Avaya Solution & Interoperability Test Lab

Application Notes for PhoneTech P20USBM Headset with Avaya Flare® Experience for Windows - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the PhoneTech P20USBM Headset with Avaya Flare® Experience for Windows. The P20USBM headset provides two-way audio, allows the audio volume to be adjusted and the audio to be muted/unmuted directly from the headset. 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 P20USBM Headset with Avaya Flare® Experience for Windows. The P20USBM headset provides two-way audio, allows the audio volume to be adjusted and the audio to be muted/unmuted directly from the headset. 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 11** 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 P20USBM 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 P20USBM 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.
- Using the volume control buttons on the headset and Flare® Experience to adjust the playback volume.
- Using the mute control button on the headset and Flare® Experience to mute and un-mute the audio.

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.

2.2. Test Results

All test cases passed with the following observations:

- There is no mute synchronization between the P20USBM headset and Flare® Experience. If a call is muted through the headset, it is not reflected on Flare® Experience, and the call needs to be un-muted through the headset, and vice versa.
- There is no call control support through the headset. Calls need to be answered and terminated through Flare® Experience.

2.3. Support

For technical support and information on PhoneTech P20USBM Headset, contact PhoneTech in Brazil at:

- Phone: 11-3717-1881
- Website: http://www.phonetech.com.br
- Email: contato@phonetech.com.br

3. 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 P20USBM Headset was connected to the desktop PC via a USB port.

