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.

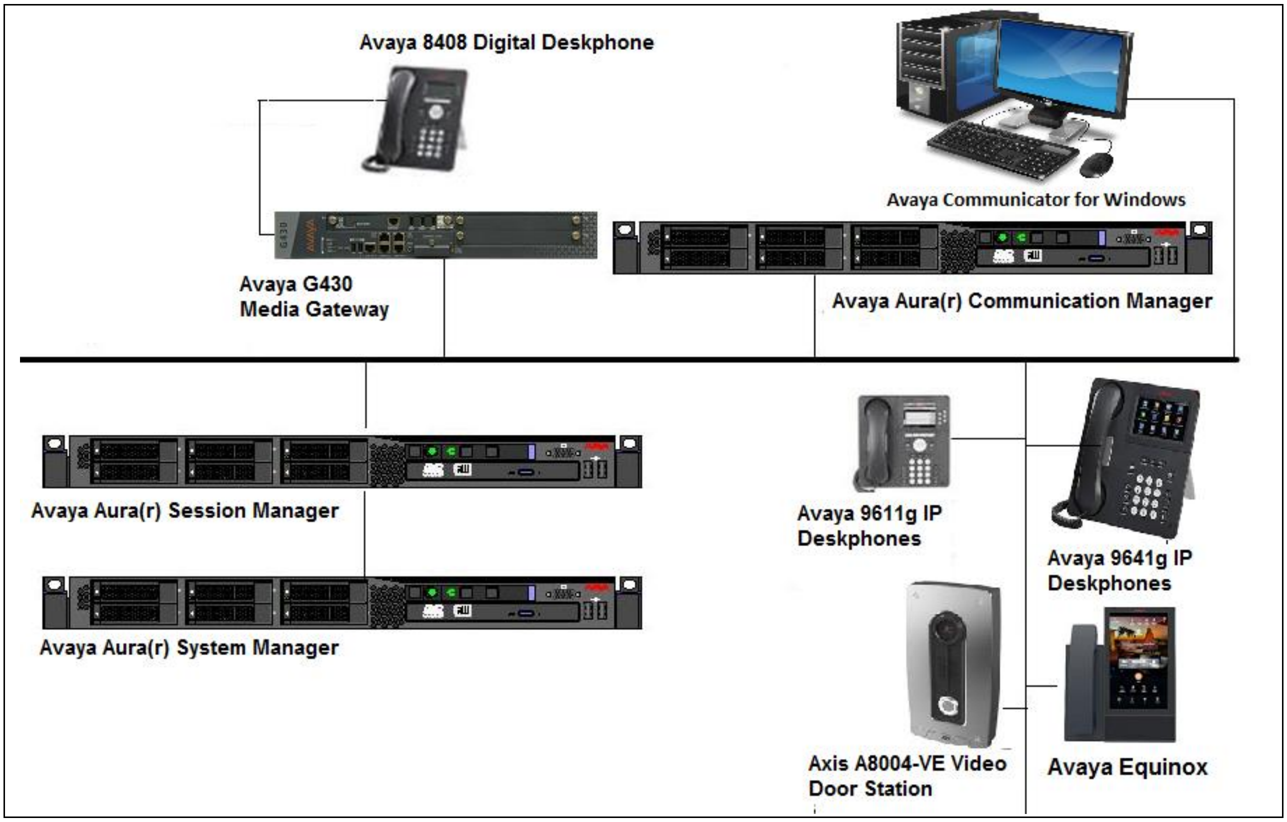


Figure 1: Connection of Axis Communications AB A8004-VE Network Video Door Station with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

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**.

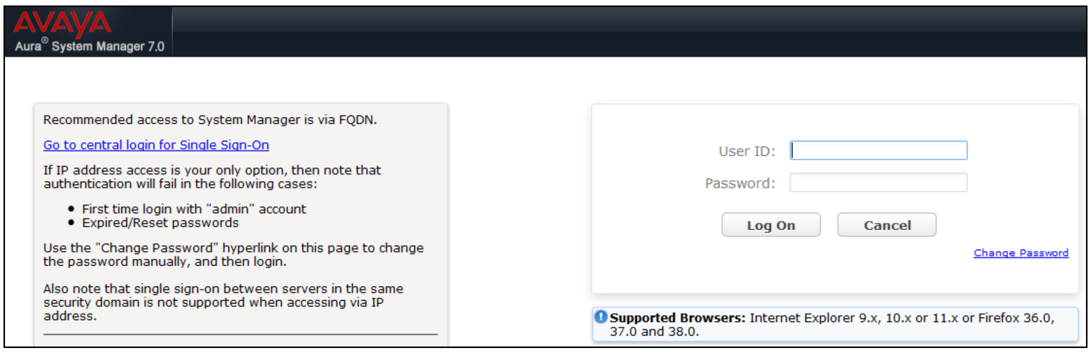
```
change dialplan analysis                                   Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                  Location: all          Percent Full: 1

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   8            7   ext
   9            3   fac
   *            3   fac
   #            3   fac
```

6. Configure Avaya Aura® Session Manager

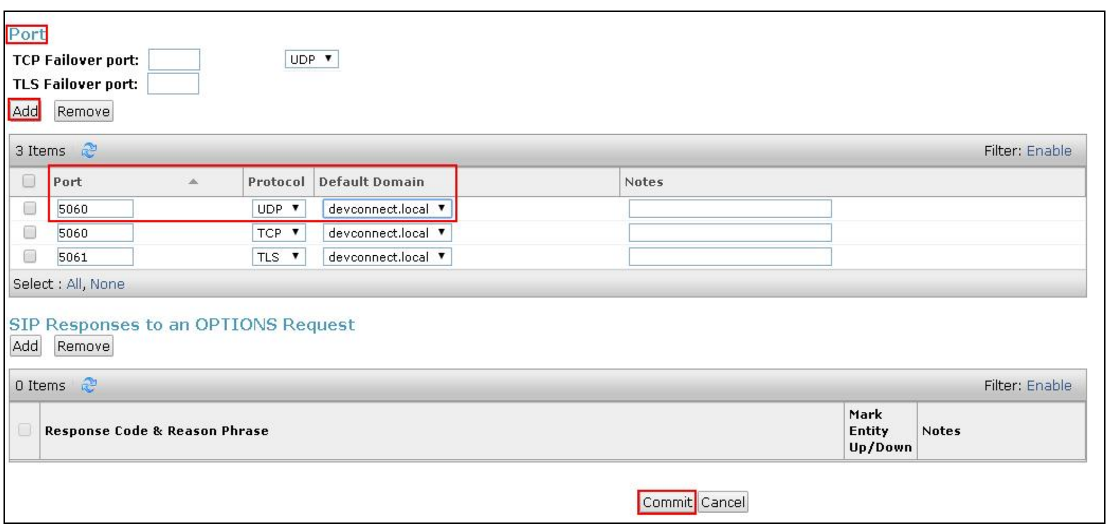
This section describes aspects of the Session Manager configuration required for interoperating with the Axis 8105-E Video Door Station. It is assumed that the Domains, Locations, SIP entities for each Session Manager, Communication Manager and Entity Links, Routing Policies, Dial Patterns and Application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



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.

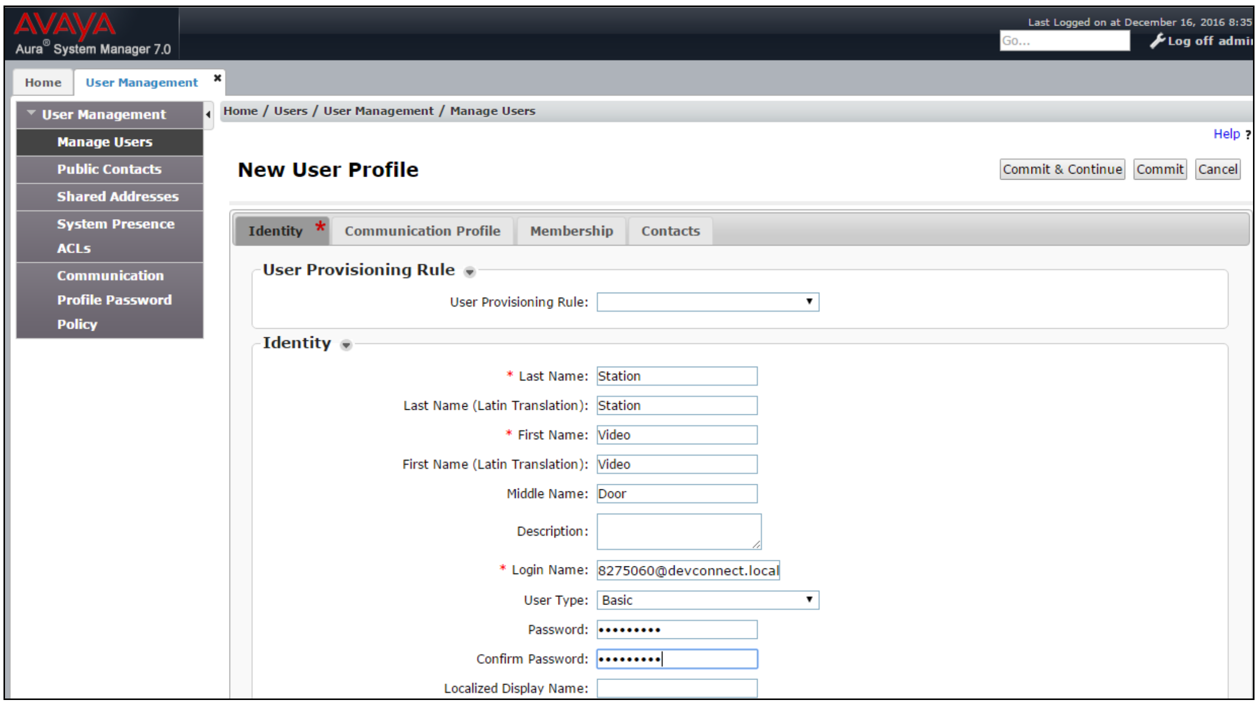


Repeat accordingly on the alternative Session Manager.

