



## Avaya Solution & Interoperability Test Lab

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# Application Notes for AudioCodes 300HD SIP IP Phone Series with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services - Issue 1.0

### Abstract

These Application Notes describe the steps required to integrate an AudioCodes 320HD SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with SIP Enablement Services, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

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 an AudioCodes 320HD SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with SIP Enablement Services, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

## 1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

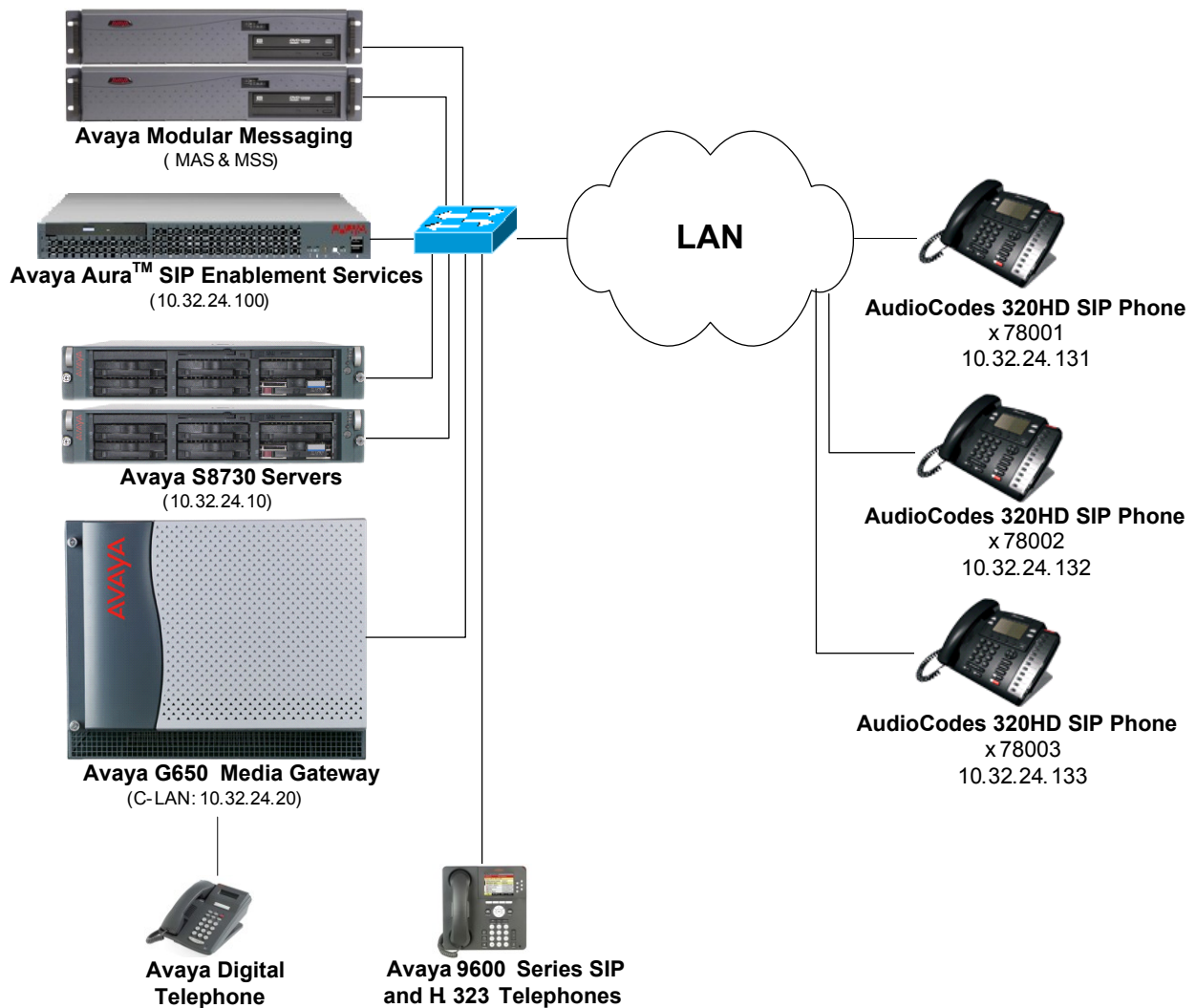
- Successful registration of the AudioCodes 320HD SIP Phone with SIP Enablement Services.
- Calls between 320HD SIP phones and Avaya SIP, H.323, and digital stations.
- G.711 and G.729A codec support.
- Proper recognition of DTMF tones by navigating voicemail menus.
- Proper operation of voicemail with Message Waiting Indication (MWI).
- Basic telephony features including Hold, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding, Call Pickup, and Send All Calls.
- Proper system recovery after a 320HD SIP phone restart and loss of IP connectivity.