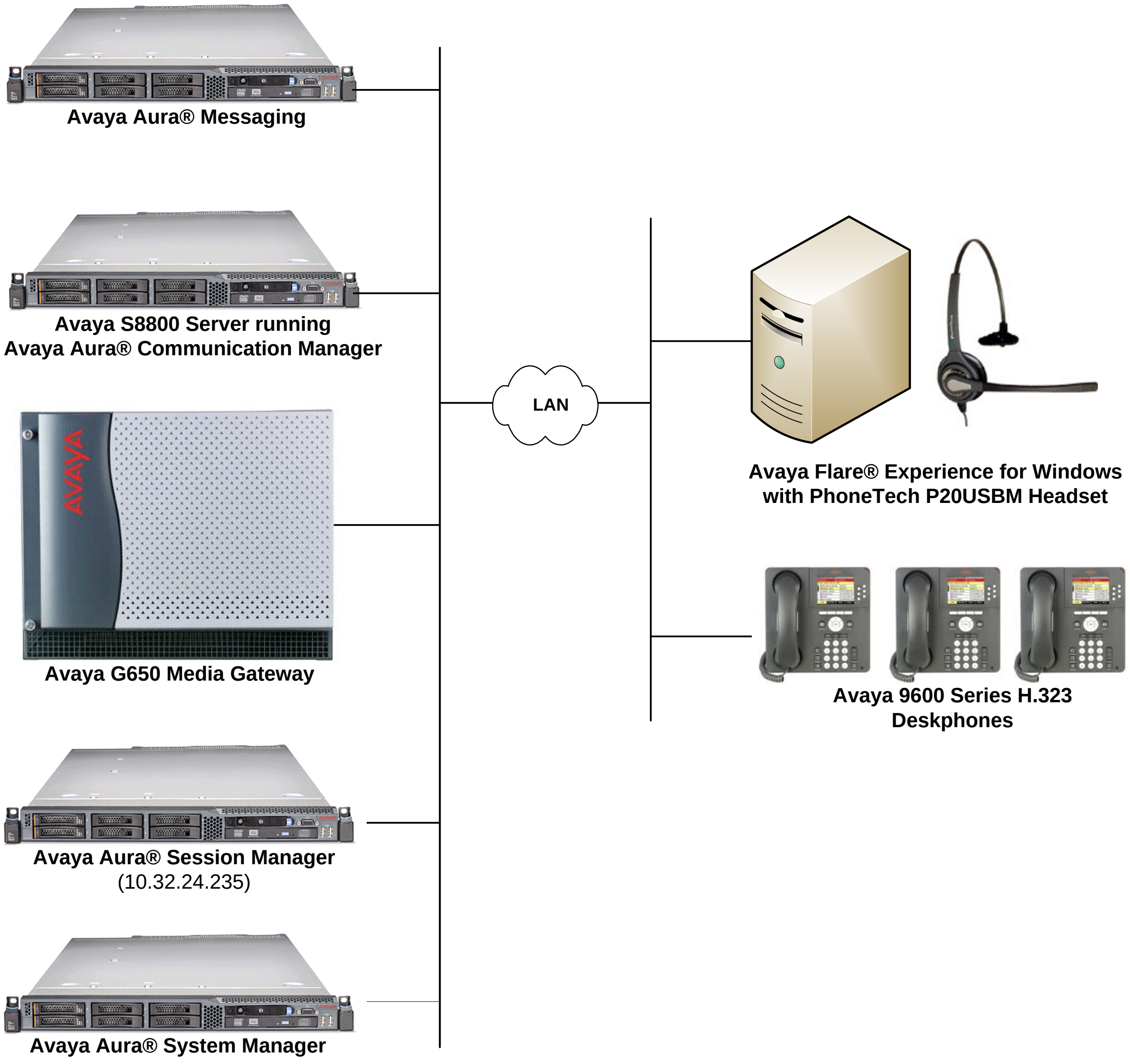

Avaya Aura® Messaging
Avaya S8800 Server running
Avaya Aura® Communication Manager
LAN
Avaya Flare® Experience for Windows
with PhoneTech P20USBM Headset
Avaya G650 Media Gateway
Avaya 9600 Series H.323
Deskphones
Avaya Aura® Session Manager
(10.32.24.235)
Avaya Aura® System Manager

Figure 1: Avaya Flare® Experience for Windows with PhoneTech P20USBM Headset

4. 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 P20USBM Headset	N/A

5. 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 6**. 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 configures 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
     Type: 9620SIP                       Security Code:                 TN: 1
     Port: IP                          Coverage Path 1:                COR: 1
     Name: Flare, Experience           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                              IP SoftPhone? y

