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Avaya Solution & Interoperability Test Lab

Application Notes for Convergys Intervoice Media Server with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement Services - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Convergys Intervoice Media Server to successfully interoperate with Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify the SIP protocol messages between Convergys Intervoice Media Server and Avaya Aura^{TM} SIP Enablement Services.

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 procedures for configuring Convergys Intervoice Media Server which was compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. The overall objective of this interoperability compliance test is to verify that Convergys Intervoice Media Server can interoperate with Communication Manager and SIP Enablement Services (here on, refers to as SES) in an Avaya IP Telephony environment.

Convergys Intervoice Media Server provides IVR and Messaging functionality via SIP/VOIP telephony interface. Callers interact with the system via DTMF or Speech input, and may be transferred to agents, as needed.

These Application Notes assume that Communication Manager and SES have already been installed and that basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [1] [2] [3].

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The focus of the interoperability compliance testing was primarily on verifying call establishment on Convergys Intervoice Media Server. Convergys Intervoice Media Server operations such as inbound calls, DTMF tone, blind transfer, and Convergys Intervoice Media Server interaction with SES, Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. Serviceability testing introduced failure scenarios to see if Convergys Intervoice Media Server can recover from failures.

1.2. Support

Technical support for the Convergys Media Server can be obtained by contacting Convergys, Inc. via http://realcare.intervoice.com or by calling 800-955-4688.

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G450 Media Gateway, a SES server, and Convergys Intervoice Media Server. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. Avaya S8720 Servers with an Avaya G650 Media Gateway were included in the test to support inter-switch calls. For completeness, Avaya 4600 Series SIP IP Telephones, Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, and Avaya 9600 Series H.323 IP Telephones were included in the compliance test.

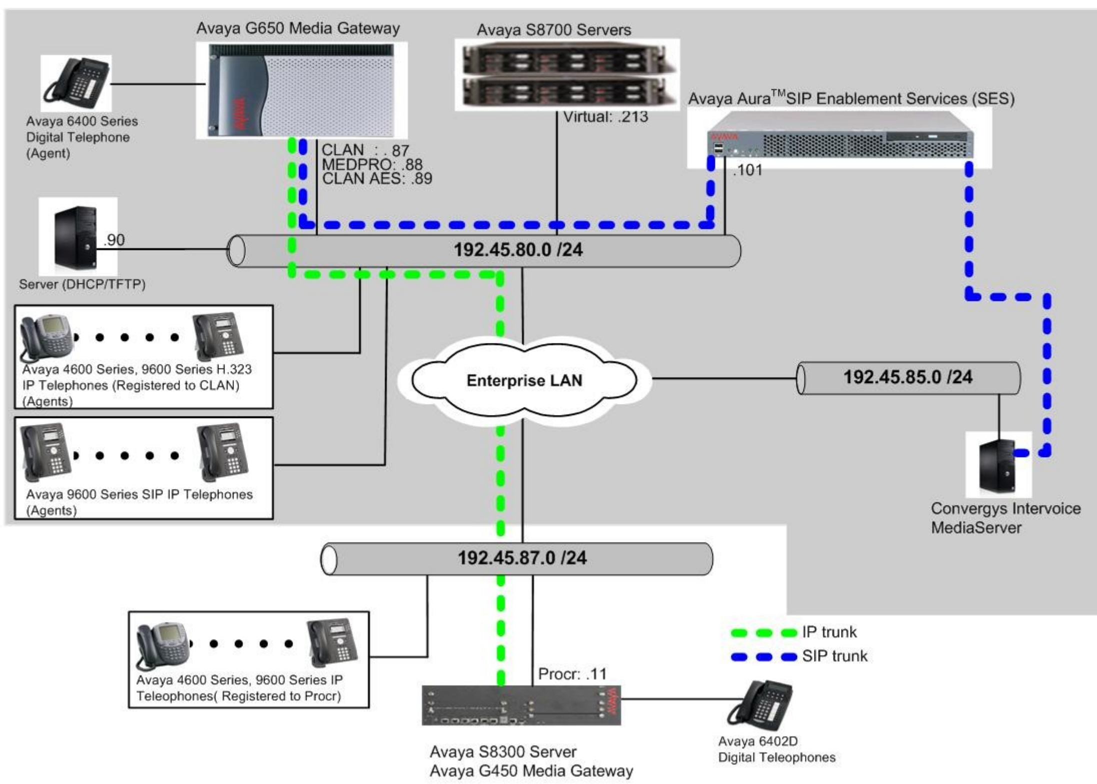


Figure 1. Test configuration of Convergys Intervoice Media Server with Avaya Aura™ SIP Enablement Services

