



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Axis Communications AB AXIS C2005 Network Ceiling Speaker with Avaya Aura® Communication Manager R7.0.1 and Avaya Aura® Session Manager R7.0.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning the AXIS C2005 Network Ceiling Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

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 C2005 Network Ceiling Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

AXIS C2005 Network Ceiling Speaker is an indoor loudspeaker that provides clear, long-range speech for remote speaking. The loudspeaker can 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 and the AXIS C2005 Network Ceiling Speaker 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 C2005 Network Ceiling 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.

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 C2005 Network Ceiling 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 C2005 Network Ceiling Speaker from Axis Communications AB with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

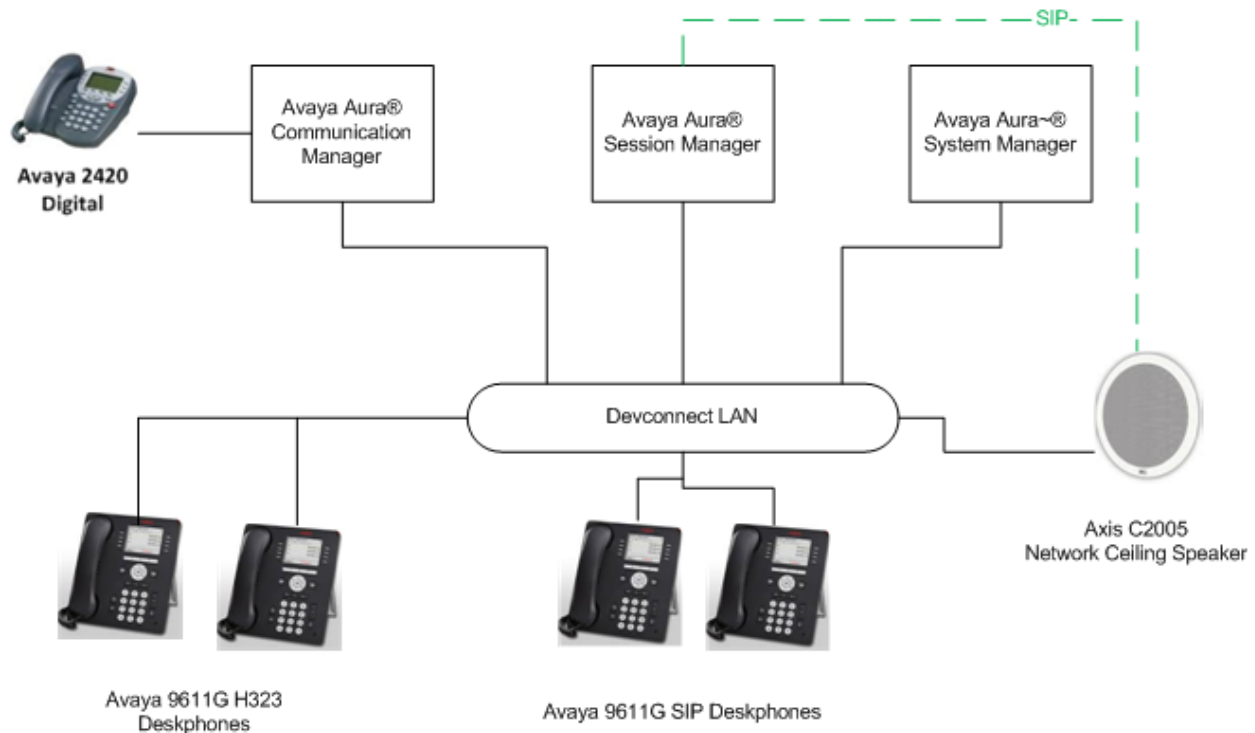


Figure 1: Connection of Axis Communications AB AXIS C2005 Network Ceiling Speaker with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

