Avaya Solution & Interoperability Test Lab

Configuring the Avaya B179 SIP Conference Phone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the steps to configure the Avaya B179 SIP Conference Phone to work with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Introduction

These Application Notes describe a basic configuration of the Avaya B179 SIP Conference Phone to work with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Figure 1 shows the configuration used for testing.

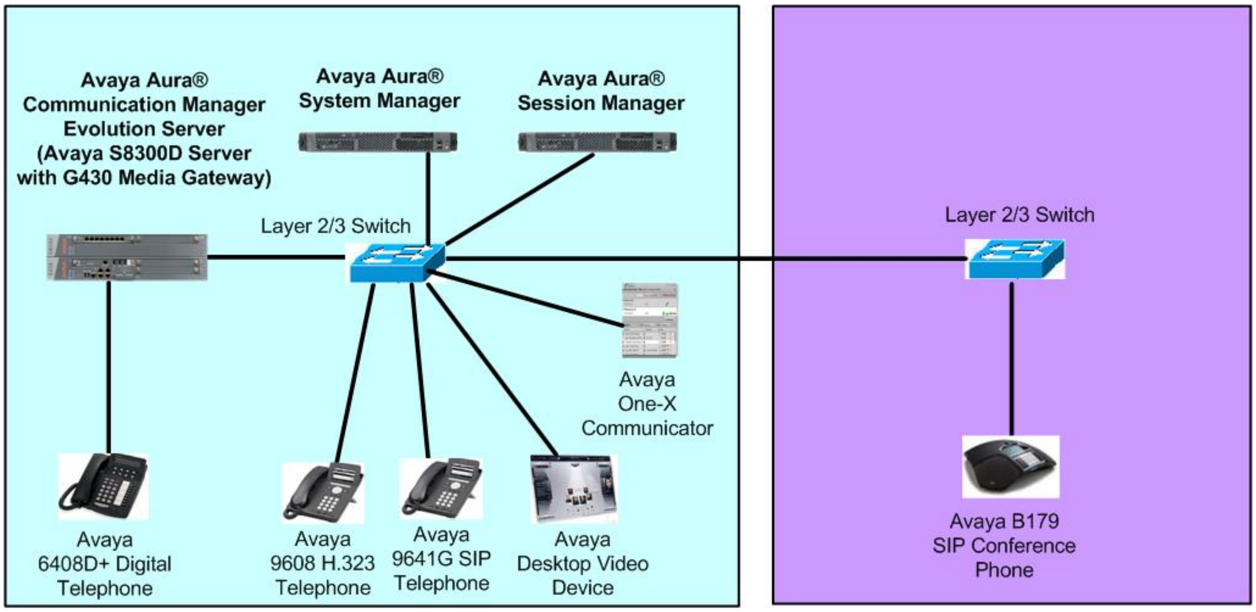


Figure 1: Network Diagram

2. Equipment and Software Validated

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

Hardware/Software Component	Software Version
Communication Manager (S8300D Server with G430 Media Gateway)	R016x.00.1.510.1
Session Manager (S8800 Server)	6.1.0.0.610023
System Manager (S8800 Server)	6.1.0 (6.1.0.4.5072-6.1.4.62)
9640G IP Telephone (SIP)	2.6.4 (SIP96xx_2_6_4_0.bin)
9641G IP Telephone (SIP)	6.0 (96x1_SALBR6_0_V4r52.tar)
9608 IP Telephone (H.323)	6.008 (S9608_11_HALBR6_0_0_8S_V4r52.tar)
6408+ Digital Telephone	N/A
Desktop Video Device	1.0.0 (1_0_0_012849)
one-X® Communicator (SIP)	6.0.1.16-SP1-25226
one-X® Communicator (H.323)	6.0.1.16-SP1-25226
300 IP SIP Telephone	2.2 & 2.2.1

3. Configuration of the Avaya B179 SIP Conference Telephone

As shown in **Figure 2** below, use the buttons on the B179 SIP Conference Phone to config the IP address settings. The B179 can be set with a static IP address or use DHCP. In this case a static IP Address was entered, and VLAN 7 was used.

