



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for configuring Axis Communications AB AXIS A8004-VE Network Video Door Station with Avaya IP Office Server Edition and IP Office 500 V2 Expansion R10.0 – 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 R10.0.

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 500 V2 expansion R10.0.

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, mobile/PSTN endpoints and a video enabled softphone.

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.

## 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.
- Transfer/Conference/Forwarding.
- Codec support.
- DTMF support.
- Door opening.
- Video Call.
- 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**

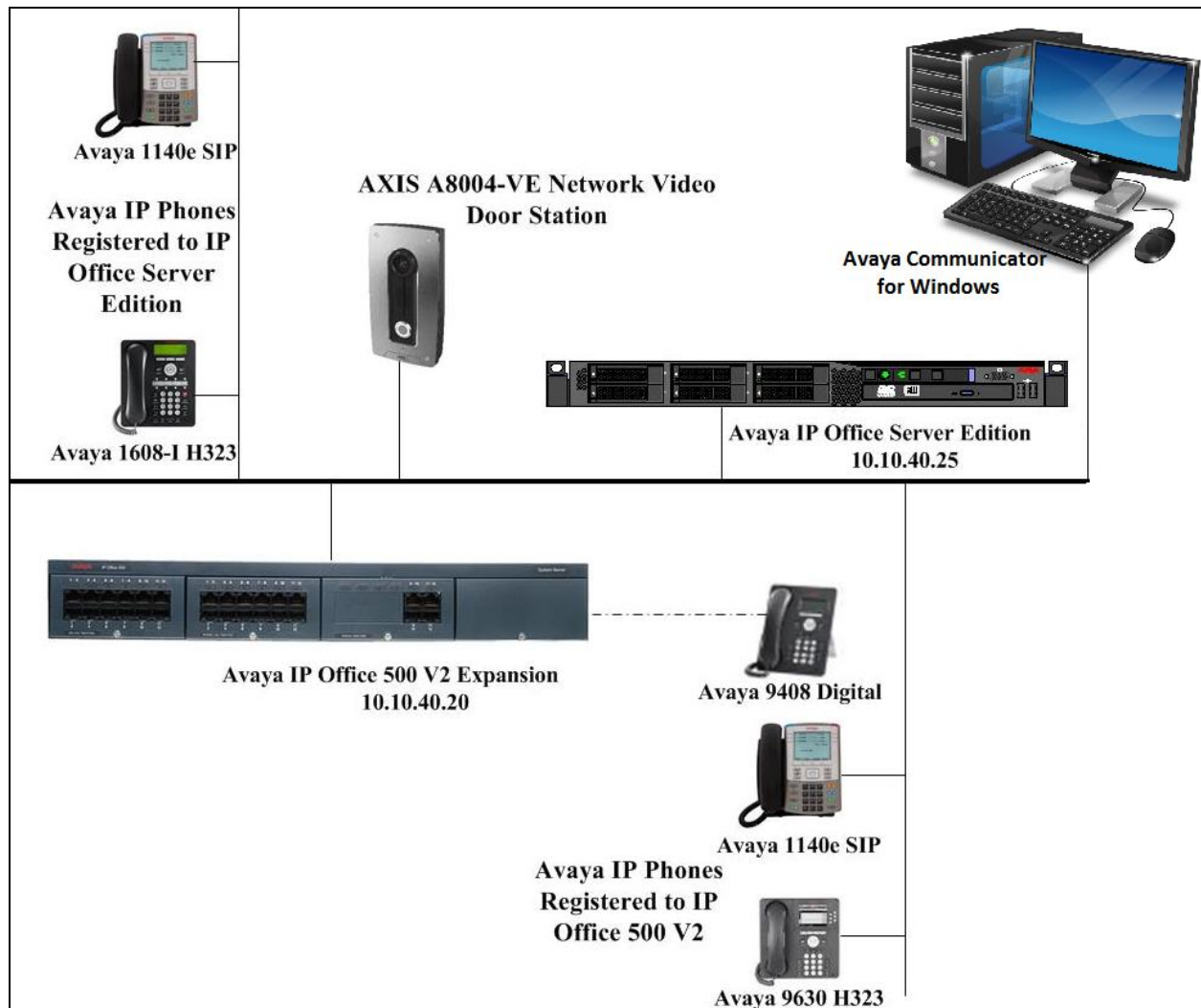
Tel: +46 46 272 18 00

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 500 V2 R10.0**

## 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	R10.0.0.0.0 Build 550
Avaya IP Office 500 V2	R10.0.0.0.0 Build 550
Avaya IP Office Manager	R10.0.0.0.0 Build 550
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
Avaya Communicator for Windows	V 2.1.3
Axis Communications AB AXIS A8004-VE Network Video Door Station	Firmware Version 1.58.2.1

