Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes MP-202 Telephone Adaptor with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the AudioCodes MP-202 Telephone Adaptor with Avaya SIP Enablement Services and Avaya Communication Manager. The AudioCodes MP-202 Telephone Adaptor serves as a gateway between legacy analog endpoints and a VoIP infrastructure using the Session Initiation Protocol (SIP).

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 procedure for configuring the AudioCodes MP-202 Telephone Adaptor with Avaya SIP Enablement Services (SES) and Avaya Communication Manager.

The MP-202 is a two-line, Session Initiation Protocol (SIP) gateway allowing residential and small office / home office (SOHO) subscribers to connect ordinary plain old telephone service (POTS) telephones or fax machines. The MP-202 registers with the Avaya SES as a SIP endpoint for each analog endpoint connected to it. When a call is placed from an analog telephone, the MP-202 will send SIP signaling messages to the Avaya SES to setup the call. Once the call has been setup, the MP-202 converts the analog signal from the analog telephone to a series of voice samples sent in data packets over the data network using the Real Time Protocol (RTP). The MP-202 extension numbers are also configured as Off PBX Stations (OPS) on Avaya Communication Manager.

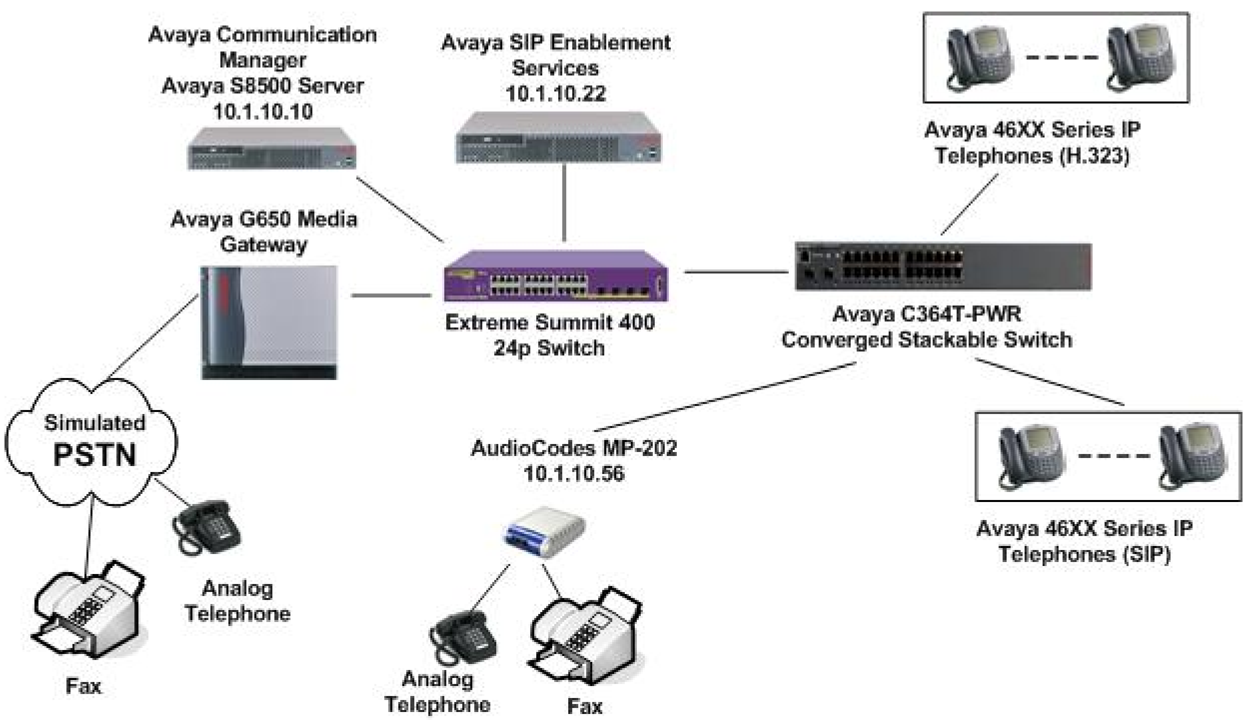


Figure 1: Test Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8500B Server	Avaya Communication Manager 5.0 (R015x.00.0.825.4), patch 15175
Avaya G650 Media Gateway C-LAN TN799DP Medpro TN2302AP	 HW 1, FW24 HW 20, FW116
Avaya SIP Enablement Services	5.0 (5.0.0.0-825.31), patch SP2d
Avaya C364T-PWR Converged Stackable Switch	4.3.12
Avaya 46XX Series IP Telephones (H.323)	2.8
Avaya 46XX Series IP Telephones (SIP)	2.3
Extreme Summit 400 24p Switch	Extremeware 7.5e.2.8
Analog Telephones	-
Analog Fax Machines	-
Windows PCs	Windows XP Professional
AudioCodes MP-202 Analog VoIP Gateway	2.6.0_Build2

3. Configure Avaya Communication Manager

The communication between Avaya Communication Manager and Avaya SES is via a SIP trunk group. All SIP signaling for calls between Avaya Communication Manager and the MP-202 passes through Avaya SES via this trunk group. This section describes the configuration of stations as OPS stations, which is required for each analog telephone, and fax machine connected to the MP-202.

