Avaya Solution & Interoperability Test Lab

Application Notes for the Interwise Connect ITS Gateway Configuration with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services (SES), and Interwise Connect ITS which offers Interactive Voice Response/Conferencing Service. The Interwise Conferencing Service is a SIP-based VoIP audio conferencing solution intended for use by large enterprises and conference service providers. During compliance testing, the Interwise Conferencing Service successfully interacted with Avaya SES as a trusted host and was able to setup conferences between SIP and non-SIP telephones. 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.

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1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and Interwise Connect's ITS (Interactive Voice Response/Conferencing Service) 7.2.78. Interwise ITS server which offers the Conferencing Service is part of the Interwise Connect platform, a SIP-based VoIP conferencing solution intended for use by large enterprises and conference service providers. During compliance testing, the Interwise Conferencing Service was able to setup conferences between SIP and non-SIP telephones. The Conferencing Service was configured as a trusted host with Avaya SES.

Figure 1 illustrates a sample configuration consisting of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the Interwise Conferencing Service. Avaya Communication Manager was installed on S8710 Media Servers. 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 conference call setup between the SIP-based Interwise Conferencing Service and Avaya SIP, H.323, and digital phones. The analog PSTN phone is also included to demonstrate calls routed by Avaya Communication Manager to the Interwise Conferencing Service.

The conference call originates from any of the phones. If the originator is a SIP phone, the call is routed via Avaya SES server over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for Interwise Conferencing Service, Avaya Communication Manager routes the call back over the SIP trunk to the Avaya SES server, which in turn delivers the call to the Interwise Conferencing Service. For all non-SIP calls arriving directly at Avaya Communication Manager destined for Interwise Conferencing Service, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server, which in turn delivers the call to the Interwise Conferencing Service. Interwise Conferencing Service then uses Interactive Voice Response (IVR) service for authentication before the originator can join the conference.

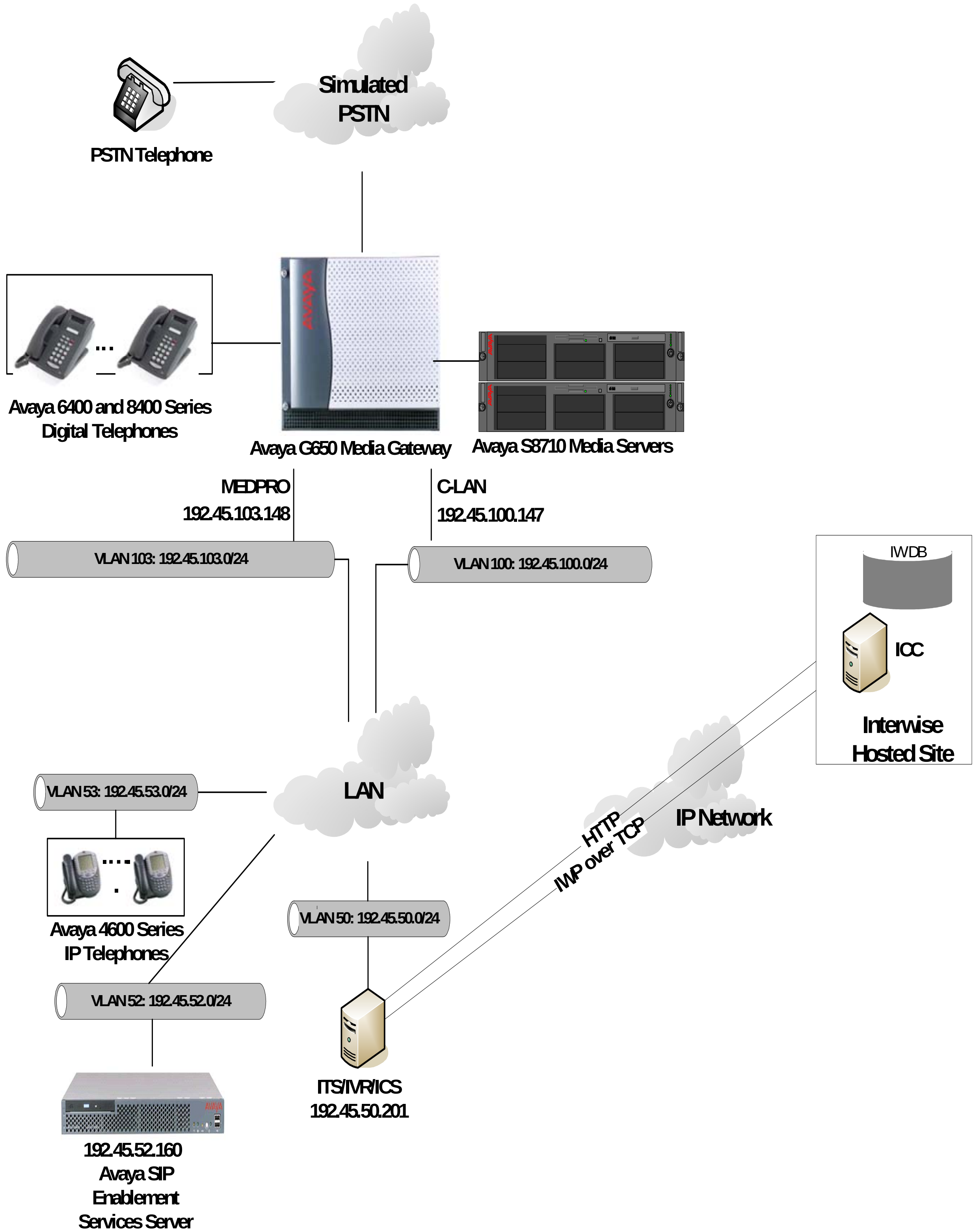

Figure 1: Sample configuration.

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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	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.3 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
Interwise IVR/Audio Conferencing Server	Interwise Connect 7.2.78
Analog Telephone	-

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3. Configure Avaya Communication Manager

This section describes the steps for configuring Avaya Communication Manager to route the calls properly for interaction with Interwise Conferencing Service via Avaya SES. System Access Terminal (SAT) interface is used to configure IP Codec Set, SIP signaling and trunking between Avaya Communication Manager and Avaya SES and setting up the dialplan for routing the calls destined for Avaya SES properly.

3.1. IP Code Set

This section describes the steps for administering the codec set in Avaya Communication Manager. This codec set is used in the IP Network Region for communications between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Enter the change ip-codec-set <c> command, where c is a number between 1 and 7, inclusive. IP codec sets are specified in the IP Network Region forms to define which codecs may be used within and between network regions. Following fields are set on this form:
• Audio Codec – Set to G.711MU</td></tr>
<tr><td><pre>
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:
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
</pre></td></tr>
<tr><td></td></tr>
</table>

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3.2. IP Network Region

This section describes the steps for administering the IP Network Region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the change ip-network-region <n> command, where n is a number between 1 and 250, inclusive and administer settings as per below.

Codec Set – Set to Codec Set as provisioned in Section 3.1.
Authoritative Domain – Set to the same value as SIP Domain on Avaya SIP Enablement Services Section 4, step 2.
Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SIP Enablement Services.

<pre>
change ip-network-region 2   Page 1 of 19
                               IP NETWORK REGION
  Region: 2
Location: Authoritative Domain: devconnect.com
    Name:
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
      Codec Set: 2   Inter-region IP-IP Direct Audio: yes
   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
</pre></td></tr>
</table>

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Step	Description
2.	Proceed to Page 3 of the **IP NETWORK REGION** form and enable inter-region connectivity between regions as per below. For purpose of these application notes, **src rgn 2** and **dst rgn 2** use **codec set 2** as configured in **Section 3.1**.

```
Page 3 of 19
                     Inter Network Region Connection Management

 src dst  codec  direct   Total           Video                          Dyn
 rgn rgn   set    WAN  WAN-BW-limits  WAN-BW-limits Intervening-regions CAC IGAR
 2   1     2      y        :NoLimit                                         n
 2   2     2
 2   3
 2   4
 2   5
 2   6
 2   7
 2   8
 2   9
 2   10
 2   11
 2   12
 2   13
 2   14
 2   15
```

3.3. IP Node Names

This section describes the steps for setting IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
1.	Issue the command "**change node-names ip**"; and administer settings as per below. • Add a node name for Avaya SIP Enablement Services along with the IP address • Verify that node-names are configured for the *C-LAN* and *MEDPRO* boards.

```
change node-names ip                                            Page 1 of 1
                                  IP NODE NAMES
    Name                IP Address
CLAN-1A06             192.45 .100.147
MEDPRO-1A13           192.45 .103.148
SES                   192.45 .52 .160
```


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3.4. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Issue the command "add signaling-group <s>", where s is an unallocated Signaling Group; and administer settings as per below.

