



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Plantronics Calisto 620-M Wireless Speakerphone with Avaya Flare® Experience for Windows - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required to integrate the Plantronics Calisto 620-M Wireless Speakerphone with Avaya Flare® Experience for Windows. The Calisto 620-M speakerphone provides two-way audio, allows the volume to be adjusted, and allows the audio to be muted directly from the speakerphone. This solution does not provide call control features directly from the speakerphone, such as call answer, call terminate, or mute sync. Calisto 620-M connects to the PC running Avaya Flare® Experience for Windows via a Bluetooth USB adapter.

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 to integrate the Plantronics Calisto 620-M Wireless Speakerphone with Avaya Flare® Experience for Windows. The Calisto 620-M speakerphone provides two-way audio, allows the volume to be adjusted, and allows the audio to be muted directly from the speakerphone. Calisto 620-M connects to the PC running Flare Experience via a Bluetooth USB adapter.

Refer to the appropriate Plantronics documentation listed in **Section 13** for additional product information.

**Note:** This solution does not provide call control integration with Avaya Flare® Experience for Windows. That is, a call cannot be answered or terminated directly from the speakerphone, nor is the mute status on the speakerphone synchronized with Flare Experience. The Plantronics Spokes Software is not installed as part of this solution.

# 2. General Test Approach

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets 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/handsets 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.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on placing calls to and from Flare Experience with the Calisto 620-M Wireless Speakerphone and verifying an audio path in both directions. The type of calls made included calls to voicemail, to local stations, and to the PSTN.

The serviceability testing focused on verifying the usability of the Calisto 620-M speakerphone after restarting Flare Experience, disconnecting and reconnecting the speakerphone, and rebooting the PC.

## 2.1. Interoperability Compliance Testing

All test cases were performed manually. The following features were verified:

- Placing calls to the voicemail system. Voice messages were recorded and played back to verify that the playback volume and recording level were good.
- Placing calls to local stations to verify two-way audio.
- Placing calls to the PSTN to verify two-way audio.
- Answering and ending calls directly from Flare Experience.
- Using the volume control buttons on the speakerphone and Flare Experience to adjust the playback volume.
- Using the mute control button on the speakerphone and Flare Experience to mute and un-mute the audio.

For the serviceability testing, the speakerphone was disconnected and reconnected to verify proper operation. Flare Experience application was also restarted for the same purpose. The desktop PC was also rebooted to verify that Flare Experience and the speakerphone were operational when the PC came back into service.

## 3. Test Results

All test cases passed with the following observations:

- There is no mute synchronization between the Calisto 620-M and Flare Experience. If a call is muted through the speakerphone, it is not reflected on Flare Experience, and the call needs to be un-muted through the speakerphone, and vice versa.
- There is no call control support through the speakerphone. Calls need to be answered and terminated through Flare Experience. The call control button on the speakerphone is not operational.

