Avaya Solution & Interoperability Test Lab

Application Notes for the ClearOne Communications MAX IP with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a compliance-tested solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services (SES), and ClearOne Communications MAX IP Tabletop Conferencing Phones. MAX IP phones are SIP-based VoIP tabletop conferencing phones intended for use in conference rooms and similar environments. During compliance testing, the MAX IP phones successfully registered with Avaya SES, placed/received calls to/from SIP and non-SIP telephones, and established conference calls. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance-tested solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services 3.1, and ClearOne Communications MAX IP Tabletop Conferencing Phones. MAX IP phones are SIP-based VoIP tabletop conferencing phones intended for use in conference rooms and similar environments. Up to four MAX IP phones may be daisy-chained together to increase physical coverage, particularly in large or unusually shaped conference rooms. Each MAX IP phone supports two lines, and can bridge calls on the two lines to establish a 3-party conference.

Figure 1 illustrates a sample configuration consisting of an Avaya S8710 Media Server, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and ClearOne Communications MAX IP Tabletop Conferencing Phones. Avaya Communication Manager runs on the S8710 Media Server. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based MAX IP phones and Avaya SIP, H.323, and digital phones. The analog PSTN phone is also included to demonstrate calls routed by Avaya Communication Manager between the MAX IP phones and the PSTN.

The MAX IP phone originates a call by sending a call request (SIP Invite message) to the Avaya SES server. The Avaya SES server routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP phone, such as another MAX IP phone or an Avaya SIP phone, then Avaya Communication Manager routes the call back over the SIP trunk to the Avaya SES server, which in turn delivers the call to the destination SIP phone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog phone, an adjunct, a vector, a hunt group, etc., depending on the destination number. For a call arriving to Avaya Communication Manager that is destined for the MAX IP phone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server, which in turn delivers the call to the MAX IP phone. These Application Notes assume that the SIP trunk between Avaya Communication Manager and the Avaya SES server has already been configured. For details on configuring SIP trunks on Avaya Communication Manager and Avaya SES, consult [1] and [4].

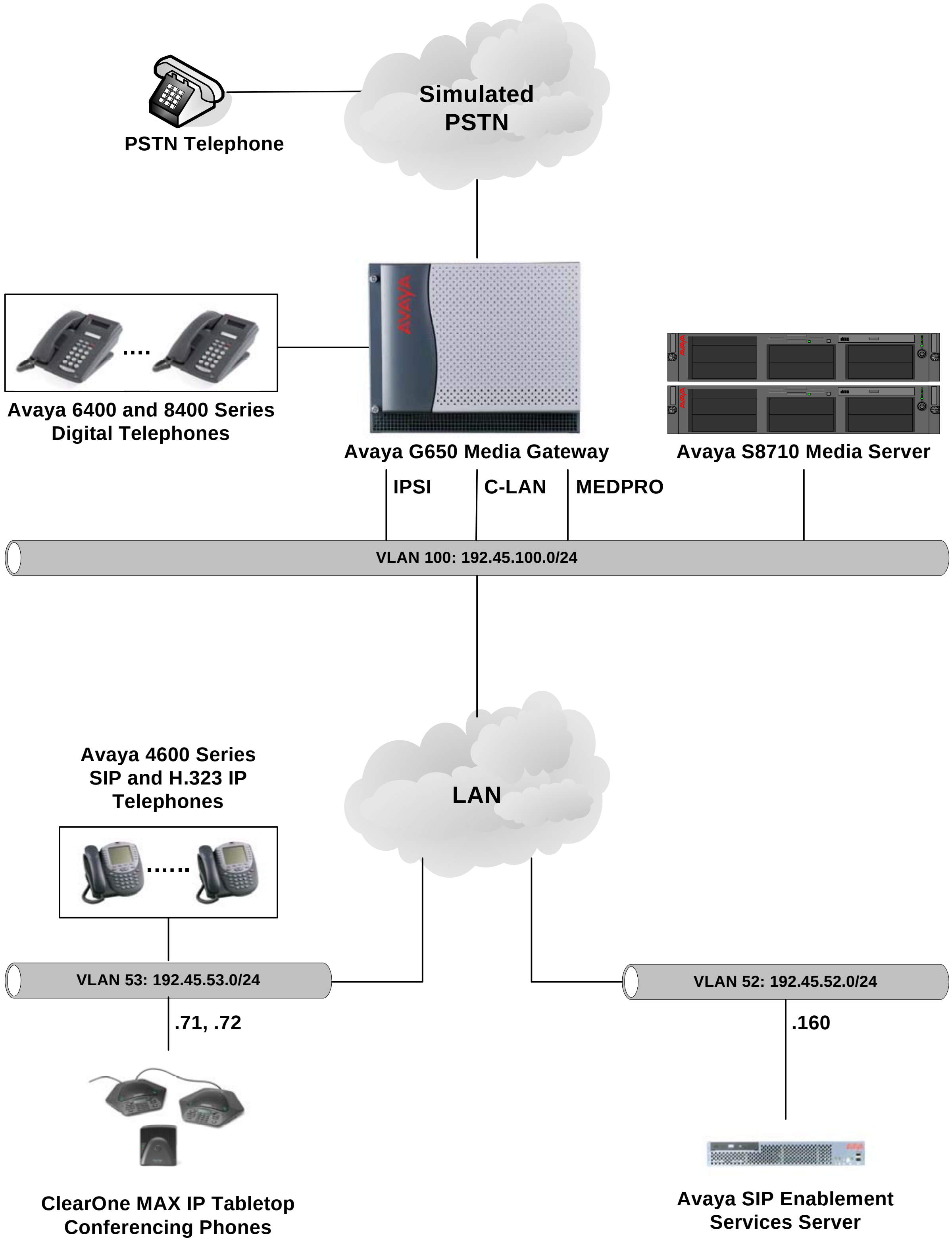


Figure 1: Sample configuration.

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW1 FW 17
TN2302AP IP Media Processor	HW20 FW 111
Avaya SIP Enablement Services Server	3.1
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.4 (4610SW H.323) 2.4 (4620SW H.323) 2.5 (4625SW H.323) 2.2.2 (4610SW SIP) 2.2.2 (4620SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
ClearOne Communications MAX IP Tabletop Conferencing Phones	06-09-06
ClearOne Communications MAXAttach IP Pods (for daisy-chaining up to 4 MAX IP phones)	11-03-05
Analog Telephone	-

3. Configure Avaya Communication Manager

This section describes the steps for configuring IP codec sets and associating SIP phone numbers with off-PBX telephone stations in Avaya Communication Manager. The steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. IP codec sets identify the codecs that may be used in calls involving VoIP endpoints. An off-PBX telephone is a phone that Avaya Communication Manager does not control, such as a cellular phone, a home phone, or a SIP phone. Avaya Communication Manager features and calling privileges, however, can be applied to an off-PBX telephone by associating a local, i.e. on-PBX, extension with the off-PBX telephone. This approach is taken for SIP phones that register with the Avaya SES server and intend to use Avaya Communication Manager for call origination and termination services. Specifically, an Administration WithOut Hardware (AWOH) on-PBX station is administered in Avaya Communication Manager and then associated with the phone number of the SIP phone. Similarly, on the Avaya SES server, the number of the SIP phone is administratively associated with the extension of the on-PBX station. Throughout the rest of this document, on-PBX stations associated with SIP phones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

3.1. IP Codec Set

Enter the **change ip-codec-set c** command, where "c" is a number between 1 and 7, inclusive. Enter at least one of the codecs supported in the ClearOne MAX IP phone (see Section 5 Step 7). IP codec sets are specified in the IP Network Region forms to define which codecs may be used within and between network regions.