Group Type – Set to sip.
Transport Method – Set to tls.
Far-end Listen Port – Set to 5061(default)
Near-end Node Name - Set to CLAN IP Address as displayed in Section 3.3.
Far-end Node Name - Set to IP Address of SES configured in Section 3.3.
Far-end Network Region - Set to the IP Network Region configured in Section 3.2.
Far-end Domain: Set to the same value as SIP Domain on Avaya SIP Enablement Services Section 4, step 2
</td></tr>
<tr><td><pre>
add signaling-group 10   Page   1 of 5
                                SIGNALING GROUP

 Group Number: 10   Group Type: sip
                       Transport Method: tls

   Near-end Node Name: CLAN-1A06   Far-end Node Name: SES
 Near-end Listen Port: 5061   Far-end Listen Port: 5061
                                        Far-end Network Region: 2
       Far-end Domain:devconnect.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload   Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
</pre></td></tr>
<tr><td></td></tr>
</table>

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3.5. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	Issue the command "**display system-parameters customer-options**", and proceed to Page 2. Verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses. **Note**: *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.*

```
Page 2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 200   148
           Maximum Concurrently Registered IP Stations: 1000  2
             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: 200   153

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     1
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 2     0
          Maximum TN2602 Boards with 320 VoIP Channels: 2     1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```


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<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Issue the command "add trunk-group <t>", where t is an unallocated Trunk Group; and administer settings as per below.

Group Type – Set to same value as Group Type configured in Section 3.4.
TAC(Trunk Access Code) – Set to any number with 1-4 digits;* and # may be used as first digit only.
Signaling Group – Set to same value as Group Number configured in Section 3.4.
Number of Members – Set to a value between 0 and 255.

<pre>
add trunk-group 10   Page   1 of 21
                                TRUNK GROUP

Group Number: 10   Group Type: sip   CDR Reports: y
  Group Name: SIP-SES-DevCon1   COR: 1   TN: 1   TAC: 110
   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: 150
</pre>
</td></tr>
</table>

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3.6. Dialplan/AAR/Route Pattern

This section describes the steps for setting the Dialplan, AAR digit analysis and Route Pattern in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES. These calls are ultimately destined for the Interwise Conferencing Service.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Issue the command “change dialplan analysis”; and add the following entries:

Dialed String – Set it to a value for routing call to Avaya AES for proper AAR digit analysis
Total Length – The dialed string and length determine whether AAR digit analysis is required.
Call Type – Set to aar

<pre>
change dialplan analysis   Page 1 of 12
                             DIAL PLAN ANALYSIS TABLE
                                                         Percent Full: 2

      Dialed Total Call   Dialed Total Call   Dialed Total Call
      String Length Type   String Length Type   String Length Type
        0   1   fac
        1   3   dac
        100   4   aar
        2   4   ext
        2   5   aar
        3   5   ext
        4   5   ars
        5   5   ext
        6   5   ext
        7   5   ext
        8   1   fac
        9   1   fac
        *   3   fac
        #   3   fac
</pre>
</td></tr>
</table>

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Step	Description
2.	Issue the command “**change aar analysis 1**”; and add the following entry: • **Dialed String –** Set it to same value as **Dialed String** in **Section 3.6, Step 1**. • **Total Min and Max –** Set it to same value as **Total Length** in **Section 3.6, Step 1.** • **Route Pattern –** Set a value for a route pattern defined in **Section 3.6, Step 3**. • **Call Type –** Set to **aar** • **ANI Reqd –** Set to **n**

```
Page 1 of 2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    100                    4    4     10      aar           n
    2                      5    5     15      aar           n
    2                      7    7     999     aar           n
    245                    5    5     33      aar           n
```

Step	Description
3.	Issue the command “**change route-pattern <r>**”, where **r** is the number of the route pattern to be administered. • **Grp No –** Set to the Trunk Group provisioned in **Section 3.5**. • **FRL –** Set to **0**.