6.2. Add A8004-VE User

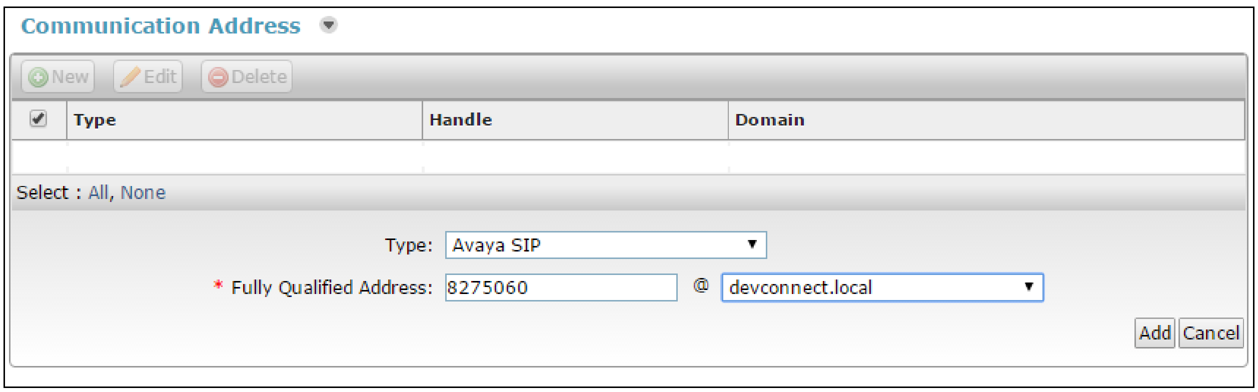
The A8004-VE Video Door Station must be added as a user. A user must be added for each A8004-VE Video Door Station. Click **User Management → Manage Users → New** (not shown) and configure as following in the **Identity** tab.

- **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

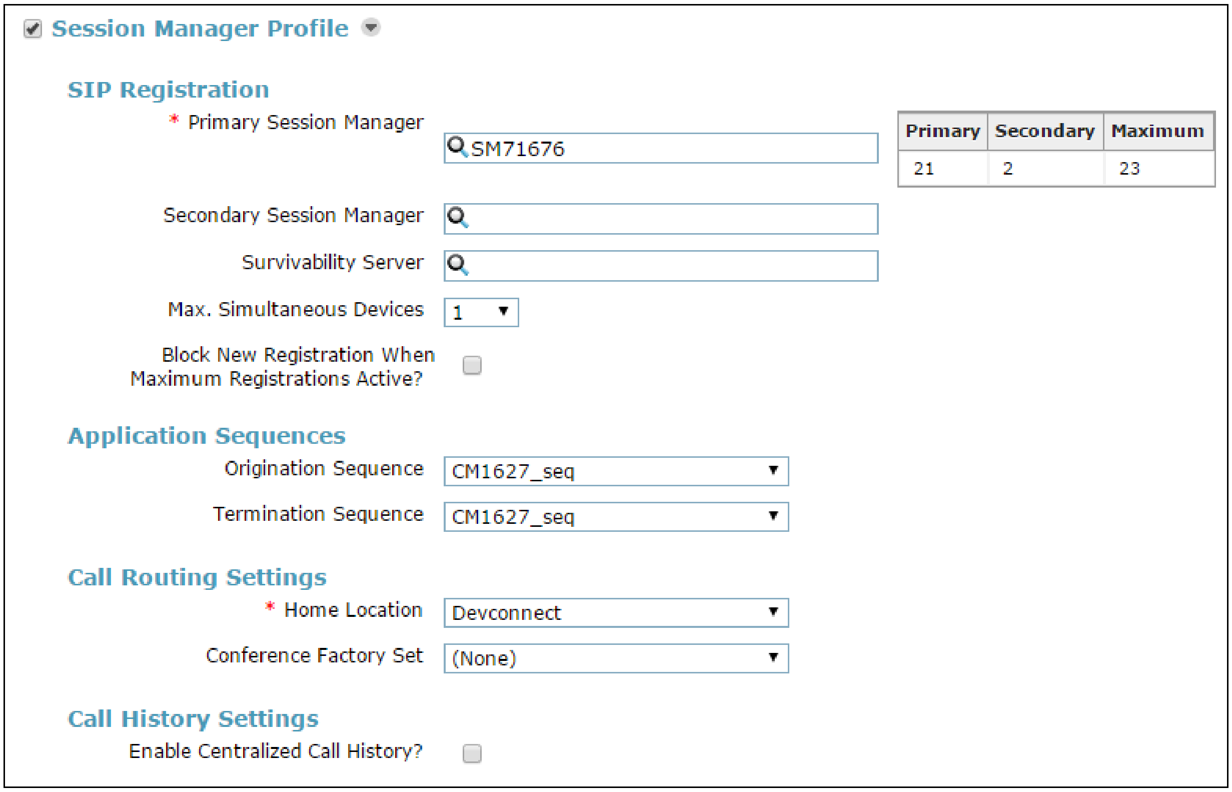


Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the Video Door Station during login.

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.



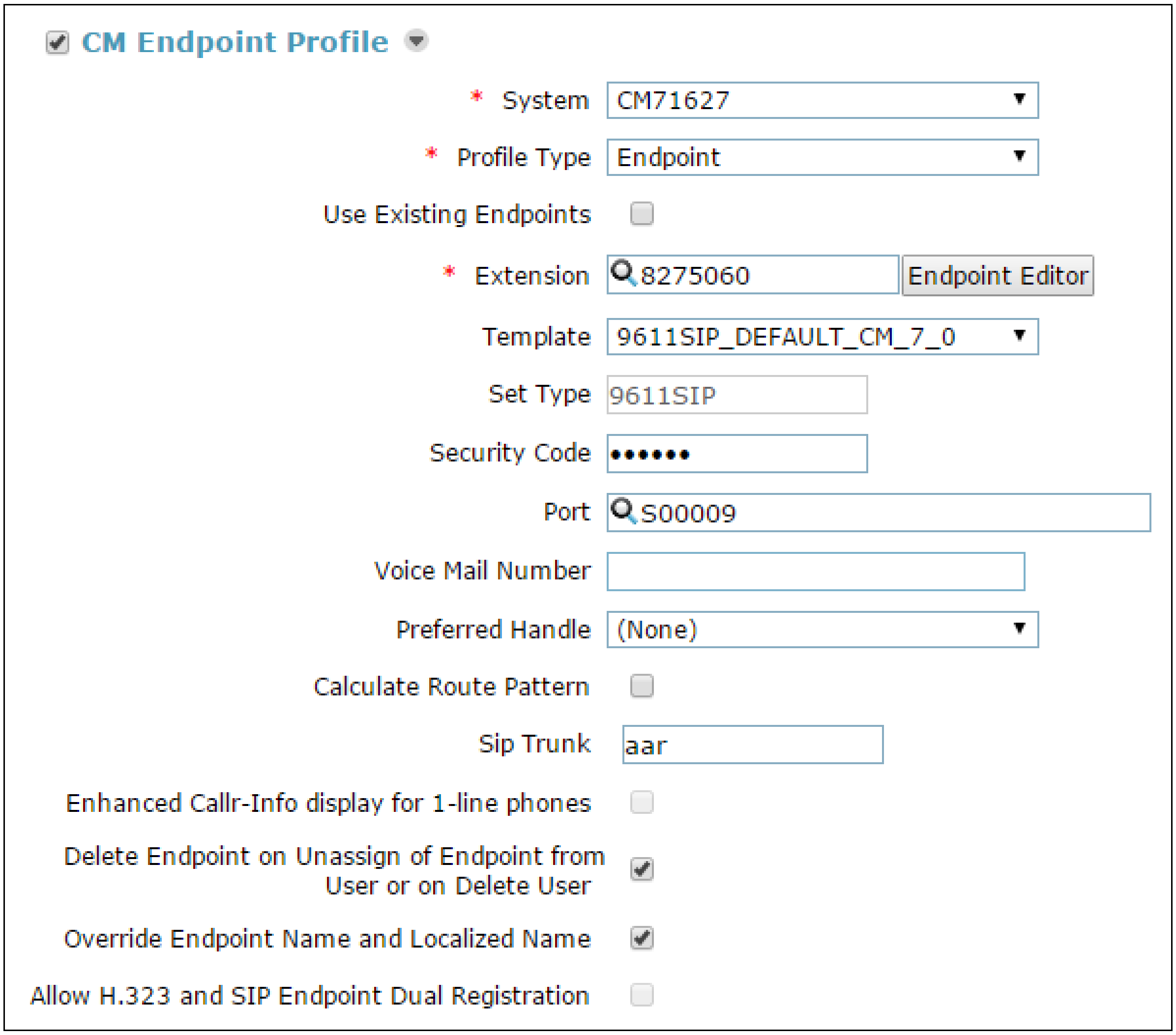
Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop down lists. The Primary Session Manager used was **SM71676**.



