1 !					⊕ EN ~	

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.

Properties				
🛋 Audio 🔮 Zones	E Device 📥 Network			
TeamConnect Ceiling 2 1 selected				•
Ethernet Settings ①	DANTE Primary Settings	DANTE Secondary Settings ①	MAC Addresses	Network Mode
IP mode Automatic IP mDNS On IP 169.254.8.16	IP mode Fixed IP → IP 169.254.1.250 Subnet 255.255.255.0	IP mode Fixed IP IP 169.254.1.250 Subnet 255.255.255.0	Ethernet 00:1B:66:1C:F3:14 DANTE Primary 00:1B:66:1C:F3:12 DANTE Secondary 00:1B:66:1C:F3:12	
Subnet 255.255.0.0 Gateway 0.0.0.0	Gateway 169.254.1.1	Gateway 169.254.1.1		DANTE Default Settings
	Ok Cancel			Click to enable reset

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.

Propertie	es											
┥ Audi	io 🕑 Zone		🔳 Device	🕂 Netv	vork							
TeamConnec	ct Ceiling 2 1 select											•
Ethernet Set	ttings	0	DANTE Prim	ary Settings	(i)	DANTE Se	condary Settings	(i)	MAC Addresses	Ō	Network Mode	١
IP mode mDNS IP	Fixed IP On 169.254.1.251	× 1	IP mode IP Subnet	Fixed IP 169.254.1.250 255.255.255.0		IP mode IP Subnet	Fixed IP 169.254.1.250 255.255.255.0		Ethernet DANTE Primary DANTE Secondary	00:1B:66:1C:F3:14 00:1B:66:1C:F3:12 00:1B:66:1C:F3:12	Switched mode	
Subnet Gateway	255.255.0.0 169.254.1.1	ncel	Gateway	169.254.1.1		Gateway	169.254.1.1				DANTE Default Settings	ОК
	Ok Car	icel										

Now, connect your computer to the local network of the Cisco device and open Dante Controller. You should see the microphone appear.

2 Dante Controller - Network	c View			- 0 ×
File Devices View Help				
📙 🗲 💼 ★ 🚠			Primary Leader Clock: TCC2-1cf312	
Search	Routing Device Info Clock Sta	tus Network Status Eve	ints	
Clear All	@Dante	1312 +		
Device Lock	Filter Transmitters	2-16		
		Ĕ		
Audio Sample Rate	Filter Receivers	L)		
E Sync to External Sync to External		itte		
E Latency	1	line in the second seco		
E Subscriptions ■		Ira		
AES67	E Receivers (1)			
E Sample Rate Pull-up	± TCC2-1cf312			
P: 📃		1 devices	Audio Multicast Bandwidth: 0 bps Event	Log: 🧧 Clock Status Monitor: 🚦

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

	TCC2-1d312 V		6
eceive Transmit Status Latency Devi	ice Config Network Config AES67 Config		
cenve mananite status catericy bevi			
LAES67 M	ode		
	Current: Disabled		
	New: Enabled ~		
This d	device must be rebooted for the changes to take effect.		
RTP Multi	icast Address Prefix		
	Current Brafiv: Not Sat		
	New Address Prefix: Set		
-Reset De	vice		
	Reboot Clear Config		

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.

ect one or more transmit chan	nels to be placed in multicast flo
Audio Flow Config (Optional)	
 Dante 	O AES67
Destination Address:	Auto 🔿 Manual
Channel Name	Add to New Flow
MainOut	
∩ FarEndOut	

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.

● •< +		TCC2-DEMO 🗸		
Transmit Status Late	ncy Device Config Network Config	AES67 Config		
Rename Device				
	TCC2-DEMO	Apply		
Sample Rate				
Sample Rate: 48k 🗸 🗸		Pull-up/down:		
AES67 enabled: sample rate configuration not supported.		This device does not support Pull-up/down configuration.		
Encoding	la	ocking -		
Preferred Enco	oding:	Unicast Delay Requests: Disabled ~		
This device	e does not support			
Preferred Er	ncoding configuration.			
Device Latericy	atency: 1.0 msec			
		<u>×</u>		
Reset Device —		-		
	Reboot	Clear Config		

Now, access the Cisco device's command line interface. If you prefer to use the web interface, configurations and statuses can be found under *Settings* and commands can be executed from the *Developer API* page. (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"

.........

CISCO

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.

Microphone Speak now to test the microph	S ones.	(\times)
Ethernet 1		
Ethernet 2	Not registered	
Ethernet 3	Not registered	
Microphone 1	Not connected	
Microphone 2	Not connected	
Microphone 3	Not connected	
USB 1	Not connected	

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".

ile Devices View Help			
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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.

July 2024

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