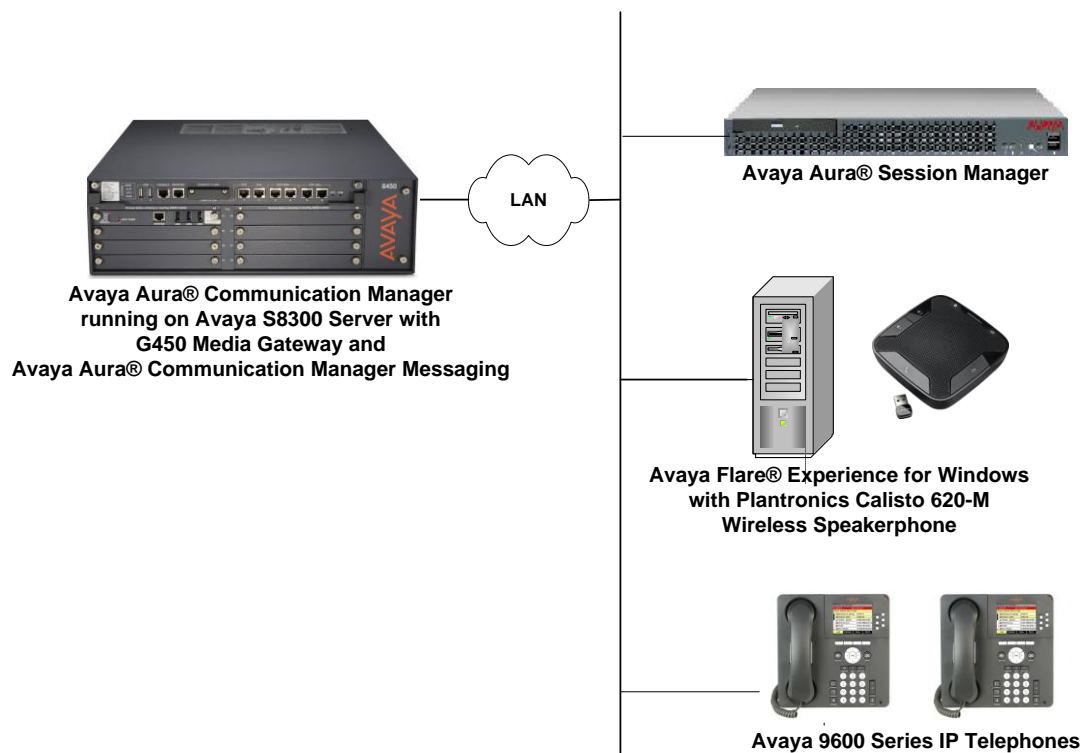
## 4. Support

For technical support and information on Plantronics Calisto 620-M Wireless Speakerphone, contact Plantronics at:

- Phone: 800-544-4660 (toll free)  
+1 831-426-5858 (International)
- Website: [http://www.plantronics.com/north\\_america/en\\_US/support/](http://www.plantronics.com/north_america/en_US/support/)

## 5. Reference Configuration

**Figure 1** illustrates the test configuration used to verify the Plantronics solution. The configuration consists of an Avaya S8300 Server running Avaya Aura® Communication Manager with an Avaya G450 Media Gateway providing connectivity to the PSTN via an ISDN-PRI trunk (not shown). Avaya Aura® Communication Manager Messaging was used as the voicemail system. Avaya Flare® Experience for Windows was installed on a desktop PC and registered to Avaya Aura® Session Manager as a SIP endpoint. Avaya Aura® Session Manager was configured using Avaya Aura® System Manager (not shown). The Plantronics Calisto 620-M Wireless Speakerphone were connected to the desktop PC via a Plantronics Bluetooth USB port.



**Figure 1: Avaya Flare® Experience for Windows with Plantronics Calisto 620-M Wireless Speakerphone**

## 6. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager running on an Avaya S8300 Server with a G450 Media Gateway and Avaya Aura® Communication Manager Messaging	6.2 SP 5 (R016x.02.0.823.0 with Patch 20396)
Avaya Aura® Session Manager	6.2 (6.2.3.0.623006)
Avaya Aura® System Manager	6.2.0 SP 3
Avaya Flare Experience for Windows on Microsoft Windows 7	1.1.1.7
Avaya 9600 Series IP Telephone	3.1 SP 5 (H.323)
Plantronics Calisto 620-M (Microsoft)	USB Firmware 921, USB Adapter 04.70, Headset 02.00

## 7. Configure Avaya Aura® Communication Manager

This section covers the station configuration for Flare Experience. The configuration is performed via the System Access Terminal (SAT) on Communication Manager.

Use the **add station** command to add a station for Flare Experience. Use *9621SIP* for the **Station Type**, provide a descriptive **Name**, and enable **IP Softphone**. Use the default values for the other fields on **Page 1**. The SIP station can also be configured automatically by System Manager as described in **Section 8**.

add station 46010		Page 1 of 6
STATION		
Extension: 46010	Lock Messages? n	BCC: 0
<b>Type: 9621SIP</b>	Security Code:	TN: 1
Port: IP	Coverage Path 1:	COR: 1
<b>Name: Flare, Experience</b>	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19		
	Message Lamp Ext: 46010	
Display Language: english		
Survivable COR: internal		
Survivable Trunk Dest? y	<b>IP SoftPhone? y</b>	
	IP Video Softphone? y	

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 46010) to the same extension configured in System Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

change off-pbx-telephone station-mapping 46010							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
46010	OPS	-		46010	aar	1	

## 8. Configure Avaya Aura® Session Manager

This section describes the procedure for configuring a SIP user for Flare Experience as defined in **Section 7**. Alternatively, use the option to automatically generate the SIP station on Communication Manager when adding a new SIP user. It is assumed that the basic installation and configuration of Session Manager has already been completed.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials.