                                                             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

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 46010            OPS          -     78010             aar        1
```

6. Configure Avaya Aura® Session Manager

This section describes the procedure for configuring a SIP user for Flare® Experience as defined in **Section 5**. 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.

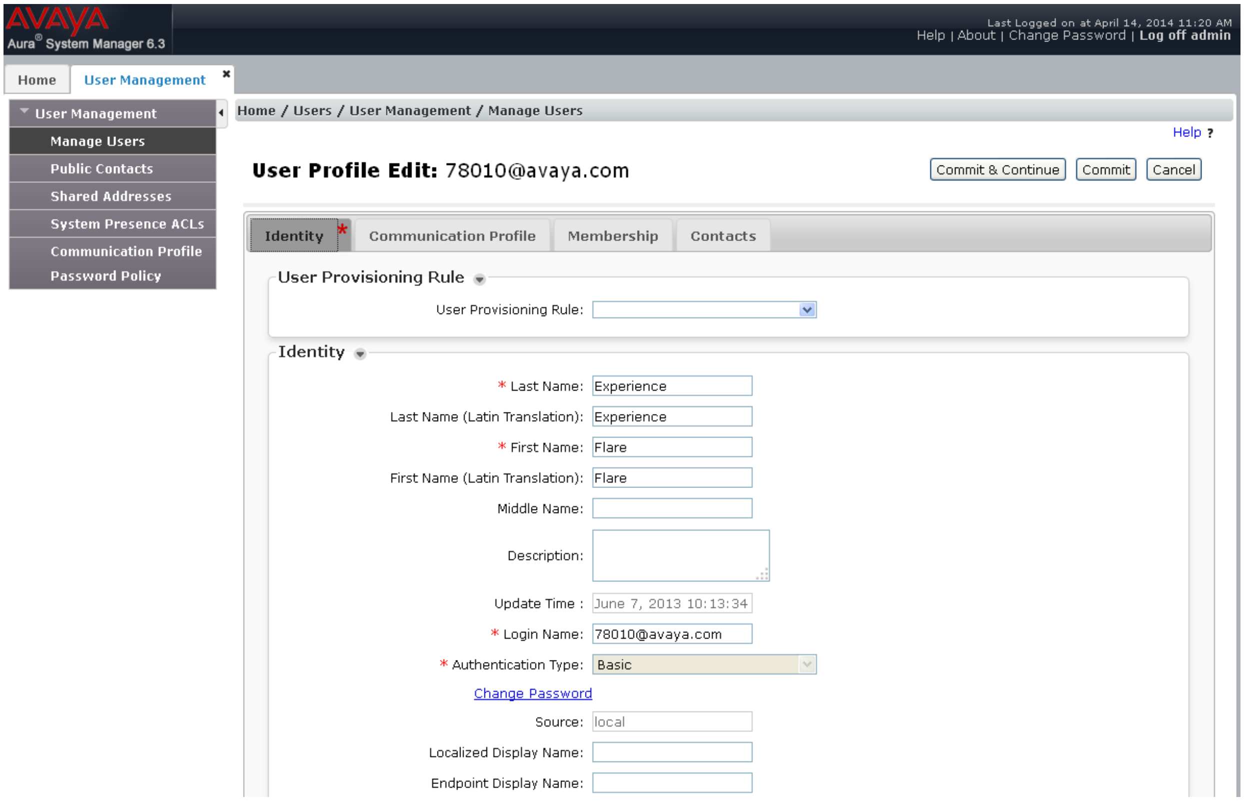
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

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.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter <*extension*>@<*sip domain*> of the user (e.g., *78010@avaya.com*).
- **Authentication Type:** Select *Basic*.
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

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



