

# How to

Register on Axis Intercoms and Axis network speakers to WebEx by Cisco

May 2024

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# 1. Introduction

Axis network speakers and Axis network intercoms integrated with Cisco WebEx Calling deliver superior communication for voice paging, alerting, and secure door entry. Tested and certified with Cisco Webex Calling, Axis solutions ensure a smooth and efficient deployment.

Axis products have a rich feature set, including secure SIP with TLS and SRTP, central provisioning, and network supervision. Multicast capabilities allow for scalable solutions across various environments and segments such as education, healthcare, government, commercial offices, manufacturing, utilities/plants, transportation, distribution/warehouse, and retail.

Migrating to a Webex Calling environment while maintaining a functional IP audio system and integration to video management systems is straightforward with Axis network speakers and Axis network intercoms.

Axis network intercoms are enabling efficient visitor management, access control, and secure door entry.

Axis network speakers are enabling audio for voice paging, emergency notifications, and scheduled announcements.

This ensures simultaneous and prioritized audio broadcast over the Axis audio management software and WebEx Calling to Axis network speakers. As organizational needs evolve, the flexible Axis IP audio system adapts to meet diverse public address (PA) requirements.

The following document highlights how to register Axis network intercoms and Axis network speakers to WebEx by Cisco. The document will provide necessary steps on how to configure the Axis product as well as how to create and associate it with a user within WebEx. It also documents how to create a paging group within WebEx and how to add Axis devices to the paging group.

The document assumes that the reader has made sure that the device to be registered fulfills the prerequisite stated below.

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## Prerequisites

Axis device must support SIP
Firmware version 11.7.30 or later
Exception is the A8105-E with FW ver. 1.65.x

**PLEASE NOTE** that Axis doesn't take any responsibility for how this configuration may affect your system. If the modification fails or if you get other unexpected results, you may have to restore the settings to default.

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## 2. Configuring WebEx

Before an Axis product can be registered to WebEx a user and device needs to be created and associated with the Axis product. If using an already existing user skip step 2.1.

### 2.1. Create a user

1. Within "Management" in the WebEx Control Hub navigate to Users and click on "Add users".
2. Select and fill in the information appropriate for your user and click "Next".

**Step 1: Basic information**

Names and Email address

Email address

You can add up to 25 users.  
Valid example: john.doe@example.com, marysue@example.com

You cannot add existing users in your organisation or users that already have a Webex account. [Learn more](#)