From the main webpage, navigate to **Users → User Management**. From the User Management webpage, click on **Manage Users** in the left pane, and then click the **New** button to display the **New User Profile** webpage.

Enter values for the following required attributes for a new SIP user in the **Identity** section of the new user form.

- |                               |  |
|-------------------------------|--|
| ▪ <b>Last Name:</b>           | Enter the last name of the user.                                     |
| ▪ <b>First Name:</b>          | Enter the first name of the user.                                    |
| ▪ <b>Login Name:</b>          | Enter <extension>@<sip domain> of the user (e.g., 46010@devcon.com). |
| ▪ <b>Authentication Type:</b> | Select <i>Basic</i> .  |
| ▪ <b>Password:</b>            | Enter the password which will be used to log into System Manager     |
| ▪ <b>Confirm Password:</b>    | Re-enter the password from above.                                    |

The screen below shows the information when adding a new SIP user.

AVAYA

Avaya Aura® System Manager 6.2

Last Logged on at April 5, 2013 6:20 PM

[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

User Management

Manage Users

Public Contacts

Shared Addresses

System Presence ACLs

Home / Users / User Management / Manage Users

User Management

Home

New User Profile

Commit & Continue

Commit

Cancel

Identity

Communication Profile

Membership

Contacts

Identity

\* Last Name:

Experience

\* First Name:

Flare

Middle Name:

Description:

\* Login Name:

46010@devcon.com

\* Authentication Type:

Basic

\* Password:

••••••••

\* Confirm Password:

••••••••

Localized Display Name:

Endpoint Display Name:

Select the **Communication Profile** tab and configure the following fields:

- **Communication Profile Password:** Enter the password which will be used by Flare Experience to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

AVAYA

Avaya Aura® System Manager 6.2

Last Logged on at April 5, 2013 6:20 PM

[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

User Management

Manage Users

Public Contacts

Shared Addresses

System Presence ACLs

Home / Users / User Management / Manage Users

User Management

Home

New User Profile

Commit & Continue

Commit

Cancel

Identity

Communication Profile

Membership

Contacts

Communication Profile

Communication Profile Password:

•••••

Confirm Password:

•••••

New

Delete

Done

Cancel

JAO; Reviewed:  
SPOC 5/20/2013

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Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

**Communication Address**

New Edit Delete

	Type	Handle	Domain
No Records found			

Type: Avaya SIP

\* Fully Qualified Address: 46010 @ devcon.com

Add Cancel

In the *Session Manager Profile* section, specify the Session Manager entity and assign the **Application Sequence** to both the originating and terminating sequence fields. Set the **Home Location** field to the appropriate **Location**.

☒ **Session Manager Profile**

\* Primary Session Manager lz-asm

Primary	Secondary	Maximum
25	0	25

Secondary Session Manager (None)

Primary	Secondary	Maximum

Origination Application Sequence devcon14

Termination Application Sequence devcon14

Conference Factory Set (None)

Survivability Server (None)

\* Home Location Lincroft

In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Endpoint on Unassign of Endpoint From User or on Delete User:** Enable field to automatically delete station when **Station Profile** is un-assigned from user.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Commit** to add the SIP user.