Step	Description
1.	Use the **display system-parameters customer-options** command to verify that sufficient SIP trunk capacity exists. On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options        Page   2 of  10
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 800    90
           Maximum Concurrently Registered IP Stations: 2400   5
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 20     0
  Max Concur Registered Unauthenticated H.323 Stations: 2400   0
                  Maximum Video Capable H.323 Stations: 2400   0
                   Maximum Video Capable IP Softphones: 2400   0
                       Maximum Administered SIP Trunks: 800    35

   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 250    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128    0
          Maximum TN2602 Boards with 320 VoIP Channels: 128    1
   Maximum Number of Expanded Meet-me Conference Ports: 300    0
```

Step	Description

2.	All SIP stations are configured as OPS stations on Avaya Communication Manager. This includes the analog telephones, and fax machine connected to the MP-202, which appear as SIP stations to Avaya Communication Manager. Use the **display system-parameters customer-options** command to verify Avaya Communication Manager has sufficient OPS capacity available to add the OPS stations needed for the SIP and analog endpoints. If there is insufficient capacity, contact an authorized Avaya sales representative or business partner to make the appropriate changes.
	<pre>display system-parameters customer-options Page 1 of 10
	                                OPTIONAL FEATURES
	     G3 Version: V14
	       Location: 2 RFA System ID (SID): 1
	       Platform: 12 RFA Module ID (MID): 1
	
	                                                                USED
	                                Platform Maximum Ports: 3200 483
	                                      Maximum Stations: 2400 253
	                              Maximum XMOBILE Stations: 2400 0
	                    Maximum Off-PBX Telephones - EC500: 2400 0
	                    Maximum Off-PBX Telephones - OPS: 2400 35
	                    Maximum Off-PBX Telephones - PBFMC: 0 0
	                    Maximum Off-PBX Telephones - PVFMC: 0 0
	                    Maximum Off-PBX Telephones - SCCAN: 2400 0</pre>

3.	Use the **change ip-codec-set *n*** command, where ***n*** is the codec set value. Multiple codecs can be listed in priority order to allow the codec to be negotiated during call establishment. The example below shows the values used in the compliance test.
	<pre>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.711A n 2 20**
	 2: **G.729AB n 2 20**</pre>

4.	On Page 2, the **FAX Mode** field must be set to **t.38-standard** to support the fax machines. The **Modem** field should be set to **off**. The screen below shows the setting used for the fax testing.
	<pre>change ip-codec-set 1 Page 2 of 2
	                          IP Codec Set
	                              Allow Direct-IP Multimedia? n
	                    Mode Redundancy
	    FAX t.38-standard 0
	    Modem off 0
	    TDD/TTY US 3
	    Clear-channel n 0</pre>

Step	Description

5.	To add a station, use the **add station *n*** command where ***n*** is an unused extension number. Use the default value of ***6408D+*** for the **Type** field. Enter an ***X*** in the **Port** field. This indicates a station is being added without identifying a physical port for the station to use. Enter a descriptive name in the **Name** field. The default values may be retained for all other fields.
	<pre>add station 10053 Page 1 of 5
	                                     STATION
	
	Extension: 10053 Lock Messages? n BCC: 0
	     Type: 6408D+ Security Code: TN: 1
	     Port: X Coverage Path 1: COR: 1
	     Name: AudioCodes Coverage Path 2: COS: 1
	                                       Hunt-to Station:
	STATION OPTIONS
	                                           Time of Day Lock Table:
	              Loss Group: 2 Personalized Ringing Pattern: 1
	             Data Module? n Message Lamp Ext: 10053
	            Speakerphone: 2-way Mute Button Enabled? y
	        Display Language: english
	
	          Survivable COR: internal Media Complex Ext:
	   Survivable Trunk Dest? y IP SoftPhone? n</pre>

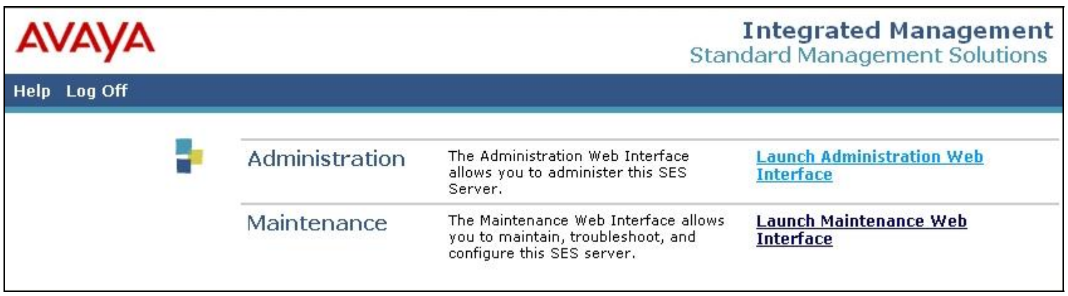
6.	Map the Avaya Communication Manager extension to the Avaya SES media server extension defined in Section 4, Step 8 with the **add off-pbx-telephone station-mapping** command. Enter the values as shown below:
	
	▪ **Station Extension**: Avaya Communication Manager extension
	▪ **Application**: OPS
	▪ **Phone Number**: Avaya SES media server extension
	▪ **Trunk Selection**: The SIP trunk group number
	▪ **Configuration Set**: Enter a valid configuration set. The compliance test used configuration set 1 which contained the default values.