## 1.2. Support

For technical support on the 300HD SIP IP Phone Series contact AudioCodes Customer Support through their website at <http://www.audiocodes.com/support>.

## 2. Reference Configuration

**Figure 1** illustrates a sample configuration with an Avaya SIP-based network, including Avaya Aura™ SIP Enablement Services, a pair of Avaya S8730 Servers with a G650 Media Gateway running Avaya Aura™ Communication Manager, and Avaya IP and digital endpoints. Avaya Modular Messaging provides voice mail service. The configuration contains three AudioCodes 320HD SIP Phones that register with SIP Enablement Services and are configured as Off-PBX Stations (OPS) on Communication Manager. The configuration contains three AudioCodes 320HD SIP Phones that register with SIP Enablement Services and are configured as Off-PBX Stations (OPS) on Communication Manager.



**Figure 1: AudioCodes 320HD SIP Phones with Avaya SIP Solution**

### 3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 3 (Patch 17959)
Avaya G650 Media Gateway TN799DP C-LAN TN2302AP IP Media Processor	HW01 FW031 HW03 FW094
Avaya Aura™ SIP Enablement Services	5.2.1 (SES-5.2.1.0-016.4)
Avaya Modular Messaging running on Avaya S3500 Servers	5.2
Avaya 9600 Series IP Telephones	3.0 (H.323) 2.0.5 (SIP)
Avaya Digital Telephones	N/A
AudioCodes 320HD SIP Phone	1.2.2_p009_Build_5

## 4. Configure Aura™ Avaya Communication Manager

This section describes the steps for configuring the AudioCodes 320HD SIP Phone as an Off-PBX Station (OPS) and configuring a SIP trunk between Communication Manager and SIP Enablement Services. **Section 4.3** covers the station configuration for the 320HD SIP Phones. For complete SIP documentation, see [2]. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

### 4.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative. On Page 1, verify that the number of OPS stations allowed in the system is sufficient.

```
display system-parameters customer-options                               Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V15                                     Software Package: Standard
Location: 1                                         RFA System ID (SID): 1
Platform: 6                                        RFA Module ID (MID): 1

                                                USED
Platform Maximum Ports: 48000 336
Maximum Stations: 36000 193
Maximum XMOBILE Stations: 0 0
Maximum Off-PBX Telephones - EC500: 5 0
Maximum Off-PBX Telephones - OPS: 100 33
Maximum Off-PBX Telephones - PBFMC: 0 0
Maximum Off-PBX Telephones - PVFMC: 0 0
Maximum Off-PBX Telephones - SCCAN: 0 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 SIP trunks supported by the system is sufficient.

```

display system-parameters customer-options                               Page 2 of 11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
      Maximum Administered H.323 Trunks: 200 60
    Maximum Concurrently Registered IP Stations: 18000 9
      Maximum Administered Remote Office Trunks: 0 0
Maximum Concurrently Registered Remote Office Stations: 0 0
      Maximum Concurrently Registered IP eCons: 0 0
    Max Concur Registered Unauthenticated H.323 Stations: 0 0
      Maximum Video Capable H.323 Stations: 0 0
      Maximum Video Capable IP Softphones: 0 0
      Maximum Administered SIP Trunks: 500 60
Maximum Administered Ad-hoc Video Conferencing Ports: 0 0
    Maximum Number of DS1 Boards with Echo Cancellation: 0 0
      Maximum TN2501 VAL Boards: 128 1
      Maximum Media Gateway VAL Sources: 0 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: 0 0

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

```

## 4.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the Avaya G650 Media Gateway and the SIP Enablement Services server. The host names will be used throughout the other configuration screens of Communication Manager.

