Avaya Solution & Interoperability Test Lab

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

Abstract

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

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

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

1. Introduction

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

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

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

2. General Test Approach and Test Results

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

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

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

2.1. Interoperability Compliance Testing

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

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

2.2. Test Results

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

2.3. Support

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

Axis Communications AB
Tel: +46 46 272 18 00
Fax: +46 46 13 61 30
http://www.axis.com/global/en/learning-and-support

3. Reference Configuration

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

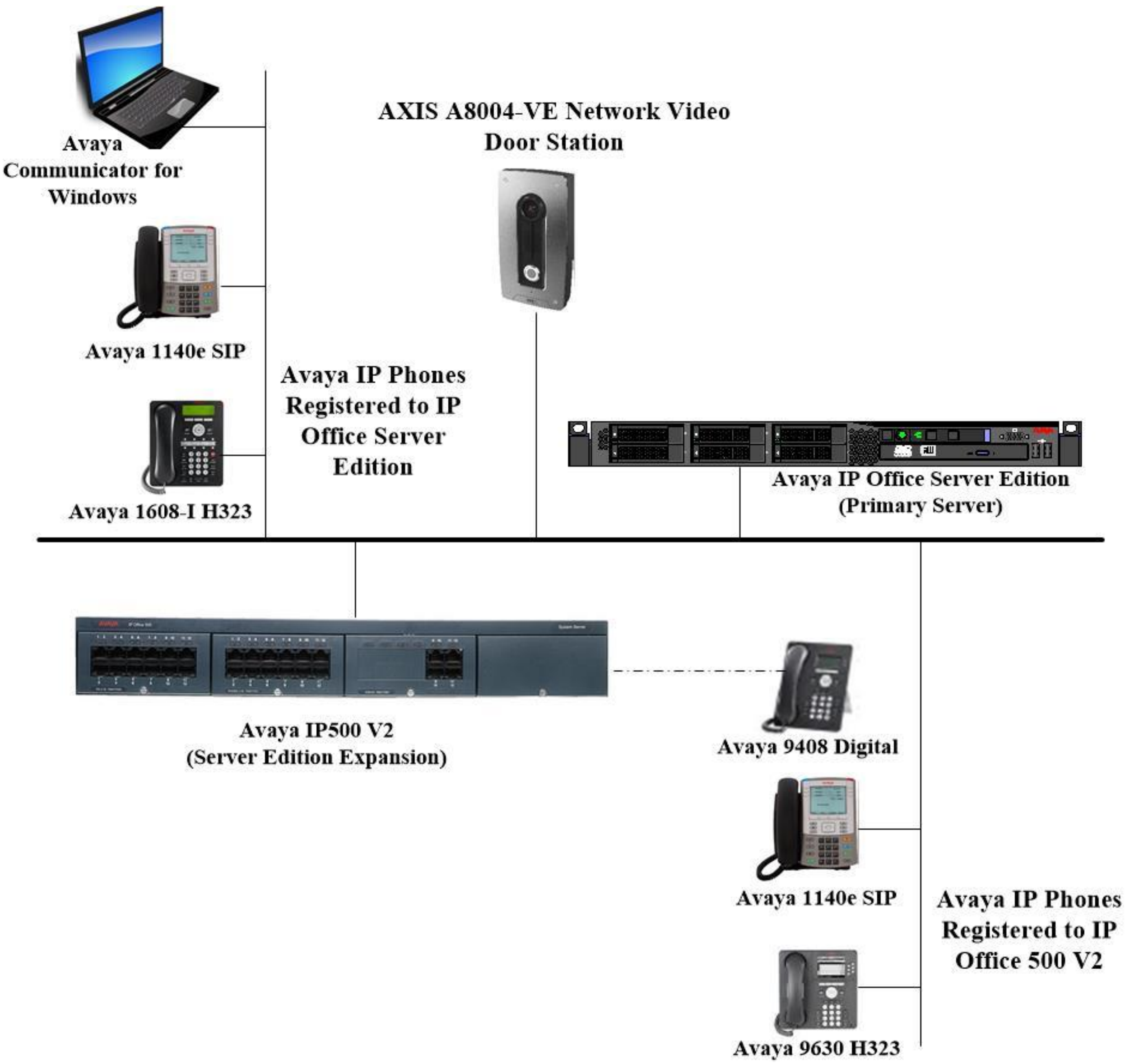


Figure 1: Connection of Axis Communications AB A8004-VE Network Video Door Station with Avaya IP Office Server Edition and IP Office IP500 V2 R9.1

4. Equipment and Software Validated

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

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

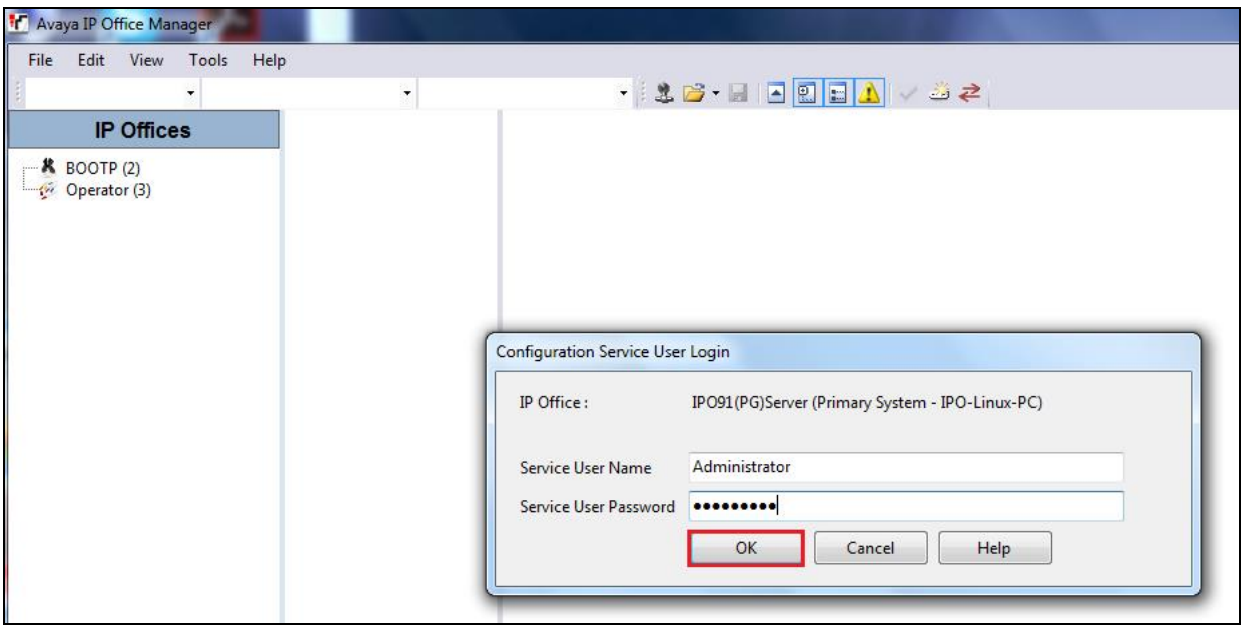
5. Configure Avaya IP Office

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

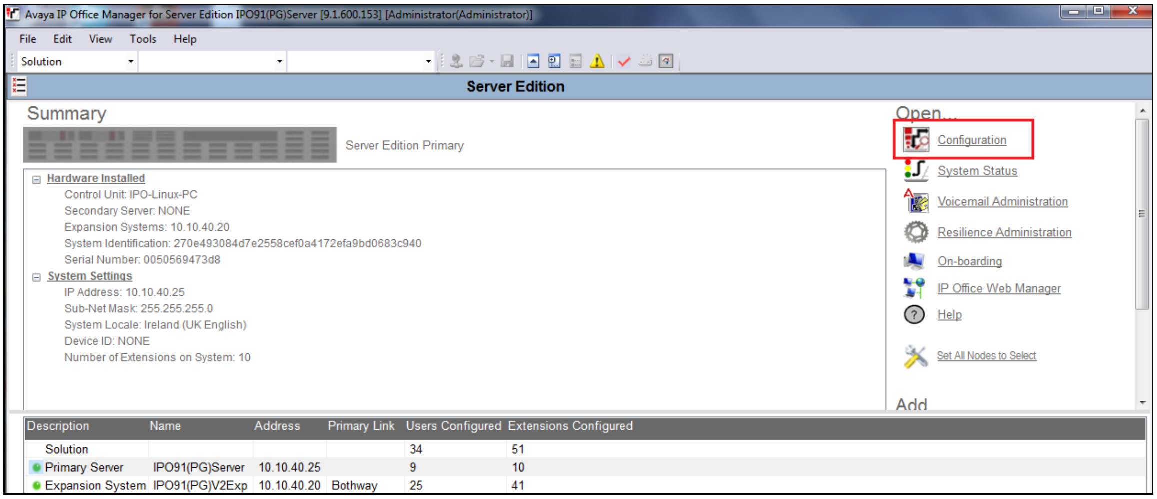
- Launch Avaya IP Office Manager.
- Display LAN Configuration.
- Configure New SIP User.
- Save Configuration.

