



**Application Notes for configuring Axis Communications AB
AXIS A8004-VE Network Video Door Station with Avaya IP
Office Server Edition and IP Office IP500 V2 Expansion
R9.1 – Issue 1.1**

Abstract

These Application Notes describe the configuration steps for provisioning the AXIS A8004-VE Network Video Door Station from Axis Communications AB to interoperate with Avaya IP Office Server Edition and IP Office 500 V2 expansion R9.1.

Readers should pay particular attention to the scope of testing as outlined in **Section 2.1**, as well as observations noted in **Section 2.2** to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps for provisioning the AXIS A8004-VE Network Video Door Station from Axis Communications AB to interoperate with Avaya IP Office Server Edition and IP Office IP500 V2 expansion R9.1

AXIS A8004-VE Network Video Door Station is an open, non-proprietary IP-based door station for two-way communication, identification and remote entry control. It is a robust outdoor unit with a high performing intercom function providing clear, uninterrupted an echo-free speech also in the most demanding situations.

The unit supports Session Initiation Protocol (SIP) for easy integration with Avaya IP Office to meet advanced audio and video communication needs. AXIS A8004-VE is equipped with multiple inputs and outputs for remote control of door locks as well as other equipment.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of the AXIS A8004-VE Network Video Door Station (Axis Door Phone) to make and receive calls to and from Avaya Digital, H.323 and SIP desk phones as well as hunt groups and mobile/PSTN endpoints. Avaya Communicator for Windows was used to receive video calls from the AXIS A8004-VE.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

2.1. Interoperability Compliance Testing

The compliance testing included the test scenarios shown below. Note that when applicable, all tests were performed with Avaya SIP phones, H.323 phones Digital phones, and PSTN endpoints.

- Registration of door phone.
- Invalid usernames/passwords for registration.
- Basic calls.
- Video Calls using Avaya Communicator for Windows.
- Transfer/Conference/Forwarding.
- Codec support.
- DTMF support.
- Door opening.
- Serviceability testing.

2.2. Test Results

The following issue was noted during testing.

Using AXIS A8004-VE Network Video Door Station to overflow calls from one IP Office phone to another in the event of a “no answer” from the initial phone, the call fails to overflow and the second phone does not ring, the call remains indefinitely on the first phone. Note this same scenario works fine for a “busy” extension. The next release of software from Axis Communications should fix this issue.

2.3. Support

Support from Avaya is available by visiting the website <http://support.avaya.com> and a list of product documentation can be found in **Section 9** of these Application Notes. Technical support for the AXIS A8004-VE Network Video Door Station product can be obtained as follows:

Axis Communications AB

Tel: +46 46 272 18 00

Fax: +46 46 13 61 30

<http://www.axis.com/global/en/learning-and-support>

3. Reference Configuration

Figure 1 shows the network topology during compliance testing, an AXIS A8004-VE Network Video Door Station from Axis Communications AB with Avaya IP Office Server Edition.

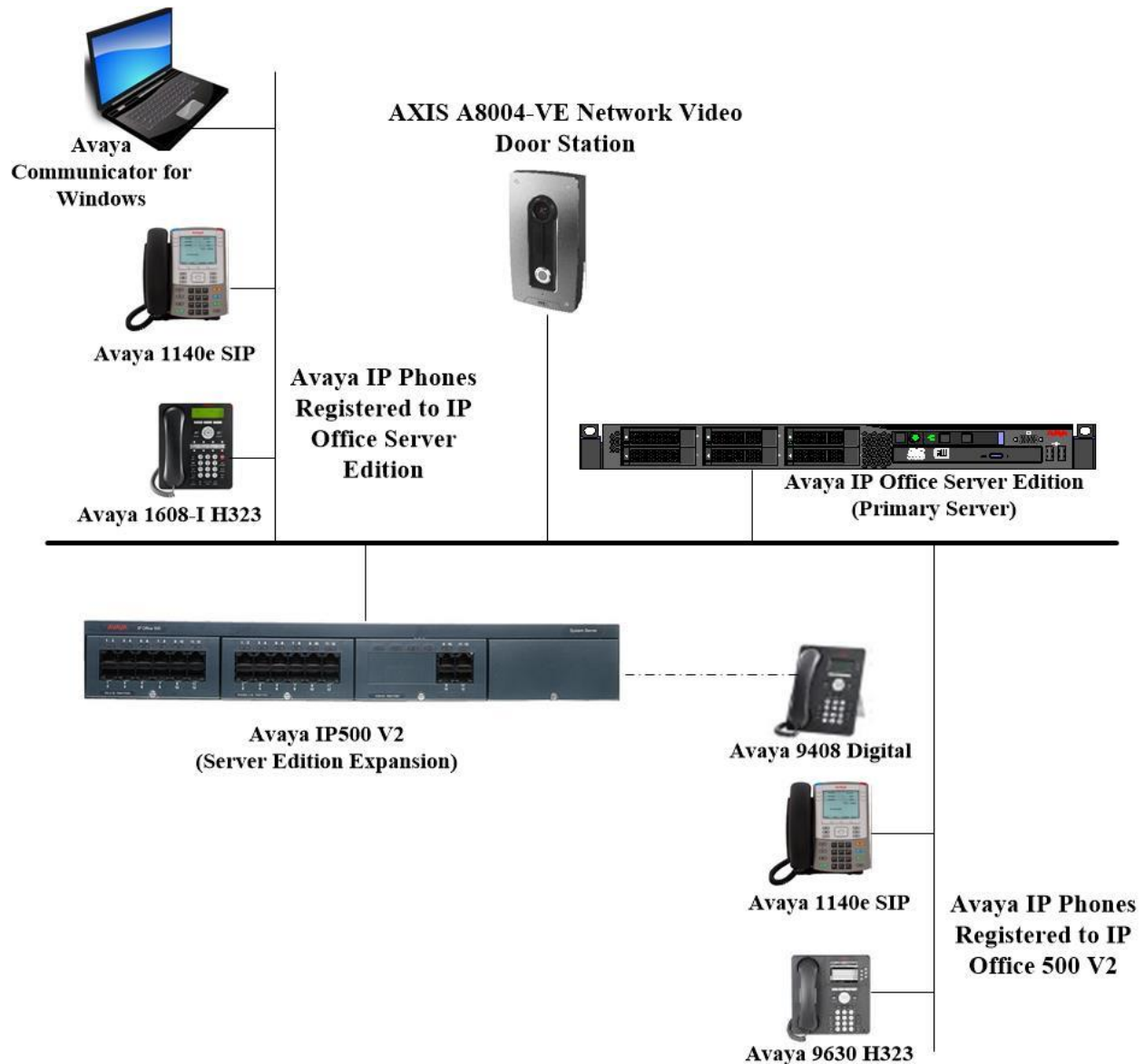


Figure 1: Connection of Axis Communications AB A8004-VE Network Video Door Station with Avaya IP Office Server Edition and IP Office IP500 V2 R9.1

4. Equipment and Software Validated

The following equipment and software was used for the compliance test.