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

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729             n         2        20
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: aes
 2: none
 3:
```

3.2. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the numbers of ClearOne MAX IP phones.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the display system-parameters customer-options command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.
<pre>
display system-parameters customer-options   Page   1 of 10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1   RFA System ID (SID): 1
       Platform: 8   RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 818
                                      Maximum Stations: 36000 397
                              Maximum XMOBILE Stations: 0   0
                    Maximum Off-PBX Telephones - EC500: 5   0
                    Maximum Off-PBX Telephones -   OPS: 200   50
                    Maximum Off-PBX Telephones - SCCAN: 0   0
</pre></td></tr>
<tr><td>2.</td><td>Enter the add station s command, where “s” is an available extension in the dial plan, to administer an OPS station. On Page 1 of the station form, set Type to “6408D+” and Port to “X”, and enter a descriptive Name.
<pre>
add station 54005   Page   1 of   4
                                     STATION

Extension: 54005   Lock Messages? n   BCC: 0
     Type: 6408D+   Security Code: TN: 1
     Port: X   Coverage Path 1:   COR: 2
     Name: SIP-54005   Coverage Path 2:   COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 2   Personalized Ringing Pattern: 1
             Data Module? n   Message Lamp Ext: 54005
            Speakerphone: 2-way   Mute Button Enabled? y
        Display Language: english

                                                    Media Complex Ext:
                                                         IP SoftPhone? n
</pre></td></tr>
</table>

Step	Description
3.	Enter the **change off-pbx-telephone station-mapping s** command, where "**s**" is the extension of the OPS station configured in Step 2. On Page 1 of the **off-pbx-telephone station-mapping** form, configure the following: • **Station Extension –** Enter the extension of the OPS station. • **Application** – Set to "**OPS**". • **Phone Number –** Enter the number that the ClearOne MAX IP phone will use for registration and call origination and termination. In the example below, the **Phone Number** is the same as the OPS **Station Extension**, but is not required to be the same. • **Trunk Selection –** Enter the number of the SIP trunk group connected to the Avaya SES server. • **Configuration Set –** Set to "**1**", which during compliance testing used the default values of the off-pbx-telephone configuration-set form.

	<pre>change off-pbx-telephone station-mapping 54005 Page 1 of 2
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	 Station Application Dial Phone Number Trunk Configuration
	 Extension Prefix Selection Set
	 54005 OPS - 54005 10 1</pre>

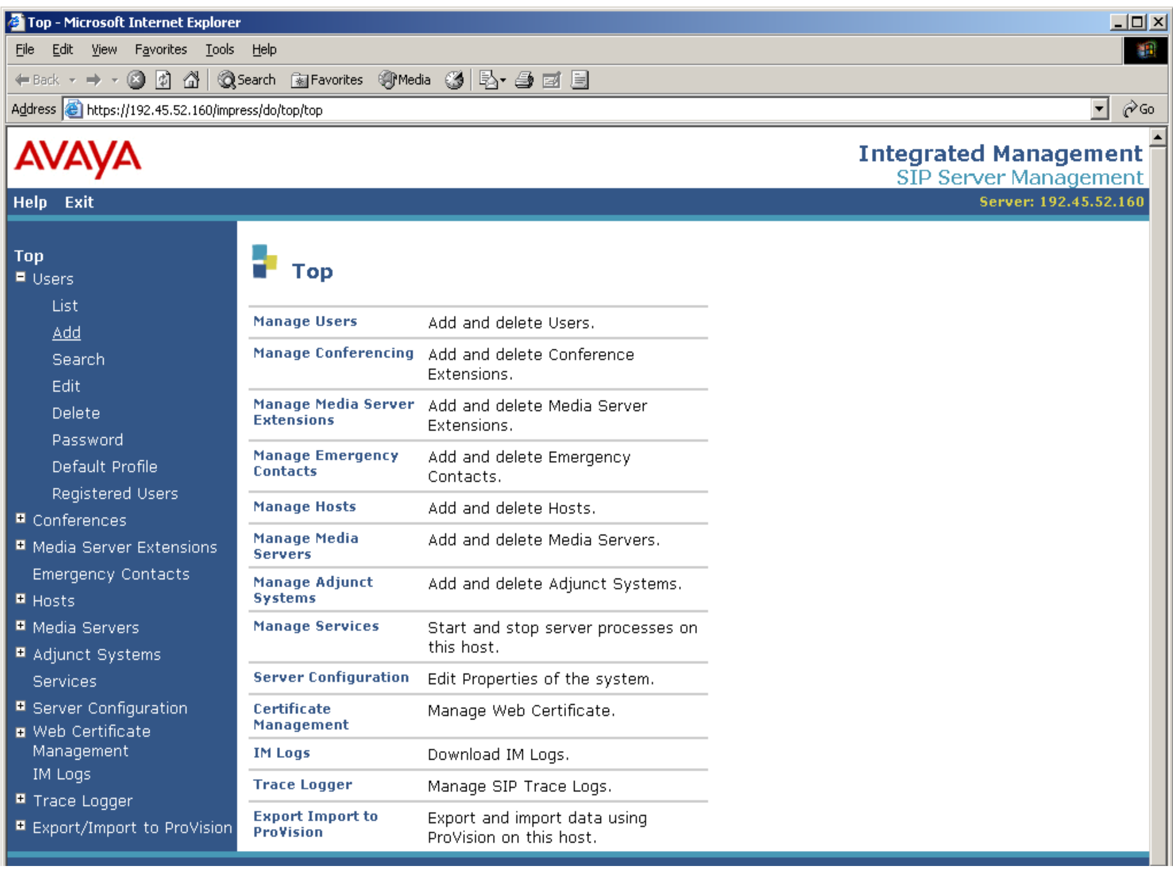
4.	Repeat Steps 2 – 3 as necessary to administer OPS stations and associations for additional ClearOne MAX IP phones.

4. Configure Avaya SIP Enablement Services

This section describes the steps for creating SIP user accounts in Avaya SIP Enablement Services (SES) and associating the SIP users with an Avaya Communication Manager OPS station extension. The ClearOne MAX IP phones will register with Avaya SES using the SIP user accounts.

This section assumes that the necessary Avaya SES configuration steps for establishing a SIP trunk with Avaya Communication Manager have been completed. For further details, consult [4].

Step	Description
1.	Open a web browser, enter http://<IP address of Avaya SES server>/admin for the URL, and log in with the appropriate credentials. Click on the "**Launch Administration Web Interface**" link upon successful login.