3. Equipment and Software Validated

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

Equipment		Software
Avaya S8720 Servers		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G650 Media Gateway		
	TN2312BP IPSI TN799DP CLAN TN2302AP MEDPRO	HW11 FW030 HW20 FW017 HW01 FW108
Avaya S8300 Server		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G450 Media Gateway		28.17
Avaya Aura™ SIP Enablement Services		Avaya Aura™ SIP Enablement Services 5.2 (R015x.02.0.947.3) with Service Pack 2 (SES-02.0.947.3-SP2a)
Avaya 4600 Series IP Telephone		
	4620SW 4625SW	2.9 2.9
Avaya 9600 Series IP Telephone		
	9630 9650	3.002 3.002
Avaya 9600 Series SIP Telephone		
	9620 9630	2.05 2.05
Avaya 64xx Series Digital Telephones		
	6408D+ 6402D	- -
Analog Telephone		-
Avaya C363T Converged Stackable Switch (Layer 3)		4.5.14
Extreme Summit 48 Switch (Layer 3)		4.1.21
Convergys Intervoice Media Server on Windows 2003 Server w/ SP2		3.6.2.246

4. Configure Avaya Aura™ Communication Manager

This section provides the procedures for configuring a SIP trunk group and a SIP signaling group on Communication Manager. All the configuration changes in Communication Manager are performed through the System Access Terminal (SAT) interface. The highlights in the following screens indicate the values used during the compliance test.

4.1. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SES. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – Enter the appropriate name for the Authoritative Domain. During the compliance test, the authoritative domain is set to **moris.mot.com**. This should match the SIP Domain value on SES, in **Section 5.1**.
- Intra-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SES in the same IP network region. The default value for this field is **yes**.
- Codec Set – Specify the codec set number as provisioned in Communication Manager.
- Inter-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SES in different IP network regions. The default value for this field is **yes**.

Default values may be used for all other fields.

```
change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: moris.mot.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

4.2. Configure a Node Name for Avaya Aura™ SIP Enablement Services

The node name configured in this section will be utilized for the SIP trunk configuration between Communication Manager and SES. Enter the **change node-names ip** command. In the compliance-tested configuration, the procr and SES IP address were utilized for Near-end Node and Far-end Node in the signaling group form configured in Section 4.3, respectively.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
IA770                   192.45.87.12
SES                     192.45.80.101
default                 0.0.0.0
procr                   192.45.87.11
```

4.3. Configure SIP Signaling Group

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and SES. For a trusted host, a second SIP signaling group is needed, on top of the SIP signaling group between Communication Manager and SES. Enter the **add signaling-group <t>** command, where **t** is an available signaling group and configure the following:

- Transport Method – Set to **tls.**
- Near-end Node Name - Set to **procr** as displayed in Section 4.2.
- Far-end Node Name - Set to the SES name, **SES**, configured in Section 4.2.
- Far-end Domain – - Set to **moris.mot.com**. This should match the SIP Domain value in **Section 4.1**.

Default values may be used for all other fields.

```
add signaling-group 3                                           Page   1 of   1

 Group Number: 3             Group Type: sip
                        Transport Method: tls




   Near-end Node Name: procr                  Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
       Far-end Domain: moris.mot.com

                                             Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3               Alternate Route Timer(sec): 6
```

4.4. Configure SIP Trunk Group

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and SES. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- Group Type – Set the Group Type field to **sip**.
- Group Name – Enter a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code.
- Signaling Group – Set to the Signaling Group Number configured in **Section 4.3**.
- Number of Members – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Default values may be used for all other fields.