Equipment/Software	Version/Release
Avaya IP Office Server Edition running on a virtual platform	R9.1 SP6
Avaya IP Office 500 V2	R9.1 SP6
Avaya IP Office Manager	R9.1 SP6
Avaya Communicator for Windows	SIP R2.0.3
Avaya 9630 Deskphone	H.323 Release 6.4014U
Avaya 1140e Deskphone	SIP R04.03.12.00
Avaya 1616-I Deskphone	H323 1608UA1_350B.bin
Avaya 9408 Digital Deskphone	V2.0
Axis Communications AB AXIS A8004-VE Network Video Door Station	V5.85.1.1

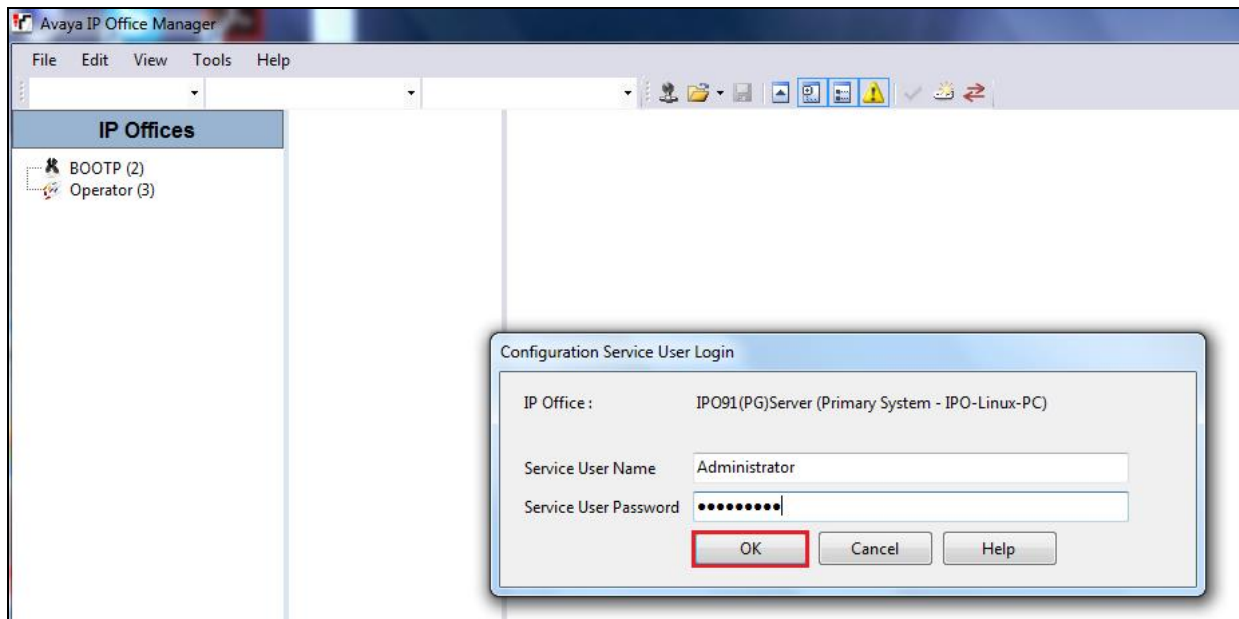
5. Configure Avaya IP Office

Configuration and verification operations on the Avaya IP Office illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration of the Avaya IP Office for this solution. It is implied a working system is already in place. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

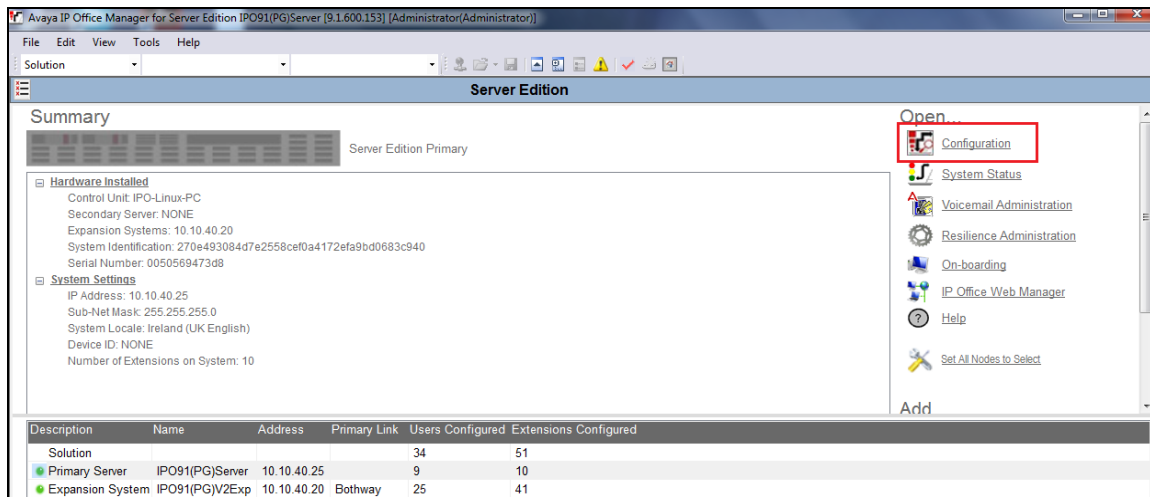
- Launch Avaya IP Office Manager.
- Display LAN Configuration.
- Configure New SIP User.
- Save Configuration.

5.1. Launch Avaya IP Office Manager

From the Avaya IP Office Manager PC, go to **Start → Programs → IP Office → Manager** to launch the Manager application or use the shortcut on the desktop (not shown). A login window will automatically appear, using the appropriate credentials click **OK** to log in.

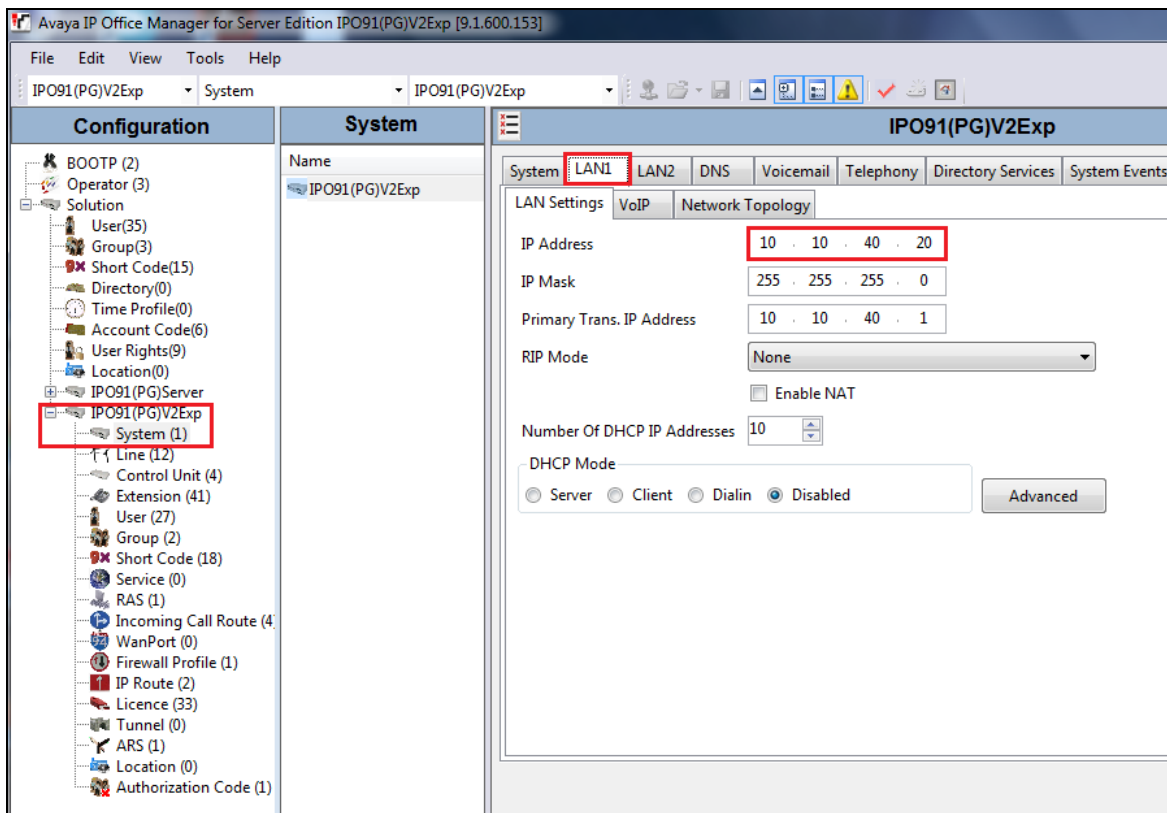


Click on **Configuration** to open the configuration GUI for both the Server Edition system and the expansion system.



