



**Avaya Solution & Interoperability Test Lab**

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## **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

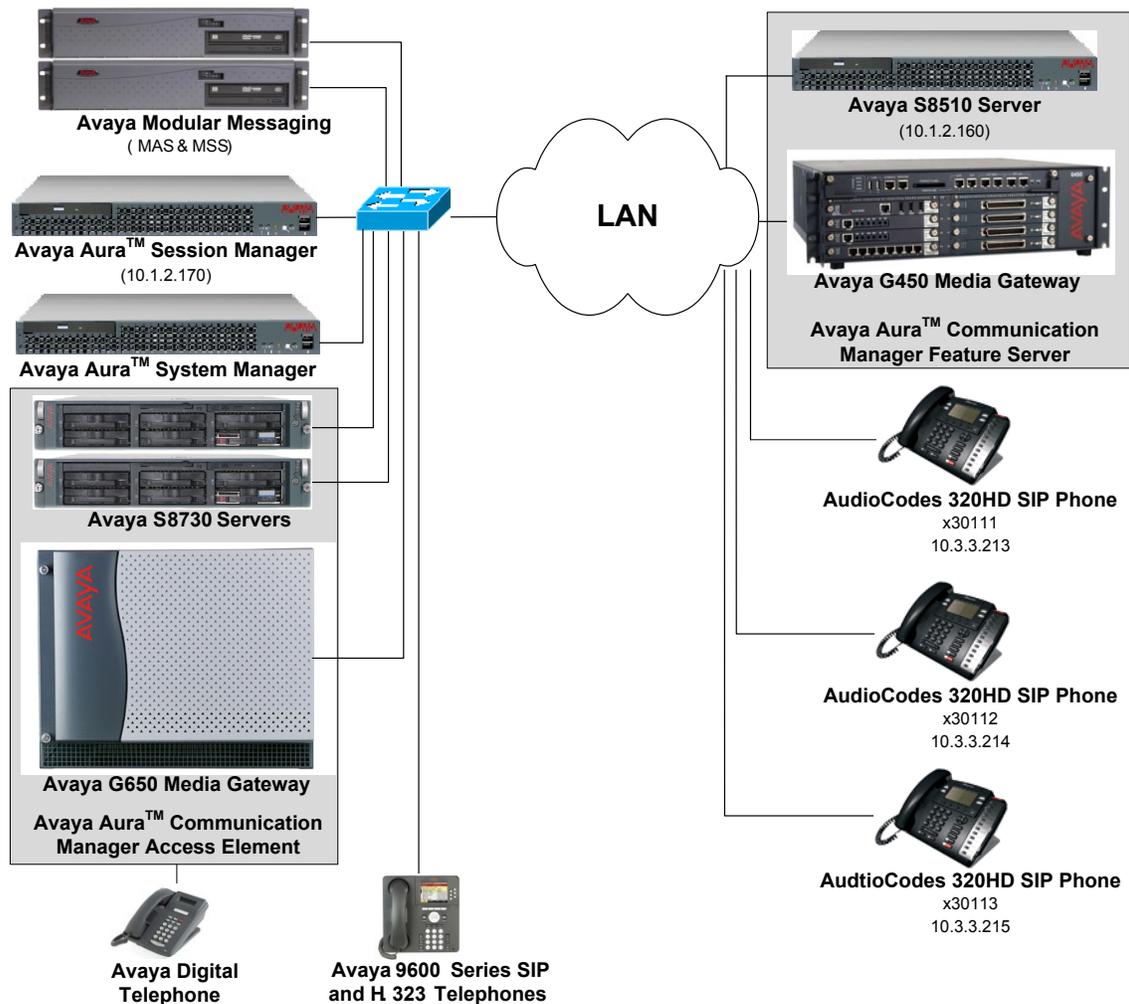
For technical support on the 300HD SIP IP Phone Series contact AudioCodes Customer Support through their website at <http://www.audiocodes.com/support>.