The screenshot shows a web-based configuration form for a 'CM Endpoint Profile'. At the top, there is a section header 'CM Endpoint Profile' with a dropdown arrow. Below this, the form contains several fields and checkboxes:

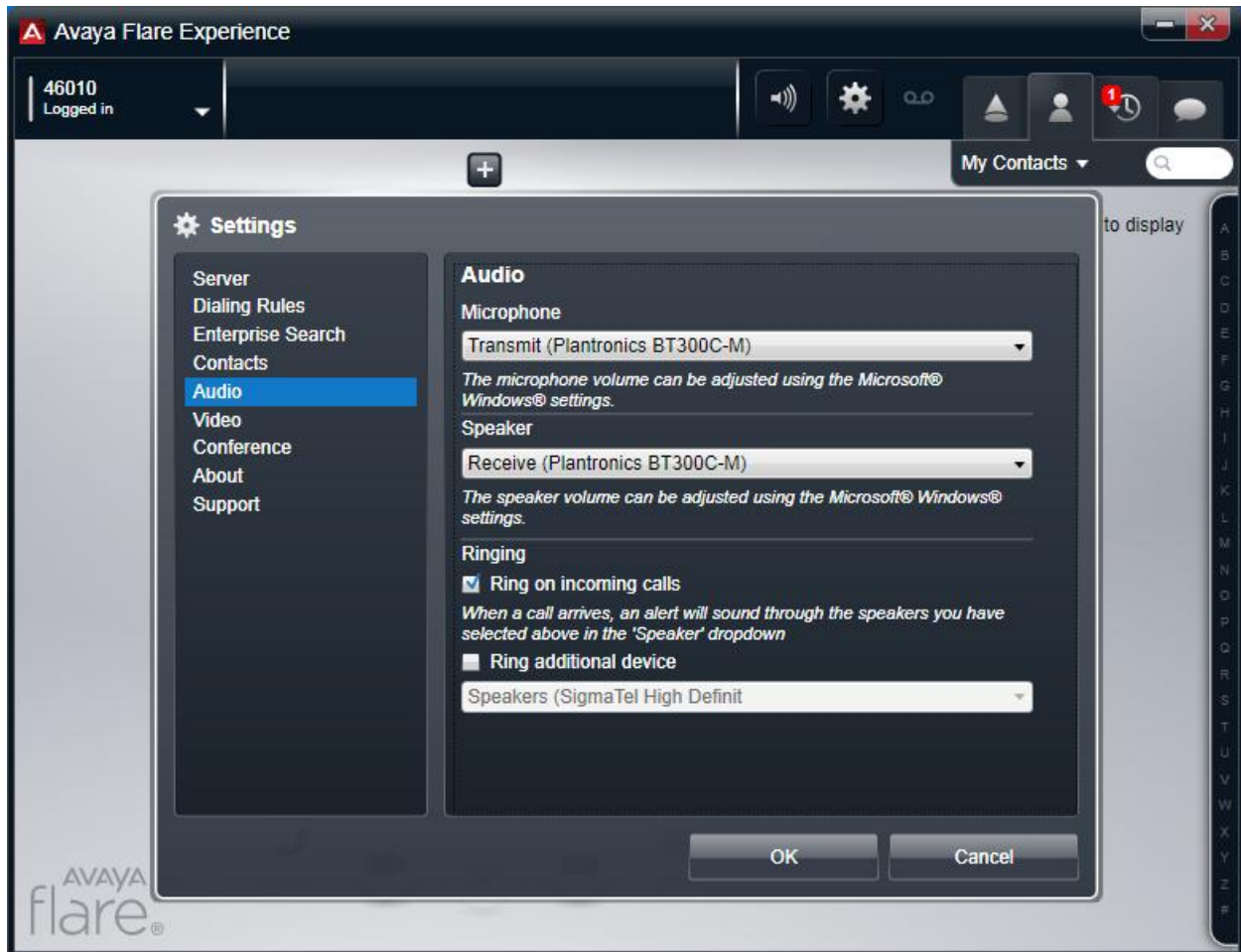
- \* System:** A dropdown menu with 'devcon14' selected.
- \* Profile Type:** A dropdown menu with 'Endpoint' selected.
- Use Existing Endpoints:** An unchecked checkbox.
- \* Extension:** A text input field containing '46010' and a magnifying glass icon. To its right is a button labeled 'Endpoint Editor'.
- \* Template:** A dropdown menu with 'DEFAULT\_9621SIP\_CM\_6\_2' selected.
- Set Type:** A text input field containing '9621SIP'.
- Security Code:** An empty text input field.
- \* Port:** A text input field containing 'IP' and a magnifying glass icon.
- Voice Mail Number:** An empty text input field.
- Preferred Handle:** A dropdown menu with '(None)' selected.
- Delete Endpoint on Unassign of Endpoint from User or on Delete User:** A checked checkbox.
- Override Endpoint Name:** A checked checkbox.