JohnDoe@axis.com ×

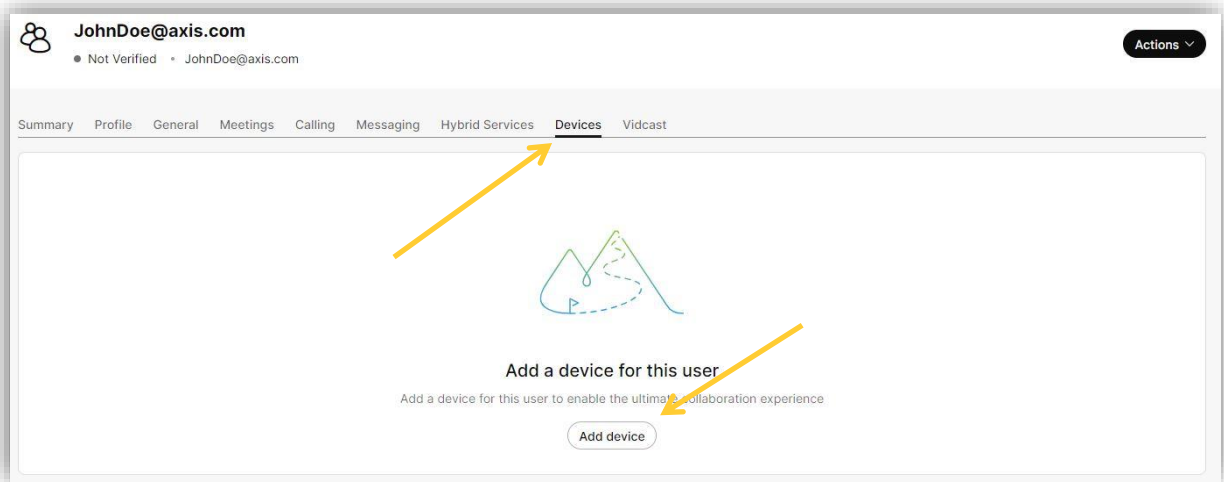
Enter user emails separated by commas

1/25 items Clear All

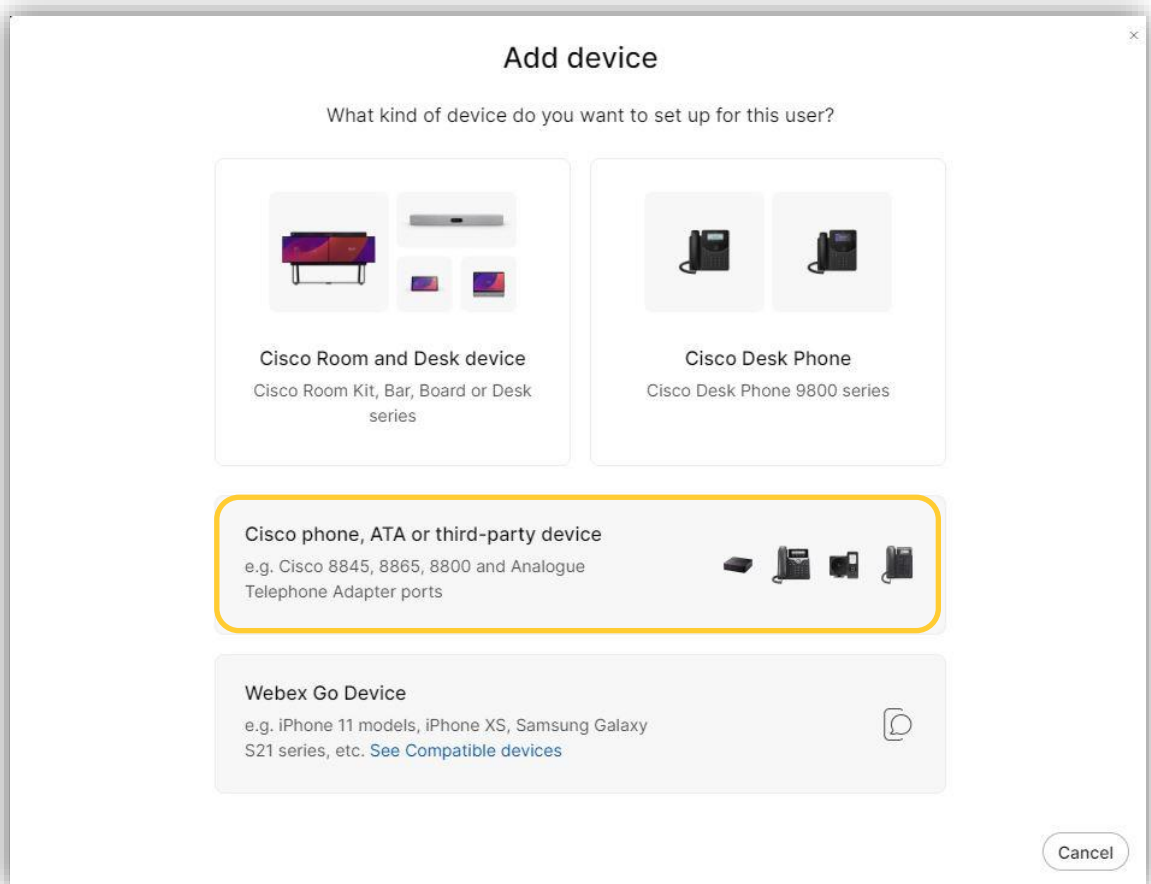
3. Assign one or more licenses to your user and then click next.
4. Review the information of the user and if the displayed information is correct click on "Add users" to finalize the creation. For troubleshooting or additional information see the following link:  
<https://help.webex.com/en-us/article/v71ztb/Add-users-manually-in-Control-Hub>

## 2.2. Associate device with user

1. Click on the newly created user and go to "Devices" and click on "Add device".



2. Click on "Cisco phone, ATA or third-party device".



3. In the "Select device" dropdown menu select "Customer-managed devices" and under "Device vendor" select "Axis Customer Managed". Fill in the MAC address of the Axis product that will be associated with this user, check the box click "Save".

**Add device**

Select the device model and activation method.

Select device  
Customer-managed devices

Device vendor  
Axis Customer Managed

Enter MAC address  
000000000000

\*I confirm that creating this device profile will expose sensitive device credentials that could be misused and exploited by users with unauthorised access. You are responsible for securing and recycling these credentials. You agree that Cisco is not responsible for any fraudulent charges or phone calls that result from the exposure of these device credentials. Cisco offers limited support for third-party devices connected via this interface. Cisco will only investigate basic issues for third-party devices that are actively registered to the Webex Calling platform. Any other third-party device-related support issues should be directed to the third-party device vendor, including issues related to onboarding, deployment, configuration or connectivity.

Previous Save

4. The association is complete and make sure to save the information displayed as it is needed to configure the Axis device.

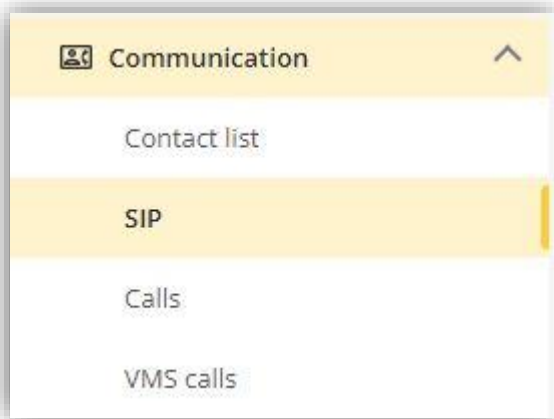
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# 3. Configuring the Axis products

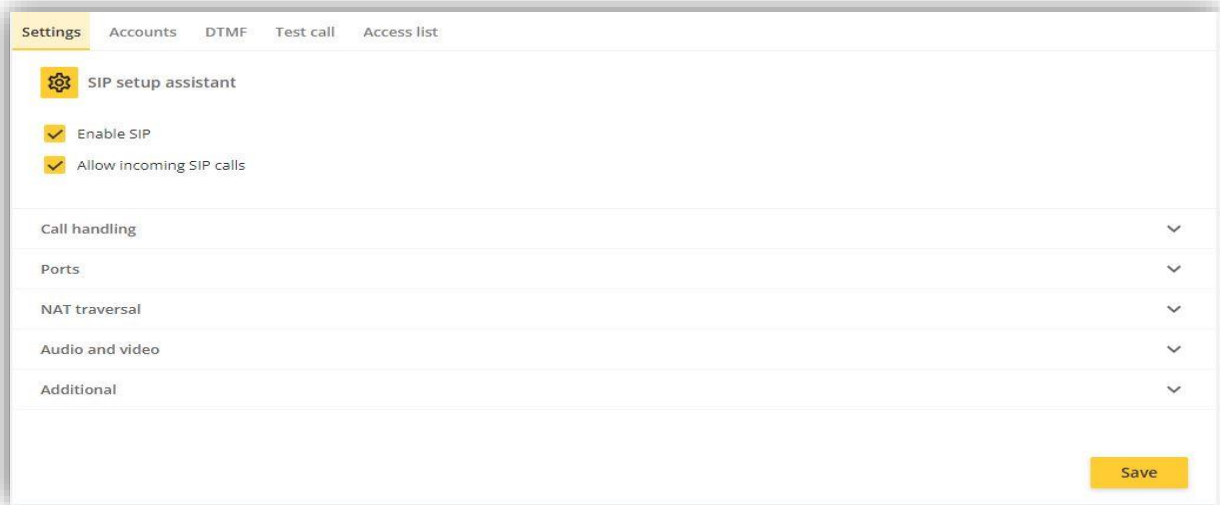
With the information from step 2.2 an Axis product can now be registered to Webex.

