



Application Notes for configuring Axis Communications AB AXIS C3003-E Network Horn Speaker with Avaya Aura® Communication Manager and Avaya Aura® Session Manager R7.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning the AXIS C3003-E Network Horn Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager R7.1.

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 C3003-E Network Horn Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager R7.1.

AXIS C3003-E Network Horn Speaker is an outdoor loudspeaker that provides clear, long-range speech for remote speaking in video surveillance applications. In live video monitoring situations, AXIS C3003-E enables an operator to remotely address people and deter unwanted activity. The loudspeaker can also play a pre-recorded audio file when it is manually or automatically triggered in response to an alarm event.

The unit supports Session Initiation Protocol (SIP) for easy integration with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The AXIS C3003-E makes announcements possible from anywhere with network connectivity. It easily integrates with video management software (VMS) that support two-way audio and with Voice over IP (VoIP) telephony systems that use SIP (Session Initiation Protocol).

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of the AXIS C3003-E Network Horn Speaker (Axis Speaker) to receive calls from Avaya Digital, H.323 and SIP desk phones as well as mobile/PSTN endpoints. The speaker is registered to Session Manager as a SIP endpoint.

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.

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 C3003-E Network Horn Speaker did not include use of any specific encryption features as requested by Axis Communications.

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 speaker.
- Invalid usernames/passwords for registration.
- Basic calls.
- Codec support.
- Serviceability testing.

2.2. Test Results

All test cases passed successfully with no issues or observations.

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 C3003-E Network Horn Speaker 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 C3003-E Network Horn Speaker from Axis Communications AB with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