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 Deskphone	H.323 Release 6.6029
Avaya 9611G Deskphone	SIP 7.0.1.1
Avaya 2420 Digital Deskphone	V 2.0
Axis Communications AB AXIS C2005 Network Ceiling Speaker	Firmware Version 1.25.1

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
- Configure IP-Codec Set

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

5.1. Verify System Capacity

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

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

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

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

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

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

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

5.2. Define the Dial Plan

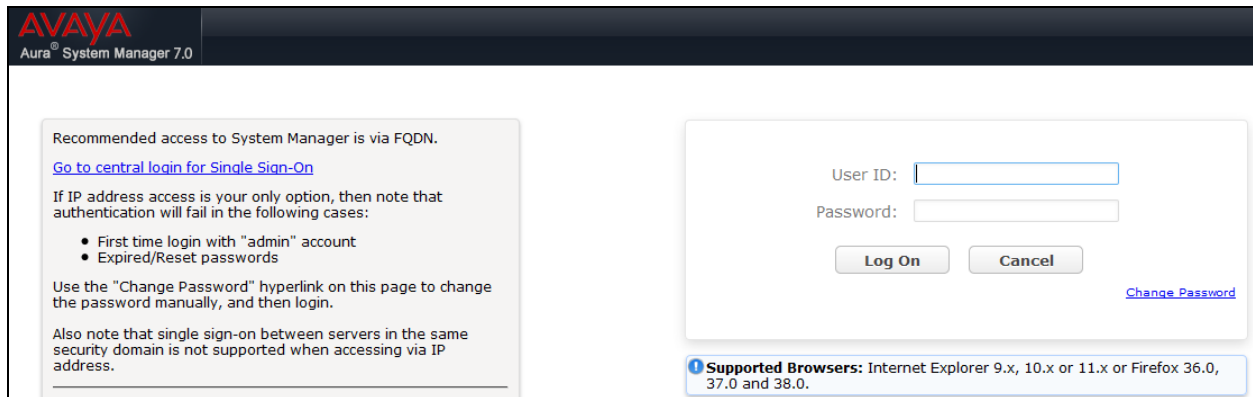
Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are seven digits long and begin with **8**.

change dialplan analysis		Page 1 of 12
DIAL PLAN ANALYSIS TABLE		
Location: all		Percent Full: 1
Dialed String	Total Length	Call Type
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 C2005 Network Ceiling 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 Session Manager ports for AXIS C2005 Registration

Each Session Manager Entity must be configured so that the Network Ceiling 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.

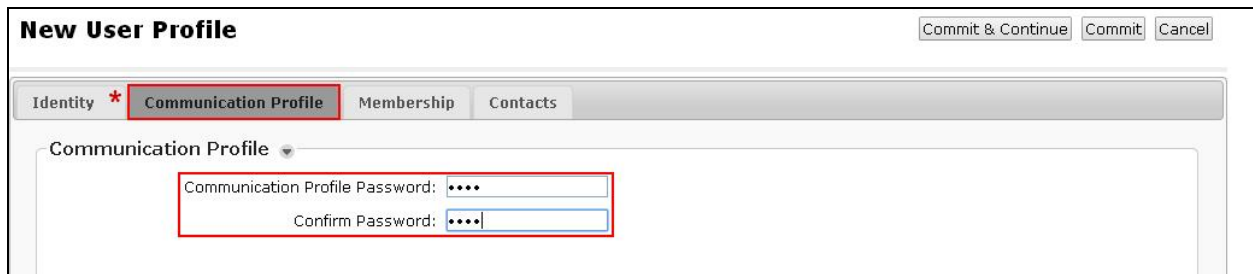
6.2. Add AXIS C2005 User

The AXIS C2005 Network Ceiling Speaker must be added as a user. A user must be added for each AXIS C2005 Network Ceiling 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 tabs for Identity (marked with a red asterisk), Communication Profile, Membership, and Contacts. The 'User Provisioning Rule' dropdown is set to a default value. The 'Identity' section is expanded, showing fields for: Last Name (Speaker), Last Name (Latin Translation) (Speaker), First Name (Ceiling), First Name (Latin Translation) (Ceiling), Middle Name (empty), Description (empty), Login Name (8275060@devconnect.local), User Type (Basic), Password (masked with dots), and Confirm Password (masked with dots).

