AXIS C6110 Network Paging Console

User manual

Table of Contents

Installation

The following video shows an example of how you can install an AXIS C6110 Network Paging Console together with an AXIS TC6901 Gooseneck Microphone.

For complete instructions on all installation scenarios as well as important safety information, see the installation guide on *axis.com/products/axis-c6110/support*.

To watch this video, go to the web version of this document.

Get started

Find the device on the network

To find Axis devices on the network and assign them IP addresses in Windows®, use AXIS IP Utility or AXIS Device Manager. Both applications are free and can be downloaded from *axis.com/support*.

For more information about how to find and assign IP addresses, go to *How to assign an IP address and access your device*.

Browser support

You can use the device with the following browsers:

	Chrome™	Edge™	Firefox®	Safari®
Windows®	✓	✓	*	*
macOS®	✓	✓	*	*
Linux®	✓	✓	*	*
Other operating systems	*	*	*	*

✓: Recommended

*: Supported with limitations

Open the device's web interface

1. Open a browser and type the IP address or host name of the Axis device.
   If you don't know the IP address, use AXIS IP Utility or AXIS Device Manager to find the device on the network.
2. Type the username and password. If you access the device for the first time, you must create an administrator account. See *Create an administrator account, on page 4*.

For descriptions of all features and settings in the web interface of devices with AXIS OS, see *AXIS OS web interface help*.

Create an administrator account

The first time you log in to your device, you must create an administrator account.

1. Enter a username.
2. Enter a password. See *Secure passwords, on page 5*.
3. Re-enter the password.
4. Accept the license agreement.
5. Click **Add account**.

Important

The device has no default account. If you lose the password for your administrator account, you must reset the device. See *Reset to factory default settings, on page 21*.

Secure passwords

Important

Use HTTPS (which is enabled by default) to set your password or other sensitive configurations over the network. HTTPS enables secure and encrypted network connections, thereby protecting sensitive data, such as passwords.

The device password is the primary protection for your data and services. Axis devices do not impose a password policy as they may be used in various types of installations.

To protect your data we strongly recommend that you:

- Use a password with at least 8 characters, preferably created by a password generator.
- Don't expose the password.
- Change the password at a recurring interval, at least once a year.

Make sure that no one has tampered with the device software

To make sure that the device has its original AXIS OS, or to take full control of the device after a security attack:

1. Reset to factory default settings. See *Reset to factory default settings, on page 21*.
   After the reset, secure boot guarantees the state of the device.
2. Configure and install the device.

Configure your device

Set up direct SIP (P2P)

Use peer-to-peer when the communication is between a few user agents within the same IP network and there is no need for extra features that a PBX-server could provide. To better understand how P2P works, see *Peer-to-peer SIP (P2PSIP), on page 14*.

For more information about setting options, see .

1. Go to **System** > **SIP** > **SIP settings** and select **Enable SIP**.
2. To allow the device to receive incoming calls, select **Allow incoming calls**.
3. Under **Call handling**, set the timeout and duration for the call.
4. Under **Ports**, enter the port numbers.
   - **SIP port** – The network port used for SIP communication. The signaling traffic through this port is non-encrypted. The default port number is 5060. Enter a different port number if required.
   - **TLS port** – The network port used for encrypted SIP communication. The signaling traffic through this port is encrypted with Transport Layer Security (TLS). The default port number is 5061. Enter a different port number if required.
   - **RTP start port** – Enter the port used for the first RTP media stream in a SIP call. The default start port for media transport is 4000. Some firewalls might block RTP traffic on certain port numbers. A port number must be between 1024 and 65535.
5. Under **NAT traversal**, select the protocols you want to enable for NAT traversal.

Note

Use NAT traversal when the device is connected to the network from behind a NAT router or a firewall. For more information see *NAT traversal, on page 15*.