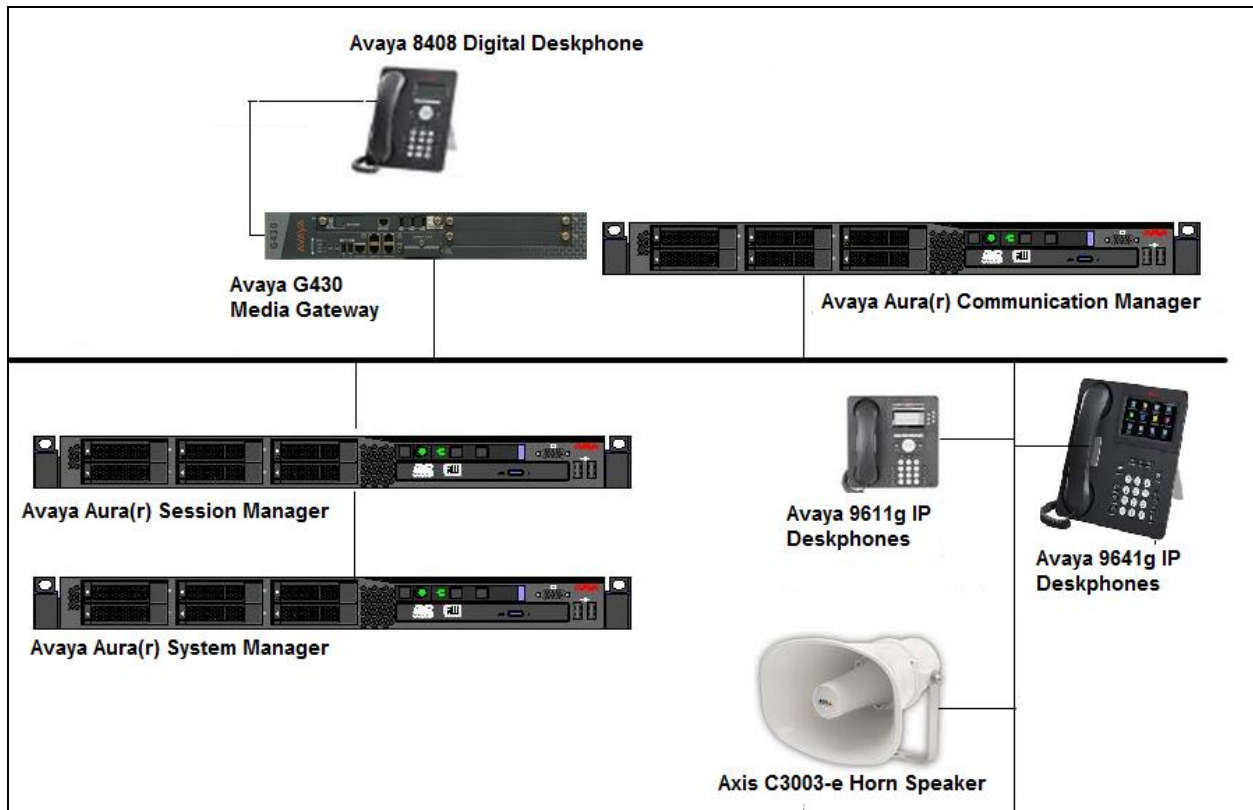


Figure 1: Connection of Axis Communications AB C3003-E Network Horn Speaker with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

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
Avaya Aura® Session Manager running on a virtual platform	R 7.0.1.1.701114
Avaya Aura® System Manager running on a virtual platform	R 7.0.1.2 Revision 7.0.1.2.075662 Service Pack 2
Avaya 9611g IP Deskphone	H.323 Release 6.6029
Avaya 9641G IP Deskphone	SIP 7.0.1.1
Avaya 2420 Digital Deskphone	V 2.0
Axis Communications AB AXIS C3003-E Network Horn speaker	Firmware Version 1.65.032

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

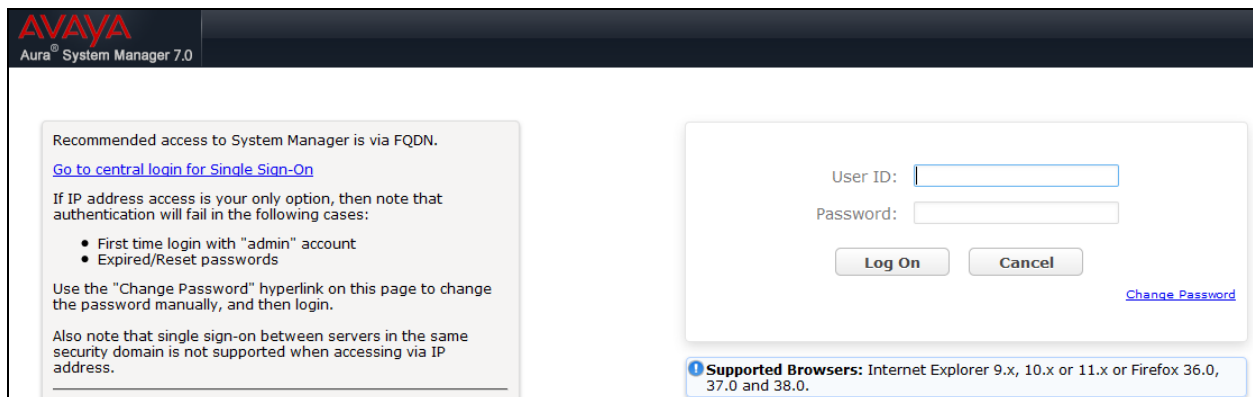
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are seven digits long and begin with **8**.

change dialplan analysis		Page 1 of 12
DIAL PLAN ANALYSIS TABLE		
Location: all		Percent Full: 1
Dialed String	Total Call Length Type	Dialed String Total Call Length Type
8	3 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 AXIS C3003-E Network Horn Speaker. It is assumed that the Domains, Locations, SIP entities for each Session Manager, Communication Manager, 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.



AVAYA
Aura® System Manager 7.0

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.

User ID:

Password:

[Change Password](#)

Supported Browsers: 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 AXIS C3003-E Registration

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

Port

TCP Failover port: UDP ▼

TLS Failover port:

Add Remove

3 Items Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	UDP ▼	devconnect.local ▼	<input type="text"/>
<input type="checkbox"/>	5060	TCP ▼	devconnect.local ▼	<input type="text"/>
<input type="checkbox"/>	5061	TLS ▼	devconnect.local ▼	<input type="text"/>

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

Repeat accordingly on the alternative Session Manager.

