



Application Notes for Configuring Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset with Avaya 96x1 Series IP Deskphones - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset with Avaya 96x1 Series IP Deskphones using H.323 and SIP protocols. The Axtel Elite HD Voice Mono/Duo NC Headsets provide two-way audio with a flexible mic. This solution does not provide call control features directly from the headset, such as answering or terminating a call from the headset. Volume control and mute are provided directly from the Avaya IP deskphone.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 configuration steps required to integrate the Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset with Avaya 96x1 Series IP Deskphones using H.323 and SIP protocols. The Axtel Elite HD Voice Mono/Duo NC Headsets provide two-way audio with a flexible mic. This solution does not provide call control features directly from the headset, such as answering or terminating a call from the headset. Volume control and mute are provided directly from the Avaya IP deskphone.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on placing calls to and from the Avaya 96x1 Series IP Deskphones with the Axtel Elite HD Voice Mono/Duo NC Headset and verifying two-way audio. The call types included calls to voicemail, local extensions, and the PSTN.

The serviceability testing focused on verifying the usability of the Axtel headsets after restarting the Avaya 96x1 Series IP Deskphones and re-connecting the headset to the deskphone.

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'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 the voicemail system. Voice messages were recorded and played back to verify that the playback volume and recording level were good.
- Placing calls to internal extensions to verify two-way audio.
- Placing calls to the PSTN to verify two-way audio.
- Incoming call alert notification.
- Hearing ring back tone for outgoing calls.
- Toggling between handset, speakerphone, and headset.
- Using the volume control buttons on the Avaya deskphone.
- Using the mute control button on the Avaya deskphone to mute and un-mute the audio.

For the serviceability testing, the Avaya IP deskphones and headset were restarted to verify proper operation of the headset after the reboot was completed.

2.2. Test Results

All test cases were executed and passed.

2.3. Support

For technical support and information on Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset, contact Axtel Support at:

- Phone: +48 22 350 96 90
- Website: <http://www.Axtelheadsets.com/en/support>

3. Reference Configuration

Figure 1 illustrates the test configuration used to verify the Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset with Avaya 96x1 Series IP Deskphones. The configuration consists of System Manager, Session Manager, Communication Manager, Aura Messaging, and Media Server appliances that were installed in virtual environment which is a VMware server. The testing used both Media Gateway G450 and Media Server to balance DSP resources that connect from/to endpoints in Communication Manager. The testing has a simulated PSTN connected from Avaya Media Gateway G450 to another PBX system in the lab via PRI trunk. Axtel AXC-HIS cable connects to the Elite HD Voice headset via quick connector and connects to Avaya 96x1 IP deskphone in RJ11 headset port.