```

change node-names ip                                                  Page 1 of 2
                                IP NODE NAMES

      Name                IP Address
ModMsg                   192.50.10.45
clancrm                 10.32.24.20
default                  0.0.0.0
medprocrm                10.32.24.21
procr                    0.0.0.0
ses                    10.32.24.100
( 6 of 16 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on SIP Enablement Services. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. *For this solution, shuffling should be disabled to prevent issues with the Hold and Conference features.* The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to SIP Enablement Services. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```

change ip-network-region 1                                     Page 1 of 19
                                                           IP NETWORK REGION
  Region: 1
Location:      Authoritative Domain: avaya.com
  Name:
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: no
  Codec Set: 1      Inter-region IP-IP Direct Audio: no
  UDP Port Min: 2048      IP Audio Hairpinning? y
  UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS      RTCP Reporting Enabled? y
  Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
  Audio PHB Value: 46      Use Default Server Parameters? y
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 7
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5

```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the 320HD SIP Phones. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. However, the **IP Codec Set** form may specify multiple codecs, including G.711 and G.729A, which are supported by the 320HD SIP Phones.

```

change ip-codec-set 1                                     Page 1 of 2
                                                           IP Codec Set

  Codec Set: 1

  Audio      Silence      Frames      Packet
  Codec      Suppression  Per Pkt    Size(ms)
1: G.711MU      n          2         20
2:
3:
4:
5:
6:
7:

```

Prior to configuring a SIP trunk group for communication with SIP Enablement Services, a SIP signaling group must be configured. Configure the Signaling Group form shown as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field will default to *tls* (Transport Layer Security).
- Specify the C-LAN board in the G650 Media Gateway and the SIP Enablement Services Server as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- Shuffling was disabled in the IP Network Region form which supersedes the setting of the **Direct IP-IP Audio Connections** field on this form. However, Shuffling could have been disabled here instead.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                     Page 1 of 1
                                                         SIGNALING GROUP
Group Number: 10                                         Group Type: sip
                                                         Transport Method: tls
IMS Enabled? n
Near-end Node Name: clancrm                               Far-end Node Name: ses
Near-end Listen Port: 5061                               Far-end Listen Port: 5061
                                                         Far-end Network Region: 1
Far-end Domain: avaya.com
Incoming Dialog Loopbacks: eliminate                    Bypass If IP Threshold Exceeded? n
                                                         RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload                               Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
Enable Layer 3 Test? n                                  Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n                  Alternate Route Timer(sec): 6
```



Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the 320HD SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```

add trunk-group 10                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 10                                     Group Type: sip                                     CDR Reports: y
  Group Name: To SES                                 COR: 1                                     TN: 1                                     TAC: 1010
  Direction: two-way                                 Outgoing Display? n
  Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                                     Auth Code? n
                                               Signaling Group: 10
                                               Number of Members: 60
  
```

On Page 3 of the trunk group form, set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

```

add trunk-group 10                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n                                 Measured: none
                                               Maintenance Tests? y
                                               Numbering Format: public
                                               UUI Treatment: service-provider
                                               Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
Show ANSWERED BY on Display? y
  
```

Configure the **Public/Unknown Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' and whose calls are routed over SIP trunk group '10' have the number sent to the far-end for display purposes.

```

change public-unknown-numbering 0                     Page 1 of 2
                                     NUMBERING - PUBLIC/UNKNOWN FORMAT
Ext Ext      Trk      CPN      Total
Len Code     Grp(s)   Prefix  CPN
Len          Len
5  7         10         5
Total Administered: 1
Maximum Entries: 9999
  
```

### 4.3. Configure Stations

Use the **add station** command to add a station for each 320HD SIP Phone to be supported. Use *9630SIP* for the **Station Type** and include the **Coverage Path** for voice mail, if applicable. The **Name** field is optional. Use the default values for the other fields on Page 1.