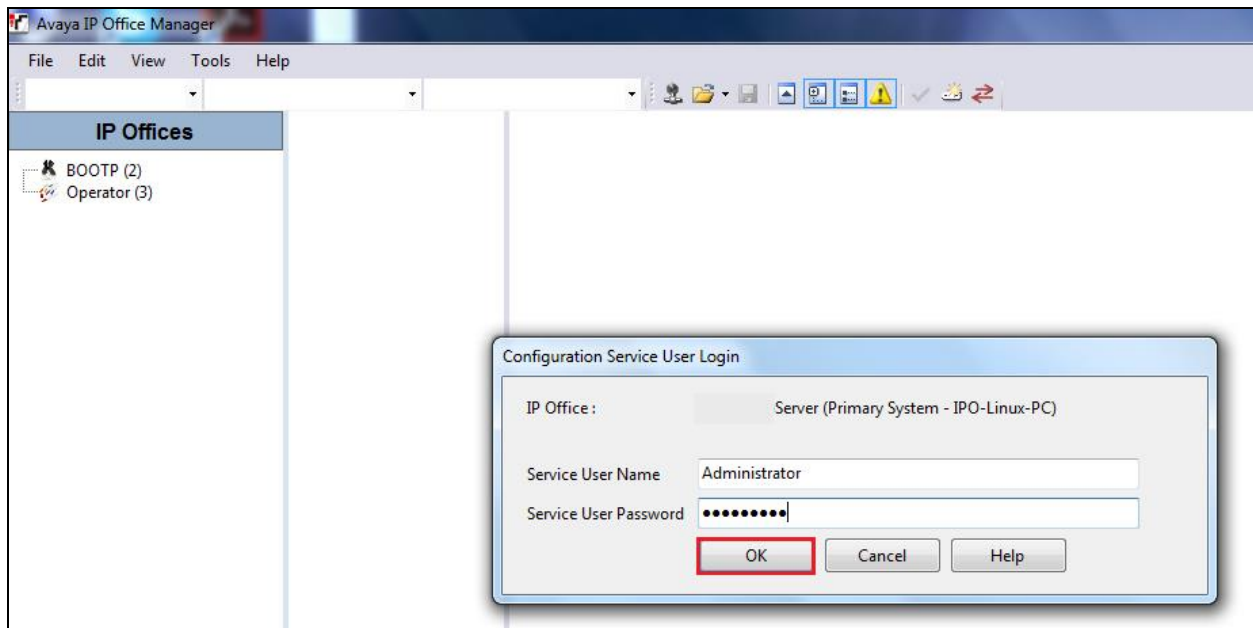
## 5. Configure Avaya IP Office

Configuration and verification operations on 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 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:

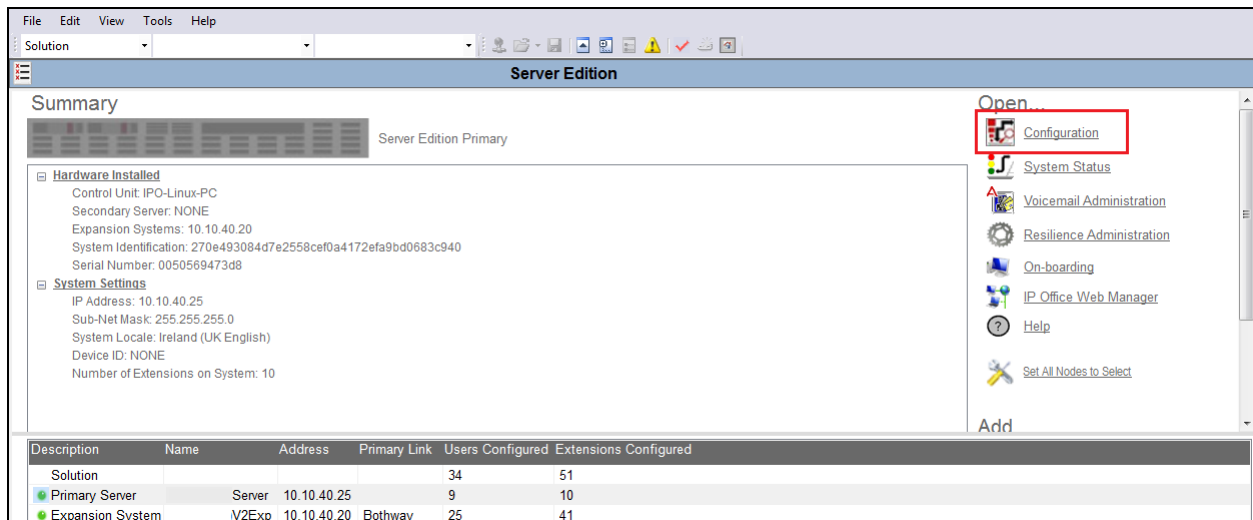
- Launch Avaya IP Office Manager.
- Display LAN Configuration.
- Configure New SIP User.
- Save Configuration.