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 Ceiling Speaker during login.



New User Profile Commit & Continue Commit Cancel

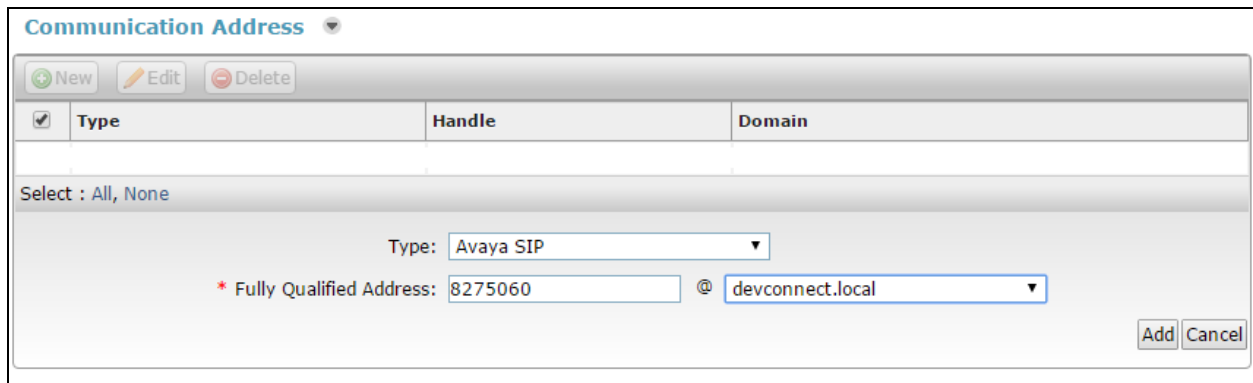
Identity * **Communication Profile** Membership Contacts

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 **DEFAULT_9611SIP_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.

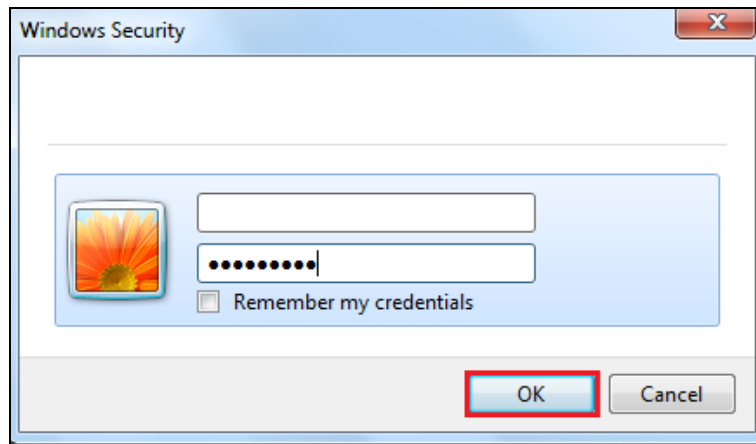
General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)		Enhanced Call Fwd (E)		Button Assignment (B)	
Group Membership (M)											
Active Station Ringing				single ▼		Auto Answer				none ▼	
MWI Served User Type				None ▼		Coverage After Forwarding				system ▼	
Per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