```
add station 78001                                     Page 1 of 6
                                                    STATION
Extension: 78001                                     Lock Messages? n          BCC: 0
  Type: 9630SIP                                     Security Code:            TN: 1
  Port: IP                                           Coverage Path 1: 20    COR: 1
  Name: 320HD 78001                                Coverage Path 2:         COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
                                                    Time of Day Lock Table:
  Loss Group: 19                                     Message Lamp Ext: 78001
  Display Language: english                          Button Modules: 0
  Survivable COR: internal
  Survivable Trunk Dest? y                           IP SoftPhone? n
```

On Page 2, set the **MWI Served User Type** field to the appropriate value to allow MWI notifications to be sent to the 320 HD SIP Phone.

**Note:** Shuffling was disabled on the IP Network Region form which supersedes the setting of the **Direct IP-IP Audio Connections** field on Page 2 of the Station form. However, Shuffling could be disabled on a station basis if the customer does not want to disable Shuffling for the entire IP network region. For this solution, Shuffling should be disabled as mentioned earlier.

```
change station 78001                                 Page 2 of 6
                                                    STATION
FEATURE OPTIONS
  LWC Reception: spe
  LWC Activation? y
  CDR Privacy? n
  Bridged Call Alerting? n
  Active Station Ringing: single
  H.320 Conversion? n
  MWI Served User Type: qsig-mwi
  Emergency Location Ext: 78001
  Coverage Msg Retrieval? y
  Auto Answer: none
  Data Restriction? n
  Idle Appearance Preference? n
  Bridged Idle Line Preference? n
  Per Station CPN - Send Calling Number?
  EC500 State: enabled
  Coverage After Forwarding? s
  Direct IP-IP Audio Connections? y
  Always Use? n IP Audio Hairpinning? n
```

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 78001) to the same SIP Enablement Services Communication Manager extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group between Communication Manager and SIP Enablement Services. The SIP trunk group was configured in **Section 4.2**. The **Configuration Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 78001                               Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
```

Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
78001	OPS	-		78001	10	1	

On Page 2, change the **Call Limit** to match the number of *call-appr* entries in the station form. Also, verify that **Mapping Mode** is set to *both* (the default value for a newly added station).

```
change off-pbx-telephone station-mapping 78001                               Page 2 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
```

Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location
78001	OPS	3	both	all	none	

## 5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of Avaya Aura™ SIP Enablement Services. SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in an Internet browser, where *<ip-addr>* is the IP address of SIP Enablement Services. Log in with the appropriate credentials and then select the Administration → SIP Enablement Services from the next screen (not shown here). The main screen is displayed as shown below.

The screenshot shows the Avaya Integrated Management SIP Server Management web interface. The header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the text 'This Server: [1] ses1'. A navigation menu on the left lists various system components. The main content area displays a 'Top' section with a list of management tasks and their descriptions.

Top	
<b>Manage Users</b>	Add and delete Users.
<b>Manage Address Map Priorities</b>	Adjust Address Map Priorities.
<b>Manage Adjunct Systems</b>	Add and delete Adjunct Systems.
<b>Manage Event Aggregators</b>	Add/Delete Event Aggregators.
<b>Certificate Management</b>	Manage Certificates.
<b>Manage Conferencing</b>	Add and delete Conference Extensions.
<b>Manage Emergency Contacts</b>	Add and delete Emergency Contacts.
<b>Export Import to ProVision</b>	Export and import data using ProVision on this host.
<b>Manage Hosts</b>	Add and delete Hosts.
<b>IM logs</b>	Download IM Logs.
<b>Manage Communication Manager Servers</b>	Add and delete Communication Manager Servers.
<b>Manage Communication Manager Extensions</b>	Add and delete Communication Manager Extensions.
<b>Server Configuration</b>	View Properties of the system.
<b>Manage SIP Phone Settings</b>	Add/Delete Phone Settings
<b>Manage Survivable Call Processors</b>	Add and delete Survivable Call Processors.
<b>System Status</b>	View System Status.
<b>Trace Logger</b>	Manage SIP Trace Logs.
<b>Manage Trusted Hosts</b>	Add and delete Trusted Hosts.

