



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

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# **Application Notes for PhoneTech P20USBF Headset with Avaya Flare® Experience for Windows - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required to integrate the PhoneTech P20USBF Headset with Avaya Flare® Experience for Windows. The P20USBF headset provides two-way audio. This solution does not provide call control features directly from the headset, such as answering or terminating a call from the headset.

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 PhoneTech P20USBF Headset with Avaya Flare® Experience for Windows. The P20USBF headset provides two-way audio. This solution does not provide call control features directly from the headset, such as answering or terminating a call from the headset.

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

**Note:** This solution does not provide call control integration with Flare Experience. That is, a call cannot be answered or terminated directly from the headset, nor is the mute status on the headset synchronized with Flare Experience.

# 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 P20USBF headset and verifying two-way audio path. 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 P20USBF headset after restarting Flare Experience, disconnecting and reconnecting the headset, 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.

For the serviceability testing, the headsets were 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 headsets were operational when the PC came back into service.

## 3. Test Results

All test cases passed with the following observations:

- There is no call control support through the headset. Calls need to be answered and terminated through Flare Experience.

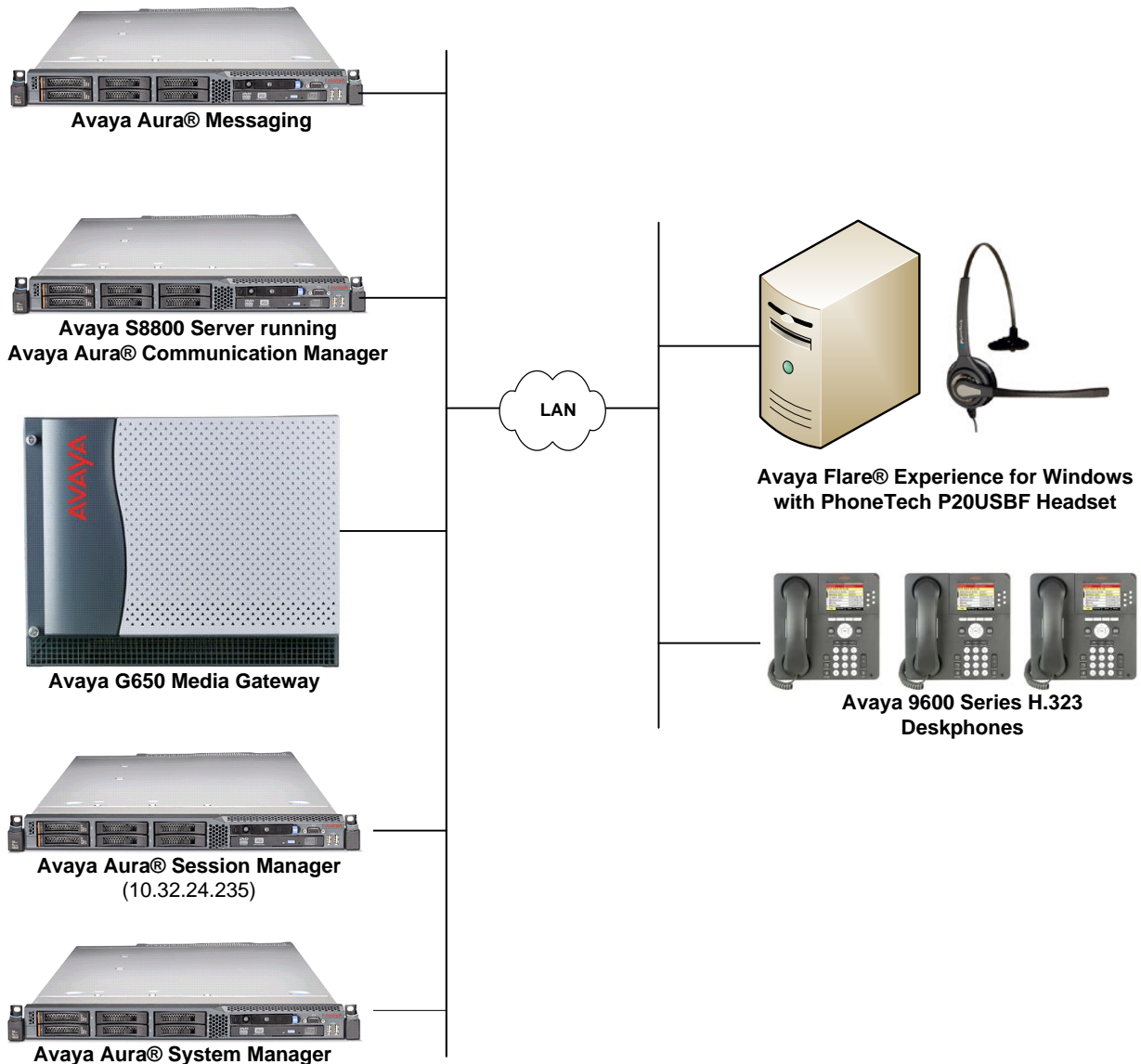
## 4. Support

For technical support and information on PhoneTech P20USBF Headset, contact PhoneTech in Brazil at:

- Phone: 11-3717-1881
- Website: <http://www.phonetech.com.br>
- Email: [contato@phonetech.com.br](mailto:contato@phonetech.com.br)

## 5. Reference Configuration

**Figure 1** illustrates the test configuration used to verify the PhoneTech solution. The configuration consists of an Avaya S8800 Server running Avaya Aura® Communication Manager with an Avaya G650 Media Gateway providing connectivity to the PSTN via an ISDN-PRI trunk (not shown). Avaya Aura® 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. The PhoneTech P20USBF Headset was connected to the desktop PC via a USB port.



**Figure 1: Avaya Flare® Experience for Windows with PhoneTech P20USBF Headset**

## 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 S8800 Server with a G650 Media Gateway and Communication Manager Messaging	6.3 SP 4 (R016x.03.0.124.0 w/Patch 21291)
Avaya Aura® Session Manager	6.3 (6.3.5.0.635005)
Avaya Aura® System Manager	6.3.5 Build No. – 6.3.0.8.5682-6.3.8.2826 Software Update Revision No: 6.3.5.5.2017
Avaya Flare Experience for Windows on Microsoft Windows 7	1.1.4.23
Avaya 9600 Series IP Telephone	S3.210A (H.323)
PhoneTech P20USBF Headset	N/A

## 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.

The SIP station was configured automatically by System Manager as described in **Section 8**. This section shows the station configuration in Communication Manager for reference only. The **display station** command below shows the station for Flare Experience. The **Station Type** was configured as *9620SIP*, a descriptive **Name** was provided, and **IP Softphone** was enabled. Default values for the other fields on **Page 1** were used.

display station 78010		Page	1 of	6
STATION				
Extension: 78010	Lock Messages? n	BCC: 0		
<b>Type: 9620SIP</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: 78010		
Display Language: english				
Survivable COR: internal				
Survivable Trunk Dest? y		<b>IP SoftPhone? y</b>		
		IP Video? y		

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 78010) 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									
<b>Station</b>	<b>Application</b>	<b>Dial</b>	<b>CC</b>	<b>Phone Number</b>	<b>Trunk</b>	<b>Config</b>	<b>Dual</b>		
<b>Extension</b>		<b>Prefix</b>			<b>Selection</b>	<b>Set</b>	<b>Mode</b>		
46010	OPS	-		78010	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 Avaya Aura® 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., 78010@avaya.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  
Aura® System Manager 6.3

Last Logged on at April 14, 2014 11:20 AM  
Help | About | Change Password | Log off admin

Home User Management x

Home / Users / User Management / Manage Users

User Profile Edit: 78010@avaya.com

Commit & Continue Commit Cancel

Identity \* Communication Profile Membership Contacts

User Provisioning Rule

User Provisioning Rule: [v]

Identity

\* Last Name: Experience

Last Name (Latin Translation): Experience

\* First Name: Flare

First Name (Latin Translation): Flare

Middle Name:

Description:

Update Time: June 7, 2013 10:13:34

\* Login Name: 78010@avaya.com

\* Authentication Type: Basic

[Change Password](#)

Source: local

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  
Aura® System Manager 6.3

Last Logged on at April 14, 2014 11:20 AM  
Help | About | Change Password | Log off admin

Home User Management x

Home / Users / User Management / Manage Users

User Profile Edit: 78010@avaya.com

Commit & Continue Commit Cancel

Identity \* Communication Profile Membership Contacts

Communication Profile

Communication Profile Password: ..... [Edit](#)

New Delete Done Cancel

Name
Primary

Select: None

\* Name: Primary

Default: ☒



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 ▼

<input type="button" value="New"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/>			
<input type="checkbox"/>	Type	Handle	Domain
<input type="checkbox"/>	Avaya SIP	78010	avaya.com
Select : All, None			

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 ▼

##### SIP Registration

\* Primary Session Manager

devcon-asm ▼

Primary	Secondary	Maximum
34	0	34

Secondary Session Manager

(None) ▼

Survivability Server

(None) ▼

Max. Simultaneous Devices

1 ▼

Block New Registration  
When Maximum Registrations  
Active?

☐

##### Application Sequences

Origination Sequence

DEVCON13 App Sequence ▼

Termination Sequence

DEVCON13 App Sequence ▼

##### Call Routing Settings

\* Home Location

BR-DevConnect ▼

Conference Factory Set

(None) ▼

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.

