Avaya Solution & Interoperability Test Lab

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

Abstract

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

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

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

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

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

1.2. Support

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

2. Reference Configuration

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

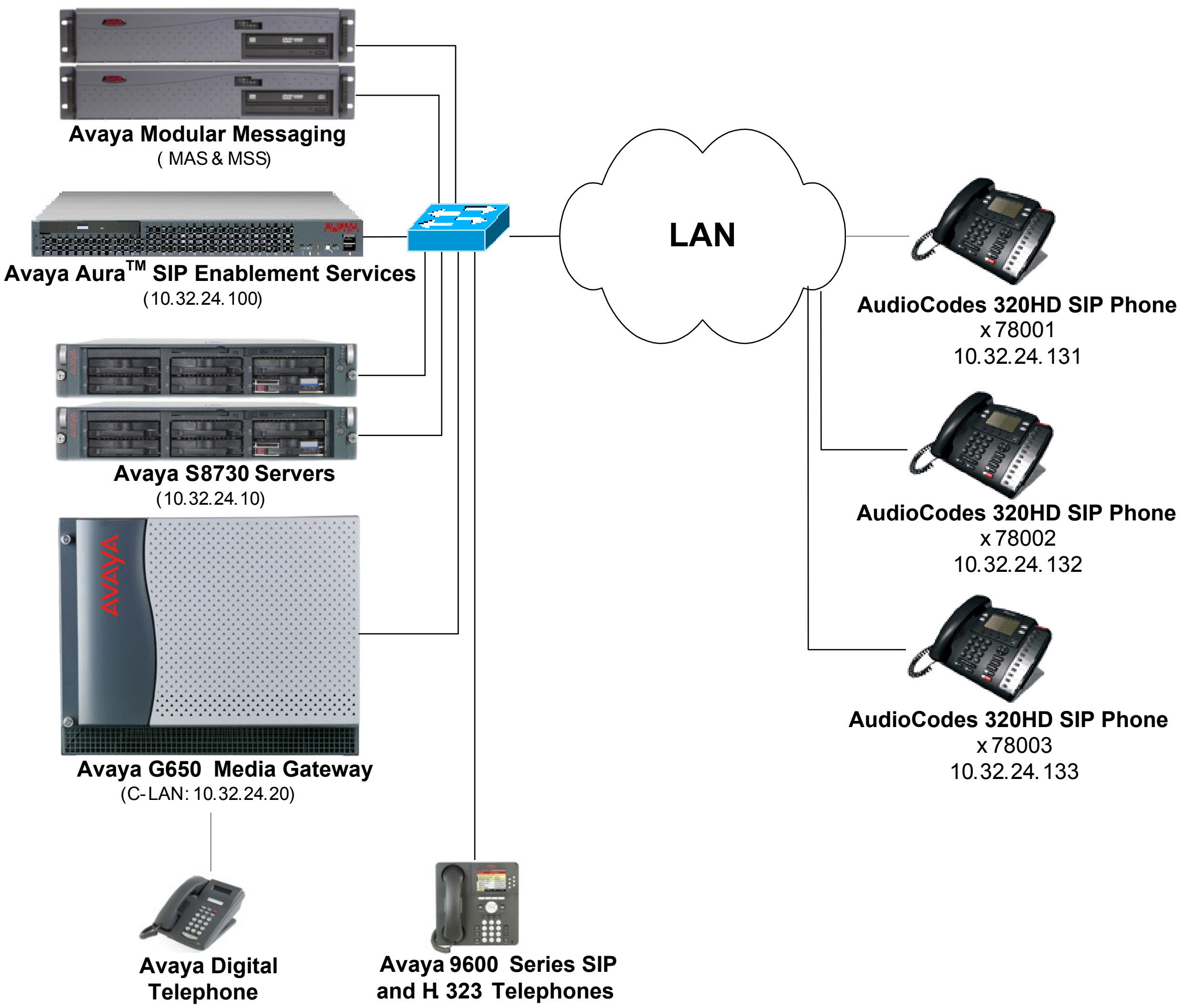


Figure 1: AudioCodes 320HD SIP Phones with Avaya SIP Solution

3. Equipment and Software Validated

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

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

4. Configure Aura™ Avaya Communication Manager

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

4.1. Verify OPS and SIP Trunk Capacity

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

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

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

                                                               USED
                                Platform Maximum Ports: 48000 336
                                      Maximum Stations: 36000 193
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones -   OPS: 100   33
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0

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

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

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 200   60
           Maximum Concurrently Registered IP Stations: 18000 9
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 500   60
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

