



Avaya Solution & Interoperability Test Lab

Application Notes for Jabra Direct and Speak 710 Bluetooth/USB Speakerphone with Avaya one-X® Communicator - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Jabra Direct and Speak 710 Bluetooth/USB speakerphone with Avaya one-X® Communicator.

Readers should pay particular attention to the scope of testing as outlined in **Section 2.1**, as well as 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 steps required to integrate Jabra Direct and Speak 710 Bluetooth/USB (Speak 710) speakerphone with Avaya one-X® Communicator. Direct is a PC application that enables Jabra USB devices to have call control over soft phones, including Speak 710 and one-X Communicator respectively. Speaker 710 has two kinds of connection to PC, one with a provided wired USB cable and another via Link 370 Bluetooth USB dongle. Link 370 also comes with the Speaker 710.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The testing utilized two connection methods of Speak 710 to Direct: USB cable and USB Bluetooth dongle to test with the one-X Communicator. The feature testing focused on placing calls to and from one-X Communicator using Direction application and Speak 710 speakerphone, answering and ending calls using the call control button on the speakerphone, and verifying two-way audio. The call types included calls between local extensions, and with the PSTN.

The serviceability testing focused on verifying the usability of the Jabra speakerphone solution after restarting one-X Communicator and reconnecting the Jabra speakerphone to the PC.

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.

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.

2.1. Interoperability Compliance Testing

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

- Placing calls to and from internal extensions to verify two-way audio.
- Placing calls to and from the PSTN and to Voicemail to verify two-way audio.
- Answering and ending calls using the call control button on the speakerphone.
- Verifying incoming call notification on Jabra speakerphone.
- Verifying call ended notification on Jabra speakerphone.

For the serviceability testing, the Jabra speakerphone was reconnected to one-X Communicator and the PC was restarted to verify proper operation of the speakerphone.

2.2. Test Results

All test cases were executed and passed.

2.3. Support

For support on this Jabra speakerphone solution, contact Jabra Technical Support at:

- Phone: (800) 697-8757
- Website: <http://www.jabra.com/NA-US/Support/pages/Default.aspx>
- Email: JabraSupport.US@jabra.com.

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Avaya Aura® Communication Manager running on a virtualized environment with an Avaya G450 Media Gateway.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.

In addition, the Direct application was installed on a PC that also has the one-X Communicator soft phone installed.