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. In the **System Properties** screen, enter the domain name assigned to the Avaya SIP-based network and the SIP License Host. For the **SIP License Host** field, enter the IP address of the SIP Enablement Services server that is running the WebLM application and has the associated license file installed. This entry should always correspond to the localhost unless the WebLM server is not co-resident with this server. After configuring the **System Properties** screen, click the **Update** button.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes 'Help' and 'Exit' on the left, and 'This Server: [1] ses1' on the right. A left-hand navigation pane lists various system management options, with 'Server Configuration' expanded to show 'System Properties' as the selected option. The main content area is titled 'View System Properties' and displays the following configuration details:

- SES Version: SES-5.2.1.0-016.4
- System Configuration: Simplex
- Host Type: SES combined home-edge
- SIP Domain\*: 
  - Note that the DNS domain is avaya.com
  - If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com
- SIP License Host\*:
- DiffServ/TOS Parameters**
  - Call Control PHB Value\*:
- 802.1 Parameters**
  - Priority Value\*:
  - Management System Access Login:
  - Management System Access Password:
  - DB Log Level:  (dropdown menu)

An **Update** button is located at the bottom of the configuration area.

After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, verify the **Host Type** field. In this example, the host server was configured as an *SES combined home/edge* during the initial setup. The default values for the other fields may be used as shown below.

**AVAYA** Integrated Management SIP Server Management  
 Help Exit This Server: [1] ses1

**Edit Host**

Host IP Address\*

Profile Service Password\*

Host Type SES combined home-edge

Parent none

Listen Protocols  UDP  TCP  TLS

Link Protocols  UDP  TCP  TLS

Access Control Policy (Default)  Allow All  Deny All

Emergency Contacts Policy  Allow  Deny

Minimum Registration (seconds)  Registration Expiration Timer (seconds)\*

Subscription Expiration Timer (seconds)\*

Line Reservation Timer (seconds)

Outbound Routing Allowed  Internal  External

OutboundProxy  Port   UDP  TCP  TLS

Outbound Direct Domains

Default Ringer Volume\*  Default Ringer Cadence

Default Receiver Volume\*  Default Speaker Volume\*

VMM Server Address

VMM Server Port  VMM Report Period

Fields marked \* are required.

**Update**

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Communication Manager and SIP Enablement Services. The screen below shows the Edit Communication Manager Server Interface screen since the server has already been added. In this screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., devcon13).
- Select the home server in the **Host** field.
- Select *TLS* (Transport Link Security) for the **SIP Trunk Link Type**.
- Enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field.

Refer to [2] for additional information on configuring the remaining fields.

The screenshot displays the Avaya Integrated Management SIP Server Management web interface. The page title is "Edit Communication Manager Server Interface". The interface includes a navigation menu on the left with categories like "Users", "Adjunct Systems", "Aggregator", "Certificate Management", "Conferences", "Emergency Contacts", "Export/Import to ProVision", "Hosts", "IM logs", "Communication Manager Servers", "Communication Manager Extensions", "Server Configuration", "SIP Phone Settings", "Survivable Call Processors", "System Status", "Trace Logger", and "Trusted Hosts". The main content area contains the following configuration fields:

- Communication Manager Server Interface Name\***: devcon13
- Host**: 10.32.24.100
- SIP Trunk Link Type**:  TCP  TLS
- SIP Trunk IP Address\***: 10.32.24.20
- Communication Manager Server Admin Address\***: 10.32.24.10 (see Help)
- Communication Manager Server Admin Port\***: 5022
- Communication Manager Server Admin Login\***: interop
- Communication Manager Server Admin Password\***: [masked]
- Communication Manager Server Admin Password Confirm\***: [masked]
- SMS Connection Type**:  SSH  Telnet  Not Available

Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.

Fields marked \* are required.

Add a user for each 320HD SIP Phone registering with SIP Enablement Services. In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** field. Enter a user password in the **Password** and **Confirm Password** fields. In the **Host** field, select the SIP Enablement Services server hosting the domain (*avaya.com*) for this user. Enter the **First Name** and **Last Name** of the user. To associate the extension for this user with a Communication Manager extension, select the **Add Communication Manager Extension** checkbox. Calls to and from this user will always be routed through Communication Manager over the SIP trunk. The **Add Communication Manager Extension** screen is displayed next after adding this user profile by clicking on the **Add** button.