## 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 Manger 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:

<b>Hardware Component</b>	<b>Version</b>
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
                                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): 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
                                                    UI 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 Ext      Trk      Private      Total
Len Code     Grp(s)    Prefix      Len
5  3         60
Total Administered: 1
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
  LWC Activation? y
  CDR Privacy? n
  Bridged Call Alerting? n
  Active Station Ringing: single
  H.320 Conversion? n
  MWI Served User Type: sip-adjunct
  Emergency Location Ext: 30111
  Precedence Call Waiting? y
  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., 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.

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 / SIP Domains

Domain Management

1 Item | Refresh Filter: Enable

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

\* Input Required

## 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.

The screenshot displays the Avaya Aura™ System Manager 5.2 interface. The top navigation bar includes the Avaya logo, the product name, and user information: "Welcome, admin Last Logged on at Mar. 18, 2010 5:05 PM". A red breadcrumb trail shows the path: Home / Network Routing Policy / Locations / Location Details. The left sidebar contains a menu with categories like Asset Management, Communication System Management, Monitoring, User Management, Network Routing Policy (expanded), Security, Applications, Settings, and Session Manager. The main content area is titled "Location Details" and has "Commit" and "Cancel" buttons. It is divided into two sections: "General" and "Location Pattern".

**General**

- \* Name: BaskingRidge HQ
- Notes: ACM & ASM's
- Managed Bandwidth: [ ]
- \* Average Bandwidth per Call: 80 kbit/sec
- \* Time to Live (secs): 3600

**Location Pattern**

[Add] [Remove]

4 Items | Refresh Filter: Enable

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

Select : All, None ( 0 of 4 Selected )

\* Input Required [Commit] [Cancel]

### 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 shows the Avaya Aura™ System Manager 5.2 interface. The top navigation bar includes the Avaya logo, the product name, and a user status message: "Welcome, admin Last Logged on at June 10, 2010 2:55 PM". There are links for "Help" and "Log off". The breadcrumb trail is "Home / Network Routing Policy / SIP Entities / SIP Entity Details". The left sidebar contains a navigation menu with categories: Asset Management, Communication System Management, Monitoring, User Management, and Network Routing Policy (expanded). Under Network Routing Policy, the following items are listed: Adaptations, Dial Patterns, Entity Links, Locations, Regular Expressions, Routing Policies, SIP Domains, SIP Entities (highlighted), Time Ranges, and Personal Settings. The main content area is titled "SIP Entity Details" and has "Commit" and "Cancel" buttons. It is divided into two sections: "General" and "SIP Link Monitoring". The "General" section contains the following fields:

- \* Name: SM1
- \* FQDN or IP Address: 10.1.2.170
- Type: Session Manager (dropdown)
- Notes: (empty text box)
- Location: BaskingRidge HQ (dropdown)
- Outbound Proxy: (empty dropdown)
- Time Zone: America/New\_York (dropdown)
- Credential name: (empty text box)

The "SIP Link Monitoring" section contains:

- SIP Link Monitoring: Use Session Manager Configuration (dropdown)

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.

The screenshot shows a web interface for configuring SIP ports. At the top, there are 'Add' and 'Remove' buttons. Below that, it says '4 Items | Refresh' and 'Filter: Enable'. A table with the following columns is displayed: Port, Protocol, Default Domain, and Notes. The first row is highlighted with a red box and contains: Port: 5060, Protocol: TCP, Default Domain: avaya.com, and an empty Notes field. Below the table, it says 'Select : All, None ( 0 of 4 Selected )'. At the bottom, there is a '\* Input Required' warning and 'Commit' and 'Cancel' buttons.

<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	

### 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.

- ▶ 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:  ▶

\* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:  ▶

Time Zone:

Override Port & Transport with DNS SRV:

\* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

**SIP Link Monitoring**

SIP Link Monitoring:

\* Proactive Monitoring Interval (in seconds):

\* Reactive Monitoring Interval (in seconds):

\* Number of Retries:

**Entity Links**

1 Item | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	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 )

## 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 Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Mar. 16, 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 Server	* 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 interface. The top navigation bar includes the Avaya logo, the product name "Avaya Aura™ System Manager 5.2", and a user status "Welcome, admin Last Logged on at Mar. 18, 2010 5:05 PM" with links for "Help" and "Log off". A red breadcrumb trail reads "Home / Session Manager / Application Configuration / Application Editor".