5.1. Launch Avaya IP Office Manager

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

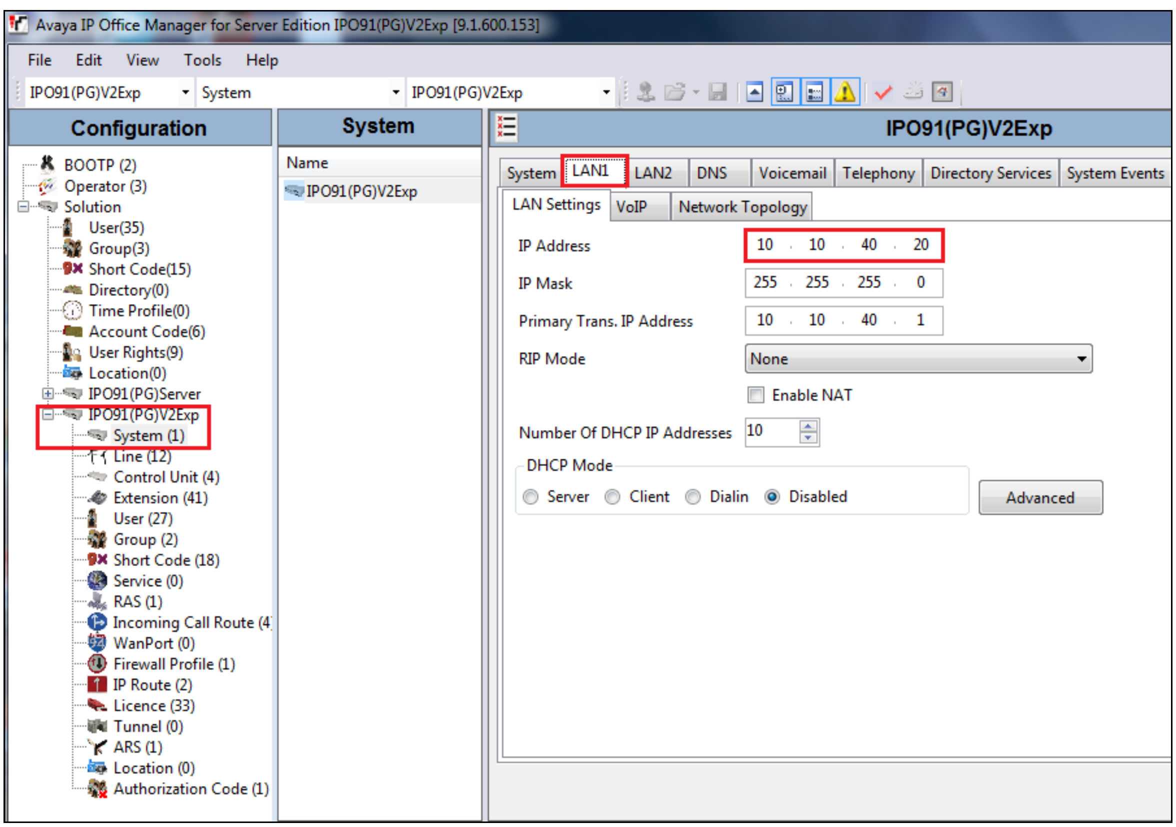


Click on **Configuration** to open the configuration GUI for both the Server Edition system and the expansion system.

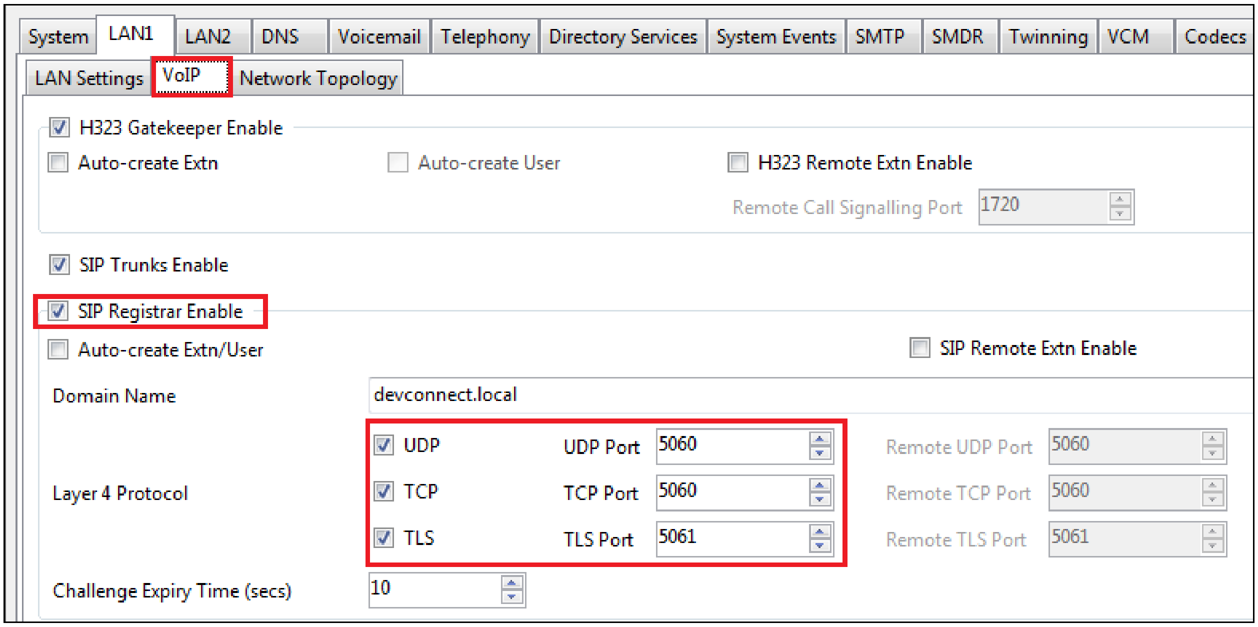


5.2. Display LAN Configuration

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

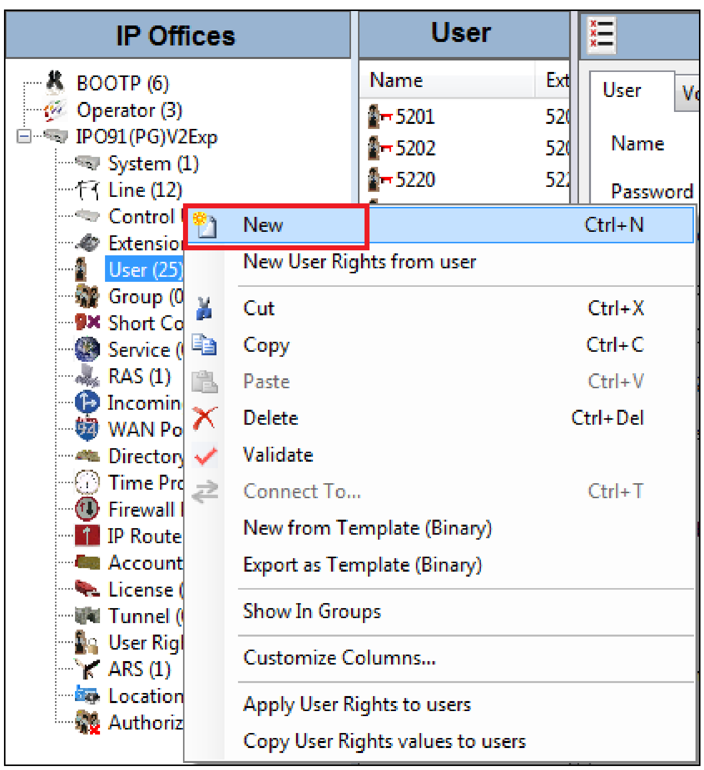


Selecting the **VoIP** tab displays the **Domain Name** and the **UDP**, **TCP** and **TLS Port** details used in the configuration of the Axis door phone in **Section 6.2**.