On the Phone select MENU then use the ▲and▼keys to get to the SETTINGS menu, then Click OK (OK is acts as the select key). In the SETTINGS menu (not shown) select ADVANCED, and enter the required password. Navigate to the NETWORK menu, and then select IP. Set the IP Address to either DHCP or STATIC. A dark box next to an entry indicates it is selected. If static, click OK and step through the prompts to enter the IP Address, Mask, and Gateway for the endpoint, clicking OK after each entry. At the end the NETWORK menu re-appears. If a VLAN is to be used, select ETHERNET, followed by VLAN. Select ENABLE VLAN, enter the VLAN number to be used, followed by OK. When done continue to press C until you reach the main menu, which will cause the phone to reboot. At this point the remaining configuration can be done via the web interface.

Figure 2: Menu Navigation Buttons

Using an HTTP browser, enter the IP address of the B179 SIP Conference Phone in the browser URL field. The Login screen will appear as shown in **Figure 3**. Select Admin for the login and enter the appropriate password, then click Login.

Figure 3: Login Screen

Once logged in, the initial display screen appears as shown in **Figure 4**.

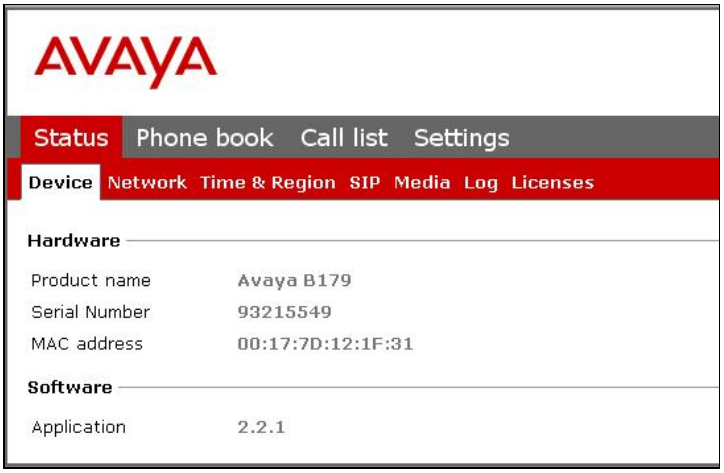


Figure 4: Initial Screen

Select **Settings** and then **Network**. **Figure 5** shows the configuration used in this test. The network section shows the IP Address information previously entered. The Quality of Service section shows the VLAN assigned. In this section the user may also assign QoS tagging if the network the phone is connected to supports QoS.

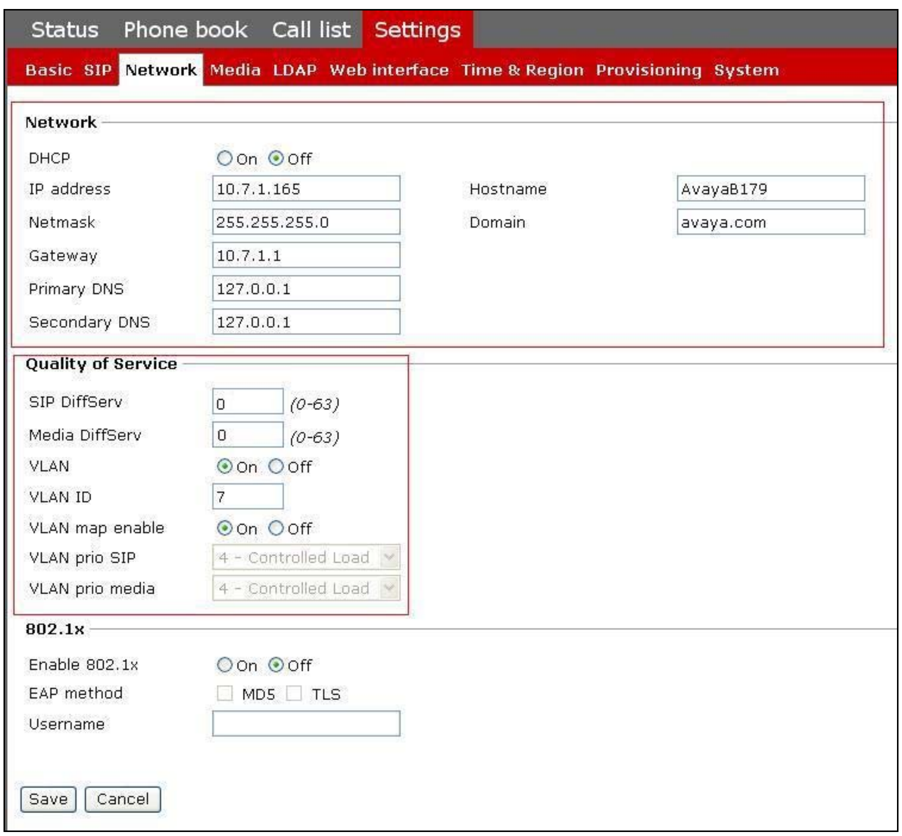


Figure 5: Network Settings Screen

Select the **Media** tab. **Figure 6** shows the configuration used in this test. The **Codec** section selects the priority for each codec supported. If a codec is not to be used the priority should be set to 0. If the G.729 codec is to be used, **VAD** should be Yes. For full feature functionality, it's required that the G.711 codec be included, even if it's the lowest priority on the list.

