



Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes 300HD SIP IP Phone Series with Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager Feature Server - 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™ Session Manager and Avaya Aura™ Communication Manager Feature Server. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with Session Manager, 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™ Session Manager and Avaya Aura™ Communication Manager Feature Server. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

These Application Notes assume that Communication Manager and Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult the appropriate document in the reference section at the end of this document.

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the AudioCodes 320HD SIP Phone with Session Manager.
- Calls between 320HD SIP phones and Avaya SIP, H.323, and digital stations.
- G.711, G.729A, and G.722-74K 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 that includes the following Avaya products:

- Communication Manager is running on an Avaya S8510 Server with a G450 Media Gateway, serving as a Feature Server for SIP endpoints.
- An Avaya S8720 Server with a G650 Media Gateway serves as an Access Element supporting H.323 and digital telephones.
- Session Manager interconnects the Feature Server and Access Element via SIP trunks and acts as a Registrar/Proxy for SIP telephones.
- Avaya Aura™ System Manager is used to configure Session Manager.
- Avaya Modular Messaging provides voice mail service.

In addition, three AudioCodes 320HD SIP Phones registered with Session Manager and were configured as Off-PBX Stations (OPS) on the Communication Manager Feature Server.

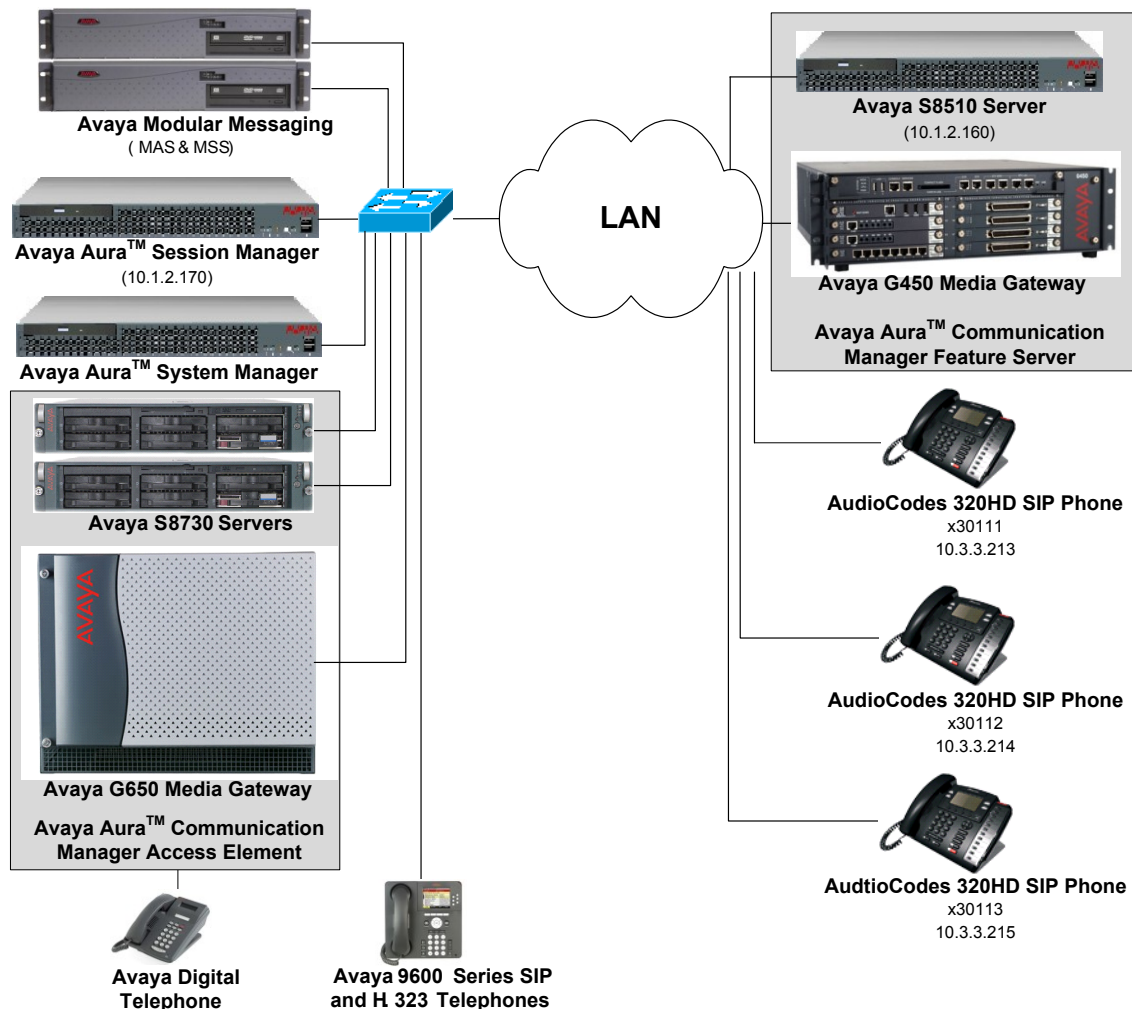


Figure 1: AudioCodes 320HD SIP Phones with Avaya SIP Solution

2.1. SIP Call Flows

The AudioCodes 320HD SIP Phone originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to the Communication Manager Feature Server for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for an H.323 or digital telephone, Communication Manager routes the call back to Session Manager for delivery to the Communication Manager Access Element supporting H.323 and digital endpoints.

For a call arriving at Communication Manager Feature Server that is destined for the AudioCodes 320HD SIP Phones, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the AudioCodes 320HD SIP Phones.

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers and G650 Media Gateway (Access Element)	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 2 (Patch 18111)
Avaya S8510 Servers with G450 Media Gateway (Feature Server)	Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4) with Patch 18151
Avaya Aura™ Session Manager	5.2.2
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 Avaya Aura™ Communication Manager Feature Server

This section describes the steps for configuring the AudioCodes 320HD SIP Phone as an Off-PBX Station (OPS) and configuring a SIP trunk between the Communication Manager Feature Server and Session Manager. **Section 4.3** covers the station configuration for the 320HD SIP Phones. See [2] for additional information on configuring SIP support on Communication Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: These Application Notes do not cover call routing to the Communication Manager Access Element.