	<pre>add off-pbx-telephone station-mapping Page 1 of 2
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	Station Application Dial CC Phone Number Trunk Config
	Extension Prefix Selection Set
	 10053 OPS - 10053 30 1</pre>

7.	Repeat Steps 5 and 6 for the remaining MP-202 endpoint.

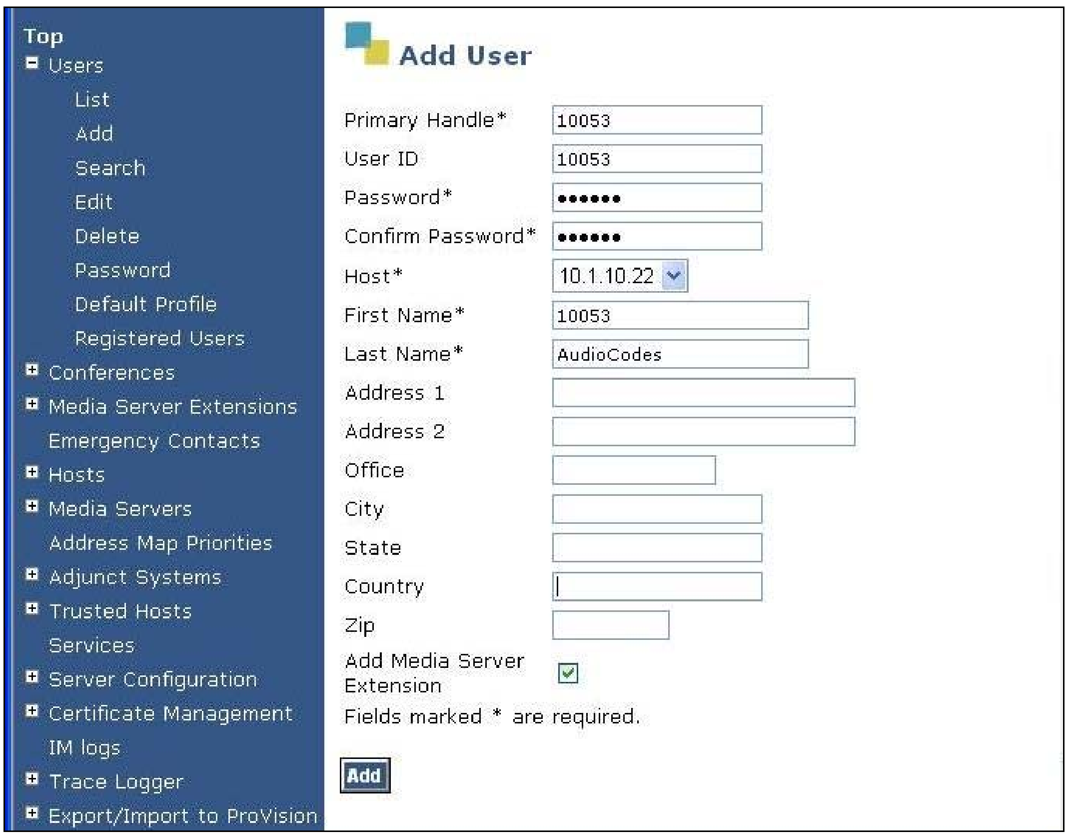
4. Configure Avaya SES

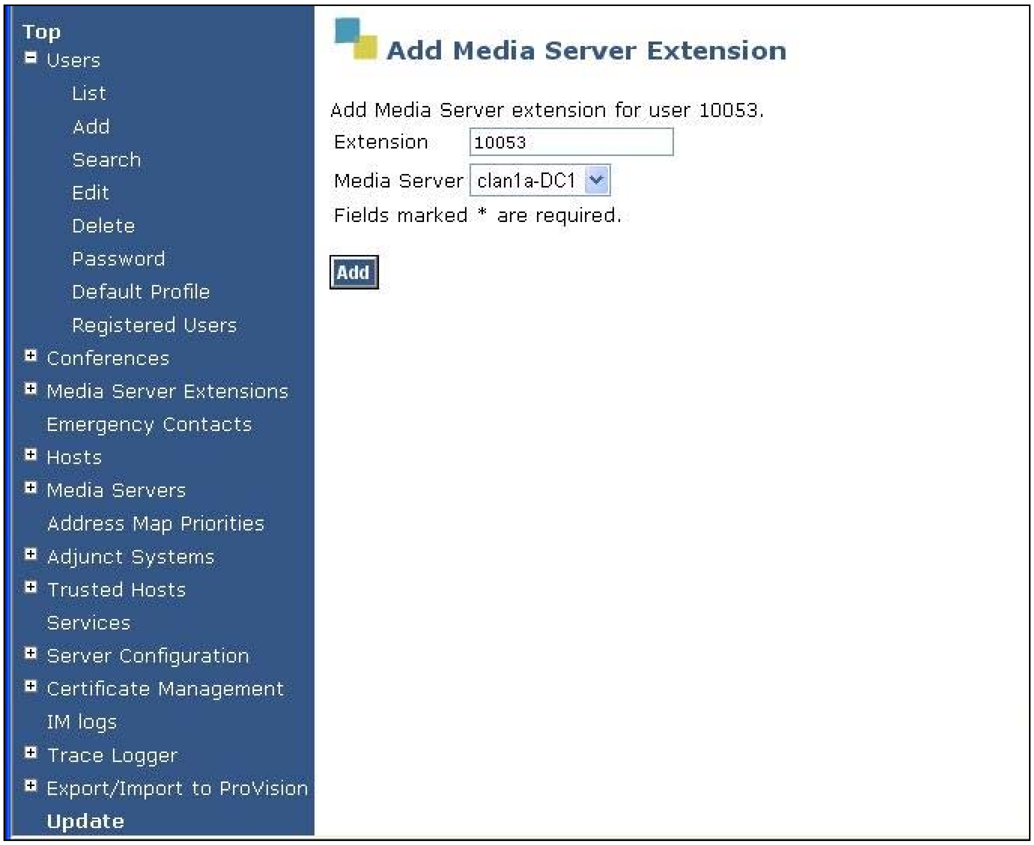
It is assumed that the connectivity between Avaya SES and Avaya Communication Manager has already been established. For information on the configuration setup between Avaya SES and Avaya Communication Manager, refer to Section 10.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Access the Avaya SES administration web interface by entering http://<<i>ip-addr</i>>/admin as the URL in an Internet browser, where <<i>ip-addr</i>> is the IP address of the Avaya SES server. Log in with the appropriate credentials and then select the Launch Administration Web Interface link from the main page as shown below.