6.2. Add AXIS C3003-E User

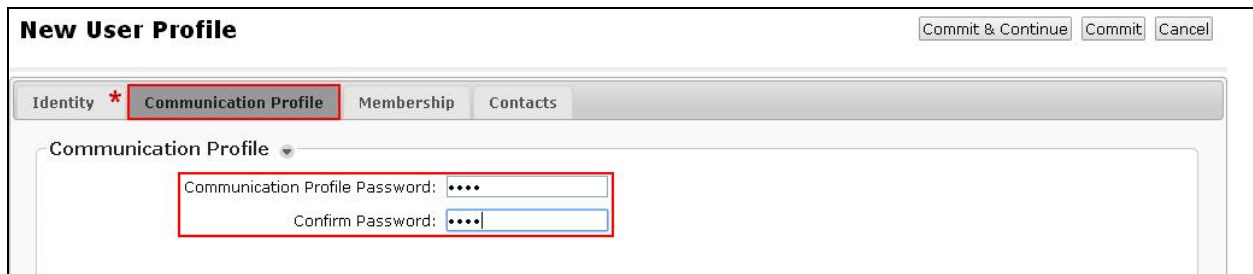
The AXIS C3003-E Network Horn Speaker must be added as a user. A user must be added for each AXIS C3003-E Network Horn Speaker. Click **User Management** → **Manage Users** → **New** (not shown) and configure as following in the **Identity** tab.

- **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 displays a web-based user provisioning interface. On the left is a sidebar menu with options: System Presence, ACLs, Communication, Profile Password, and Policy. The main area has four tabs: Identity (marked with a red asterisk), Communication Profile, Membership, and Contacts. The 'Identity' tab is active, showing a 'User Provisioning Rule' dropdown and an 'Identity' section with various input fields. The fields are filled with the following values: Last Name: Speaker, Last Name (Latin Translation): Speaker, First Name: Horn, First Name (Latin Translation): Horn, Middle Name: (empty), Description: (empty), Login Name: 8275060@devconnect.local, User Type: Basic, Password: (masked with dots), and Confirm Password: (masked with dots).

Field	Value
User Provisioning Rule	[Dropdown]
Last Name	Speaker
Last Name (Latin Translation)	Speaker
First Name	Horn
First Name (Latin Translation)	Horn
Middle Name	
Description	
Login Name	8275060@devconnect.local
User Type	Basic
Password	*****
Confirm 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 Network Horn Speaker during login.



New User Profile Commit & Continue Commit Cancel

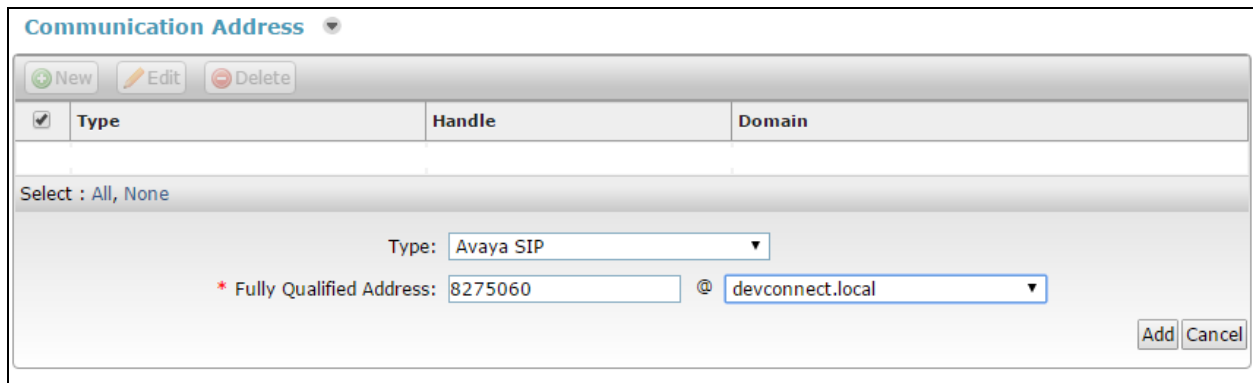
Identity * **Communication Profile** Membership Contacts

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.