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 for the number of SIP endpoints that will be deployed.

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

G3 Version: V15                                     Software Package: Standard
Location: 1                                           RFA System ID (SID): 1
Platform: 12                                         RFA Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 44000 316
                                Maximum Stations: 36000 86
                                Maximum XMOBILE Stations: 0 0
Maximum Off-PBX Telephones - EC500: 10 0
Maximum Off-PBX Telephones - OPS: 200 80
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 10
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	450	100
Maximum Concurrently Registered IP Stations:	18000	0
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:	5	0
Maximum Video Capable H.323 Stations:	0	0
Maximum Video Capable IP Softphones:	5	0
Maximum Administered SIP Trunks:	300	130
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	5	1
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 S8510 Server processor and the Session Manager 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	
SM1	10.1.2.170	
default	0.0.0.0	
procr	10.1.2.160	
(3 of 3 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 Session Manager. 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 G450 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 Session Manager. 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: HQ CM and SIP Phones
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: 65535
DIFFSERV/TOS PARAMETERS      RTCP Reporting Enabled? y
  Call Control PHB Value: 46      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: 6
  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, G.729A, and G.722-64K, 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 Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- Set the **Group Type** field to *sip*.
- The **Transport Method** field was set to *tcp*.
- Specify the S8510 Server processor and the Session Manager 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 *5060* 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 Session Manager 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 so that only calls using this signaling group are impacted.
- 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 60		Page 1 of 1
SIGNALING GROUP		
Group Number: 60	Group Type: sip	
	Transport Method: tcp	
IMS Enabled? y		
Near-end Node Name: procr	Far-end Node Name: SM1	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: avaya.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early Media? n	
	Alternate Route Timer(sec): 10	

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 60		Page 1 of 21	
TRUNK GROUP			
Group Number: 60	Group Type: sip	CDR Reports: y	
Group Name: SMI	COR: 1	TN: 1	TAC: 160
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
Signaling Group: 60			
Number of Members: 100			

