



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Axis Communications AB AXIS A8004-VE Network Video Door Station with Avaya Aura® Communication Manager 7.x and Avaya Aura® Session Manager 7.x – Issue 1.2

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 Aura® Communication Manager 7.x and Avaya Aura® Session Manager 7.x.

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 Aura® Communication Manager 7.x and Avaya Aura® Session Manager 7.x.

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 Aura® Communication Manager and Avaya Aura® Session Manager 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and the Axis 8105-E did not include use of any specific encryption features as requested by Axis.

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

All Test cases were successfully executed, with the following issue noted.

With Initial IP-IP Direct Media set to Y on the Communication Manager a video call is offered to the Equinox extension. If this call is answered on Equinox or Avaya Communicator using the “Video Icon” thus establishing the video call the video is not displayed on the Avaya endpoint. The Axis phone should have ideally added a bandwidth line for video media in the invite but it did not do so. Because there is no bandwidth allocated for the video portion of the call Communication Manager allocates 20kbps. This is because session level is 84kbps, 64kbps is given to audio and hence for video 20 kbps is remaining. This is not enough to sustain the video portion of the call. However, if the call is answered using the “Audio Icon” and then once that call is established the video icon is pressed the video is displayed correctly. This is because now the Avaya endpoint has initiated the video call and it has correctly allocated the correct bandwidth. The Avaya endpoint must initiate the video portion of the call as Axis does not allocate bandwidth correctly for the video call.

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 10** 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 Aura® Communication Manager and Avaya Aura® Session Manager.