**AVAYA** Integrated Management SIP Server Management  
 Help Exit This Server: [1] ses1

**Add User**

Primary Handle\* 78001  
 User ID  
 Password\* ●●●●●●  
 Confirm Password\* ●●●●●●  
 Host\* 10.32.24.100  
 First Name\* AudioCodes  
 Last Name\* 78001  
 Address 1 211 Mount Airy Rd  
 Address 2  
 Office  
 City Basking Ridge  
 State NJ  
 Country USA  
 Zip 07920  
 Survivable Call Processor none  
 Add Communication Manager Extension   
 Fields marked \* are required.

**Add**

**Top**  
 Users  
 Add  
 Default Profile  
 Delete  
 Edit  
 List  
 Password  
 Search  
 Manage All Registered Users  
 Search Registered Devices  
 Search Registered Users  
 Address Map Priorities  
 Adjunct Systems  
 Aggregator  
 Certificate Management  
 Conferences  
 Emergency Contacts  
 Export/Import to ProVision  
 Hosts  
 IM logs  
 Communication Manager Servers  
 Communication Manager Extensions



In the **Add Communication Manager Extension** screen, enter the **Extension** configured in Communication Manager for the previously added user. Usually, the Communication Manager extension and the user extension are the same. Click the **Add** button.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. At the top left is the Avaya logo. The top right shows the title 'Integrated Management SIP Server Management' and the server status 'This Server: [1] ses1'. Below the title is a navigation bar with 'Help' and 'Exit' links. A left-hand menu is visible, listing various user management options under the 'Users' category. The main content area is titled 'Add Communication Manager Extension' and contains a form for adding an extension for user 78001. The form includes an 'Extension' text box with the value '78001' and a 'Communication Manager Server' dropdown menu with 'devcon13' selected. A note below the form states 'Fields marked \* are required.' and an 'Add' button is positioned at the bottom left of the form area.

**AVAYA** Integrated Management SIP Server Management  
This Server: [1] ses1

Help Exit

Top  
Users  
Add  
Default Profile  
Delete  
Edit  
List  
Password  
Search  
Manage All Registered Users  
Search Registered Devices  
Search Registered Users  
Address Map Priorities

### Add Communication Manager Extension

Add Communication Manager extension for user 78001.

Extension

Communication Manager Server

Fields marked \* are required.

**Add**

## 6. Configure AudioCodes 320HD SIP Phone

The configuration of the 320HD SIP Phone was performed via the phone's menu-driven LCD user interface and its embedded Web interface. The phone's LAN connection interface was initially configured via the phone's LCD screen. To access the **Network Settings** menu, click the MENU key on the phone and navigate to Administration→Network Settings to configure the IP parameters for the phone. A valid password will be required. The rest of the configuration was performed through the phone's embedded Web interface. Refer to [4] for additional information on configuring the 320HD SIP Phone.

From an internet browser, enter `http://<ip-addr>` in the URL field, where `<ip-addr>` is the phone's IP address. Navigate to the **Quick Setup** screen shown below. The **LAN Setup** section was previously configured through phone's LCD screen. In the **SIP Proxy and Registrar** section, set the **Use SIP Proxy** field to *Enable* and set the **Proxy IP Address or Host Name** field to the IP address of SIP Enablement Services. Configure the **Proxy Port** field with the appropriate port value (i.e., 5060). The **Line Settings** section should be configured with the phone's extension and password that were configured for the user on SIP Enablement Services. This section provides the authentication information required to register with SIP Enablement Services. Click **Submit**.

The screenshot displays the 'Quick Setup' web interface for an AudioCodes 320HD SIP phone. The interface is organized into three main sections:

- LAN Setup:** This section is expanded and shows the following fields:
  - IP Type:  Static IP  Automatic IP (DHCP)
  - IP Address:
  - Subnet Mask:
  - Default Gateway Address:
  - Primary DNS:
  - Secondary DNS:
- SIP Proxy and Registrar:** This section is expanded and shows the following fields:
  - Use SIP Proxy:
  - Proxy IP Address or Host Name:
  - Proxy Port:
  - Use SIP Proxy IP and Port for Registration:
  - Use SIP Registrar:
- Line Settings:** This section is expanded and shows the following fields:
  - Line Activate:
  - User ID:
  - Authentication User Name:
  - Authentication Password:

A 'Submit' button is located in the bottom right corner of the form.