4.2. Configure SIP Trunk

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

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
medprocrm           10.32.24.21
procr               0.0.0.0
ses                 10.32.24.100
 ( 6 of 16   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

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

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

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

```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

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

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

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

```
add signaling-group 10                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
                        Transport Method: tls
  IMS Enabled? n

   Near-end Node Name: clancrm             Far-end Node Name: ses
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 1
Far-end Domain: avaya.com

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

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

```
add trunk-group 10                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip          CDR Reports: y
  Group Name: To SES                     COR: 1        TN: 1        TAC: 1010
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                     Signaling Group: 10
                                                   Number of Members: 60
```

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

```
add trunk-group 10                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


 Show ANSWERED BY on Display? y
```

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

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

4.3. Configure Stations

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

```
add station 78001                                            Page   1 of   6
                                     STATION

Extension: 78001                        Lock Messages? n               BCC: 0
     Type: 9630SIP                      Security Code:                  TN: 1
     Port: IP                         Coverage Path 1: 20              COR: 1
     Name: 320HD 78001                Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 78001

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n
```

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

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

```
change station 78001                                            Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                        Coverage Msg Retrieval? 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: qsig-mwi

                                                Coverage After Forwarding? s

                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 78001         Always Use? n IP Audio Hairpinning? n
```

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

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

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

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

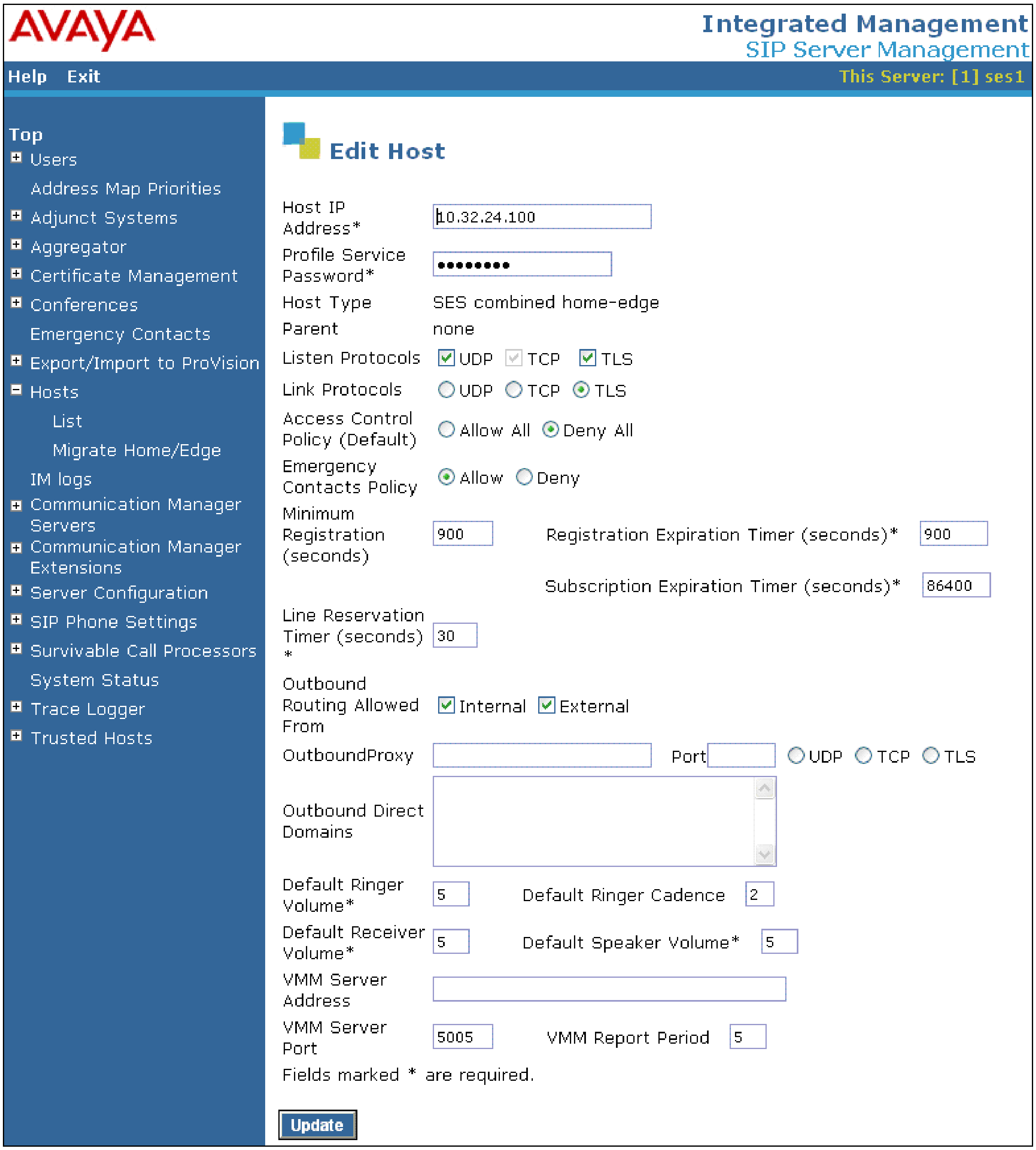
```
change off-pbx-telephone station-mapping 78001                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

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

5. Configure Avaya Aura™ SIP Enablement Services

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

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

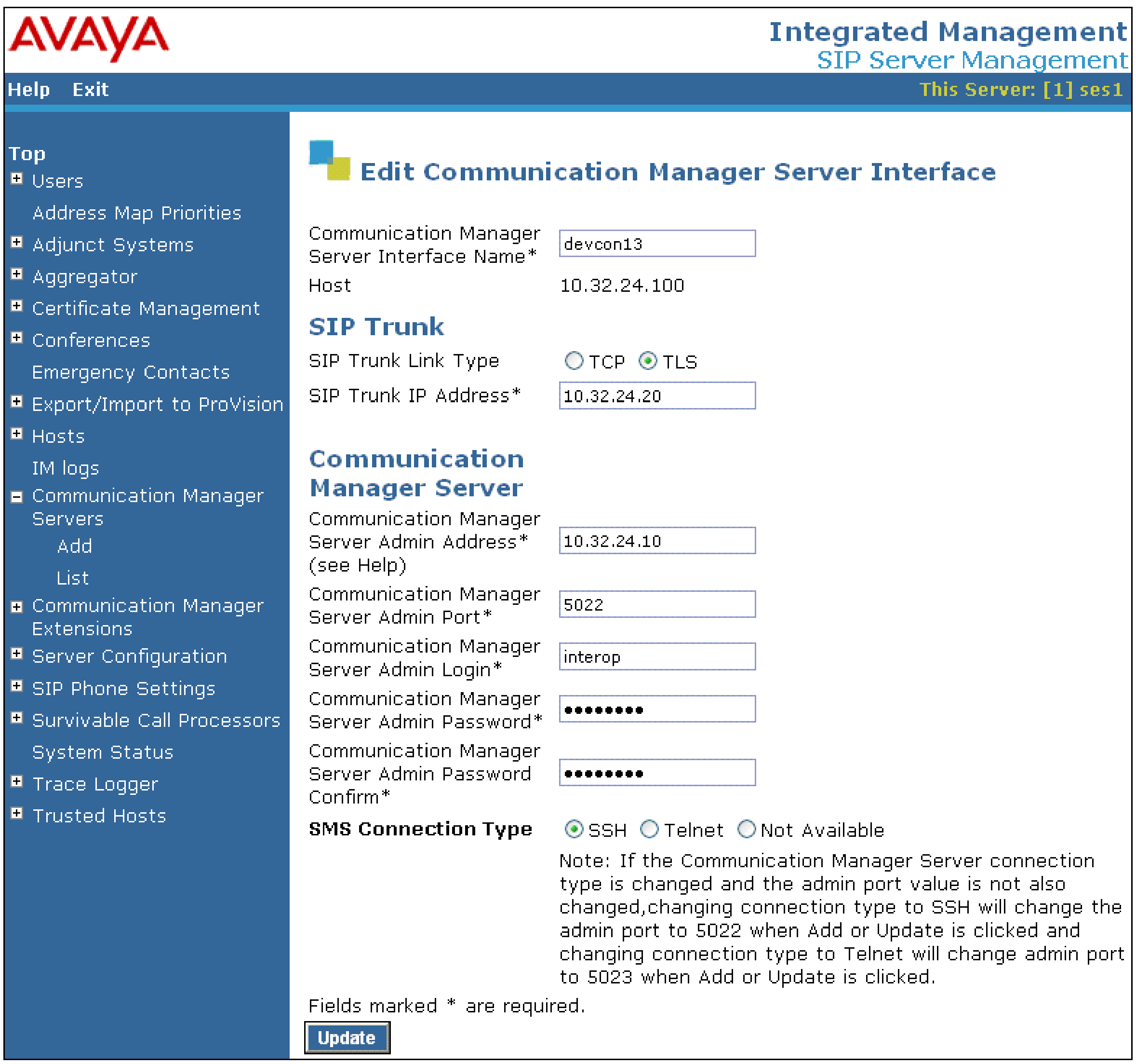
After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile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



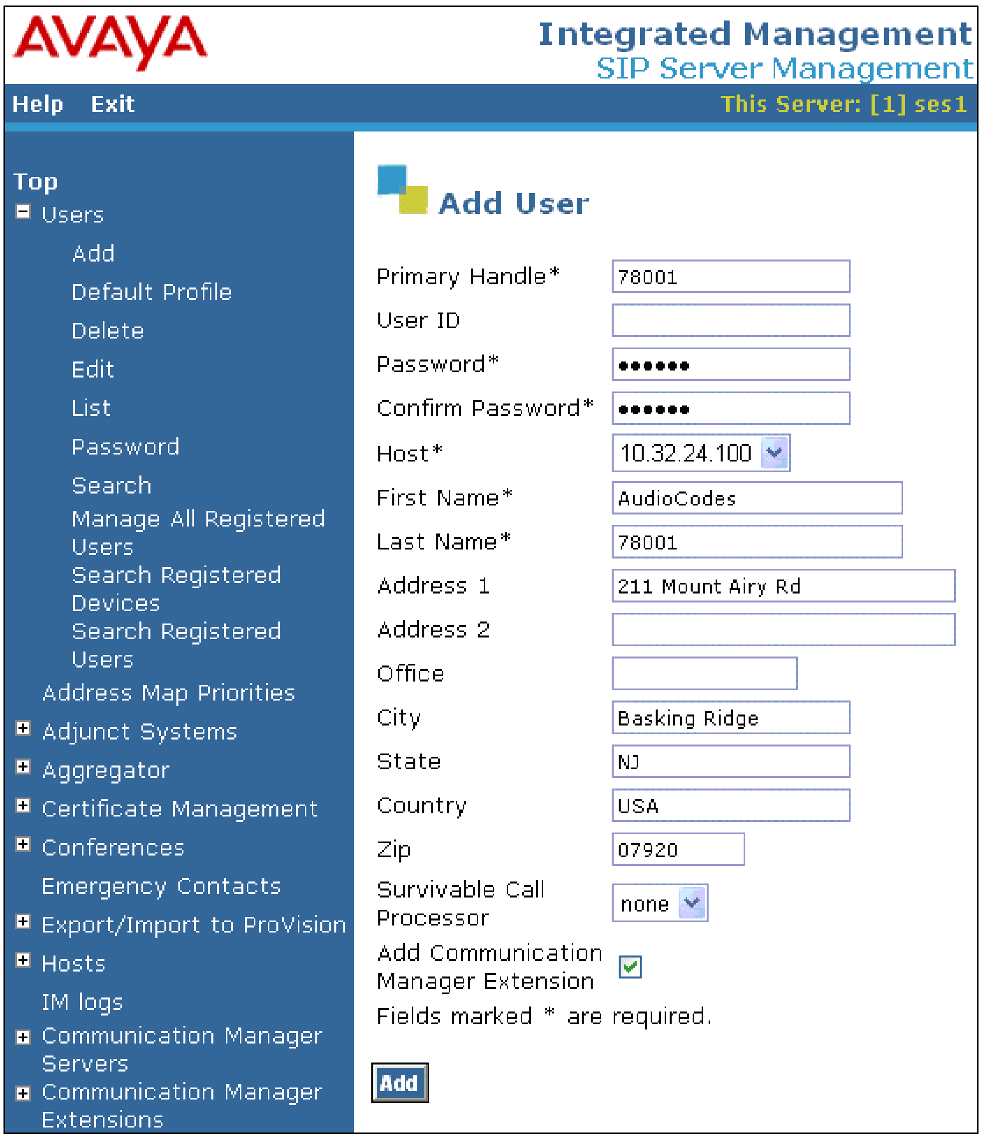
Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server in the enterprise site since a SIP trunk is required between Communication Manager and SIP Enablement Services. The screen below shows the Edit Communication Manager Server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