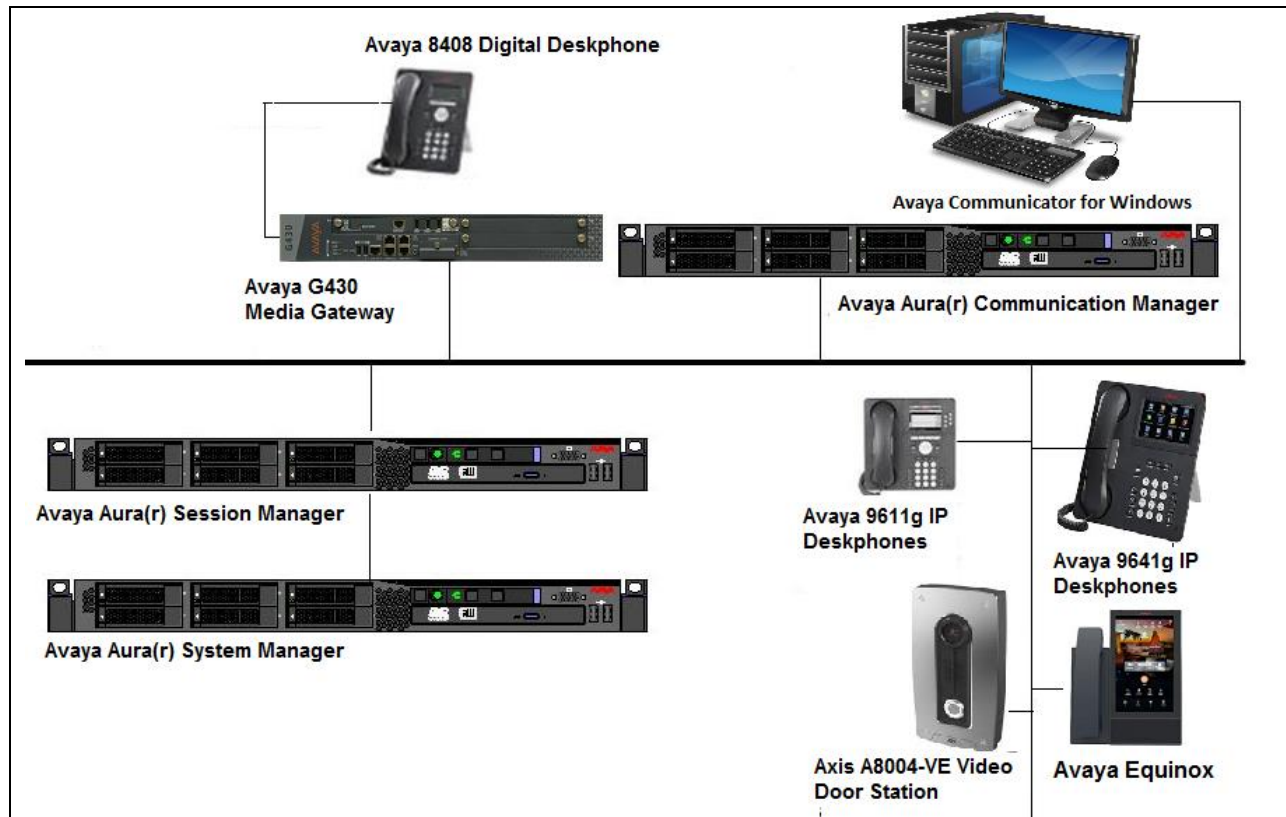


Figure 1: Connection of Axis Communications AB A8004-VE Network Video Door Station with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Version/Release
Avaya Aura® Communication Manager running on a virtual platform	R 7.0.1.1.0.441.23169 & R017x.01.0.532.0 R7.1.1.0.0 - FP1 -01.0.532.0-23985
Avaya Aura® Session Manager running on a virtual platform	R 7.0.1.1.701114 & R 7.1 FP2 Build No. – 7.1.2.0.712004
Avaya Aura® System Manager running on a virtual platform	R 7.0.1.2 Revision 7.0.1.2.075662 Service Pack 2 & R 7.1.2.0 Build No. - 7.1.0.0.1125193 Software Update Revision No: 7.1.2.0.057353 Feature Pack 2
Avaya 9611G Deskphone	H.323 Release 6.6029
Avaya 9611G Deskphone	SIP 7.0.1.1
Avaya 2420 Digital Deskphone	V 2.0
Avaya Communicator for Windows	V 2.1.3
Avaya Equinox for Windows	V3.3.1.60
Avaya Equinox for iOS	V3.3.1.1
Avaya Equinox for Vantage	V3.3.1
Axis Communications AB AXIS A8004-VE Network Video Door Station	Firmware Version 1.58.2.1 & 1.65.1.1

NOTE: Testing with Equinox was done after the initial test for a special request, and Axis engineers were not involved in the testing. Only Equinox was tested with the 7.1 versions of Aura, and with the 1.65.1.1 version of the A8004-VE. All other testing was done with previous releases.

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to a Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Define the Dial Plan

Note: Any settings not in **Bold** in the following screen shots may be left as default.