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 60		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private			
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 '3' and whose calls are routed over SIP trunk group "60" have the number sent to the far-end for display purposes.

change private-numbering 0		Page 1 of 2	
NUMBERING - PRIVATE FORMAT			
Ext Len	Ext Code	Trk Grp (s)	Private Prefix
5	3	60	
Total Len	Total Administered: 1		
5	Maximum Entries: 540		

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. The SIP station can also be configured automatically by Session Manager as described in **Section 5.8**.

add station 30111		Page 1 of 6
STATION		
Extension: 30111	Lock Messages? n	BCC: 0
Type: 9630SIP	Security Code:	TN: 1
Port: IP	Coverage Path 1: 60	COR: 1
Name: AudioCodes 30111	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
	Time of Day Lock Table:	
Loss Group: 19		
	Message Lamp Ext: 30111	
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.

add station 30111		Page 2 of 6
STATION		
FEATURE OPTIONS		
LWC Reception: spe	Coverage Msg Retrieval? y	
LWC Activation? y	Auto Answer: none	
CDR Privacy? n	Data Restriction? n	
	Idle Appearance Preference? n	
	Bridged Idle Line Preference? n	
Bridged Call Alerting? n		
Active Station Ringing: single		
H.320 Conversion? n	Per Station CPN - Send Calling Number?	
	EC500 State: enabled	
MWI Served User Type: sip-adjunct	Coverage After Forwarding? s	
	Direct IP-IP Audio Connections? y	
Emergency Location Ext: 30111	Always Use? n IP Audio Hairpinning? n	
Precedence Call Waiting? y		

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 30111) to the same extension configured in Session Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group between the Communication Manager Feature Server and Session Manager. 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 30111							Page 1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode
30111	OPS	-		30111	60	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 30111							Page 2 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station Extension	Appl Name	Call Limit	Mapping Mode	Calls Allowed	Bridged Calls	Location	
30111	OPS	3	both	all	none		

5. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and the Communication Manager Feature Server
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Application Sequence
- Define Feature Server as Administrable Entity
- Session Manager, corresponding to the Avaya Aura™ Session Manager Server to be managed by Avaya Aura™ System Manager
- Add SIP Users

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 Avaya Aura™ System Manager. Log in with the appropriate credentials and accept the Copyright Notice.

Note: These Application Notes do not cover the configuration of the Communication Manager Access Element, but the configuration would be similar to that of the Feature Server which is covered.

5.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **SIP Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.

The screenshot shows the Avaya Aura System Manager 5.2 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 5.2", and a welcome message for user "admin" with a "Log off" link. A red breadcrumb trail indicates the path: Home / Network Routing Policy / SIP Domains. On the left, a sidebar menu lists various management categories, with "SIP Domains" highlighted under "Network Routing Policy". The main content area is titled "Domain Management" and contains a table with one item, "avaya.com". The table has columns for Name, Type (set to "sip"), Default (unchecked), and Notes. Below the table, a red asterisk indicates "Input Required". "Commit" and "Cancel" buttons are present at the top right and bottom right of the form area.

Name	Type	Default	Notes
* avaya.com	sip	<input type="checkbox"/>	

5.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *Basking Ridge HQ* location, which includes the Avaya Aura™ Communication Manager Feature Server and Avaya Aura™ Session Manager. Click **Commit** to save the Location definition.

The screen below shows the information for Communication Manager Feature Server in the sample configuration.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Mar. 18, 2010 5:05 PM [Help](#) | [Log off](#)

[Home](#) / [Network Routing Policy](#) / [Locations](#) / [Location Details](#)

Location Details Commit Cancel

General

* **Name:**

Notes:

Managed Bandwidth:

* **Average Bandwidth per Call:**

* **Time to Live (secs):**

Location Pattern

Add Remove

4 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.1.2.*	<input type="text"/>

Select : [All](#), [None](#) (0 of 4 Selected)

* **Input Required** Commit Cancel

Shortcuts

- [Change Password](#)
- [Help for Locations Details fields](#)
- [Help for Committing configuration changes](#)

5.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and the S8510 Server associated with the Feature Server.