### 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.

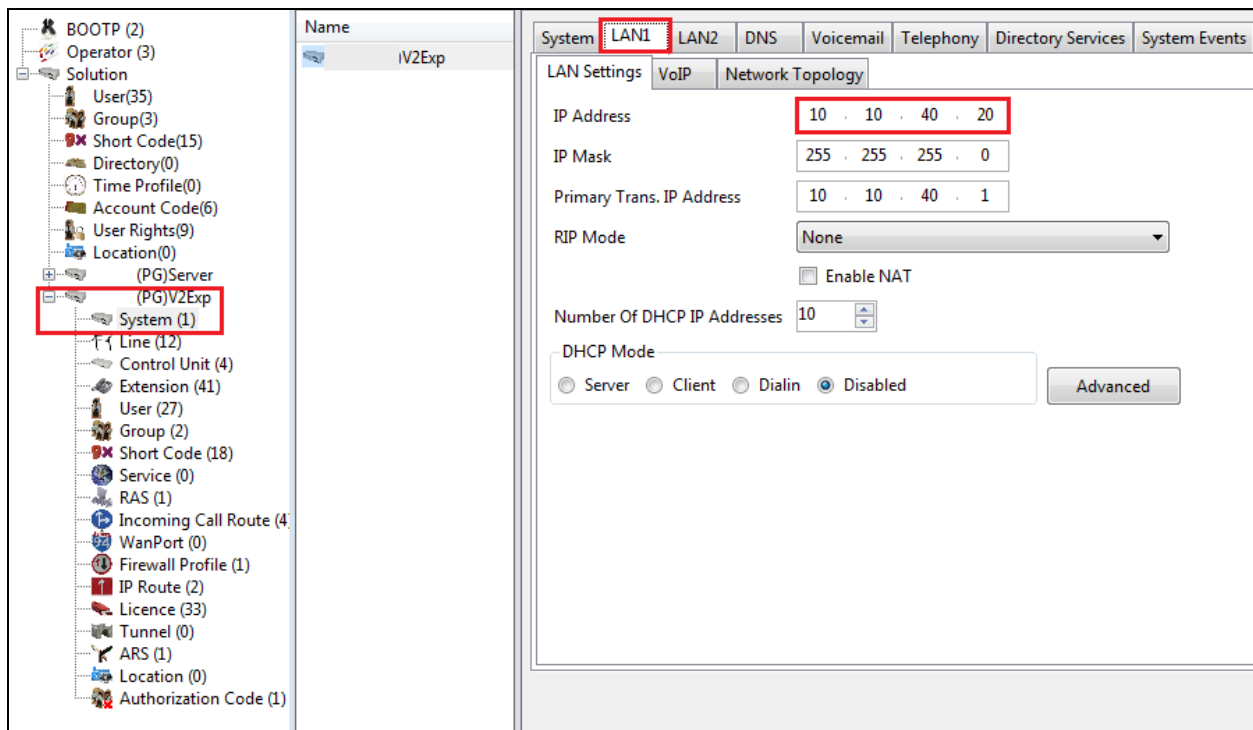


Click on **Configuration** to open the configuration GUI for both the Server Edition system and the expansion system.



## 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**.

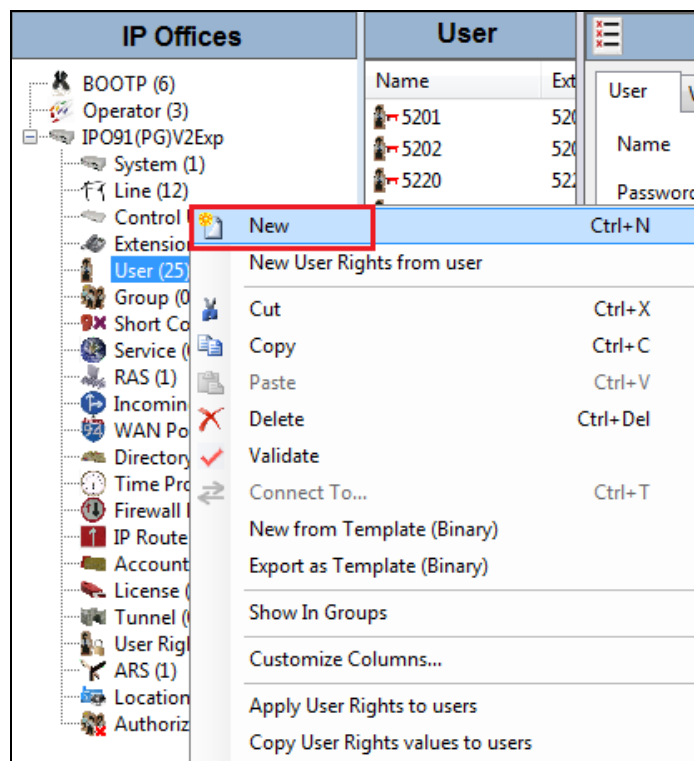


Selecting the **VoIP** tab displays the **Domain Name** and the **UDP**, **TCP** and **TLS Port** details used in the configuration of the Axis door phone in **Section 6.2**.

The screenshot shows the VoIP configuration interface. The 'VoIP' tab is selected. The 'SIP Registrar Enable' checkbox is checked. The 'Domain Name' is 'devconnect.local'. The 'UDP', 'TCP', and 'TLS' protocols are enabled, with their respective ports set to 5060, 5060, and 5061. Remote ports are also set to 5060, 5060, and 5061. The 'Challenge Expiry Time (secs)' is set to 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.