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

```
display system-parameters customer-options                               Page 1 of 10
                                OPTIONAL FEATURES

G3 Version: V16                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                         Module ID (MID): 1

                                USED
Platform Maximum Ports: 65000 290
Maximum Stations: 41000 44
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 14
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 41000 0
Maximum Survivable Processors: 313 0

(NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **System-Parameters Customer-Options form**, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

display system-parameters customer-options		Page	2 of 10
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	16
Maximum Concurrently Registered IP Stations:		18000	2
Maximum Administered Remote Office Trunks:		12000	0
Maximum Concurrently Registered Remote Office Stations:		18000	0
Maximum Concurrently Registered IP eCons:		414	0
Max Concur Registered Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		41000	1
Maximum Video Capable IP Softphones:		18000	4
Maximum Administered SIP Trunks:		24000	180
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0
Maximum Number of DS1 Boards with Echo Cancellation:		522	0
Maximum TN2501 VAL Boards:		128	0
Maximum Media Gateway VAL Sources:		250	0
Maximum TN2602 Boards with 80 VoIP Channels:		128	0
Maximum TN2602 Boards with 320 VoIP Channels:		128	0
Maximum Number of Expanded Meet-me Conference Ports:		300	0
(NOTE: You must logoff & login to effect the permission changes.)			

5.2. Define the Dial Plan

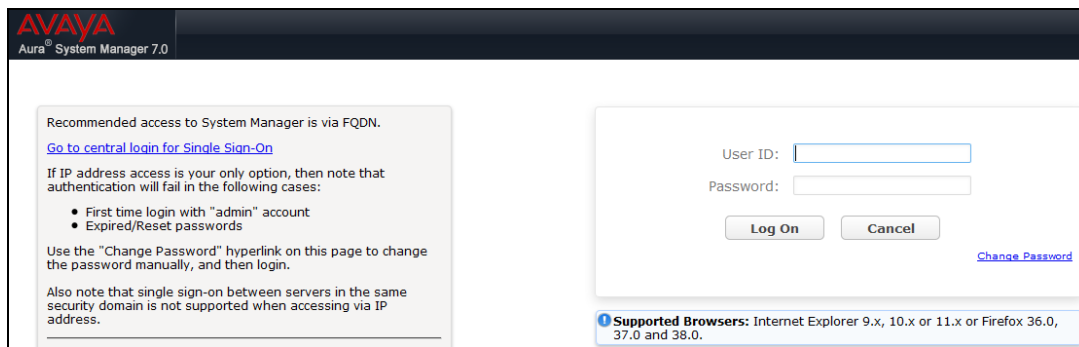
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are seven digits long and begin with **8**.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
Location: all						Percent Full: 1			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
8	7	ext							
9	3	fac							
*	3	fac							
#	3	fac							

6. Configure Avaya Aura® Session Manager

This section describes aspects of the Session Manager configuration required for interoperating with the Axis 8105-E Video Door Station. It is assumed that the Domains, Locations, SIP entities for each Session Manager, Communication Manager and Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured.

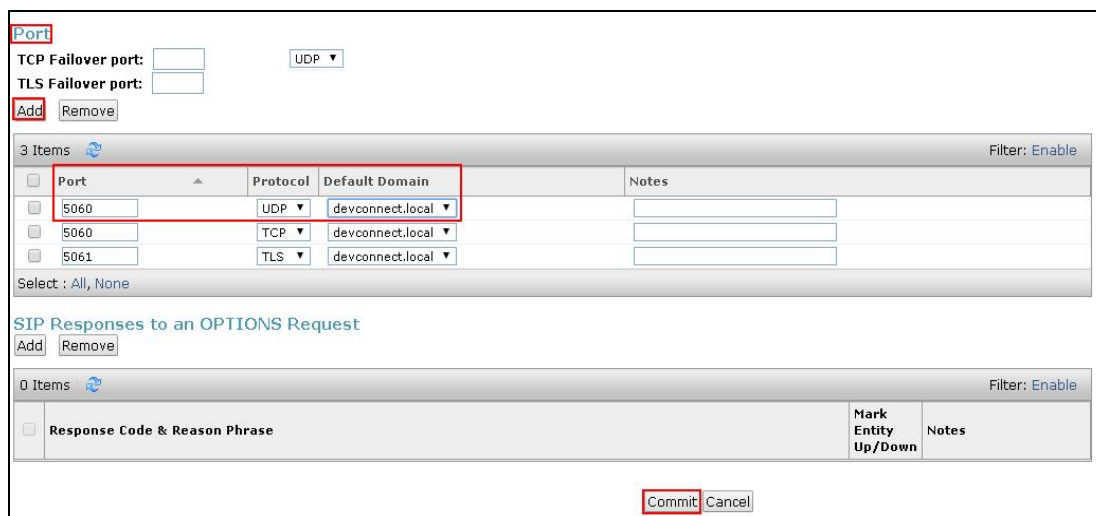
Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



The screenshot shows the Avaya Aura System Manager 7.0 Log On screen. It includes a header with the Avaya logo and 'Aura System Manager 7.0'. The main content area has a left sidebar with instructions: 'Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On. If IP address access is your only option, then note that authentication will fail in the following cases: • First time login with "admin" account • Expired/Reset passwords. Use the "Change Password" hyperlink on this page to change the password manually, and then login. Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.' The right sidebar contains a login form with fields for 'User ID' and 'Password', and buttons for 'Log On', 'Cancel', and 'Change Password'. At the bottom, a 'Supported Browsers' section lists Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

6.1. Check Avaya Aura® Session Manager ports for A8004-VE Registration

Each Session Manager Entity must be configured so that the Video Door Station can register to it using UDP/TCP. From the web interface click **Routing** → **SIP Entities** (not shown) select the Session Manager entity used for registration. Make sure that **TCP** and **UDP** entries are present. The UDP entry is highlighted below.