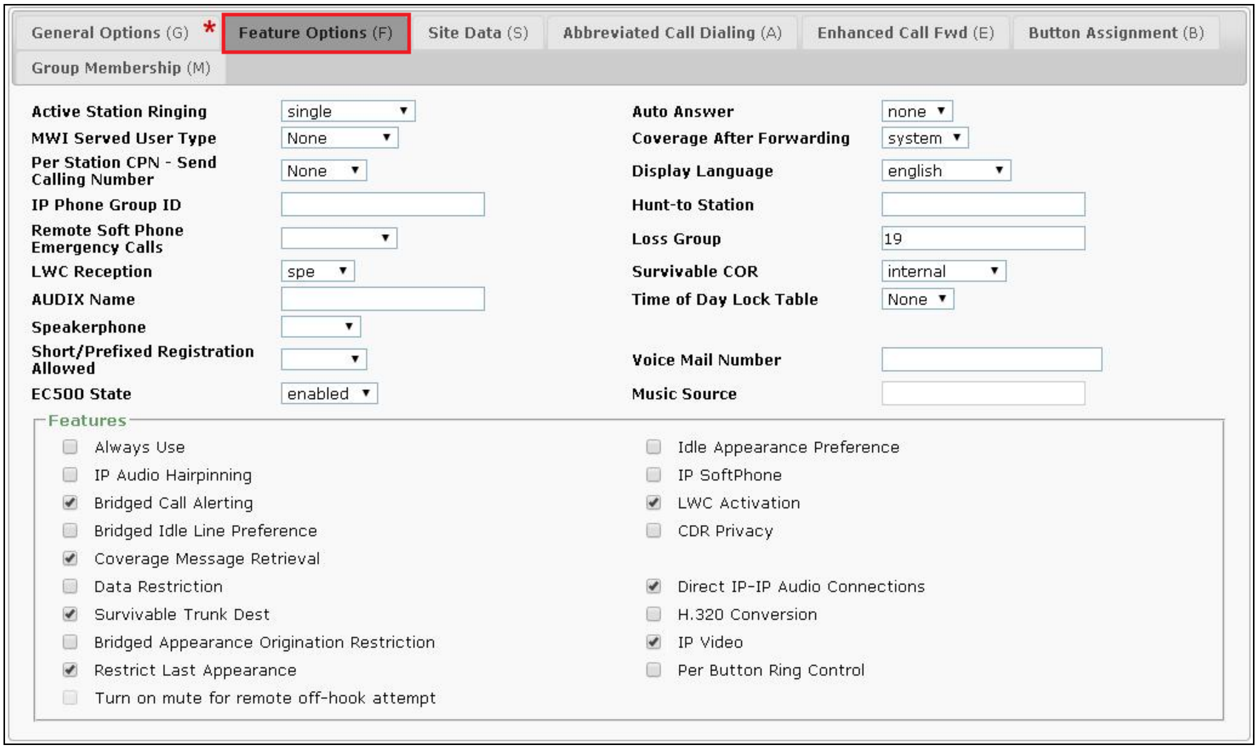
Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **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**.



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

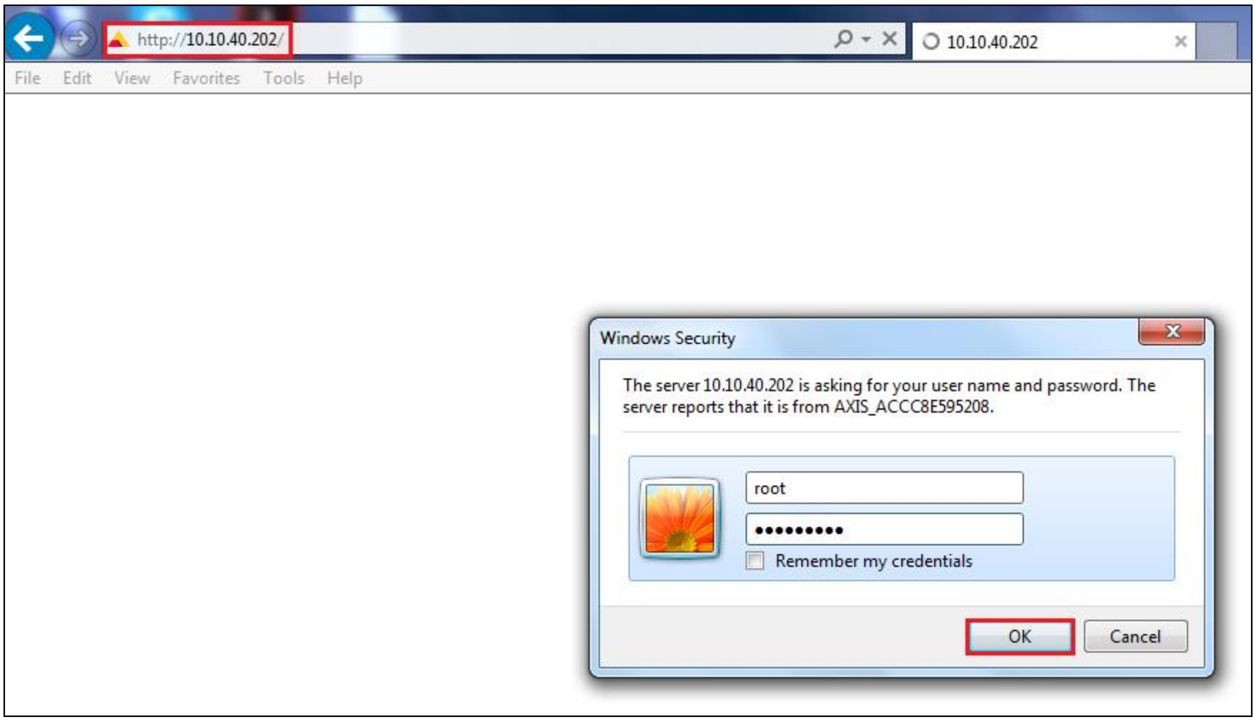


7. Configure AXIS A8004-VE Network Video Door Station

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

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

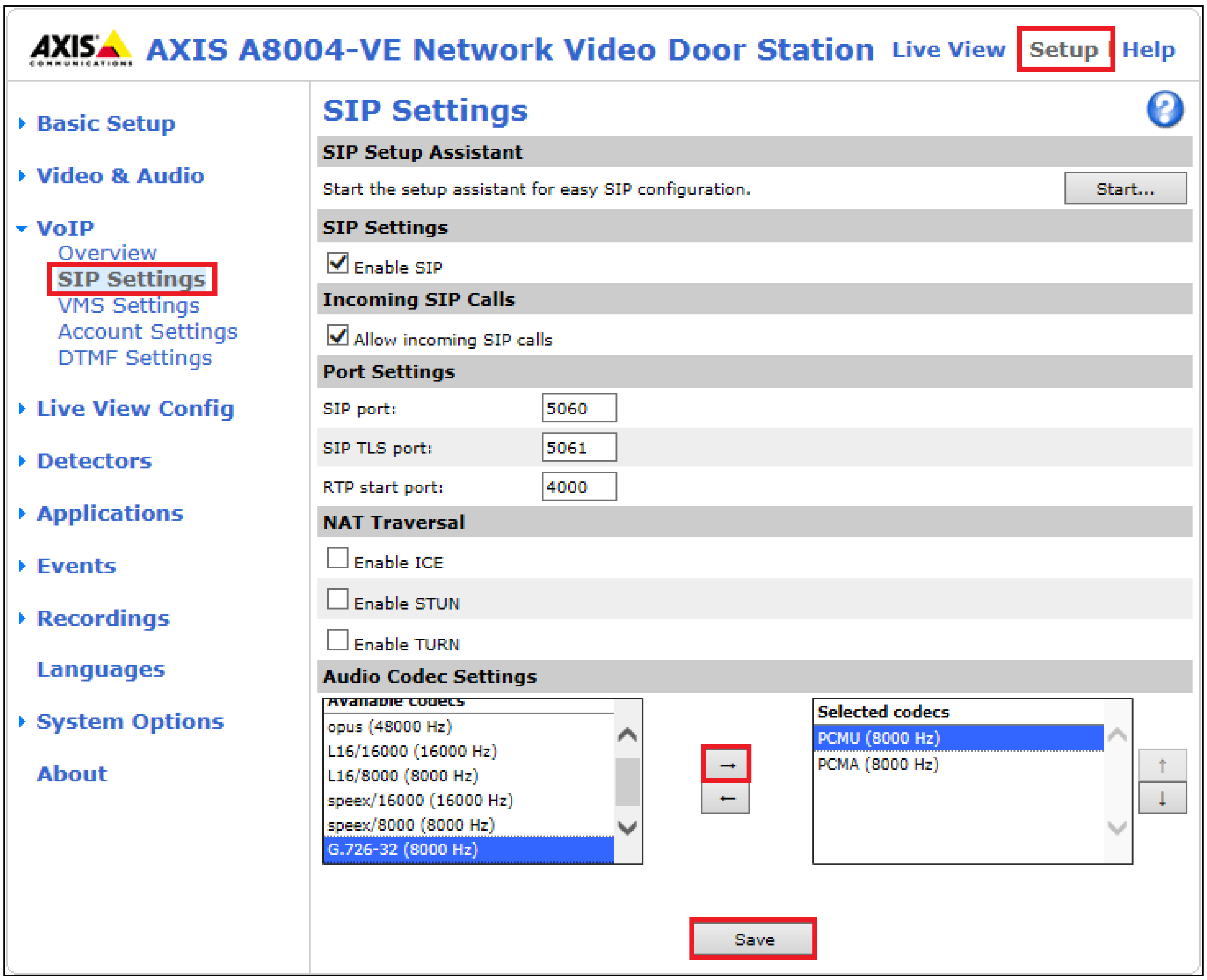
Open a web session to the IP address of the Axis door phone, enter the appropriate credentials and click on **OK**.