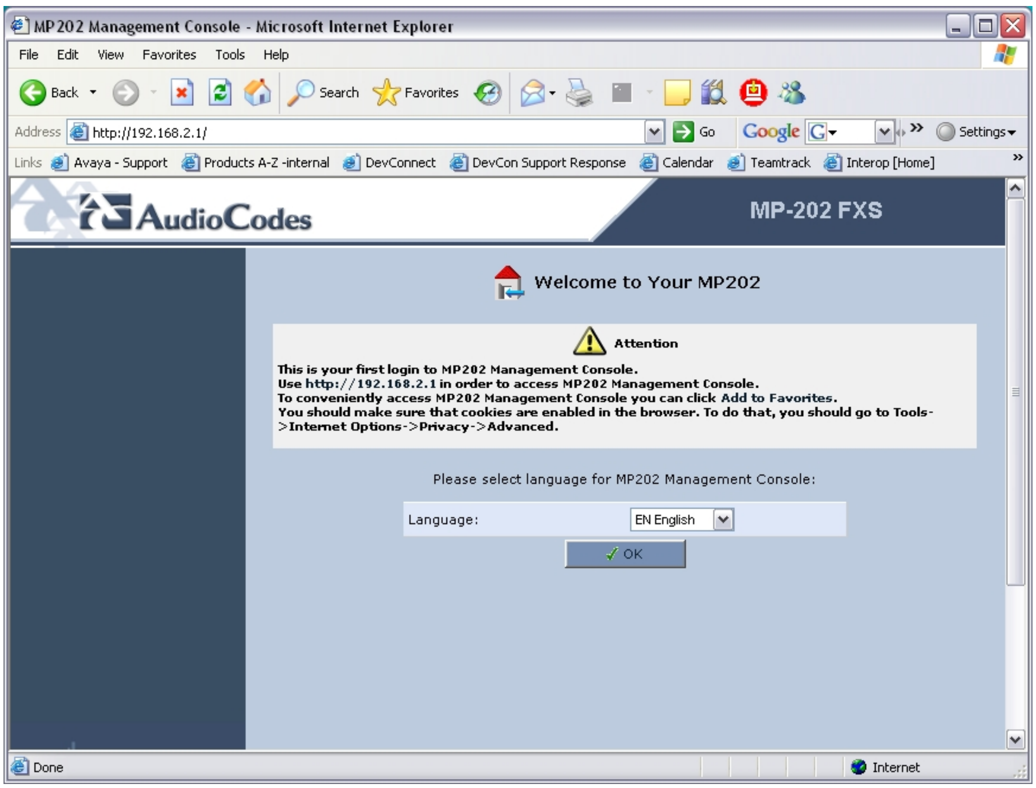
</td></tr>
</table>

Step	Description
2.	A user must be added on Avaya SES for each MP-202 endpoint and also be created in Avaya Communication Manager in Section 3, Step 5. From the left pane, select **Users** → **Add**. Enter the values as shown below. ▪ **Primary Handle**: Enter the extension for this user. ▪ **UserID**: Enter a descriptive name for the User ID. ▪ **Password**: Enter a valid password for logging into the SIP endpoint. ▪ **Confirm Password**: Re-enter the password. ▪ **Host**: Select the Avaya SES server from the pull-down menu. ▪ **First Name**: Any descriptive name. ▪ **Last Name**: Any descriptive name. Check the **Add Media Server Extension** checkbox. Click the **Add** button to proceed. A confirmation window will appear. Click **Continue** on this new page to proceed. 