---

☒ **CM Endpoint Profile** ▼

\* System

devcon13-CM-ES ▼

\* Profile Type

Endpoint ▼

Use Existing Endpoints

☐

\* Extension

78010

Endpoint Editor

Template

Select/Reset ▼

Set Type

9620SIP

Security Code

Port

S00003

Voice Mail Number

Preferred Handle

(None) ▼

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


☒

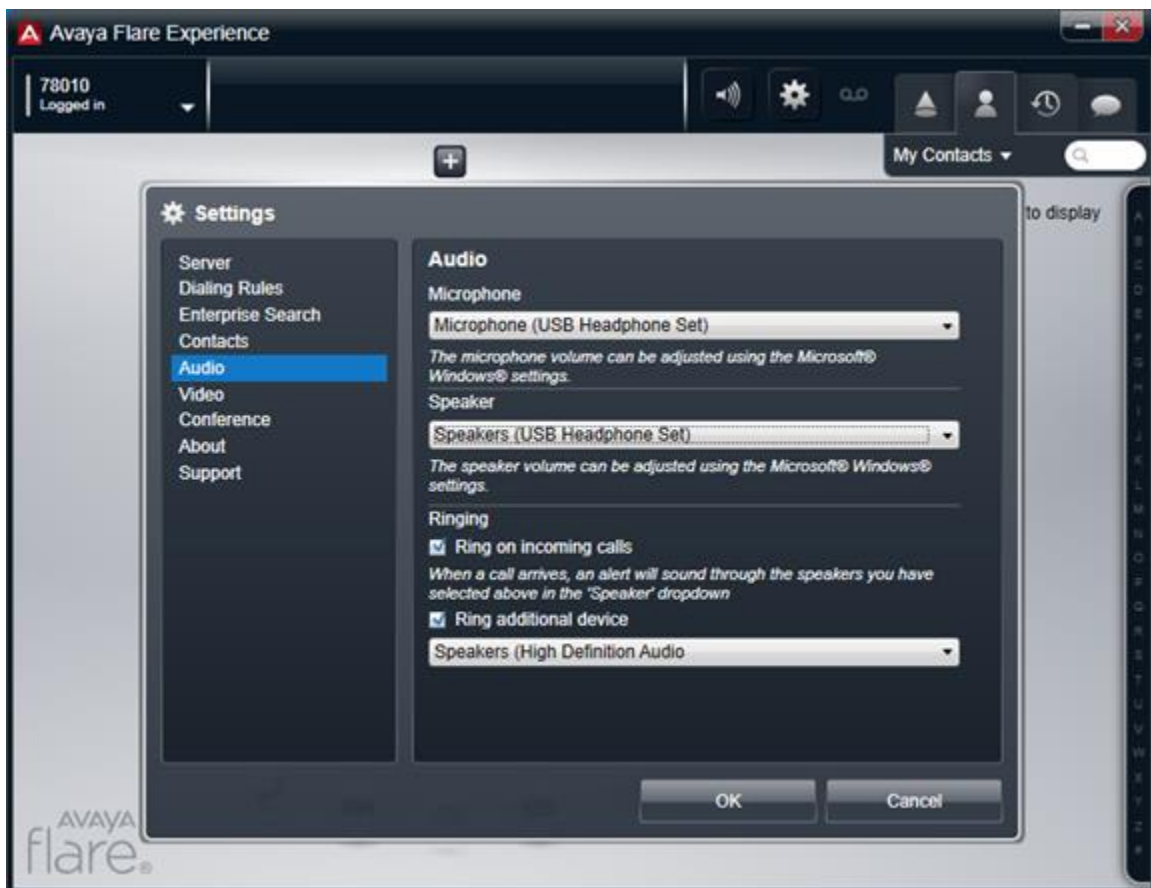
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Click the **Endpoint Editor** button, and in the subsequent webpage, navigate to the **Feature Options** tab. Enable **IP Softphone** as shown below. Click **Done** (not shown) to return to the previous webpage, and then click **Commit**.