5.2. Display LAN Configuration

Once logged in navigate to **System** in the left window and this will display the IP Office system properties in the main window. Select the **LAN1** tab in the main window and within that tab select the **LAN Settings** tab. This displays the **IP Address** information for the Axis door phone to register to in **Section 6.2**.



Selecting the **VoIP** tab displays the **Domain Name** and the **UDP**, **TCP** and **TLS Port** details used in the configuration of the Axis door phone in **Section 6.2**.

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning VCM Codecs

LAN Settings **VoIP** Network Topology

☒ H323 Gatekeeper Enable

☐ Auto-create Extn ☐ Auto-create User ☐ H323 Remote Extn Enable

Remote Call Signalling Port 1720

☒ SIP Trunks Enable

☒ **SIP Registrar Enable**

☐ Auto-create Extn/User ☐ SIP Remote Extn Enable

Domain Name devconnect.local

Layer 4 Protocol

☒ UDP UDP Port 5060 Remote UDP Port 5060

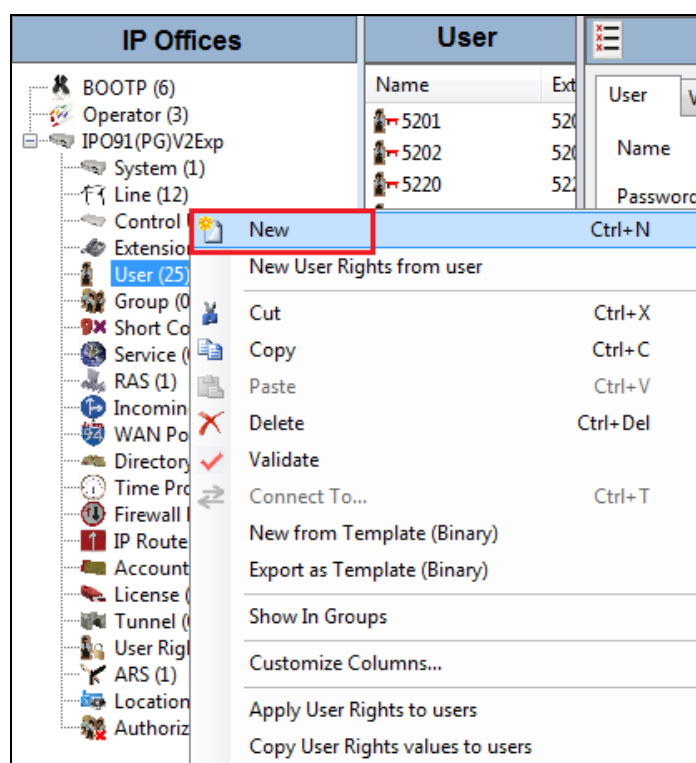
☒ TCP TCP Port 5060 Remote TCP Port 5060

☒ TLS TLS Port 5061 Remote TLS Port 5061

Challenge Expiry Time (secs) 10

5.3. Configure New SIP User

From the left window right click on **Users** and select **New** as shown below, this will allow a new user to be added to IP Office, this new user will be a SIP user.



Within the **User** tab at the top of the screen, enter a suitable **Name** and **Password** for the user. Add the **Extension** number as shown below.

The screenshot displays the 'User' configuration tab for 'Door2 5200: 5200'. The 'User' tab is highlighted with a red box. The form contains the following fields and options:

Field	Value
Name	Door2 5200
Password	••••
Confirm Password	••••
Conference PIN	
Confirm Conference PIN	
Account Status	Enabled
Full Name	Axis Door Phone 500V2
Extension	5200
Email Address	
Locale	
Priority	5
System Phone Rights	None
ACCS Agent Type	None
Profile	Power User

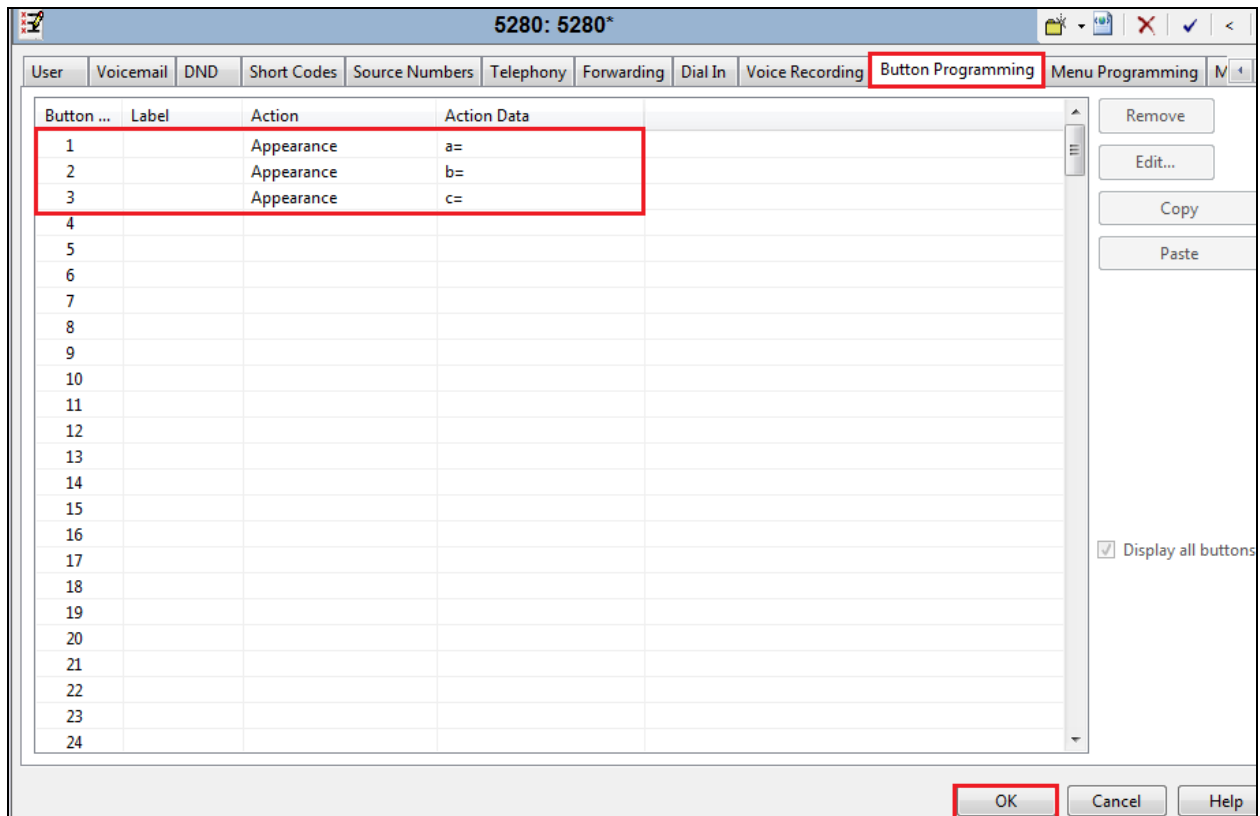
Below the Profile dropdown, there are four checkboxes:

- ☐ Receptionist
- ☒ Enable Softphone
- ☒ Enable one-X Portal Services
- ☒ Enable one-X TeleCommuter

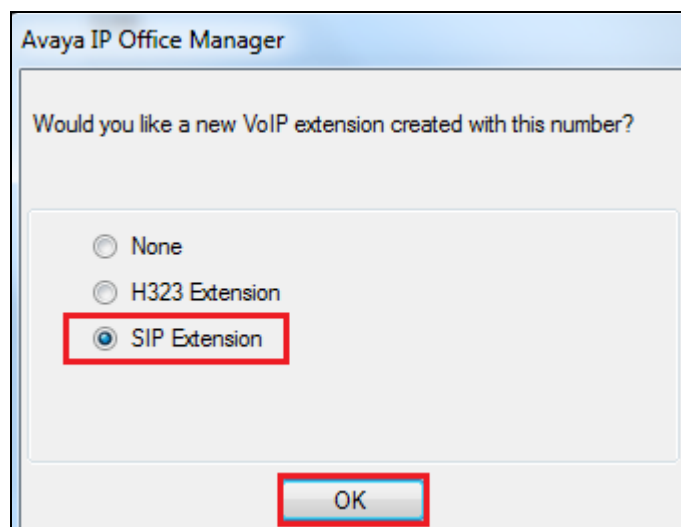
Navigate to the **Supervisor Settings** tab, enter the **Login Code** for the SIP user and note that this password will be required for the Axis door phone configuration in **Section 6.3**. Click on **OK** to save the configuration.

The screenshot shows the 'Door2 5200: 5200*' configuration window. The 'Telephony' tab is selected in the top navigation bar. Within the 'Telephony' tab, the 'Supervisor Settings' sub-tab is active. The settings are organized into two columns. The left column contains: 'Login Code' (masked with four dots), 'Confirm Login Code' (masked with four dots), 'Login Idle Period (secs)' (empty field), 'Monitor Group' (dropdown menu showing '<None>'), 'Coverage Group' (dropdown menu showing '<None>'), 'Status on No-Answer' (dropdown menu showing 'Logged On (No change)'), and a 'Reset Longest Idle Time' section with two radio buttons: 'All Calls' (selected) and 'External Incoming'. The right column contains a list of checkboxes: 'Force Login', 'Force Account Code', 'Force Authorization Code', 'Incoming Call Bar', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', and 'Deny Auto Intercom Calls'. At the bottom right, there are three buttons: 'OK' (highlighted with a red box), 'Cancel', and 'Help'.

Navigate to **Button Programming** and the three call appearance buttons should already be programmed, click on **OK**. If not create the appearance buttons (not shown) and click on **OK**.

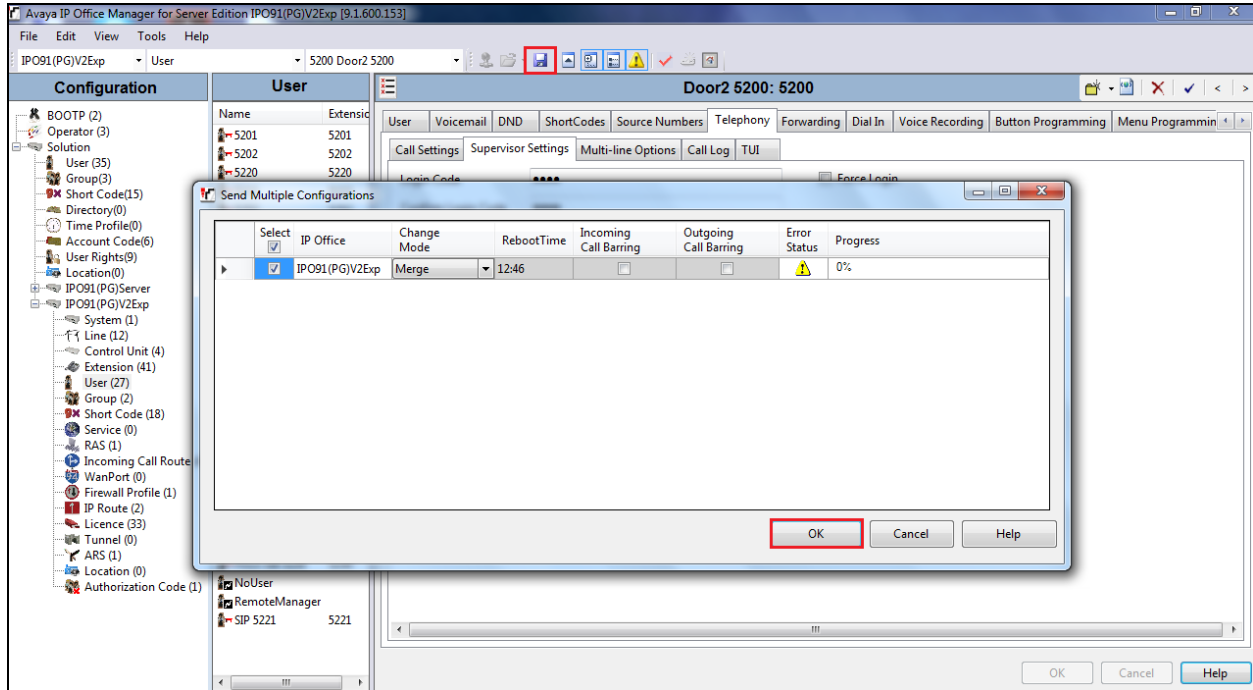


On the subsequent screen, ensure that **SIP Extension** is selected and click on **OK** to create the SIP extension along with the new user.



5.4. Save Configuration

Once all the users and extensions have been created click on the **Save** icon at the top of the screen, which will bring up a new window and click on **OK** to save the new configuration.