```
Change route-pattern 10                                           Page   1  of  3
                    Pattern Number: 1   Pattern Name: SES SIP
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 10   0                                                                 n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                            none
 2: y y y y y n  n            rest                                            none
 3: y y y y y n  n            rest                                            none
 4: y y y y y n  n            rest                                            none
 5: y y y y y n  n            rest                                            none
 6: y y y y y n  n            rest                                            none
```


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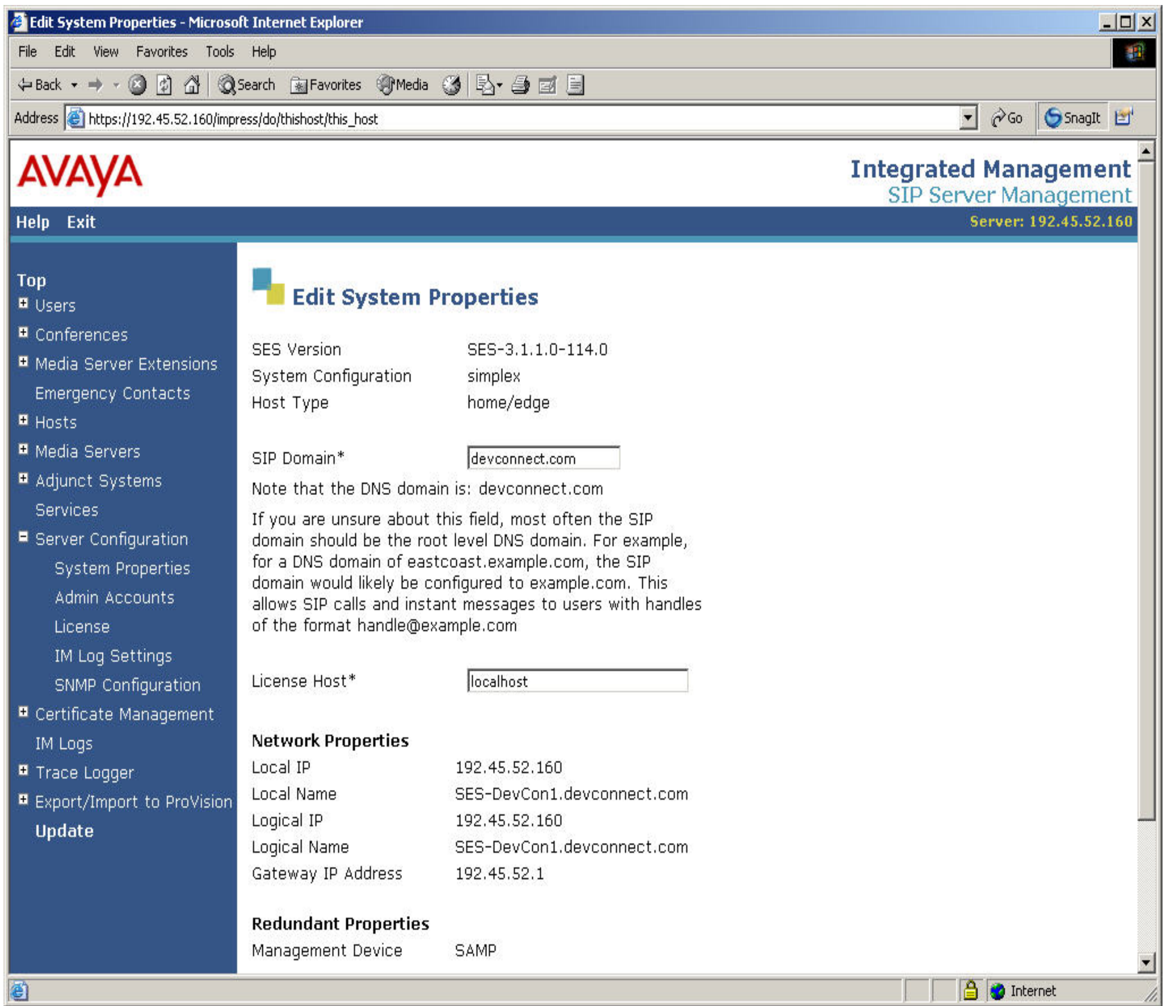
4. Configure Avaya SIP Enablement Services

This section describes steps for creating Media Server entries for communication between Avaya Communication Manager and Avaya SES. Once the media server entry is created, the Host Address Map entry along with the contact information for Interwise Conferencing Service was created in Avaya SES. Additionally, Interwise Conferencing Service is configured as a trusted host in Avaya SES.

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.	From the **Administration Web Interface**: • Click the **+** sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Authoritative Domain** configured for the **IP NETWORK REGION** on Avaya Communication Manager in **Section 3.2**.



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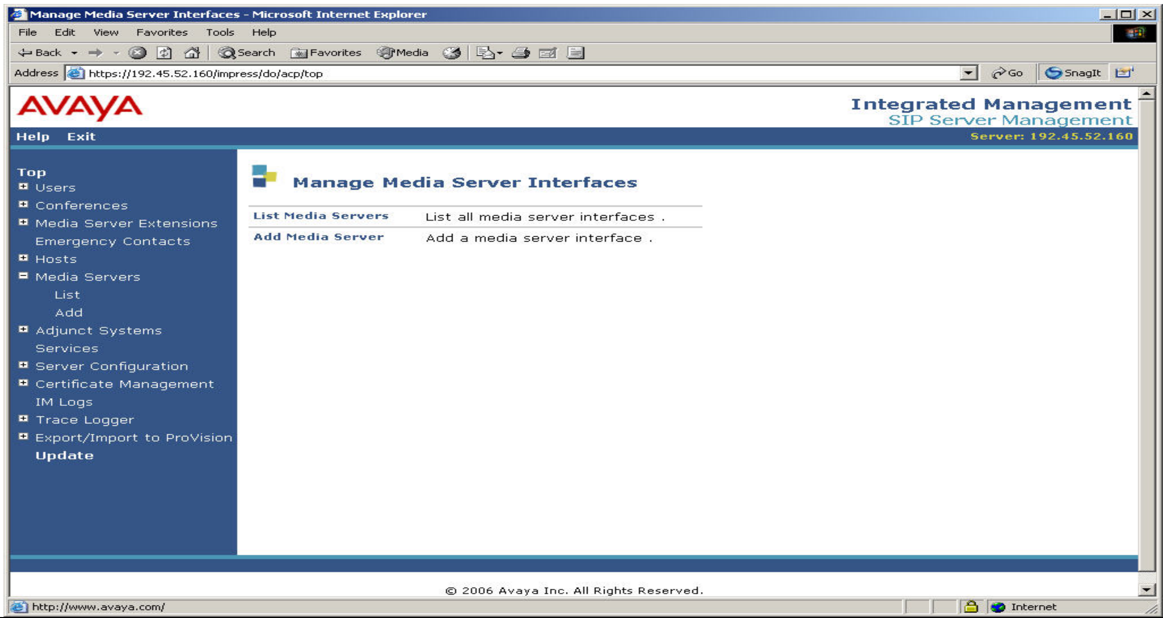
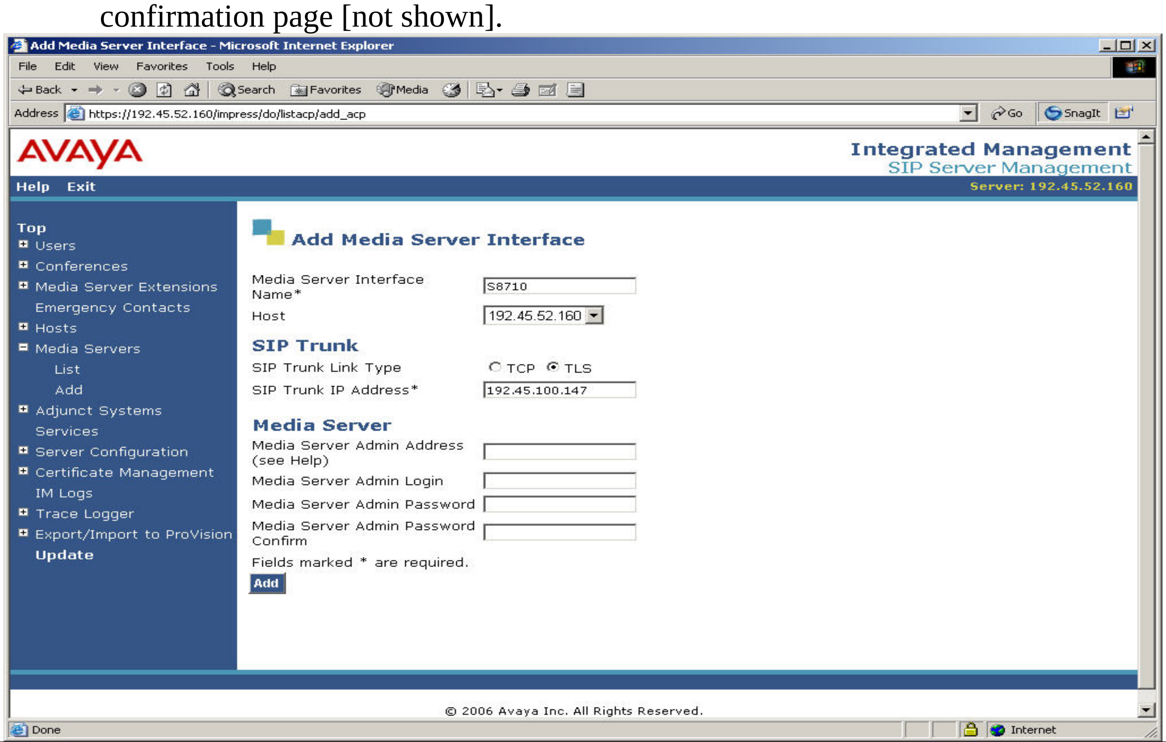
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a Media Server corresponding to Avaya Communication Manager from the Administration Web Interface:

Click the + sign to expand the options under Media Servers.
Click Add.



</td></tr>
<tr><td>4.</td><td>
At the Add Media Server Interface page, provision SIP Trunk parameters as follows for connectivity to Avaya Communications Manager:

SIP Trunk Link Type - Set to same value as Transport Method in Section 3.4.
SIP Trunk IP Address - Set to same value as CLAN address in Section 3.3.
Click the Add button when finished and hit the Continue button on the confirmation page [not shown].



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

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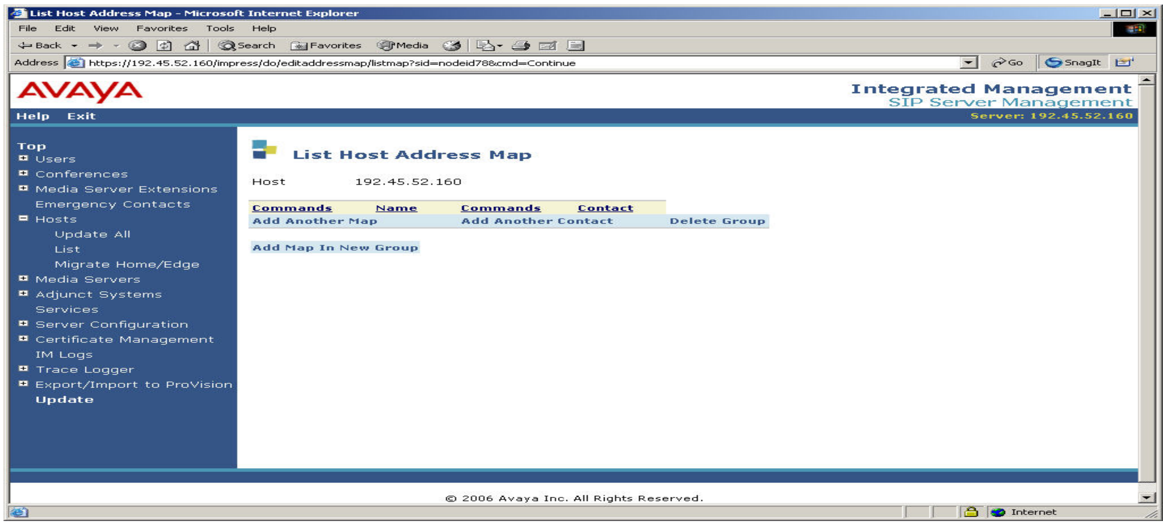
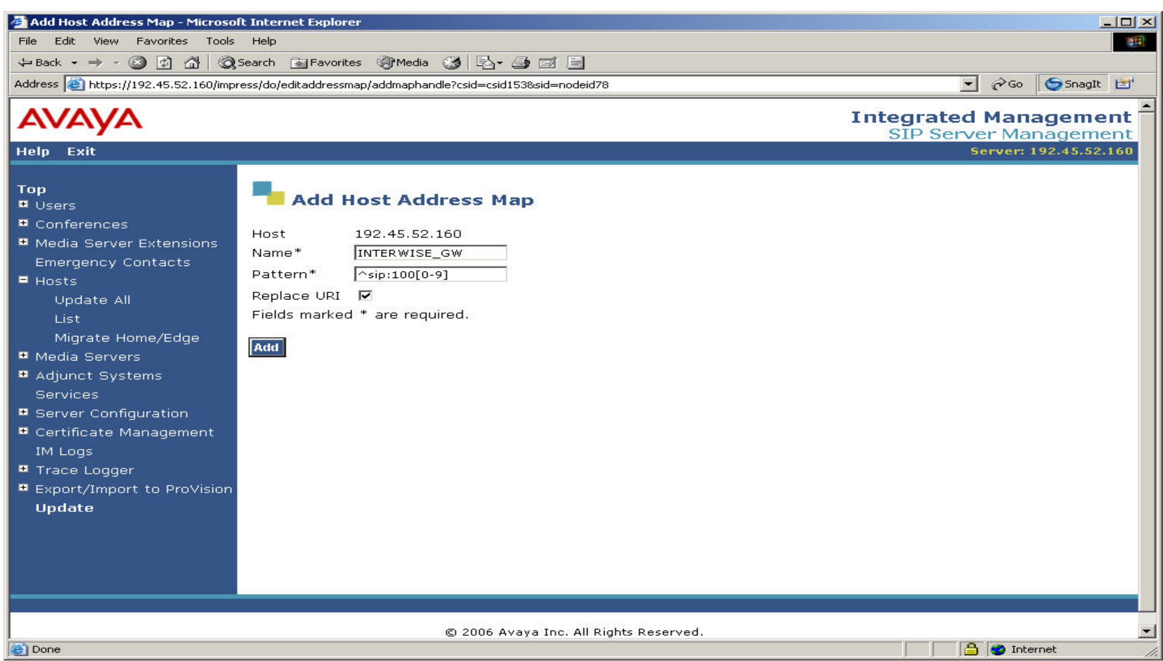
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>
A Host Address Map is required on Avaya SES to direct outbound calls from Avaya Communication Manager to Interwise Conferencing Service. An Address Map is used to route the calls based on the contents of SIP INVITE URI. To configure Host Address Map, do the following:

Click the + sign to expand the options under Hosts.
Click on Add Another Map.



</td></tr>
<tr><td>6.</td><td>
On the Add Host Address Map page enter the following:

Name – Descriptive Name
Pattern – Expression to match the beginning of URI.
Click on Add to add the map



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

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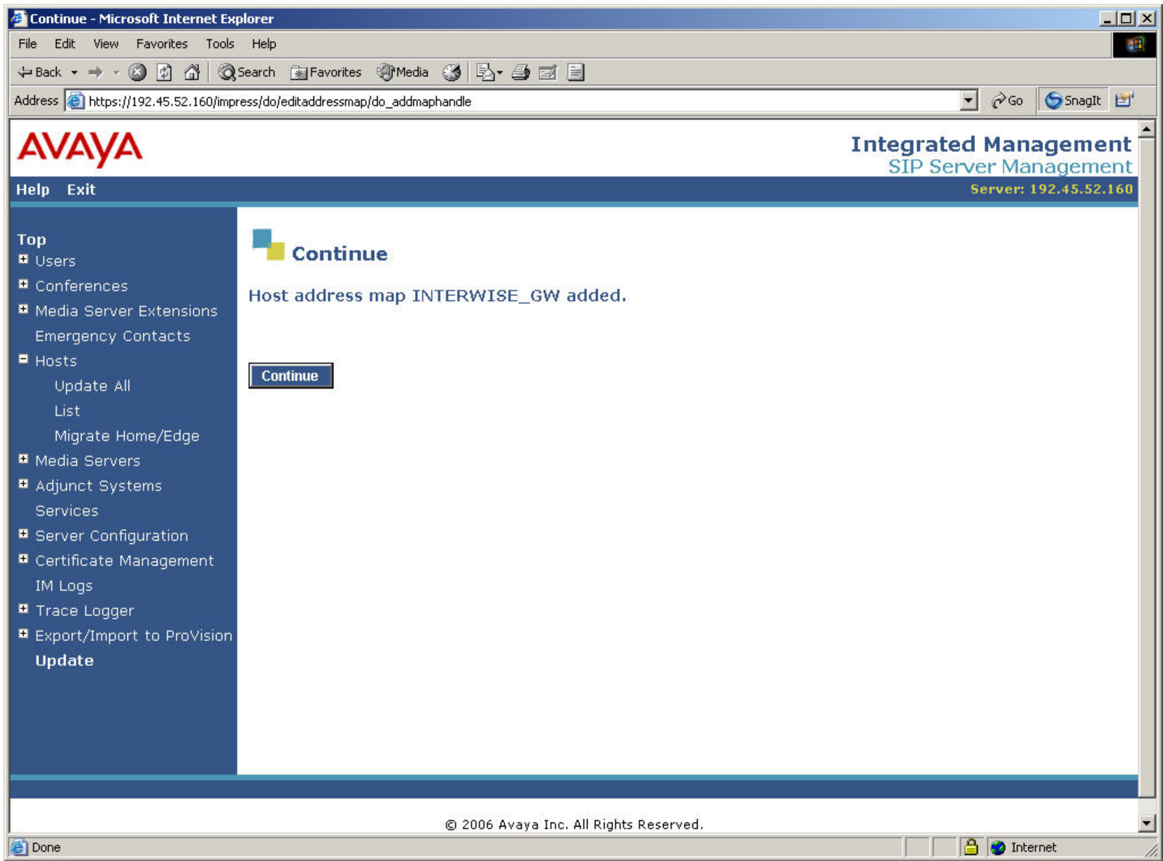
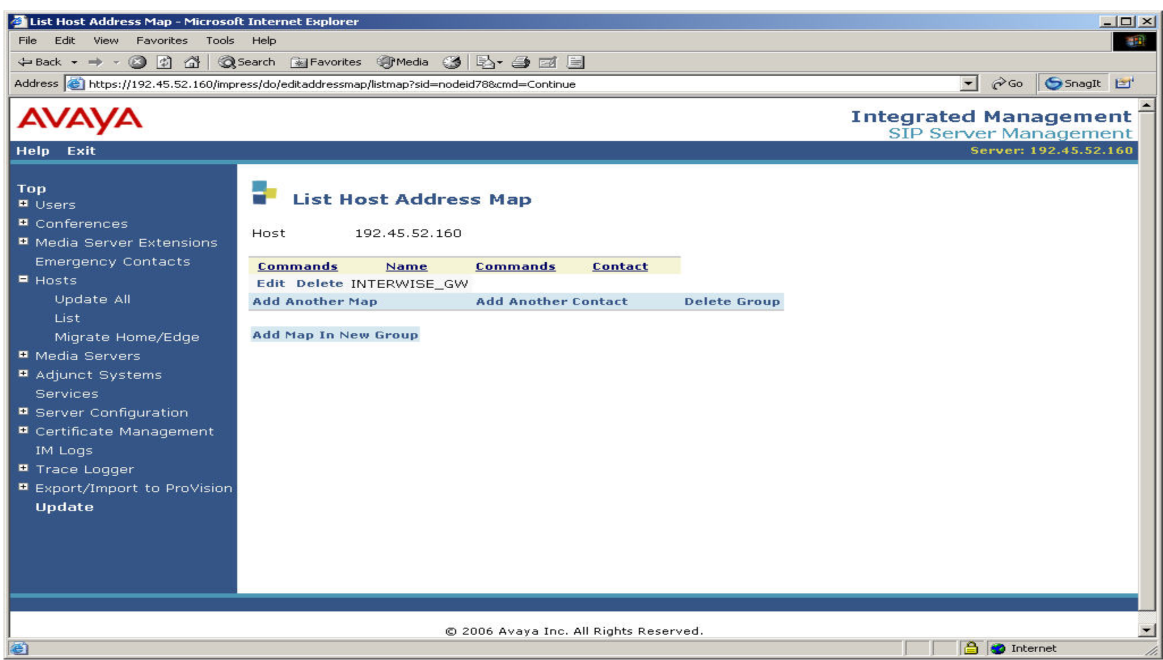
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Click on Continue.



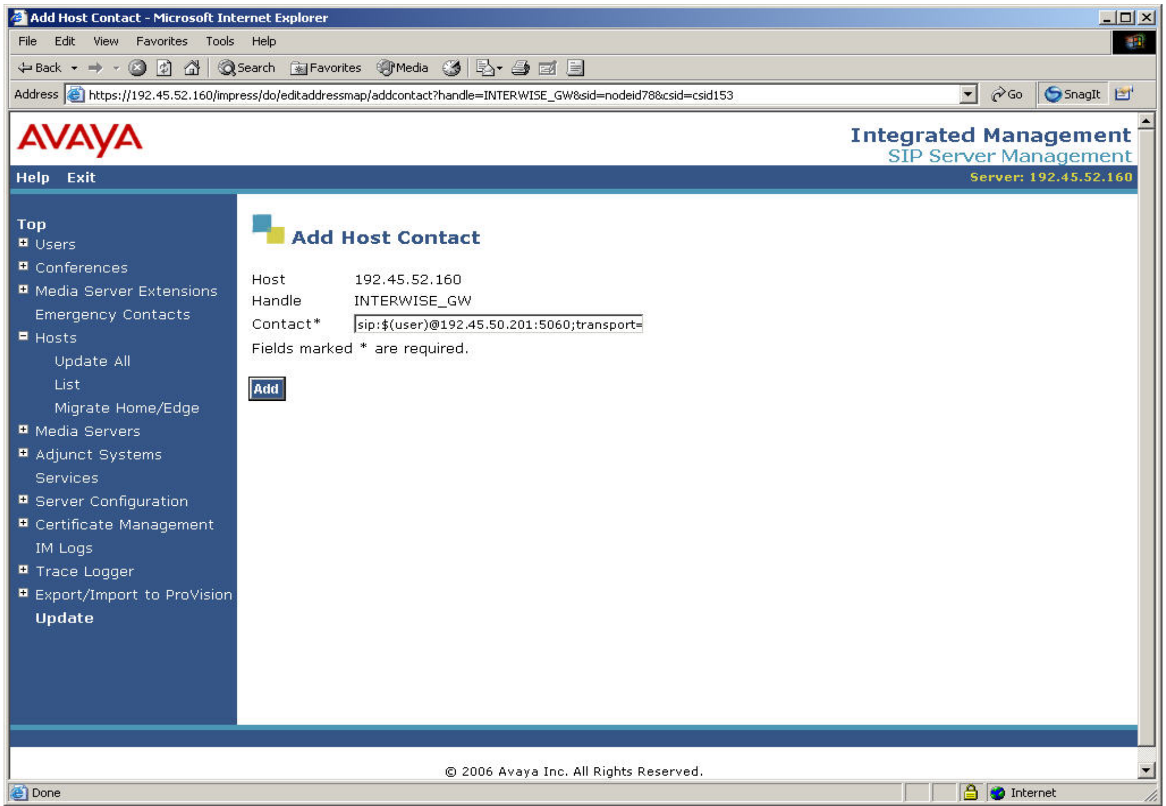
</td></tr>
<tr><td>8.</td><td>On the List Host Address Map page, click on Add Another Contact.

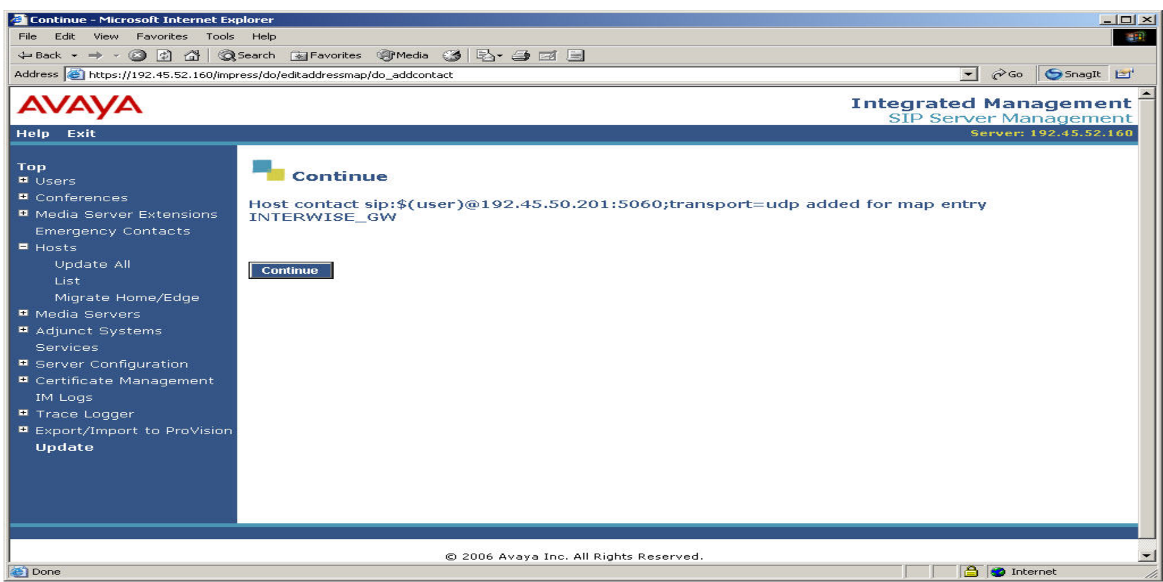


</td></tr>
</table>

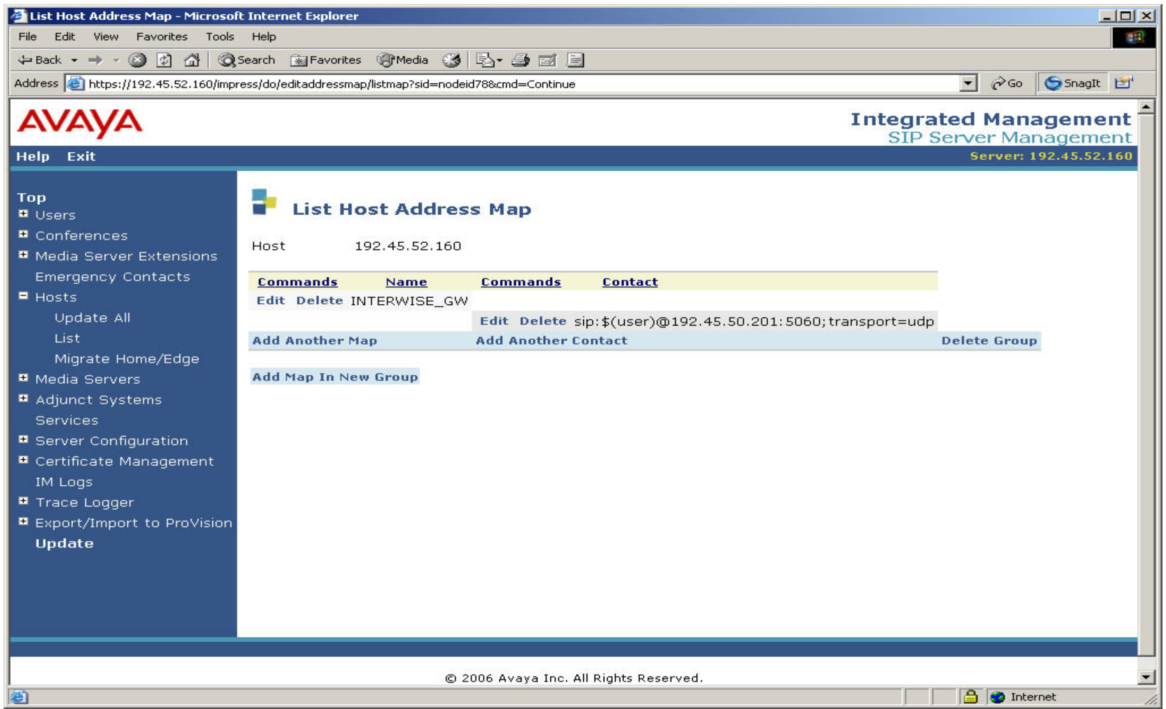
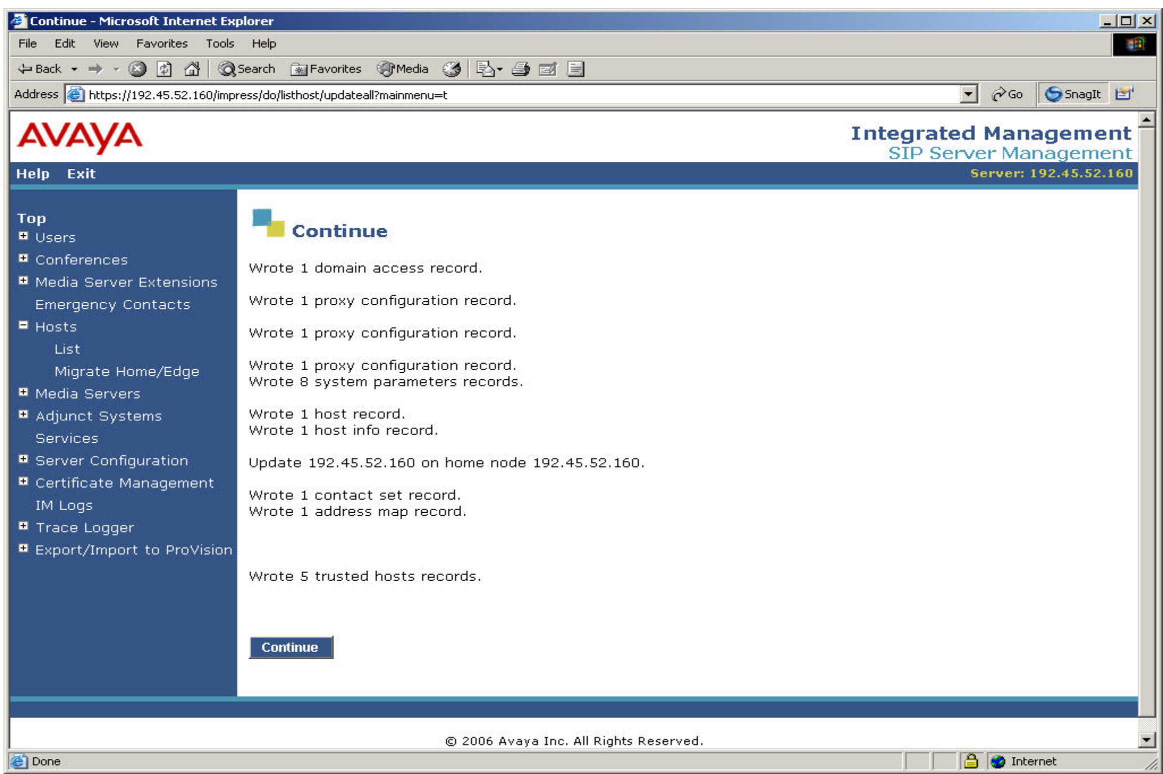
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>The host contact must be entered for the Address Map defined in Section 4, step 6.

Contact – Enter the destination IP address (ip_addr), port number (port) and transport protocol (protocol) as follows:
sip:$(user)@ip_addr:port;transport=protocol
Click on Add to configure this contact.



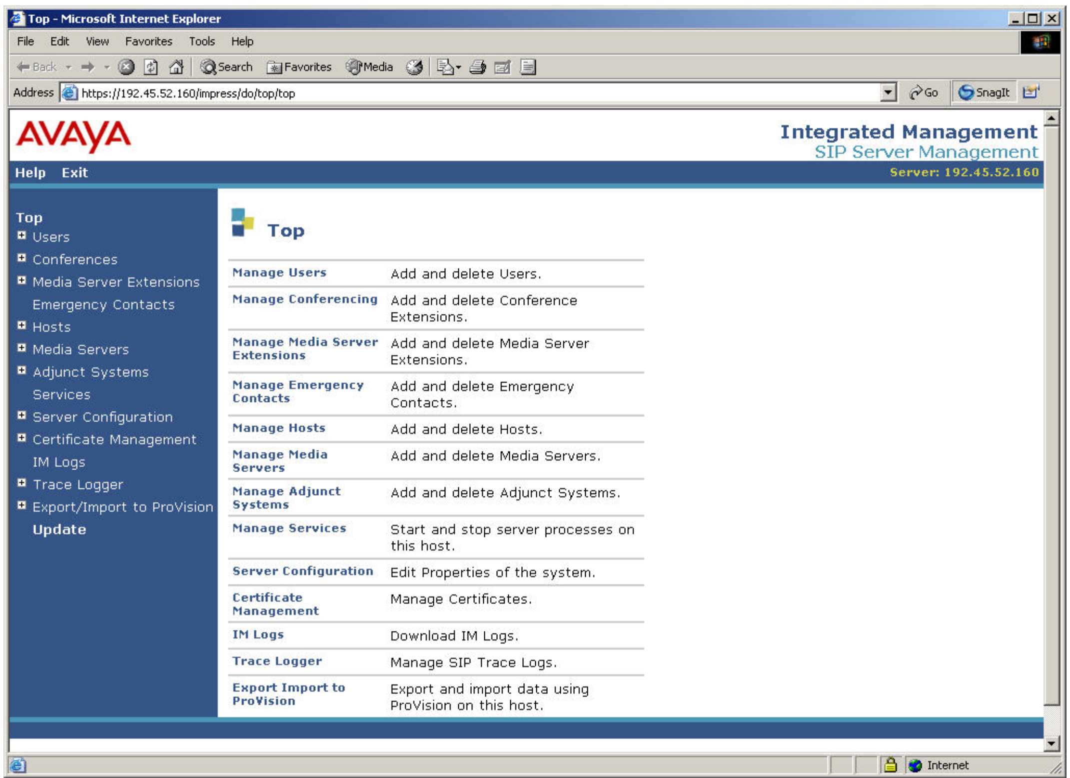
</td></tr>
<tr><td>10.</td><td>Click on Continue


