



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Dolby Conference Phone 3.4 with Avaya Aura® Session Manager 7.1 and Avaya Aura® Communication Manager 7.1 – Issue 1.0

### Abstract

These Application Notes describe the configuration steps required for Dolby Conference Phone 3.4 to interoperate with Avaya Aura® Session Manager 7.1 and Avaya Aura® Communication Manager 7.1.

Dolby Conference Phone is a conference speaker phone with a touch screen user interface. In the compliance testing, Dolby Conference Phone registered to Avaya Aura® Session Manager as a SIP endpoint.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 required for Dolby Conference Phone 3.4 to interoperate with Avaya Aura® Session Manager 7.1 and Avaya Aura® Communication Manager 7.1.

Dolby Conference Phone is a conference speaker phone with a touch screen user interface. In the compliance testing, Dolby Conference Phone registered to Avaya Aura® Session Manager as a SIP endpoint.

# 2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established between the Dolby Conference Phone user with Avaya SIP, Avaya H.323, and/or PSTN users. Call controls were performed from various users to verify call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to Dolby Conference Phone.

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

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

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

For the testing associated with these Application Notes, the interface between Session Manager and Dolby Conference Phone did not include use of any specific encryption features as requested by Dolby.

## 2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing.

The feature testing included registration, basic call, display, mute/unmute, hold/resume, drop, media shuffling, G.711, G.729, G.722, codec negotiation, blind transfer, attended conference, music on hold, DTMF, long hold and held call reminder, session refresh, long duration, coverage, and multiple calls. In addition, testing also included feature access code dialing with asterisk and pound, and for extended telephony features of call forwarding, call park/unpark, and call pickup.

The serviceability testing focused on verifying the ability of Dolby Conference Phone to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to Dolby Conference Phone.

## 2.2. Test Results

All test cases were executed and verified. The following were observations on Dolby Conference Phone from the compliance testing.

- By design, Dolby Conference Phone does not support attended transfer, blind conference, and message waiting indicator.
- By design, only one incoming call can be ringing at Dolby Conference Phone at any one time, with 486 Busy Here returned for additional simultaneous incoming calls.

## 2.3. Support

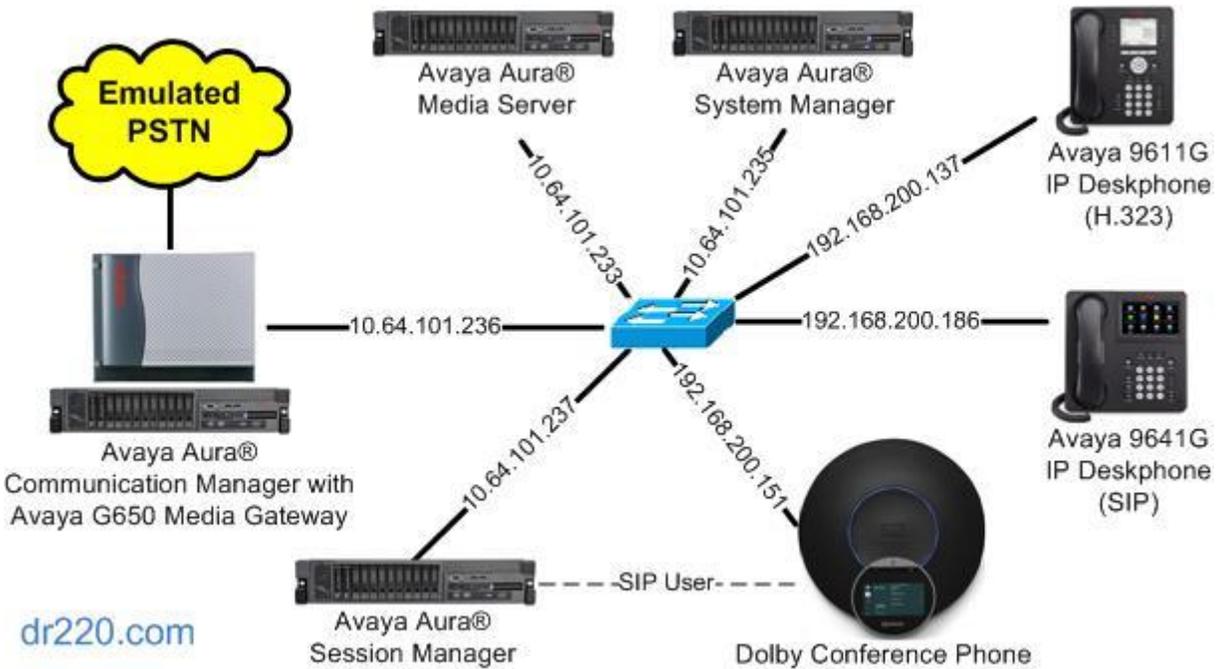
Technical support on Dolby Conference Phone can be obtained through the following:

- **Web:** <http://www.dolby.com/us/en/support/conferencing-support.html>

### 3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**, with the domain name used in the testing being “dr220.com”.