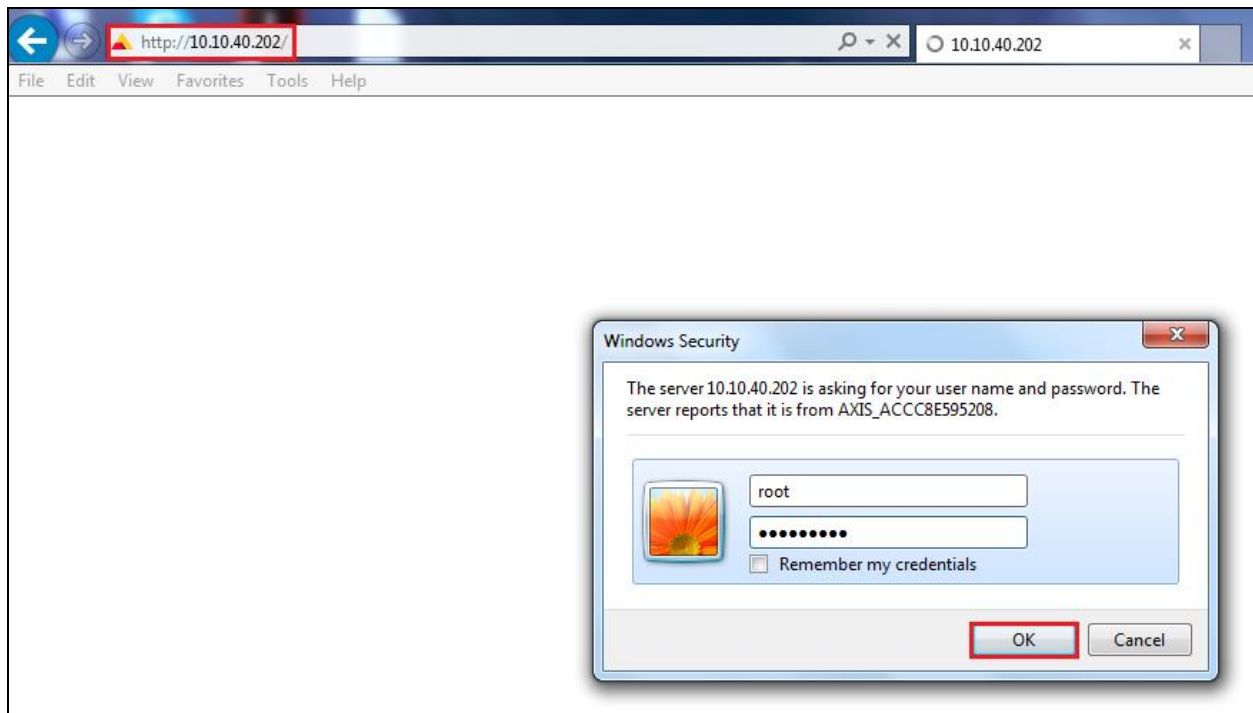


6. Configure AXIS A8004-VE Network Video Door Station

The configuration of the Axis door phone uses a web interface.

Note: The door phone obtains its IP address using DHCP and this was the way in which an IP address was given to the device during compliance testing.

Open a web session to the IP address of the Axis door phone, enter the proper credentials and click on **OK**.



Please refer to Axis Communications documentation listed in **Section 9** of these Application Notes for further information about the Axis door phone configuration. The following sections cover specific settings concerning SIP and the connection to IP Office.

6.1. Configure SIP Settings

The initial step is to enable SIP-functionality as shown below. Some AXIS products have a SIP Setup Assistant that provides an easy setup for the entire product (like button-initiated calls on Network Video Door Station). This guide only shows how to set up an account in the AXIS product not the specific product capabilities. If a Setup Assistant is available, it's recommended to be used. The same configuration specified below can be applied in the assistant separate pages. Click on **VoIP** → **SIP Settings** in the left window, in the main window ensure that **Enable SIP** is ticked under **SIP Settings** and **Allow incoming SIP calls** under **Incoming SIP Calls**. Under the **Audio Codec Settings**, select the codecs that are to be used and click on **Save** once all is configured correctly.

AXIS A8004-VE Network Video Door Station Live View **Setup** Help

Basic Setup
Video & Audio
VoIP
 Overview
 SIP Settings
 VMS Settings
 Account Settings
 DTMF Settings
Live View Config
Detectors
Applications
Events
Recordings
Languages
System Options
About

SIP Settings

SIP Setup Assistant
Start the setup assistant for easy SIP configuration. [Start...](#)

SIP Settings
☒ Enable SIP

Incoming SIP Calls
☒ Allow incoming SIP calls

Port Settings
SIP port: 5060
SIP TLS port: 5061
RTP start port: 4000

NAT Traversal
☐ Enable ICE
☐ Enable STUN
☐ Enable TURN

Audio Codec Settings

Available codecs	Selected codecs
opus (48000 Hz)	PCMU (8000 Hz)
L16/16000 (16000 Hz)	PCMA (8000 Hz)
L16/8000 (8000 Hz)	
speex/16000 (16000 Hz)	
speex/8000 (8000 Hz)	
G.726-32 (8000 Hz)	

Save

6.2. Configure Account

Click on **Account Settings** under **VoIP** in the left window. Click on the **Add** button in the main window.

The screenshot shows the configuration interface for the AXIS A8004-VE Network Video Door Station. The top navigation bar includes the AXIS logo, the device name, and links for Live View, Setup, and Help. The left sidebar contains a tree view of configuration categories: Basic Setup, Video & Audio, VoIP (expanded), Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. Under the VoIP category, the sub-items are Overview, SIP Settings, VMS Settings, Account Settings (highlighted with a red box), and DTMF Settings. The main content area is titled 'Account Settings' and features a table with columns: Name, SIP address, Transport, Default, and Reg. status. The table is currently empty. Below the table are three buttons: 'Add...' (highlighted with a red box), 'Modify...', and 'Remove'. At the bottom, there is a 'Test SIP Call' section with a text input field labeled 'Enter SIP address: sip(s):extension@domain' and a 'Test call' button.

Name	SIP address	Transport	Default	Reg. status
------	-------------	-----------	---------	-------------

Add... **Modify...** **Remove**

Test SIP Call

Make a test call from the selected SIP account to the specified SIP address.

Enter SIP address: sip(s):extension@domain **Test call**

Enter the following details under the **General** tab:

- **Name:** Enter a suitable name for the SIP account.
- **User ID:** Enter the SIP user number configured in **Section 5.3**.
- **Password:** Enter the password for the SIP user created in **Section 5.3**.
- **Caller ID:** This should be the extension number created in **Section 5.3**.
- **Domain Name:** The domain as per **Section 5.2**, the IP Office telephony domain.
- **Registrar address:** The IP address of the IP Office, as per **Section 5.2**.

AXIS A8004-VE Network Video Door Station - Internet Explorer

http://10.10.40.202/admin/account_set.shtml?doAction=mod&id=sip_account_2

Modify Account

General Network Video

Account Information

Name: 500V2 Door

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

Account Credentials

User ID: 5200

☒ Use User ID as Authentication ID

Authentication ID: 5200

Password: ••••

Caller ID: 5200

SIP Server Settings

Domain name: devconnect.local

Registrar address: 10.10.40.20

Select the **Network** tab and select the transport mode to be used, this can be UDP, **TCP** or TLS, all three protocols were tested and work correctly with IP Office. Click on **Save** to save the Account information.

The screenshot shows a web browser window titled "AXIS A8004-VE Network Video Door Station - Internet Explorer". The address bar displays the URL: `http://10.10.40.202/admin/account_set.shtml?doAction=mod&id=sip_account_2#`. The main content area is titled "Modify Account" and features three tabs: "General", "Network" (which is highlighted with a red box), and "Video".

Under the "Network" tab, there are two main sections:

- Transport Settings:** This section includes a checkbox for "Enable SIPs", a "Transport mode:" dropdown menu currently set to "TCP", and another checkbox for "Allow port update messages through MWI".
- Proxy Settings:** This section contains a table with two columns, "Address" and "Username". The table is currently empty. To the right of the table are up and down arrow buttons. Below the table is an "Add..." button.

At the bottom of the form, there is an "Account Status" section. At the very bottom of the page, there are two buttons: "Save" (highlighted with a red box) and "Cancel".

6.3. Configure DTMF Settings

Staying within the **VoIP** menu on the left window, select **DTMF Settings**. In the main window select the SIP account that was created in **Section 6.2** and click on the edit icon, as shown below.

