



## TechNote: AXIS and CyberGate

Version: 1.0.2 ENG  
Date: 26-08-2022



**Configure the AXIS Network Video  
Door Station for CyberGate**

## CyberGate

Microsoft Teams is the hub for team collaboration in Microsoft Office 365 that integrates people, content, conversations and tools your team needs. Via the CyberGate application that runs in Microsoft Azure you can now connect an AXIS Network Video Door Station to your Microsoft Teams environment. Microsoft Teams users can answer incoming calls from- or place outgoing calls to the intercom – with 2-way audio and live video – on the Teams desktop client, Teams desk phone or Teams Smartphone app and open the door for visitors.

***! Note: For instructions on how to purchase and configure the CyberGate service, see our Tech Note: 'Connect a SIP Intercom to MS Teams using the CyberGate service'. !***

## AXIS Network Video Doorstation

For this document the AXIS A8105-E Network Video Doorstation (from now on named 'AXIS') is used to connect to the CyberGate service (from now on named 'CyberGate').

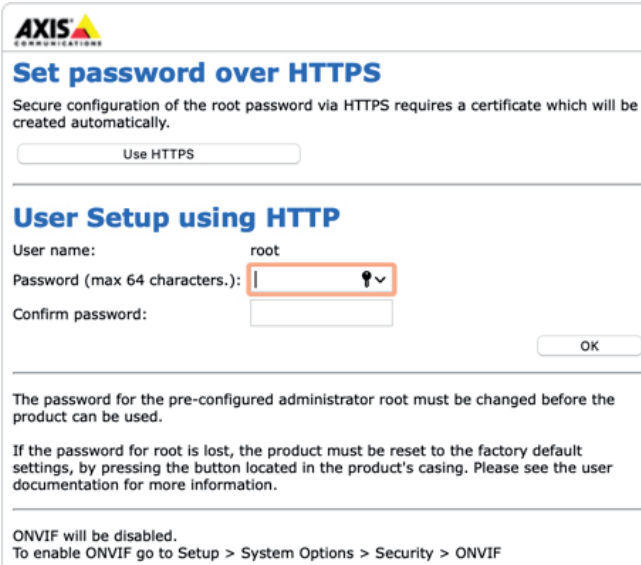
This manual also applies to the following Axis Video Doorstations:

- Axis A8004-VE
- Axis A8207-VE
- Axis I8016-LVE

Follow the next steps to configure the AXIS to connect it to CyberGate.

### Connect the AXIS

Connect the AXIS to the network, power it on and open a webbrowser to its IP-address. Create a password for the 'root' user and click 'OK'



The screenshot shows the AXIS web interface. At the top, there is the AXIS logo and the title "Set password over HTTPS". Below this, a message states: "Secure configuration of the root password via HTTPS requires a certificate which will be created automatically." There is a button labeled "Use HTTPS".

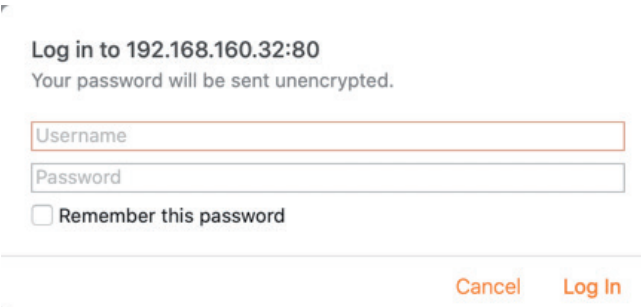
The next section is titled "User Setup using HTTP". It contains the following fields and text:

- User name: root
- Password (max 64 characters.): [input field with a key icon]
- Confirm password: [input field]
- OK button

Below the form, there is a warning: "The password for the pre-configured administrator root must be changed before the product can be used." Another warning follows: "If the password for root is lost, the product must be reset to the factory default settings, by pressing the button located in the product's casing. Please see the user documentation for more information."