## 3.1. Register an Axis network intercom

1. Log into the Intercom by typing its IP address in a web browser. On the left-hand side click on "Communication" and then "SIP".

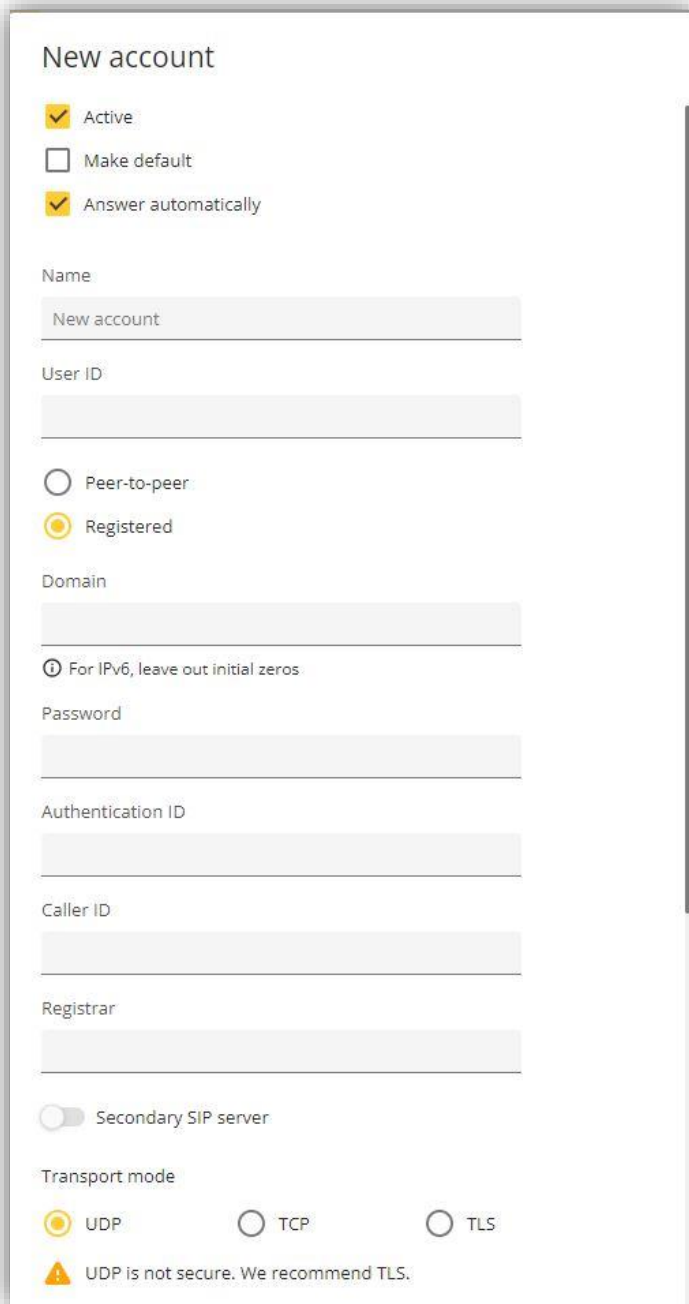


2. Click on "Enable SIP" and if the Axis network intercom should accept incoming call, click on "Allow incoming SIP calls" and then on "Save".





3. Go to "Accounts" at the top and click on "Add account". The highlighted fields below must be filled out to make a correct registration.



New account

Active

Make default

Answer automatically

Name

New account

User ID

Peer-to-peer

Registered

Domain

For IPv6, leave out initial zeros

Password

Authentication ID


Caller ID

Registrar

Secondary SIP server

Transport mode

UDP     TCP     TLS

 UDP is not secure. We recommend TLS.

- **Active** – Enables or disables the SIP account and should be left as enabled to use the account.
- **Make default** – If this account should be the default account. Only one account can be the default account. For WebEx this can be either enabled or disabled.

- 
- **Answer automatically** – If enabled the account will automatically answer any calls made to it. For WebEx this can be either enabled or disabled.
  - **Name** – The name of the account shown in the Intercom and APIs.
  - **User ID** – The ID associated with the device in WebEx. In this field type the first part, everything before the “@”, of the “Line ID” displayed in step 2.2.
  - **Domain** – The domain address associated with the WebEx Control hub. In this field type the second part, everything after the “@”, of the “Line ID” displayed in step 2.2.
  - **Password** – The password associated with the device in WebEx. In this field type the “SIP password” displayed in step 2.2.
  - **Authentication ID** – The authentication ID associated with the device in WebEx. In this field type the “SIP username” displayed in step 2.2.
  - **Caller ID** – What will be displayed on the receiving end when a call from this account is made. Can be empty.
  - **Registrar** – Not needed and can be left empty.
  - **Secondary SIP server** – Fields to add a backup server. Not needed and can be left empty.
  - **Transport mode** – What protocol to use when making calls. All transport can be used however TLS is highly recommended due to its being secure.