The screenshot shows the Avaya Aura System Manager 7.0 SIP Entities configuration screen. It includes a header with the Avaya logo and 'Aura System Manager 7.0'. The main content area has a left sidebar with instructions: 'Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On. If IP address access is your only option, then note that authentication will fail in the following cases: • First time login with "admin" account • Expired/Reset passwords. Use the "Change Password" hyperlink on this page to change the password manually, and then login. Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.' The right sidebar contains a login form with fields for 'User ID' and 'Password', and buttons for 'Log On', 'Cancel', and 'Change Password'. At the bottom, a 'Supported Browsers' section lists Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

Repeat accordingly on the alternative Session Manager.

6.2. Add A8004-VE User

The A8004-VE Video Door Station must be added as a user. A user must be added for each A8004-VE Video Door Station. Click **User Management** → **Manage Users** → **New** (not shown) and configure as following in the **Identity** tab.

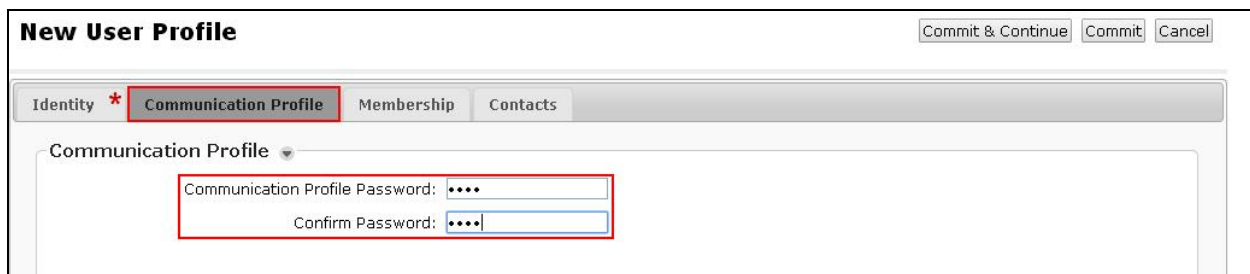
- **First Name and Last Name** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **8275060@devconnect.local**
- **Authentication Type** Select **Basic** from the drop down list
- **Password and Confirm Password** Enter and confirm a password

The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 7.0 interface. The 'Identity' tab is selected, and the form contains the following fields:

- User Provisioning Rule:** A dropdown menu.
- Identity:**
 - Last Name:** Station
 - Last Name (Latin Translation):** Station
 - First Name:** Video
 - First Name (Latin Translation):** Video
 - Middle Name:** Door
 - Description:** (empty text area)
 - Login Name:** 8275060@devconnect.local
 - User Type:** Basic (dropdown menu)
 - Password:** (masked with dots)
 - Confirm Password:** (masked with dots)
 - Localized Display Name:** (empty text field)

Buttons at the top right include 'Commit & Continue', 'Commit', and 'Cancel'. The top navigation bar shows 'Home / Users / User Management / Manage Users'.

Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the Video Door Station during login.



New User Profile Commit & Continue Commit Cancel

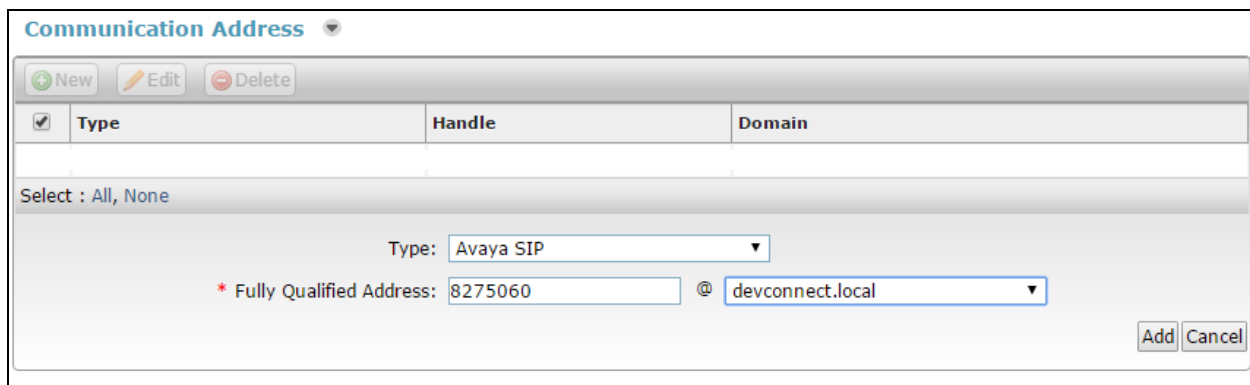
Identity * **Communication Profile** Membership Contacts

Communication Profile

Communication Profile Password:

Confirm Password:

Select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number as required, and select the appropriate **Domain** from the drop down list. Click **Add** when done.



Communication Address

New Edit Delete

<input checked="" type="checkbox"/>	Type	Handle	Domain
Select : All, None			

Type: Avaya SIP

* Fully Qualified Address: 8275060 @ devconnect.local

Add Cancel

Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence** and **Home Location**, from the respective drop down lists. The Primary Session Manager used was **SM71676**.

☒ **Session Manager Profile** ▼

SIP Registration

* Primary Session Manager

SM71676

Primary	Secondary	Maximum
21	2	23

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

1 ▼

Block New Registration When Maximum Registrations Active?

☐

Application Sequences

Origination Sequence

CM1627_seq ▼

Termination Sequence

CM1627_seq ▼

Call Routing Settings