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.

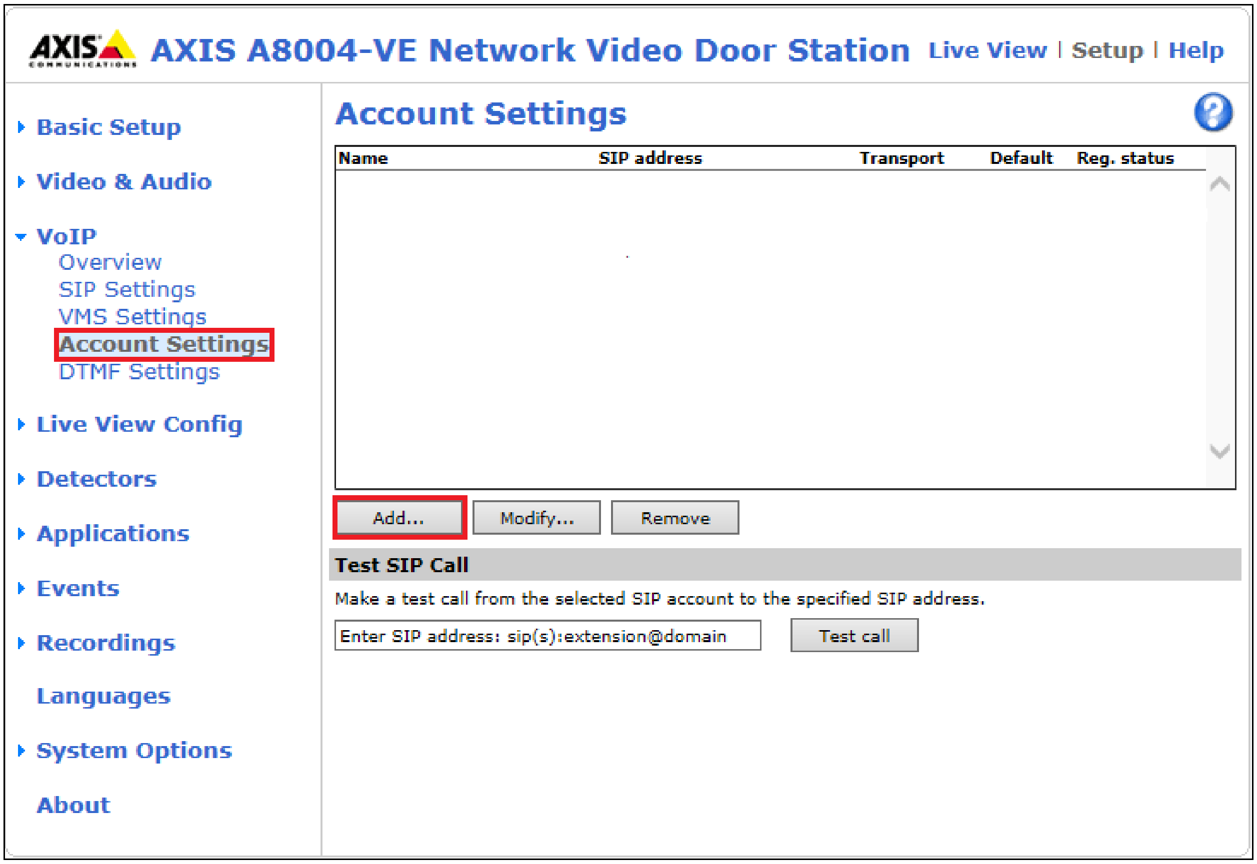
7.1. Configure SIP Settings

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.