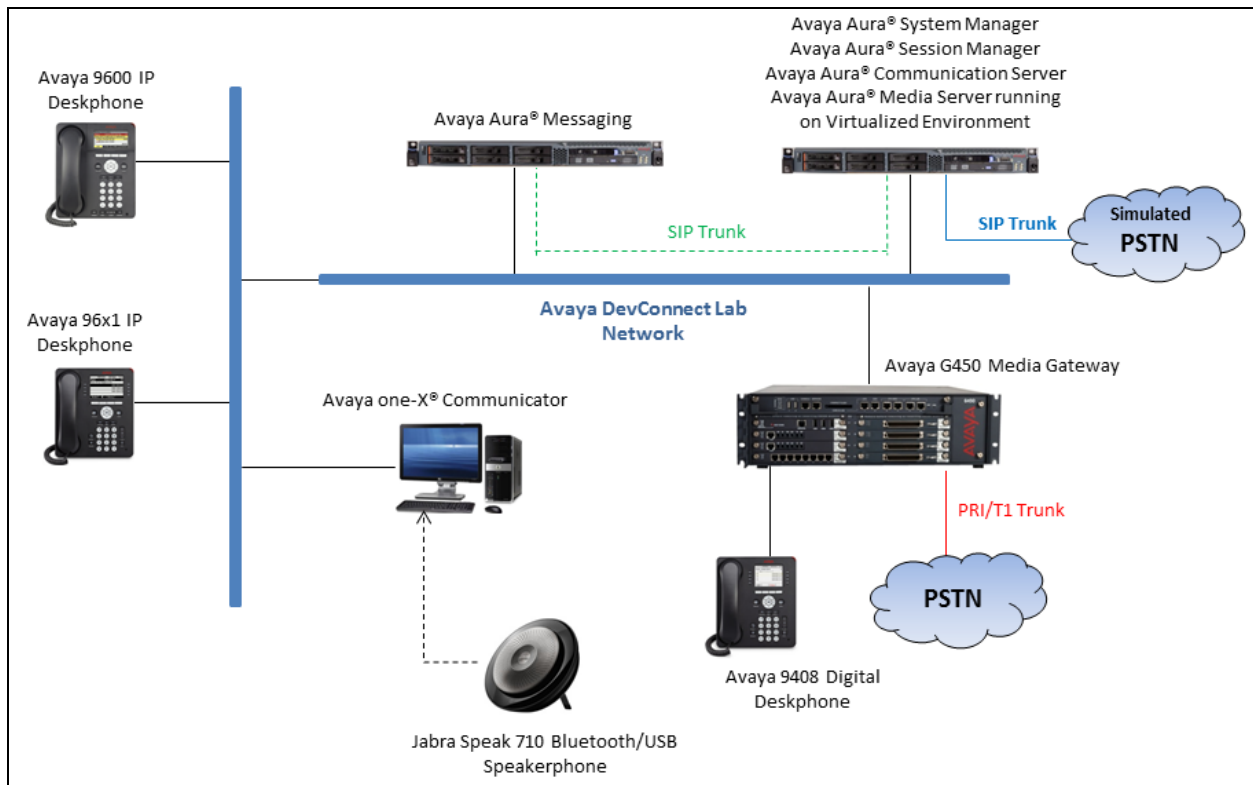


Figure 1: Test Configuration Diagram

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on a virtual platform	7.1.0.532.0
Avaya Aura® Session Manager running on a virtual platform	7.1.0.0.710028
Avaya Aura® System Manager running on a virtual platform	7.1.0.0.1
Avaya Aura® Messaging running on a virtual platform	7.0 SP0
Avaya Aura® Media Server running on a virtual platform	7.8.0.309
Avaya one-X® Communicator running on Microsoft Windows 10 Professional	6.2.12
Avaya 9611G IP Deskphone	7.0.1.2 (SIP) 6.64 (H323)
Avaya 9640 IP Deskphone	3.25
Avaya 9408 Digital Deskphone	V 2.0
Jabra Direct	3.8.689
Jabra Link 370 (USB Bluetooth dongle)	1.14.0
Jabra Speak 710	1.10.0

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator emulation mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Define the Dial Plan

Note: Any settings not in **Bold** in the following screen shots may be left as default.

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per one-X Communicator.

```
display system-parameters customer-options                               Page 1 of 10
                                OPTIONAL FEATURES

G3 Version: V16                                     Software Package: Enterprise
Location: 2                                         System ID (SID): 1
Platform: 28                                       Module ID (MID): 1

                                                USED
Platform Maximum Ports: 65000 290
Maximum Stations: 41000 44
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 14
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 41000 0
Maximum Survivable Processors: 313 0

(NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options form**, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

```

display system-parameters customer-options                               Page 2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
    Maximum Administered H.323 Trunks: 12000 16
    Maximum Concurrently Registered IP Stations: 18000 2
    Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
    Maximum Concurrently Registered IP eCons: 414 0
    Max Concur Registered Unauthenticated H.323 Stations: 100 0
    Maximum Video Capable Stations: 41000 1
    Maximum Video Capable IP Softphones: 18000 4
    Maximum Administered SIP Trunks: 24000 180
Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
    Maximum Number of DS1 Boards with Echo Cancellation: 522 0
    Maximum TN2501 VAL Boards: 128 0
    Maximum Media Gateway VAL Sources: 250 0
    Maximum TN2602 Boards with 80 VoIP Channels: 128 0
    Maximum TN2602 Boards with 320 VoIP Channels: 128 0
    Maximum Number of Expanded Meet-me Conference Ports: 300 0

(NOTE: You must logoff & login to effect the permission changes.)
  
```

5.2. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are 4 digits long and begin with **33** and **34**.