Navigate to **Voice Over IP**→**Signaling Protocols** and set the **Gateway Name** field to the SIP domain (e.g., *avaya.com*). All of the other fields should be properly set based on the **Quick Setup** configuration above. Click **Submit**.

The screenshot shows the AudioCodes 320HD web interface. The top navigation bar includes the AudioCodes logo, the model number 320HD, and links for Home and Log Off. The main interface is divided into three tabs: Configuration, Management, and Status & Diagnostics. The left sidebar contains a tree view of configuration options, with 'Voice Over IP' expanded to show 'Signaling Protocols'. The main content area is titled 'Signaling Protocol' and contains two sections:

- SIP General**
  - SIP Transport Protocol: UDP
  - SIP Local Port: 5060
  - Gateway Name: avaya.com
  - PRACK Mode: Enable
  - Enable RPORT: Enable
  - Include PTIME in SDP: Disable
  - Enable Keep Alive using OPTIONS: Disable
  - Connect Media on 180 Response: Disable
- SIP Proxy and Registrar**
  - Use SIP Proxy: Enable
  - Proxy IP Address or Host Name: 10.32.24.100
  - Proxy Port: 5060
  - Maximum Number of Authentication Retries: 4
  - Use SIP Proxy IP and Port for Registration: Enable
  - Use SIP Registrar: Disable
  - Registration Expires: 3600 Seconds
  - Use SIP Outbound Proxy: Disable
  - Use Redundant Proxy: Disable

A 'Submit' button is located at the bottom right of the configuration area.

Navigate to **Voice Over IP**→**Dialing** and set the **Digit Map** field to **xxxxx** to allow 5-digit extensions to be dialed without waiting for an inter-digit timeout to expire. This corresponds to the dial plan used during the compliance test. Click **Submit**.

The screenshot shows the 'Dialing' configuration page in the AudioCodes web interface. The left sidebar contains a tree view with 'Voice Over IP' expanded to 'Dialing'. The main content area is titled 'Dialing' and contains two sections: 'Dialing Parameters' and 'Automatic Dialing'.

**Dialing Parameters:**

- Dialing Timeout: 5 Seconds
- Phone Number Length: 19 Digits
- Enable Dialing Complete Key: Enable
- Dialing Complete Key: #
- Dial Tone Timeout: 30 Seconds
- Reorder Tone Timeout: 40 Seconds
- No Answer Call Timeout: 60 Seconds
- Howler Tone Timeout: 120 Seconds
- Secondary Dial Tone: Enable
- Secondary Dial Tone Key: 9
- DTMF Transport Mode: RFC 2833
- Digit Map: xxxxx
- Dial Plan: (empty)

**Automatic Dialing:**

- Activate: Disable

A 'Submit' button is located at the bottom right of the configuration area.

The following screen simply shows the codecs supported by the endpoint. No additional configuration is required here.

The screenshot shows the 'Media Streaming' configuration page in the AudioCodes web interface. The left sidebar contains a tree view with 'Voice Over IP' expanded to 'Media Streaming'. The main content area is titled 'Media Streaming' and contains three sections: 'Media Streaming Parameters', 'Quality of Service Parameters', and 'Codecs'.

**Media Streaming Parameters:**

- RTP Port Range - Contiguous Series of 4 Ports Starting From: 4000
- DTMF Relay RFC 2833 Payload Type: 101

**Quality of Service Parameters:**

- Type of Service (ToS): 0xb8 Hex

**Codecs:**

Codec Priority	Codec Type	Packetization Time (milliseconds)
1st Codec	G.722	20
2nd Codec	G.711, 64 Kbps, μ-Law	20
3rd Codec	G.711, 64 Kbps, A-Law	20
4th Codec	G.729, 8 Kbps	20
5th Codec	G.723, 5.3/6.3 Kbps	30

**G.723 Bitrate:**

- G.723 Bitrate: High

A 'Submit' button is located at the bottom right of the configuration area.