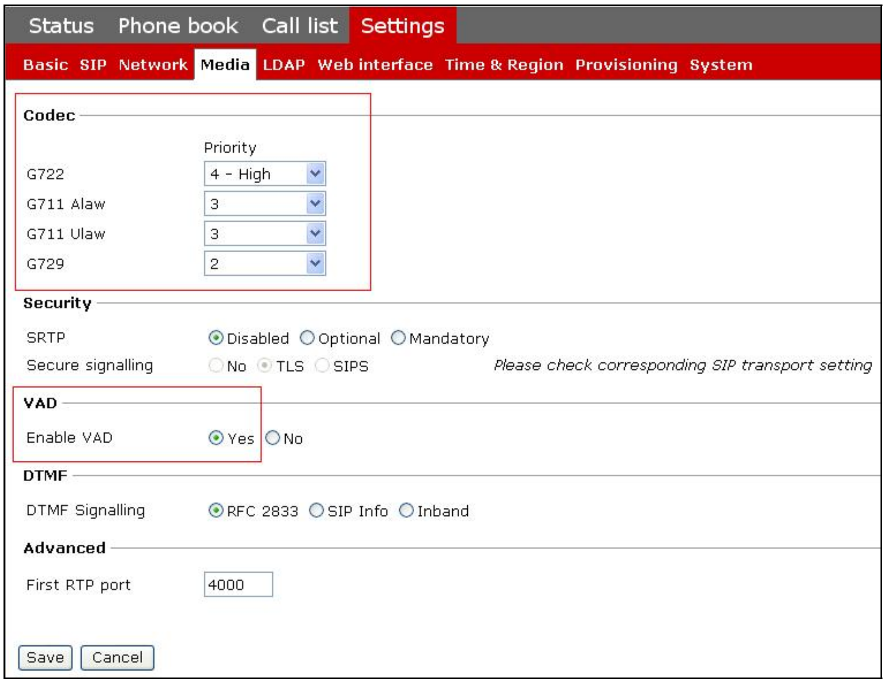


Figure 6: Media Settings Screen

Select the **SIP** tab. **Figure 7** shows the configuration used in this test.

Account 1:
- **Enable account**: Yes
- **Account Name**: Use a meaningful name.
- **User**: Enter the extension assigned in Session Manager, in this case 36052.
- **Realm:** * (the default)
- **Authentication Name and Password**: Enter the extension and password assigned in Session Manager.
- **Registrar and Proxy**: Enter the IP Address for Session Manager, in this case 10.1.2.210.
- **Registration interval**: Use the default 300