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager are not the focus of these Application Notes and will not be described.



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

<b>Equipment/Software</b>	<b>Release/Version</b>
Avaya Aura® Communication Manager in Virtual Environment	7.1.2 (7.1.2.0.0.532.24184)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	7.8.0.333
Avaya Aura® Session Manager in Virtual Environment	7.1.2 (7.1.2.0.712004)
Avaya Aura® System Manager in Virtual Environment	7.1.2 (7.1.2.0.057353)
Avaya 9611G IP Deskphone (H.323)	6.6604
Avaya 9641G IP Deskphone (SIP)	7.1.1.0.9
Dolby Conference Phone VCP 9000	3.4.1.19

## 5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer IP codec set

### 5.1. Verify License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command to verify that there is sufficient license for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V17                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 65000 194
                                Maximum Stations: 41000 29
                                Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 1
Maximum Off-PBX Telephones - OPS: 41000 2
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
                                Maximum Survivable Processors: 313 0
```

## 5.2. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is an existing codec set number used for integration with Dolby Conference Phone.

Update the audio codec types in the **Audio Codec** fields as needed. Besides the two codecs shown below, G.722-64K was also verified in the compliance testing.

For customer network that uses encrypted media, make certain that “none” is included for **Media Encryption**, and that **Encrypted SRTP** is set to “best-effort”, these settings are needed for support of non-encrypted media from Dolby Conference Phone.

In the compliance testing, this IP codec set was used by Dolby Conference Phone and by all Avaya endpoints.

```
change ip-codec-set 1                                     Page 1 of 2

                                IP Codec Set

Codec Set: 1

Audio          Silence      Frames      Packet
Codec          Suppression Per Pkt      Size (ms)
1: G.711MU     n                2           20
2: G.729      n                2           20
3:
4:
5:
6:
7:

Media Encryption                                Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: aes
3: none
4:
5:
```

## 6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer users
- Administer Session Manager entity

### 6.1. Launch System Manager

Access the System Manager web-based interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.

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.

User ID:

Password:

[Change Password](#)

### 6.2. Administer Users

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.

AVAYA Aura System Manager 7.1

Last Logged on at Go...

Home / User Management

Home / Users / User Management / Manage Users

Search

Help ?

### User Management

Users

More Actions

Advanced Search

3 Items Show All Filter: Enable

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	Avaya	SIP 2	Avaya, SIP 2	66002@dr220.com	66002	
<input type="checkbox"/>	Avaya	SIP 4	Avaya, SIP 4	66004@dr220.com	66004	

## 6.2.1. Identity

The **New User Profile** screen is displayed. In the **Identity** sub-section, enter desired **Last Name** and **First Name**. For **Login Name**, enter “n@x”, where “n” is the desired user extension and “x” is the applicable domain name from **Section 3**. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.1', and a search bar. The main navigation menu on the left lists various management options, with 'User Management' expanded to show 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The 'Manage Users' option is selected, leading to the 'New User Profile' screen. The breadcrumb trail indicates the path: Home / Users / User Management / Manage Users. The 'New User Profile' form is divided into four tabs: 'Identity' (marked with a red asterisk), 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active and contains a 'User Provisioning Rule' dropdown menu. Below this, the 'Identity' section includes several input fields: 'Last Name' (Dolby), 'Last Name (Latin Translation)' (Dolby), 'First Name' (VCP), 'First Name (Latin Translation)' (VCP), 'Middle Name', 'Description', 'Login Name' (66009@dr220.com), 'Email Address', 'User Type' (Basic), 'Password', 'Confirm Password', 'Localized Display Name', 'Endpoint Display Name', and 'Title'. A 'Commit & Continue' button is located in the top right corner of the form area.

## 6.2.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.

In the **Communication Address** sub-section, click **New** to add a new address. The sub-section is updated with additional fields, as shown below. For **Type**, retain “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.2.1** respectively. Click **Add**.

The screenshot displays the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.1', and a search bar. The main navigation menu on the left lists 'User Management' with sub-items: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The 'Manage Users' sub-item is selected. The breadcrumb trail shows 'Home / Users / User Management / Manage Users'. The main content area is titled 'New User Profile' and contains several tabs: 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active. Below the tabs, there are fields for 'Communication Profile Password' and 'Confirm Password', both masked with dots. A 'Generate' button is next to the 'Confirm Password' field. Below this is a 'Name' section with a 'New' button, a 'Delete' button, a 'Done' button, and a 'Cancel' button. The 'Name' field is set to 'Primary' and is marked as the default. Below the 'Name' section is the 'Communication Address' section, which has a 'New' button circled in red, an 'Edit' button, and a 'Delete' button. Below these buttons is a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'. Below the table, there are fields for 'Type' (set to 'Avaya SIP'), 'Fully Qualified Address' (set to '66009 @ dr220.com'), and an 'Add' button circled in red. At the bottom of the form, there is a 'Session Manager Profile' section.

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location**, select values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

**Communication Address** ▾

<input type="checkbox"/>	Type	Handle	Domain
<input type="checkbox"/>	Avaya SIP	66009	dr220.com

Select : All, None

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**Session Manager Profile** ▾

**SIP Registration**

\* Primary Session Manager 

Primary	Secondary	Maximum
4	0	4

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

Block New Registration When Maximum Registrations Active?