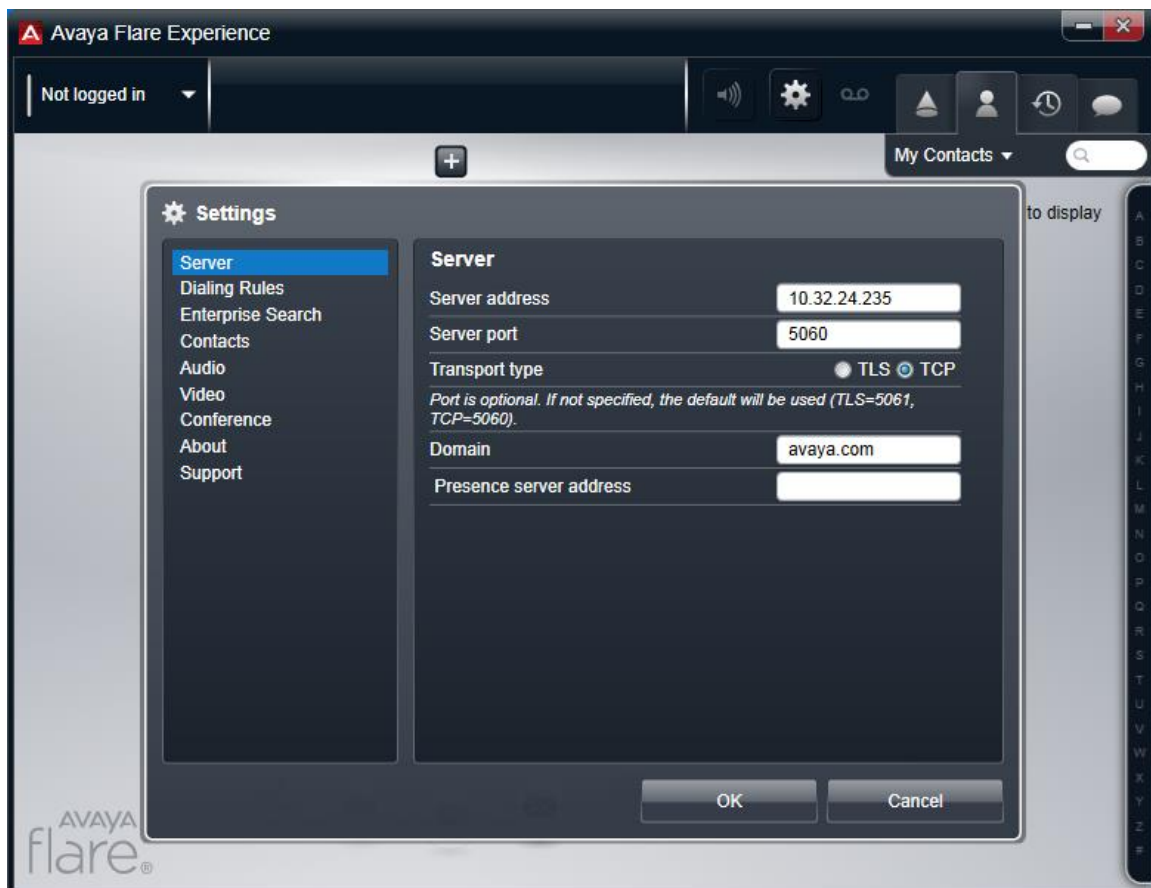
General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)		Enhanced Call Fwd (E)																					
Button Assignment (B)		Group Membership (M)																											
Active Station Ringing	single	Auto Answer	none																										
MWI Served User Type	sip-adjunct	Coverage After Forwarding	system																										
Per Station CPN - Send Calling Number	None	Display Language	english																										
AUDIX Name	None	Hunt-to Station																											
Remote Soft Phone Emergency Calls		Loss Group	19																										
LWC Reception	spe	Survivable COR	internal																										
IP Phone Group ID		Time of Day Lock Table	None																										
Speakerphone		Voice Mail Number																											
Short/Prefixed Registration Allowed																													
EC500 State	enabled																												
<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> <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 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"/> Turn on mute for remote off-hook attempt	
<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference																												
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<input checked="" type="checkbox"/> Restrict Last Appearance																													
<input type="checkbox"/> Turn on mute for remote off-hook attempt																													

## 9. Configure Avaya Flare® Experience for Windows

After logging into Flare Experience, click on  and then select the **Audio** settings as shown below. The PhoneTech P20USBF Headset is automatically detected in Flare Experience. Under **Audio**, set the **Microphone** and **Speaker** fields to the appropriate device as shown below. The example below is configured for the P20USBF headset. 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 PhoneTech P20USBF Headset

Simply connect the P20USBF headset to a USB port on the PC running Flare Experience. Flare Experience will automatically detect it. Refer to [4] for instructions on using the PhoneTech headset.

## 11. Verification Steps

This section provides the tests that can be performed to verify proper installation and configuration of the PhoneTech P20USBF Headset 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 headset and phone.
5. Disconnect the call from Flare Experience.
6. Verify that the call is properly disconnected.

## 12. Conclusion

These Application Notes describe the configuration steps required to integrate the PhoneTech P20USBF Headset 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 PhoneTech 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 6.3, Issue 9, October 2013, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 3, October 2013.
- [3] *Administering Avaya Flare® Experience for Windows*, Release 1.1, Issue 2, February 2013, Document Number 18-604156.

The following PhoneTech product documentation is available with the headset.

- [4] *Manual do Usuário P20USBF*.

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