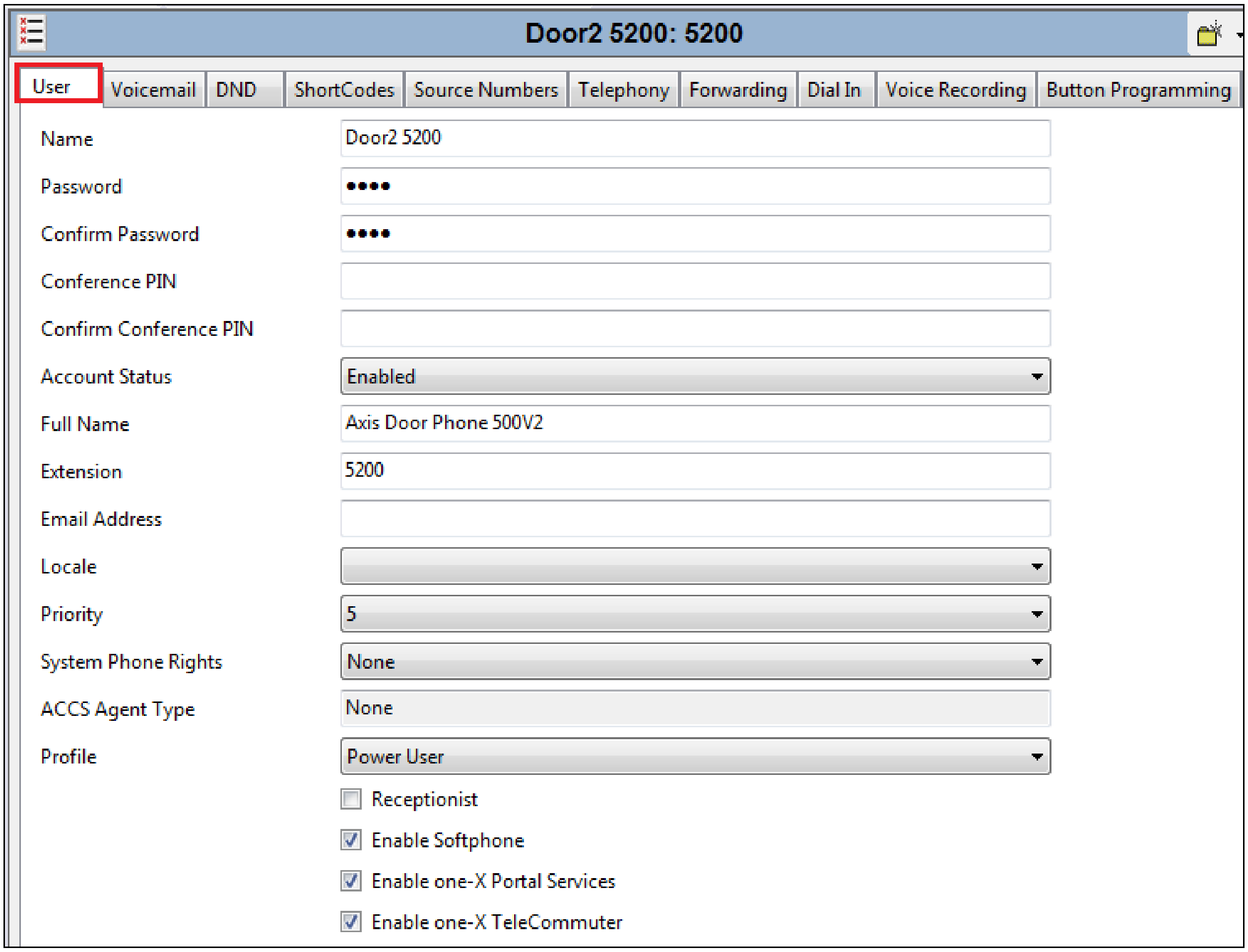


5.3. Configure New SIP User

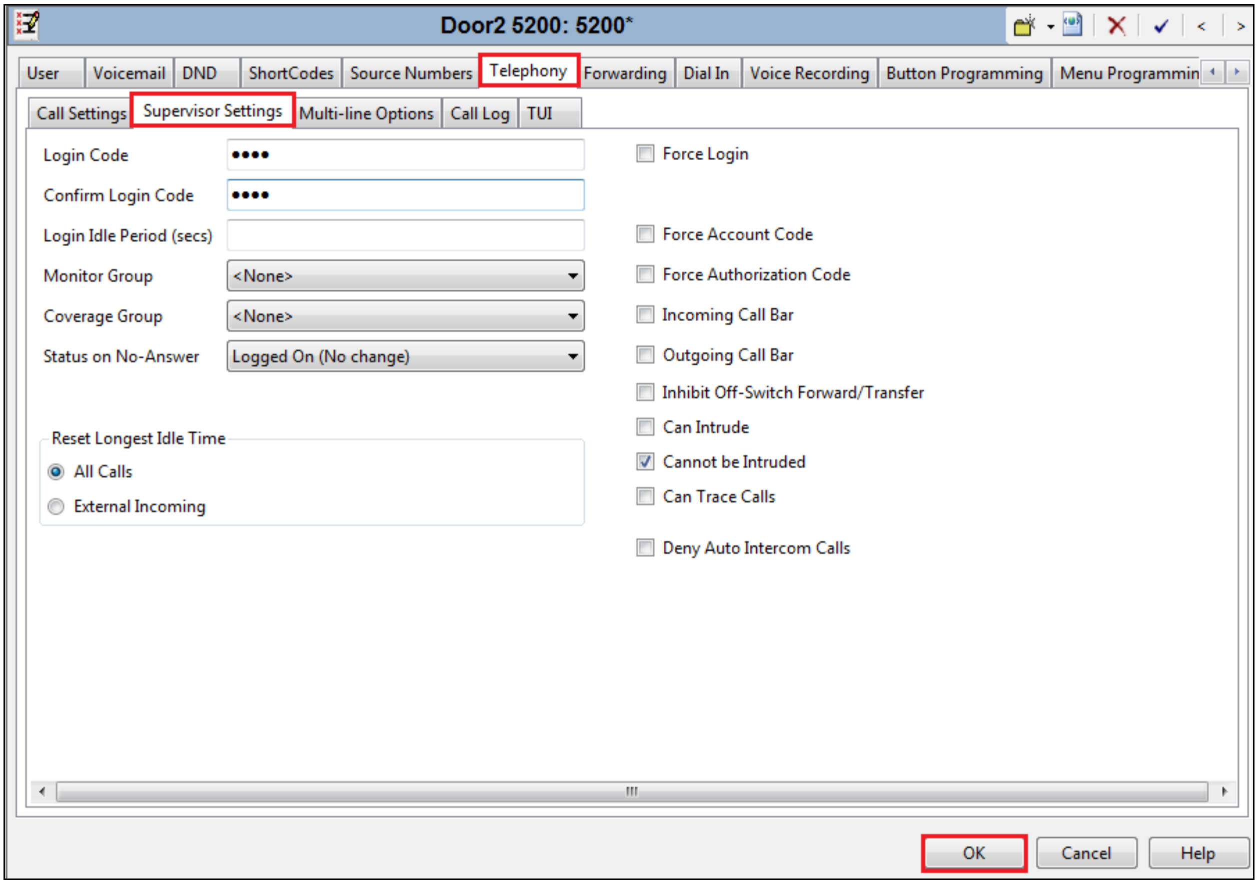
From the left window right click on **Users** and select **New** as shown below, this will allow a new user to be added to IP Office, this new user will be a SIP user.



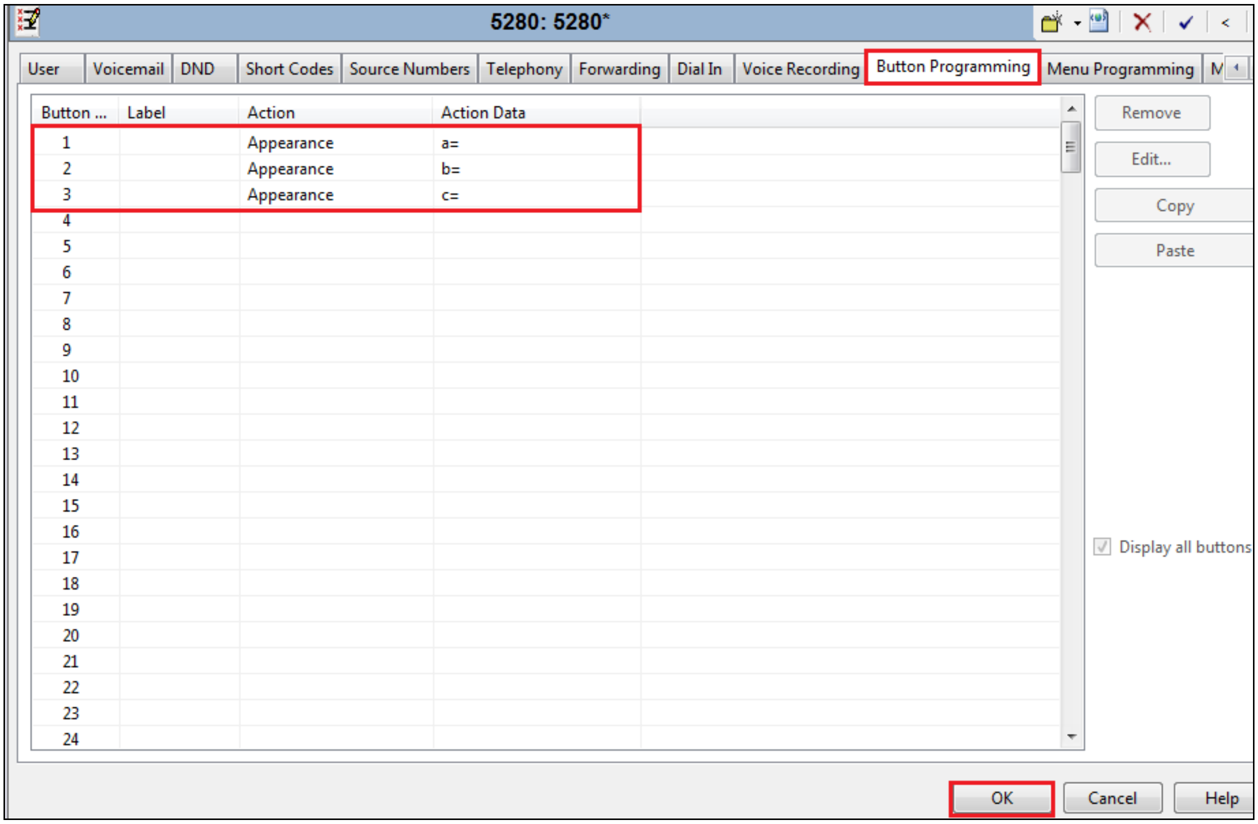
Within the **User** tab at the top of the screen, enter a suitable **Name** and **Password** for the user. Add the **Extension** number as shown below.



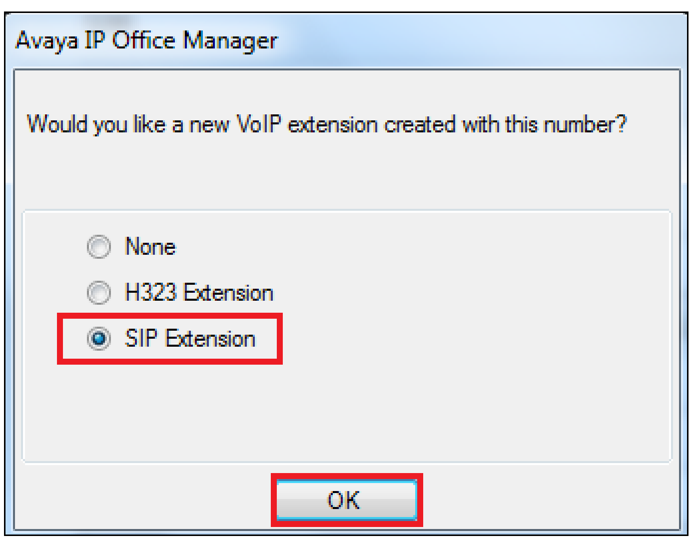
Navigate to the **Supervisor Settings** tab, enter the **Login Code** for the SIP user and note that this password will be required for the Axis door phone configuration in **Section 6.3**. Click on **OK** to save the configuration.



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**.

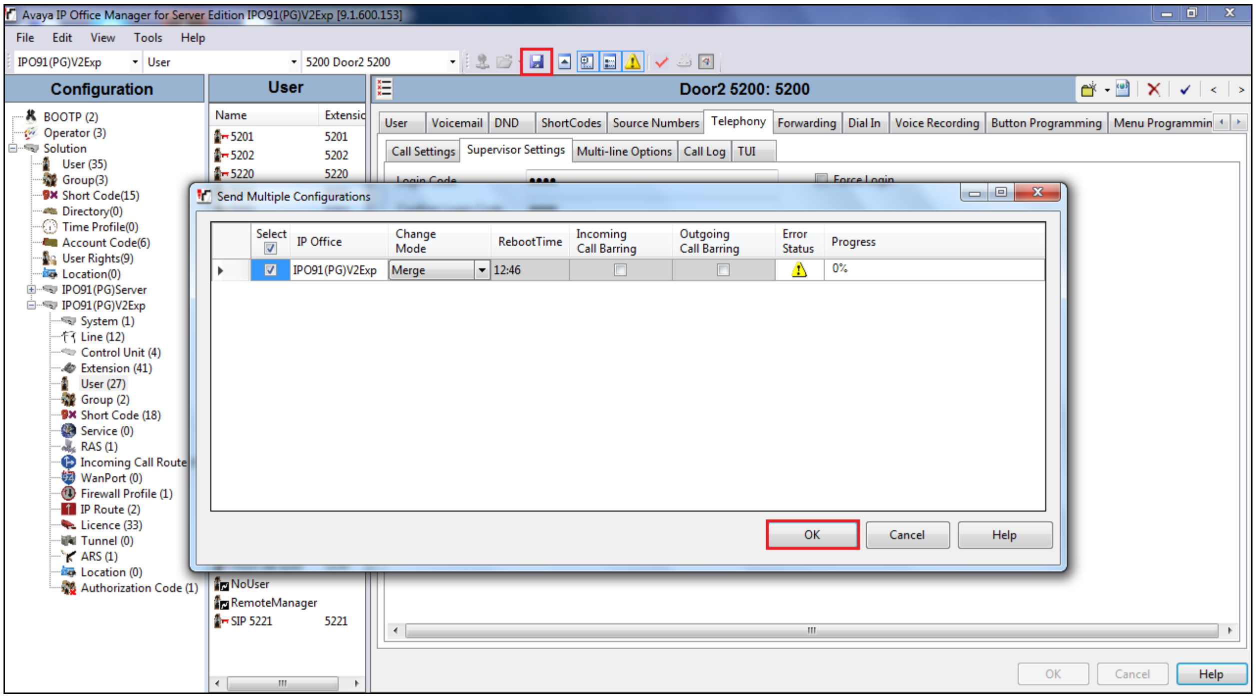


On the subsequent screen, ensure that **SIP Extension** is selected and click on **OK** to create the SIP extension along with the new user.