**Application Sequences**

Origination Sequence

Termination Sequence

**Emergency Calling Application Sequences**

Emergency Calling Origination Sequence

Emergency Calling Termination Sequence

**Call Routing Settings**

\* Home Location

Conference Factory Set

**Call History Settings**

Enable Centralized Call History?

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**CM Endpoint Profile** ▸

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select value corresponding to the applicable Communication Manager. For **Extension**, click and select the SIP user extension from **Section 6.2.1**. For **Template**, select “9608SIP\_DEFAULT\_CM\_7\_1”. Retain the default values in the remaining fields.

The screenshot shows a configuration page for a Session Manager Profile. The 'CM Endpoint Profile' section is expanded, showing various fields and checkboxes. The 'System' field is set to 'DR220-CM7-ES', 'Profile Type' is 'Endpoint', and 'Extension' is '66009'. The 'Template' is set to '9608SIP\_DEFAULT\_CM\_7\_1'. Other fields include 'Set Type' (9608SIP), 'Security Code', 'Port' (IP), 'Voice Mail Number', and 'Preferred Handle' (None). There are several checkboxes for advanced options, with 'Delete Endpoint on Unassign of Endpoint from User or on Delete User', 'Override Endpoint Name and Localized Name', and 'Allow H.323 and SIP Endpoint Dual Registration' checked.

Session Manager Profile ▾

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CM Endpoint Profile ▾

\* System

\* Profile Type

Use Existing Endpoints

\* Extension  [Display Extension Ranges](#)

\* Template

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Calculate Route Pattern

Sip Trunk

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

### 6.3. Administer Session Manager Entity

Select **Routing** → **SIP Entities** from the left pane, followed by the applicable SIP entity for Session Manager, in this case “DR-SM7”. The **SIP Entity Details** screen is displayed.

AVAYA  
Aura® System Manager 7.1

Home / Elements / Routing / SIP Entities

### SIP Entity Details

Commit

**General**

\* Name: DR-SM7

\* FQDN or IP Address: 10.64.101.238

Type: Session Manager

Notes: TLT DR SM7

Location: DR-Loc

Outbound Proxy:

Time Zone: America/New\_York

Minimum TLS Version: Use Global Setting

Credential name:

Scroll down to **Listen Ports** and verify that the transport protocol used by Dolby Conference Phone is specified in the list, as shown below. Note that Dolby Conference Phone can support UDP or TCP with Session Manager, and the compliance testing used UDP.

**Failover Ports**

TCP Failover port:

TLS Failover port:

**Listen Ports**

Add Remove

3 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	dr220.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	dr220.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	dr220.com	<input checked="" type="checkbox"/>	

Select : All, None

## 7. Configure Dolby Conference Phone

This section provides the procedures for configuring Dolby Conference Phone. The procedures include the following areas:

- Launch web interface
- Administer IP PBX settings

Prior to configuration, follow the procedures in reference [4] to manually set the IP address of Dolby Conference Phone.

### 7.1. Launch Web Interface

Access the Dolby Conference Phone web-based interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the phone. Log in using the appropriate credentials.



The screenshot shows the login page for the Dolby Conference Phone web interface. At the top center is the Dolby logo, consisting of a square with a stylized 'D' inside, followed by the word 'DOLBY.' in a bold, sans-serif font. Below the logo is the text 'Conference Phone' in a smaller, regular font. Underneath this are two input fields: the first is labeled 'Username' and the second is labeled 'Password'. Below the input fields is a blue rectangular button with the text 'Log In' in white. At the bottom of the page, centered, is the copyright notice: 'Dolby Laboratories© 2014 - 2016. All rights reserved.'

## 7.2. Administer IP PBX Settings

The **Settings** screen is displayed. Expand **IP PBX Settings** followed by **Account**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Display Name:** A desired name.
- **Extension Number/Address:** The user extension from **Section 6.2.1**.
- **Display Number:** The user extension from **Section 6.2.1**.
- **Transport Type:** The applicable transport protocol from **Section 6.3**.