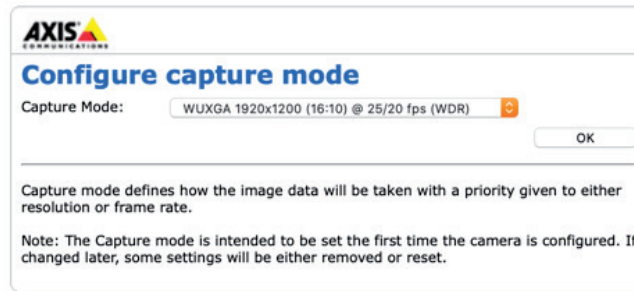
At the bottom, it states: "ONVIF will be disabled. To enable ONVIF go to Setup > System Options > Security > ONVIF"

Login as 'root' with the just created password

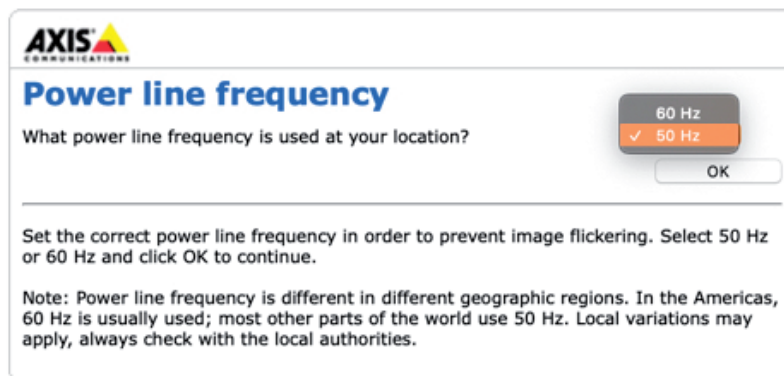


The screenshot shows a login page titled "Log in to 192.168.160.32:80". Below the title, it says "Your password will be sent unencrypted." There are two input fields: "Username" and "Password". Below the "Password" field, there is a checkbox labeled "Remember this password". At the bottom right, there are two buttons: "Cancel" and "Log In".

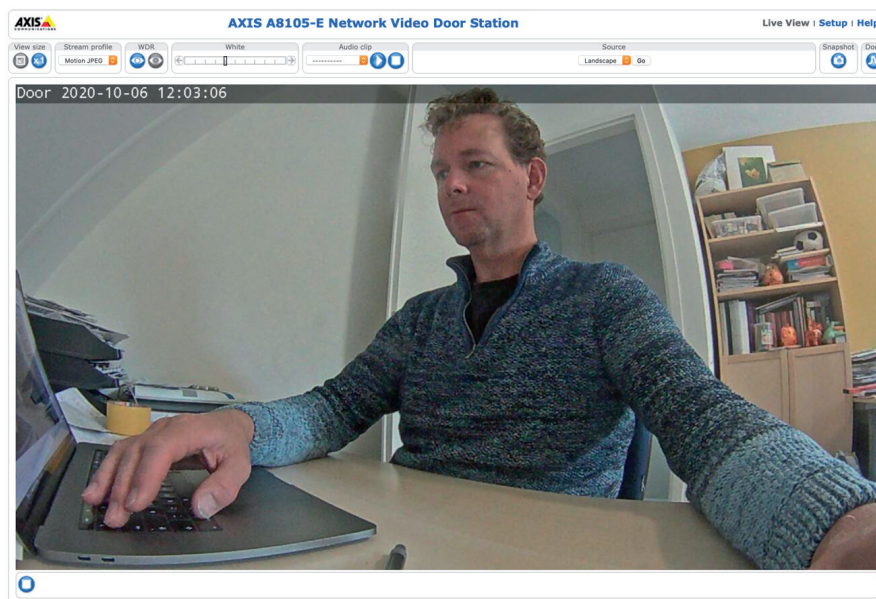
Set the Capture Mode:



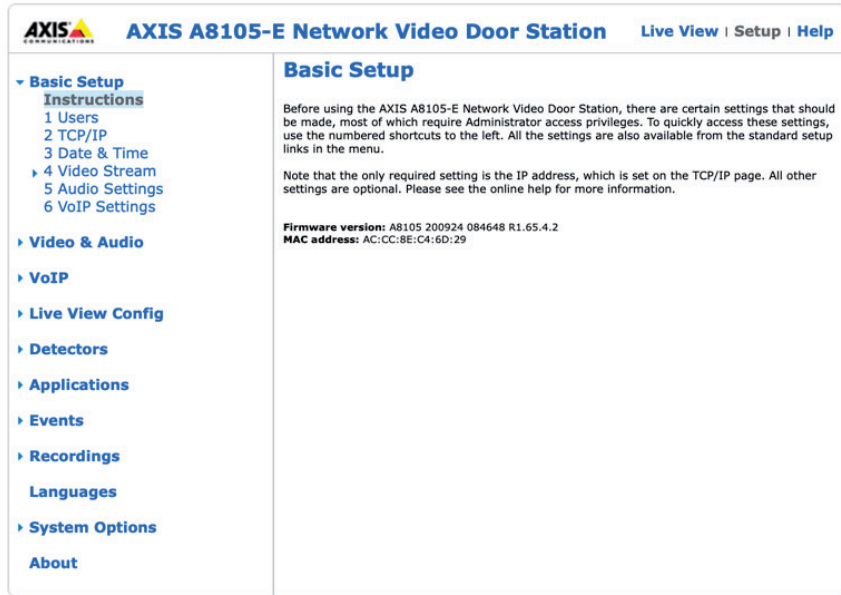
Set the used power line frequency:



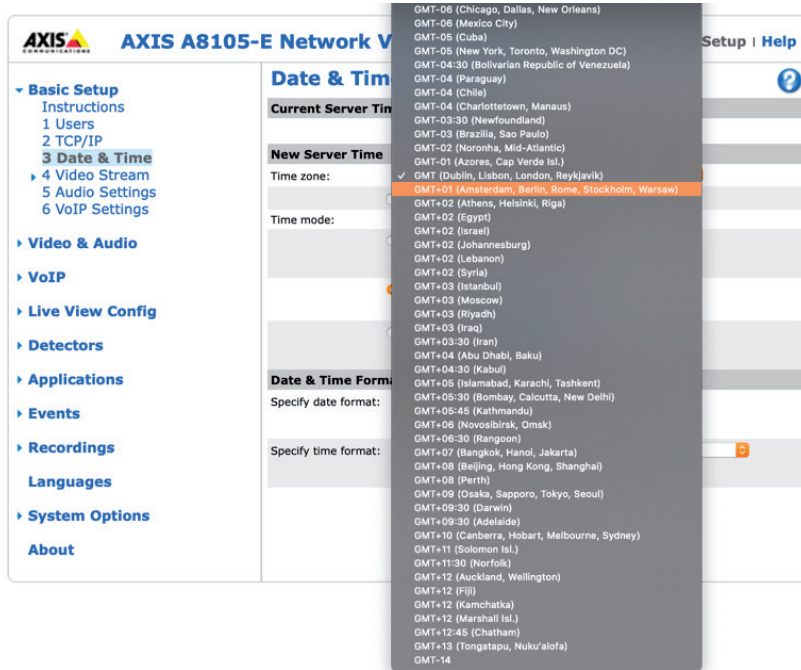
When logged-in, the live view opens



At the top, click 'Setup'



Click on 'Date& Time' to set the correct Time zone.



Click the Save button to confirm.

Click on 'VoIP Settings' and at Account settings 'Configure'

The screenshot shows the 'VoIP Overview' page. On the left is a navigation menu with categories like Basic Setup, Video & Audio, VoIP, Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. The 'VoIP' category is expanded, showing sub-items: Overview, SIP Settings, VMS Settings, Account Settings, and DTMF Settings. The main content area is titled 'VoIP Overview' and includes a 'SIP Setup Assistant' button, 'SIP Settings' (SIP: Disabled, Allow incoming SIP calls: Disabled, SIP port: 5060, SIP TLS port: 5061), 'VoIP Action Rules' table, and 'Account Settings' table.