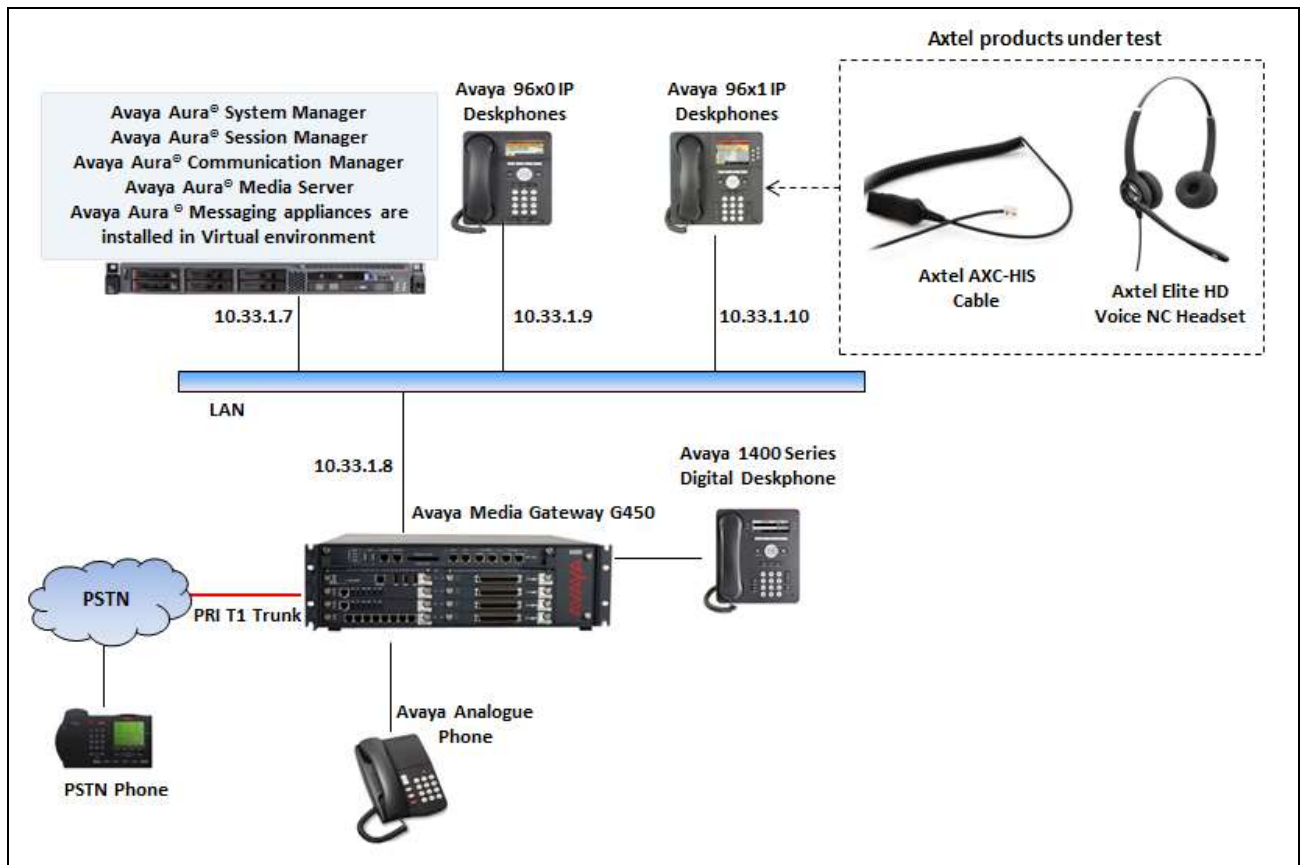


Figure 1: Avaya 96x1 Series IP Deskphones with Axtel AXC-HIS Cable and Axtel Elite HD Voice Mono/Duo NC Headset

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager in Virtual Environment	7.0.0.0.16266
Avaya Aura® Session Manager in Virtual Environment	7.0.0.0.700007
Avaya Aura® Communication in Virtual Environment	7.0 (R017x.00.0.441.0) Patch 22477
Avaya Aura® Messaging in Virtual Environment	6.3.1
Avaya Aura® Media Server in Virtual Environment	7.7.0.236
Avaya Media Gateway G450	37.19.0
Avaya 96x1 H323 IP Deskphones (9608, 9611G, 9621G, 9641G)	6.6115 (H323) 7.0.39 (SIP)
Axtel AXC-HIS Cable	P/N: AXC-HIS
Axtel Elite HD Voice Mono/Duo Headsets	P/N: AXH-EHDD

5. Configure Avaya Aura® Communication Manager

This section covers the station configuration for the Avaya 96x1 IP Deskphones. Configuration is performed via the System Access Terminal (SAT) on Communication Manager or via System Manager for SIP stations.

5.1. Configure a Station for Avaya 96x1 Series IP Deskphone

Use the **add station** command to create a station for the 9641 IP deskphone. Set the **Type** field to the station type to be emulated. In this example, *9641* was used. Set the **Port** field to *IP* and configure a **Security Code** as that password to be used by the Avaya deskphone to log in.

Note: To enable Auto Answer on the IP deskphone set the **Auto Answer** field on **Page 2** (not shown) to the appropriate value, such as *all*.

```
add station 3301                                     Page 1 of 5
                                                    STATION
Extension: 3301                                     Lock Messages? n          BCC: M
  Type: 9641                                         Security Code: 1234       TN: 1
  Port: IP                                           Coverage Path 1:         COR: 1
  Name: H.323 3301                                   Coverage Path 2:         COS: 1
                                                    Hunt-to Station:        Tests? y
STATION OPTIONS
Loss Group: 19                                     Time of Day Lock Table:
Speakerphone: 2-way                               Personalized Ringing Pattern: 1
Display Language: english                         Message Lamp Ext: 3301
Survivable GK Node Name:                          Mute Button Enabled? y
Survivable COR: internal                           Button Modules: 0
Survivable Trunk Dest? y                           Media Complex Ext:
                                                    IP SoftPhone? y
                                                    IP Video Softphone? y
Short/Prefixed Registration Allowed: default
                                                    Customizable Labels? y
```

5.2. Configure a Station for Avaya 96x1 Series SIP Deskphone

The SIP station was previously configured through System Manager. Use the **display station** command to view the station for the 9611G IP deskphone. The **Station Type** was set to *9611SIP* and a descriptive **Name** was also provided. Use the default values for the other fields on **Page 1**.