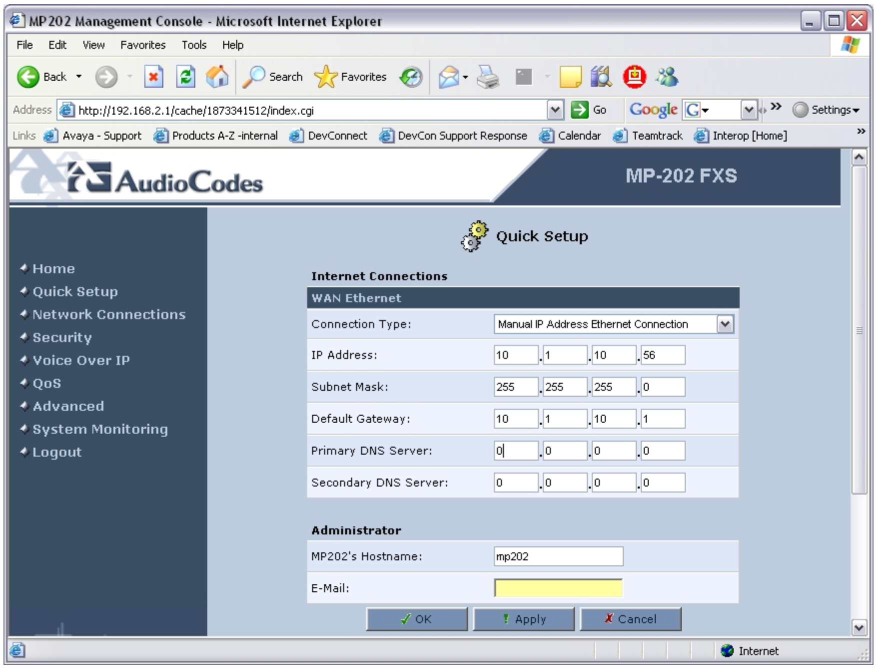
Step	Description
3.	The **Add Media Server Extension** page will appear. In the **Extension** field, enter the same extension used in the previous step. From the pull down **Media Server** field, select the name of the pre-configured Avaya Communication Manager CLAN that the extension configured will be linked with. Click the **Add** button to complete the operation. In the bottom left hand pane click **Update**. 
4.	Repeat Steps 2 and 3 for the remaining MP-202 endpoint.

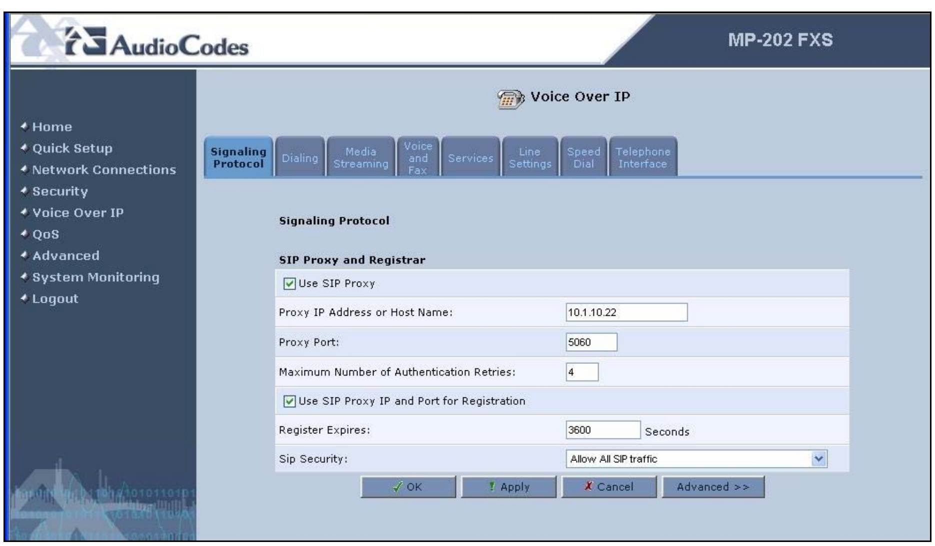
5. Configure the MP-202 Telephone Adaptor

This section describes the procedures for configuring the MP-202. The configuration of the MP-202 is done via a Web browser.

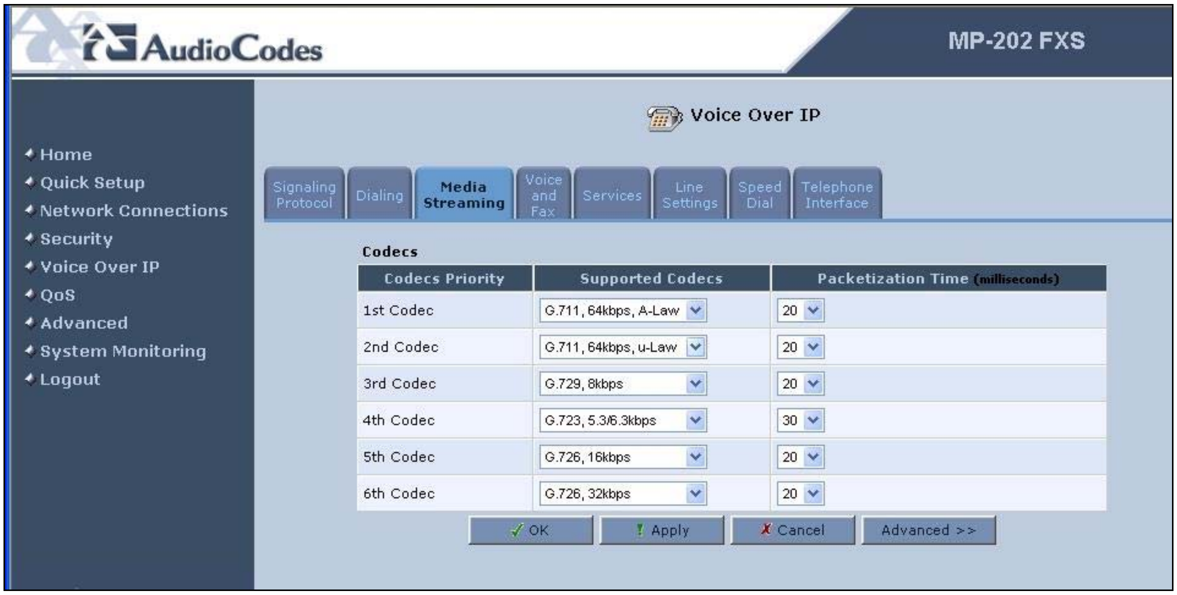
Step	Description
1.	Using an Ethernet cable, connect a PC to MP-202's Ethernet port labeled LAN/PC. By default the LAN/PC Ethernet has DHCP enabled. To access the device, enter the default IP address *192.168.2.1* of the MP-202 in the **Address** field of the browser. Select the language and click **OK**. 