* Home Location

Devconnect ▼

Conference Factory Set

(None) ▼

Call History Settings

Enable Centralized Call History?

☐

Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **System** Select the relevant Communication Manager SIP Entity from the drop down list
- **Profile Type** Select **Endpoint** from the drop down list
- **Extension** Enter the required extension number, in this case **8275060**
- **Template** Select 9611SIP_DEFAULT_CM_7_0 from the drop down list
- **Port** Enter **IP**

Click on **Endpoint Editor**.

☒ **CM Endpoint Profile** ▼

* System

CM71627 ▼

* Profile Type

Endpoint ▼

Use Existing Endpoints

☐

* Extension

8275060

Endpoint Editor

Template

9611SIP_DEFAULT_CM_7_0 ▼

Set Type

9611SIP

Security Code

•••••

Port

S00009

Voice Mail Number

Preferred Handle

(None) ▼

Calculate Route Pattern

☐

Sip Trunk

aar

Enhanced Callr-Info display for 1-line phones

☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User

☒

Override Endpoint Name and Localized Name

☒

Allow H.323 and SIP Endpoint Dual Registration

☐

Click on the **Feature Options** tab. The screen shot below shows the Feature Options that were used during compliance testing.

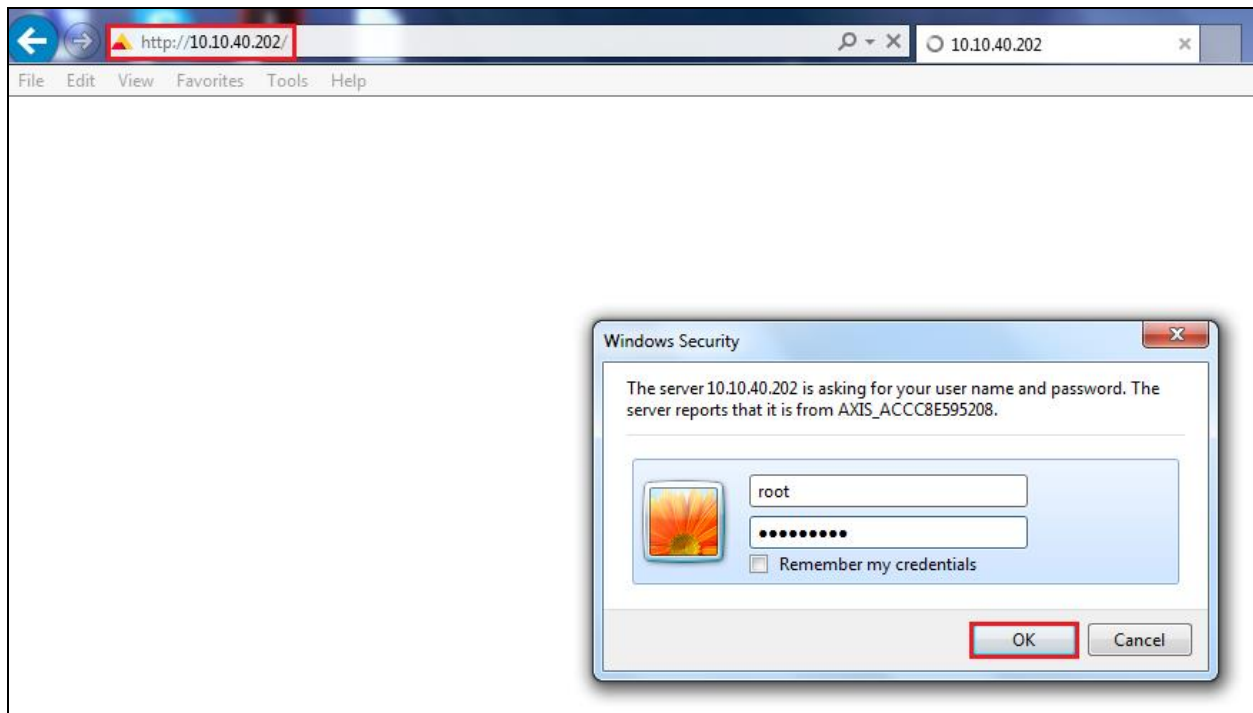
General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)		Enhanced Call Fwd (E)		Button Assignment (B)																					
Group Membership (M)																															
Active Station Ringing MWI Served User Type Per Station CPN - Send Calling Number IP Phone Group ID Remote Soft Phone Emergency Calls LWC Reception AUDIX Name Speakerphone Short/Prefixed Registration Allowed EC500 State				single None None spe enabled				Auto Answer Coverage After Forwarding Display Language Hunt-to Station Loss Group Survivable COR Time of Day Lock Table Voice Mail Number Music Source				none system english 19 internal None 																			
Features <table border="0"> <tr> <td><input type="checkbox"/> Always Use</td> <td><input type="checkbox"/> Idle Appearance Preference</td> </tr> <tr> <td><input type="checkbox"/> IP Audio Hairpinning</td> <td><input type="checkbox"/> IP SoftPhone</td> </tr> <tr> <td><input checked="" type="checkbox"/> Bridged Call Alerting</td> <td><input checked="" type="checkbox"/> LWC Activation</td> </tr> <tr> <td><input type="checkbox"/> Bridged Idle Line Preference</td> <td><input type="checkbox"/> CDR Privacy</td> </tr> <tr> <td><input checked="" type="checkbox"/> Coverage Message Retrieval</td> <td><input checked="" type="checkbox"/> Direct IP-IP Audio Connections</td> </tr> <tr> <td><input type="checkbox"/> Data Restriction</td> <td><input type="checkbox"/> H.320 Conversion</td> </tr> <tr> <td><input checked="" type="checkbox"/> Survivable Trunk Dest</td> <td><input checked="" type="checkbox"/> IP Video</td> </tr> <tr> <td><input type="checkbox"/> Bridged Appearance Origination Restriction</td> <td><input type="checkbox"/> Per Button Ring Control</td> </tr> <tr> <td><input checked="" type="checkbox"/> Restrict Last Appearance</td> <td></td> </tr> <tr> <td><input type="checkbox"/> Turn on mute for remote off-hook attempt</td> <td></td> </tr> </table>												<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference	<input type="checkbox"/> IP Audio Hairpinning	<input type="checkbox"/> IP SoftPhone	<input checked="" type="checkbox"/> Bridged Call Alerting	<input checked="" type="checkbox"/> LWC Activation	<input type="checkbox"/> Bridged Idle Line Preference	<input type="checkbox"/> CDR Privacy	<input checked="" type="checkbox"/> Coverage Message Retrieval	<input checked="" type="checkbox"/> Direct IP-IP Audio Connections	<input type="checkbox"/> Data Restriction	<input type="checkbox"/> H.320 Conversion	<input checked="" type="checkbox"/> Survivable Trunk Dest	<input checked="" type="checkbox"/> IP Video	<input type="checkbox"/> Bridged Appearance Origination Restriction	<input type="checkbox"/> Per Button Ring Control	<input checked="" type="checkbox"/> Restrict Last Appearance		<input type="checkbox"/> Turn on mute for remote off-hook attempt	