Name	Trigger	Action	Recipient	Schedule
AUDIO: Calling	Call - State	Play Audio Clip	-	-
AUDIO: Stop on Active call	Call - State	Stop Audio Clip	-	-
AUDIO: Stop on Idle call	Call - State	Stop Audio Clip	-	-
BUTTON: VMS call	Input Signal - Digital Input Port	Make Call	VMS	-
LIGHT: Active call	Call - State	Activate Light on Active	-	-
LIGHT: Calling	Call - State	Activate Light on Calling	-	-

Name	SIP address	Reg. status	Call status
peer-to-peer	sip:192.168.160.32	<input type="checkbox"/>	<input type="checkbox"/>

Remove the default 'peer-to-peer entry with 'Remove', after that add a new entry by clicking on 'Add'

The screenshot shows the 'Account Settings' page. The left navigation menu is similar to the previous screenshot, but 'Account Settings' is highlighted. The main content area is titled 'Account Settings' and features a table with columns: Name, SIP address, Transport, Default, and Reg. status. Below the table are 'Add...', 'Modify...', and 'Remove' buttons. A 'Test SIP Call' section contains a note: 'Note that SIP is disabled. Enable SIP in the SIP Settings configuration.'

Name	SIP address	Transport	Default	Reg. status
peer-to-peer	sip:192.168.160.32	UDP	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Provide the following information:

Name	Name of the SIP account, in this example 'CyberGate'
Default account	Enable by checking the checkbox
User ID	Use the Username provided by the CyberGate service
Authentication ID	Will be the same as the User ID
Password	Use the Password provided by the CyberGate service
Caller ID	Use the Username provided by the CyberGate service
Domain name	Use the domain name of your organisation
Register address	cybergate.cybertwice.com

Click the Save button to confirm.

### 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

Click 'Network' to set the SIP connection parameters

By default it uses UDP to connect to CyberGate. You can change it to TCP when needed.

**Modify Account** ?

**General** **Network** **Video**

**Transport Settings**

Prioritize IPv6

Enable SIPS

Transport mode:

Media encryption:

Allow port update messages through MWI

**Proxy Settings**

Address	Username

↑  
↓

**Account Status**

Reg. status: ● OK (200)

To establish a secure connection to CyberGate, modify the following settings:

Transport Mode	Change to TLS
Media Encryption	Change this to SRTP Mandatory

Click the Save button to confirm.

These settings ensure that the SIP traffic and the audio / video are encrypted.

Click 'Video' to set the video parameters for SIP  
Set the Resolution to 720p and set the View area to Landscape.

### Modify Account ?

**General** **Network** **Video**

**Video Settings:**

Resolution:  pixels

Frame rate:

View area:

**Account Status**

Reg. status: ● OK (200)

### Modify Account ?

**General** **Network** **Video**

**Video Settings:**

Resolution:  pixels

Frame rate:

View area:

**Account Status**

Reg. status: ● OK (200)

Click the Save button to confirm.

Click on 'SIP Settings' and enable SIP by checking 'Enable SIP':

The screenshot shows the configuration interface for the AXIS A8105-E Network Video Door Station. The page is titled "SIP Settings" and includes a sidebar with navigation options: Basic Setup, Video & Audio, VoIP (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 is divided into several sections:

- SIP Setup Assistant:** A section with a "Start..." button to begin the configuration process.
- SIP Settings:** A section where the "Enable SIP" checkbox is checked.
- Calling timeout (seconds):** A text input field set to 60.
- Incoming SIP Calls:** A section with an unchecked "Allow incoming SIP calls" checkbox.
- Port Settings:** A section with three text input fields: "SIP port:" (5060), "SIP TLS port:" (5061), and "RTP start port:" (4000).
- NAT Traversal:** A section with a link "Enable IPv6 here" and three unchecked checkboxes: "Enable ICE", "Enable STUN", and "Enable TURN".
- Audio Codec Settings:** A section with two lists of codecs. The "Available codecs" list includes L16/8000 (8000 Hz), speex/16000 (16000 Hz), speex/8000 (8000 Hz), and G.726-32 (8000 Hz). The "Selected codecs" list includes PCMU (8000 Hz), PCMA (8000 Hz), opus (48000 Hz), and L16/16000 (16000 Hz). Arrows indicate the movement of codecs between the lists.
- Advanced SIP Settings:** A section with three unchecked checkboxes: "Disable Automatic UDP to TCP Switch", "Disable contact rewrite", and "Disable via rewrite". It also includes text input fields for "Max call duration (seconds):" (3600), "Registration Interval (seconds):" (300), and "DTMF payload type:" (96). There are also dropdown menus for "H.264 packetization mode:" (Auto), "Audio direction:" (Send and receive), and "Video direction:" (Send only).

A "Save" button is located at the bottom right of the configuration area.

Click the Save button to confirm.

Click on 'Events':

The screenshot shows the configuration interface for the AXIS A8105-E Network Video Door Station. The left sidebar contains a navigation menu with categories: Basic Setup, Video & Audio, VoIP, Live View Config, Detectors, Applications, Events (expanded), Recordings, Languages, System Options, and About. Under the 'Events' category, 'Action Rules' is selected. The main content area is titled 'Action Rules' and contains an 'Action Rule List' table. The table has columns for Name, Trigger, Schedule, Action, and Recipient. Below the table are buttons for 'Add...', 'Copy...', 'Modify...', and 'Remove'.

Name	Trigger	Schedule	Action	Recipient
<input checked="" type="checkbox"/> AUDIO: Calling	Call - State	-	Play Audio Clip	-
<input checked="" type="checkbox"/> AUDIO: Stop on Active call	Call - State	-	Stop Audio Clip	-
<input checked="" type="checkbox"/> AUDIO: Stop on Idle call	Call - State	-	Stop Audio Clip	-
<input checked="" type="checkbox"/> BUTTON: VMS call	Input Signal - Digital Input Port	-	Make Call	VMS
<input checked="" type="checkbox"/> LIGHT: Active call	Call - State	-	Activate Light	-
<input checked="" type="checkbox"/> LIGHT: Calling	Call - State	-	Activate Light	-
<input checked="" type="checkbox"/> LIGHT: Idle	Call - State	-	Activate Light	-
<input checked="" type="checkbox"/> TAMPERING: Shock detected	Detectors - Shock Detection	-	Output Port	-
<input checked="" type="checkbox"/> TAMPERING: Tilt detected	Detectors - Tilt Detection	-	Output Port	-

Click on 'Recipients' and 'Add' to add the recipient to call

The screenshot shows the configuration interface for the AXIS A8105-E Network Video Door Station. The left sidebar is the same as in the previous screenshot, but 'Recipients' is now selected under the 'Events' category. The main content area is titled 'Recipients' and contains a 'Recipients List' table. The table has columns for Name, Type, Address, Upload path, and User name. Below the table are buttons for 'Add...', 'View...', 'Copy...', and 'Remove'.

Name	Type	Address	Upload path	User name
------	------	---------	-------------	-----------

Provide the following information:

Name	Name of the recipient, in this example 'Teams User'
Type	Select 'SIP'
To SIP address	Use the Teams user address with after the @ cybergate.cybertwice.com instead of your own domain. (name.name@cybergate.cybertwice.com)

**Recipient Setup**

Name:

Type:

From SIP account:

To SIP address:

**Test**

Test the connection between the selected SIP account and the specified SIP address. The call will end automatically.

Select SIP account:

Click the Close button.