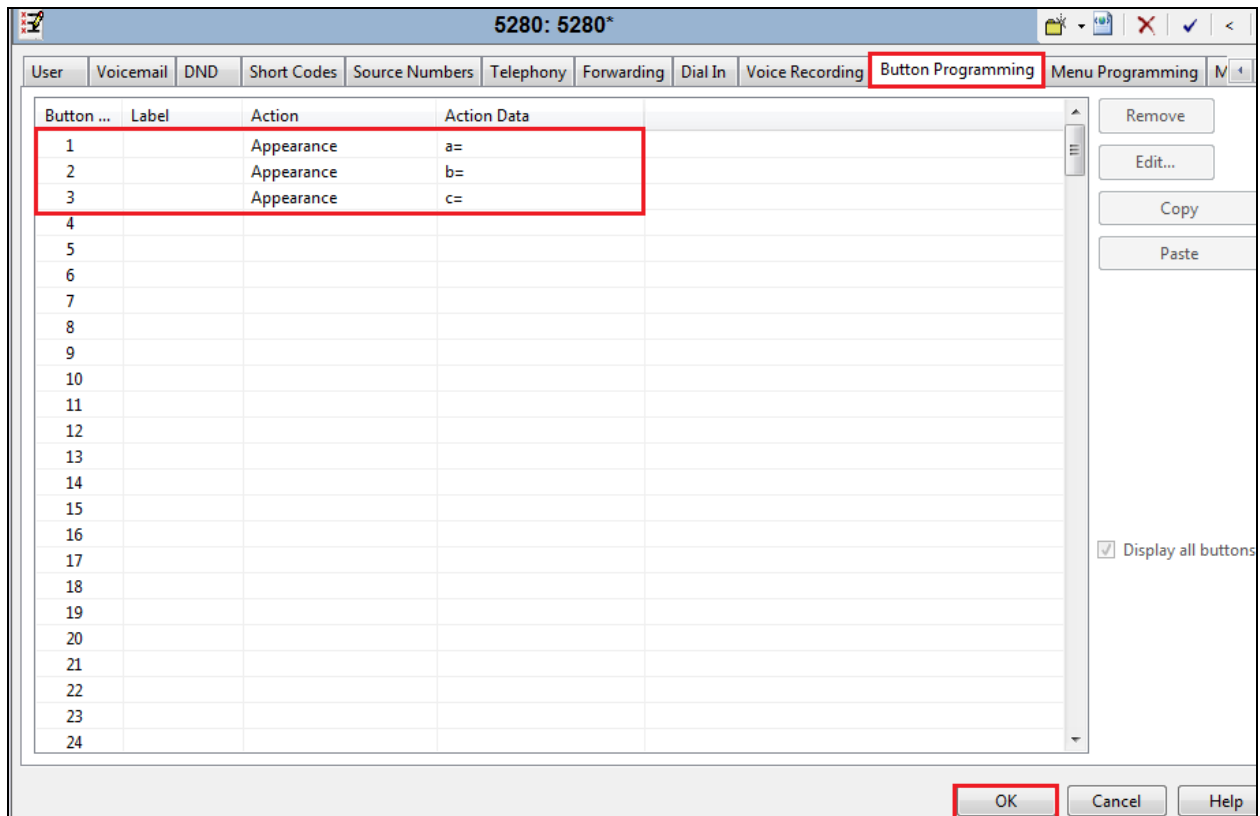
The screenshot displays the AxisDoor configuration web interface. At the top, a blue header bar contains the text "Door2 5200: 5200" and a small icon on the right. Below the header is a horizontal tab bar with the following tabs: "User", "Voicemail", "DND", "ShortCodes", "Source Numbers", "Telephony", "Forwarding", "Dial In", "Voice Recording", and "Button Programming". The "User" tab is selected and highlighted with a red border. The main content area is a form for configuring a user. It includes the following fields and options:

- Name:** Text input field containing "Door2 5200".
- Password:** Password input field with four black dots.
- Confirm Password:** Password input field with four black dots.
- Conference PIN:** Empty text input field.
- Confirm Conference PIN:** Empty text input field.
- Account Status:** Dropdown menu showing "Enabled".
- Full Name:** Text input field containing "Axis Door Phone 500V2".
- Extension:** Text input field containing "5200".
- Email Address:** Empty text input field.
- Locale:** Dropdown menu.
- Priority:** Dropdown menu showing "5".
- System Phone Rights:** Dropdown menu showing "None".
- ACCS Agent Type:** Text input field containing "None".
- Profile:** Dropdown menu showing "Power User".
- Receptionist:** Checkbox, currently unchecked.
- Enable Softphone:** Checkbox, currently checked.
- Enable one-X Portal Services:** Checkbox, currently checked.
- Enable one-X TeleCommuter:** Checkbox, currently checked.

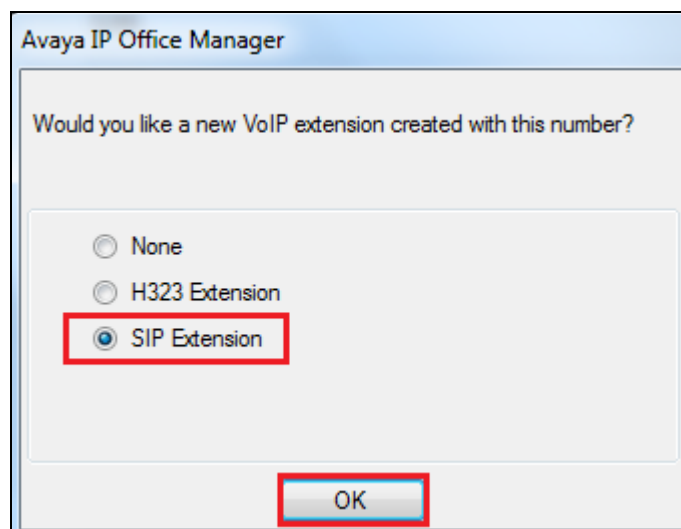
Navigate to the **Supervisor Settings** tab, enter the **Login Code** for the SIP user and note that this password will be required for the Axis door phone configuration in **Section 6.3**. Click on **OK** to save the configuration.

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**.