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.

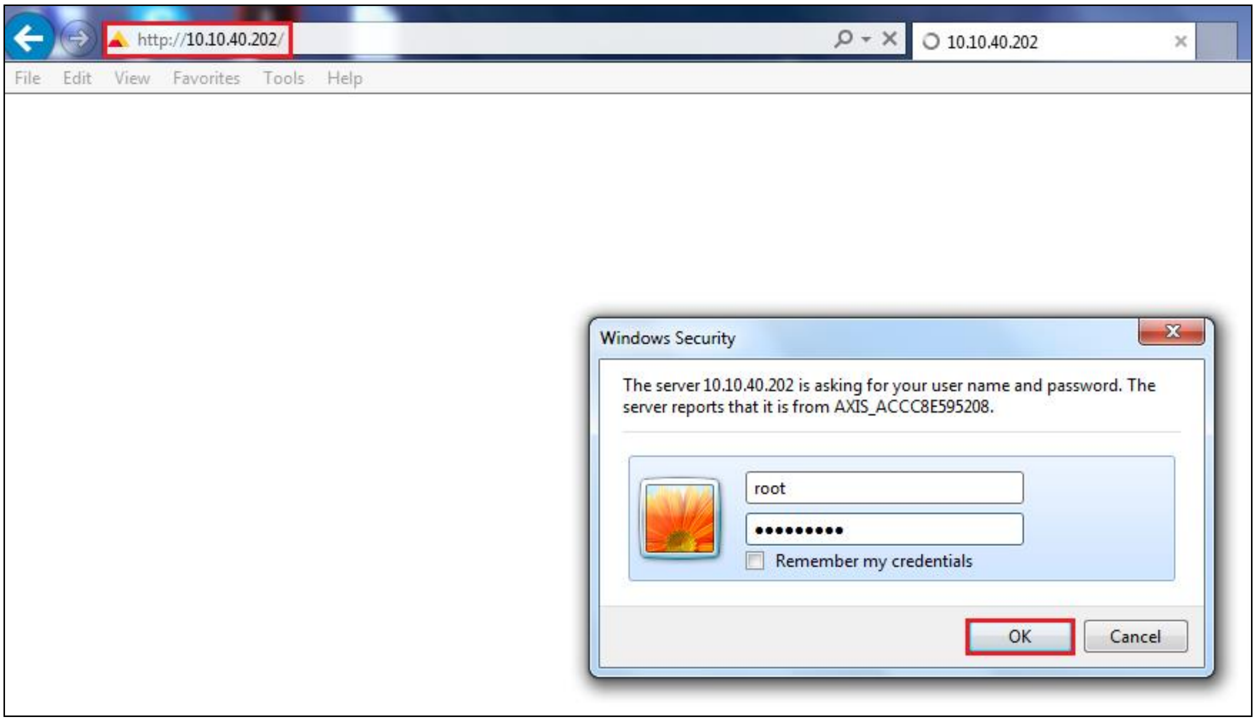


6. Configure AXIS A8004-VE Network Video Door Station

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

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

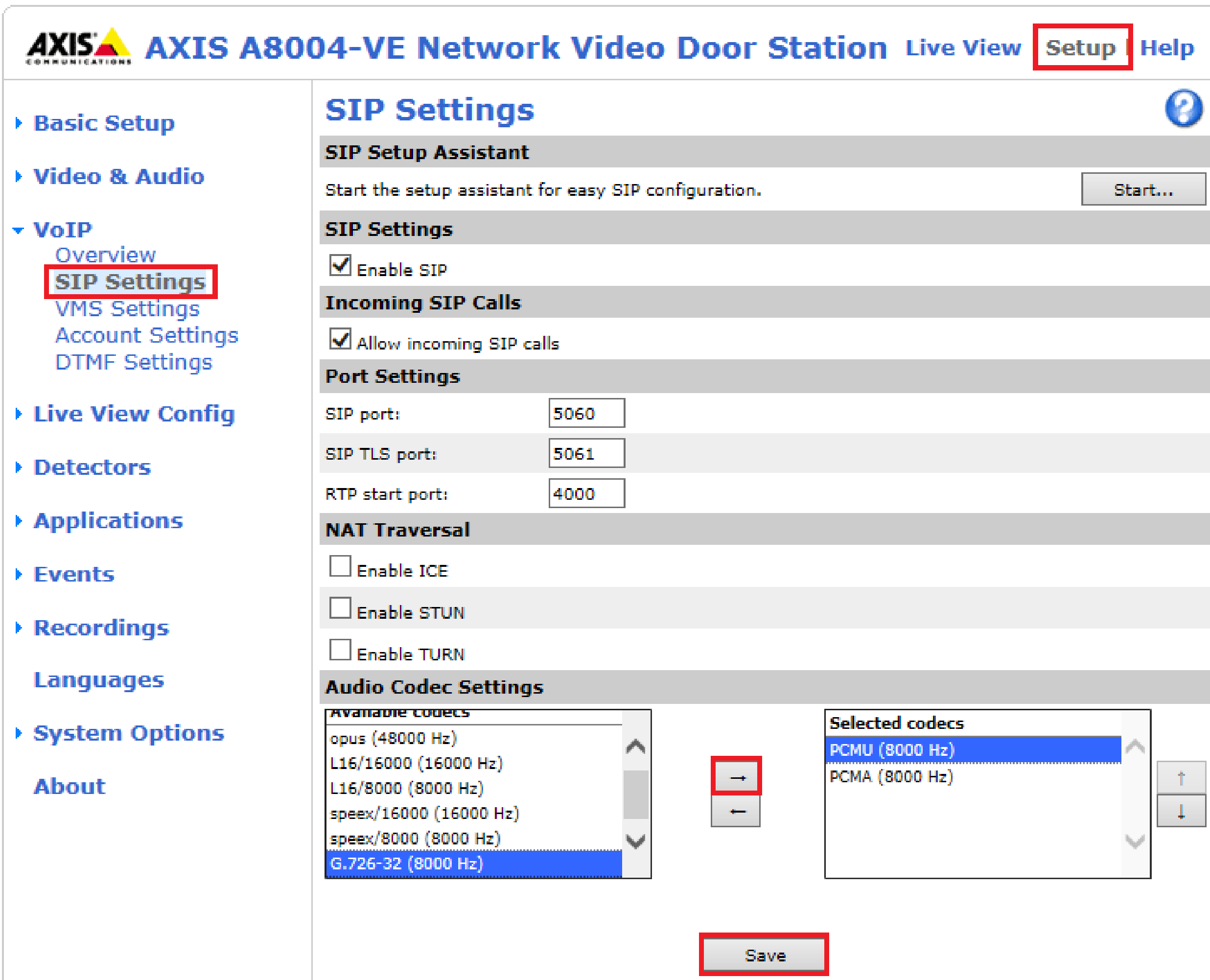
Open a web session to the IP address of the Axis door phone, enter the proper credentials and click on **OK**.



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.

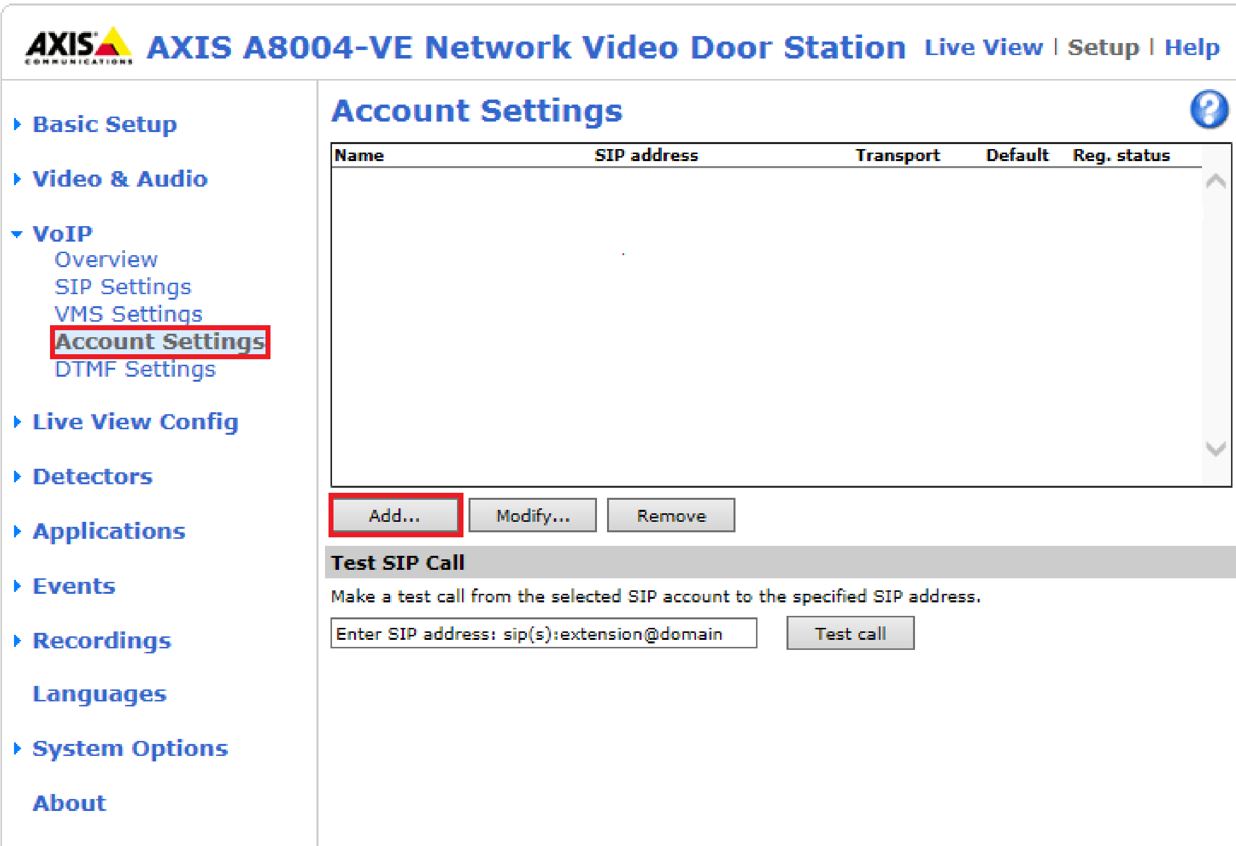
6.1. Configure SIP Settings

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.