Transport mode

UDP     TCP     TLS

TLS version

Automatic ▾

Media encryption

SRTP mandatory ▾

Client certificate ⓘ

Default (self-signed) ▾

Verify server certificate ⓘ

SIPS ⓘ

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Proxies ^

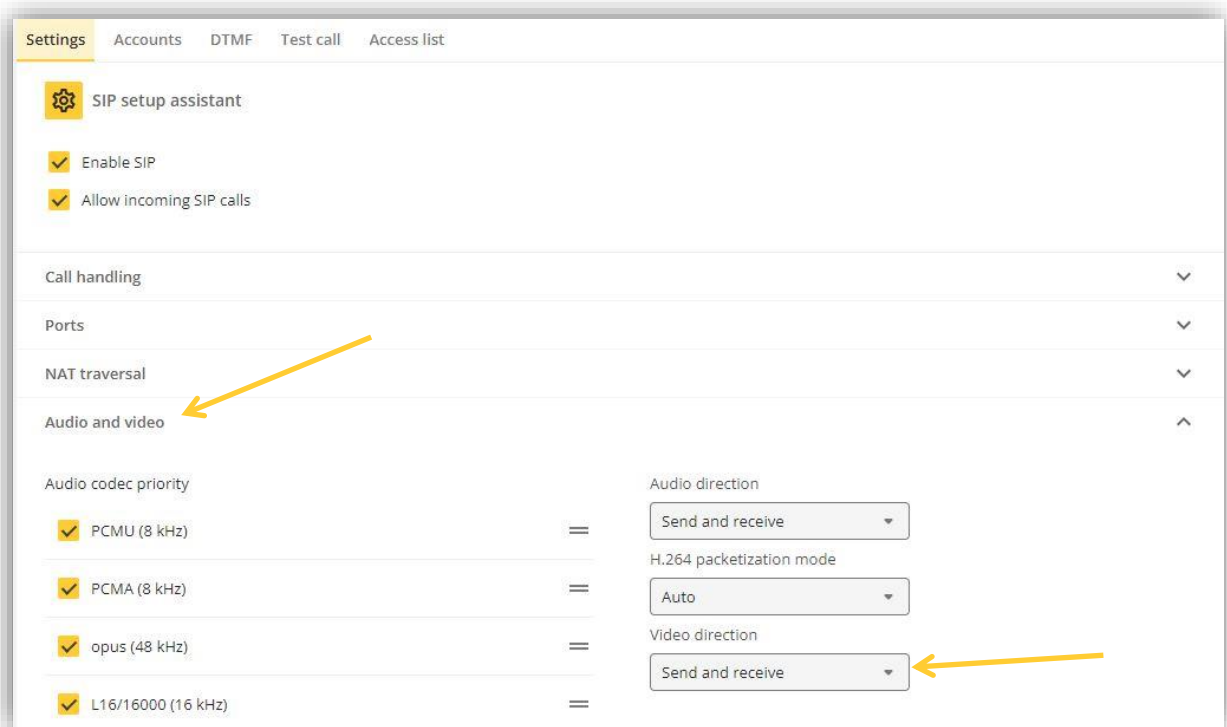
+ Add proxy    Prioritize

Server address	Username	Password	Transport
<input type="text"/>	<input type="text"/>	<input type="text"/>	UDP ▾

#### Settings needed if using TLS and proxies:

- **TLS version** – To specify TLS version. Automatic will dynamically change depending on the receiver.
- **Media encryption** – How the media shall be encrypted.
- **Client certificate** – Certificate to use when verifying the client.
- **Verify server certificate** – If enabled the Axis product will verify the server.
- **SIPS** – If enabled, they will use SIPS instead of SIP.

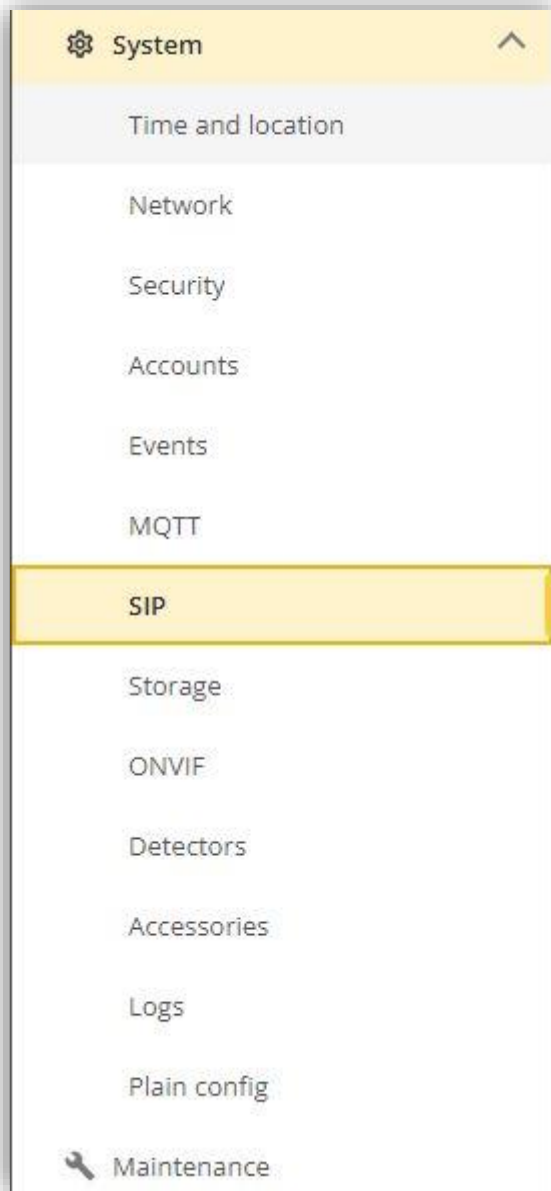
- **Server address** - The address of the proxy. If WebEx is using a proxy, type in this field the address shown under "Outbound proxy" displayed in step 2.2.
  - **Username** – Username used during authentication. If WebEx is using a proxy, type in this field the first part, everything before the "@", of the "Line ID" displayed in step 2.2.
  - **Password** – Password used during authentication. If WebEx is using a proxy, type in this field the text shown under "SIP password" displayed in step 2.2.
  - **Transport** – The transport to use with the proxy. Must match the transport of the account.
4. Click on "Save" and the status of the account should show green and "200 OK" for a successful registration. If shown red make sure that all the necessary fields are filled in correctly.
  5. To use video calls, click on "Settings" followed by "Audio and video". On the drop-down menu "Video direction" to the right click on "Send and receive" and then click on "Save".