5.3.1. Avaya Aura™ Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Specify *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

The screenshot displays the Avaya Aura™ System Manager 5.2 web interface. The top header includes the Avaya logo, the product name, and a welcome message for the 'admin' user. A red navigation bar contains the breadcrumb 'Home / Network Routing Policy / SIP Entities / SIP Entity Details'. On the left, a sidebar menu lists various management categories, with 'SIP Entities' highlighted under 'Network Routing Policy'. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form contains several fields: 'Name' (text input with 'SM1'), 'FQDN or IP Address' (text input with '10.1.2.170'), 'Type' (dropdown menu set to 'Session Manager'), 'Notes' (text area), 'Location' (dropdown menu set to 'BaskingRidge HQ'), 'Outbound Proxy' (text input), 'Time Zone' (dropdown menu set to 'America/New_York'), and 'Credential name' (text input). At the bottom, there is a 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located in the top right corner of the form area.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

Port

Add Remove

4 Items | Refresh Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	
<input type="checkbox"/>	5060	UDP	avaya.com	
<input type="checkbox"/>	5061	TLS	avaya.com	
<input type="checkbox"/>	5070	TCP	avocs.contoso.com	

Select : All, None (0 of 4 Selected)

* Input Required Commit Cancel

5.3.2. Avaya Aura™ Communication Manager Feature Server

A SIP Entity must be added for the Communication Manager Feature Server. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the ASM or the signaling interface on the telephony system.
- **Type:** Specify *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 18, 2010 5:05 PM
[Help](#) | [Log off](#)

Home / Network Routing Policy / SIP Entities / SIP Entity Details

Asset Management
Communication System Management
Monitoring
User Management
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
Settings
Session Manager

Shortcuts
Change Password
Help for SIP Entity Details fields
Help for Committing configuration changes

SIP Entity Details
General

Name:
CM Feature Server

FQDN or IP Address:
10.1.2.160

Type:
CM

Notes:

Adaptation:

Location:
BaskingRidge HQ

Time Zone:
America/New_York

Override Port & Transport with DNS SRV:
☐

SIP Timer B/F (in seconds):
4

Credential name:

Call Detail Recording:
none

SIP Link Monitoring
SIP Link Monitoring:
Link Monitoring Enabled

Proactive Monitoring Interval (in seconds):
30

Reactive Monitoring Interval (in seconds):
30

Number of Retries:
1

Entity Links
Add Remove

1 Item Refresh
Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	SM1	TCP	* 5060	CM Feature Server	* 5060	<input checked="" type="checkbox"/>

Select : All, None (0 of 1 Selected)

Commit Cancel

5.4. Add Entity Link

The SIP trunk from Session Manager to the Communication Manager Feature Server is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *SM1_CM Feature Server*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of the Communication Manager Feature Server.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in Section 0 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 18, 2010 5:05 PM

Help | Log off

Home / Network Routing Policy / Entity Links

Entity Links

Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* SM1_CM Feature S	* SM1	TCP	* 5060	* CM Feature Server	* 5060	<input checked="" type="checkbox"/>	

* Input Required

Commit Cancel

5.5. Add Routing Policies

Since the SIP users are register with Session Manager, a routing policy is not required to be defined for the Communication Manager Feature Server.

5.6. Add Application Sequence

Define an application for the Communication Manager Feature Server. Fill in the following fields:

- **SIP Entity:** Select the Communication Manager Feature Server.

Click **Commit** to save the Application definition.

The screenshot shows the Avaya Aura System Manager 5.2 web interface. The top header includes the Avaya logo, the product name "Avaya Aura™ System Manager 5.2", and a user status bar indicating "Welcome, admin" and "Last Logged on at Mar. 18, 2010 5:05 PM". A navigation breadcrumb trail reads "Home / Session Manager / Application Configuration / Application Editor".