Tranport:
- **Protocol**: TCP
- **Local Port:** 5060

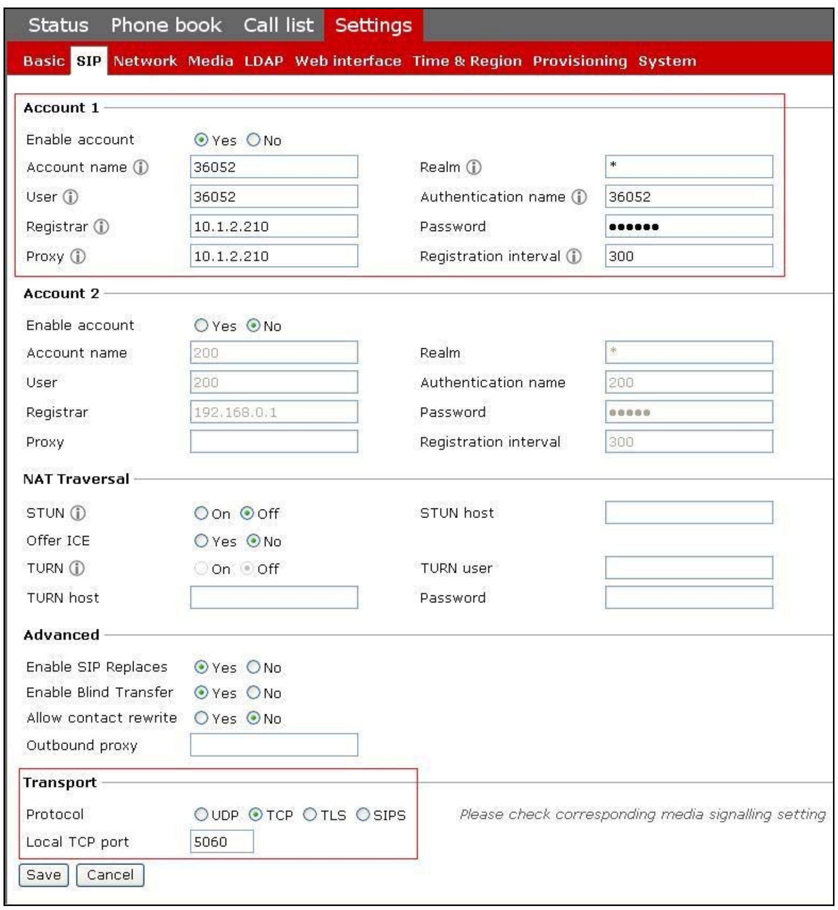


Figure 7: SIP Settings Screen

The rest of the configuration was left at default values. When complete, click **Save**. This will cause the phone to reboot.

4. Configuration of the Avaya Aura® Session Manager

This section shows the Session Manager screens used to configure an endpoint with extension 36052, for the B179 SIP Conference Phone. For additional information on configuring Session Manager, see reference [3].

Login to System Manager and select User Management → Manage Users. A list of users will appear. Click New to create a new endpoint. The Identity and Comunication Profile tabs need to be configured for a new user.

Figure 8 shows the **Identity** configuration used. Enter a meaningful **Name** for the user, as well as a meaningful **Localized** and **Endpoint Display Name**. For **Login Name** enter the extension and domain that the endpoint will use to register to Session Manager. Enter a password for the user. Note that this password is not the password the endpoint will use to register to Session Manager.

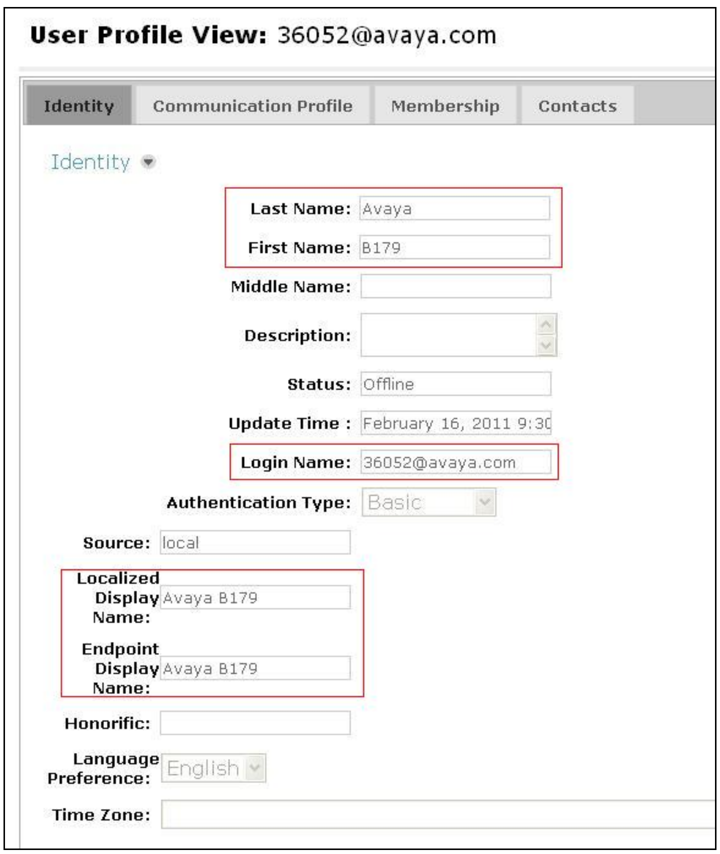


Figure 8: User Profile Identity

Figure 9 shows the **Communication Profile** configuration used.

Communication Profile:

- **Communication Profile Password**: Enter the password the endpoint will use to register to Session Manager
- **Confirm Password**: Re-enter the same password.