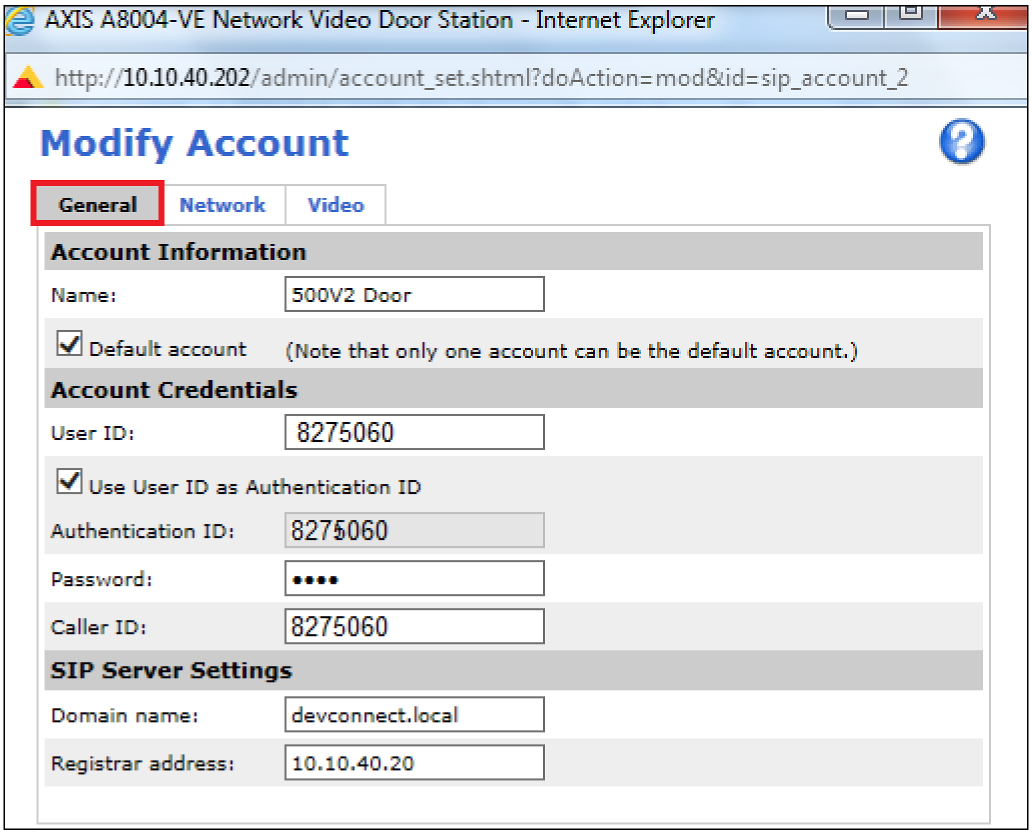
7.2. Configure Account

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

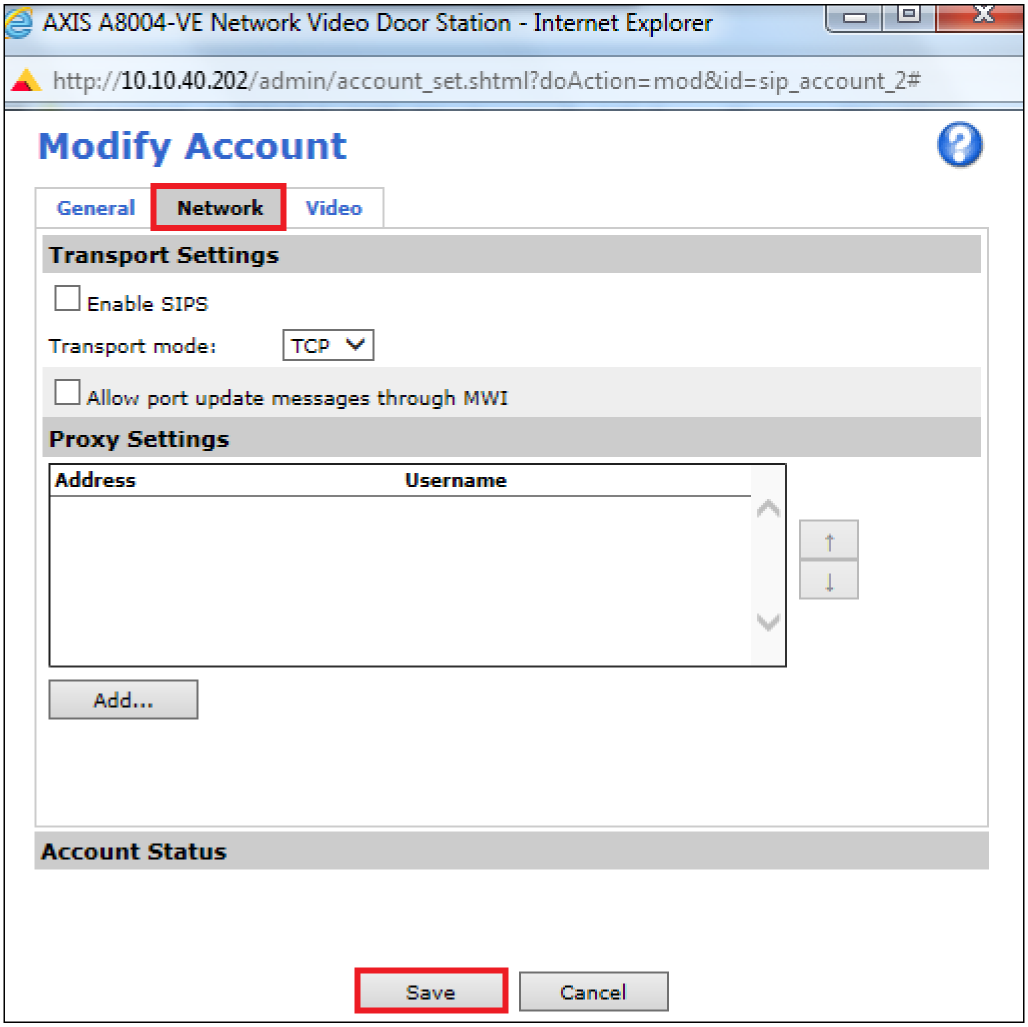


Enter the following details under the **General** tab:

- **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.

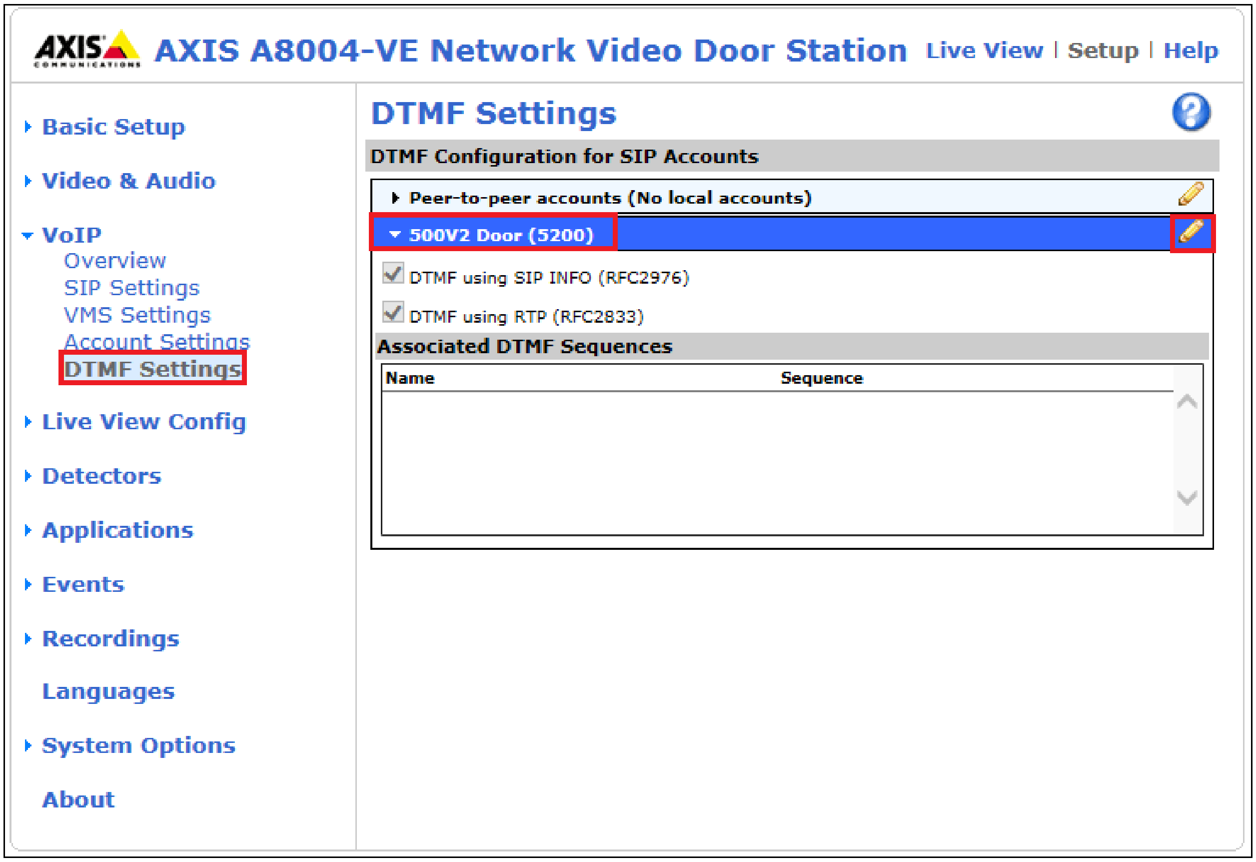


Select the **Network** tab and select the transport mode to be used, this can be UDP, **TCP** or TLS, all three protocols were tested and work correctly with Session Manager. Click on **Save** to save the Account information.

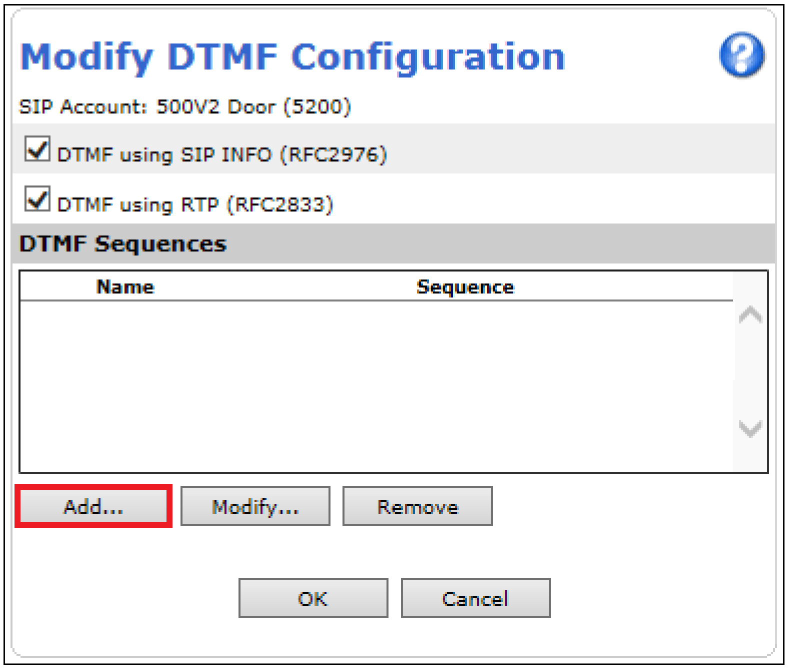


7.3. Configure DTMF Settings

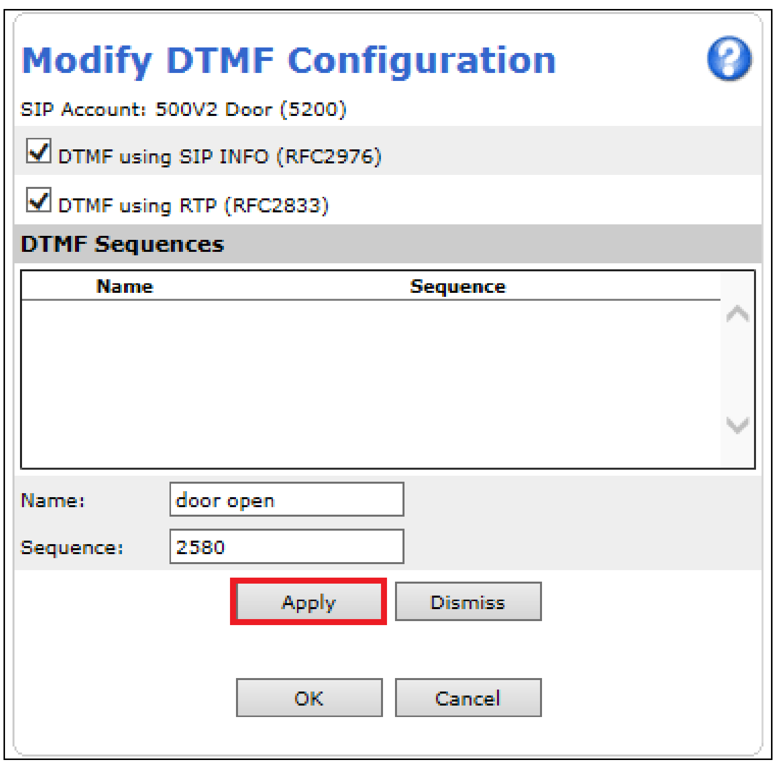
Staying within the **VoIP** menu on the left window, select **DTMF Settings**. In the main window select the SIP account that was created in **Section 6.2** and click on the edit icon, as shown below.



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.