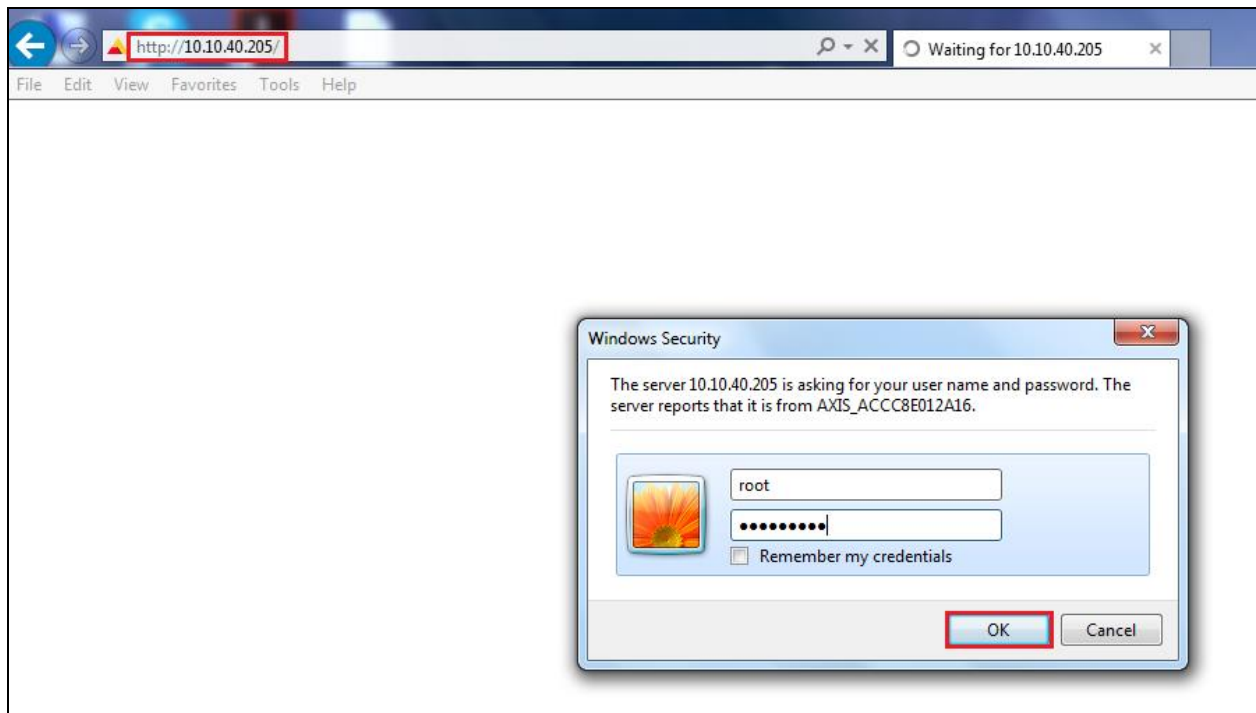
General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)		Enhanced Call Fwd (E)		Button Assignment (B)																					
Group Membership (M)																															
Active Station Ringing MWI Served User Type Per Station CPN - Send Calling Number IP Phone Group 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 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 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																														
<input type="checkbox"/> IP Audio Hairpinning	<input type="checkbox"/> IP SoftPhone																														
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<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																															

7. Configure AXIS C3003-E Network Horn Speaker

The configuration of the Axis speaker uses a web interface.

Note: The speaker 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 speaker, enter the proper 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 speaker configuration. The following sections cover specific settings concerning SIP and the connection to Session Manager.

7.1. Audio Settings

Although the audio settings are not relevant to the SIP connection with Communication Manager it is important as it governs the volume from the speaker and so it is shown below how to adjust this under **Audio** → **Audio Settings**.

AXIS C3003-E Network Speaker Setup | Help

Audio Settings

Auto Speaker Test
Test Status: The Auto Speaker Test must be calibrated before use.

Calibrate Auto Speaker Test
Calibrate Status: The Auto Speaker Test must be calibrated before use.