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

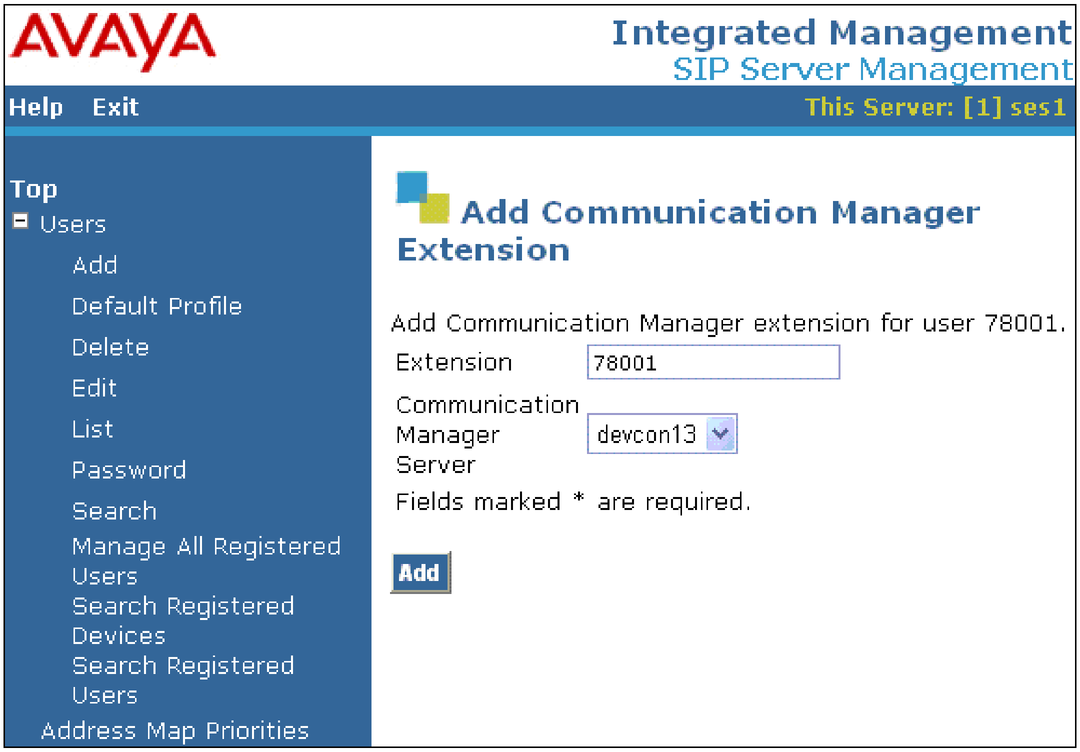
Refer to [2] for additional information on configuring the remaining fields.



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



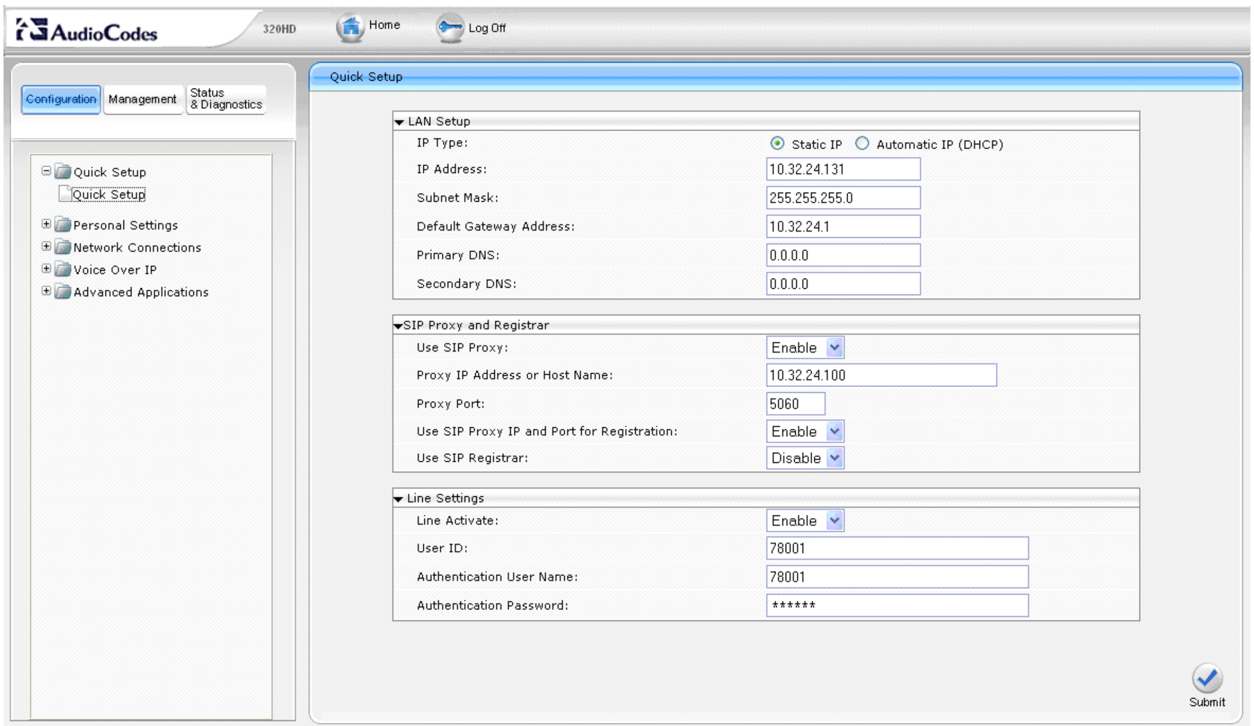
In the **Add Communication Manager Extension** screen, enter the **Extension** configured in Communication Manager for the previously added user. Usually, the Communication Manager extension and the user extension are the same. Click the **Add** button.



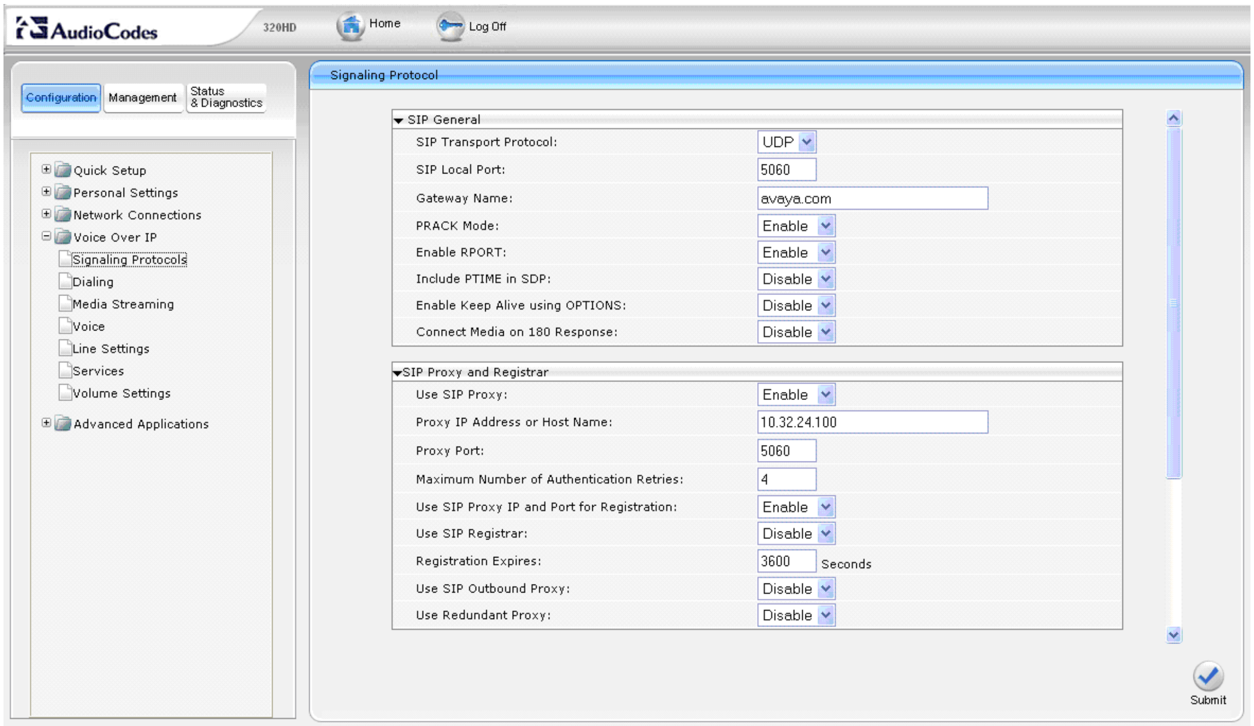
6. Configure AudioCodes 320HD SIP Phone

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

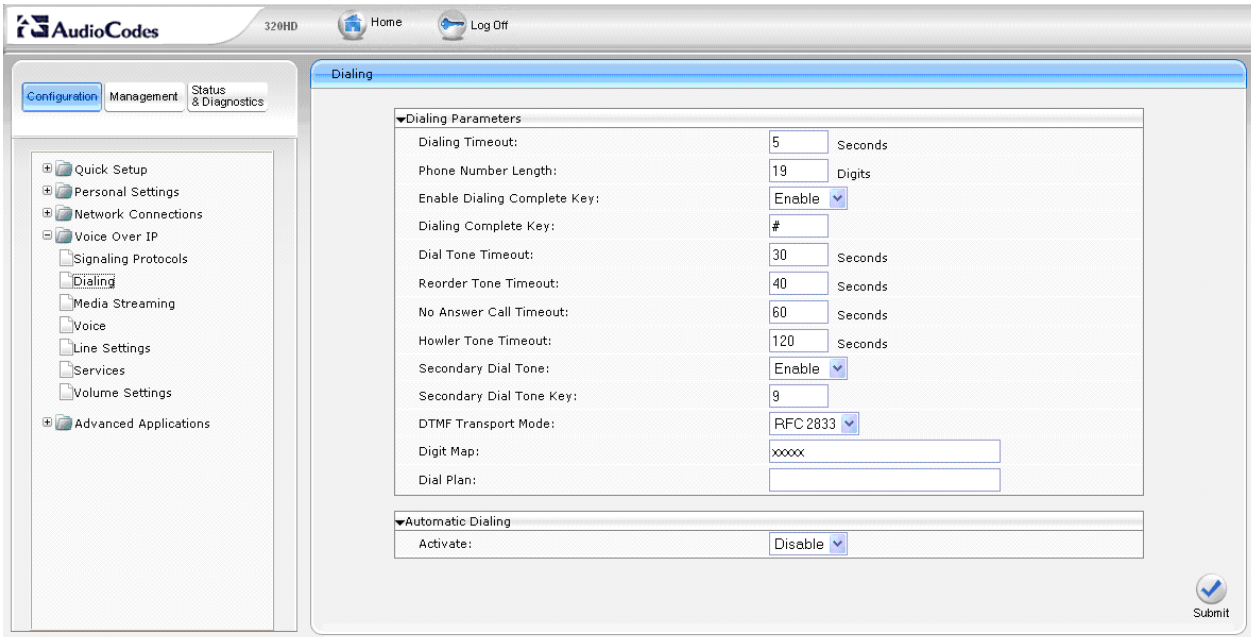