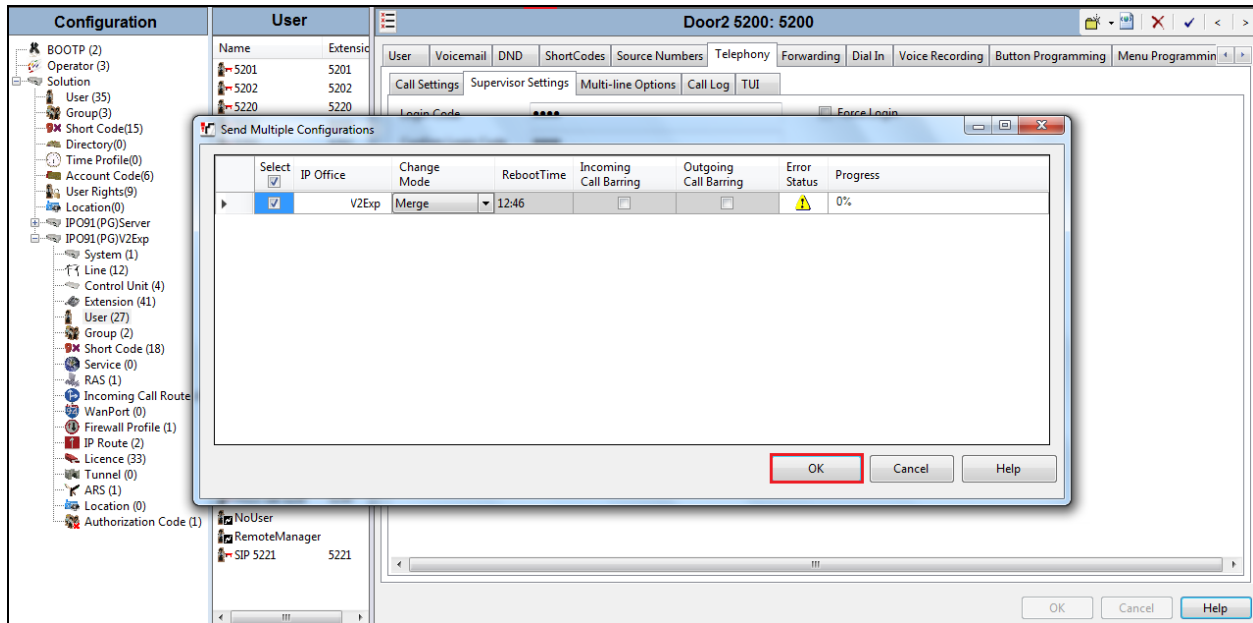


On the subsequent screen, ensure that **SIP Extension** is selected and click on **OK** to create the SIP extension along with the new user.



## 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.

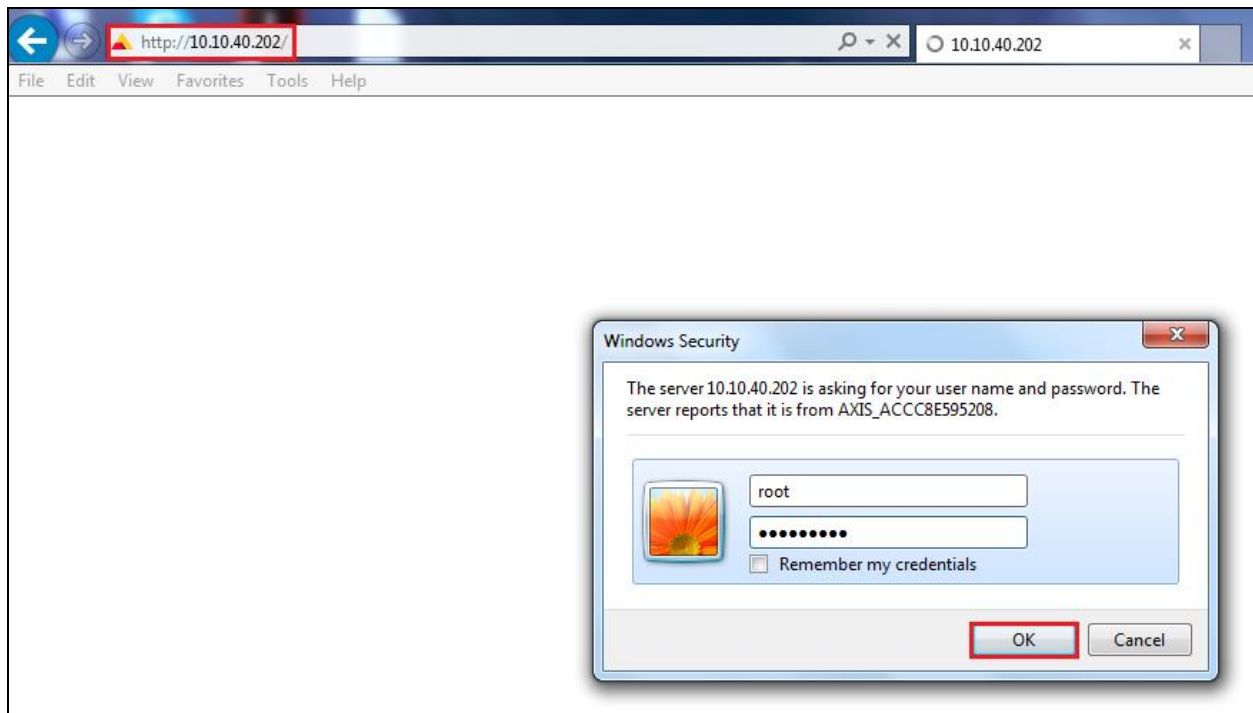


## 6. Configure AXIS A8004-VE Network Video Door Station

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 COMMUNICATIONS** **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:

SIP TLS port:

RTP start port:

**NAT Traversal**

☐ Enable ICE

☐ Enable STUN

☐ Enable TURN

**Audio Codec Settings**

**Available codecs**

- opus (48000 Hz)
- L16/16000 (16000 Hz)
- L16/8000 (8000 Hz)
- speex/16000 (16000 Hz)
- speex/8000 (8000 Hz)
- G.726-32 (8000 Hz)**

**Selected codecs**

- PCMU (8000 Hz)**
- PCMA (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 of the main area is a 'Test SIP Call' section with the instruction 'Make a test call from the selected SIP account to the specified SIP address.' It includes a text input field with the placeholder '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.

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

### Transport Settings

☐ Enable SIPS

Transport mode: TCP

☐ Allow port update messages through MWI

### Proxy Settings

Address	Username
---------	----------

Add...