Note: To enable Auto Answer on the IP deskphone set the **Auto Answer** field on **Page 2** (not shown) to the appropriate value, such as *all*.

```
display station 3400                                     Page 1 of 6
                                                         STATION
Extension: 3400                                         Lock Messages? n          BCC: M
Type: 9611SIP                                           Security Code:            TN: 1
Port: S00004                                           Coverage Path 1:         COR: 1
Name: SIP, 3400                                         Coverage Path 2:         COS: 1
                                                         Hunt-to Station:
STATION OPTIONS
                                                         Time ofDay Lock Table:
Loss Group: 19                                          Message Lamp Ext: 3400
                                                         Button Modules: 0
Display Language: english
Survivable COR: internal
Survivable Trunk Dest? y                               IP SoftPhone? y
                                                         IP Video Softphone? y
Short/Prefixed Registration Allowed: default
```

5.3. Configure 46xxsettings.txt File

In the 46xxsettings.txt file, the HEADSETBIDIR parameter needs to be set to '1' so that switch hook and alerting are enabled for the 96x1 IP deskphone. This allows incoming call alert to be heard through the headset. Alternatively, the switch hook and alerting options can be enabled through the 96x1 phone menu. Press the **Menu** button on the phone and then navigate to **Options & Settings → Call Settings → Headset Signaling...** Select the **Switchhook & Alerting** option. Below is an example for setting this parameter in the 46xxsettings.txt file.

```
##### Headset Signaling #####
##
## HEADSETBIDIR specifies whether bidirectional signaling
## on the headset interface will be enabled or disabled.
## Value Operation
## 0 Disabled (default)
## 1 Switchhook and alerting signaling are both enabled
## 2 Only switchhook signaling is enabled
## This parameter is supported by:
## H1xx SIP R1.0 and later
## 96x1 H.323 R6.3 and later (values 0-2)
## 96x1 H.323 R6.2.1 and later (values 0-1)
## Note that 96x1 H.323 R6.2 only supported signaling for alerting.
SET HEADSETBIDIR 1
```


For H.323 and SIP Deskphones

In the 46xxsettings.txt file, the HEADSYS parameter needs to be set appropriately depending on whether the headset button should be deactivated automatically if the far-end drops the call. For stations configured with auto-answer, set this parameter to '1' so that the headset button is not deactivated when the far-end drops the call. This allows the subsequent call to be answered automatically through the headset. Below is an example for setting this parameter. In this example, the parameter is set to '0', which would cause the headset button to be deactivated when the far-end drops the call first.

```
##### CALL CENTER SETTINGS #####
##
## HEADSYS specifies whether the telephone will go on-hook if the
## headset is active
## when a Disconnect message is received.
## Value Operation
## 0 The telephone will go on-hook if a Disconnect message is
## received when the headset is active
## 1 Disconnect messages are ignored when the headset is active
## Note: a value of 2 has the same effect as a value of 0, and
## a value of 3 has the same effect as a value of 1.
## This parameter is supported by:
## 96x1 H.323 R6.2.1 and later (the default value is 0 unless
## the value
## of CALLCTRSTAT is set to 1, in which case the default
## value is 1)
## 96x1 H.323 R6.1 and R6.2 ignore this parameter, and will
## ignore Disconnect messages
## if the user is logged in as a call center agent. If the
## user is not logged in
## as a call center agent, the telephone will go on-hook if
## a Disconnect message
## is received when the headset is active.
## 96x1 H.323 releases prior to R6.1 (the default value is 1)
## 96x1 SIP R6.4 and later (the default value is 0)
## 96x1 SIP R6.0 and later up to R6.4 (not included) (the
## default value is 1)
## 96x0 H.323 R1.2 and later (the default value is 1)
## 96x0 SIP R1.0 and later (the default value is 1)
## 16xx H.323 R1.3 and later (the default value is 1)
SET HEADSYS 0
```

6. Configure Axtel AXC-HIS Cable & Axtel Elite HD Voice Mono/Duo NC Headset

Connect the Elite HD Voice headset to the AXC-HIS adapter cable, and then connect the HIS cable directly to the headset port of the Avaya 96x1 Series IP Deskphones.

7. Verification Steps

Verify that the Axtel Elite HD Voice Mono/Duo NC Headsets and the AXC-HIS adapter cable have been connected to the Avaya 96x1 Series IP Deskphones. Once the headset is connected to the phone, verify that incoming and outgoing calls are established with two-way audio to the headset.

8. Conclusion

These Application Notes describe the configuration steps required to integrate the Axtel AXC-HIS Cable & Axtel Elite HD Voice Mono/Duo NC Headset with Avaya 96x1 Series IP Deskphones. All test cases were completed successfully.

9. Additional References

This section references the Avaya and Axtel 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 7.0, Issue 1, August 2015, Document Number 03-300509.
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.0, June 2015
- [3] Administering Avaya 9601/9608/9611G/9621G/9641G/9641GS IP Deskphones SIP, Release 7.0, Issue 2, August 2015
- [4] Administering Avaya 9608/9608G/9611G/9621G/9641G/9641GS IP Deskphones H.323, Release 6.6, Issue 1, April 2015

The Axtel product documentation can be found at <http://www.Axtelheadsets.com/en/>

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