2.	In the login screen enter the appropriate username and password (not shown). Click **OK**.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>From the left menu click on Quick Setup. For the WAN Ethernet connection enter an IP address, subnet mask and default gateway that can access the SES Server. Default values may be retained for all other values.

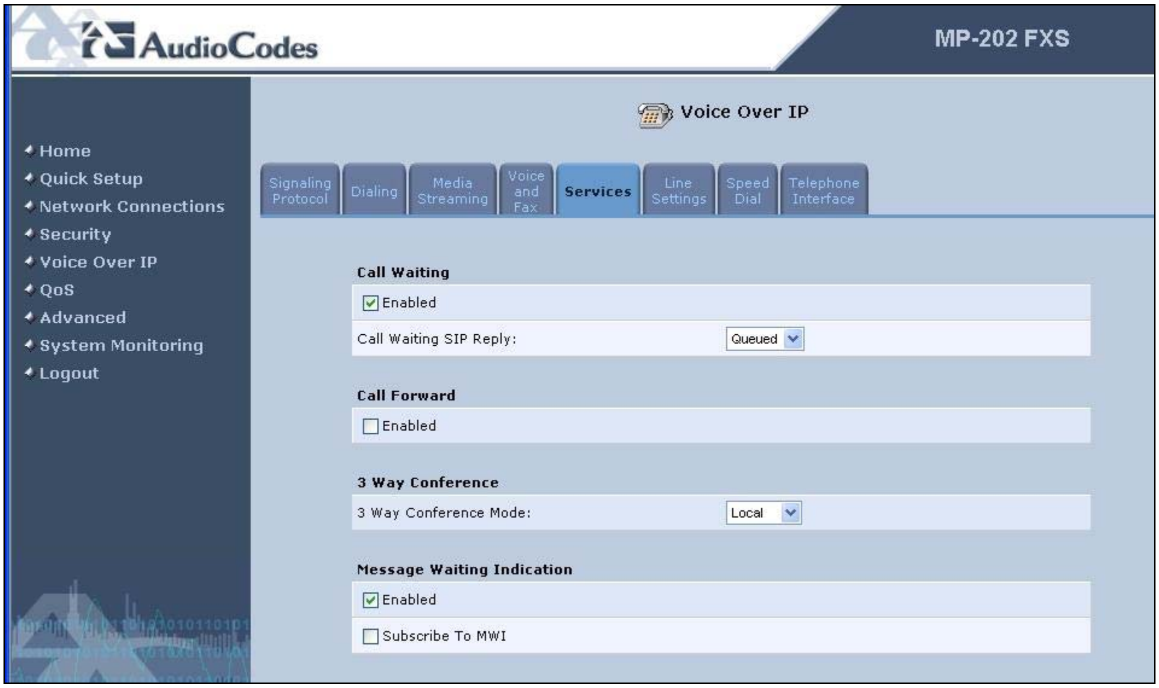
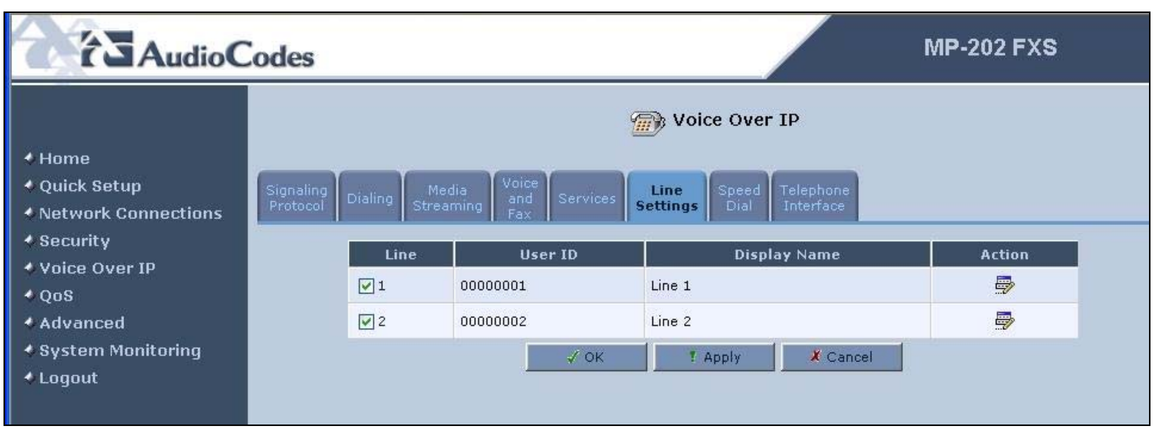

</td></tr>
<tr><td>4.</td><td>From the left menu select Voice Over IP and then select the Signaling and Protocol tab. For the SIP Proxy and Registrar settings, check the Use SIP Proxy check box and enter the SES IP address in the Proxy IP Address or Host Name field. Check the Use SIP Proxy IP and Port Registration check box. Default values may be retained for all other fields. Click Apply.