<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference																														
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<input checked="" type="checkbox"/> Restrict Last Appearance																															
<input type="checkbox"/> Turn on mute for remote off-hook attempt																															

7. 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 appropriate credentials and click on **OK**.



Please refer to Axis Communications documentation listed in **Section 10** 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 Session Manager.

7.1. 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 (for example 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. 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.

The screenshot displays the 'Setup' page 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 (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 'SIP Settings' item is highlighted with a red box. The main content area is titled 'SIP Settings' and includes a 'SIP Setup Assistant' section with a 'Start...' button. Below this are three sections: 'SIP Settings' with a checked 'Enable SIP' checkbox; 'Incoming SIP Calls' with a checked 'Allow incoming SIP calls' checkbox; and 'Port Settings' with input fields for 'SIP port' (5060), 'SIP TLS port' (5061), and 'RTP start port' (4000). The 'NAT Traversal' section has unchecked checkboxes for 'Enable ICE', 'Enable STUN', and 'Enable TURN'. The 'Audio Codec Settings' section features a list of 'Available codecs' (opus, L16/16000, L16/8000, speex/16000, speex/8000, and G.726-32) and a 'Selected codecs' list containing 'PCMU (8000 Hz)' and 'PCMA (8000 Hz)'. Arrows between the lists allow for moving items, and a 'Save' button is at the bottom.

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

7.2. Configure Account

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

AXIS A8004-VE Network Video Door Station [Live View](#) | [Setup](#) | [Help](#)

Account Settings ?

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: [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.
- **Password:** Enter the password for the SIP user created.
- **Caller ID:** This should be the extension number created
- **Domain Name:** The domain used for SIP Telephony.
- **Registrar address:** The IP address of the Session Manager.

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: 8275060

☒ Use User ID as Authentication ID

Authentication ID: 8275060

Password:

Caller ID: 8275060

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 Session Manager. 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", and "Video". The "Network" tab is currently selected and highlighted with a red box. Below the tabs, the "Transport Settings" section includes a checkbox for "Enable SIPs", a "Transport mode:" dropdown menu set to "TCP", and a checkbox for "Allow port update messages through MWI". The "Proxy Settings" section contains a table with two columns: "Address" and "Username". The table is currently empty, and there are up and down arrow buttons to its right. Below the table is an "Add..." button. At the bottom of the form, there is an "Account Status" section. The "Save" button is highlighted with a red box, and a "Cancel" button is located next to it.