7. Configure AXIS C2005 Network Ceiling 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

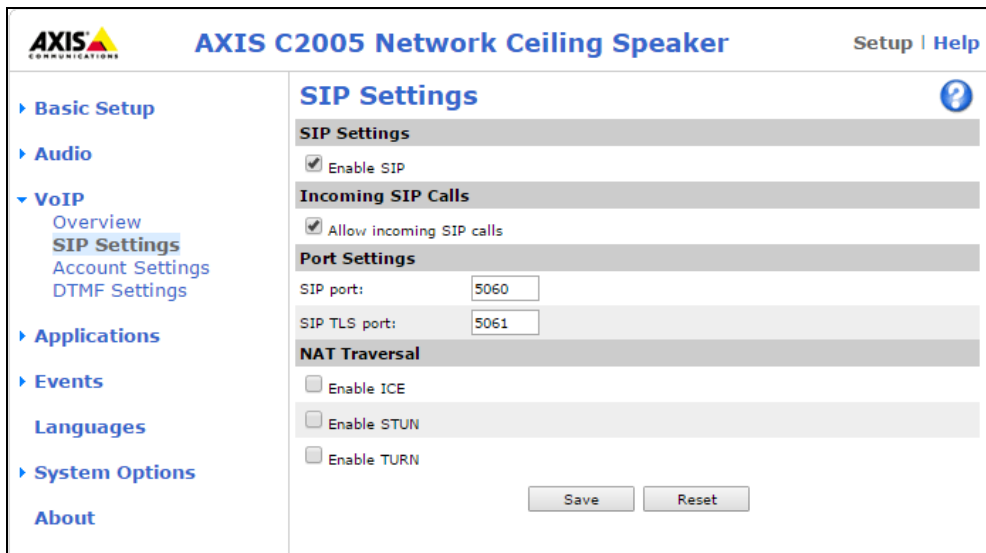
Although the audio settings are not relevant to the SIP connection with Session Manager it is important as it governs the volume from the speaker and so it is shown below how to adjust this under **Audio** → **Audio Settings**.



The screenshot shows the web interface for the AXIS C2005 Network Ceiling Speaker. The left sidebar contains a navigation menu with the following items: Basic Setup, Audio (selected), VoIP, Applications, Events, Languages, System Options, and About. Under the 'Audio' section, the sub-menu items are Audio Overview, Audio Settings (highlighted), and Audio Clips. The main content area is titled 'Audio Settings' and includes a help icon. It contains three sections: 'Auto Speaker Test' with 'Test' and 'Calibrate' buttons and a status message; 'Audio Channels' with an 'Audio mode' dropdown set to 'Simplex - Speaker only'; and 'Audio Output' with an 'Output gain' slider set to -10 dB. 'Save' and 'Reset' buttons are at the bottom.

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.



The screenshot shows the web interface for the AXIS C2005 Network Ceiling Speaker with the 'SIP Settings' page selected. The left sidebar navigation menu is the same as in the previous screenshot, but the 'VoIP' section is expanded, showing 'Overview', 'SIP Settings' (highlighted), 'Account Settings', and 'DTMF Settings'. The main content area is titled 'SIP Settings' and includes a help icon. It contains three sections: 'SIP Settings' with a checked 'Enable SIP' checkbox; 'Incoming SIP Calls' with a checked 'Allow incoming SIP calls' checkbox; and 'Port Settings' with input fields for 'SIP port' (5060) and 'SIP TLS port' (5061). A 'NAT Traversal' section has unchecked checkboxes for 'Enable ICE', 'Enable STUN', and 'Enable TURN'. 'Save' and 'Reset' buttons are at the bottom.

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 C2005 Network Ceiling 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 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, there is a section titled 'Test SIP Call' 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 COMMUNICATIONS **AXIS C2005 Network Ceiling 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 in **Section 6.2**.
- **Password:** Enter the password for the SIP user created in **Section 6.2**.
- **Caller ID:** This should be the extension number created in **Section 6.2**.
- **Domain Name:** The Session Manager SIP domain.
- **Registrar address:** The IP address of the Session Manager
- **Transport mode** This can be **UDP**, TCP or TLS. TLS was not tested as no 3rd Party Certificates were exchanged.