Click the **Endpoint Editor** button, and in the subsequent webpage, navigate to the **Feature Options** tab. Enable **IP Softphone** as shown below. Click **Done** (not shown) to return to the previous webpage, and then click **Commit**.

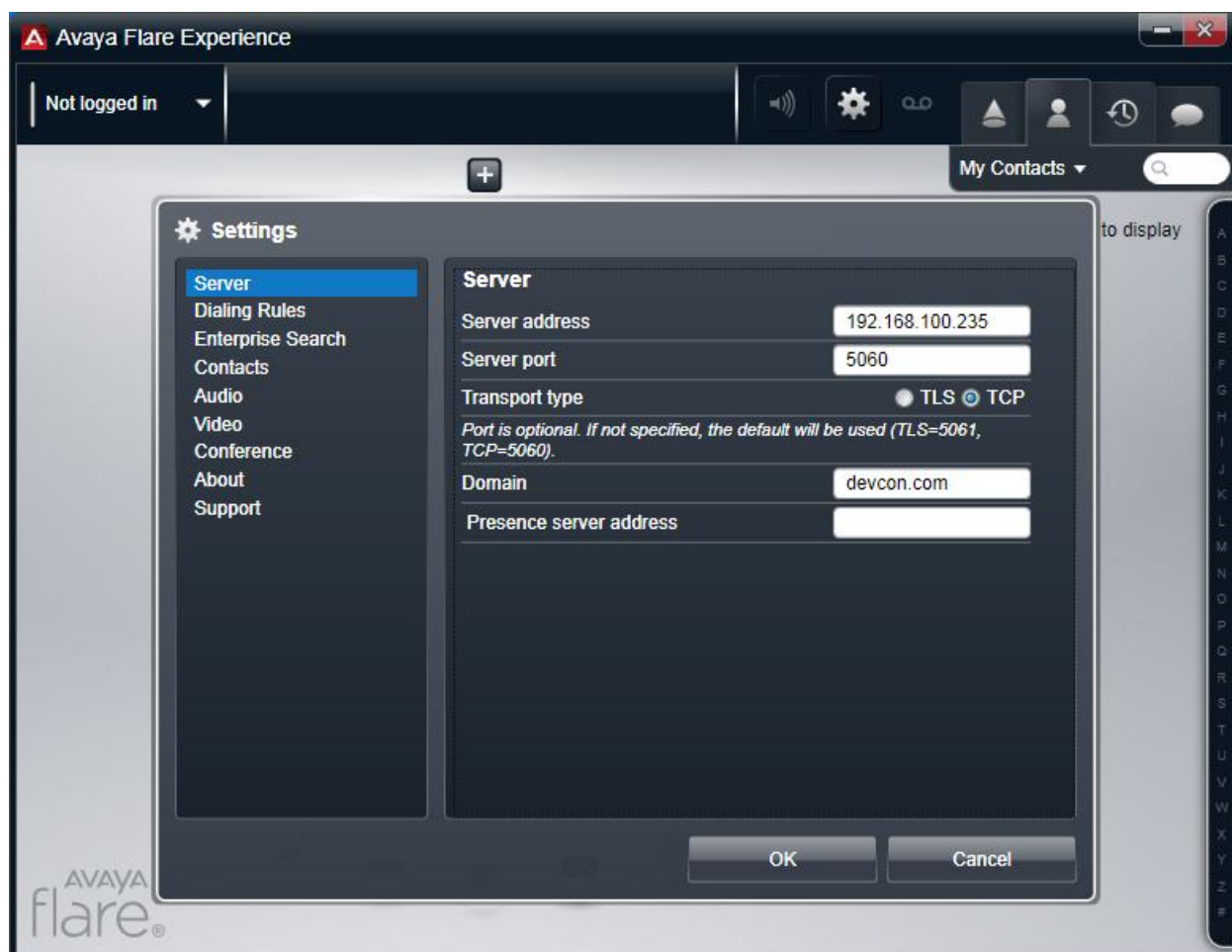
General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)																			
Enhanced Call Fwd (E)		Button Assignment (B)		Group Membership (M)																					
Active Station Ringing	single	Auto Answer	none																						
MWI Served User Type	Select	Coverage After Forwarding	system																						
Per Station CPN - Send Calling Number	Select	Display Language	english																						
IP Phone Group ID		Hunt-to Station																							
Remote Soft Phone Emergency Calls	as-on-local	Loss Group	19																						
LWC Reception	spe	Survivable COR	internal																						
AUDIX Name		Time of Day Lock Table	Select																						
EC500 State	enabled	Voice Mail Number																							
Short/Prefixed Registration Allowed	default																								
<b>Features</b> <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 checked="" type="checkbox"/> IP SoftPhone</td> </tr> <tr> <td><input 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 checked="" type="checkbox"/> IP Video Softphone</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> </table>								<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference	<input type="checkbox"/> IP Audio Hairpinning	<input checked="" type="checkbox"/> IP SoftPhone	<input 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 checked="" type="checkbox"/> IP Video Softphone	<input type="checkbox"/> Bridged Appearance Origination Restriction	<input type="checkbox"/> Per Button Ring Control	<input checked="" type="checkbox"/> Restrict Last Appearance	
<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference																								
<input type="checkbox"/> IP Audio Hairpinning	<input checked="" type="checkbox"/> IP SoftPhone																								
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<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 checked="" type="checkbox"/> IP Video Softphone																								
<input type="checkbox"/> Bridged Appearance Origination Restriction	<input type="checkbox"/> Per Button Ring Control																								
<input checked="" type="checkbox"/> Restrict Last Appearance																									