6. Under **Audio**, select at least one audio codec with the desired audio quality for SIP calls. Drag-and-drop to change the priority.
7. Under **Additional**, select additional options.
   - **UDP-to-TCP switching** – Select to allow calls to switch transport protocols from UDP (User Datagram Protocol) to TCP (Transmission Control Protocol) temporarily. The reason for switching is to avoid fragmentation, and the switch can take place if a request is within 200 bytes of the maximum transmission unit (MTU) or larger than 1300 bytes.
   - **Allow via rewrite** – Select to send the local IP address instead of the router's public IP address.
   - **Allow contact rewrite** – Select to send the local IP address instead of the router's public IP address.
   - **Register with server every** – Set how often you want the device to register with the SIP server for the existing SIP accounts.
   - **DTMF payload type** – Changes the default payload type for DTMF.
8. Click **Save**.

Set up SIP through a server (PBX)

Use a PBX-server when user agents will communicate within and outside the IP network. Additional features could be added to the setup depending on the PBX-provider. To better understand how P2P works, see *Private Branch Exchange (PBX), on page 14*.

For more information about setting options, see .

1. Request the following information from your PBX provider:

- User ID
- Domain

- Password
- Authentication ID
- Caller ID
- Registrar
- RTP start port

2. To add a new account, go to **System** > **SIP** > **SIP accounts** and click **+ Account**.
3. Enter the details you received from your PBX provider.
4. Select **Registered**.
5. Select a transport mode.
6. Click **Save**.
7. Set up the SIP settings the same way as for peer-to-peer. See *Set up direct SIP (P2P), on page 6* for more information.

Add contacts and recipient devices

To add contacts, open the web interface by entering the IP address of your paging console in a web browser.

Note

Only recipients of the type "contacts" will appear in the contact list on the display of your AXIS C6110 Network Paging Console.

Recipients of the type "device" will not show up in the contact list, but you can configure a button on the display to target the device directly.

Note

Only VAPIX devices can be used for recipient groups.

Add an individual device as a recipient:

1. Go to **Communication** > **Recipients** > **Devices**.
2. Click **Add device**.
3. Enter the details and click **Save**.
   For information about the options under **Protocol**, see .

Add an individual person as a recipient:

1. Go to **Communication** > **Recipients** > **Contacts**.
2. Click **Add contact**.
3. Enter the details and click **Save**.
   For information about the options under **Protocol**, see .

Create a group of VAPIX recipients:

1. Go to **Communication** > **Recipients** > **Groups**.
2. Click **Add group**.
3. Enter the details and click **Save**.

Configure buttons, folders and pages

To configure buttons and folders, open the web interface by entering the IP address of your paging console in a web browser.

Create a new button or folder:

1. Go to the location where you want to add the button or folder.
   This will either be on the **Home** view or inside one of your folders.
2. Click a white button.
   White color indicates that the button has not been configured.

3. Select if you want to create an action or a folder.

Note

If you have a view that is located deep down in the folder structure, a good practice is to add a **Home** button that makes it easy to return to the home view.

4. Enter the details and click **Save**.

Edit or delete an existing button or folder:

- Click ⋮ and select **Edit** or **Delete**.

Rename the home view title:

1. Click ⋮ next to the home view title.
2. Select **Rename title**.
3. Enter the new title and click **Save**.

Add a new page:

- Click **Add page**.
  This will add a page to the same location, i.e. at the **Home** view or inside the current folder.

Note

If you create many pages, a good practice is to add a **Home** button that makes it easy to return to the home view.

You can add up to 10 pages per folder.

Configure a button for two-way VAPIX paging

1. Create a VAPIX recipient:
   1.1. Go to **Communication > Recipients**.
   1.2. If you want to add a device, go to **Devices**.
   If you want to add a contact, go to **Contacts**.
   1.3. Click **+ Add device** or **+ Add contact**.
   1.4. Name your recipient.
   1.5. Under **Protocol**, select **VAPIX**.
   1.6. Enter the IP address for the recipient.
   1.7. Enter the user name and password for the recipient.
   1.8. Click **Save**.
2. Create a two-way action:
   2.1. Go to **Display > Configuration > Actions**.
   2.2. Click **+ Add action**.
   2.3. Under **Action**, select **Two-way**.
   2.4. Under **Contact**, select your VAPIX recipient.
   2.5. Click **Save**.
3. Configure a button:
   3.1. Go to **Display > Configuration > Buttons**.
   3.2. Click an available button.
   3.3. Under **Select button type**, select **Action**.
   3.4. Under **Select an action to be triggered by the button**, select **Use an existing action**.
   3.5. Click the row of your two-way action in the list.