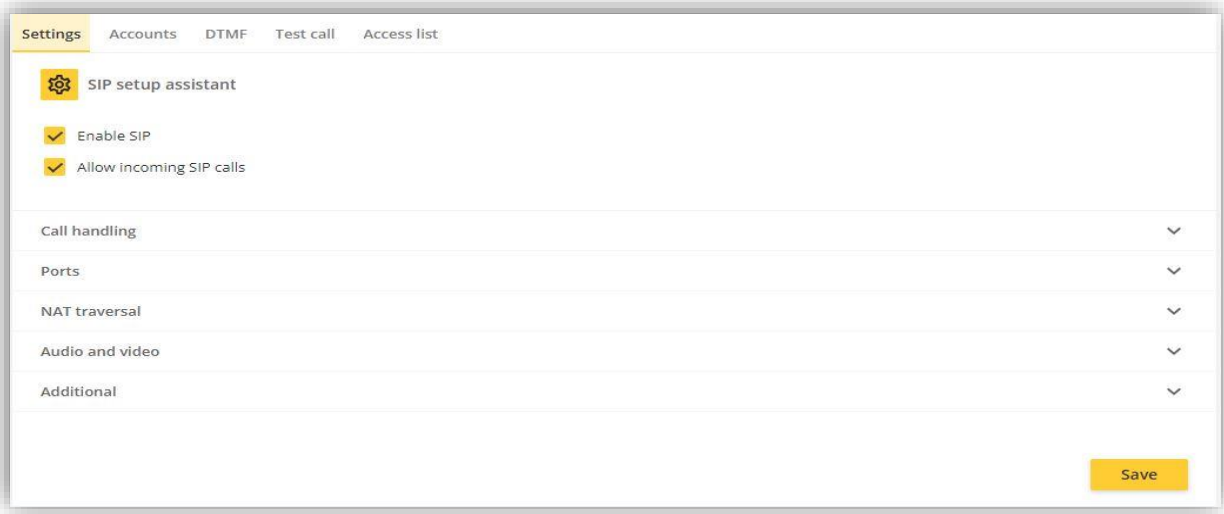
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## 3.2. Register an Axis network speaker

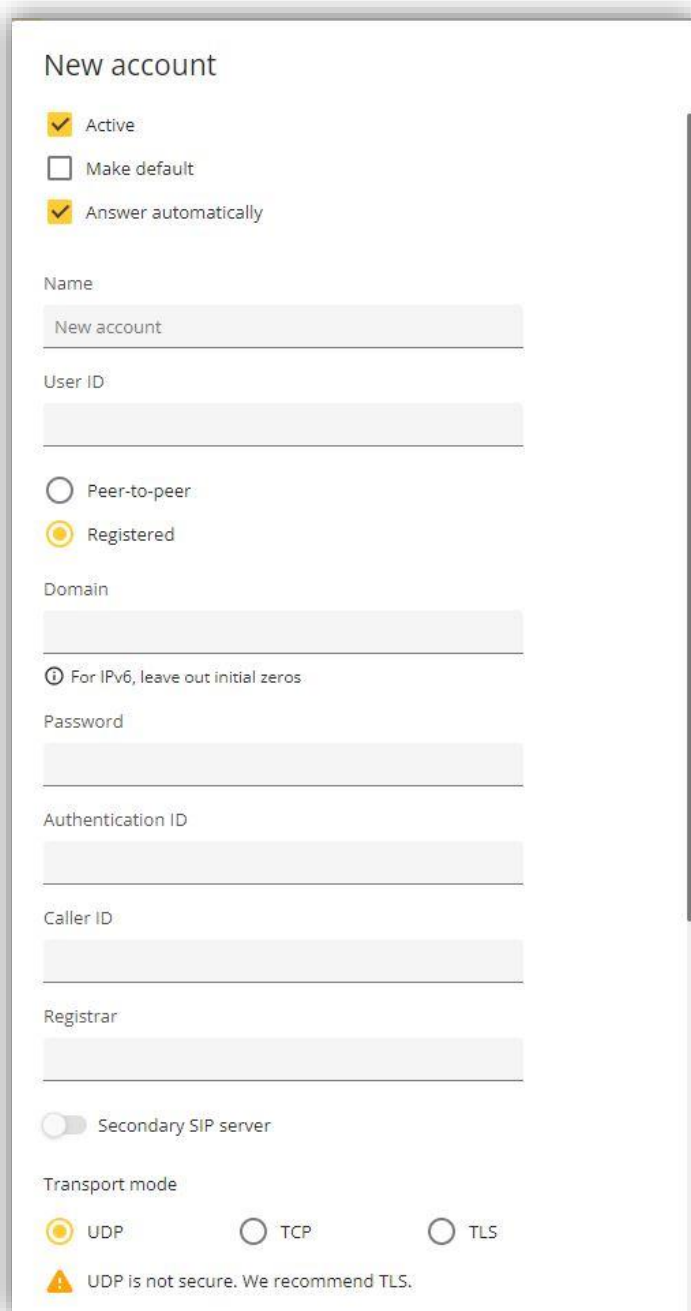
1. Log into the Axis network speaker by typing its IP address in a web browser. On the left-hand side click on "System" and then "SIP".



2. Click on "Enable SIP" and if the Axis network speaker should accept incoming call, click on "Allow incoming SIP calls" and then on "Save".



3. Go to "Accounts" at the top and click on "Add account". The highlighted fields below must be filled out to make a correct registration.



New account

Active

Make default

Answer automatically

Name

New account

User ID

Peer-to-peer

Registered

Domain

For IPv6, leave out initial zeros

Password

Authentication ID


Caller ID

Registrar

Secondary SIP server

Transport mode

UDP     TCP     TLS

 UDP is not secure. We recommend TLS.

- **Active** – Enables or disables the SIP account and should be left as enabled to use the account.
- **Make default** – If this account should be the default account. Only one account can be the default account. For WebEx this can be either enabled or disabled.

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- **Answer automatically** – If enabled the account will automatically answer any calls made to it. For WebEx this can be either enabled or disabled.
  - **Name** – The name of the account shown in the Intercom and APIs.
  - **User ID** – The ID associated with the device in WebEx. In this field type the first part, everything before the “@”, of the “Line ID” displayed in step 2.2.
  - **Domain** – The domain address associated with the WebEx Control hub. In this field type the second part, everything after the “@”, of the “Line ID” displayed in step 2.2.
  - **Password** – The password associated with the device in WebEx. In this field type the “SIP password” displayed in step 2.2.
  - **Authentication ID** – The authentication ID associated with the device in WebEx. In this field type the “SIP username” displayed in step 2.2.
  - **Caller ID** – What will be displayed on the receiving end when a call from this account is made. Can be empty.
  - **Registrar** – Not needed and can be left empty.
  - **Secondary SIP server** – Fields to add a backup server. Not needed and can be left empty.
  - **Transport mode** – What protocol to use when making calls. Each transport can be used however TLS is highly recommended due to its being secure.