```

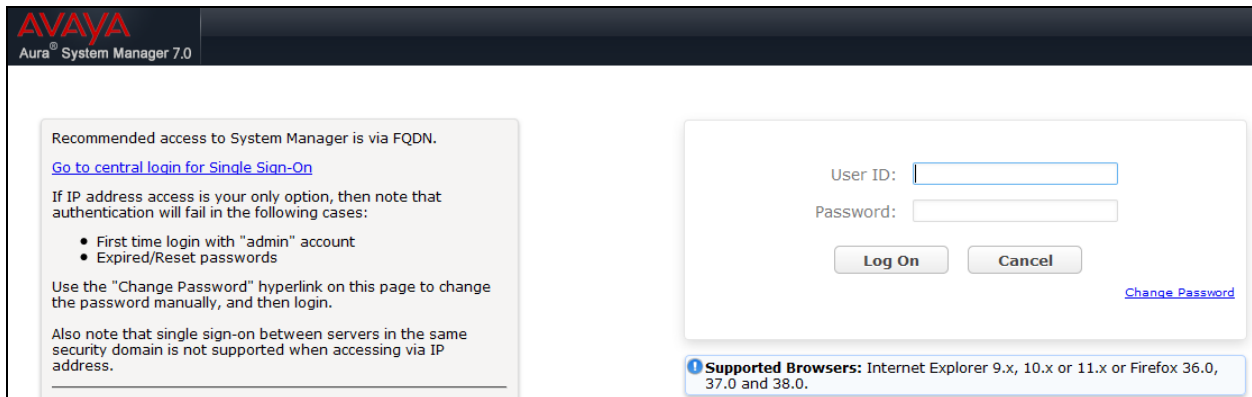
change dialplan analysis                                               Page 1 of 12
                                DIAL PLAN ANALYSIS TABLE
                                Location: all                            Percent Full: 1

    Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
    String   Length  Type   String   Length  Type   String   Length  Type
33         4   ext
34         4   ext
    *         3   fac
    #         3   fac
  
```

6. Configure Avaya Aura® Session Manager

This section describes aspects of the Session Manager configuration required for one-X Communicator to register. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication Manager, Session Manager and Messaging.

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



AVAYA
Aura® System Manager 7.0

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.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

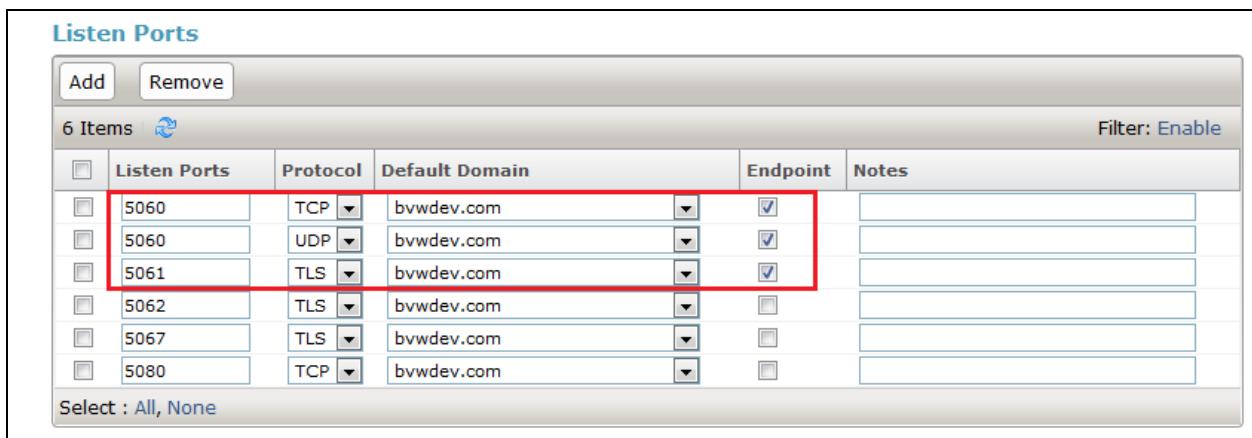
Password:

[Change Password](#)

Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

6.1. Check Avaya Aura® Session Manager Ports for SIP Endpoint Registration

Each Session Manager Entity must be configured so that the SIP Endpoint can register to it. From the web interface click **Routing** → **SIP Entities** (not shown) and select the Session Manager entity used for registration. Make sure that TCP, UDP and TLS entries are present. During the compliance test, one-X Communicator registered to Session Manager using TLS transport.



Listen Ports

6 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5062	TLS	bvwdev.com	<input type="checkbox"/>	
<input type="checkbox"/>	5067	TLS	bvwdev.com	<input type="checkbox"/>	
<input type="checkbox"/>	5080	TCP	bvwdev.com	<input type="checkbox"/>	

Select : All, None

6.2. Add a SIP User

A SIP user for one-X Communicator must be added. Navigate to **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

- **Last Name and First Name** Enter identifying names.
- **Login Name** Enter the extension number followed by the domain, in this case **3407@bvwdev.com**.
- **Authentication Type** Select **Basic** from the drop down list.