3.6. Click **Save**.

When you press the configured button on your AXIS C6110 Paging Console, a two-way VAPIX call will be made to the recipient.

Remember that the microphone on the recipient device must be activated. Enable echo cancellation on the recipient device to enhance the quality of the two way call. See .

Use AXIS Audio Manager Edge to configure a button for one-way paging

You can use AAM Edge to configure a button on the C6110 to page one or several physical zones.

1. Open AXIS Audio Manager Edge.
2. Create a paging recipient.
3. Open the web interface
4. set it to be one-way
5. assign the desired zone/zones
6. Open the recipient to see which Intermediary device has been chosen
7. Copy the IP adress of the intermediary device.
8. Return to the web interface of C6110.
9. Go to **Communication > Recipients > Devices** and click **+ Add device**
10. Give the Contact a name, select SIP as a protocol, enter the IP adress in the SIP Address field, and select the peer-to-peer account on the C6110.
11. Go to Display -> Configuration and add a new button.
12. Create new action -> Action: One way, Contact: The Contact created in the step above. Save the button.

Change the display settings

To change the display settings, open the web interface by entering the IP address of your paging console in a web browser.

- To adjust brightness, timers and presence detection, go to **Display settings** > **Display**.
- To adjust language and clock settings for the display of your paging console, go to **Display** > **Localization**.

For more information about the individual options, see .

Set up rules for events

You can create rules to make your device perform actions when certain events occur. A rule consists of conditions and actions. The conditions can be used to trigger the actions. For example, the device can play an audio clip according to a schedule or when it receives a call, or send an email if the device changes IP address.

To learn more, see *Get started with rules for events*.

Place and receive a call

Place a call

1. Navigate to the page on the display where the contact is located.

   Contacts are indicated by .

2. To place a call, press the button for the contact.
3. To mute or unmute your microphone, press the **Mute** or **Unmute** button.
4. To regulate the volume level of your speaker, press the volume button on the left side of the paging console.
5. To end the call, press the button for **Hang up**.

Receive a call

When you receive a call, the display shows **Incoming call** and a ringing signal is heard.

1. To answer the call, press the **Answer** button.
2. To hang up or reject the call, press the **Hang up** button.

If you have missed a call, is shown at the top right corner of the display. To see who called, press the **Call history** button.

Page a message

To page a one-way live callout:

1. Navigate to the page on the display where the target is located.
   The target can be an individual person or device, or a group. Targets are indicated by .
2. Press the button of the target.
3. Wait for the pre-announcement message to be played, if such message is configured for the target.
4. Press and hold the push-to-talk button, and speak your message.
5. When you are done, press **Cancel**.

Play an announcement

Play a pre-recorded audio file:

1. Navigate to the page on the display where the announcement is located.

   Announcements are indicated by .

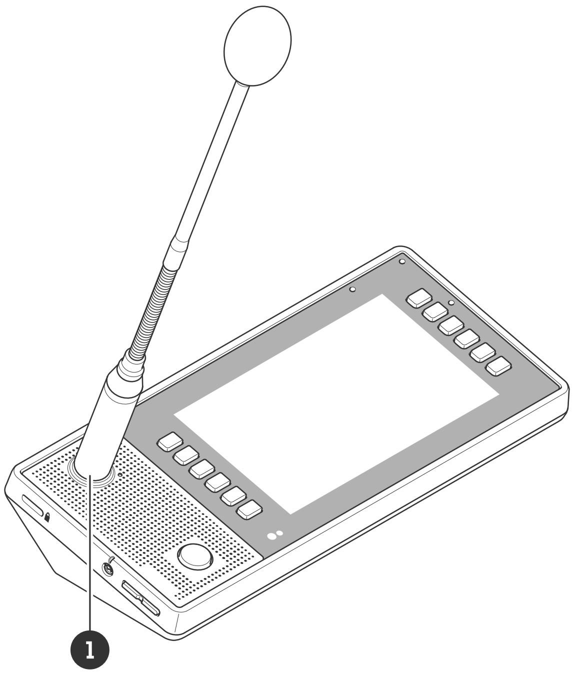
2. Press the button for the announcement.