The image shows a configuration panel with the following sections:

- Transport mode:** Three radio buttons for UDP, TCP, and TLS. The TLS option is selected and highlighted with a yellow circle.
- TLS version:** A dropdown menu currently set to "Automatic".
- Media encryption:** A dropdown menu currently set to "SRTP mandatory".
- Client certificate:** A dropdown menu currently set to "Default (self-signed)".
- Verify server certificate:** An unchecked checkbox with an information icon.
- SIPS:** An unchecked checkbox with an information icon.
- Proxies:** A section with a yellow "+" button, an "Add proxy" button, and a "Prioritize" button. Below this is a table with columns for "Server address", "Username", "Password", and "Transport". The "Transport" column has a dropdown menu currently set to "UDP".

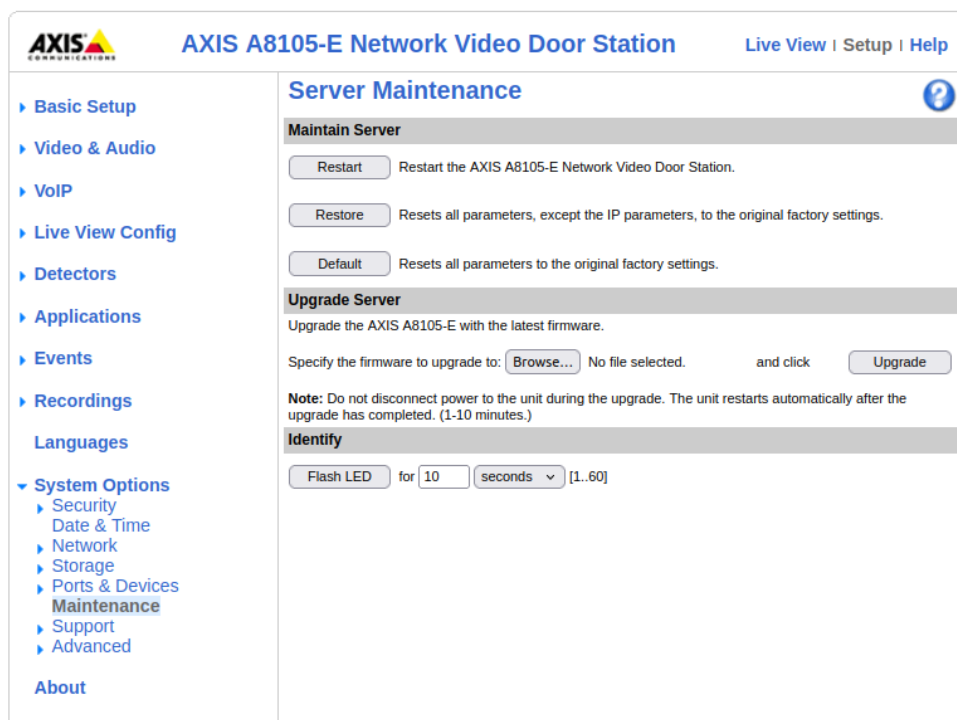
**Setting needed if using TLS and proxies:**

- **TLS version** – To specify TLS version. Automatic will dynamically change depending on the receiver.
- **Media encryption** – How the media shall be encrypted.
- **Client certificate** – Certificate to use when verifying the client.
- **Verify server certificate** – If enabled the Axis product will verify the server.
- **SIPS** – If enabled, they will use SIPS instead of SIP.

- 
- **Server address** - The address of the proxy. If WebEx is using a proxy, type in this field the address shown under "Outbound proxy" displayed in step 2.2.
  - **Username** – Username used during authentication. If WebEx is using a proxy, type in this field the first part, everything before the "@", of the "Line ID" displayed in step 2.2.
  - **Password** – Password used during authentication. If WebEx is using a proxy, type in this field the text shown under "SIP password" displayed in step 2.2.
  - **Transport** – The transport to use with the proxy. Must match the transport of the account.
4. Click on "Save" and the status of the account should show green and "200 OK" for a successful registration. If shown red make sure that all the necessary fields are filled in correctly.

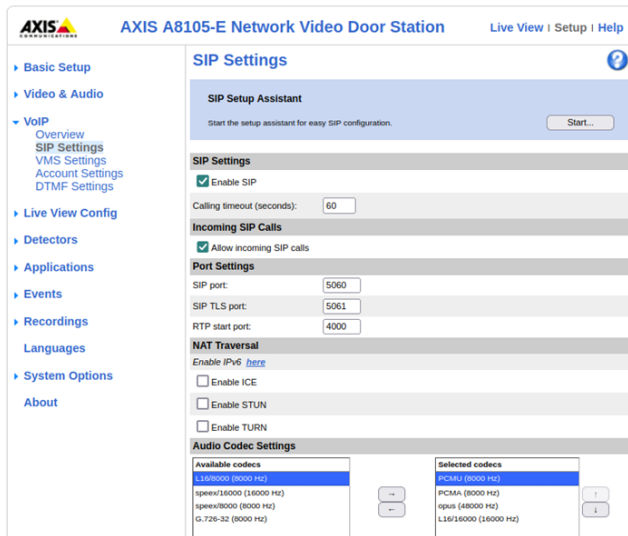
# 4. Register an Axis product with FW ver. (1.65.X)