The left sidebar contains a tree view of system management categories, with "Application Configuration" expanded to show "Applications".

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

- Name:** CM Feature Server (Fred)
- \* SIP Entity:** CM Feature Server (dropdown menu)
- Description:** (empty text box)

Below these fields is a section for "Application Attributes (optional)" which includes a table:

Name	Value
Application Handle	(empty text box)
URI Parameters	(empty text box)

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

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

The screenshot shows the Avaya Aura System Manager 5.2 interface. The top header includes the Avaya logo, the system name 'Avaya Aura™ System Manager 5.2', and a welcome message for user 'admin' last logged on at Mar. 18, 2010 5:05 PM. The navigation menu on the left is expanded to 'Session Manager', which includes 'Application Configuration' and 'Application Sequences'. The main content area is titled 'Application Sequence Editor' and contains the following elements:

- Sequence Name:** A text box containing 'CM FS App Sequence'.
- Description:** An empty text box.
- Applications in this Sequence:** A table with one row:
 

Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
1	CM Feature Server	CM Feature Server	<input checked="" type="checkbox"/>	
- Available Applications:** A table with three items, including 'CM Feature Server':
 

Name	SIP Entity	Description
CM Feature Server	CM Feature Server	

## 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 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**
Commit
Cancel

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

---

Application ▼

\* Name

\* Type

Description

\* Node

---

Port ▶

---

Access Point ▶

---

Attributes ▼

\* Login

Password

Confirm Password

Is SSH Connection

\* Port

Alternate IP Address

RSA SSH Fingerprint (Primary IP)

RSA SSH Fingerprint (Alternate IP)

## 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.

The screenshot shows the Avaya Aura System Manager 5.2 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 5.2', and a user status 'Welcome, admin Last Logged on at Mar. 19, 2010 11:07 AM'. Below the navigation bar is a breadcrumb trail: 'Home / User Management / User Management / New User'. The left sidebar contains a tree view of system management options, with 'User Management' expanded to show 'New User'. The main content area is titled 'New User Profile' and has 'Commit' and 'Cancel' buttons. The 'General' tab is selected, showing the following fields: 'Last Name' (30111), 'First Name' (AudioCodes), 'Middle Name', 'Description', and 'User Type' (with radio buttons for administrator, communication\_user, agent, supervisor, resident\_expert, service\_technician, and lobby\_phone). Below the 'General' tab are sections for 'Identity', 'Communication Profile', and 'Roles', each with a right-pointing arrow.

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.

The screenshot shows the Avaya Aura System Manager 5.2 interface. The top navigation bar includes the Avaya logo, the system name, and a user status message: "Welcome, admin Last Logged on at Mar. 19, 2010 11:07 AM". Below the navigation bar is a breadcrumb trail: "Home / User Management / User Management / New User".

The main content area is titled "New User Profile" and contains several tabs: "General", "Identity", "Communication Profile", "Roles", "Override Permissions", "Group Membership", "Attribute Sets", "Default Contact List", and "Private Contacts". The "Identity" tab is currently selected.

The "Identity" section contains the following fields:

- \* Login Name:** 30111@avaya.com
- \* Authentication Type:** Basic
- SMGR Login Password:** [Password field]
- \* Password:** [Password field]
- \* Confirm Password:** [Password field]
- Shared Communication Profile Password:** [Password field]
- Confirm Password:** [Password field]
- Localized Display Name:** AudioCodes 30111
- Endpoint Display Name:** AudioCodes 30111
- Honorific:** [Text field]
- Language Preference:** English
- Time Zone:** Eastern Time (US & Canada)

On the left side of the interface, there is a navigation menu with categories like "Asset Management", "Communication System Management", "Monitoring", "User Management", "Network Routing Policy", "Security", "Applications", "Settings", and "Session Manager". A "Shortcuts" section is also visible at the bottom of the menu.