Connect external equipment

Use an AXIS TC6901 Gooseneck Microphone

The AXIS TC6901 Gooseneck Microphone is an accessory that is sold separately.

For mounting instructions, see the installation guide for AXIS TC6901 Gooseneck Microphone.



1 AXIS TC6901 Gooseneck Microphone

To use a gooseneck microphone:

1. Open the web interface by entering the IP address of your paging console in a web browser.
2. Go to **Device settings**.
3. Set **Input type** to **Balanced microphone**.

Use a headset

You can connect a headset to the 3.5 mm audio connector located on the side of the AXIS C6110 Network Paging Console.

You can adjust the volume for the headset using the volume buttons.

If you connect headphones without a microphone, the internal microphone will stay active.

Learn more

Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) is used to set up, maintain and terminate VoIP calls. You can make calls between two or more parties, called SIP user agents. To make a SIP call you can use, for example, SIP phones, softphones or SIP-enabled Axis devices.

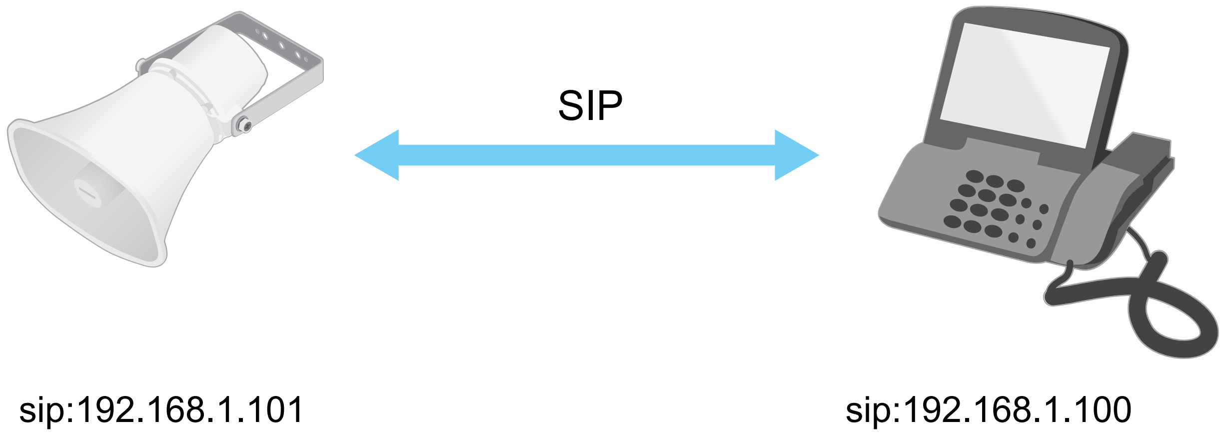
The actual audio or video is exchanged between the SIP user agents with a transport protocol, for example RTP (Real-Time Transport Protocol).

You can make calls on local networks using a peer-to-peer setup, or across networks using a PBX.

Peer-to-peer SIP (P2PSIP)

The most basic type of SIP communication takes place directly between two or more SIP user agents. This is called peer-to-peer SIP (P2PSIP). If it takes place on a local network, all that's needed are the SIP addresses of the user agents. A typical SIP address in this case would be `sip:<local-ip>`.

Example:



You can set up a SIP-enabled phone to call an audio device on the same network using a peer-to-peer SIP setup.