## 4.1. Reset the Axis product to factory default

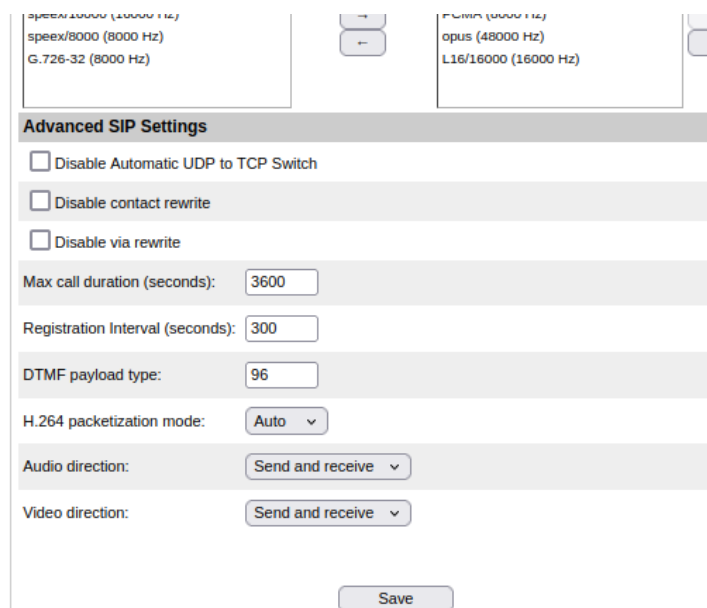


1. Log into the Axis product by typing its IP address in a web browser. On the upper right corner click on "Setup".
2. On the left menu, press System Options and go to "Maintenance"
3. Under the section Maintain Server, press Default. This will factory default the device and all settings on the device will be reset to default.
4. After the device has been reset to factory default, open the browser and enter the IP of the device.
5. The device will ask for entering new password to the root user, enter the desired password.
6. Configure the desired setting for the capture mode.
7. Configure the desired setting for the power line frequency.

## 4.2. Setting up SIP on the Axis product



1. Log into the Axis product by typing its IP address in a web browser. On the upper right corner click on "Setup".
2. Go to VoIP found on the left menu and press SIP Settings.
3. Click on "Enable SIP" and if the Axis product should accept incoming call, click on "Allow incoming SIP calls".
4. To use video calls, at the bottom on the SIP settings change the "Video direction" from "Send only" to "Send and receive" and then click on "Save".



## 4.3. Register a new SIP account using the web interface

The screenshot shows the web interface for the Axis A8105-E Network Video Door Station. The page title is "Account Settings". On the left, there is a navigation menu with the following items: Basic Setup, Video & Audio, VoIP (with sub-items: Overview, SIP Settings, VMS Settings, Account Settings, DTMF Settings), Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. The main content area displays a table of SIP accounts:

Name	SIP address	Transport	Default	Reg. status
peer-to-peer (peer-to-peer)	sip:172.25.9.64:5060	UDP	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Below the table are three buttons: "Add...", "Modify...", and "Remove". Below these buttons is a section titled "Test SIP Call" with the instruction: "Make a test call from the selected SIP account to the specified SIP address." There is a link "Enable IPv6 here" and a text input field with the placeholder "Enter SIP address: sip(s):extension@domain" and a "Test call" button.

1. Log into the Axis product by typing its IP address in a web browser. On the upper right corner click on "Setup".
2. Go to VoIP found on the left menu and press Account Settings.
3. Press Add button to add new SIP account on the device.
4. A new popup window will open, asking for entering the data of the account that will be added.

**Add Account**

General Network Video

**Account Information**

Name:

Account enabled

Default account (Note that only one account can be the default account.)

**Account Credentials**

User ID:  This field is required.

Use User ID as Authentication ID

Authentication ID:  This field is required.

Password:

Caller ID:

**SIP Server Settings**

IPv6 is not enabled

Domain name:

Registrar address:

Secondary SIP Server

**Incoming SIP Calls**

Auto answer

Save Cancel

Under the General tab, enter the following:

- Name – The name of the account shown in the Intercom and APIs.
- User ID – The ID associated with the device in WebEx. In this field type the first part, everything before the “@”, of the “Line ID” displayed in step 2.2.
- Authentication ID – The authentication ID associated with the device in WebEx. In this field type the “SIP username” displayed in step 2.2.
- Password – The password associated with the device in WebEx. In this field type the “SIP password” displayed in step 2.2.
- Domain – The domain address associated with the WebEx Control hub. In this field type the second part, everything after the “@”, of the “Line ID” displayed in step 2.2.

Under the Network tab, enter the following:

- Transport mode – What protocol to use when making calls. Each transport can be used however TLS is highly recommended due to being secure.
- Media encryption – How the media shall be encrypted (SRTP).
- Certificate – Certificate to use when verifying the client.
- Verify server certificate – If enabled the Axis product will verify the server.
- SIPS – If enabled will use SIPS instead of SIP.

**Add Account**

General **Network** Video

**Transport Settings**

Prioritize IPv6

Enable SIPS

Transport mode: TLS

Media encryption: SRTP Mandatory TLS transport mode required to ensure security

Allow port update messages through MWI

**Certificates**

[If no certificates are available go to certificates to manage.](#)

Certificate: self-signed

Verify server certificate

**Proxy Settings**

Address	Username

Add...

Save Cancel

5. If outbound proxy is needed, under the network tab, there is a section for the Proxy Settings, press the Add button on that section.

Allow port update messages through MWI

**Proxy Settings**

Address	Username

Address:

Username:

Password:

Apply Dismiss

Save Cancel

Three new input fields will appear:

- Address - Enter the outbound proxy you got from Webex (ex. hs17.hosted-int.bclid.webex.com)
- Username - Username used during authentication. Type in this field the first part, everything before the "@", of the "Line ID" displayed in step 2.2.