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

The screenshot shows a dialog box titled "Add Account" with a help icon in the top right corner. The dialog is organized into several sections:

- Account Information:** Includes a "Name:" field with the value "Speaker" and a checkbox for "Default account" (Note that only one account can be the default account.).
- Account Credentials:** Includes a "User ID:" field with the value "8275060", a checkbox for "Use User ID as Authentication ID", an "Authentication ID:" field with the value "8275060", a "Password:" field with masked characters "*****", and a "Caller ID:" field with the value "123456".
- SIP Server Settings:** Includes a "Domain name:" field with the value "devconnect.local" and a "Registrar address:" field with the value "10.10.16.77".
- Transport Settings:** Includes a checkbox for "Enable SIPs", a "Transport mode:" dropdown menu currently set to "UDP", and a checkbox for "Allow port update messages through MWI".
- Proxy Settings:** Includes a table with two columns: "Address" and "Username". The table is currently empty. To the right of the table are up and down arrow buttons. Below the table is an "Add..." button.

At the bottom of the dialog are "OK" and "Cancel" buttons.

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 and 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 AXIS C2005 Network Ceiling Speaker is listed and a tick under **Registered** for the Session Manager it is registered to.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left-hand navigation menu includes options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, and Application Configuration. The main content area is titled 'User Registrations' and shows a list of 24 items. The table below represents the data shown in the screenshot:

	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 C2005 Network Ceiling Speaker

Log in to the speaker 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**. 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 for the AXIS C2005 Network Ceiling Speaker. The top header includes the AXIS logo, the product name "AXIS C2005 Network Ceiling Speaker", and links for "Setup" and "Help". A left sidebar contains a navigation menu with options: Basic Setup, Audio, VoIP (expanded), Applications, Events, Languages, System Options, and About. Under the VoIP section, the sub-menu 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 main area is a section titled "Test SIP Call" with the instruction "Make a test call from the selected SIP account to the specified SIP address." This section includes a text input field labeled "Enter SIP address: sip(s):extension@domain" and a "Test call" button.

In the event of an issue with a call to the Axis speaker there are logs that can be accessed that 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 shows the web interface for the AXIS C2005 Network Ceiling Speaker. The left sidebar contains a navigation menu with the following items: Basic Setup, Audio, VoIP, Applications, Events, Languages, System Options (expanded), Security, Date & Time, Network, Storage, Ports & Devices, Maintenance, Support (expanded), Support Overview, System Overview, Logs & Reports (highlighted), Information, Advanced, and About. The main content area is titled 'AXIS C2005 Network Ceiling 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.' Below this, a 'Note' states: 'Depending on your connection, these pages may take a while to load.' The 'Logs' section includes buttons for 'System Log' (System log information) and 'Access Log' (Access log information). The 'Reports' section includes buttons for 'View Server Report' (Important information about the server's status), 'Download Server Report' (with an unchecked checkbox for 'Include snapshot from Live View'), 'Parameter List' (The unit's parameters and their current settings), 'Connection List' (Connection list information), and '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.). At the bottom, a link to the 'Privacy statement' is provided.

AXIS C2005 Network Ceiling 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 ☐ Include snapshot from Live View

Parameter List The unit's parameters and their current settings.

Connection List Connection list information.

Crash Report Detailed information about the server's internal status. This report may contain sensitive information. It may take several minutes to download this report, please wait for the download to finish.

For more information, please read Axis [Privacy statement](#).

9. Conclusion

These Application Notes describe the configuration steps for provisioning the AXIS C2005 Network Ceiling Speaker from Axis Communications AB to interoperate with Avaya Aura® Communication Manager R7.0.1 and Avaya Aura® Session Manager R7.0.1. 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 7.0, August 2015, 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 7.0, Issue 1 August 2015*

[4] *Administering Avaya Aura® System Manager, Release 7.0, Issue 1, August, 2015*

Technical information for the AXIS C2005 Network Ceiling Speaker can be obtained from:

Axis Communications AB

Tel: +46 46 272 18 00

Fax: +46 46 13 61 30

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

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