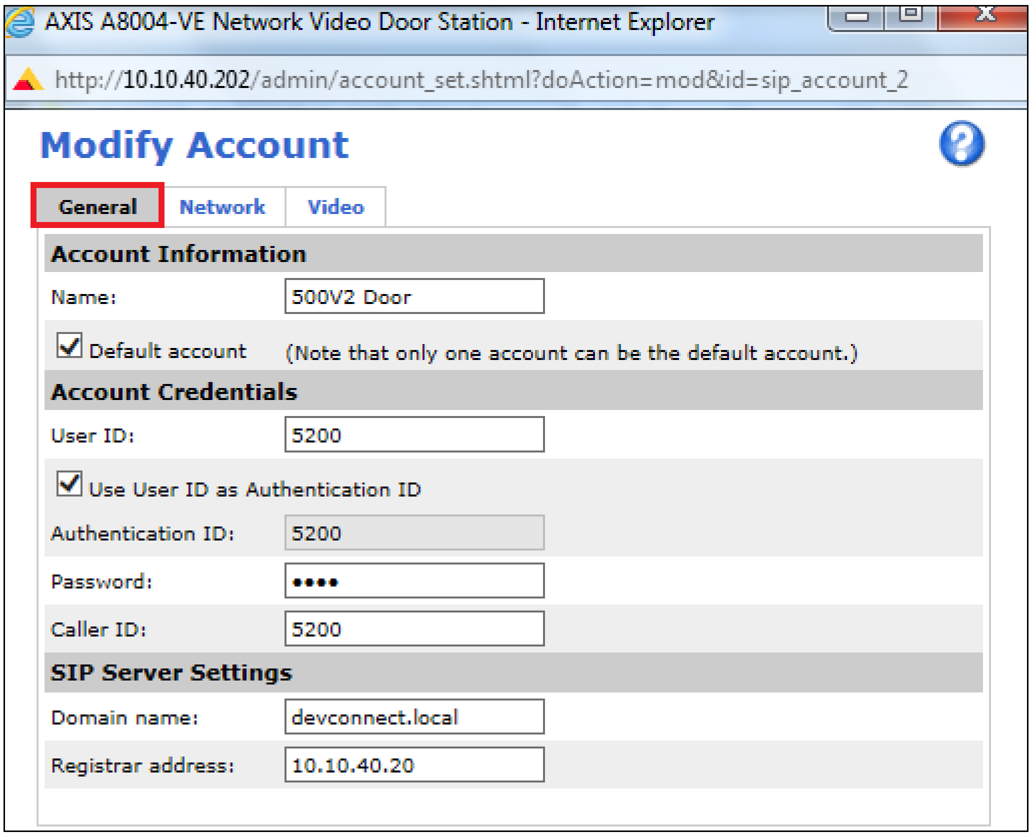
6.2. Configure Account

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

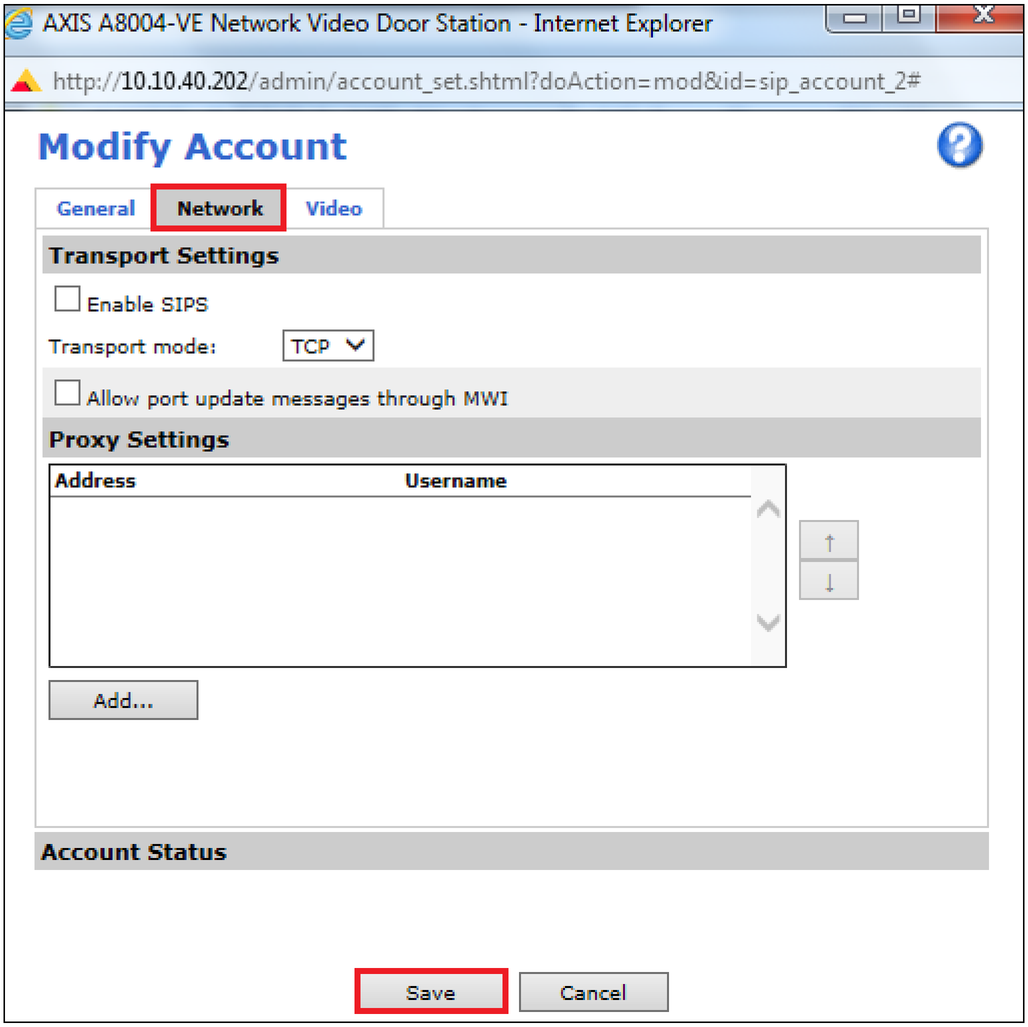


Enter the following details under the **General** tab:

- **Name:** Enter a suitable name for the SIP account.
- **User ID**: Enter the SIP user number configured 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**.

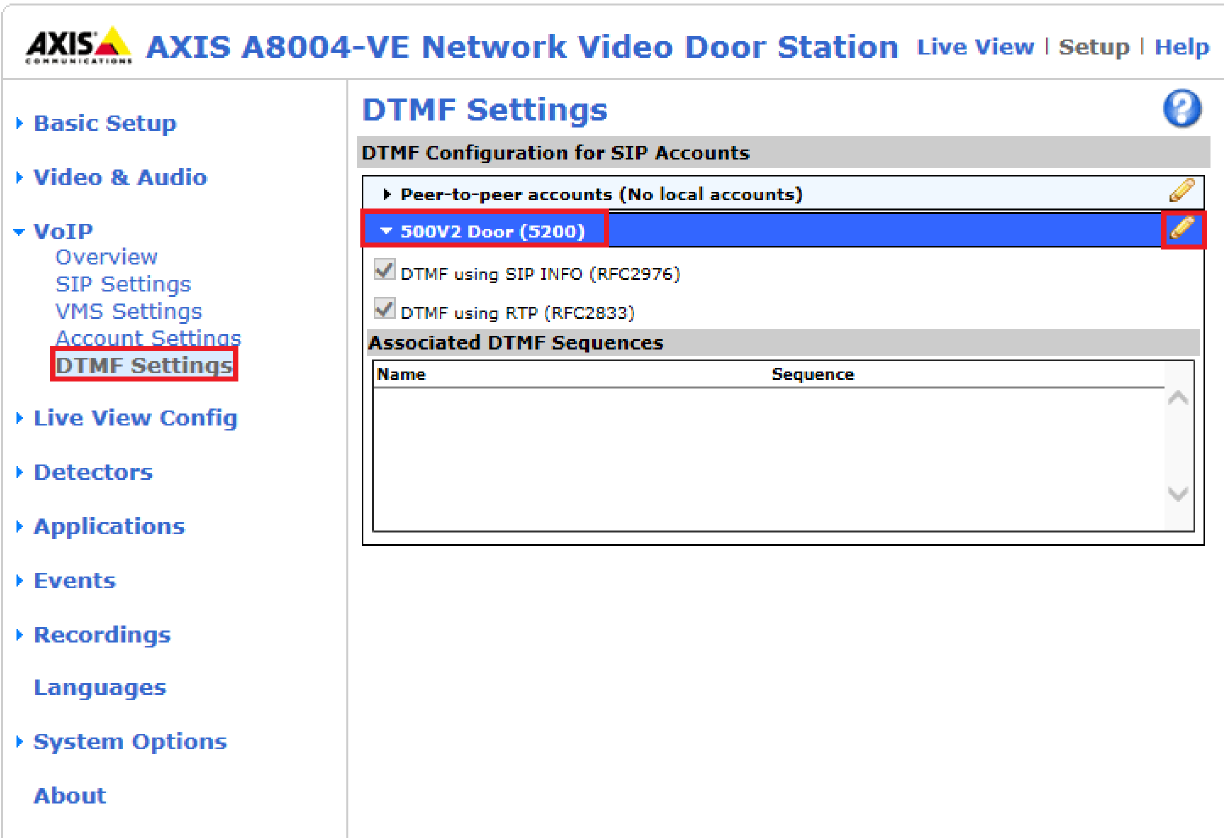


Select the **Network** tab and select the transport mode to be used, this can be UDP, **TCP** or TLS, all three protocols were tested and work correctly with IP Office. Click on **Save** to save the Account information.