</td></tr>
</table>

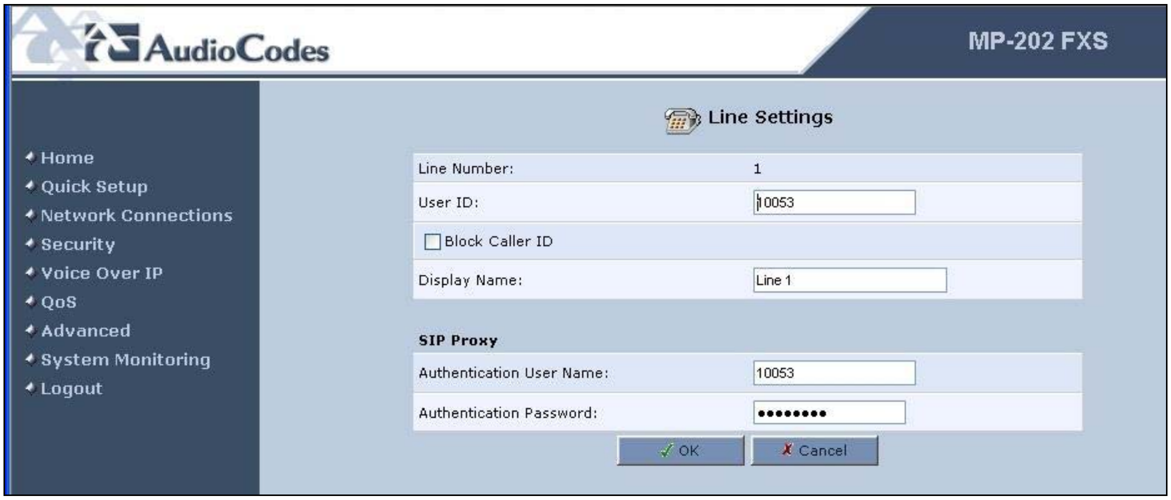
Step	Description
5.	Select the **Media Streaming** tab. In the screen below, select the list of preferred codecs to be used by the MP-202 with the most preferred codec at the top and working downward to the least preferred. This list must have an overlap with the list provided on Avaya Communication Manager in Section 3, Step 3. The codec is selected from the pull-down menu under the **Supported Codecs**. The codec list used for the compliance test is shown in the example below. ***G.711A-law*** was selected as the most preferred followed by ***G.729***. Default values were retained for all other fields. Click **Apply**.



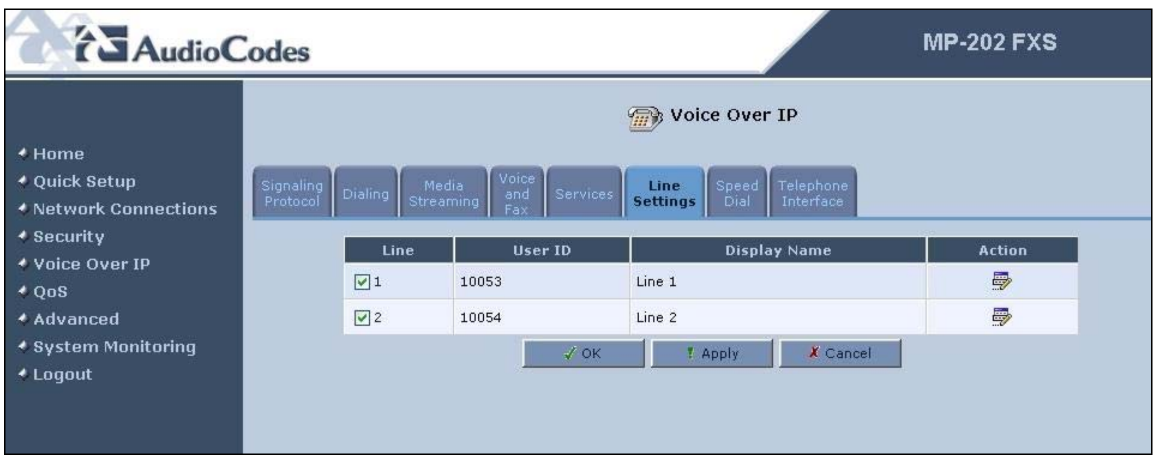
Step	Description
6.	Select the Voice and Fax tab. With silence suppression enabled, the MP-202 uses the equivalent of a G.729AB codec. For the G.729 codec, check the **Enable Silence Suppression** check box. For the Fax and Modem Settings. ▪ Select ***T.38 Relay*** for the **Fax Transport Mode** to support faxing across the data WAN. ▪ Select ***Bypass*** for the **Modem Transport Mode** to support modem calls over SIP. Default values may be retained for all other fields. Click **Apply**. 

Step	Description
7.	Select **Services** tab. Configure the parameters as described below. ▪ Call Waiting is enabled by default. ▪ Call forward was disabled by default but enabled during testing. ▪ Check the **Enabled** check box for Message Waiting Indication (MWI). For the compliance MWI was only tested for stutter dial tone. Default values may be retained for all other fields. Scroll to the bottom and click **Apply**. 
8.	Select the **Line Settings** tab and click on the **Action** icon for Line 1. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>In the User ID field enter the user ID configured on the SES Section 4, Step 2. Enter Authentication User Name and Authentication Password that matches the values configured on Avaya SES in Section 3, Step 2. Block caller ID is left unchecked, as it is not presently supported. Default values may be retained for all other fields. Click OK.



</td></tr>
<tr><td>10.</td><td>Repeat the previous step for the remaining MP-202 endpoint and click OK.