2.	In the left pane of the SES Administration Web Interface, expand "**Users**" and click on "**Add**".

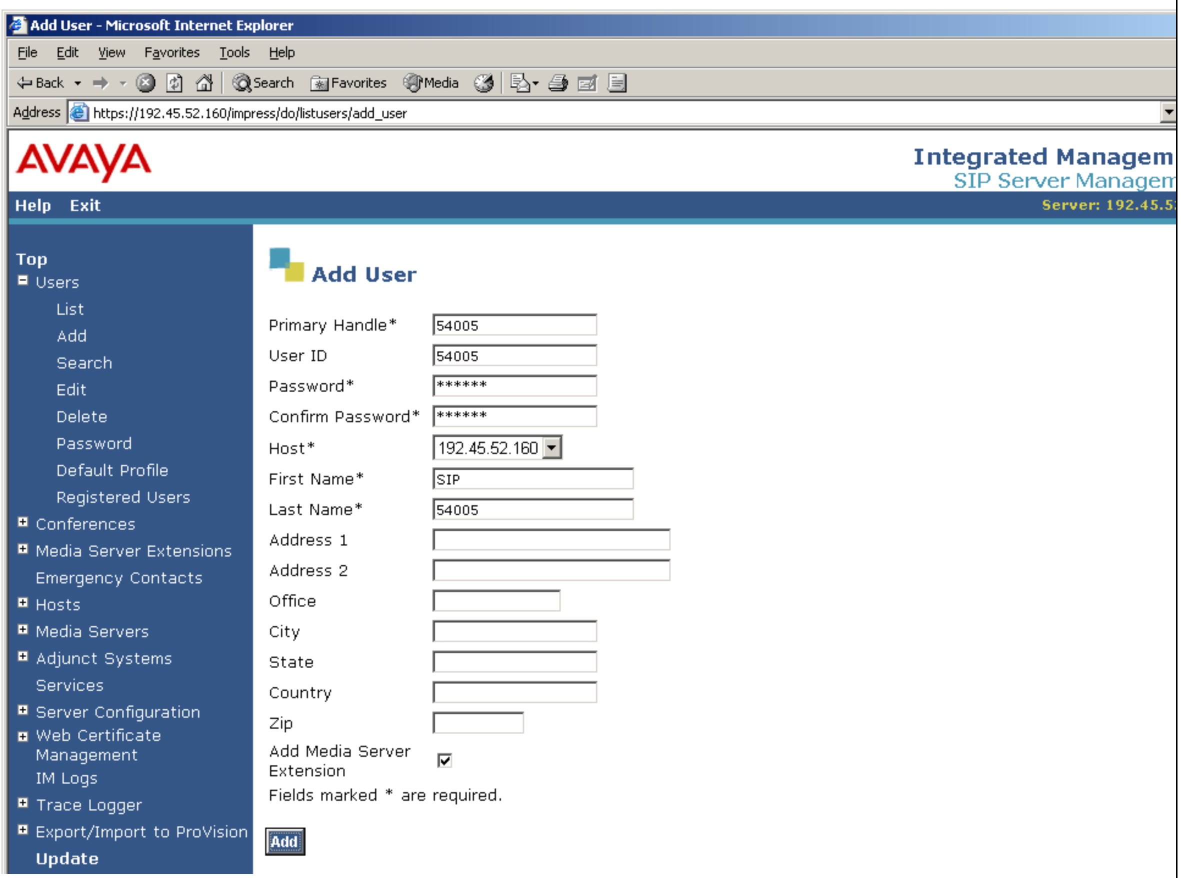


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

In the **Add User** page, configure the following:

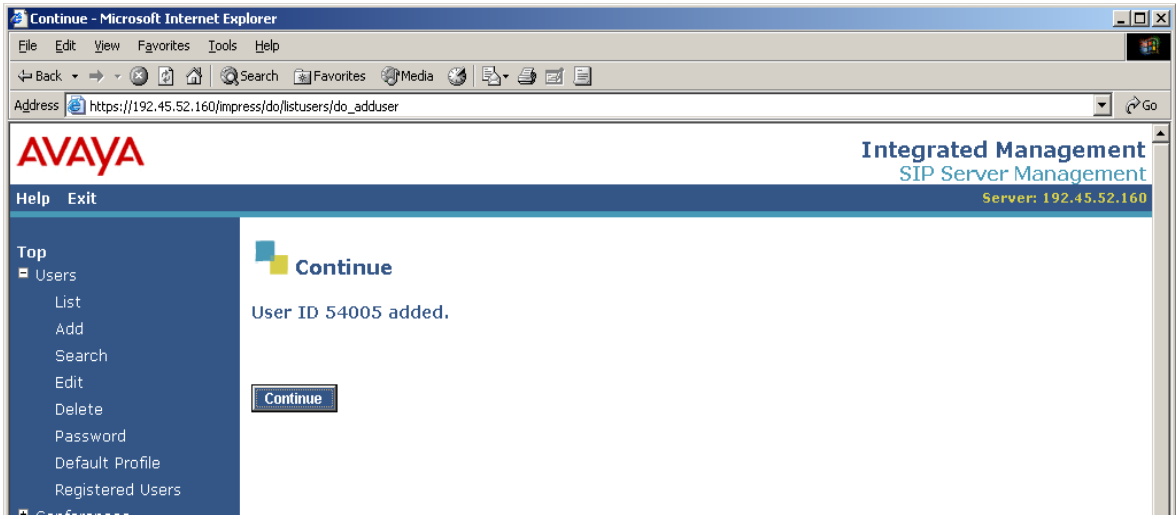
- **Primary Handle –** Enter the phone number of the ClearOne MAX IP phone. The number must match the phone number entered in Section 3.2 Step 3.
- **Password** and **Confirm Password –** Specify a password that the ClearOne MAX IP phone must use to successfully register with Avaya SES.
- **Host –** Select the IP address or FQDN of the Avaya SES server.
- **First Name** and **Last Name –** Enter descriptive names.
- Check the **Add Media Server Extension** checkbox.

Click on “**Add**”.