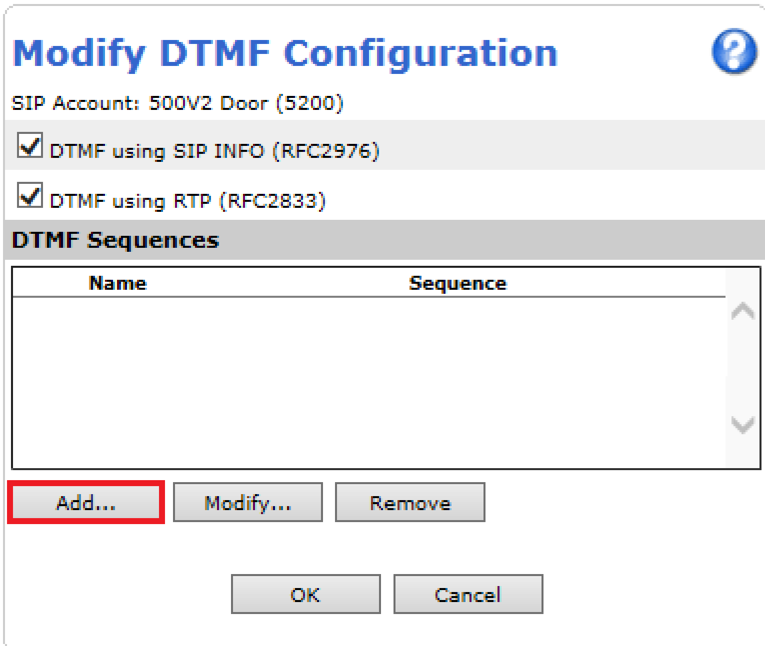


6.3. Configure DTMF Settings

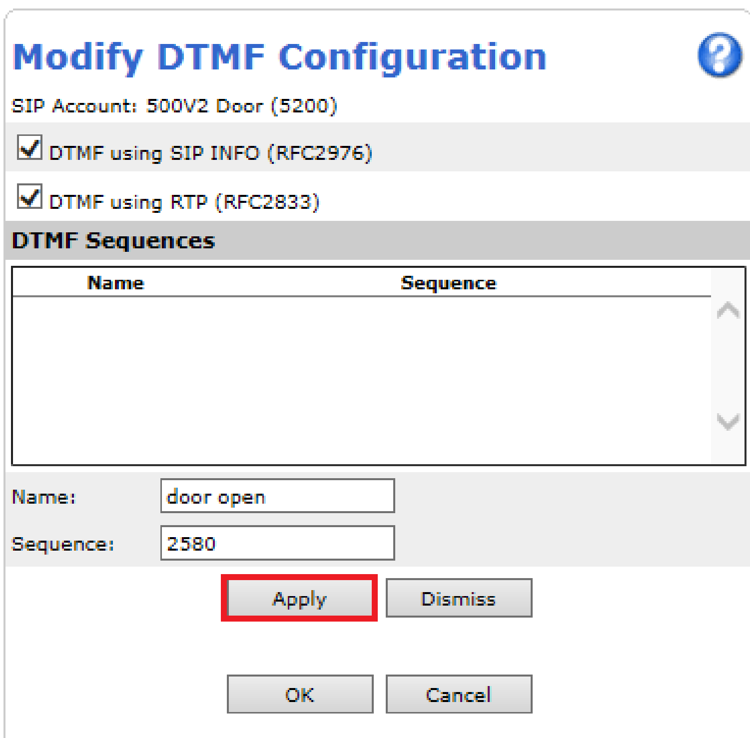
Staying within the **VoIP** menu on the left window, select **DTMF Settings**. In the main window select the SIP account that was created in **Section 6.2** and click on the edit icon, as shown below.



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.

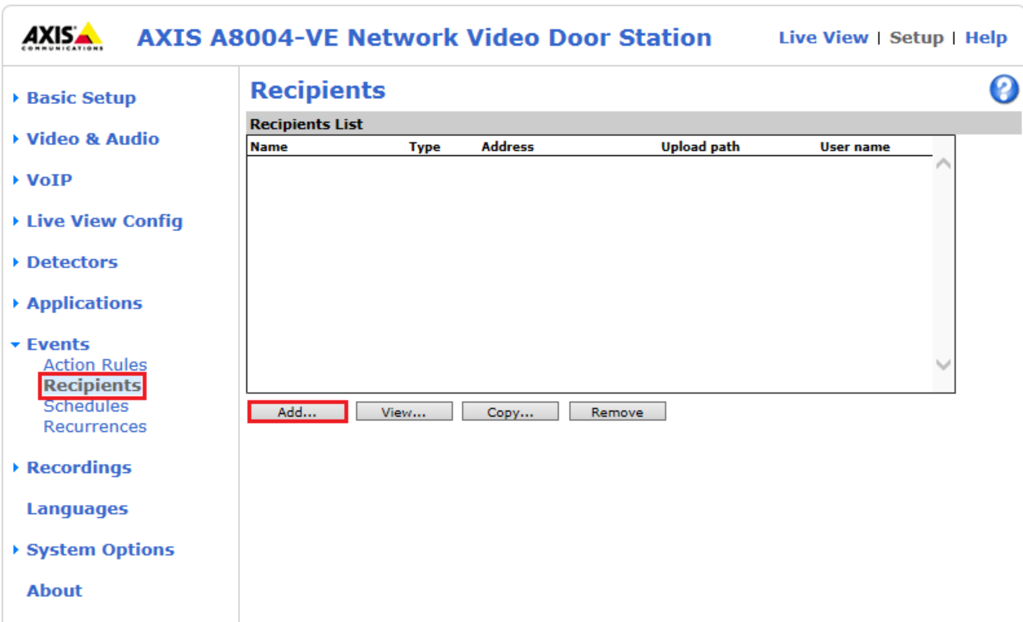


6.4. Configure Events

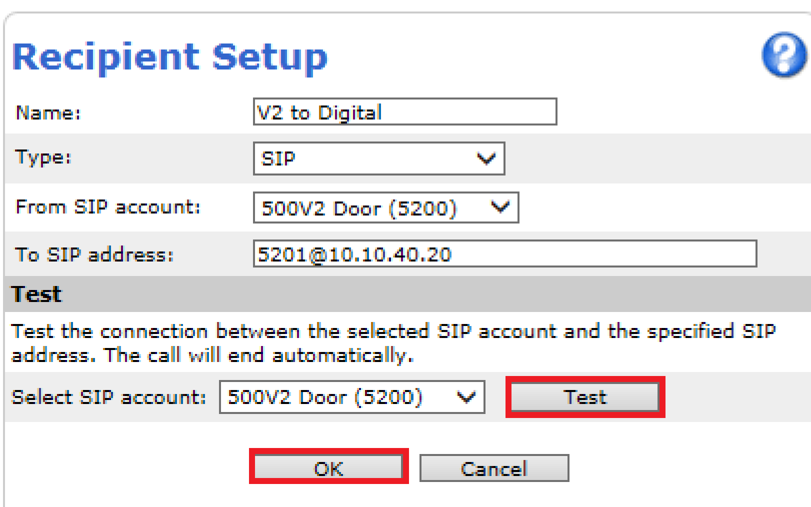
In order to create an event both a recipient and an action rule must be created. A recipient is created before an action rule.

6.4.1. Add a new recipient

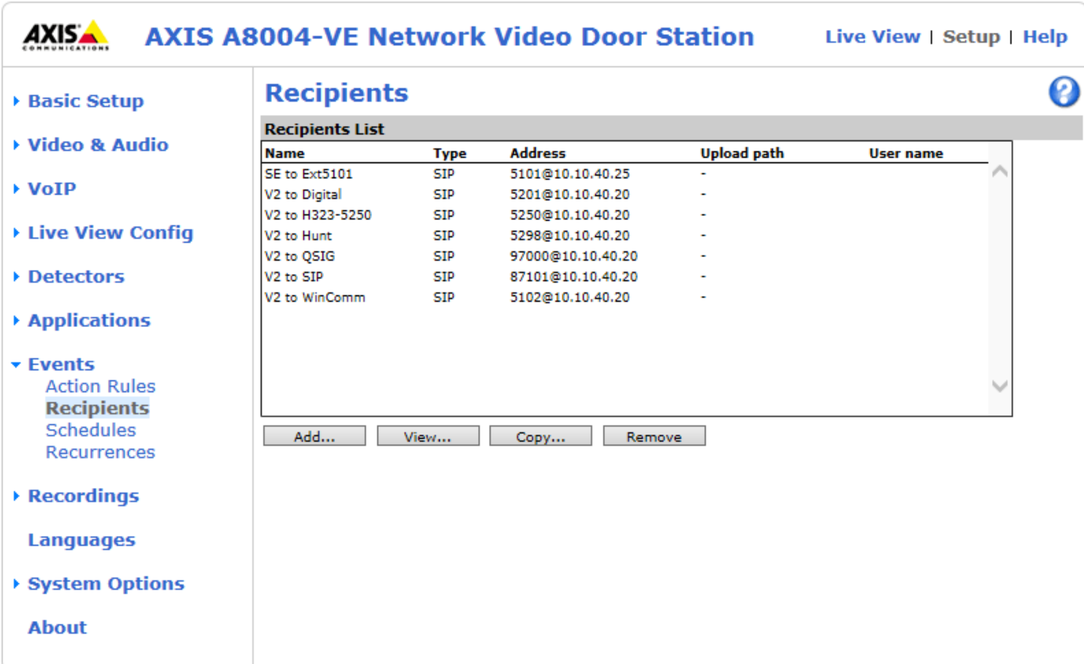
Click on **Events → Recipients** in the left window and in the main window, click on **Add**.