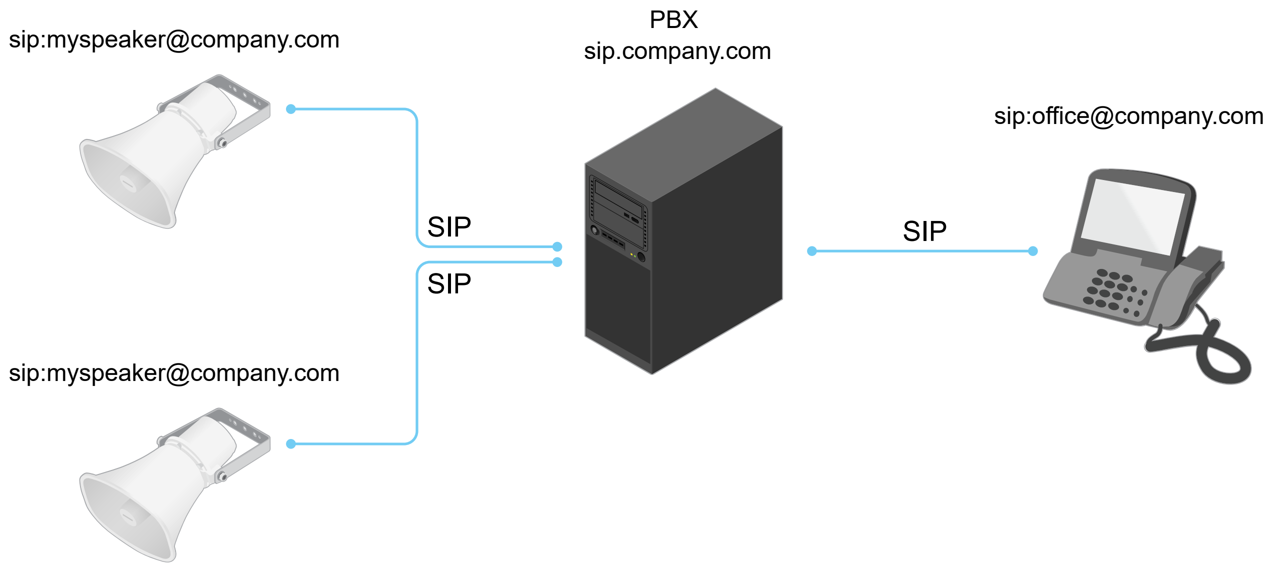
Private Branch Exchange (PBX)

When you make SIP calls outside your local IP network, a Private Branch Exchange (PBX) can act as a central hub. The main component of a PBX is a SIP server, which is also referred to as a SIP proxy or a registrar. A PBX works like a traditional switchboard, showing the client's current status and allowing for example call transfers, voicemail, and redirections.

The PBX SIP server can be set up as a local entity or offsite. It can be hosted on an intranet or by a third party provider. When you make SIP calls between networks, calls are routed through a set of PBXs, that query the location of the SIP address to be reached.

Each SIP user agent registers with the PBX, and can then reach the others by dialing the correct extension. A typical SIP address in this case would be `sip:<user>@<domain>` or `sip:<user>@<registrar-ip>`. The SIP address is independent of its IP address and the PBX makes the device accessible as long as it is registered to the PBX.

Example:



NAT traversal

Use NAT (Network Address Translation) traversal when the Axis device is located on an private network (LAN) and you want to access it from outside of that network.

Note

The router must support NAT traversal and UPnP®.

Each NAT traversal protocol can be used separately or in different combinations depending on the network environment.