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>11.</td><td>This screen displays the completely configured Host Address Map. Click on Update in the left pane to effect these changes.

</td></tr>
<tr><td>12.</td><td>Click on “Continue” at the bottom of the right pane.

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>
Lastly, the IP Address of the Interwise Conferencing Service (ICS) must be configured as a trusted host on Avaya SES. As a trusted host, Avaya SES will not issue SIP authentication challenges for incoming requests from the IP address of Interwise Conferencing Service.
The following steps are required to configure Interwise Conferencing Service as a trusted host on Avaya SES:

Connect to Avaya SES using secure shell and log in using proper credentials.
Issue the trustedhost command at the Linux shell prompt.
trustedhost -a ICS-IP-address -n SES-IP-address [-c 'comment text']
Important Step: Complete the trusted host configuration by returning to the Avaya SES administration web page and click on Update link in the left pane on the Top screen.



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

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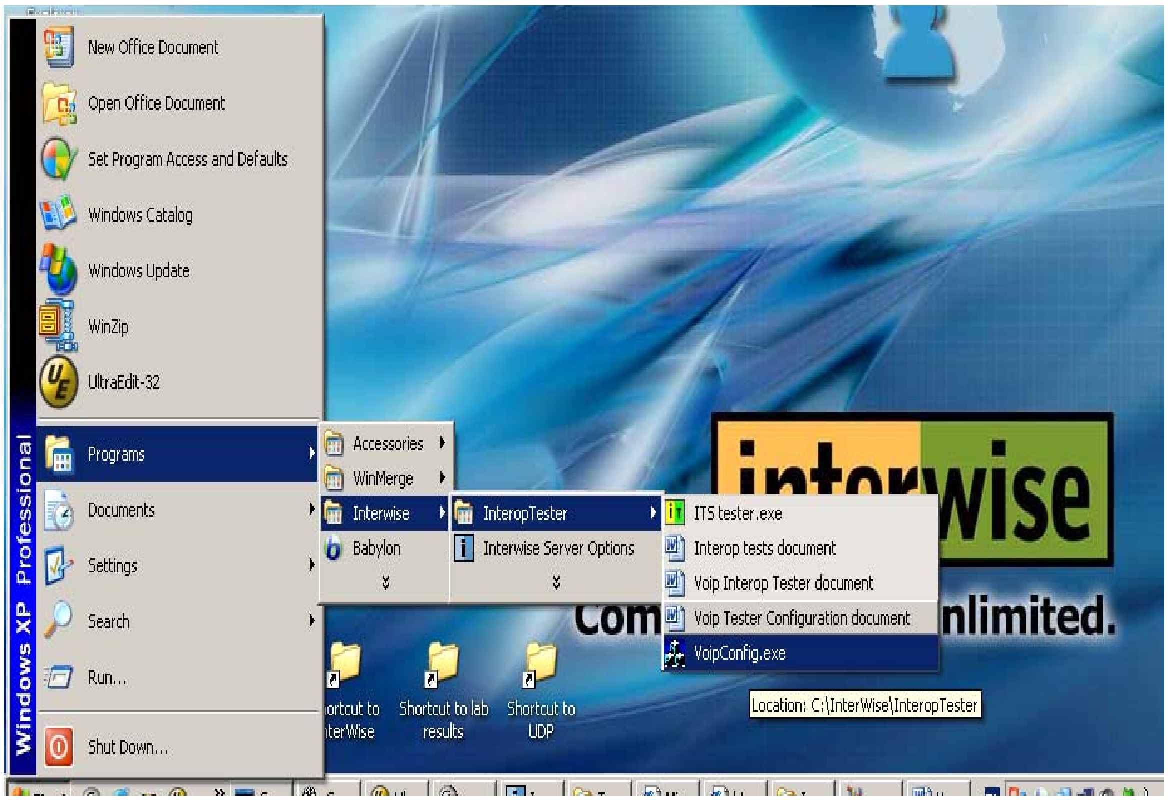
5. Configure the Interwise Conferencing Service

This section describes the steps for configuring the Interwise Conferencing Service. This section assumes that the Interwise Conferencing Service application software is already installed on the conference server (Interwise ITS installation).

Step	Description
1.	Click on **Start->Programs->Interwise->InteropTester->VoipConfig.exe** to start the configuration process. 

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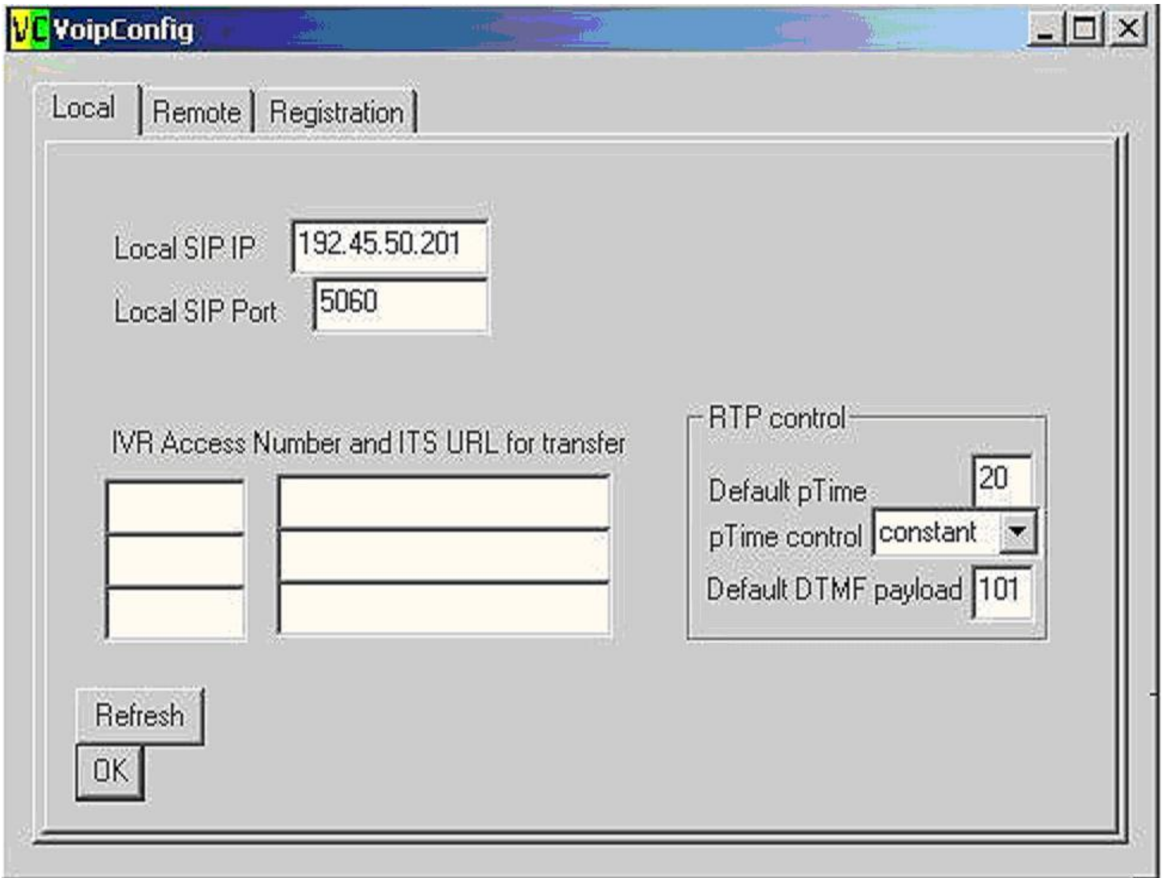
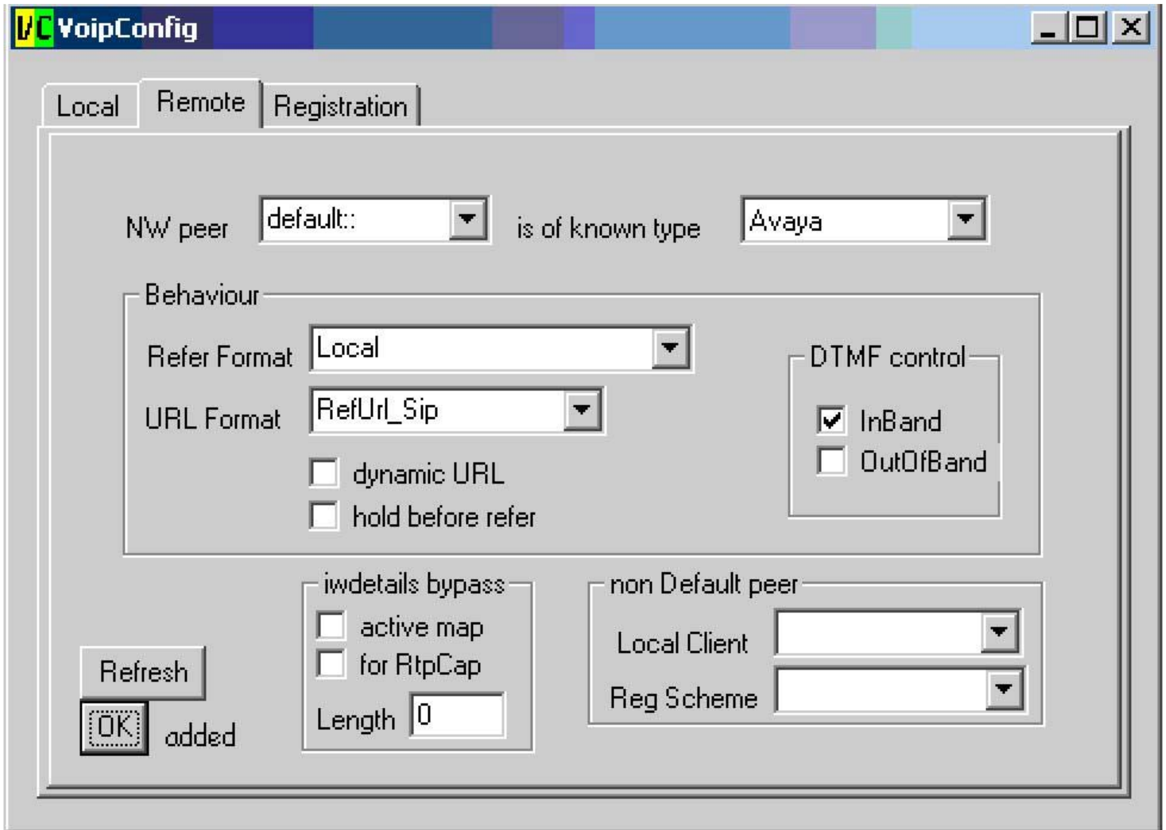
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>
On VoipConfig screen, click on “Local” and enter the following fields:

Local SIP IP – Set to IP Address of the conferencing server.
Local SIP Port – set to 5060 (default).
Hit the OK button to save the changes

Note: No value is required for the IVR Access Number and ITS URL for transfer.



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