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.

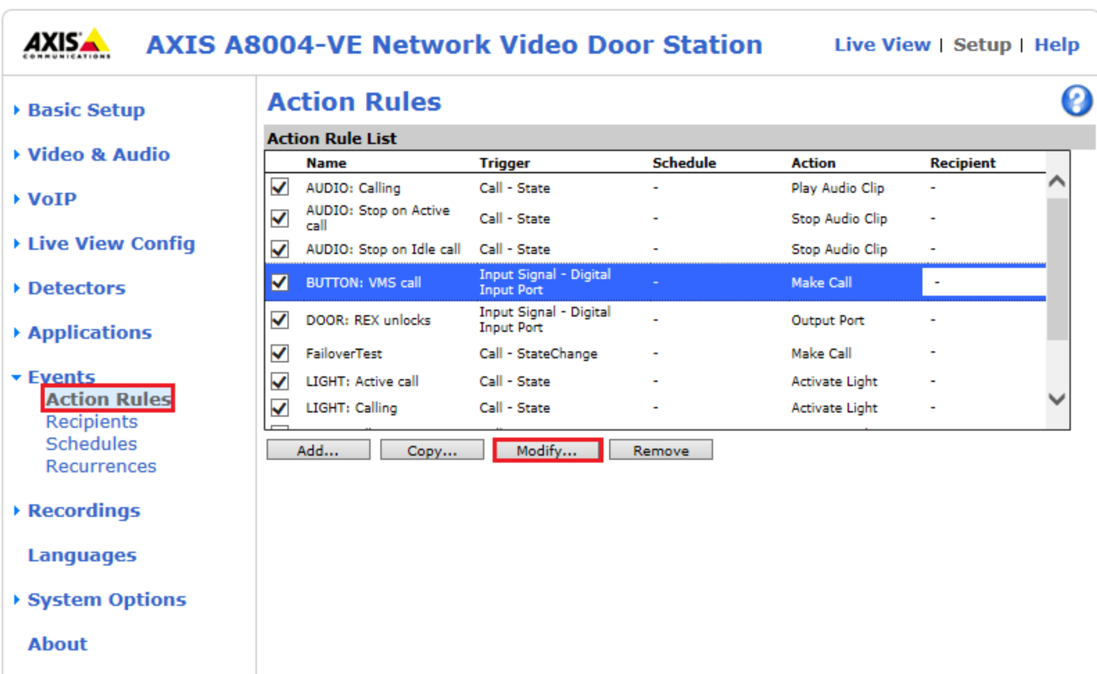


A number of different recipients are normal for such a test, where various IP Office endpoints can be called, or perhaps a number of hunt groups.

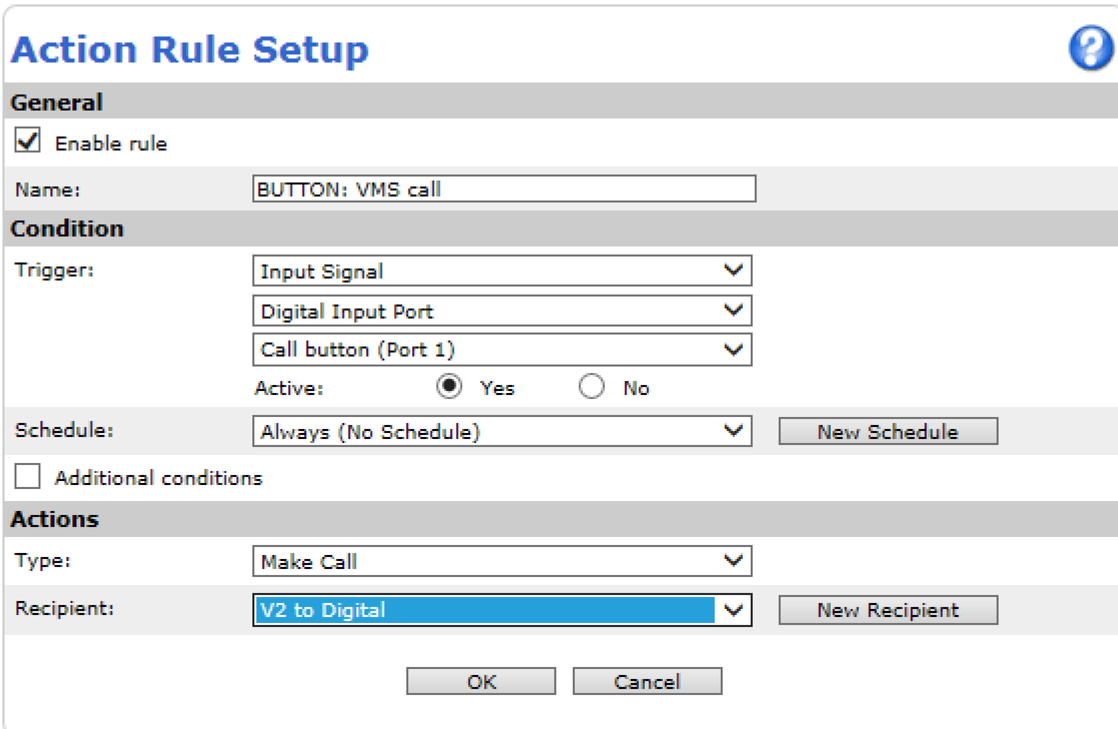


6.4.2. Modify Action Rule

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 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 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.