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 configuration interface. On the left is a navigation menu with categories: Network Routing Policy, Security, Applications, Settings, and Session Manager. Below this is a 'Shortcuts' section with links for password changes and user management. The main area is titled 'Communication Profile' and contains a 'New' button. Below the button is a table with one row: 'Primary'. A 'Select: None' dropdown is present. The 'Name' field is set to 'Primary' and is marked as required. The 'Default' checkbox is checked. Below this is the 'Communication Address' section, which has 'New', 'Edit', and 'Delete' buttons. A table below shows a 'No Records found' message. The 'Type' dropdown is set to 'sip', and the 'SubType' dropdown is set to 'username'. The 'Fully Qualified Address' field is filled with '30111' and '@ 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

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**

**Name**

Primary

Select : None

\* **Name:** Primary

**Default:**

**Communication Address**

<input type="checkbox"/>	Type	SubType	Handle	Domain
<input type="checkbox"/>	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**

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**.

Quick Setup	
<b>LAN Setup</b>	
IP Type:	<input checked="" type="radio"/> Static IP <input type="radio"/> Automatic IP (DHCP)
IP Address:	<input type="text" value="10.3.3.213"/>
Subnet Mask:	<input type="text" value="255.255.255.0"/>
Default Gateway Address:	<input type="text" value="10.3.3.1"/>
Primary DNS:	<input type="text" value="0.0.0.0"/>
Secondary DNS:	<input type="text" value="0.0.0.0"/>
<b>SIP Proxy and Registrar</b>	
Use SIP Proxy:	<input type="text" value="Enable"/>
Proxy IP Address or Host Name:	<input type="text" value="10.1.2.170"/>
Proxy Port:	<input type="text" value="5060"/>
Use SIP Proxy IP and Port for Registration:	<input type="text" value="Enable"/>
Use SIP Registrar:	<input type="text" value="Disable"/>
<b>Line Settings</b>	
Line Activate:	<input type="text" value="Enable"/>
User ID:	<input type="text" value="30111"/>
Authentication User Name:	<input type="text" value="30111"/>
Authentication Password:	<input type="text" value="*****"/>

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 Configuration tab is active, and a sidebar on the left lists various configuration categories, with 'Voice Over IP' and 'Signaling Protocols' expanded. The main content area is titled 'Signaling Protocol' and contains two sections of configuration options:

- 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

A 'Submit' button with a checkmark icon 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 navigation menu with 'Dialing' selected under 'Voice Over IP'. The main content area is titled 'Dialing' and contains two sections: 'Dialing Parameters' and 'Automatic Dialing'. The 'Dialing Parameters' section includes the following fields:

- 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)

The 'Automatic Dialing' section includes:

- 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 navigation menu with 'Media Streaming' selected under 'Voice Over IP'. The main content area is titled 'Media Streaming' and contains three sections: 'Media Streaming Parameters', 'Quality of Service Parameters', and 'Codecs'. The 'Media Streaming Parameters' section includes:

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

The 'Quality of Service Parameters' section includes:

- Type of Service (ToS): 0xb8 Hex

The 'Codecs' section is a table with the following data:

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

The 'G.723 Bitrate' section includes:

- 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 Session Manager, 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 'AudioCodes', '320HD', 'Home', and 'Log Off'. The left sidebar contains a tree view with categories: Configuration, Management, and Status & Diagnostics. Under Configuration, there are sub-items: Quick Setup, Personal Settings, Network Connections, Voice Over IP (expanded), Signaling Protocols, Dialing, Media Streaming, Voice, Line Settings, Services, Volume Settings, and Advanced Applications. The main content area is titled 'Services' and contains several 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
- 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

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 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 Aura<sup>™</sup> Communication Manager and Avaya Aura<sup>™</sup> 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 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] *Administering Avaya Aura<sup>TM</sup> Communication Manager as a Feature Server*, May 2010, Issue 1.3, Release 5.2, Document Number 03-603479.
- [4] *Administering Avaya Aura<sup>TM</sup> 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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