Step	Description
3.	On **VoipConfig** screen, click on **"Remote"** and enter the following fields: • **NW peer** – Set to **default::** • Choose **"Avaya"** from the list in the next field. • **DTMF control** – Set to **InBand.** • Hit the **OK** button to save the changes 

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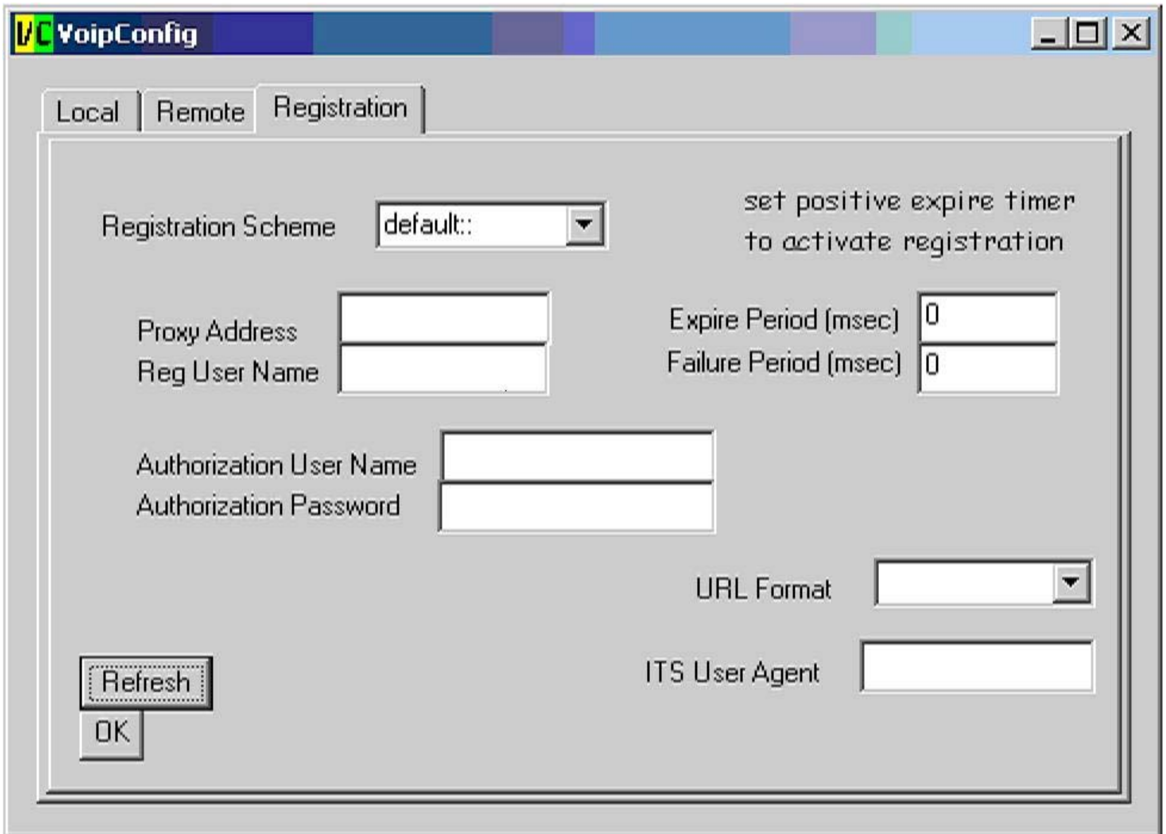
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>On VoipConfig screen, click on “Registration” and enter the following fields:

Registration Scheme – Set to default::
Expiration Period – Set to 0 as Interwise Conferencing Service is a trusted host.
Failure Period - Set to 0 as Interwise Conferencing Service is a trusted host.
Hit the OK button to save the changes



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

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6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily on multi-party conference establishment on the Interwise Conferencing Service. The tests verified its features such as help, mute, raising hand, stepping out, verifying number of people in the conference using interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, digital, analog and PSTN phones.

6.1. General Test Approach

The general test approach was to place calls from any phone to establish a conference and exercise the features supported by the Interwise Conferencing Service. The main objectives were to verify that:

- The Interwise Conferencing Service acts as a trusted host to the Avaya SES.
- The Interwise Conferencing Service successfully establishes a conference call with Avaya SIP, H.323, and digital phones connected to Avaya SES or Avaya Communication Manager.
- The Interwise Conferencing Service successfully establishes a conference with PSTN phones through Avaya Communication Manager.
- The Interwise Voice Response System successfully executes a blind transfer.
- The Interwise Conferencing Service successfully handles multiple parties in the conference.
- The Interwise Conferencing Service successfully shuffles for VoIP calls.