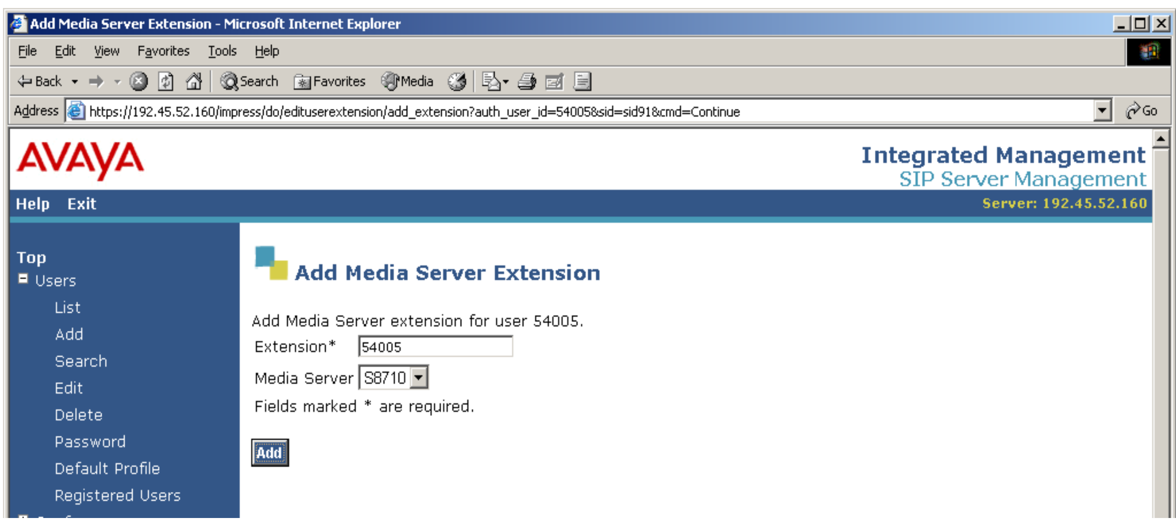


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>Click on “Continue”.

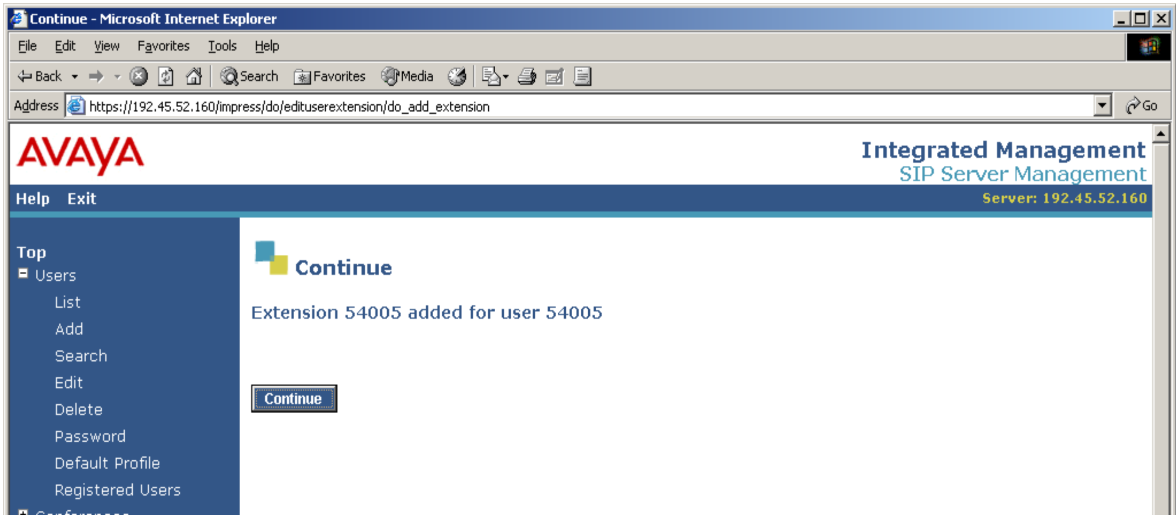


</td></tr>
<tr><td>5.</td><td>Enter the Extension of the corresponding Avaya Communication Manager OPS station configured in Section 3.2 Step 3 and select the Media Server on which the OPS station is configured. Calls from this user will always be routed to the selected Avaya Communication Manager media server for origination services. Click on “Add”.

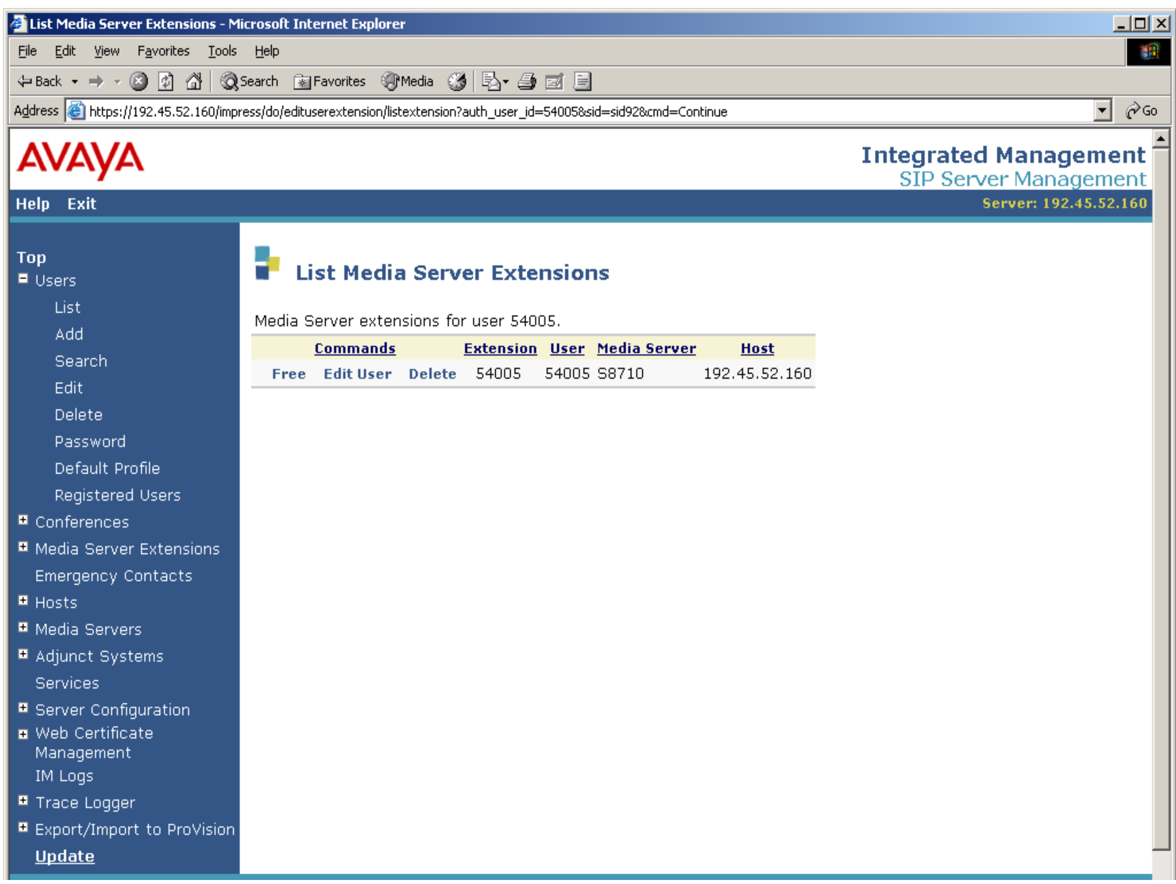


</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>Click on "Continue".