7.3. 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 sub-items are 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 includes a help icon. Below the title is a section for 'DTMF Configuration for SIP Accounts'. It shows a list of accounts under the heading 'Peer-to-peer accounts (No local accounts)'. One account, '500V2 Door (5200)', is selected and highlighted with a blue bar and a red box. To the right of this account is an edit icon (pencil). Below the account list, there are two checked options: 'DTMF using SIP INFO (RFC2976)' and 'DTMF using RTP (RFC2833)'. At the bottom, there 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)

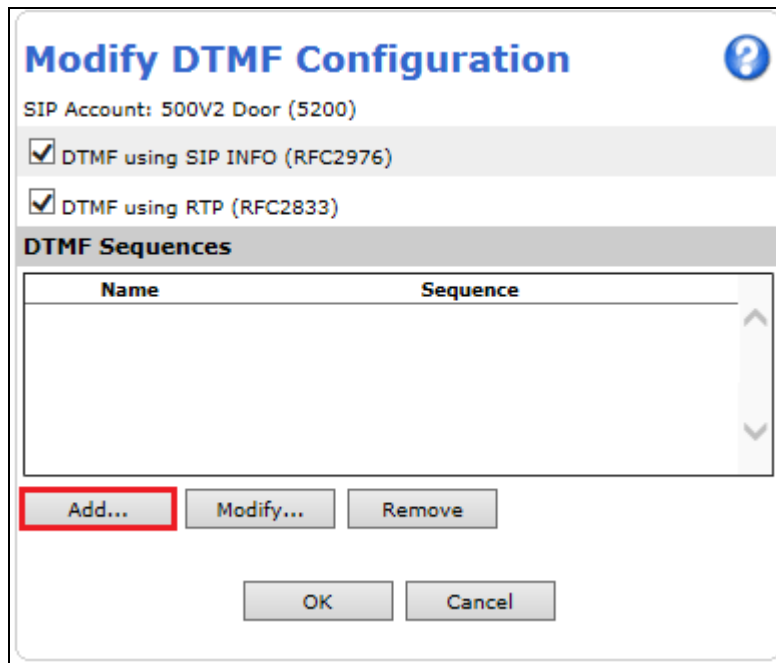
☒ DTMF using SIP INFO (RFC2976)

☒ 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)

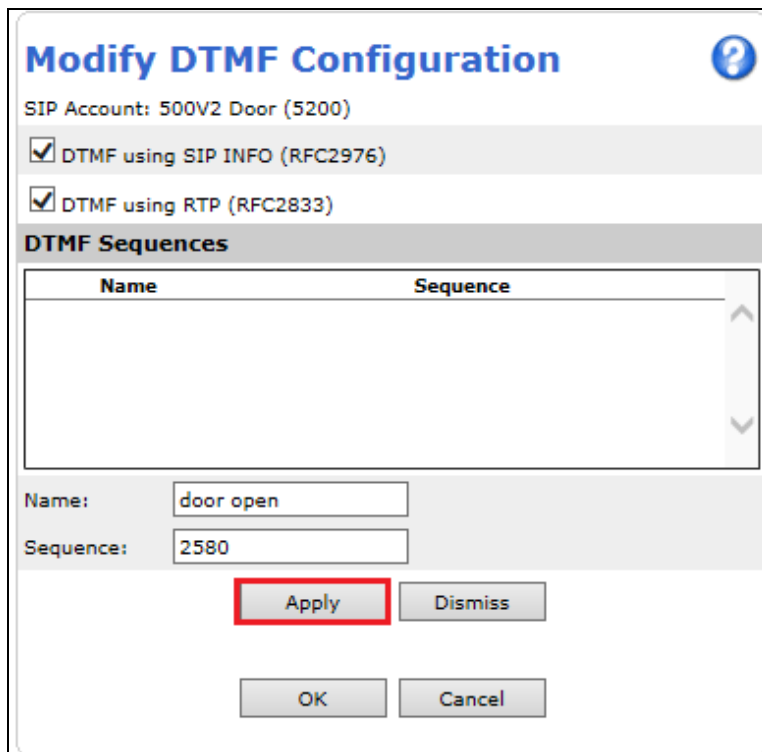
DTMF Sequences

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

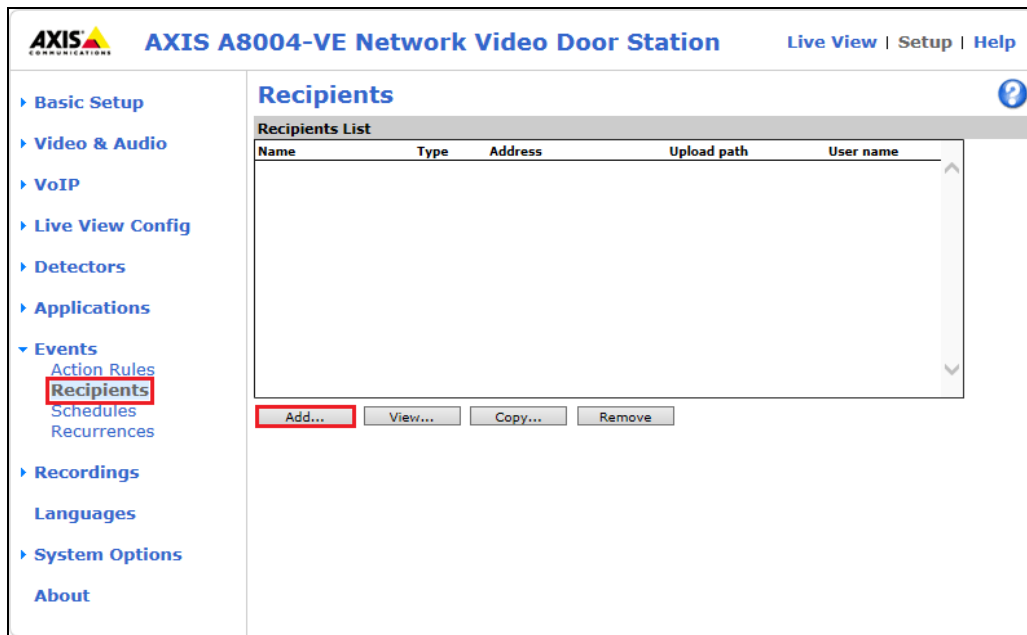
OK Cancel

7.4. 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.

7.4.1. 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 Communication Manager 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 contains the following fields and controls:

- Name:** Text input field with the value '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 with the value '5201@10.10.40.20'.
- Test:** A section with a description: '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 Communication Manager 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. The table contains eight entries, all of type 'SIP'. 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	-	

7.4.2. Modify Action Rule

An action rule can now be modified to include the participant created in **Section 7.4.1**. 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' item under 'Events' is selected and highlighted with a red box. The main area displays a table titled 'Action Rule List' with columns: Name, Trigger, Schedule, Action, and Recipient. The table contains eight entries. The entry 'BUTTON: VMS call' is selected and highlighted with a blue background. Below the table are buttons for 'Add...', 'Copy...', 'Modify...' (highlighted with a red box), 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	-
<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 7.4.1**. 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:

8. Verification Steps

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

8.1. Verify Registration to Avaya Aura® Session Manager

From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status**→**User Registrations** (not shown). The 8105-E Video Station is listed and a tick under **Registered** for the Session Manager it is registered to.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 7.0', and a 'Last Logged on at December 16, 2016 8:35' timestamp. The left sidebar contains a menu with 'Session Manager' selected, showing sub-items like 'Dashboard', 'Session Manager Administration', 'Communication Profile Editor', 'Network Configuration', 'Device and Location Configuration', and 'Application Configuration'. The main content area is titled 'User Registrations' and includes a breadcrumb trail: 'Home / Elements / Session Manager / System Status / User Registrations'. Below the title, there are controls for 'View' (Default), 'Force Unregister', 'AST Device Notifications', 'Reboot', 'Reload', 'Failback', and a timestamp 'As of 11:46 AM'. A table below shows 24 items, with the first row having a checked 'Prim' status under the 'Registered' column. The table columns are: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv).