Navigate to **Voice Over IP→Services** to enable MWI. Under **Message Waiting Indication (MWI)**, set the **Activate** field to *Enable* and set the **Voice Mail Number** field to the phone's extension.

**Note:** The **Voice Mail Number** field is usually set to the voicemail pilot number so that when the Voicemail button on the phone is pressed, the voicemail system is called for the user to check or send voicemail messages. However, for MWI to work, this field should be configured as specified above. The voicemail system can still be dialed manually by the user.

Next, set the **Subscribe To MWI** field to *Enable*, set the **MWI Server IP Address or Host Name** field to the IP address of SIP Enablement Services, and increase the **MWI Subscribe Expiry Time** field to a minimum of 3600. Click **Submit**.

The screenshot shows the AudioCodes configuration interface. The top navigation bar includes the AudioCodes logo, the user '320HD', and links for 'Home' and 'Log Off'. The main interface is divided into three tabs: 'Configuration', 'Management', and 'Status & Diagnostics'. The 'Configuration' tab is active, and a left-hand sidebar contains a tree view of configuration categories: Quick Setup, Personal Settings, Network Connections, Voice Over IP, Signaling Protocols, Dialing, Media Streaming, Voice, Line Settings, Services, Volume Settings, and Advanced Applications. The 'Services' page is displayed, showing several configuration sections: 'Application Server' (Application Server Type: Generic), 'Call Waiting' (Activate: Enable, Call Waiting SIP Reply: Queued), 'Call Forward' (Activate: Enable, Call Forward Type: No Reply, Forward on No Reply Timeout: 6 Seconds), '3-Way Conference' (3-Way Conference Mode: Local), and 'Message Waiting Indication (MWI)' (Activate: Enable, Voice Mail Number: 78001, Subscribe To MWI: Enable, MWI Server IP Address or Host Name: 10.32.24.100, MWI Server Port: 5060, MWI Subscribe Expiry Time: 3600 Seconds). A 'Submit' button is located at the bottom right of the configuration area.

## 7. General Test Approach and Test Results

To verify interoperability of the AudioCodes 300HD SIP IP Phone Series with Communication Manager and SIP Enablement Services, calls were made between 320HD SIP Phones and Avaya SIP, H.323, and digital stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using phone buttons and FNEs. The PBX features listed in **Section 1.1** were covered.

The following observations were noted during the compliance test:

- Shuffling should be disabled for calls to the 300HD SIP IP Phone Series to prevent issues with Hold and Conference.
- The Voice Mail Number on the 300HD SIP IP Phone Series should be set to the phone's extension for MWI to work. As a result, the user will have to manually dial the voicemail pilot number instead of simply pressing the Voicemail button on the phone.
- When a 320HD SIP Phone calls another phone with Call Forwarding enabled, there is no indication on the 320HD SIP Phone that the call was forwarded.

## 8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the 320HD SIP Phones have successfully registered with SIP Enablement Services.
2. Verify basic telephony features by establishing calls between a 320HD SIP Phone and another phone.
3. Call a 320HDSIP phone that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., Voicemail button) illuminates. Call the voicemail system and retrieve voice messages. Verify that after hearing all messages, that the message waiting indicator is extinguished.

## 9. Conclusion

These Application Notes have described the administration steps required to integrate the AudioCodes 300HD SIP IP Phone Series with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. The 320HD SIP Phone successfully registered with SIP Enablement Services and basic telephony features were verified.

## 10. References

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

- [1] *Administering Avaya Aura<sup>TM</sup> Communication Manager*, Release 5.2, May 2009, Issue 5.0, Document Number 03-300509.
- [2] *SIP Support in Avaya Aura<sup>TM</sup> Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [3] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura<sup>TM</sup> SIP Enablement Services*, May 2009, Issue 7.0, Document Number 03-600768.
- [4] *AudioCodes Administrator Manual 320HD IP Phone*, Version 1.2.2, April 2010, Document Number LTRT-08105.

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