The screenshot shows the Dolby VCP 1 administration interface. At the top, the user is logged in as 'admin' and can click 'Log Out'. The 'Settings' menu is active, and the 'IP PBX Settings' section is expanded to show the 'Account' configuration. The fields are as follows:

Field	Value
Display Name	Dolby VCP 1
Extension Number/Address	66009
Display Number	66009
Transport Type	UDP
Secure Media	Mandatory
Transport Port	5060

Expand **Server** and **Credential**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **SIP Domain Name:** The domain name from **Section 3**.
- **Primary Call Server:** IP address of the Session Manager signaling interface.
- **PBX Codec List:** The applicable and desired codecs.
- **User Credential Name:** The user extension from **Section 6.2.1**.
- **User Credential Password:** The user password from **Section 6.2.2**.

The screenshot displays the 'IP PBX Settings' configuration page. It is organized into several sections: 'Account', 'Server', 'Credential', and 'NAT'. The 'Server' section is expanded, showing various SIP-related parameters. The 'Credential' section is also expanded, showing user authentication details. The 'NAT' section is collapsed.

Section	Field Name	Value
Server	SIP Domain Name	dr220.com
	Primary Call Server/Outbound Proxy	10.64.101.238
	Primary Server/Outbound Proxy Port	5060
	Secondary Call Server/Outbound Proxy	
	Secondary Server/Outbound Proxy Port	5060
	SIP URI Scheme	On
	Only accept call server SIP events	On
	SIP Registration Timeout (seconds)	32
	PBX Codec List	G711U,G729AB,G722
	In band DTMF	Off
Credential	User Credential Name	66009
	User Credential Password	.....
	Server Realm	*
NAT		

## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Session Manager and Dolby Conference Phone.

### 8.1. Verify Avaya Aura® Session Manager

From the System Manager web-based interface, select **Elements** → **Session Manager** → **System Status** → **User Registrations** (not shown) to display the **User Registrations** screen.

Verify that the user from **Section 6.2** is registered, as shown below with a check in the **Registered Prim** column.

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

### User Registrations

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

Customize ▾

View ▾ Default Export Force Unregister **AST Device Notifications:** Reboot Reload ▾ Failback **As of 2:02 PM**

2 Items Show All ▾ Filter: Enable

<input type="checkbox"/>	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered	
											Prim	Sec
<input type="checkbox"/>	► Show	66002@dr220.com	SIP 2	Avaya	NJ-Loc	192.168.200.186	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)
<input type="checkbox"/>	► Show	66004@dr220.com	SIP 4	Avaya	NJ-Loc	---	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

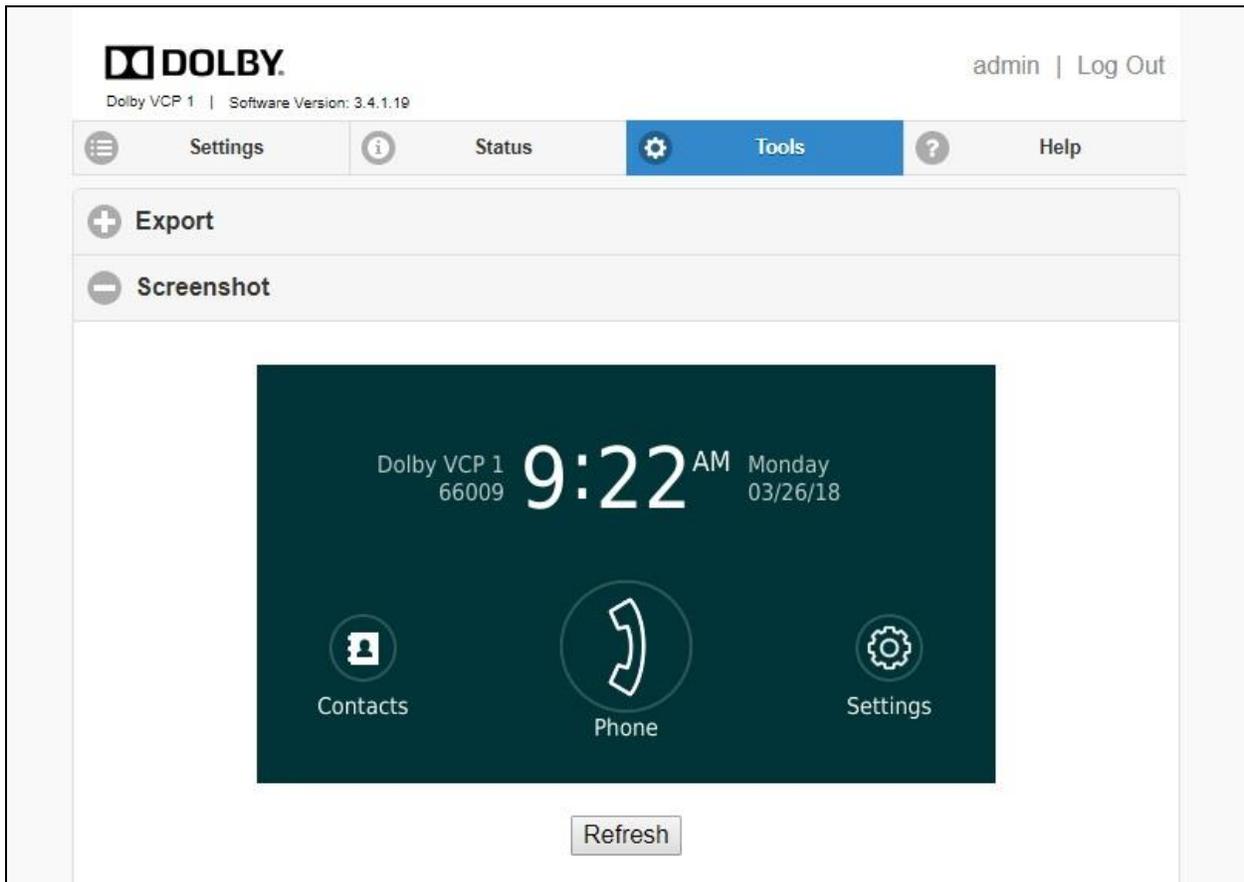
Select : All, None

## 8.2. Verify Dolby Conference phone

From the Dolby Conference Phone web-based interface, select **Tools** from the top menu. Expand **Screenshot** for a snapshot of the current display on the Dolby Conference Phone.

Verify that the phone displays the proper user name and extension in the upper left portion of screen, and that there is a **Phone** icon in the middle lower portion of screen as shown in the snapshot below, which is an indication that the phone has registered successfully.

In the case of unsuccessful registration, the **Phone** icon would not be shown, and there will be a red exclamation mark on top of the **Settings** icon.



## 9. Conclusion

These Application Notes describe the configuration steps required for Dolby Conference Phone 3.4 to successfully interoperate with Avaya Aura® Session Manager 7.1 and Avaya Aura® Communication Manager 7.1. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

## 10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura® Communication Manager*, Release 7.1.2, Issue 5, February 2018, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 7.1.2, Issue 4, March 2018, available at <http://support.avaya.com>.
3. *Administering Avaya Aura® System Manager for Release 7.1.2*, Issue 11, March 2018, available at <http://support.avaya.com>.
4. *Dolby Conference Phone Administrator's Guide*, Version 3.3.1, September 14, 2017, available at <http://www.dolby.com>.
5. *Dolby Conference Phone Configuration Guide for Avaya Aura Platform 6.x*, Version 3.3, July 31, 2017, available at <http://www.dolby.com>.

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