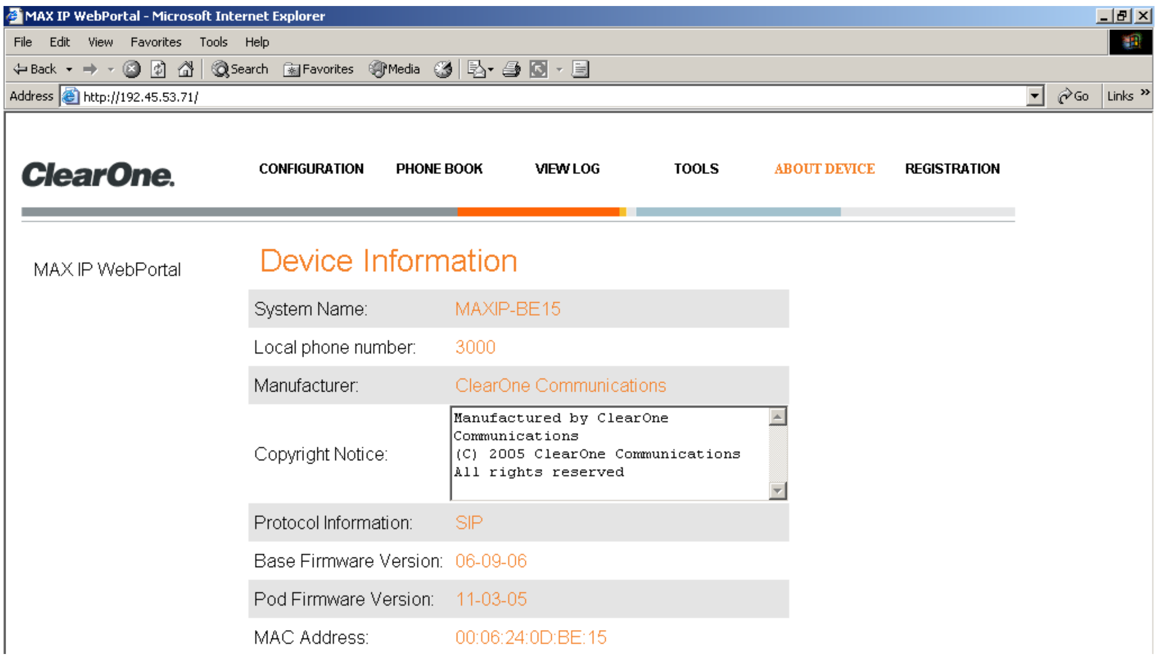
</td></tr>
<tr><td>7.</td><td>Click on "Update" at the bottom of the left pane.



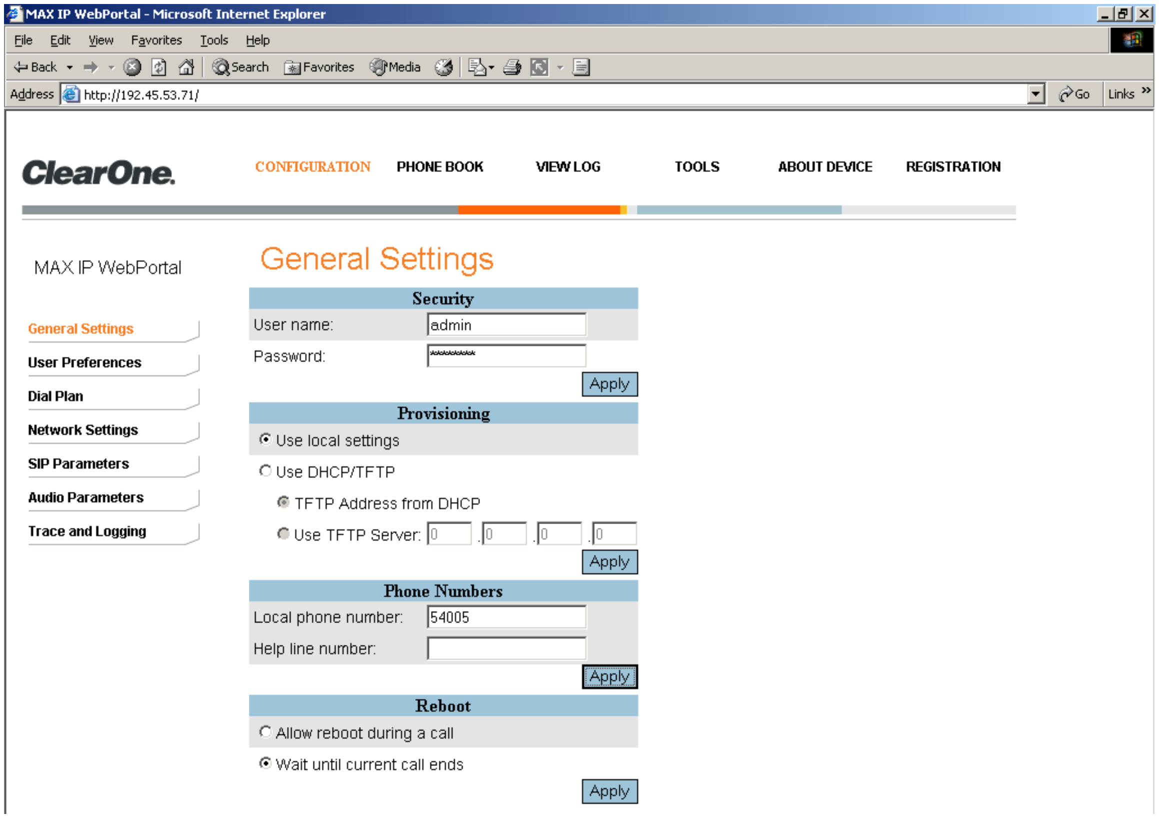
</td></tr>
<tr><td>8.</td><td>Repeat Steps 2 – 7 as necessary to configure SIP users for additional MAX IP phones.</td></tr>
</table>

5. Configure the ClearOne MAX IP Phone

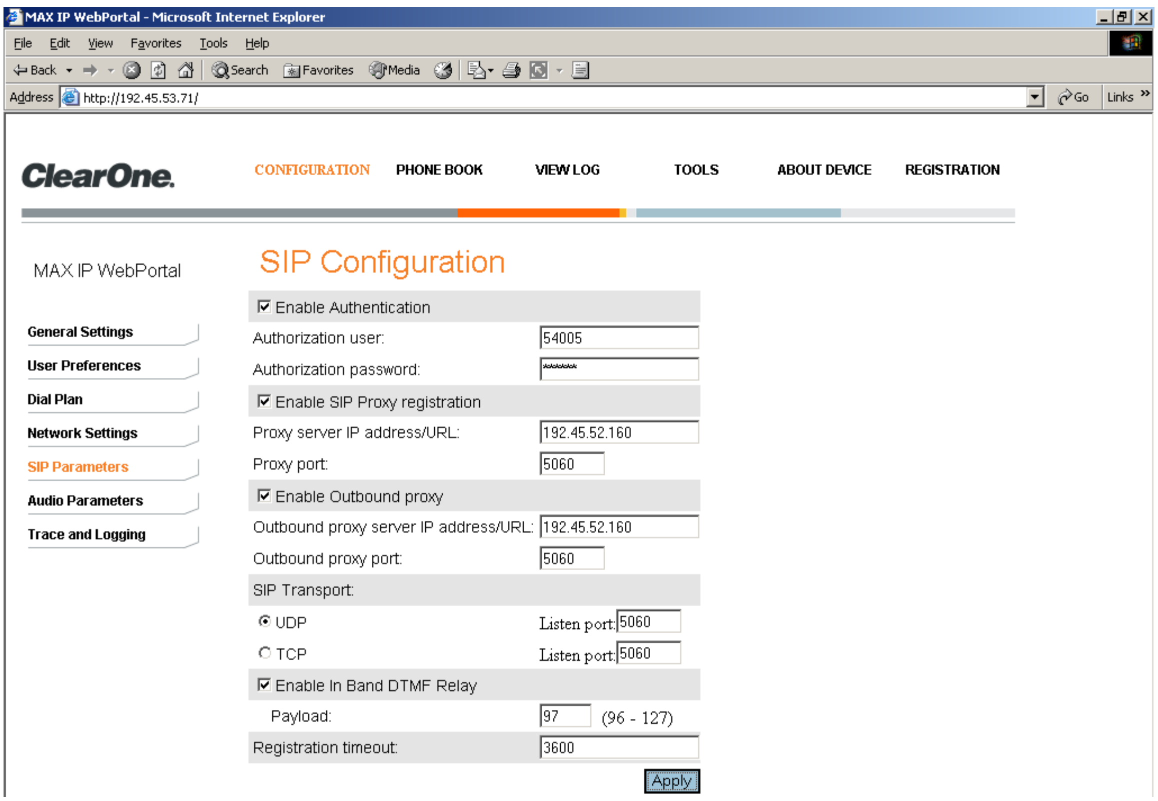
This section describes the steps for configuring the ClearOne MAX IP phone. This section assumes that the MAX IP phone has already been configured with an IP address.