- The Interwise Conferencing Service successfully handles DTMF for establishment of a conference.
- The Interwise Conferencing Service successfully handles holds and transfer of calls.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two Avaya SIP Phones and two Avaya H323 phones, two Avaya digital phones and an analog PSTN was formed as follows. Each phone called into the Interwise Voice Response (IVR) system which played an announcement to prompt for the conference ID. Once the conference ID was correctly entered, each calling party was welcomed and they joined the conference.

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6.2. Test Results

The test objectives of Section 6.1 were verified. For serviceability testing, the Interwise Conferencing Service operated properly after recovering from failures such as cable disconnects, and resets of the Interwise Conferencing Service, the Avaya SES server, and Avaya Communication Manager. For performance testing, each of the participants was successfully able to interact with the IVR and join the conference call which was successfully maintained for approximately two hours.

The following observations were made during testing:

- Interwise Conferencing Service does not support de-registration but it re-registers automatically with SES once the service is re-started.
- Interwise Conferencing Service operates with UDP as the SIP transport protocol.
- Interwise Conferencing Service only supports the G.711MU codec.
- Interwise Conferencing Service does not support QOS parameters.
- If Music on Hold is configured in Avaya Communication Manager, Interwise Conferencing Service will replay the music while the conference is put on hold by any caller. A workaround is to enable the Interwise Conferencing Service mute feature before putting the call on hold or transferring the call.

Interwise Inc. expects to resolve the above observations in future releases. Contact Interwise Inc. (www.interwise.com) for further updates.

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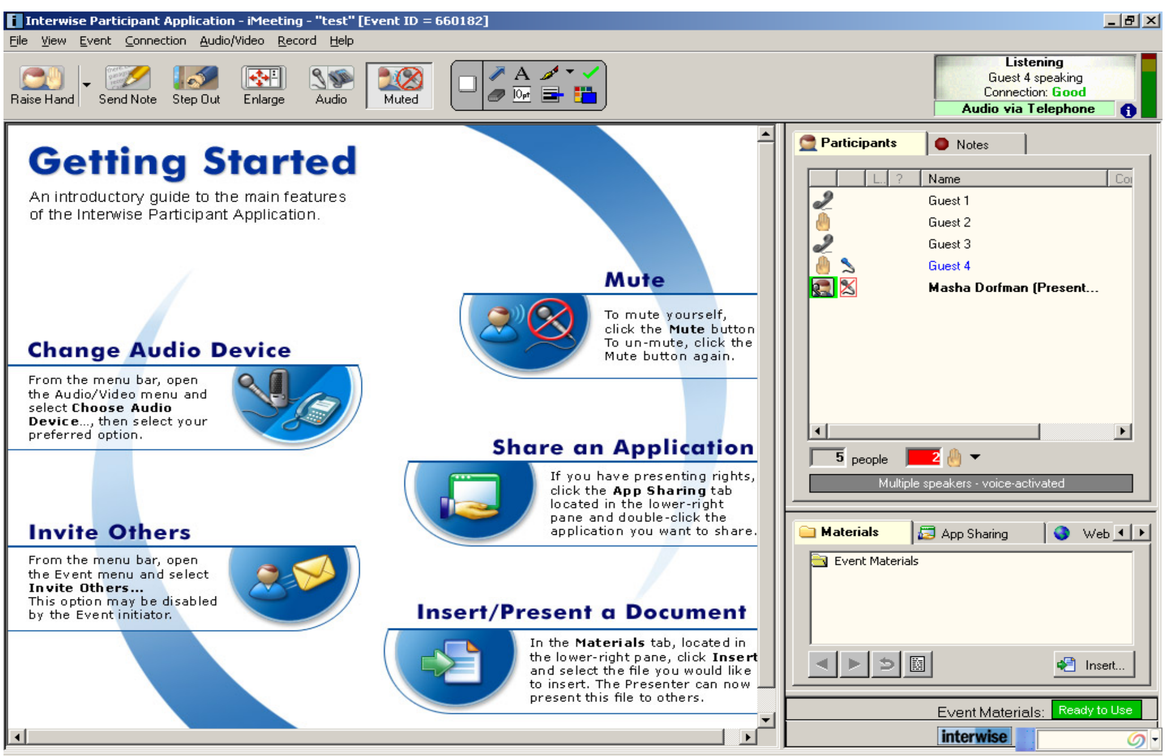
7. Verification Steps

Step	Description