The screenshot displays the web interface for the AXIS A8004-VE Network Video Door Station. The top navigation bar includes the AXIS logo, the device name, and links for Live View, Setup, and Help. A left-hand menu lists various configuration categories: Basic Setup, Video & Audio, VoIP, Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. The VoIP section is expanded, showing sub-items: Overview, SIP Settings, VMS Settings, Account Settings, and DTMF Settings (which is highlighted with a red box). The main content area is titled 'DTMF Settings' and contains a section for 'DTMF Configuration for SIP Accounts'. Under this, there is a collapsible list of 'Peer-to-peer accounts (No local accounts)'. One account, '500V2 Door (5200)', is expanded and highlighted with a blue bar and a red box. To the right of this account name is an edit icon (a pencil inside a red square). Below the account list, there are two checked options: 'DTMF using SIP INFO (RFC2976)' and 'DTMF using RTP (RFC2833)'. At the bottom of the main content area is a section titled 'Associated DTMF Sequences' which contains an empty table with columns for 'Name' and 'Sequence'.

AXIS A8004-VE Network Video Door Station Live View | Setup | Help

- Basic Setup
- Video & Audio
- VoIP**
 - Overview
 - SIP Settings
 - VMS Settings
 - Account Settings
 - DTMF Settings**
- Live View Config
- Detectors
- Applications
- Events
- Recordings
- Languages
- System Options
- About

DTMF Settings

DTMF Configuration for SIP Accounts

- Peer-to-peer accounts (No local accounts)
 - 500V2 Door (5200)** [Edit]

☒ DTMF using SIP INFO (RFC2976)

☒ DTMF using RTP (RFC2833)

Associated DTMF Sequences

Name	Sequence
------	----------

Tick the required way in which DTMF will be sent. **SIP INFO** packets or as specially marked events in the RTP stream using **RFC 2833**. Click on **Add** at the bottom of the screen to add the digits required to utilise the “open door” function.

Modify DTMF Configuration ?

SIP Account: 500V2 Door (5200)

☒ DTMF using SIP INFO (RFC2976)

☒ DTMF using RTP (RFC2833)

DTMF Sequences

Name	Sequence
------	----------

Add... Modify... Remove

OK Cancel

Enter a suitable **Name** and the number **Sequence** to open the door, click on **Apply** and **OK** to save.

Modify DTMF Configuration ?

SIP Account: 500V2 Door (5200)

☒ DTMF using SIP INFO (RFC2976)

☒ DTMF using RTP (RFC2833)

DTMF Sequences

Name	Sequence
------	----------

Name:

Sequence:

Apply Dismiss

OK Cancel

6.4. Configure Events

In order to create an event both a recipient and an action rule must be created. A recipient is created before an action rule.

6.4.1. Add a new recipient

Click on **Events** → **Recipients** in the left window and in the main window, click on **Add**.

The screenshot shows the web interface for the AXIS A8004-VE Network Video Door Station. The left sidebar contains a menu with options: Basic Setup, Video & Audio, VoIP, Live View Config, Detectors, Applications, Events (expanded), Recordings, Languages, System Options, and About. Under the Events section, 'Recipients' is highlighted with a red box. The main content area is titled 'Recipients' and contains a 'Recipients List' table with columns: Name, Type, Address, Upload path, and User name. Below the table are buttons: Add... (highlighted with a red box), View..., Copy..., and Remove. A help icon (?) is in the top right corner.

Enter a suitable **Name** for the **Recipient** and ensure that **Type** is set to **SIP**. The **From** and **To** must be chosen. The **From SIP account** should be that created in **Section 6.2**. The **To SIP address** will be the IP Office extension that is to be called. A **Test** can be done to verify the call to the extension before it is saved.

The screenshot shows the 'Recipient Setup' dialog box. It has a title bar with a question mark icon. The fields are: Name (V2 to Digital), Type (SIP dropdown), From SIP account (500V2 Door (5200) dropdown), and To SIP address (5201@10.10.40.20). Below these is a 'Test' section with the text: 'Test the connection between the selected SIP account and the specified SIP address. The call will end automatically.' There is a 'Select SIP account' dropdown (500V2 Door (5200)) and a 'Test' button (highlighted with a red box). At the bottom are 'OK' (highlighted with a red box) and 'Cancel' buttons.

A number of different recipients are normal for such a test, where various IP Office endpoints can be called, or perhaps a number of hunt groups.

The screenshot shows the 'Recipients' configuration page in the AXIS A8004-VE Network Video Door Station web interface. The left sidebar contains a menu with options: Basic Setup, Video & Audio, VoIP, Live View Config, Detectors, Applications, Events (with sub-items: Action Rules, Recipients, Schedules, Recurrences), Recordings, Languages, System Options, and About. The main area is titled 'Recipients' and contains a 'Recipients List' table. Below the table are buttons for 'Add...', 'View...', 'Copy...', and 'Remove'.

Name	Type	Address	Upload path	User name
SE to Ext5101	SIP	5101@10.10.40.25	-	-
V2 to Digital	SIP	5201@10.10.40.20	-	-
V2 to H323-5250	SIP	5250@10.10.40.20	-	-
V2 to Hunt	SIP	5298@10.10.40.20	-	-
V2 to QSIG	SIP	97000@10.10.40.20	-	-
V2 to SIP	SIP	87101@10.10.40.20	-	-
V2 to WinComm	SIP	5102@10.10.40.20	-	-

6.4.2. Modify Action Rule

An action rule can now be modified to include the participant created in **Section 6.4.1**. Under **Events** in the left window click on **Action Rules** and in the main window select the **BUTTON: VMS call** rule and click **Modify** as shown below.

The screenshot shows the 'Action Rules' configuration page in the AXIS A8004-VE Network Video Door Station web interface. The left sidebar is the same as in the previous screenshot, but the 'Action Rules' option under 'Events' is highlighted with a red box. The main area is titled 'Action Rules' and contains an 'Action Rule List' table. Below the table are buttons for 'Add...', 'Copy...', 'Modify...', and 'Remove'. The 'BUTTON: VMS call' rule is highlighted in blue.

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	-
<input checked="" type="checkbox"/> DOOR: REX unlocks	Input Signal - Digital Input Port	-	Output Port	-
<input checked="" type="checkbox"/> FailoverTest	Call - StateChange	-	Make Call	-
<input checked="" type="checkbox"/> LIGHT: Active call	Call - State	-	Activate Light	-
<input checked="" type="checkbox"/> LIGHT: Calling	Call - State	-	Activate Light	-

The information should reflect what is displayed below, the **General** section should display what is shown by default, and if not change it to what is displayed below or to what condition is required. Under the **Actions** section the **Type** is set to **Make Call** and the **Recipient** is set to that recipient created in **Section 6.4.1**. This will ensure that when the button is pressed a call is made to the recipient. Click on **OK** to save the configuration.

Action Rule Setup

?