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

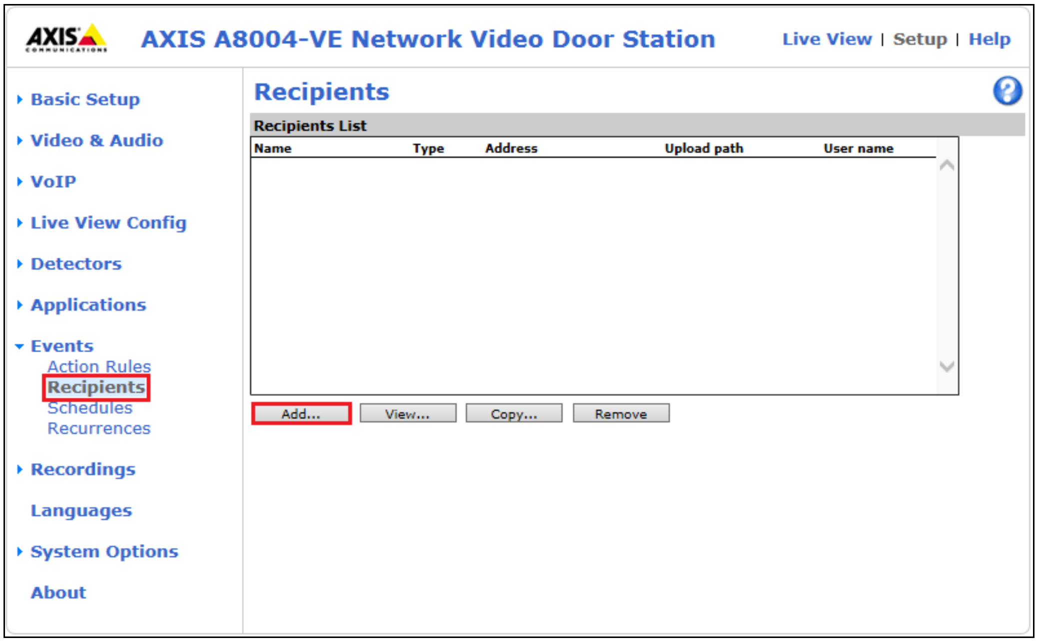


7.4. Configure Events

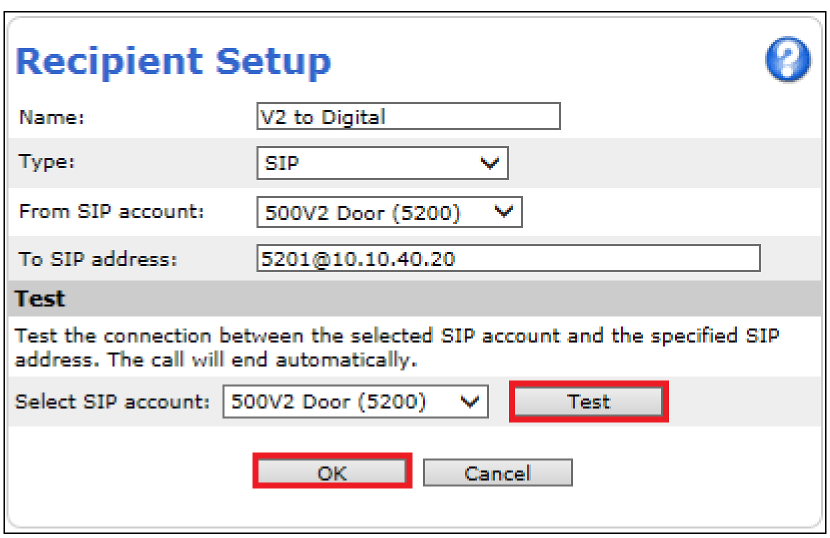
In order to create an event, both a recipient and an action rule must be created. A recipient is created before an action rule.

7.4.1. Add a new recipient

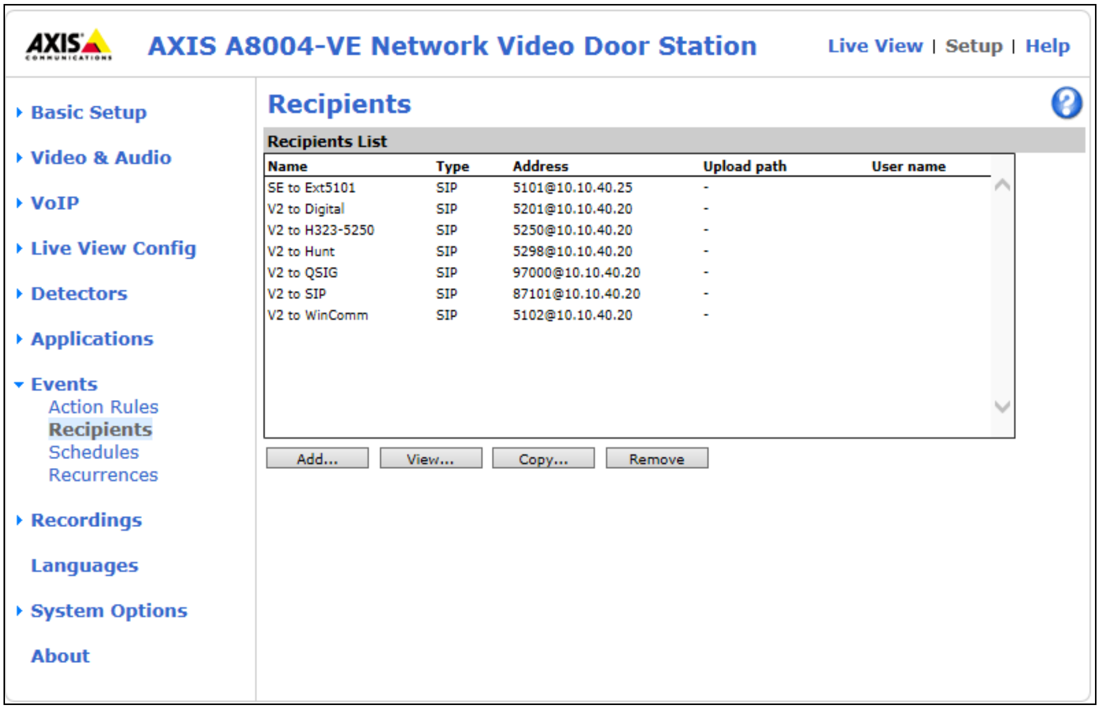
Click on **Events → Recipients** in the left window and in the main window, click on **Add**.



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.

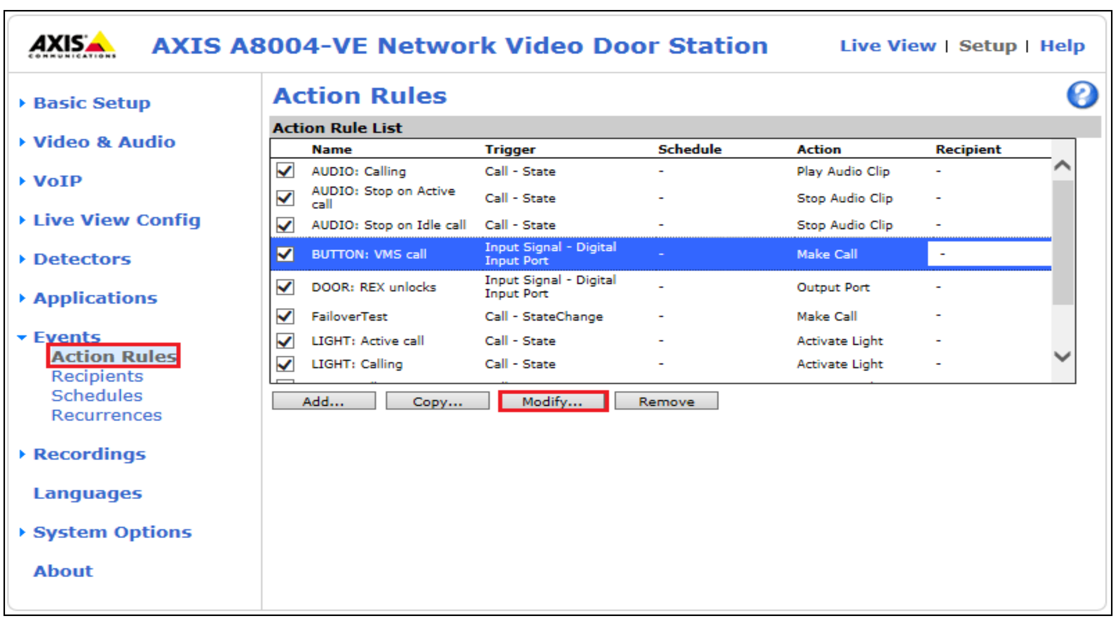


A number of different recipients are normal for such a test, where various Communication Manager endpoints can be called, or perhaps a number of hunt groups.