The recipient is added

**AXIS A8105-E Network Video Door Station**

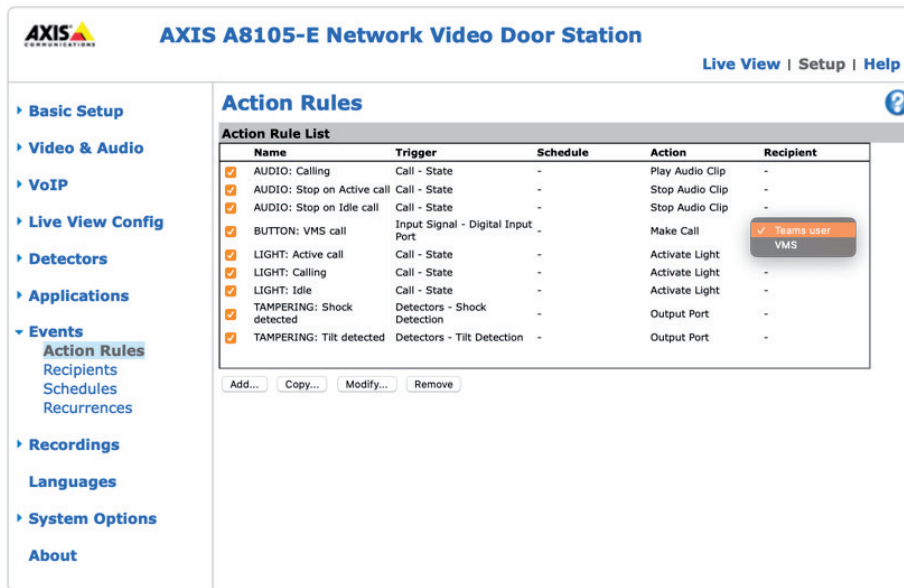
Live View | Setup | Help

**Recipients**

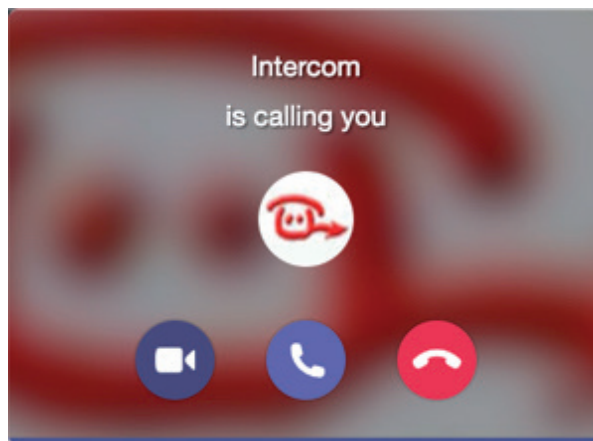
**Recipients List**

Name	Type	Address	Upload path	User name
Teams user	SIP	koos.ridder@cybergate.cybertwice.com	-	-

Click on 'Action Rules' and select the Teams user at the 'Button' action. Doing this will result in the calling of the Teams recipient when the button on the AXIS is pressed.



After this, press the button on the AXIS to call the Teams recipient. If configured correctly, the Teams client will notify you of an incoming call. Answer it by clicking the answer button.



The call will be established and video will be displayed within ±3 seconds.



To run configured actions (such as door opening) from the Teams call, click on the three dots (...) in the call screen and select the 'Keypad'. Use the dtmf code (as configured in the menu VoIP-DTMF Settings), this will trigger any action to use dtmf (as configured in the menu Events-Action Rules).

**Note:**

For instructions on how to configure DTMF on the AXIS A8105-E, please see page 40 of the AXIS A8105-E manual, downloadable on the CyberTwice support site ([Link](#)).

## Document History

Document Version	Date	Author	Change
1.0.0	06-10-2020	KR	Initial version
1.0.1	13-01-2021	KR	Added page numbers to the document and changed recipient SIP address (page 8)
1.0.2	26-08-2022	KR	Added Secure SIP chapter