User Profile Edit: 78010@avaya.com

User Provisioning Rule

User Provisioning Rule:

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

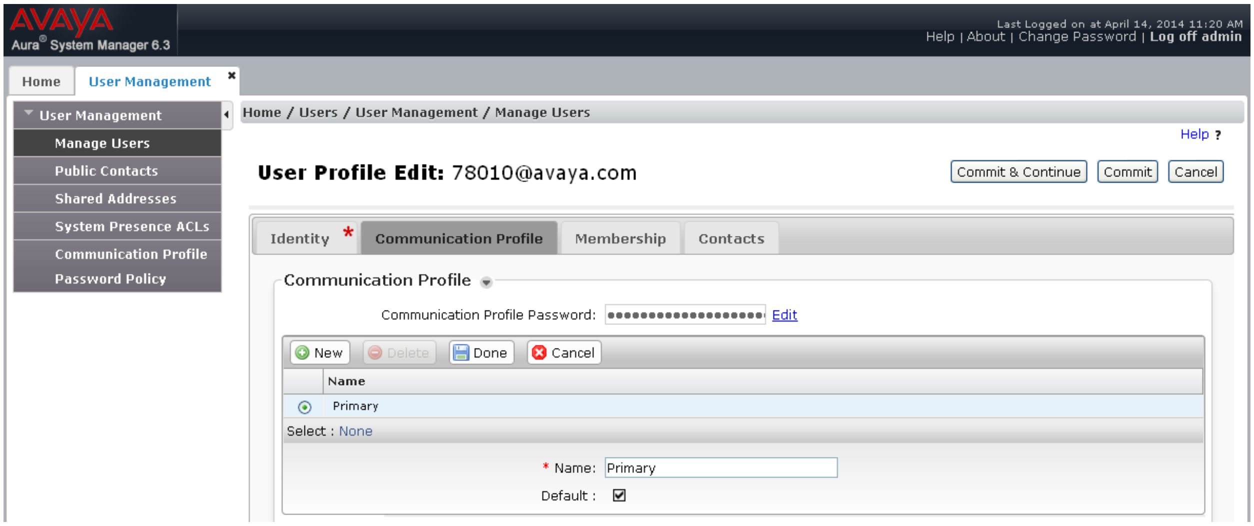
Source: local

Localized Display Name:

Endpoint Display Name:

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

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



User Profile Edit: 78010@avaya.com

Communication Profile

Communication Profile Password: ●●●●●●●●●●●●●●●●●●●● Edit

	Name
●	Primary

Select : None

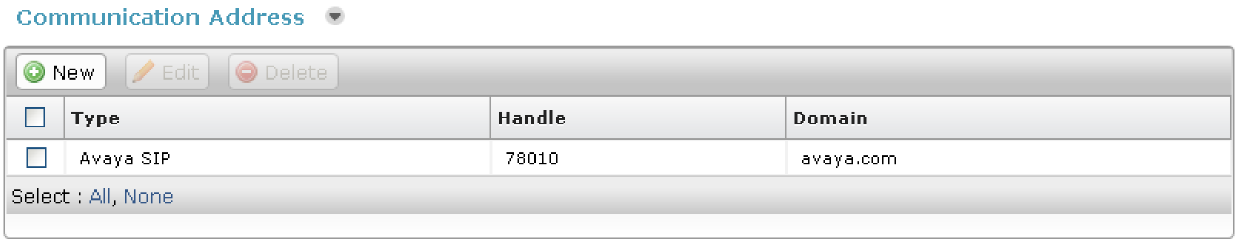
* Name: Primary

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**.

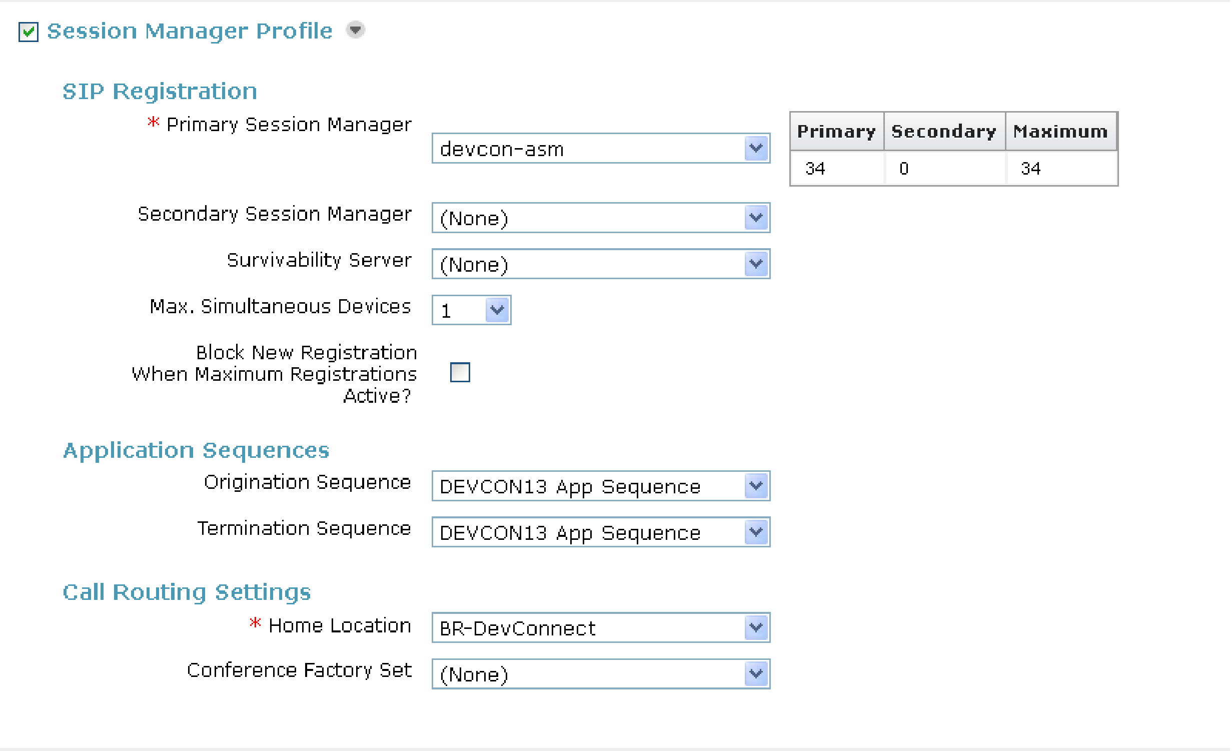


Communication Address

□	Type	Handle	Domain
□	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
Secondary Session Manager	(None)
Survivability Server	(None)
Max. Simultaneous Devices	1