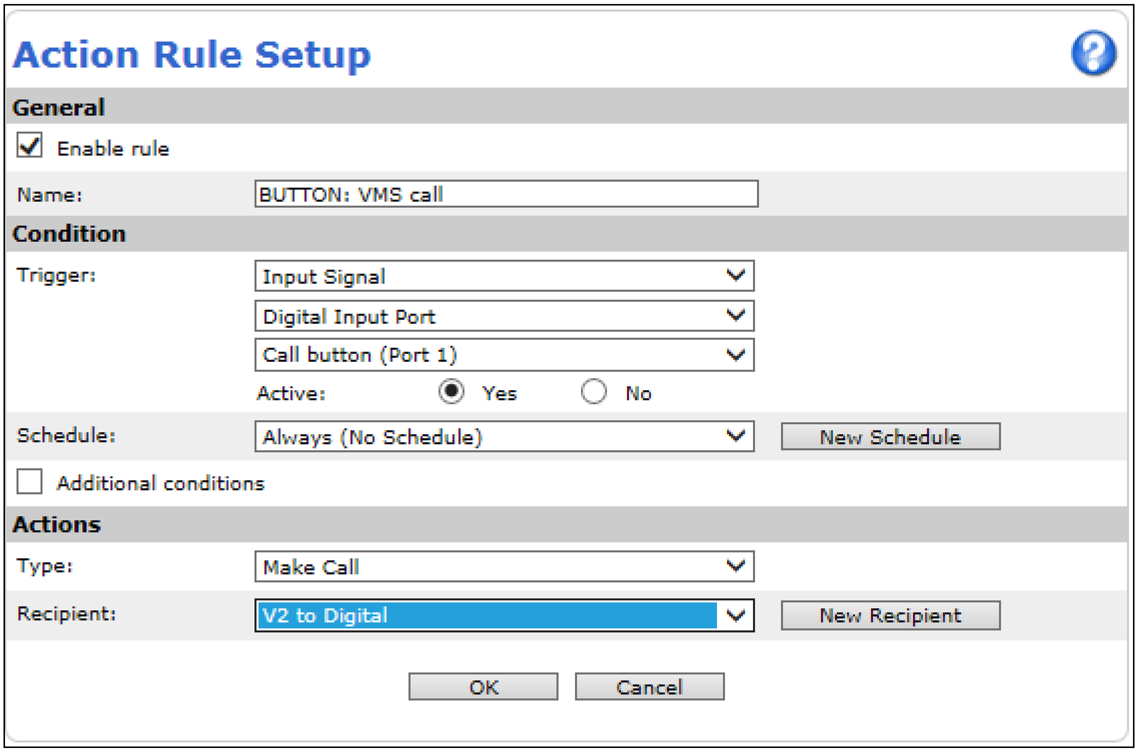


7.4.2. Modify Action Rule

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 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 **Action**s 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.

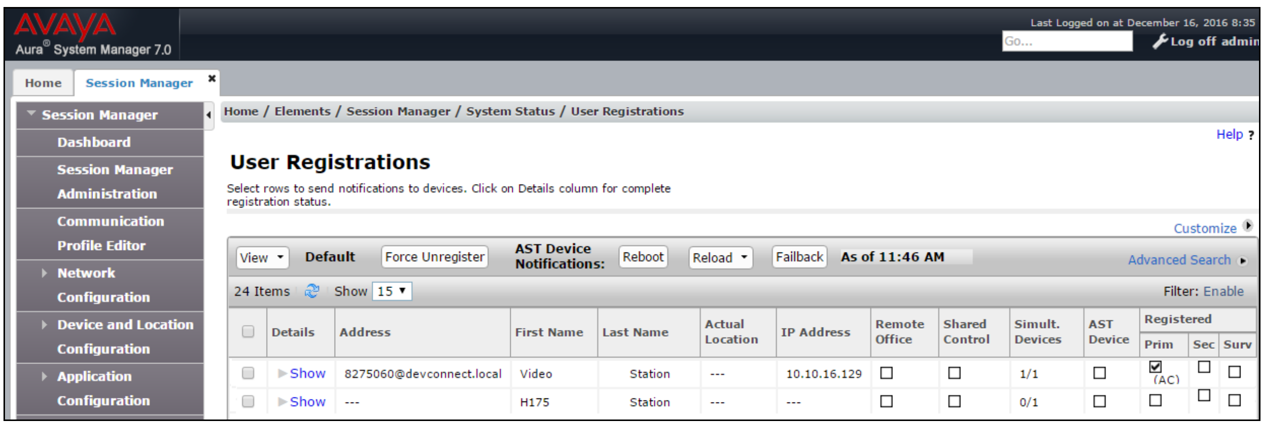


8. Verification Steps

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.

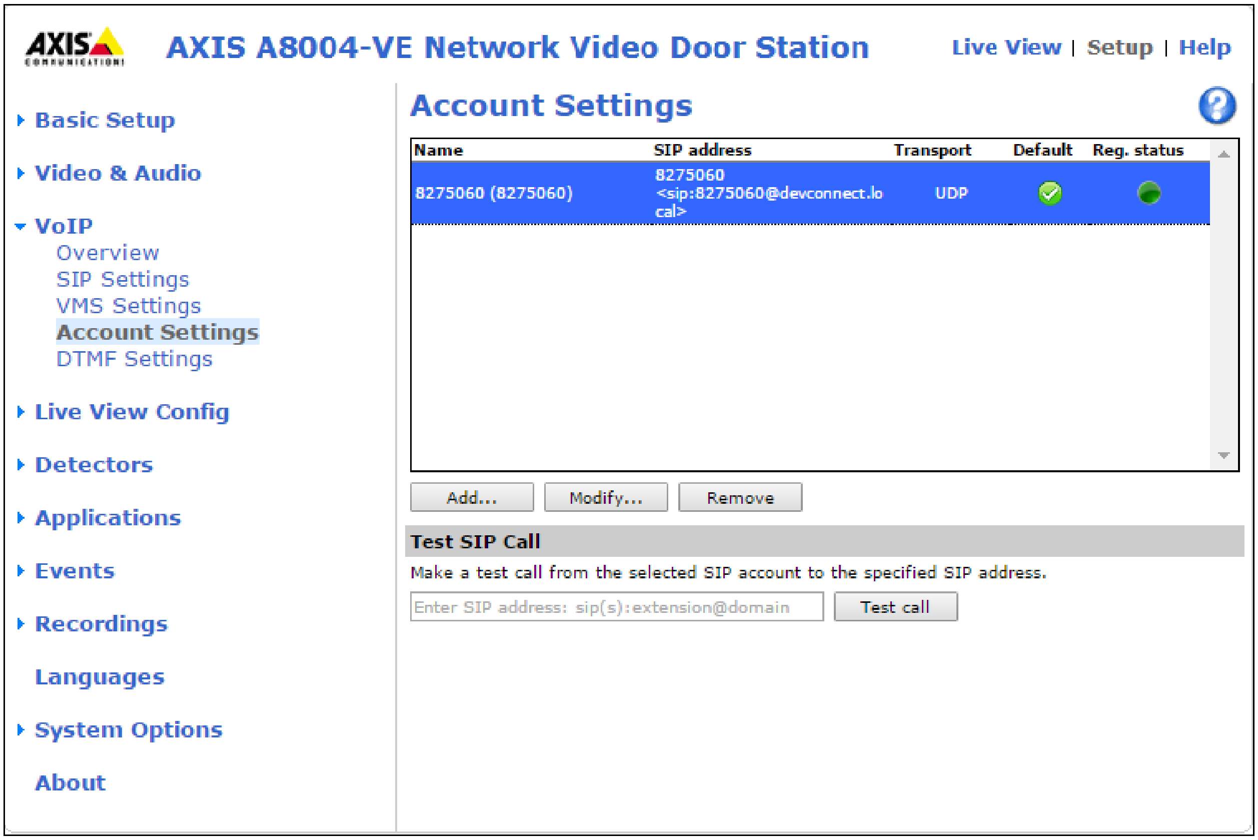
8.1. Verify Registration to Avaya Aura® Session Manager

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.