Step	Description
1.	Open a web browser, enter http://a.b.c.d for the URL, where a.b.c.d is the IP address of the MAX IP phone, and log in with the appropriate credentials.
2.	Click on "**CONFIGURATION**". 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>For Local phone number, enter the SIP user (Primary Handle) created in Section 4 Step 3 and click on “Apply”. Click on “SIP Parameters” in the left menu.



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

Step	Description
4.	Check the "**Enable Authentication**", "**Enable SIP Proxy registration**", and "**Enable Outbound proxy**" checkboxes and configure the following parameters: • **Authorization user** and **Authorization password –** Enter the SIP user and password administered in Section 4 Step 3. • **Proxy server IP address/URL** and **Outbound proxy server IP address/URL –** Enter the IP address of the Avaya SES server. The remaining fields may be left at the defaults. Click on "**Apply**" and then "**Dial Plan**" in the left menu. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>

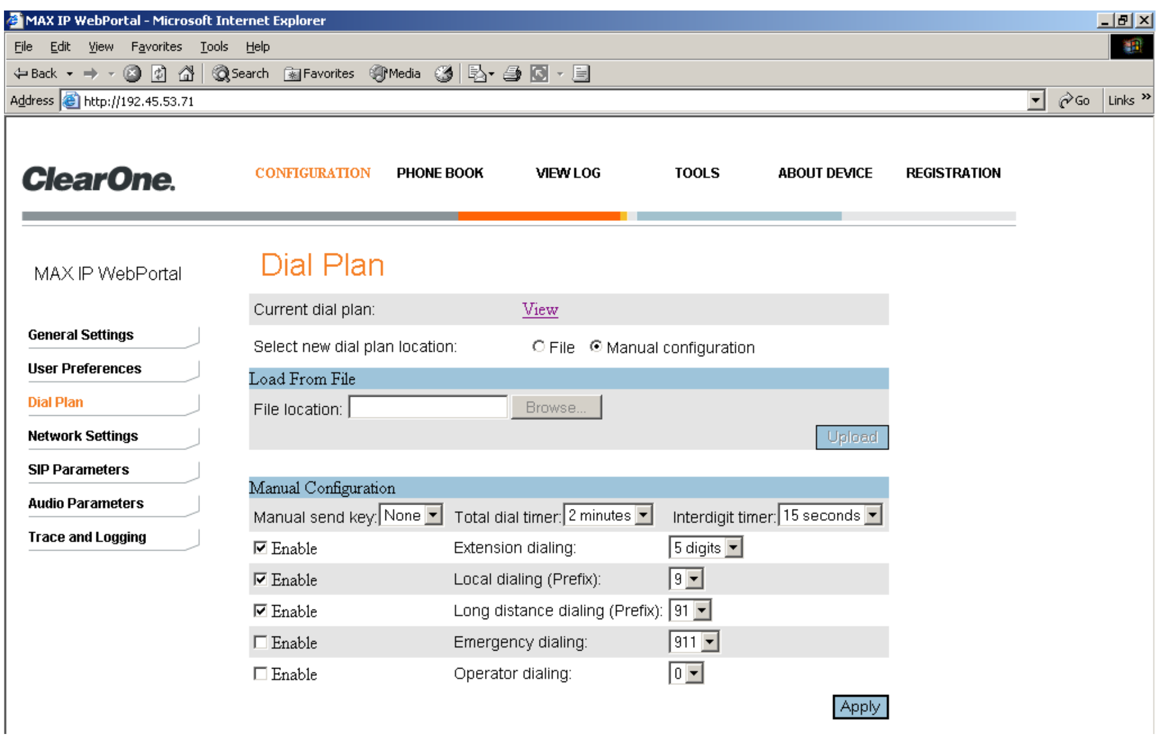
The MAX IP phone dial plan can be either loaded from a file or manually configured. Consult [6] for guidelines on creating the dial plan file or manually configuring the dial plan. The following briefly explains the settings in the screen below:

- **Manual send key** is set to "**None**" so that a "Send" key does not have to be entered after entering the destination number.
- **Extension dialing** is set to "**5 digits**" because 5-digit extensions were used.
- **Local dialing (Prefix)** and **Long distance dialing (Prefix)** are set to "**9**" and "**91**", respectively, because "9" was configured as the ARS Feature Access Code in Avaya Communication Manager.

Taken together, the settings mean that:

- If "9" followed by "1" is entered, then the MAX IP telephone will dial out after 10 more digits are entered.
- If "9" followed by any digit other than "1" is entered, then the MAX IP telephone will dial out after 7 more digits are entered. Note that this presents a problem where 10-digit dialing is required for local phone calls. In that case, a custom dial plan file must be created and loaded.
- If any digit other than "9" is dialed, then the MAX IP telephone will dial out after 4 more digits are entered.

Click on "**Apply**" and then "**Network Settings**" in the left menu.