↑  
↓

### Account Status

Save 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 configuration interface for the AXIS A8004-VE Network Video Door Station. The left sidebar contains a navigation menu with the following items: Basic Setup, Video & Audio, VoIP (expanded), Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. Under the VoIP menu, the following sub-items are listed: Overview, SIP Settings, VMS Settings, Account Settings, and DTMF Settings (highlighted with a red box). The main content area is titled "DTMF Settings" and includes a help icon. Below the title is a section for "DTMF Configuration for SIP Accounts". Under this section, there is a collapsible menu for "Peer-to-peer accounts (No local accounts)". The "500V2 Door (5200)" account is selected and highlighted with a blue bar and a red box. To the right of this bar is an edit icon (pencil). Below the account selection, there are two checked options: "DTMF using SIP INFO (RFC2976)" and "DTMF using RTP (RFC2833)". Below these options is a section titled "Associated DTMF Sequences" which contains a table with two columns: "Name" and "Sequence". The table is currently empty.

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

**DTMF Settings** ?

DTMF Configuration for SIP Accounts

▸ Peer-to-peer accounts (No local accounts)

▾ 500V2 Door (5200) [Edit]

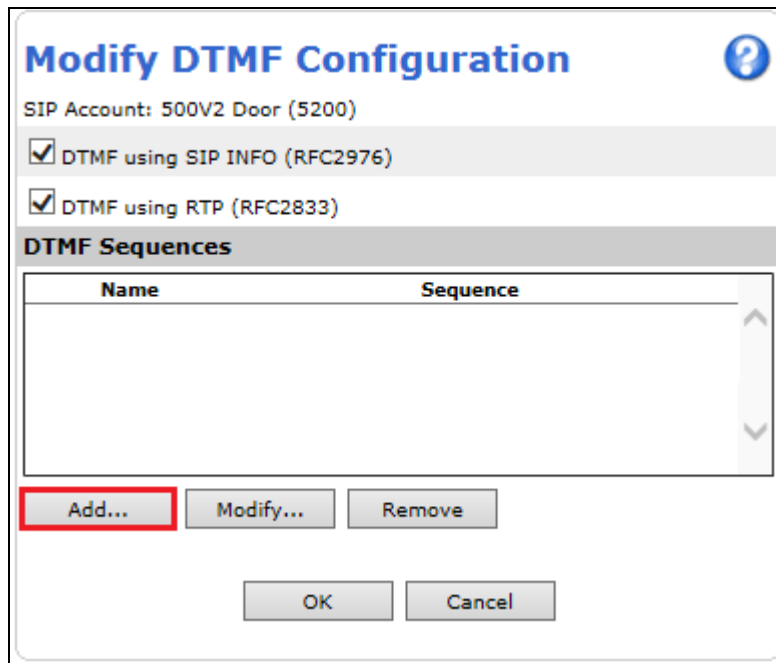
☒ DTMF using SIP INFO (RFC2976)

☒ 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)