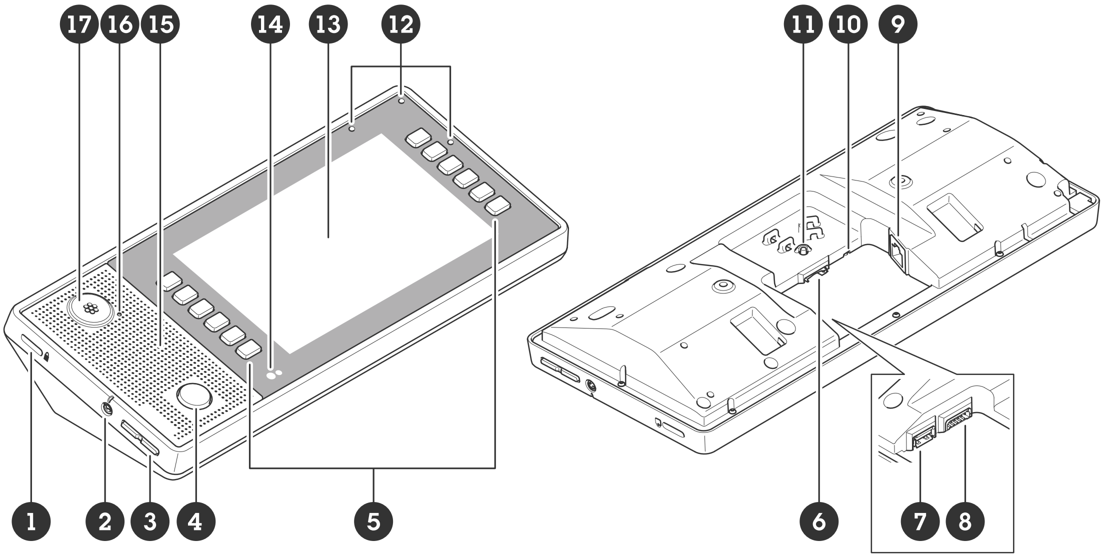
- **ICE** The ICE Interactive Connectivity Establishment) protocol increases the chances of finding the most efficient path to successful communication between peer devices. If you also enable STUN and TURN, you improve the ICE protocol's chances.
- **STUN** - STUN (Session Traversal Utilities for NAT) is a client-server network protocol that lets the Axis device determine if it is located behind a NAT or firewall, and if so obtain the mapped public IP address and port number allocated for connections to remote hosts. Enter the STUN server address, for example, an IP address.
- **TURN** - TURN (Traversal Using Relays around NAT) is a protocol that lets a device behind a NAT router or firewall receive incoming data from other hosts over TCP or UDP. Enter TURN server address and the login information.

The web interface

To read about all the features and settings available in the web interface of devices with AXIS OS, go to *AXIS OS web interface help*.

Specifications

Product overview



1 Security slot
2 Headset connector (3.5 mm audio connector)
See Audio connector, on page 18
3 Volume buttons
4 Push-to-talk button
5 Soft keys
6 SD card slot, on page 18
7 USB connector (not in use)
8 I/O connector, on page 19
9 Network connector, on page 18 (PoE)
10 Status LED
11 Control button, on page 18
12 Built-in beamforming microphone
13 7-inch color display
14 Light and presence sensor
15 Speaker
16 Microphone status LED
17 XLR connector for gooseneck microphone
The connector is placed underneath the cover, which is replaced if you connect a gooseneck microphone. For more information, see XLR connector, on page 18

LED indicators

Status LED	Indication
Unlit	Unlit for normal operation.
Green	Steady for 10 seconds for normal operation after startup completed.
Amber	Steady during startup. Flashes during device software upgrade or reset to factory default.
Amber/Red	Flashes if network connection is unavailable or lost.
Red	Flashes slowly if upgrade failed.
Red/Green	Flashes fast when **Locate device** is selected.

SD card slot

NOTICE

- Risk of damage to SD card. Don't use sharp tools, metal objects, or excessive force when inserting or removing the SD card. Use your fingers to insert and remove the card.
- Risk of data loss and corrupted recordings. Unmount the SD card from the device's web interface before removing it. Don't remove the SD card while the product is running.

For SD card recommendations, see *axis.com*.

SD, SDHC, and SDXC Logos are trademarks of SD-3C LLC. SD, SDHC and SDXC are trademarks or registered trademarks of SD-3C, LLC in the United States, other countries or both.

Buttons

Control button

The control button is used for:

- Calibrating the speaker test. Press and release the control button and a test tone is played.
- Resetting the product to factory default settings. See *Reset to factory default settings, on page 21*.

Connectors

Network connector

RJ45 Ethernet connector with Power over Ethernet (PoE).

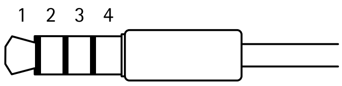
NOTICE

The device shall be connected using a shielded network cable (STP). All cables connecting the device to the network shall be intended for their specific use. Make sure that the network devices are installed in accordance with the manufacturer's instructions. For information about regulatory requirements, see the Installation Guide at *www.axis.com*.

Audio connector

3.5 mm input/output connector for headset (4-pole TRRS) or headphone (3-pole TRS).