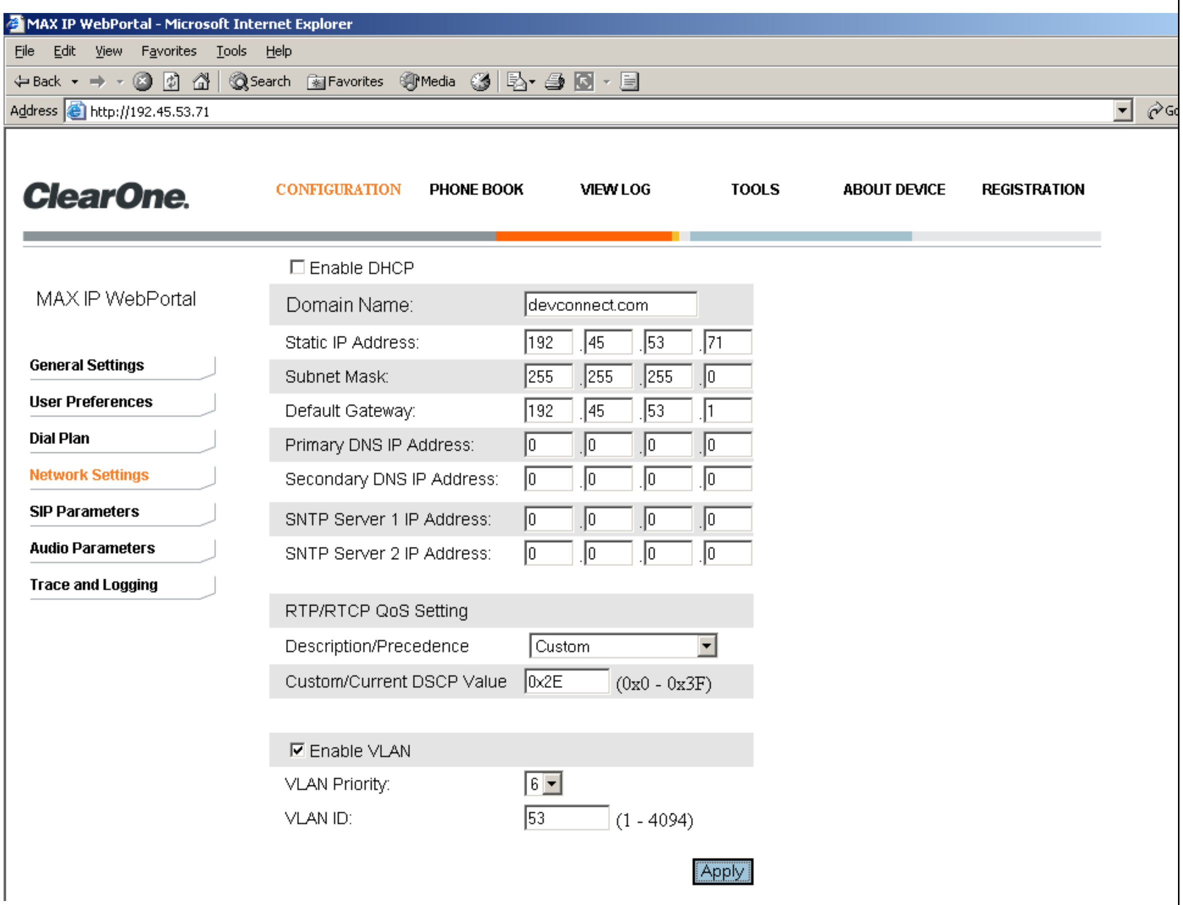
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>
To set 802.1p/Q QoS values in SIP and RTP frames originated by the MAX IP phone, configure the following:

Check the Enable VLAN checkbox.
VLAN Priority - Select an appropriate priority value in the range 0 – 7, where 7 is the highest priority.
VLAN ID - Enter the ID of the VLAN on which the MAX IP phone resides.

To set specific DiffServ[1] QoS values in RTP packets originated by the MAX IP phone, configure the following:

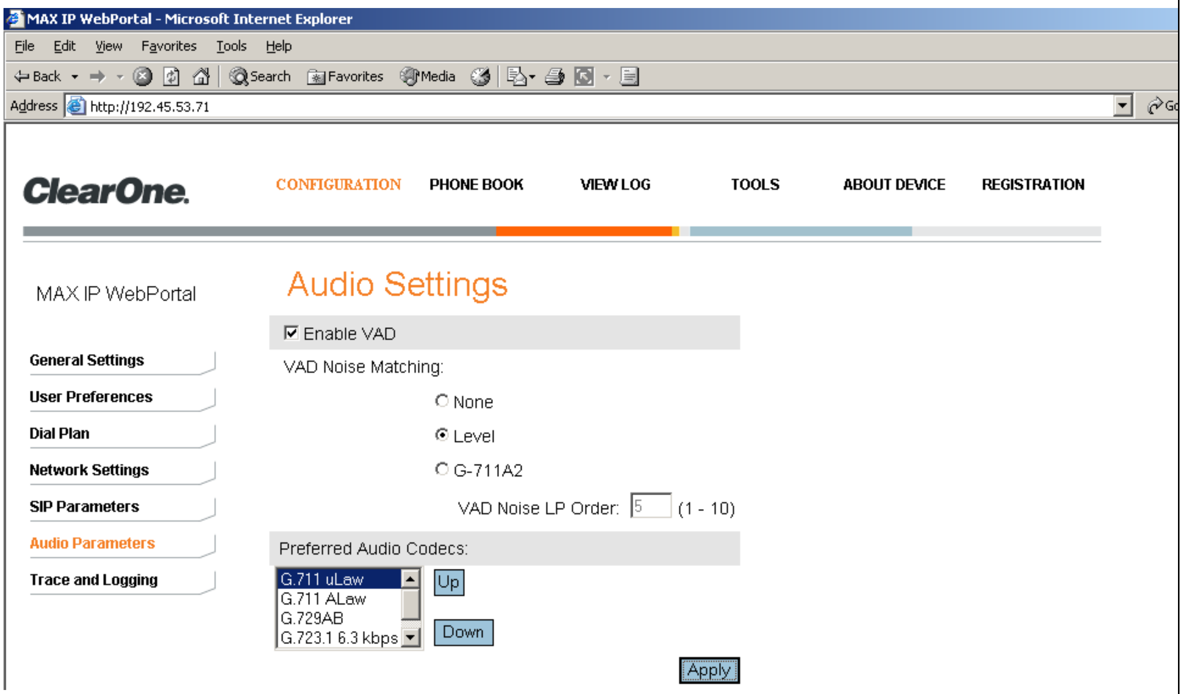
Description/Precedence - Select “Custom”.
Custom/Current DSCP Value - Enter the hexadecimal equivalent of the desired DiffServ QoS value for VoIP traffic. The DiffServ QoS value should be set in accordance with the customer’s QoS policies. In the example below, 0x2E is equivalent to the decimal value 46.

Click on “Apply” and then “Audio Parameters” in the left menu.


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

[1] The ClearOne MAX IP phone does not set DiffServ QoS values in SIP signaling packets.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Order the Preferred Audio Codecs list if necessary and click “Apply”. Codecs listed higher in the list have higher preference.



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

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on verifying call establishment on the ClearOne MAX IP phones, MAX IP phone operations such as dialing methods (manual, re-dial, and phone book), hold, mute, and conference, and MAX IP phone interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, and digital phones.