Block New Registration When Maximum Registrations Active?	□

Primary	Secondary	Maximum
34	0	34

Application Sequences

Origination Sequence	DEVCON13 App Sequence
Termination Sequence	DEVCON13 App Sequence

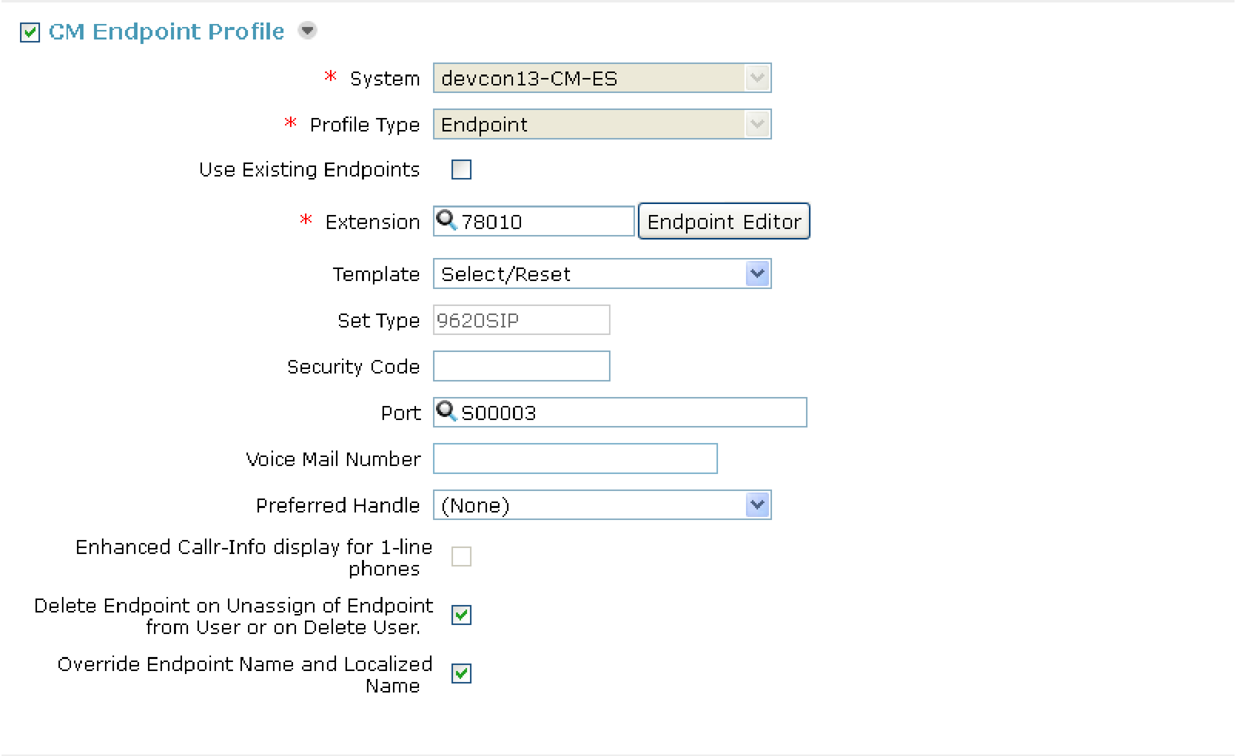
Call Routing Settings

* 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.

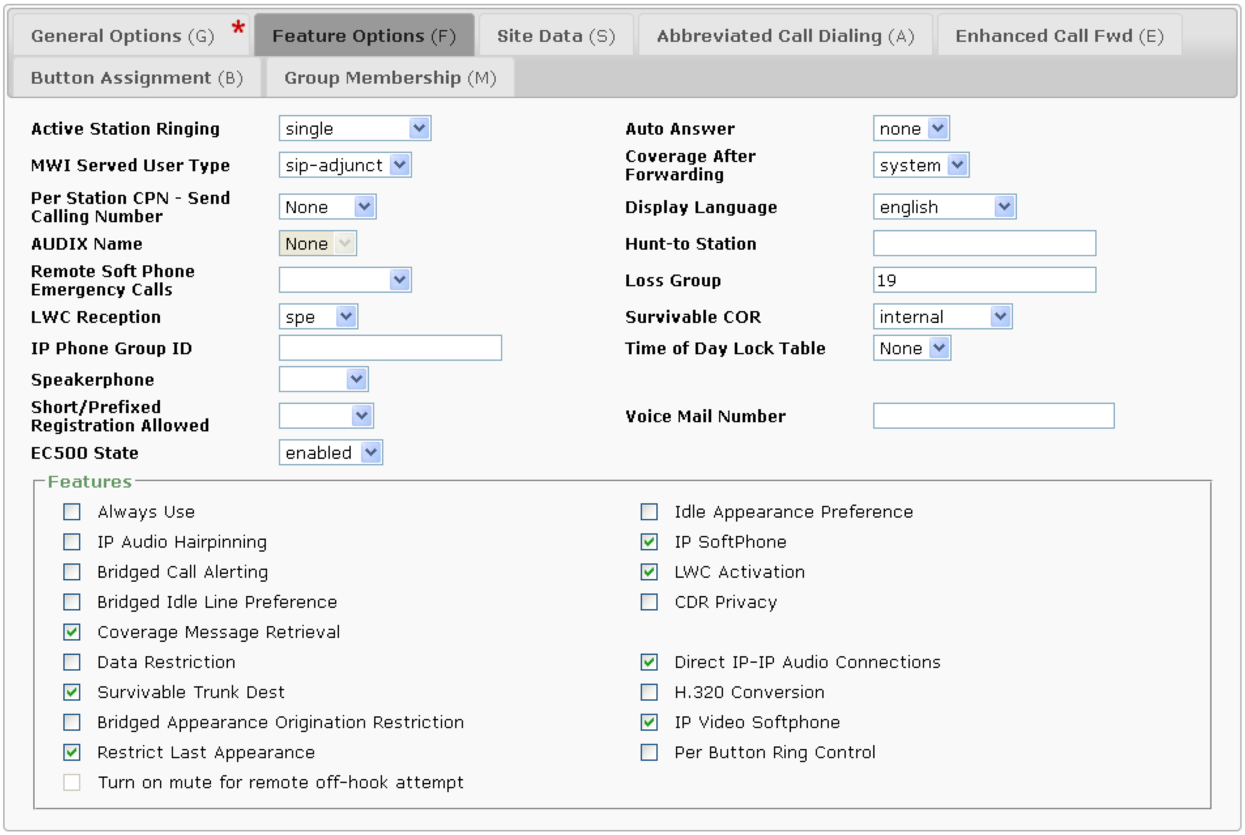
The screen below shows the information when adding a new SIP user to the sample configuration. Click **Commit** to add the SIP user (not shown).



☑ CM Endpoint Profile

* System	devcon13-CM-ES
* Profile Type	Endpoint
Use Existing Endpoints	□
* Extension	78010
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	☑

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**.



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			
Short/Prefixed Registration Allowed		**Voice Mail Number**	