- **Password and Confirm Password** Enter and confirm a password.

The screenshot shows the 'New User Profile' form in the 'User Management' application. The form is divided into four tabs: Identity, Communication Profile, Membership, and Contacts. The 'Identity' tab is selected and contains the following fields:

- User Provisioning Rule:** A dropdown menu.
- Identity:**
 - Last Name:** SIP
 - Last Name (Latin Translation):** SIP
 - First Name:** 3407
 - First Name (Latin Translation):** 3407
 - Middle Name:** (empty)
 - Description:** (empty text area)
 - Login Name:** 3407@bvwdev.com
 - User Type:** Basic
 - Password:** (masked with dots)
 - Confirm Password:** (masked with dots)
 - Localized Display Name:** (empty)

Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the one-X Communicator during login.

The screenshot shows the 'New User Profile' form with the 'Communication Profile' tab selected. The form has three tabs: 'Identity *', 'Communication Profile', and 'Membership'. Below the tabs, there are two input fields: 'Communication Profile Password' and 'Confirm Password', both containing four dots. At the top right of the form, there are three buttons: 'Commit & Continue', 'Commit', and 'Cancel'.

In the **Communication Address** section, for **Type** select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate **Domain** from the drop down list. Click **Add** when done.

The screenshot shows the 'Communication Address' section of the user profile form. At the top, there are buttons for 'New', 'Delete', 'Done', and 'Cancel'. Below these is a 'Name' section with a 'Primary' radio button selected and a 'Select : None' dropdown. The 'Name' field contains 'Primary' and is marked with an asterisk. Below the name field is a 'Default' checkbox which is checked. The 'Communication Address' section is highlighted with a red box. It contains a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty, showing 