6.1. General Test Approach

The general test approach was to place calls to and from the MAX IP phones and exercise basic phone operations on the MAX IP phones. The main objectives were to verify that:

- The MAX IP phone successfully registers with Avaya SES.
- The MAX IP phone successfully establishes calls with Avaya SIP, H.323, and digital phones attached to Avaya SES or Avaya Communication Manager.
- The MAX IP phone successfully establishes calls with PSTN phones through Avaya Communication Manager.
- The MAX IP phone is able to hold and retrieve calls.
- The MAX IP phone successfully handles concurrent calls on its two lines and is able to switch between and bridge the two lines.
- The MAX IP phone successfully transmits DTMF during a call.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two MAX IP phones and two Avaya phones was formed as follows. A call was established between an Avaya phone and a MAX IP phone. The MAX IP phone then used its second line to establish a call with another MAX IP phone, and bridged the two lines together, forming a 3-party conference. The second MAX IP phone then used its second line to establish a call with another Avaya phone, and bridged its two lines together, effectively forming a 4-party conference.

6.2. Test Results

The test objectives of Section 6.1 were verified. For serviceability testing, the MAX IP phone operated properly after recovering from failures such as cable disconnects, and resets of the MAX IP phone, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours.

The following observations were made during testing:

- An unattended transfer where the transfer target is a MAX IP phone and the transferring phone is a SIP phone does not complete and causes the call to drop. The workaround is to perform attended transfers instead for such situations.
- When the MAX IP phone places a call on hold, the held party does not hear the holding party (MAX IP phone) as expected, but also does not hear Music On Hold.

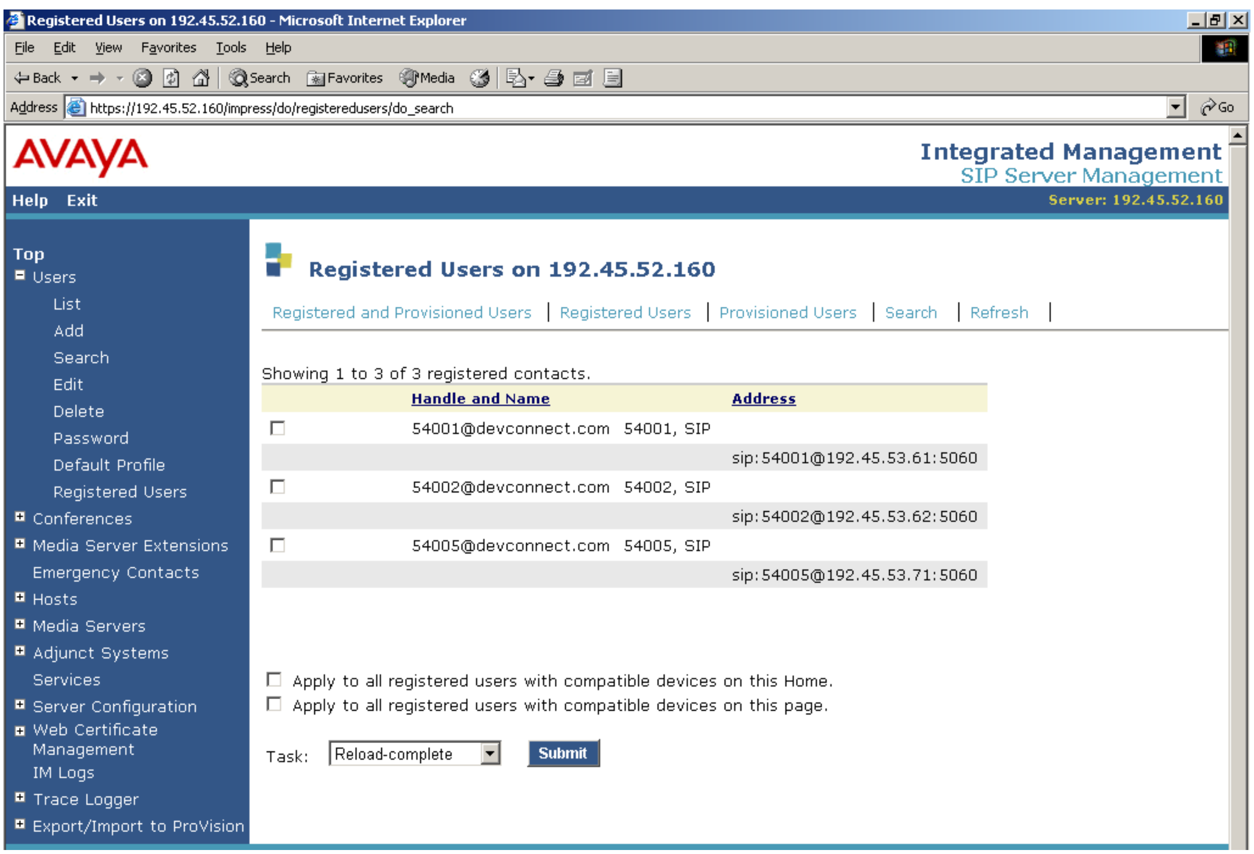
- When the MAX IP phone attempts to unregister, it does not respond to the authentication challenge from Avaya SES. Thus, Avaya SES continues to consider the MAX IP phone registered until the registration timer expires or the MAX IP phone registers again.
- The MAX IP phone codec list includes G.729AB (see screenshot in Section 5 Step 7), but advertises G.729 in the SIP signaling. Therefore, if G.729 is desired, G.729 must be included in the Avaya Communication Manager IP codec set list.
- The MAX IP phone does not operate properly when the MAX IP phone is configured to use TCP as the SIP transport protocol. Therefore, the MAX IP phone must use UDP (the default configuration) as the SIP transport protocol.

ClearOne Communications expects to resolve the above observations in future releases. Contact ClearOne Communications (www.clearone.com) for further updates.

7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the MAX IP phones successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the SES Administration Web Interface.



- Place calls to and from the MAX IP phones and verify that the calls are successfully established with two-way talk path.

8. Support

For technical support on ClearOne Communications MAX IP phones, consult the support pages at http://www.clearone.com/support or contact ClearOne Communications technical support at:

- Phone: 1.800.283.5936
- E-mail: tech.support@clearone.com

9. Conclusion

These Application Notes described a compliance-tested solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1, and ClearOne Communications MAX IP Tabletop Conferencing Phones. MAX IP phones are SIP-based VoIP tabletop conferencing phones intended for use in conference rooms and similar environments. During compliance testing, the MAX IP phones successfully registered with Avaya SES, placed/received calls to/from SIP and non-SIP telephones, and established 3-party conference calls.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 11, February 2006, Document Number 555-233-504
[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206
[4] *Installing and Administering SIP Enablement Services R3.1*, Issue 1.5, February 2006, Document Number 03-600768

Product documentation for ClearOne Communications products may be found at http://www.clearone.com.

[5] *MAX IP User's Guide*, January 2006 (Rev 1.0), Part No. 800-158-301
[6] *MAX IP Administrator's Guide*, January 2006 (Rev 1.0), Part No. 800-158-302

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 Developer*Connection* Program at devconnect@avaya.com.