EC500 State	enabled		

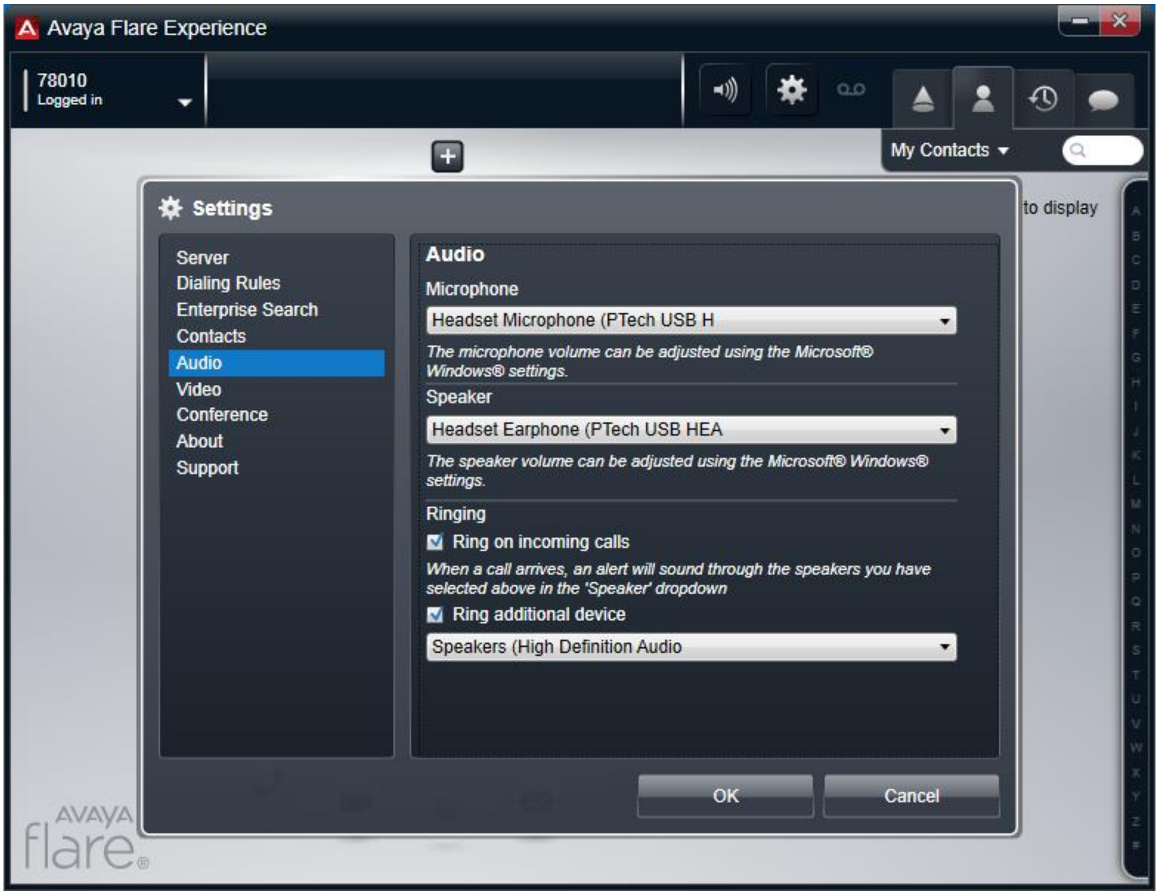
Features

☐ Always Use
☐ IP Audio Hairpinning
☐ Bridged Call Alerting
☐ Bridged Idle Line Preference
☑ Coverage Message Retrieval
☐ Data Restriction
☑ Survivable Trunk Dest
☐ Bridged Appearance Origination Restriction
☑ Restrict Last Appearance
☐ Turn on mute for remote off-hook attempt

☐ Idle Appearance Preference
☑ IP SoftPhone
☑ LWC Activation
☐ CDR Privacy
☑ Direct IP-IP Audio Connections
☐ H.320 Conversion
☑ IP Video Softphone
☐ Per Button Ring Control

7. Configure Avaya Flare® Experience for Windows

After logging into Flare® Experience, click on ⚙ and then select the **Audio** settings as shown below. The PhoneTech P20USBM 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 P20USBM headset. Click **OK**.



Audio

Microphone

Headset Microphone (PTech USB H