1.	Verify all members for the SIP trunk group provisioned in **Section 3.5** are **in-service/idle** using the following command from SAT session on Avaya Communication Manager: • Issue the command “**status trunk 10**”. • Verify that all members in Trunk Group **10** are **in-service/idle**.

2.	Verify that the Interwise Conferencing Service is properly configured as a trusted host using secure shell to connect to the Avaya SES server.
	`SES> trustedhost -L`
	`Third party trusted hosts.`
	` Trusted Host IP address  |   SES Host IP address     |         Comment`
	`-------------------------+--------------------------+------------------------`
	`192.45.50.201            | 192.45.52.160            | INTERWISE_GW`
	• Place calls into the Interwise Conferencing Service and verify that the conference is successfully established.

3.	Verify that all the parties involved in the conference call appear on the Interwise Conferencing Service. The following screen on the Interwise Conference Service platform verifies all the conference parties connected to a particular event. 

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8. Support

For technical support on Interwise Inc. Connect's ITS, consult the support pages at http://www.interwise.com/support or contact Interwise Inc. technical support at:

- Phone: 1-617-475-2200
- E-mail: support@interwise.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and Interwise Connect's ITS 7.2.78 which offers Interactive Voice Response/Conferencing Service. The Interwise Conferencing Service is a SIP-based VoIP audio conferencing solution intended for use by large enterprises and conference service providers. During compliance testing, the Interwise Conferencing Service successfully interacted with Avaya SES as a trusted host and was able to setup conferences between SIP and non-SIP telephones.

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 R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

Product documentation for Interwise Inc. products may be found at http://www.interwise.com.

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