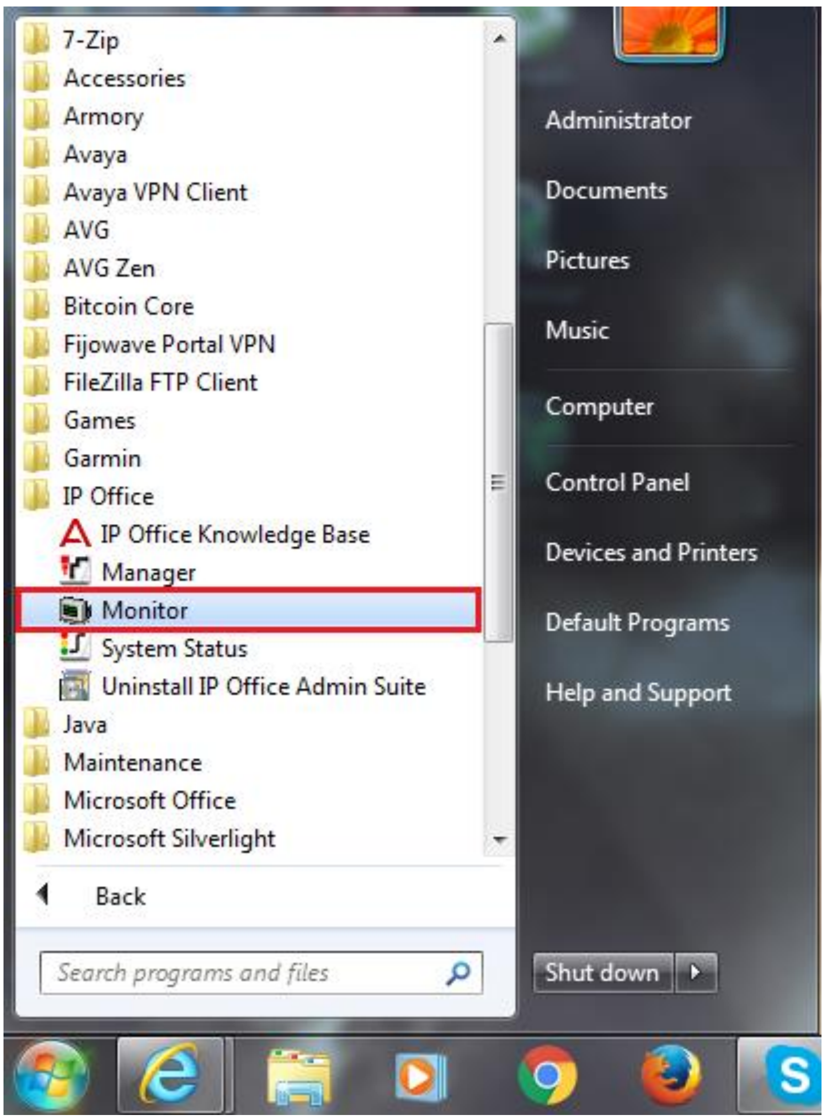


7. Verification Steps

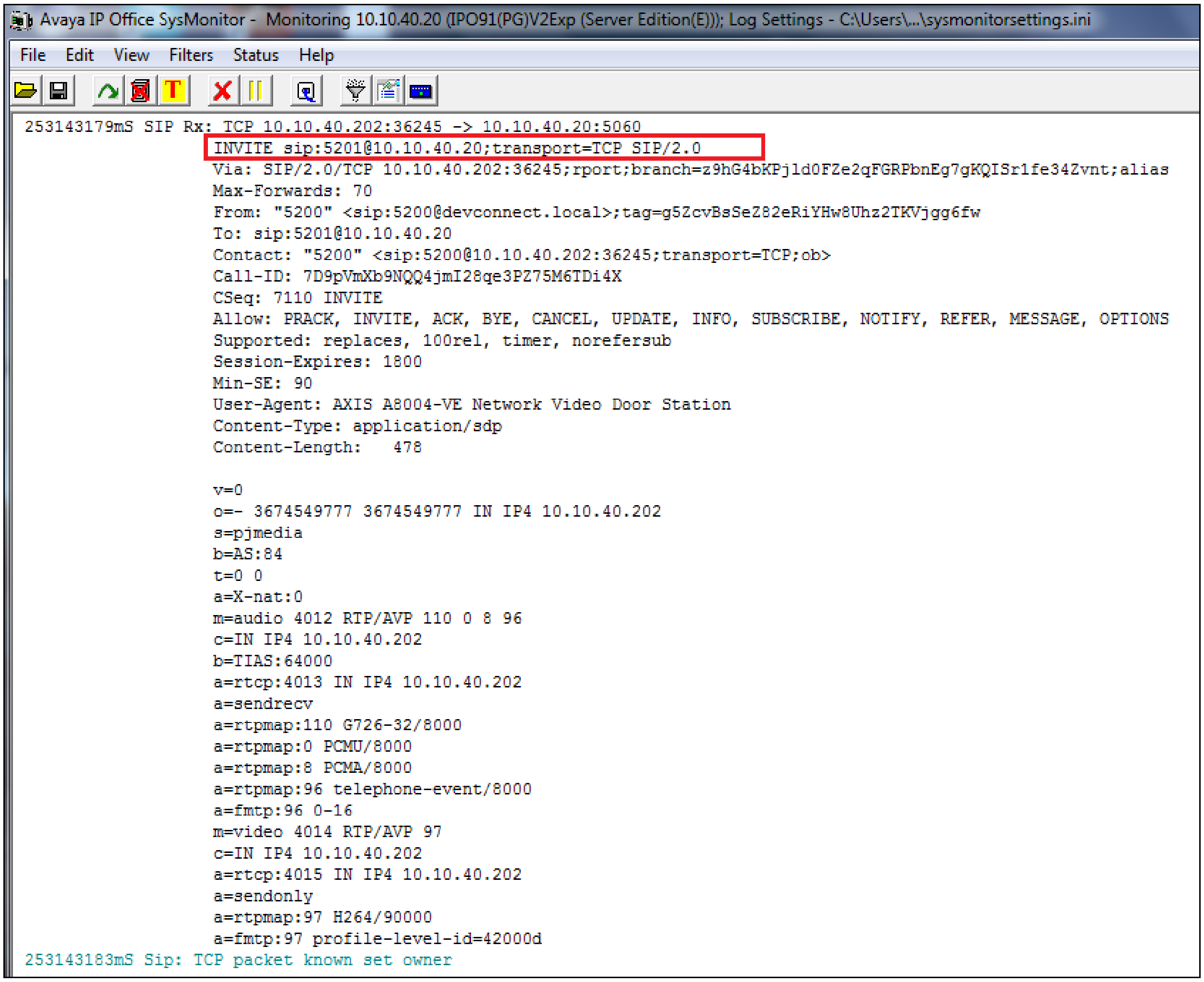
Pressing the Axis door phone button and answering the call from the IP Office set and ensuring there is two-way speech and video (where possible) is the ultimate verification that the product works and is connected and configured correctly. The steps below can also be taken to ensure that the Axis door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

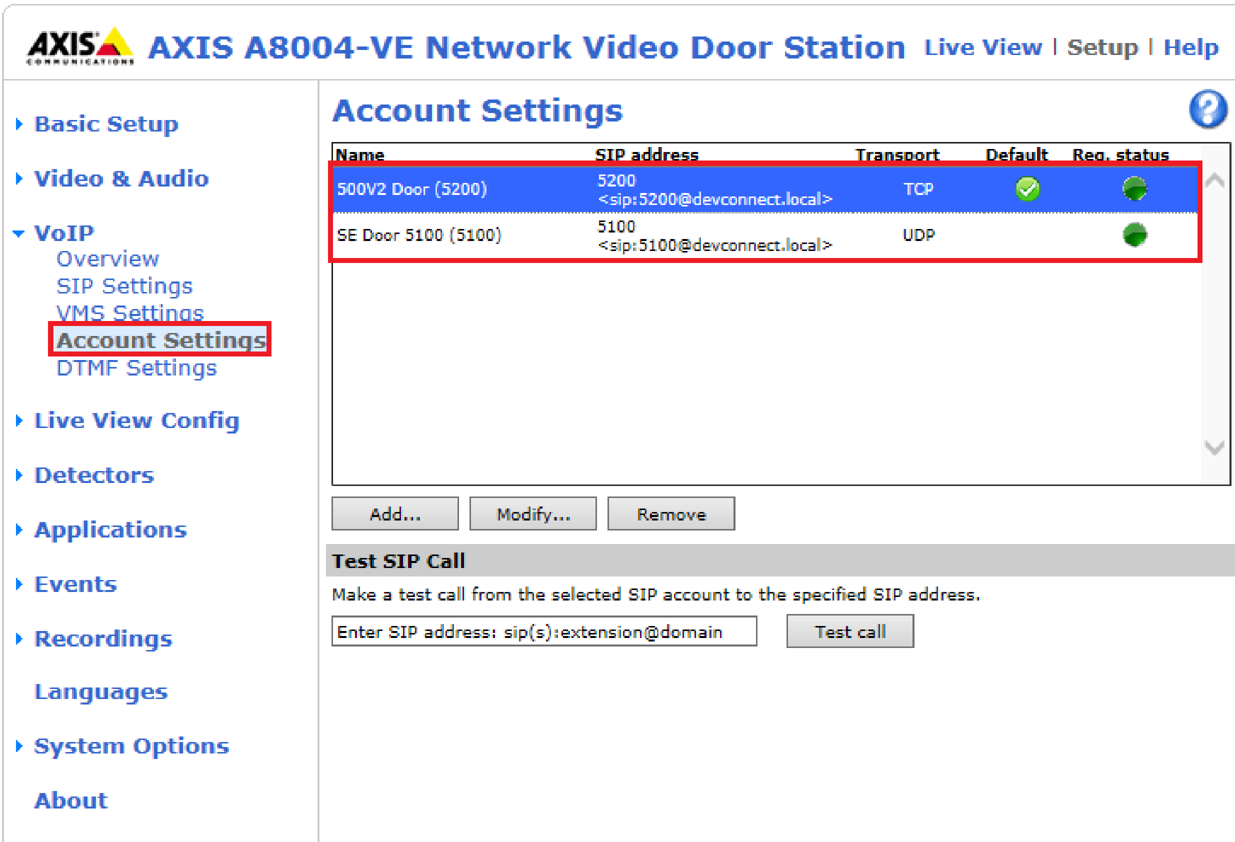


Once connected to the desired IP Office information on SIP calls and registrations will be shown (as long as the correct filter is applies for SIP messaging (not shown)). Below is an example of a message being displayed when a call is made from the door phone to extension **5201** which is a digital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

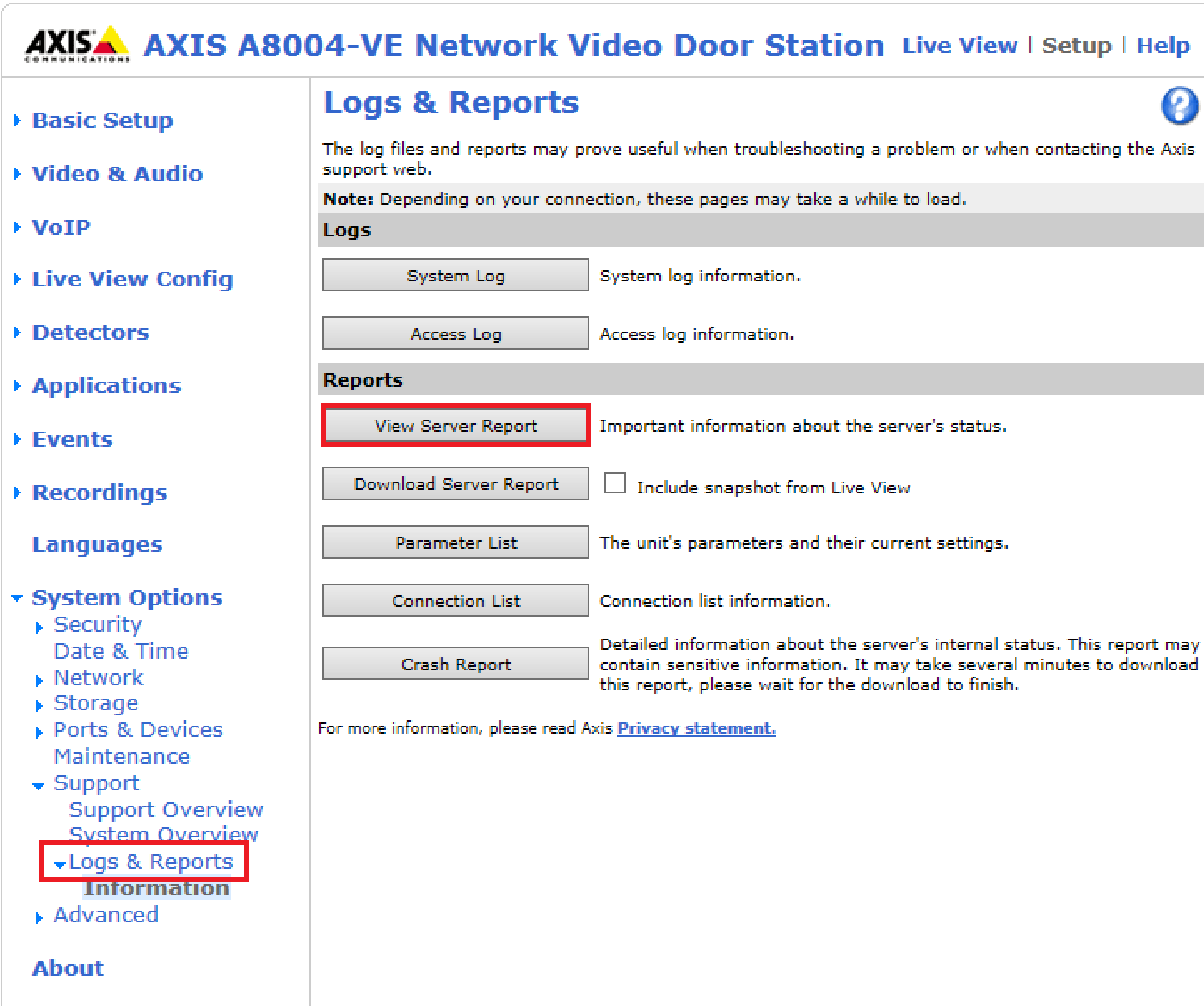


7.2. Verify Registration from AXIS A8004-VE Network Video Door Station

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.



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.



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"
```

8. Conclusion

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

9. Additional References

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

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

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

[2] Avaya IP Office R9.1 Doc library

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

Axis Communications AB
Tel: +46 46 272 18 00
Fax: +46 46 13 61 30
http://www.axis.com/global/en/learning-and-support

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.