General

☒ Enable rule

Name:

BUTTON: VMS call

Condition

Trigger:

Input Signal

Digital Input Port

Call button (Port 1)

Active:

☒ Yes

☐ No

Schedule:

Always (No Schedule)

New Schedule

☐ Additional conditions

Actions

Type:

Make Call

Recipient:

V2 to Digital

New Recipient

OK

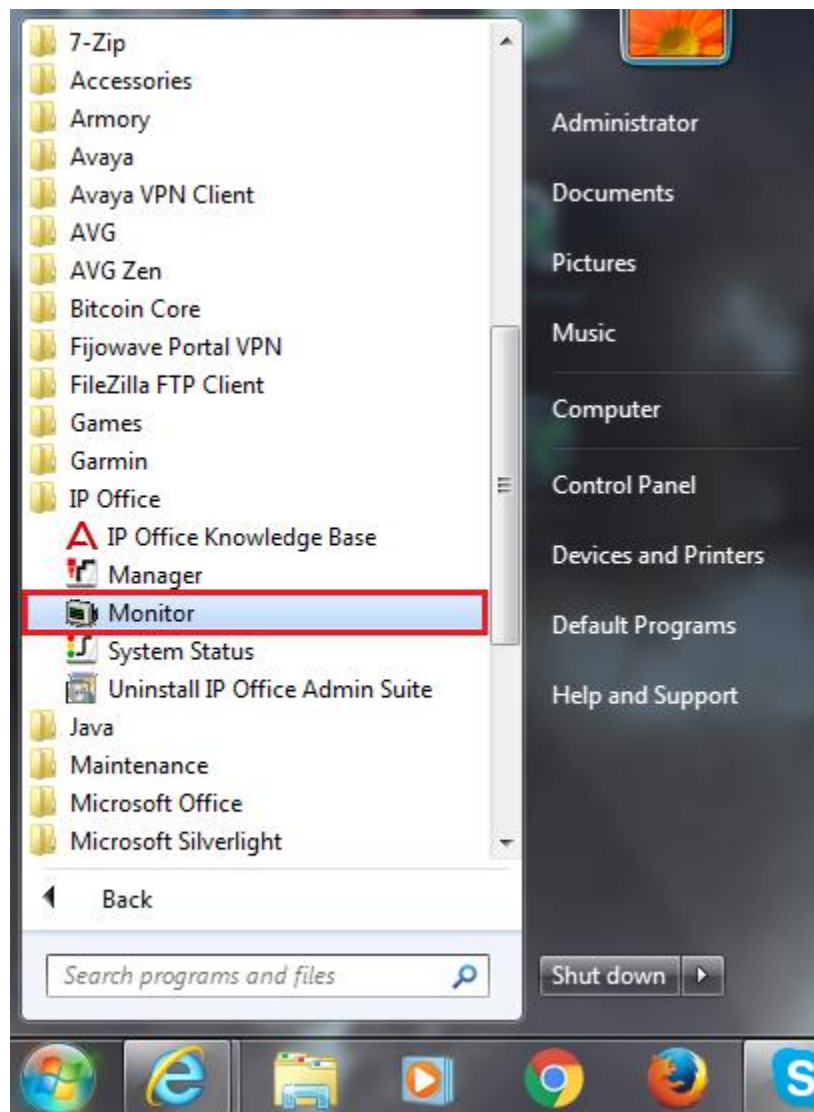
Cancel

7. Verification Steps

Pressing the Axis door phone button and answering the call from the IP Office set and ensuring there is two-way speech and video (where possible) is the ultimate verification that the product works and is connected and configured correctly. The steps below can also be taken to ensure that the Axis door phone is registered correctly with IP Office and some monitoring tips to see that this is the case.

7.1. Verify Registration from IP Office

Open IP Office **Monitor** as shown below.



Once connected to the desired IP Office information on SIP calls and registrations will be shown (as long as the correct filter is applied for SIP messaging (not shown)). Below is an example of a message being displayed when a call is made from the door phone to extension **5201** which is a digital phone on IP Office. It clearly shows from **5200** which is the door phone extension number.

The screenshot shows the Avaya IP Office SysMonitor application window. The title bar reads "Avaya IP Office SysMonitor - Monitoring 10.10.40.20 (IPO91(PG)V2Exp (Server Edition(E))); Log Settings - C:\Users\...\sysmonitorsettings.ini". The menu bar includes File, Edit, View, Filters, Status, and Help. Below the menu is a toolbar with various icons. The main display area shows a log of SIP messages. A red box highlights the "INVITE sip:5201@10.10.40.20;transport=TCP SIP/2.0" line. The log entry starts with "253143179mS SIP Rx: TCP 10.10.40.202:36245 -> 10.10.40.20:5060". The message body includes fields like Via, Max-Forwards, From, To, Contact, Call-ID, CSeq, Allow, Supported, Session-Expires, Min-SE, User-Agent, Content-Type, and Content-Length. It also contains SDP lines for audio and video codecs. The log ends with "253143183mS Sip: TCP packet known set owner".

```

253143179mS SIP Rx: TCP 10.10.40.202:36245 -> 10.10.40.20:5060
INVITE sip:5201@10.10.40.20;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.10.40.202:36245;rport;branch=z9hG4bKPjld0FZe2qFGRPbnEg7gKQISr1fe34Zvnt;alias
Max-Forwards: 70
From: "5200" <sip:5200@devconnect.local>;tag=g5ZcvBsSeZ82eRiYHw8Uhz2TKVjgg6fw
To: sip:5201@10.10.40.20
Contact: "5200" <sip:5200@10.10.40.202:36245;transport=TCP;ob>
Call-ID: 7D9pVmXb9NQQ4jmI28qe3PZ75M6TDi4X
CSeq: 7110 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: AXIS A8004-VE Network Video Door Station
Content-Type: application/sdp
Content-Length: 478

v=0
o=- 3674549777 3674549777 IN IP4 10.10.40.202
s=pjmedia
b=AS:84
t=0 0
a=X-nat:0
m=audio 4012 RTP/AVP 110 0 8 96
c=IN IP4 10.10.40.202
b=TIAS:64000
a=rtcp:4013 IN IP4 10.10.40.202
a=sendrecv
a=rtpmap:110 G726-32/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-16
m=video 4014 RTP/AVP 97
c=IN IP4 10.10.40.202
a=rtcp:4015 IN IP4 10.10.40.202
a=sendonly
a=rtpmap:97 H264/90000
a=fmtp:97 profile-level-id=42000d
253143183mS Sip: TCP packet known set owner

```


7.2. Verify Registration from AXIS A8004-VE Network Video Door Station

Log in to the door phone as per **Section 6**. Navigate to **VoIP → Account Settings** in the left window and the registration information should be displayed in the main window as shown below. The green lights show a successful registration of both **5200** and **5100**. Test call can be made from each account to a specific phone number using the **Test SIP Call** at the bottom of the screen.

The screenshot displays the web interface of the AXIS A8004-VE Network Video Door Station. The top navigation bar includes the AXIS logo, the product name, and links for Live View, Setup, and Help. The left sidebar contains a 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, and 'Account Settings' is highlighted with a red box. The main content area, titled 'Account Settings', features a table with columns for Name, SIP address, Transport, Default, and Reg. status. Two accounts are listed: '500V2 Door (5200)' and 'SE Door 5100 (5100)'. Both accounts show a green checkmark in the 'Reg. status' column, indicating successful registration. Below the table are buttons for 'Add...', 'Modify...', and 'Remove'. At the bottom, there is a 'Test SIP Call' section with a text input field for the SIP address and a 'Test call' button.

Name	SIP address	Transport	Default	Reg. status
500V2 Door (5200)	<sip:5200@devconnect.local>	TCP	✓	✓
SE Door 5100 (5100)	<sip:5100@devconnect.local>	UDP		✓

If there is an issue with a call from the Axis door phone then there are logs that can be accessed that may show some further information on where the issue may lie. Navigate to **System Options → Support → Logs & Reports** in the left window and from the main window select **View Server Report** under the **Reports** section.