'No Records found'. Below the table, there are three input fields: 'Type' (set to 'Avaya SIP'), 'Fully Qualified Address' (set to '3407'), and 'Domain' (set to 'bvwdev.com'). The 'Fully Qualified Address' field is marked with an asterisk. At the bottom right of this section are 'Add' and 'Cancel' buttons.

Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager**, **Origination Sequence**, **Termination Sequence** and **Home Location**, from the respective drop down lists. The primary Session Manager used was **ASM70A**.

Session Manager Profile ▼

SIP Registration

* Primary Session Manager

Primary	Secondary	Maximum
13	0	13

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

* Home Location ▼

Conference Factory Set ▼

Call History Settings

Enable Centralized Call History?

Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **System** Select the relevant Communication Manager SIP Entity from the drop down list.
- **Profile Type** Select **Endpoint** from the drop down list.
- **Extension** Enter the required extension number, in this case **3407**.
- **Template** Select **9611SIP_DEFAULT_CM_7_0** from the drop down list.
Note that the template can be any SIP template in this example the 9611SIP used.
- **Port** The “IP” is auto filled out by the system.

Click on **Endpoint Editor**.

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

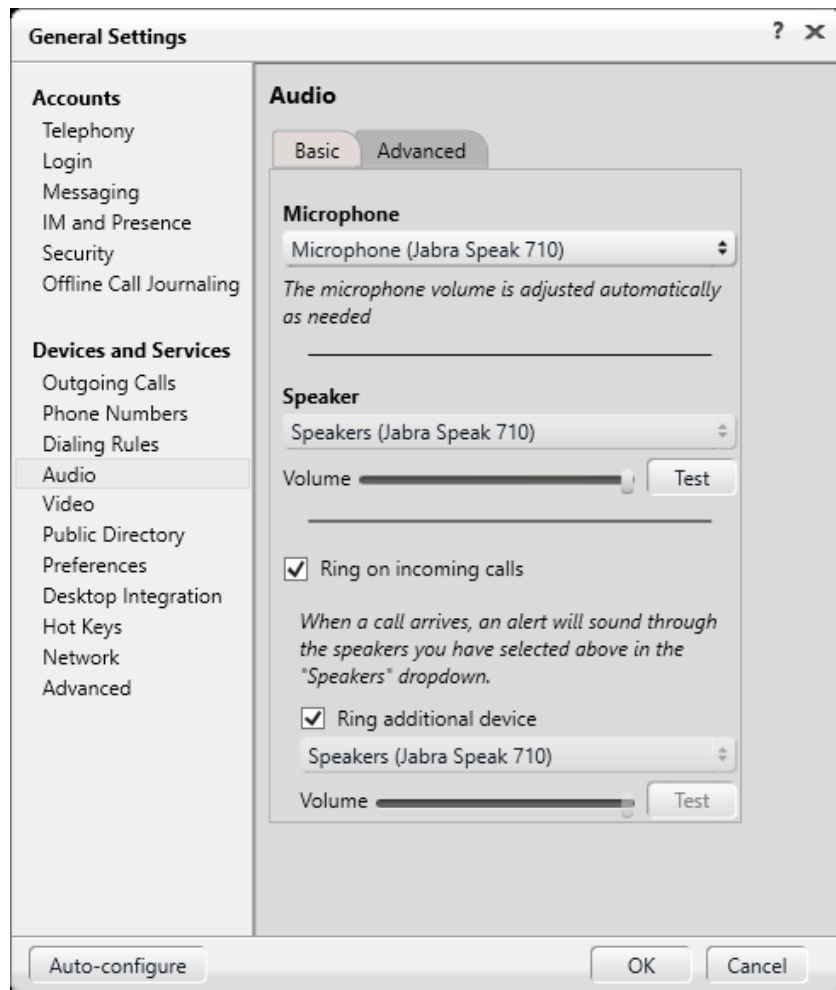
Click on the **Feature Options** tab, the screen shot below shows the Feature options that were used during compliance testing. Make sure the **IP SoftPhone** checkbox is selected.

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	None	Coverage After Forwarding		
Per Station CPN - Send Calling Number	None	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 CDR	internal	
AUDIX Name	None	Time of Day Lock Table	None	
Speakerphone		Voice Mail Number		
Short/Prefixed Registration Allowed	default	Music Source		