On the left is a sidebar menu with categories like Asset Management, Communication System Management, Monitoring, User Management, Network Routing Policy, Security, Applications, and Settings. The "Session Manager" category is expanded, showing sub-items like Session Manager Administration, Network Configuration, Device and Location Configuration, Application Configuration (which is selected), and System Status.

The main content area is titled "Application Editor" and contains the following fields:

- Name:** A text field containing "CM Feature Server (Fred)".
- * SIP Entity:** A dropdown menu with "CM Feature Server" selected.
- Description:** An empty text field.

Below these fields is a section titled "Application Attributes (optional)" containing a table:

Name	Value
Application Handle	<input type="text"/>
URI Parameters	<input type="text"/>

At the bottom of the form, there is a legend for "* Required" and two buttons: "Commit" and "Cancel".

Next, define the Application Sequence for Communication Manager Feature Server as shown below.

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 18, 2010 5:05 PM [Help](#) [Log off](#)

Home / Session Manager / Application Configuration / **Application Sequence Editor**

Application Sequence Editor [Commit](#) [Cancel](#)

Sequence Name

Name

Description

Applications in this Sequence

[Move First](#) [Move Last](#) [Remove](#)

1 Item

<input type="checkbox"/>	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>		CM Feature Server	CM Feature Server	<input checked="" type="checkbox"/>	

Select : All, None (0 of 1 Selected)

Available Applications

3 Items [Refresh](#) [Filter: Enable](#)

<input type="checkbox"/>	Name	SIP Entity	Description
<input checked="" type="checkbox"/>	CM Feature Server	CM Feature Server	

*Required [Commit](#) [Cancel](#)

5.7. Define Feature Server as Administrable Entity

Before adding SIP users, the Communication Manager Feature Server must be added to System Manager as an administrable entity. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify the Communication Manager Feature Server when new SIP users are added.

To define the Feature Server as an administrable entity, select **Entities** on the left and click on the **New** button on the right. Fill in the following fields as follows:

Under *Application*:

- **Name:** Enter an identifier for the Communication Manager Feature Server.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for the Feature Server.

Under *Attributes*:

- **Login / Password:** Enter the login and password used for administration access.

- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Click **Commit** to save the settings.

AVAYA

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 18, 2010 5:05 PM
[Help](#) | [Log off](#)

Home / Applications / Application Management / Applications Details

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▶ Network Routing Policy

▶ Security

▼ Applications

Session Manager 5.2

Other Applications

SMGR

SIP AS 8.0

Entities

▶ Settings

▶ Session Manager

Shortcuts
[Change Password](#)
[Application Instance Fields](#)

Edit CM: S8510-FS

[Application](#) | [Port](#) | [Access Point](#) | [Attributes](#) |
[Expand All](#) | [Collapse All](#)

[Application](#) ▼

* Name

S8510-FS

* Type

CM

Description

S8510 CM as Feature Server

* Node

10.1.2.160

[Port](#) ▶

[Access Point](#) ▶

[Attributes](#) ▼

* Login

admin

Password

●●●●●●

Confirm Password

●●●●●●

Is SSH Connection

☒

* Port

5022

Alternate IP Address

RSA SSH Fingerprint (Primary IP)

RSA SSH Fingerprint (Alternate IP)

JAO; Reviewed:
SPOC 6/23/2010

Solution & Interoperability Test Lab Application Notes
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22 of 35
AudioCodes-SM

5.8. Add SIP Users

Add SIP users corresponding to the AudioCodes 320HD SIP Phone defined in **Section 4.3**. Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature Server when adding a new SIP user.

Enter values for the following required attributes for a new SIP user in the **General** section of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.

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

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Mar. 19, 2010 11:07 AM [Status](#) | [Help](#) | [Log off](#)

Home / User Management / User Management / New User

New User Profile Commit Cancel

General | Identity | Communication Profile | Roles | Override Permissions | Group Membership | Attribute Sets | Default Contact List | Private Contacts |
Expand All | Collapse All

General ▾

* **Last Name:**

* **First Name:**

Middle Name:

Description:

☐ administrator
☐ communication_user
☐ agent
☐ supervisor
☐ resident_expert
☐ service_technician
☐ lobby_phone

User Type:

Identity ▸

Communication Profile ▸

Roles ▸

Shortcuts

- Change Password
- Help for Create User
- Help for New Private Contact
- Help for Edit Private Contact
- Help for Delete Private Contact
- Help for adding contact into contact list
- Help for editing contact from contact list
- Help for deleting contact from contact list

Enter values for the following required attributes for a new SIP user in the *Identity* section of the new user form.

- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., 30111@avaya.com).
- **Authentication Type:** Select *Basic*.
- **SMGR Login Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.
- **Shared Communication Profile Password:** Enter the password which will be by the SIP phone to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

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

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 19, 2010 11:07 AM

Home / User Management / User Management / New User

New User Profile [Commit] [Cancel]

General | Identity | Communication Profile | Roles | Override Permissions | Group Membership | Attribute Sets | Default Contact List | Private Contacts | Expand All | Collapse All

General

Identity

* Login Name: 30111@avaya.com

* Authentication Type: Basic

SMGR Login Password:

* Password:

* Confirm Password:

Shared Communication Profile Password:

Confirm Password:

Localized Display Name: AudioCodes 30111

Endpoint Display Name: AudioCodes 30111

Honorific:

Language Preference: English

Time Zone: Eastern Time (US & Canada)

Scroll down to the *Communication Profile* section and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required fields:

- **Name:** Enter name of communication profile.
- **Default:** Select field to indicate that this is the default profile.

Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *sip*.
- **SubType:** Select *username*.
- **Fully Qualified Address:** Enter extension number and SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

The screenshot displays the Avaya SIP configuration interface. On the left is a sidebar with navigation links: Network Routing Policy, Security, Applications, Settings, Session Manager, and a Shortcuts section with links for password changes and user/contact management. The main area is titled 'Communication Profile' and contains a 'New' button. Below this is a table with one row labeled 'Primary' and a 'Select : None' dropdown. The 'Name' field is set to 'Primary' and the 'Default' checkbox is checked. The 'Communication Address' section has 'New', 'Edit', and 'Delete' buttons. Below is a table with columns 'Type', 'SubType', 'Handle', and 'Domain', currently showing 'No Records found'. The 'Type' dropdown is set to 'sip' and the 'SubType' dropdown is set to 'username'. The 'Fully Qualified Address' field contains '30111' and the domain dropdown is set to 'avaya.com'. 'Add' and 'Cancel' buttons are at the bottom right.

Name
Primary

Select : None

* Name: Primary

Default : ☒

Type	SubType	Handle	Domain
No Records found			

Type: sip

SubType: username

* Fully Qualified Address: 30111 @ avaya.com

Add Cancel

In the *Session Manager* section, specify the Session Manager entity and assign the **Application Sequence** defined in **Section 5.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence.

Communication Profile

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default: ☒

Communication Address

New Edit Delete

Type	SubType	Handle	Domain
sip	username	30111	avaya.com

Select : All, None (0 of 1 Selected)

☒ Session Manager

* Session Manager Instance: SM1

Origination Application Sequence: CM FS App Sequence

Termination Application Sequence: CM FS App Sequence

In the **Communication Profile** section, fill in the following fields:

- **System:** Select the SIP Entity corresponding to the Communication Manager Feature Server.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in the Feature Server.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Station on Unassign of Station:** 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.

Communication Profile

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default: ☒

Communication Address

☒ Session Manager

☒ Station Profile

* System: S8510-FS

Use Existing Stations: ☐

* Extension: 30111

* Template: DEFAULT_9630SIP

Set Type: 9630SIP

Security Code:

* Port: IP

Delete Station on Unassign of Station from User: ☒

5.9. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between Avaya Aura™ System Manager and Avaya Aura™ Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, and fill in the fields as described below and shown in the following screen:

Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Avaya Aura™ Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Avaya Aura™ Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Avaya Aura™ Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Avaya Aura™ Session Manager

Use default values for the remaining fields. Click **Save** to add this Session Manager.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at June 14, 2010 10:23 AM [Help](#) [Log off](#)