The screenshot displays the web interface for the Axis A8004-VE Network Video Door Station. The top navigation bar includes the Axis logo, the device name, and links for Live View, Setup, and Help. A left-hand sidebar contains a tree view of system options, with 'Logs & Reports' highlighted under the 'Support' category. The main content area is titled 'Logs & Reports' and includes a help icon. It contains a note about loading times, a 'Logs' section with buttons for 'System Log' and 'Access Log', and a 'Reports' section with buttons for 'View Server Report', 'Download Server Report', 'Parameter List', 'Connection List', and 'Crash Report'. The 'View Server Report' button is highlighted with a red rectangle. Below the reports section is a link to the Axis Privacy statement.

AXIS A8004-VE Network Video Door Station Live View | Setup | Help

Logs & Reports ?

The log files and reports may prove useful when troubleshooting a problem or when contacting the Axis support web.

Note: Depending on your connection, these pages may take a while to load.

Logs

System Log System log information.

Access Log Access log information.

Reports

View Server Report Important information about the server's status.

Download Server Report ☐ Include snapshot from Live View

Parameter List The unit's parameters and their current settings.

Connection List Connection list information.

Crash Report Detailed information about the server's internal status. This report may contain sensitive information. It may take several minutes to download this report, please wait for the download to finish.

For more information, please read Axis [Privacy statement](#).

This should open a report something like that shown below.

```
http://10.10.40.202/axis-cgi/admin/serverreport.cgi?id=119 - Internet Explorer
http://10.10.40.202/axis-cgi/admin/serverreport.cgi?id=119

2016-06-10T11:44:09.656+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "TAMPERING: Casing open" is starting action "Output Port"
2016-06-10T11:44:10.415+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Removing "TAMPERING: Shock detected" action rule
2016-06-10T11:44:10.498+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Creating "TAMPERING: Shock detected" action rule
2016-06-10T11:47:29.021+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Validating "Make Call" action
2016-06-10T11:47:29.130+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Creating "BUTTON: VMS call" action rule
2016-06-10T11:47:29.220+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Removing "Make Call" action
2016-06-10T11:47:29.221+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Removing "BUTTON: VMS call" action rule
2016-06-10T11:54:58.417+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP image/jpeg session created from 10.10.40.203
2016-06-10T11:54:59.277+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP audio/mpeg session created from 10.10.40.203
2016-06-10T11:55:04.474+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP audio/mpeg session terminated from 10.10.40.203
2016-06-10T11:55:05.591+01:00 axis-acc08e595208 [ INFO ] sipd: Terminated incoming call: In-7-1465556099.336039-VMS
2016-06-10T11:55:05.216+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP audio/mpeg session created from 10.10.40.203
2016-06-10T11:55:19.091+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP image/jpeg session terminated from 10.10.40.203
2016-06-10T11:55:19.173+01:00 axis-acc08e595208 [ INFO ] sipd: Terminated incoming call: In-6-1465556099.513016-VMS
2016-06-10T11:55:19.191+01:00 axis-acc08e595208 [ INFO ] monolith[305]: monolith[305]: HTTP audio/mpeg session terminated from 10.10.40.203
2016-06-10T11:55:19.289+01:00 axis-acc08e595208 [ INFO ] sipd: Terminated incoming call: In-7-1465556105.292005-VMS
2016-06-10T11:56:42.445+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "BUTTON: VMS call" is starting action "Make Call"
2016-06-10T11:56:42.471+01:00 axis-acc08e595208 [ INFO ] sipd[1690]: Making call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq from sip account
2016-06-10T11:56:42.490+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "AUDIO: Calling" is starting action "Play Audio Clip"
2016-06-10T11:56:42.543+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Calling" is starting action "Activate Light on Calling"
2016-06-10T11:56:42.628+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Idle" is stopping action "Activate Light on Idle"
2016-06-10T11:56:45.735+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "AUDIO: Stop on Active call" is starting action "Stop Audio Clip"
2016-06-10T11:56:45.793+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Active call" is starting action "Activate Light on Active"
2016-06-10T11:56:45.919+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Calling" is stopping action "Activate Light on Calling"
2016-06-10T11:56:46.720+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "TAMPERING: Shock detected" is starting action "Output Port"
2016-06-10T11:56:49.248+01:00 axis-acc08e595208 [ INFO ] sipd: DTMF event door open in call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq.
2016-06-10T11:56:51.381+01:00 axis-acc08e595208 [ INFO ] sipd: Terminated outgoing call: Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq
2016-06-10T11:56:51.410+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "AUDIO: Stop on Idle call" is starting action "Stop Audio Clip"
2016-06-10T11:56:51.446+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Active call" is stopping action "Activate Light on Active"
2016-06-10T11:56:51.520+01:00 axis-acc08e595208 [ NOTICE ] actionengined: Action rule "LIGHT: Idle" is starting action "Activate Light on Idle"

----- Kernel log -----
<6>Initializing cgroup subsys cpu
<6>Initializing cgroup subsys cpusacct
<5>Linux version 3.18.0 (svcg@aster-x) (gcc version 4.7.2 20120820 (prerelease) [gcc-4_7-branch revision 190527] (GCC 4.7.2 Axis release R25/1.25) ) #1 SMP F
<6>bootconsole [early0] enabled
<6>CPU0 revision is: 01019550 (MIPS 34Kc)
<6>Determined physical RAM map:
<6> memory: 0c000000 @ 00000000 (usable)
<6>Initrd not found or empty - disabling initrd
<4>Zone ranges:
```

Information on the call made and the door opening is displayed in the log file.

```
[ INFO ] sipd: Terminated incoming call: In-7-1465556105.292005-VMS
[ NOTICE ] actionengined: Action rule "BUTTON: VMS call" is starting action "Make Call"
[ INFO ] sipd[1690]: Making call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq from sip account
[ NOTICE ] actionengined: Action rule "AUDIO: Calling" is starting action "Play Audio Clip"
[ NOTICE ] actionengined: Action rule "LIGHT: Calling" is starting action "Activate Light on Calling"
[ NOTICE ] actionengined: Action rule "LIGHT: Idle" is stopping action "Activate Light on Idle"
[ NOTICE ] actionengined: Action rule "AUDIO: Stop on Active call" is starting action "Stop Audio Clip"
[ NOTICE ] actionengined: Action rule "LIGHT: Active call" is starting action "Activate Light on Active"
[ NOTICE ] actionengined: Action rule "LIGHT: Calling" is stopping action "Activate Light on Calling"
[ NOTICE ] actionengined: Action rule "TAMPERING: Shock detected" is starting action "Output Port"
[ INFO ] sipd: DTMF event door open in call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq.
[ INFO ] sipd: Terminated outgoing call: Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTcl70-Yr.vq
[ NOTICE ] actionengined: Action rule "AUDIO: Stop on Idle call" is starting action "Stop Audio Clip"
[ NOTICE ] actionengined: Action rule "LIGHT: Active call" is stopping action "Activate Light on Active"
[ NOTICE ] actionengined: Action rule "LIGHT: Idle" is starting action "Activate Light on Idle"
```

8. Conclusion

These Application Notes describe the configuration steps for provisioning the AXIS A8004-VE Network Video Door Station from Axis Communications AB to interoperate with Avaya IP Office Server Edition and IP Office 500 V2 expansion R9.1. Please refer to **Section 2.2** for test results and observations.

9. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com> where the following documents can be obtained.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] Avaya IP Office R9.1 Manager 10.1, Document Number 15-601011
- [2] Avaya IP Office R9.1 Doc library

Technical information for the AXIS A8004-VE Network Video Door Station can be obtained from:

Axis Communications AB

Tel: +46 46 272 18 00

Fax: +46 46 13 61 30

<http://www.axis.com/global/en/learning-and-support>

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