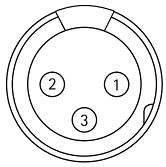
Audio input/output for headset (standard)



1 Tip	2 Ring	3 Ring	4 Sleeve
Channel 1, unbalanced line, mono	Channel 1, unbalanced line, mono	Ground	Micro-phone
Balanced line, "hot" signal	Balanced line, "cold" signal	Ground	Micro-phone
Stereo unbalanced line, "left"	Stereo unbalanced line, "right"	Ground	Micro-phone
Channel 1, unbalanced line	Channel 2, unbalanced line	Ground	Micro-phone

XLR connector

For more information, see *Use an AXIS TC6901 Gooseneck Microphone, on page 13*



Pin	1	2	3
Function	Ground	Balanced Microphone Hot (+) In	Balanced Microphone Cold (-) In

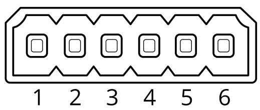
I/O connector

Use the I/O connector with external devices in combination with, for example, motion detection, event triggering, and alarm notifications. In addition to the 0 VDC reference point and power (12 V DC output), the I/O connector provides the interface to:

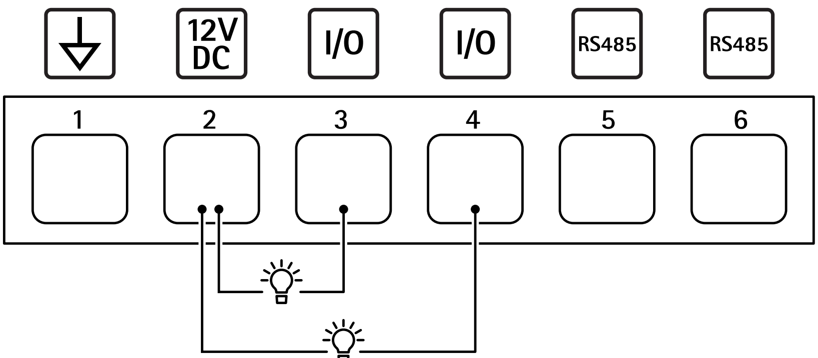
Digital input – For connecting devices that can toggle between an open and closed circuit, for example PIR sensors, door/window contacts, and glass break detectors.

Digital output – For connecting external devices such as relays and LEDs. Connected devices can be activated by the VAPIX® Application Programming Interface, through an event or from the device's web interface.

6-pin terminal block



Function	Pin	Notes	Specifications
DC ground	1		0 V DC
DC output	2	Can be used to power auxiliary equipment. Note: This pin can only be used as power out.	12 V DC Max load = 25 mA
Digital I/O	3	Connect to pin 1 to activate, or leave floating (unconnected) to deactivate.	0 to max 30 V DC
Digital I/O	4	Internally connected to pin 1 (DC ground) when active, and floating (unconnected) when inactive. If used with an inductive load, e.g., a relay, connect a diode in parallel with the load, to protect against voltage transients.	0 to max 30 V DC, open drain, 100 mA
RS485	5	RS485: A+	
RS485	6	RS485: B+	



1. *DC ground*
2. *DC output 12 V, max 50 mA*
3. *Digital I/O*
4. *Digital I/O*
5. *Configurable I/O (RS485)*
6. *Configurable I/O (RS485)*

Troubleshooting

Reset to factory default settings

Important

Reset to factory default should be used with caution. A reset to factory default resets all settings, including the IP address, to the factory default values.

To reset the product to the factory default settings:

1. Disconnect power from the product.
2. Press and hold the control button while reconnecting power. See *Product overview, on page 17*.
3. Keep the control button pressed for 10 seconds until the status LED indicator turns amber for the second time.
4. Release the control button. The process is complete when the status LED indicator turns green. If no DHCP server is available on the network, the device IP address will default to one of the following:
   - **Devices with AXIS OS 12.0 and later:** Obtained from the link-local address subnet (169.254.0.0/16)
   - **Devices with AXIS OS 11.11 and earlier:** 192.168.0.90/24
5. Use the installation and management software tools, assign an IP address, set the password, and access the product.

You can also reset parameters to factory default through the device's web interface. Go to **Maintenance** > **Factory default** and click **Default**.

Contact support

If you need more help, go to *axis.com/support*.

T10201145

2026-02 (M16.2)