Home / Session Manager / Session Manager Administration / Edit Session Manager

Edit Session Manager [Commit](#) [Cancel](#)

General | Security Module | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server | Expand All | Collapse All

General

SIP Entity Name

Description

*Management Access Point Host Name/IP

*Direct Routing to Endpoints

Security Module

SIP Entity IP Address

*Network Mask

*Default Gateway

*Call Control PHB

*QOS Priority

*Speed & Duplex

VLAN ID

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 [5] 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 Session Manager. 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 Session Manager. This section provides the authentication information required to register with Session Manager. 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, SIP Proxy and Registrar, and Line Settings. The LAN Setup section is expanded, showing fields for IP Type (Static IP selected), IP Address (10.3.3.213), Subnet Mask (255.255.255.0), Default Gateway Address (10.3.3.1), Primary DNS (0.0.0.0), and Secondary DNS (0.0.0.0). The SIP Proxy and Registrar section is expanded, showing fields for Use SIP Proxy (Enable), Proxy IP Address or Host Name (10.1.2.170), Proxy Port (5060), Use SIP Proxy IP and Port for Registration (Enable), and Use SIP Registrar (Disable). The Line Settings section is expanded, showing fields for Line Activate (Enable), User ID (30111), Authentication User Name (30111), and Authentication Password (*****). A Submit button is located at the bottom right.

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**.

AudioCodes 320HD Home Log Off

Configuration Management Status & Diagnostics

Signaling Protocol

SIP General

SIP Transport Protocol:	UDP
SIP Local Port:	5060
Gateway Name:	avaya.com
PRACK Mode:	Enable
Enable RPORT:	Enable
Include PTIME in SDP:	Enable
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:	10.1.2.170
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

Submit

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**.

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:		

Automatic Dialing

Activate:	Disable
-----------	---------

Submit

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

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
----------------	------

Submit

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 Session Manager, and increase the **MWI Subscribe Expiry Time** field to a minimum of 3600. 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 left sidebar contains a tree view with categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are links for Quick Setup, Personal Settings, Network Connections, Voice Over IP, Signaling Protocols, Dialing, Media Streaming, Voice, Line Settings, Services, Volume Settings, and Advanced Applications. The main content area is titled 'Services' and contains several expandable sections. The 'Message Waiting Indication (MWI)' section is expanded, showing the following fields: 'Activate' (set to 'Enable'), 'Voice Mail Number' (set to '30111'), 'Subscribe To MWI' (set to 'Enable'), 'MWI Server IP Address or Host Name' (set to '10.1.2.170'), 'MWI Server Port' (set to '5060'), and 'MWI Subscribe Expiry Time' (set to '3600' Seconds). A 'Submit' button is located at the bottom right of the form.

Section	Field	Value
Application Server	Application Server Type	Generic
	Call Waiting	
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
	Message Waiting Indication (MWI)	
Message Waiting Indication (MWI)	Activate	Enable
	Voice Mail Number	30111
	Subscribe To MWI	Enable
	MWI Server IP Address or Host Name	10.1.2.170
	MWI Server Port	5060
	MWI Subscribe Expiry Time	3600 Seconds

7. General Test Approach and Test Results

To verify interoperability of the AudioCodes 300HD SIP IP Phone Series with Communication Manager Feature Server and Session Manager, 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 Session Manager.
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 AuraTM Communication Manager and Avaya AuraTM Session Manager. The 320HD SIP Phone successfully registered with Session Manager and basic telephony features were verified. Noted observations during the compliance test were also covered.

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 AuraTM Communication Manager*, Release 5.2, May 2009, Issue 5.0, Document Number 03-300509.
- [2] *SIP Support in Avaya AuraTM Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [3] *Administering Avaya AuraTM Communication Manager as a Feature Server*, May 2010, Issue 1.3, Release 5.2, Document Number 03-603479.
- [4] *Administering Avaya AuraTM Session Manager*, November 2009, Issue 2, Release 5.2, Document Number 03-603324.
- [5] *AudioCodes Administrator Manual 320HD IP Phone*, Version 1.2.2, April 2010, Document Number LTRT-08105.

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