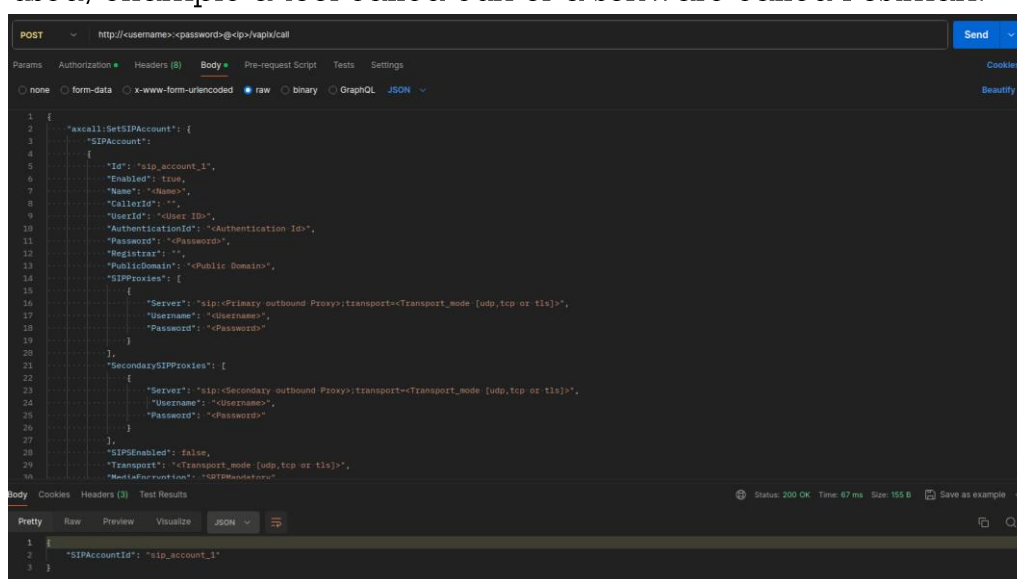
- Password - Password used during authentication. Type in this field the text shown under "SIP password" displayed in step 2.2.

Press Apply

6. For adding the account, press Save
7. If a secondary outbound proxy is needed, it can be added using the API.

## 4.4. Register a new SIP account using the API

For sending HTTP/HTTPS API requests, different third-party tools can be used, example a tool called curl or a software called Postman.





2. Select "raw" from the drop down menu.
3. Choose "JSON" (application/json) from the second dropdown menu.
4. Paste the following JSON code into the text area, replacing placeholders with your desired values:

```
{
  "axcall:SetSIPAccount": {
    "SIPAccount": {
      {
        "Id": "sip_account_1",
        "Enabled": true,
        "Name": "<Name>",
        "CallerId": "",
        "UserId": "<User ID>",
        "AuthenticationId": "<Authentication Id>",
        "Password": "<Password>",
        "Registrar": "",
        "PublicDomain": "<Public Domain>",
        "SIPProxies": [
          {
            "Server": "sip:<Primary outbound Proxy>;transport=<Transport_mode [udp,tcp or tls]>",
            "Username": "<Username>",
            "Password": "<Password>"
          }
        ],
        "SecondarySIPProxies": [
          {
            "Server": "sip:<Secondary outbound Proxy>;transport=<Transport_mode [udp,tcp or tls]>",
            "Username": "<Username>",
            "Password": "<Password>"
          }
        ],
        "SIPSEnabled": false,
        "Transport": "<Transport_mode [udp,tcp or tls]>",
        "MediaEncryption": "SRTPMandatory",
        "Certificate": "<Certificate_Name>",
        "VerifyServerCertificate": false
      }
    }
  }
}
```

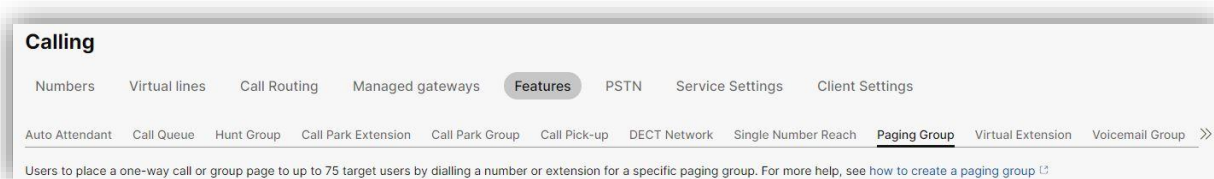
6. Send the request:
  1. Click the "Send" button located at the top right corner of the Postman window.
  2. Postman will send the request and display the response from the server in a new tab. The response will contain a status code (indicating success or failure) and any response body sent by the server.

# 5. Paging groups

Once the Axis products are registered an optional option is to create a paging group in WebEx. A paging group allows multiple products to be called simultaneously through WebEx.

## 5.1. Creating paging group

1. In the WebEx Control Hub navigate to "Calling" under "Services" on the menu to the left.
2. Click on "Features" and then on "Paging Group".



3. Click on "Create Paging Group". In the new window, select where the paging group shall be located. Give the group a new and associate a phone number or extension that when called, will call everyone within the group. Add a language to be used and the ID that will be shown on the receivers' side when the group is called.

Location  
Assign your paging group to a location.  
SI10AxisCom  
ⓘ The selected location is not associated with calling.

Paging Group Name  
The name is used to default Caller ID and reference the paging group later in the process.  
IT

Phone Number  
Assign the Paging Group to a Webex Calling primary line. A phone number and/or extension is required.  
Search and/or 8000

Language  
Select the paging group language  
Swedish (Sweden)

Calling Line ID  
This ID displays on the target user's phones when a group page is performed.

Calling ID First Name      Calling ID Last Name

Calling ID Label  
This determines what is shown on a paging target user's caller ID when a group page is performed

Paging Group ID ⓘ  
 Page Originator ⓘ

4. Add the users to be called when calling the paging group. Click on "Next".

Paging Targets ⓘ

Add users, workspaces or virtual lines

Search for and add up to 75 users, workspaces or virtual lines by name, phone number or extension. Click the name to view more details.

Add a user, workspace or virtual line

Name ^	Phone Number	Extension	
AC User8	+17138614008	4008	
AC User9	+17138614009	4009	
AC User3	+17134614003	4003	
AC User1	+17138614001	4001	

4/75 users

Copy my paging targets to my paging originators

5. Add the users that should be able to call the paging group. Users can simultaneously be targets and originators. Click on "Next".

Paging Originators ⓘ

Add users, workspaces or virtual lines

Search for and add users, workspaces or virtual lines by name, phone number or extension.

Add a user, workspace or virtual line

Name ^	Phone Number	Extension	
AC User6	+17138614006	4006	
AC User3	+17134614003	4003	
AC User10	+17138614010	4010	

3 users

6. Review the information displayed and if correct click on "Create" to finalize the creation of the paging group.