EC500 State	enabled			
Features				
<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"/> Precedence Call Waiting		
<input type="checkbox"/> Data Restriction		<input checked="" type="checkbox"/> Direct IP-IP Audio Connections		
<input checked="" type="checkbox"/> Survivable Trunk Dest		<input type="checkbox"/> H.320 Conversion		
<input type="checkbox"/> Bridged Appearance Origination Restriction		<input type="checkbox"/> IP Video Softphone		
<input checked="" type="checkbox"/> Restrict Last Appearance		<input type="checkbox"/> Per Button Ring Control		

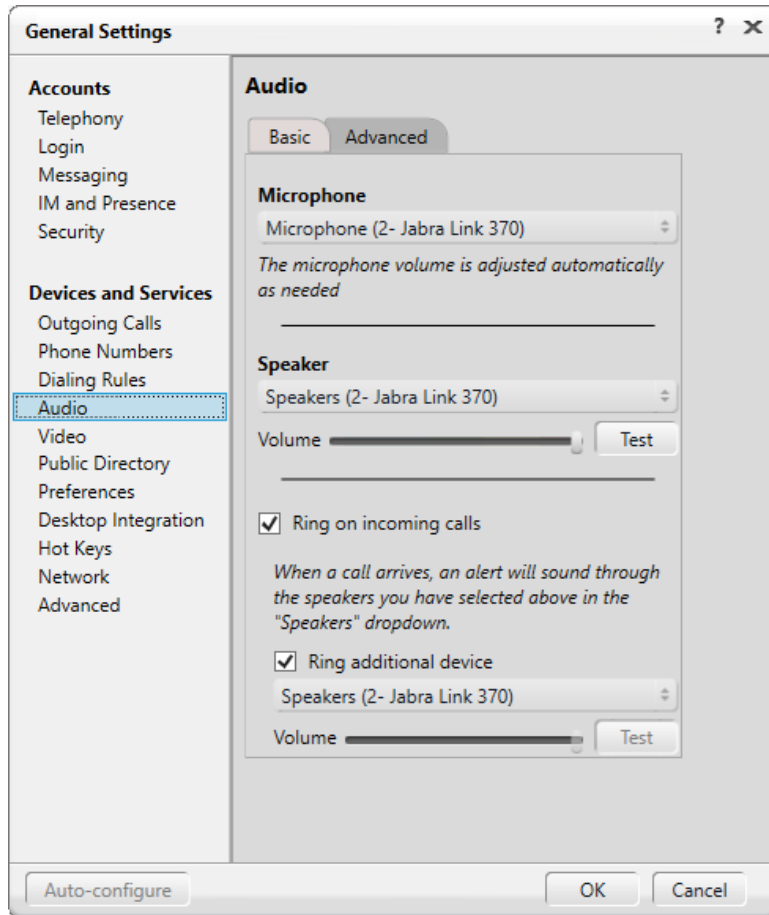
7. Configure Avaya one-X Communicator

This section shows how to configure one-X Communicator to use Speak 710.

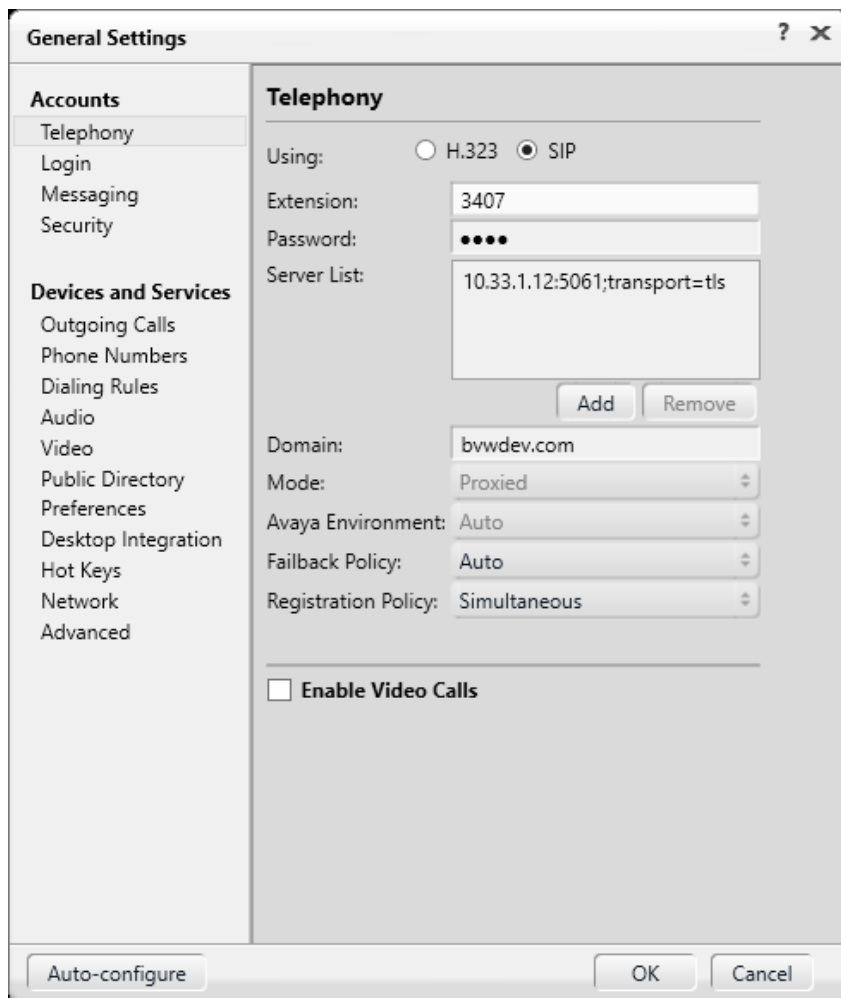
From the one-X Communicator window, navigate to the setting icon  → **Settings** → **Devices and Services** → **Audio** and select *Microphone (Jabra Speak 710)* and *Speakers (Jabra Speak 710)* in the **Microphone** and **Speaker** fields as shown below.



In case of Jabra Link 370 USB Bluetooth dongle being used, select the Jabra Link 370 values respectively as shown below.



From the one-X Communicator window, click  → **Settings** → **Accounts** → **Telephony** and enter the SIP user and password as configured in **Section 6**.



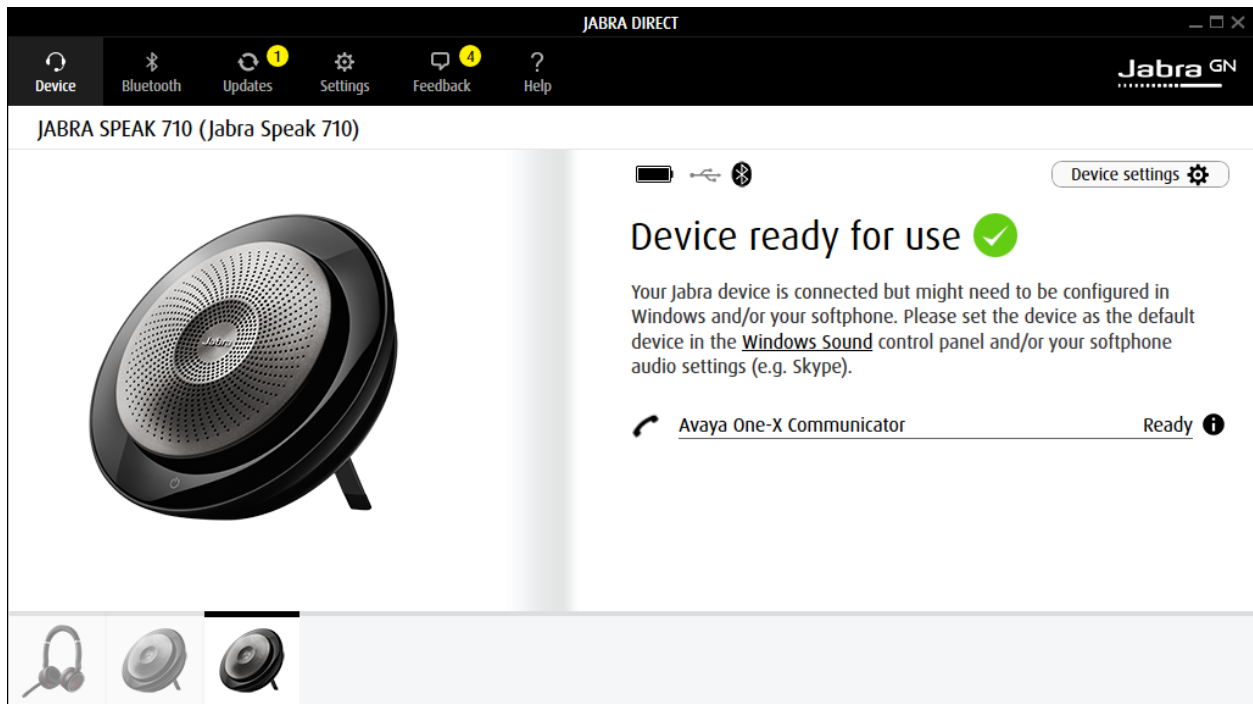
This window shows the one-X Communicator successfully registers to Session Manager.



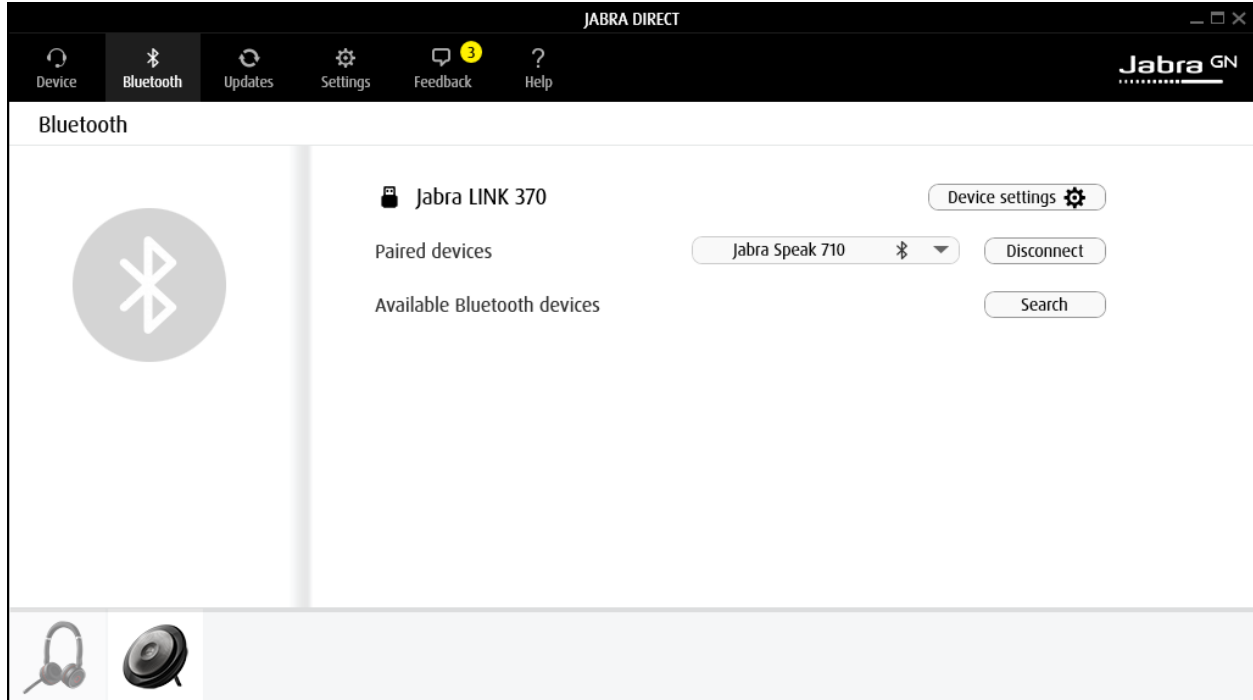
8. Configure Jabra Direct

Direct is PC software and can be downloaded from Jabra website at <http://www.jabra.ca/software-and-services/jabra-direct>. Direct enables Speak 710 to have call control over one-X Communicator. This section assumes Direct is already installed and updated to the latest version. Direct automatically detects Speaker 710 and the connection method to the PC.