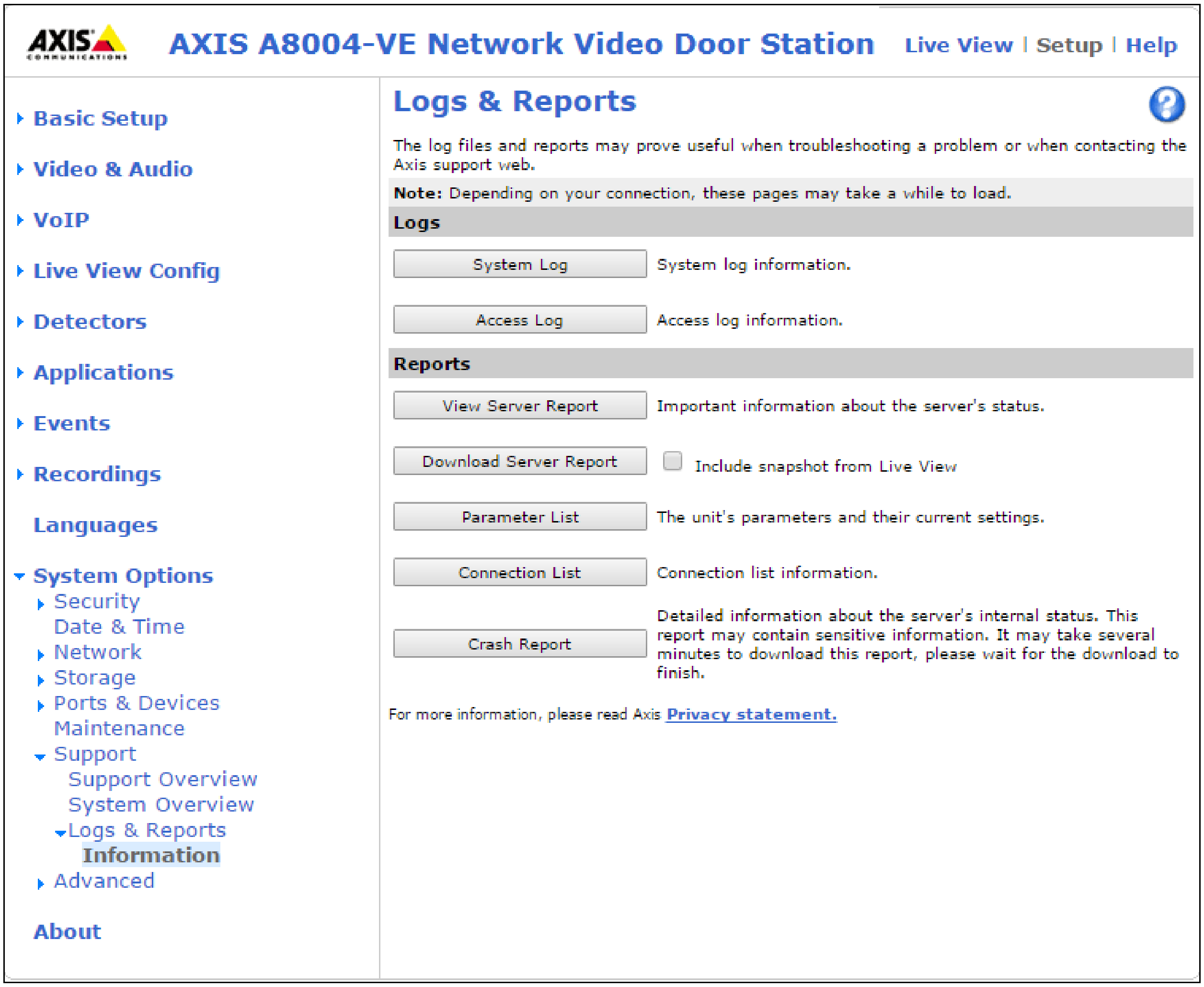


8.2. Verify Registration from AXIS A8004-VE Network Video Door Station

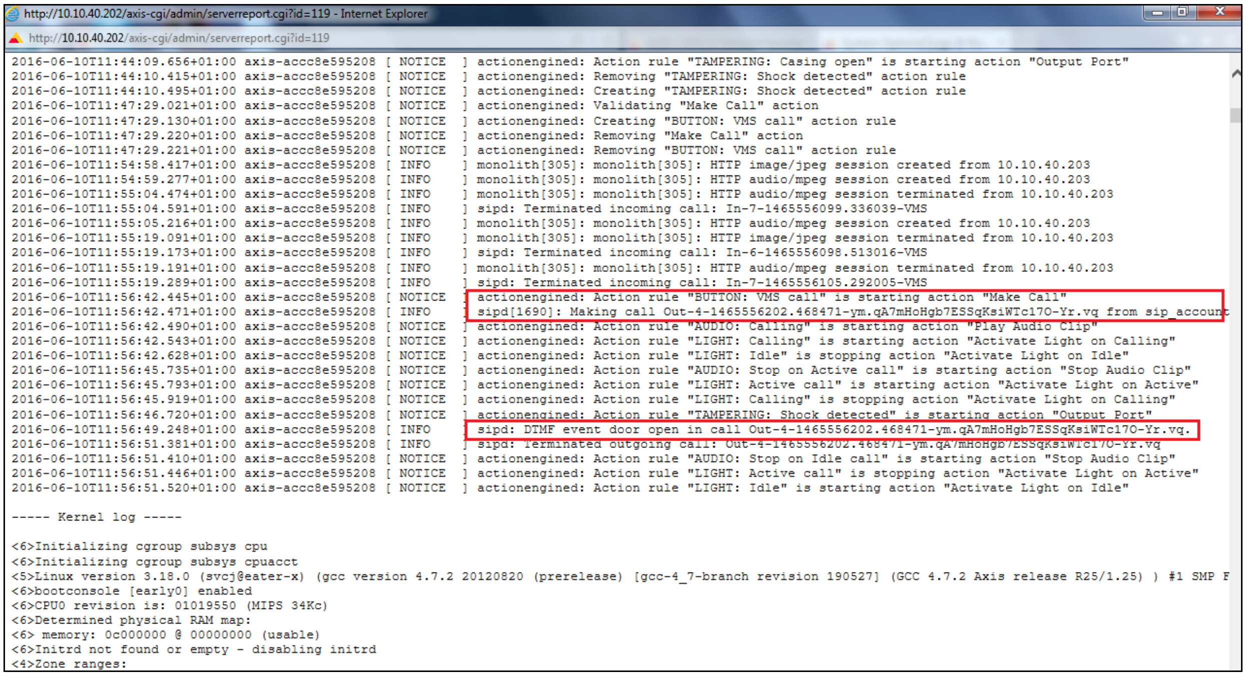
Log in to the door phone as per **Section 7**. Navigate to **VoIP → Account Settings** in the left window and the registration information is displayed in the main window as shown below. The green lights show a successful registration of 8275060. A test call can be made from each account to a specific phone number using the **Test SIP Call** at the bottom of the screen.



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.



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-accc8e595208 [ NOTICE  ] actionengined: Action rule "TAMPERING: Casing open" is starting action "Output Port"
2016-06-10T11:44:10.415+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Removing "TAMPERING: Shock detected" action rule
2016-06-10T11:44:10.495+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Creating "TAMPERING: Shock detected" action rule
2016-06-10T11:47:29.021+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Validating "Make Call" action
2016-06-10T11:47:29.130+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Creating "BUTTON: VMS call" action rule
2016-06-10T11:47:29.220+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Removing "Make Call" action
2016-06-10T11:47:29.221+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Removing "BUTTON: VMS call" action rule
2016-06-10T11:54:58.417+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP image/jpeg session created from 10.10.40.203
2016-06-10T11:54:59.277+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP audio/mpeg session created from 10.10.40.203
2016-06-10T11:55:04.474+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP audio/mpeg session terminated from 10.10.40.203
2016-06-10T11:55:04.591+01:00 axis-accc8e595208 [ INFO    ] sipd: Terminated incoming call: In-7-1465556099.336039-VMS
2016-06-10T11:55:05.216+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP audio/mpeg session created from 10.10.40.203
2016-06-10T11:55:19.091+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP image/jpeg session terminated from 10.10.40.203
2016-06-10T11:55:19.173+01:00 axis-accc8e595208 [ INFO    ] sipd: Terminated incoming call: In-6-1465556098.513016-VMS
2016-06-10T11:55:19.191+01:00 axis-accc8e595208 [ INFO    ] monolith[305]: monolith[305]: HTTP audio/mpeg session terminated from 10.10.40.203
2016-06-10T11:55:19.289+01:00 axis-accc8e595208 [ INFO    ] sipd: Terminated incoming call: In-7-1465556105.292005-VMS
2016-06-10T11:56:42.445+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "BUTTON: VMS call" is starting action "Make Call"
2016-06-10T11:56:42.471+01:00 axis-accc8e595208 [ INFO    ] sipd[1690]: Making call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTc17O-Yr.vq from sip_account
2016-06-10T11:56:42.490+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "AUDIO: Calling" is starting action "Play Audio Clip"
2016-06-10T11:56:42.543+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Calling" is starting action "Activate Light on Calling"
2016-06-10T11:56:42.628+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Idle" is stopping action "Activate Light on Idle"
2016-06-10T11:56:45.735+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "AUDIO: Stop on Active call" is starting action "Stop Audio Clip"
2016-06-10T11:56:45.793+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Active call" is starting action "Activate Light on Active"
2016-06-10T11:56:45.919+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Calling" is stopping action "Activate Light on Calling"
2016-06-10T11:56:46.720+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "TAMPERING: Shock detected" is starting action "Output Port"
2016-06-10T11:56:49.248+01:00 axis-accc8e595208 [ INFO    ] sipd: DTMF event door open in call Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTc17O-Yr.vq.
2016-06-10T11:56:51.381+01:00 axis-accc8e595208 [ INFO    ] sipd: Terminated outgoing call: Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTc17O-Yr.vq
2016-06-10T11:56:51.410+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "AUDIO: Stop on Idle call" is starting action "Stop Audio Clip"
2016-06-10T11:56:51.446+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Active call" is stopping action "Activate Light on Active"
2016-06-10T11:56:51.520+01:00 axis-accc8e595208 [ NOTICE  ] actionengined: Action rule "LIGHT: Idle" is starting action "Activate Light on Idle"

----- Kernel log -----

<6>Initializing cgroup subsys cpu
<6>Initializing cgroup subsys cpuacct
<5>Linux version 3.18.0 (svcj@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.qA7mHoHgb7ESSqKsiWTc17O-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.qA7mHoHgb7ESSqKsiWTc17O-Yr.vq.
[ INFO    ] sipd: Terminated outgoing call: Out-4-1465556202.468471-ym.qA7mHoHgb7ESSqKsiWTc17O-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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.