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered (Prim, Sec, Surv)			
<input type="checkbox"/>	Show	8275060@devconnect.local	Video	Station	---	10.10.16.129	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	H175	Station	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

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

Log in to the door phone as per **Section 7**. Navigate to **VoIP → Account Settings** in the left window and the registration information is displayed in the main window as shown below. The green lights show a successful registration of 8275060. A 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 "AXIS A8004-VE Network Video Door Station", and links for "Live View", "Setup", and "Help". The left sidebar contains a menu with categories: 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" section is active, showing a table with the following data:

Name	SIP address	Transport	Default	Reg. status
8275060 (8275060)	8275060 <sip:8275060@devconnect.io cal>	UDP	✓	✓

Below the table are buttons for "Add...", "Modify...", and "Remove". A "Test SIP Call" section follows, with the instruction "Make a test call from the selected SIP account to the specified SIP address." and a text input field containing "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 for the issue. 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 header includes the AXIS logo, the product name, and navigation links for Live View, Setup, and Help. A left sidebar contains a tree view of configuration options, with 'Logs & Reports' highlighted under 'System Options'. The main content area is titled 'Logs & Reports' and includes a help icon. It contains a note about loading times, sections for Logs (System Log, Access Log) and Reports (View Server Report, Download Server Report, Parameter List, Connection List, Crash Report), and a link to the 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](#).

Left Sidebar Navigation:

- Basic Setup
- Video & Audio
- VoIP
- Live View Config
- Detectors
- Applications
- Events
- Recordings
- Languages
- System Options
 - Security
 - Date & Time
 - Network
 - Storage
 - Ports & Devices
 - Maintenance
 - Support
 - Support Overview
 - System Overview
 - Logs & Reports
 - Information**
 - Advanced
- About

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
2016-06-10T11:44:10.495+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
2016-06-10T11:55:04.591+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 cpuctt
<5>Linux version 3.18.0 (svcc@eater-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"
```

9. 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 Aura® Communication Manager R7.1 and Avaya Aura® Session Manager R7.1. Please refer to **Section 2.2** for test results and observations.

10. 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] *Administering Avaya Aura® Communication Manager, Release 7.1, August 7 Document Number 03-300509, Issue 1.*

[2] *Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.0, August 2015, Document Number 555-245-205, Issue 1.*

[3] *Administering Avaya Aura® Session Manager, Release 7.1 Issue 1 August 2017*

[4] *Administering Avaya Aura® System Manager, Release 7.1, Issue 1, August, 2017*

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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