Audio Channels
Audio mode: Simplex - Speaker only

Audio Output
Output gain: -35 dB

Save Reset

7.2. Configure SIP Settings

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 **Port Settings** select the SIP ports that are to be used and click on **Save** once all is configured correctly.

AXIS C3003-E Network Speaker Setup | Help

SIP Settings

SIP Settings
☒ Enable SIP

Incoming SIP Calls
☒ Allow incoming SIP calls

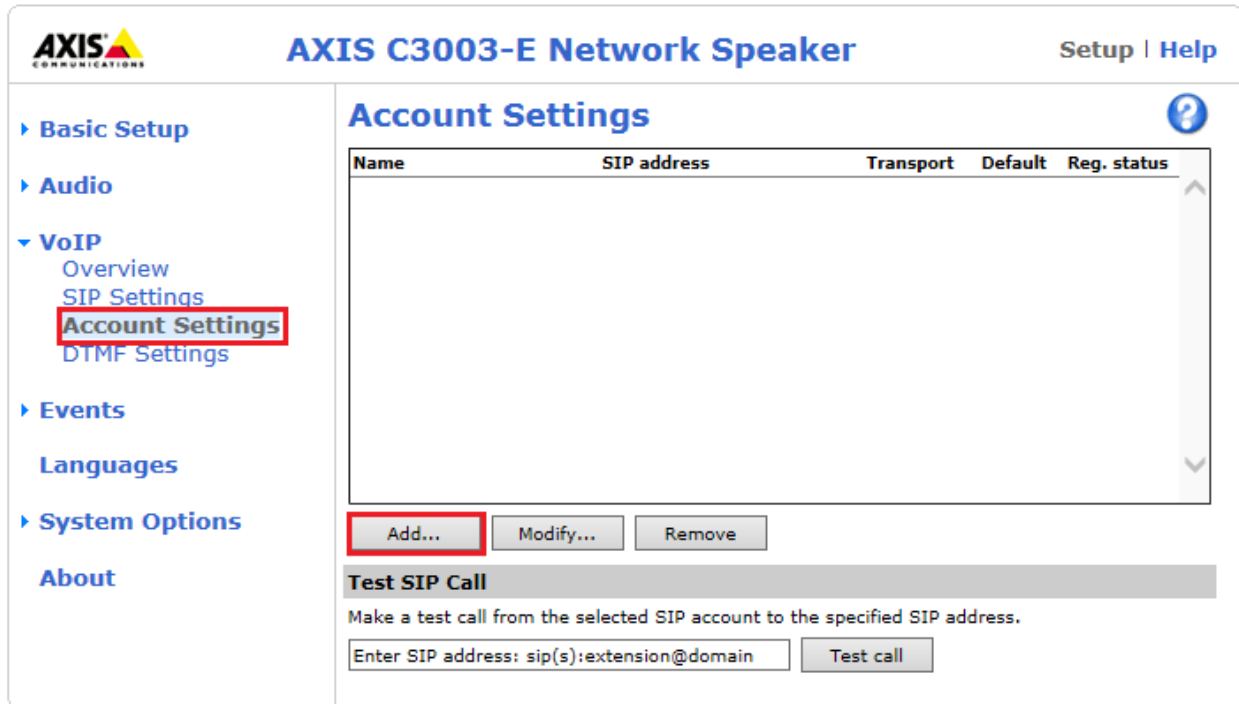
Port Settings
SIP port: 5060
SIP TLS port: 5061

NAT Traversal
☐ Enable ICE
☐ Enable STUN
☐ Enable TURN

Save Reset

7.3. Configure Account

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



The screenshot shows the web interface for the AXIS C3003-E Network Speaker. The left sidebar contains a navigation menu with the following items: Basic Setup, Audio, VoIP (expanded), Events, Languages, System Options, and About. Under the VoIP section, the sub-items are Overview, SIP Settings, Account Settings (highlighted with a red box), and DTMF Settings. The main content area is titled "Account Settings" and features a table with the following headers: Name, SIP address, Transport, Default, and Reg. status. The table is currently empty. Below the table are three buttons: Add... (highlighted with a red box), Modify..., and Remove. At the bottom of the main area is a "Test SIP Call" section with the instruction "Make a test call from the selected SIP account to the specified SIP address." Below this instruction is a text input field labeled "Enter SIP address: sip(s):extension@domain" and a "Test call" button.