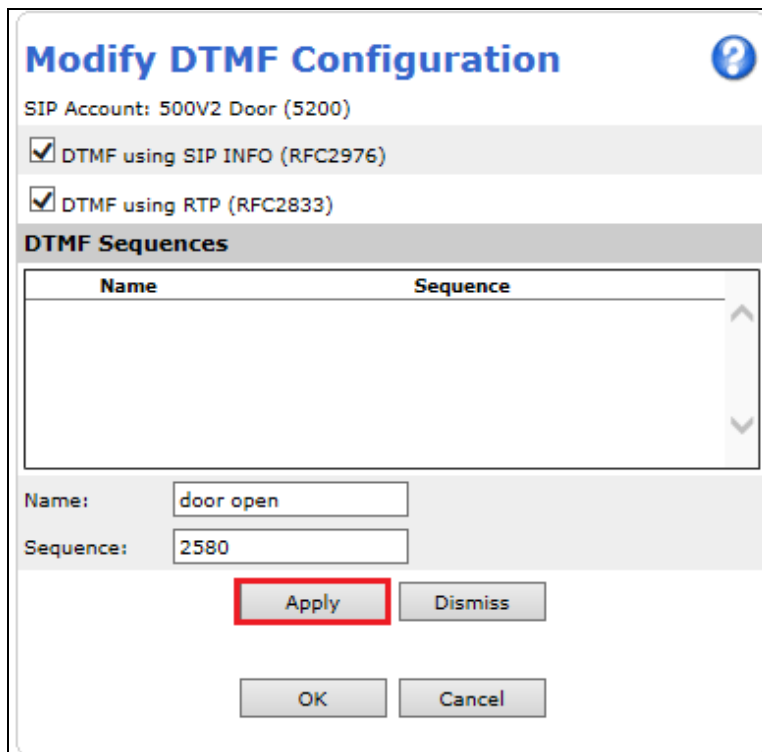
**DTMF Sequences**

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: door open

Sequence: 2580

**Apply** Dismiss

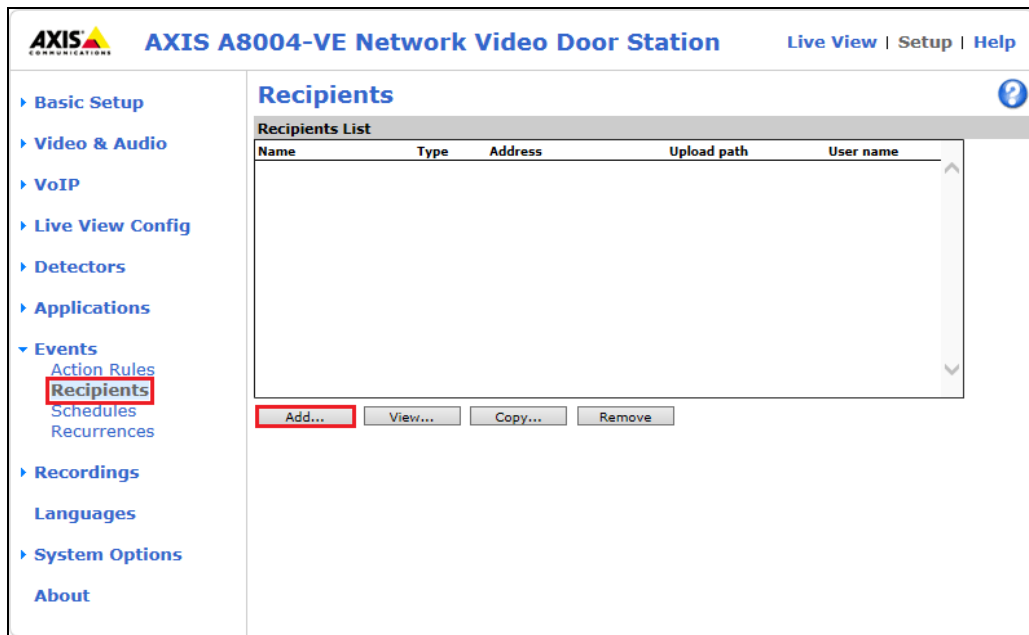
OK Cancel

## 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**.



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 'Recipient Setup' dialog box is shown. It contains the following fields and controls:

- Name:** Text input field containing 'V2 to Digital'.
- Type:** Dropdown menu set to 'SIP'.
- From SIP account:** Dropdown menu set to '500V2 Door (5200)'.
- To SIP address:** Text input field containing '5201@10.10.40.20'.
- Test:** A section with the text: 'Test the connection between the selected SIP account and the specified SIP address. The call will end automatically.'
- Select SIP account:** Dropdown menu set to '500V2 Door (5200)'.
- Test:** A button highlighted with a red box.
- OK:** A button highlighted with a red box.
- Cancel:** A button.

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 'Recipients' item is selected. The main area displays a table titled 'Recipients List' with columns: 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
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 the previous screenshot, but the 'Action Rules' item under 'Events' is selected. The main area displays a table titled 'Action Rule List' with columns: Name, Trigger, Schedule, Action, and Recipient. 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"/> <b>BUTTON: VMS call</b>	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:

**Condition**

Trigger:

Active: ☒ Yes ☐ No

Schedule:

☐ Additional conditions

**Actions**

Type:

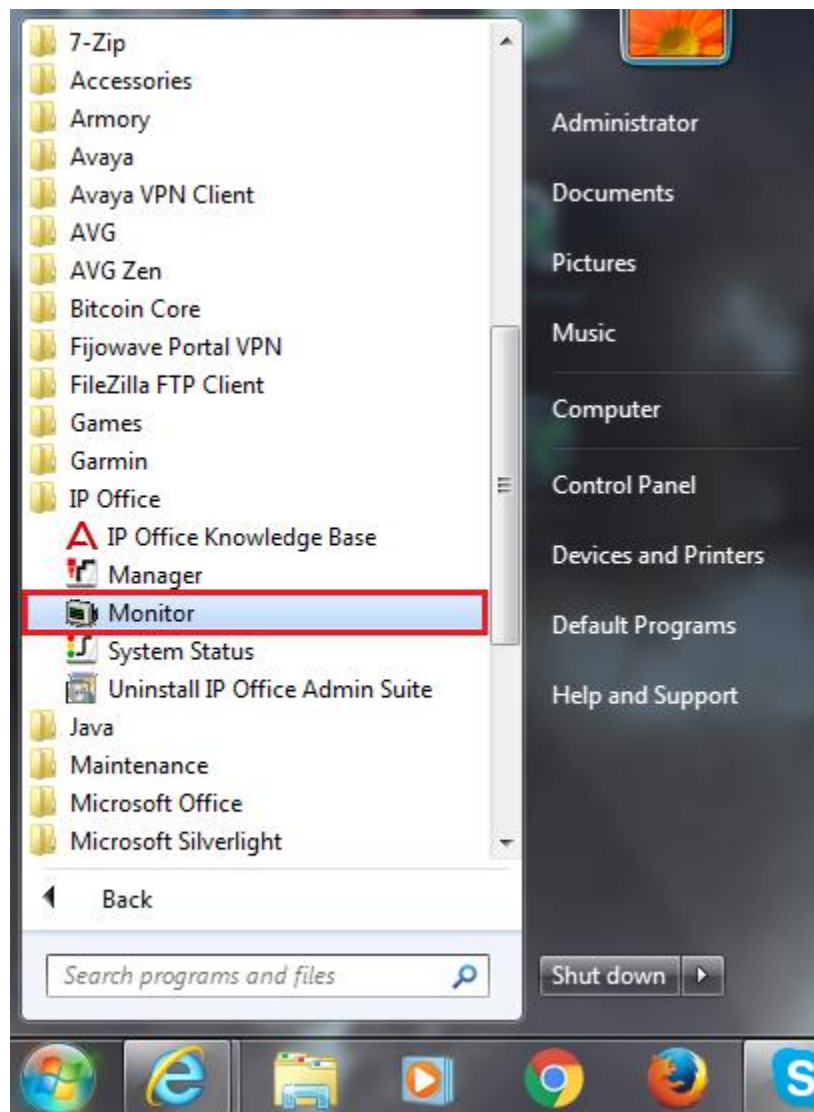
Recipient:

## 7. Verification Steps

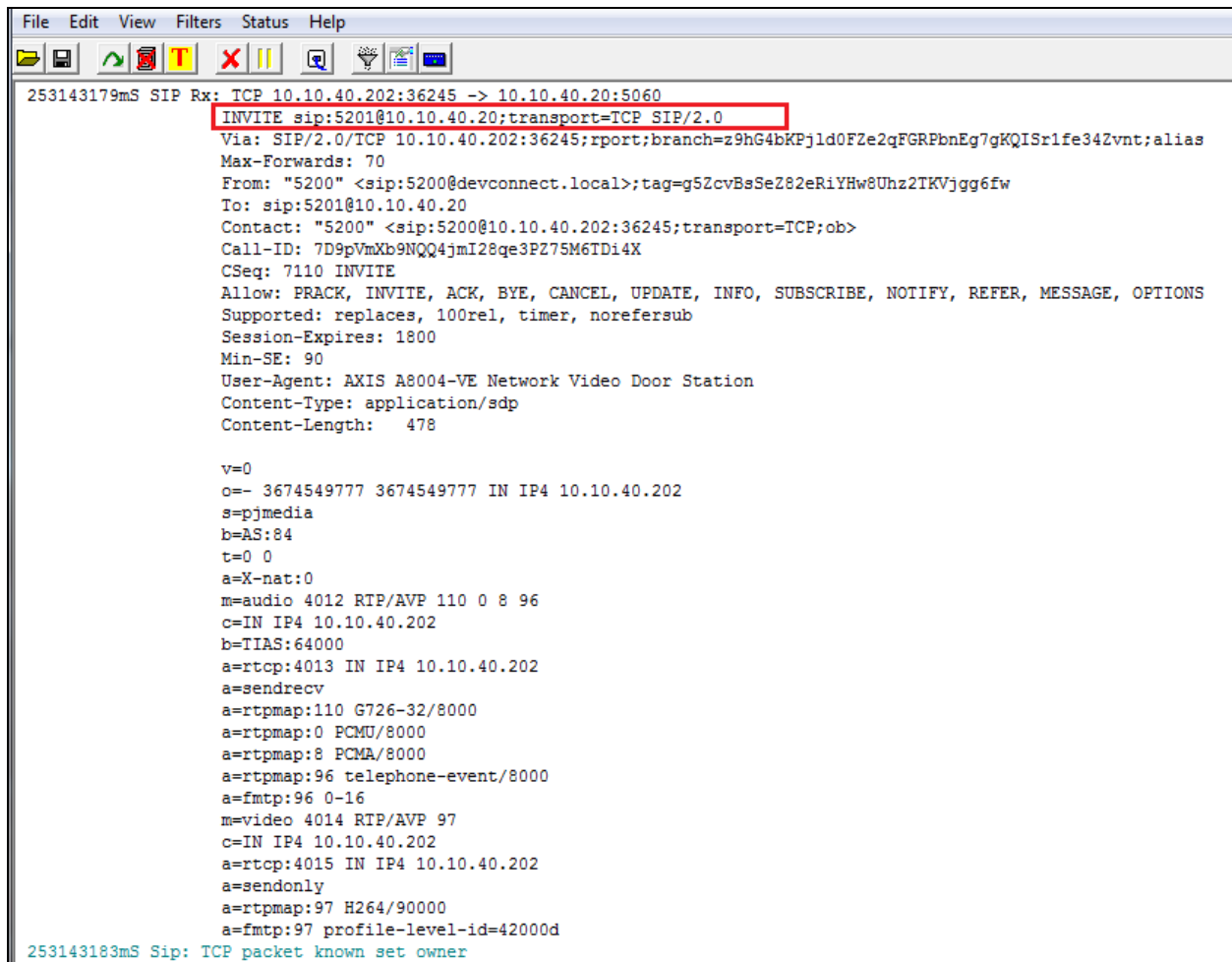
Pressing the Axis door phone button and answering the call from the IP Office set and ensuring there is two-way speech and video (where possible) is the ultimate verification that the product works and is connected and configured correctly. The steps below can also be taken to ensure that the Axis door 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 applies 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 a network traffic capture window with a menu bar (File, Edit, View, Filters, Status, Help) and a toolbar. The main display area shows a SIP message capture. The first line is '253143179mS SIP Rx: TCP 10.10.40.202:36245 -> 10.10.40.20:5060'. The second line, 'INVITE sip:5201@10.10.40.20:transport=TCP SIP/2.0', is highlighted with a red rectangle. The message body follows, including headers like 'Via: SIP/2.0/TCP 10.10.40.202:36245;rport;branch=z9hG4bKPjld0FZe2qFGRPbnEg7gKQISr1fe34Zvnt;alias', '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', and 'Content-Length: 478'. The message body is an SDP offer for audio and video. The third line is '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 left sidebar contains a navigation menu with the following items: Basic Setup, Video & Audio, VoIP (with sub-items: Overview, SIP Settings, VMS Settings, Account Settings, and DTMF Settings), Live View Config, Detectors, Applications, Events, Recordings, Languages, System Options, and About. The 'Account Settings' item is highlighted with a red box. The main content area is titled 'Account Settings' and features a table with the following data:

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

Below the table are three buttons: 'Add...', 'Modify...', and 'Remove'. At the bottom of the main content area, there is a 'Test SIP Call' section with the instruction 'Make a test call from the selected SIP account to the specified SIP address.' It includes a text input field labeled 'Enter SIP address: sip(s):extension@domain' and a 'Test call' button.

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 Axis A8004-VE Network Video Door Station web interface. 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, with 'System Options' expanded to show 'Logs & Reports' as the selected item. The main content area is titled 'Logs & Reports' and contains a note about log file usage, a 'Logs' section with buttons for System Log and Access Log, and a 'Reports' section. In the 'Reports' section, the 'View Server Report' button is highlighted with a red rectangle. Other report options include Download Server Report (with an option to include a snapshot), Parameter List, Connection List, and Crash Report. A link to the Axis Privacy statement is provided at the bottom of the main content area.

**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-6-1465556098.513016-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 R10.0. 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 R10.0 Manager, Document Number 15-601011
- [2] Avaya IP Office R10.0 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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