The microphone volume can be adjusted using the Microsoft® Windows® settings.

Speaker

Headset Earphone (PTech USB HEA

The speaker volume can be adjusted using the Microsoft® Windows® settings.

Ringing

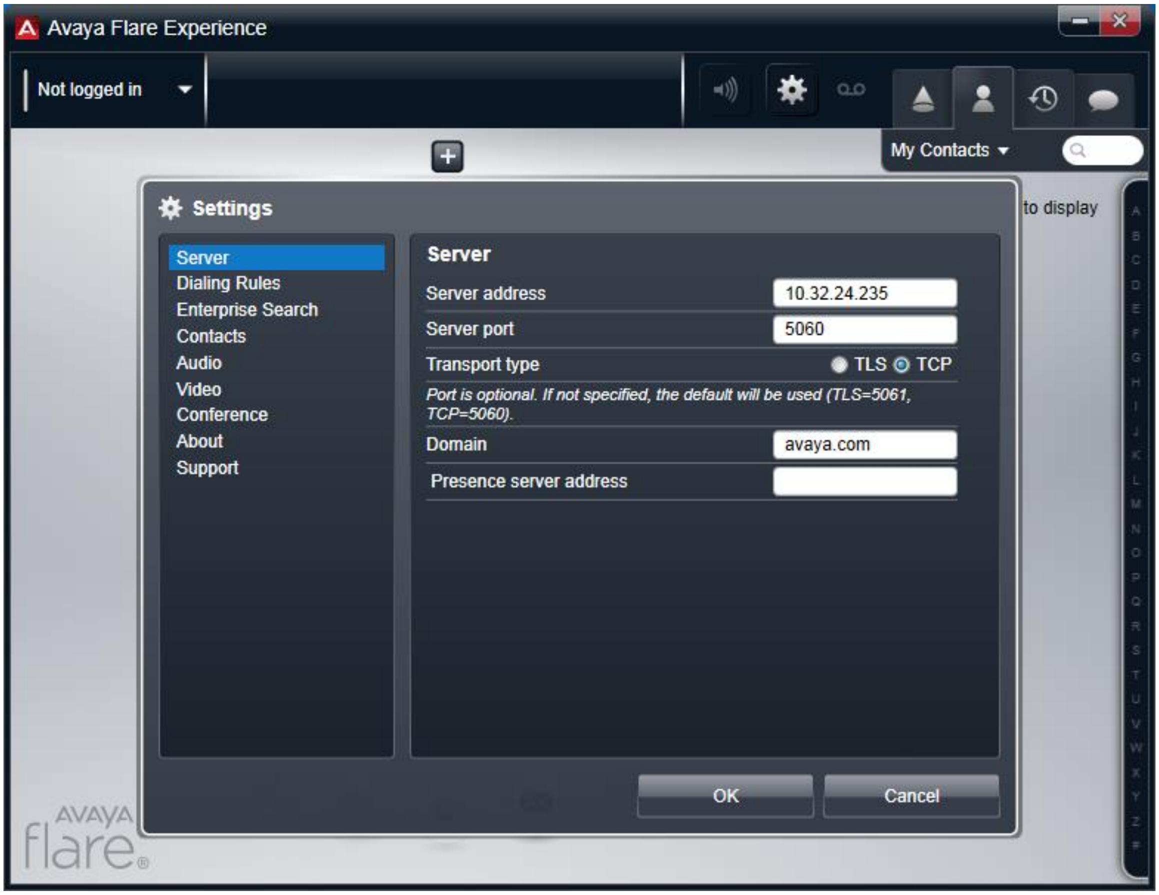
☑ Ring on incoming calls

When a call arrives, an alert will sound through the speakers you have selected above in the 'Speaker' dropdown

☑ Ring additional device

Speakers (High Definition Audio

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.



Server

Server address: 10.32.24.235

Server port: 5060

Transport type: ○ TLS ● TCP

Port is optional. If not specified, the default will be used (TLS=5061, TCP=5060).

Domain: avaya.com

Presence server address:

8. Connect PhoneTech P20USBM Headset

Simply connect the P20USBM headset to a USB port on the PC running Flare® Experience. Flare® Experience will automatically detect it. Refer to [4] in **Section 11** for instructions on using the PhoneTech headset.

9. Verification Steps

This section provides the tests that can be performed to verify proper installation and configuration of the PhoneTech P20USBM 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. Verify that the audio can be muted and the volume can be adjusted directly from the headset.
6. Disconnect the call from Flare® Experience.
7. Verify that the call is properly disconnected.

10. Conclusion

These Application Notes describe the configuration steps required to integrate the PhoneTech P20USBM Headset with Avaya Flare® Experience. All test cases were completed successfully with observations noted in **Section 2.2**.

11. 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 P20USB.*

The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.