Communication Address:

- Click **New** to create a new entry.
- **Type**: Avaya SIP
- **Fully Qualified Address**: Enter appropriate Handle (extension) and select appropriate Domain.
- Click **Add** to add the new entry.

Session Manager Profile:

- **Primary Session Manager**: Select appropriate Session Manager
- **Origination and Termination Application Sequence**: Select appropriate sequence
- **Home Location**: Select appropriate location.

Endpoint Profile:

- **System**: Select appropriate Communication Manager System
- **Profile Type**: Endpoint
- **Extension**: Enter extension for this endpoint, 36052
- **Template**: Select the appropriate template. DEFAULT_9630SIP_CM_6_0 was used.
- **Port**: IP (this will change to a virtual port when form is submitted)

Click **Commit** (not shown) when done.

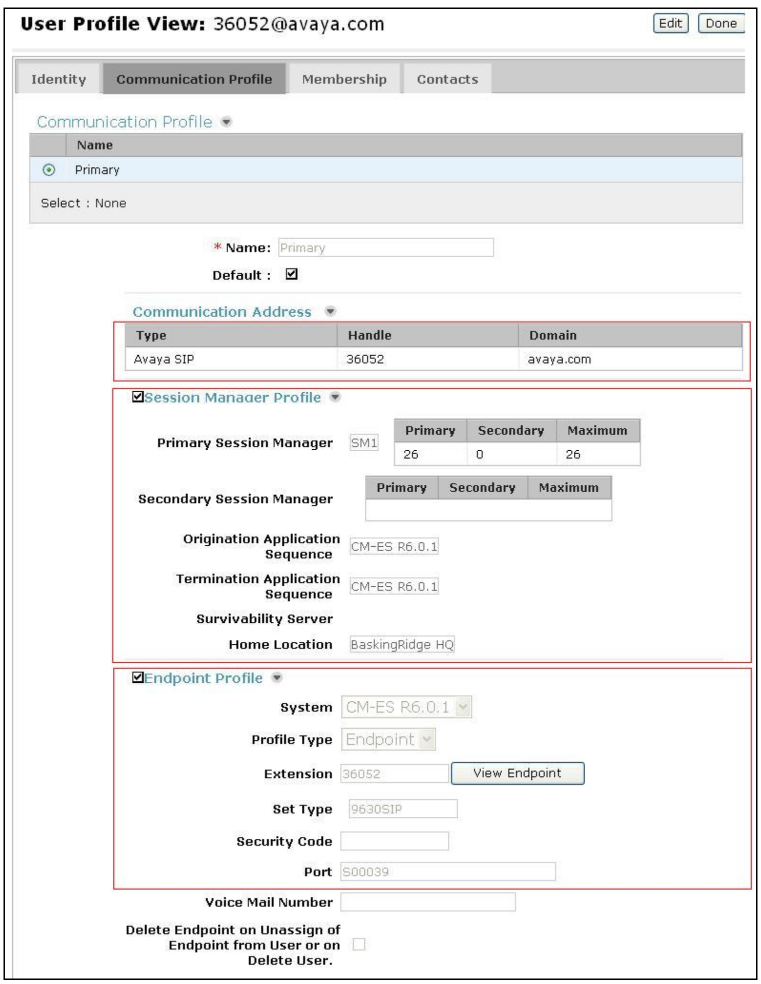
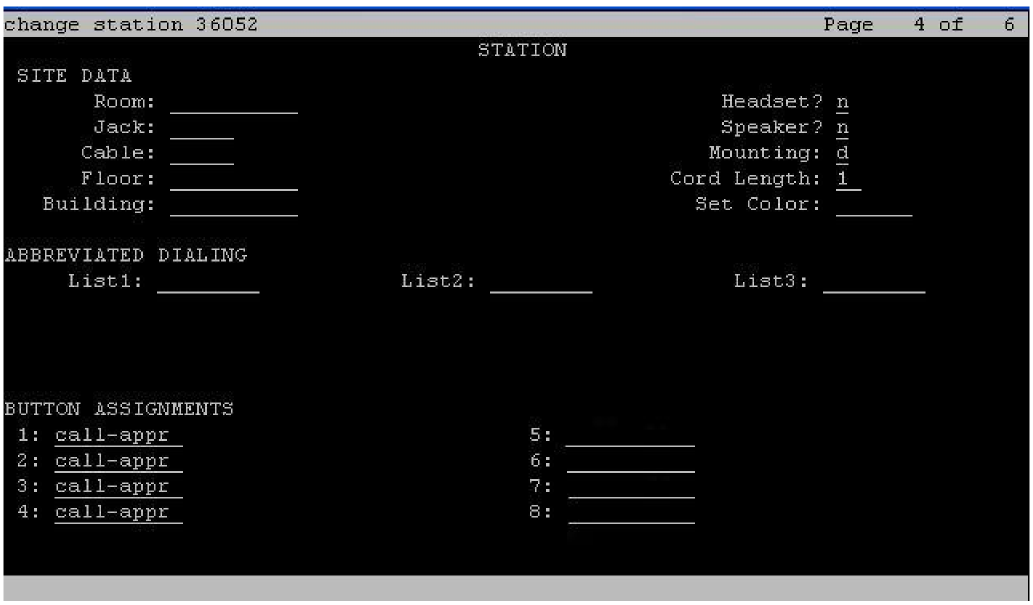