## 9. Configure Avaya Flare® Experience for Windows

After logging into Flare Experience, click on  and then select the **Audio** settings as shown below. The Plantronics Calisto 620-M wireless speakerphone is automatically detected in Flare Experience. Under **Audio**, set the **Microphone** and **Speaker** fields to *Transmit (Plantronics BT300C-M)* and *Receive (Plantronics BT300C-M)* as shown below. Click **OK**.



For Flare Experience to register successfully with Session Manager, the **Server** settings must be configured with the Session Manager IP address, server port, transport type, and domain name as shown below.



## 10. Connect Plantronics Calisto 620-M speakerphone

Connect the Plantronics Calisto 620 Bluetooth USB adapter to the desktop PC running Flare Experience and turn on the speakerphone. When the speakerphone is paired via Bluetooth, a chime should be heard and the Bluetooth LED on the speakerphone should blink blue once. If the speakerphone needs to be paired again, follow the instructions in [4].

## 11. Verification Steps

This section provides the tests that can be performed to verify proper installation and configuration of the Plantronics Calisto 620-M Wireless Speakerphone with Avaya Flare® Experience.

1. Start the Flare Experience application.
2. Place an incoming call to Flare Experience from any local phone.
3. Answer the call from Flare Experience.
4. Verify two-way talk path between the Calisto 620-M and phone.
5. Verify that the audio can be muted and the volume can be adjusted directly from the speakerphone.
6. Disconnect the call from Flare Experience.
7. Verify that the call is properly disconnected.

## 12. Conclusion

These Application Notes describe the configuration steps required to integrate the Plantronics Calisto 620-M Wireless Speakerphone with Avaya Flare® Experience. All test cases were completed successfully with observations noted in **Section 3**.

## 13. Additional References

This section references the Avaya and Plantronics documentation that are relevant to these Application Notes.

The following Avaya product documentation can be found at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7, December 2012, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, July 2012, Issue 3, Release 6.2, Document Number 03-603324.
- [3] *Administering Avaya Flare® Experience for Windows*, Release 1.1, Issue 2, February 2013, Document Number 18-604156.

The following Plantronics product documentation can be found at <http://www.plantronics.com>.

*Plantronics Calisto 620 Bluetooth Wireless Speakerphone + Bluetooth USB Adapter Quick Start Guide.*

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