AXIS C3003-E Network Speaker 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: sip(s):extension@domain **Test call**

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

- **Name:** Enter a suitable name for the SIP account.
- **User ID:** Enter the SIP user number configured.
- **Password:** Enter the password for the SIP user created.
- **Caller ID:** This should be the extension number created.
- **Domain Name:** The Session Manager telephony domain.
- **Registrar address:** The IP address of Session Manager.
- **Transport mode** This can be UDP, **TCP**.

Click on **OK** to save the configuration.

Modify Account ?

Account Information

Name:

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

Account Credentials

User ID:

☒ Use User ID as Authentication ID

Authentication ID:

Password:

Caller ID:

SIP Server Settings

Domain name:

Registrar address:

Transport Settings

☐ Enable SIPS

Transport mode:

☐ Allow port update messages through MWI

Proxy Settings

Address	Username

Account Status

8. Verification Steps

Calling from the Communication Manager set and ensuring there is clear audio heard at the Horn Speaker 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 Horn Speaker 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 AXIS C3003-E Network Horn Speaker is listed and a tick under **Registered** for the Session Manager it is registered to.

AVAYA
Aura® System Manager 7.0

Last Logged on at December 16, 2016 8:35
GO... Log off admin

Home Session Manager

Session Manager
Dashboard
Session Manager
Administration
Communication
Profile Editor
Network
Configuration
Device and Location
Configuration
Application
Configuration

Home / Elements / Session Manager / System Status / User Registrations

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View Default Force Unregister AST Device Notifications: Reboot Reload Fallback As of 11:46 AM Advanced Search Customize

24 Items Show 15 Filter: Enable

	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 type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<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 C3003-E Network Horn Speaker

Log in to the speaker as per the user details in **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 **8275061** 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 C3003-E Network Speaker. The left sidebar contains a navigation menu with the following items: Basic Setup, Audio, VoIP (expanded), Applications, Events, Languages, System Options, and About. Under the VoIP section, the sub-items are Overview, SIP Settings, Account Settings (highlighted), and DTMF Settings. The main content area is titled 'Account Settings' and features a table with the following data:

Name	SIP address	Transport	Default	Reg. status
Speaker (8275061)	8275061 <sip:8275061@devconnect.l ocal>	UDP	✓	●

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

If there is an issue with a call to the Axis speaker 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 also the System Log is available as shown below.

The screenshot displays the web interface for the AXIS C3003-E Network Speaker. The left sidebar contains a navigation menu with the following items: Basic Setup, Audio, VoIP, Events, Languages, System Options (expanded), Security, Date & Time, Network, Ports & Devices, Maintenance, Support (expanded), Support Overview, System Overview, Logs & Reports Information (highlighted with a red box), Advanced, and About. The main content area is titled "AXIS C3003-E Network Speaker" and "Setup | Help". The "Logs & Reports" section is active, showing a help icon and a note: "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." Below this, there are two sections: "Logs" and "Reports". The "Logs" section contains two buttons: "System Log" (highlighted with a red box) and "Access Log". The "Reports" section contains four buttons: "View Server Report" (highlighted with a red box), "Download Server Report", "Parameter List", and "Connection List". A "Crash Report" button is also present, with a detailed description: "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." At the bottom, there is a link to the Axis Privacy statement.

AXIS C3003-E Network Speaker 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

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](#).

9. Conclusion

These Application Notes describe the configuration steps for provisioning the AXIS C3003-E Network Horn Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager 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, May 2017, Document Number 03-300509, Issue 1.*

[2] *Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.1, May 2017, Document Number 555-245-205, Issue 1.*

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

Administering Avaya Aura® System Manager, Release 7.1, Issue 1, August, 2017

Technical information for the AXIS C3003-E Network Horn Speaker can be obtained from:

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

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