</td></tr>
</table>

6. Interoperability Compliance Testing

This section describes the compliance testing used to verify the interoperability between the AudioCodes MP-202, Avaya SIP Enablement Services (SES) and Avaya Communication Manager. This section covers the general test approach and the test results.

6.1. General Test Approach

The general test approach was to make calls to/from the analog telephones connected through the MP-202 using various codec settings and exercising common PBX features such as hold, transfer and conference. This testing included the analog telephones, Avaya SIP telephones and Avaya H323 telephones.

6.2. Test Results

The AudioCodes MP-202 successfully passed compliance testing. The following features and functionality were verified using an MP-202 analog endpoint.

- Calls to/from the MP-202 analog telephones
- Calls to/from the PSTN
- Intra-branch calls
- G.711A and G.729AB codec support
- Proper recognition of DTMF transmissions
- Hold/Retrieve
- Transfer
- Call Waiting
- Call Forwarding.
- 3-Way Conferencing
- Proper operation of voicemail with message waiting indicators (MWI). For the analog phones, MWI was provided via stutter dial tone.
- Extended telephony features using Avaya Communication Manager Feature Name Extensions.
- T.38 fax support
- Proper system recovery after a MP-202 restart

The following observations were made during the compliance test:

- When using Avaya Communication Manager Feature Access Codes (FAC), from an MP-202 analog endpoint the user is required to wait a couple of seconds before hanging up after hearing the feature confirmation tones, otherwise the FAC will not be activated.
- MP-202 analog endpoint cannot use # to enable or disable FACs. Compliance testing was carried out using *XX to enable and disable FACs.

7. Verification Steps

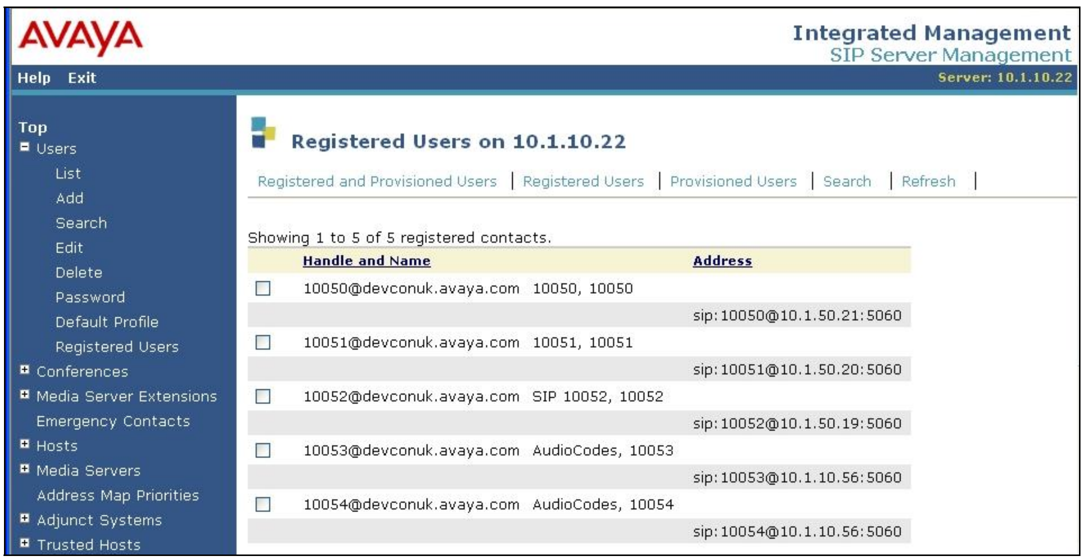
The following steps may be used to verify the configuration:

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>

- From the Avaya Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is in-service.
- From the Avaya Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is in-service.
- Verify that calls can be placed to/from the MP-202 analog endpoints.

</td></tr>
<tr><td>2.</td><td>

From the Avaya SES web administration interface menu, select **Users → Registered Users**. Verify that all endpoints behind the MP-202 are registered with the Avaya SES.



</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>From the MP-202 web administration menu, select System Monitoring and then select the Traffic tab. Verify the WAN Ethernet is Connected and packets are sent and received

</td></tr>
<tr><td>4.</td><td>From the MP-202 web administration menu, select System Monitoring and then select the Voice Over IP tab. Verify both MP-202 endpoints are registered.

</td></tr>
</table>

8. Support

Technical support for the AudioCodes MP-202 Telephone Adaptor can be obtained from AudioCodes. See the Support link at www.audiocodes.com for contact information.

9. Conclusion

These Application Notes describe the configuration steps required for AudioCodes MP-202 to successfully interoperate with Avaya SIP Enablement Services 5.0 and Avaya Communication Manager 5.0. AudioCodes MP-202 Telephone Adaptor can successfully register to Avaya SIP Enablement Services, place/receive calls and utilize telephony features of Avaya Communication Manager.

10. Additional References

This section references the Avaya and AudioCodes product documentation that are relevant to these Application Notes.

The following documentation is available at: http://support.avaya.com.

- *Documentation for Avaya Communication Manager (4.0), Media Gateways and Servers, Document* ID 03-300151, Issue 6, February 2007.
- *Installing and Administering SIP Enablement Services*, Document ID 03-600768, Issue May 2007.

The following documentation is available on request from AudioCodes: www.audiocodes.com

- *MP-202 Telephone Adapter User's Manual*: Version 2.4.0: Document: LTRT-50604
- *MP-202 Telephone Adapter Quick Installation Guide*: Document LTRT-50404

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.