From an internet browser, enter http://*<ip-addr>* in the URL field, where *<ip-addr>* is the phone's IP address. Navigate to the **Quick Setup** screen shown below. The **LAN Setup** section was previously configured through phone's LCD screen. In the **SIP Proxy and Registrar** section, set the **Use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



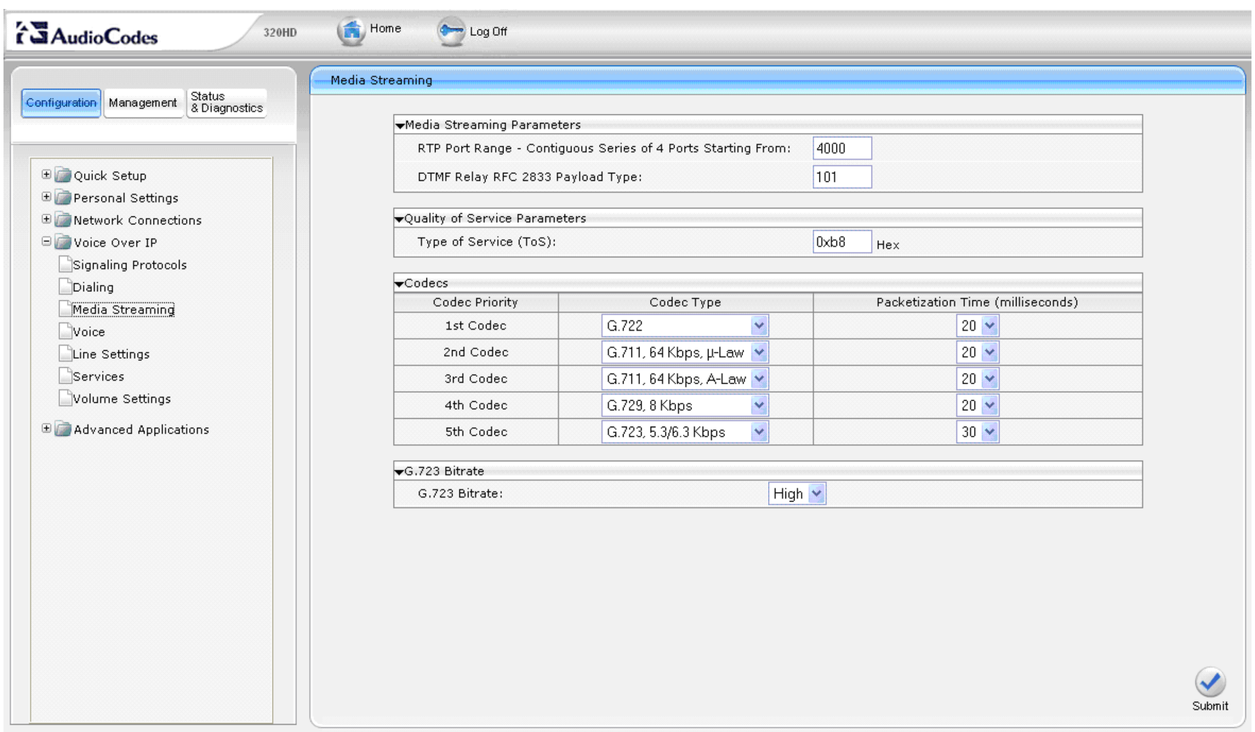
Navigate to **Voice Over IP→Signaling Protocols** and set the **Gateway Name** field to the SIP domain (e.g., *avaya.com*). All of the other fields should be properly set based on the **Quick Setup** configuration above. Click **Submit**.



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**.



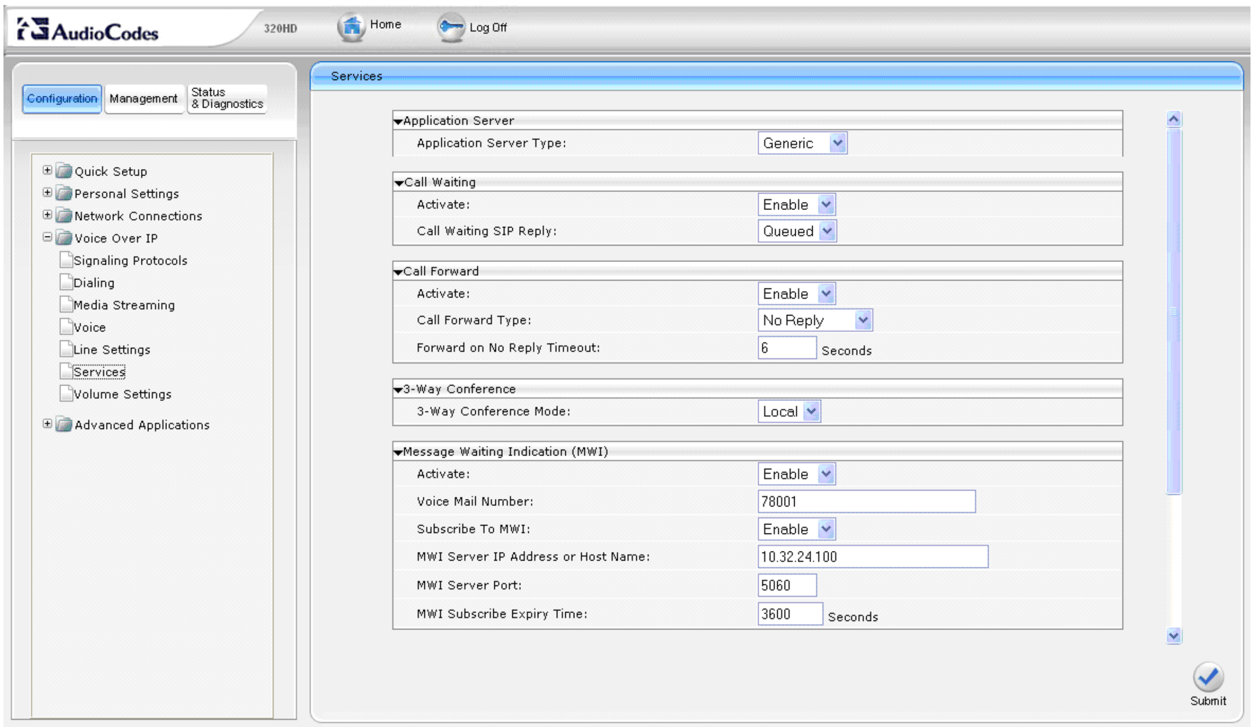
The following screen simply shows the codecs supported by the endpoint. No additional configuration is required here.



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

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

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



7. General Test Approach and Test Results

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

The following observations were noted during the compliance test:

- Shuffling should be disabled for calls to the 300HD SIP IP Phone Series to prevent issues with Hold and Conference.

- The Voice Mail Number on the 300HD SIP IP Phone Series should be set to the phone's extension for MWI to work. As a result, the user will have to manually dial the voicemail pilot number instead of simply pressing the Voicemail button on the phone.

- When a 320HD SIP Phone calls another phone with Call Forwarding enabled, there is no indication on the 320HD SIP Phone that the call was forwarded.

8. Verification Steps

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

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

9. Conclusion

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

10. References

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

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.