Figure 9: User Communication Profile

5. Configuration of the Avaya Aura® Communication Manager

Basic configuration of Communication manager is outside the scope of this document. A station to support the B179 SIP Conference Phone is created in Communication Manager automatically when using System Manager to add a User in Session Manager, as described in **Section 4**.

If the native conference feature of the B179 SIP Conference phone is to be used, it's necessary that the station in Communication Manager have at least 4 call appearances, as each call initiated by the B179 SIP Conference Phone will use a call apperance. This was manually done via the SAT interface in Communication Manager, using the "change station 36052" command, as shown in **Figure 10**. For additional information on configuring Configuration Manager, see references [4] and [5]. System Manager can also be used to add/change an endpoint's button assignments.



```
change station 36052                                              Page   4 of   6
                                     STATION
 SITE DATA
       Room: __________                               Headset? n
       Jack: _____                                    Speaker? n
      Cable: _____                                   Mounting: d
      Floor: __________                          Cord Length: 1
   Building: __________                            Set Color: ______

ABBREVIATED DIALING
     List1: ________          List2: ________         List3: ________



BUTTON ASSIGNMENTS
 1: call-appr                            5: __________
 2: call-appr                            6: __________
 3: call-appr                            7: __________
 4: call-appr                            8: __________
```

Figure 10: Change Station

6. Observations

At this time, for full feature functionality, it is recommended that multiple codecs be configured using the priority value to determine the preference. At a minimum one of the G.711 codecs must be selected. This may not be required in future releases.

UDP cannot be used as the SIP signaling transport protocol if VLAN tagging is used.

For full functionality with the conference feature, native to the B179 SIP Conference Phone, the station should be administered in Communication Manager with at least 4 call appearances.

7. Verification Steps

Look at the screen of the B179 SIP Conference Phone. A dark square next to the extension indicates the endpoint has succesfully registered. A clear/empty square indicates the endpoint is not registered.

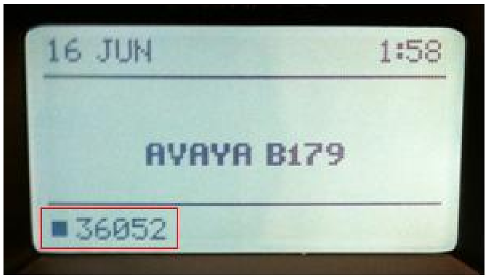


Figure 11: Registered Endpoint

If the endpoint is not registered, verify that the configuration was entered correctly. When the endpoint registers verify it is working properly by placing calls to and from the enpoint.

8. Conclusion

These Application Notes described the configuration for the Avaya B179 SIP Conference Phone to connect to Session Manager and Communication Manager. The Avaya B179 SIP Conference Phone sucessfully registed and was able to place and receive calls. Some observations were noted in section 6.

9. Additional References

The Avaya product documentation is available at http://support.avaya.com.

[1] Avaya B179 SIP Conference Phone User Guide

[2] Avaya B179 SIP Conference Phone Installation and Administration Guide

[3] Administering Avaya Aura® Session Manager; Issue 1, 03-603324, Release 6.1, Nov 2010

[4] Administering Avaya Aura® Communication Manager; Issue 6.0, 03-300509, Release 6.0, June 2010

[5] Avaya Aura® Communication Manager Screen Reference; Issue 2.0, 03-602878, Release 6.0, June 2010

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com