The picture below shows the **Device** tab in Direct dashboard having the Speaker 710 speakerphone connected to the PC via USB Bluetooth dongle, it displays “Device ready for use” with one-X Communicator in **Ready** state.



The **Bluetooth** tab displays the Link 370 USB dongle that paired with Speak 710.



9. Verification Steps

This section verifies that Jabra Direct and Speak 710 Bluetooth/USB speakerphone has been successfully integrated Avaya one-X Communicator.

1. Verify that the Speak 710 speakerphone has been successfully installed and is ready for use with the one-X Communicator.
2. Once the Speaker 710 speakerphone is selected in the one-X Communicator, verify that incoming and outgoing calls can be established with two-way audio to the speakerphone.
3. Verify that the Speaker 710 speakerphone is able to control call functions from the speakerphone such as mute/un-mute, hold/un-hold and disconnect the call.

10. Conclusion

These Application Notes describe the configuration steps required to integrate Jabra Direct and Speak 710 Bluetooth/USB speakerphone with Avaya one-X® Communicator. All test cases were completed successfully; refer to **Section 2.2** for details.

11. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com> where the following documents can be obtained.

- [1] Administering Avaya Aura® Communication Manager, Release 7.1, May 2017, Document Number 03-300509, Issue 1.
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.1, May 2017, Document Number 555-245-205, Issue 1.
- [3] Administering Avaya Aura® Session Manager, Release 7.1, May 2017
- [4] Administering Avaya Aura® System Manager, Release 7.1, May 2017
- [5] Administering Avaya one-X® Communicator, Release 6.2, June 2016

The Jabra Speaker 710 documentation is available at <https://www.jabra.com/supportpages/jabra-speak-710#/#7710-409>

- [1] Technical Specification
- [2] User Manual

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