Note: *Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted*

```
add trunk-group 3                                                 Page   1 of  21
                                TRUNK GROUP

Group Number: 3                   Group Type: sip          CDR Reports: y
  Group Name: ToSES                          COR: 1        TN: 1         TAC: 1003
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                         Signaling Group: 3
                                                       Number of Members: 10
```

4.5. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of Convergys Intervoice Media Server. Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **9600SIP**.
- Name – Enter a descriptive name

Repeat this step as necessary to configure additional SIP endpoint extensions.

```
add station 27004                                            Page   1 of   5
                                     STATION

Extension: 27004                         Lock Messages? n             BCC: 0
     Type: 9600SIP                       Security Code:                TN: 1
     Port: IP                          Coverage Path 1:               COR: 1
     Name: SIP 27004                   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: 27004
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

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

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Set the extension of the OPS station as configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that Convergys Intervoice Media Server will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- Trunk Selection – Set to the trunk group number configured in Section 4.4.
- Config Set – Set to **1**

Repeat this step as necessary to configure additional off-pbx-telephone station-mapping.

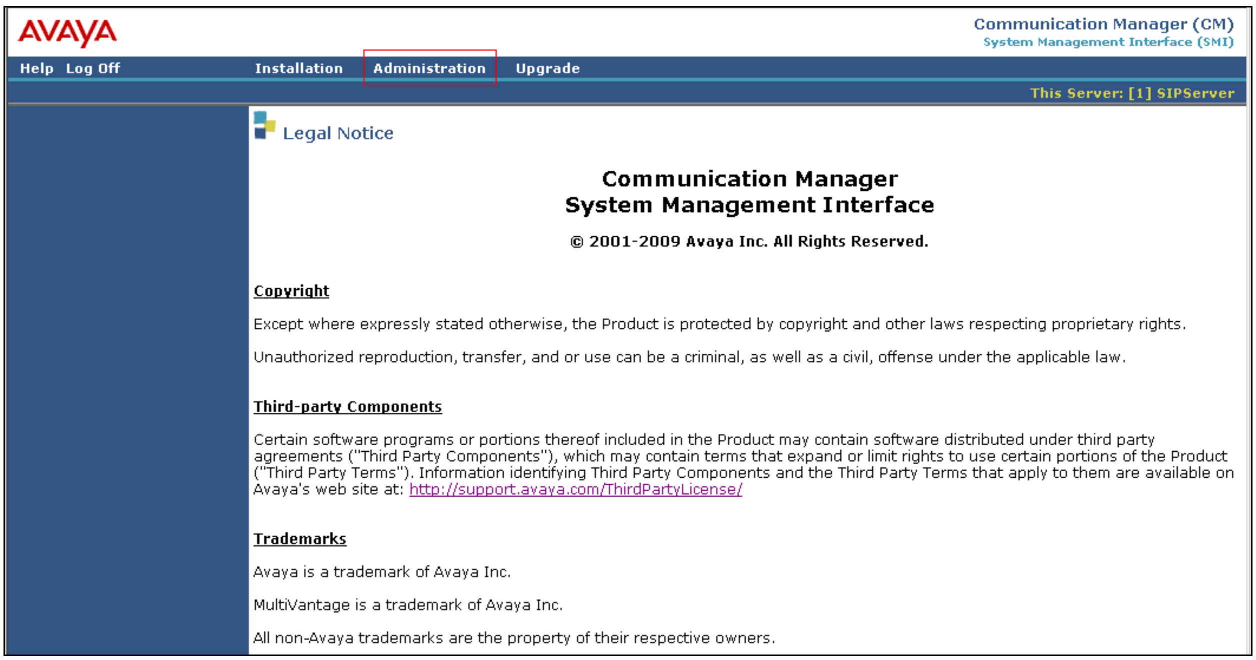
```
add off-pbx-telephone station-mapping                          Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station         Application Dial   CC  Phone Number       Trunk       Config
 Extension                   Prefix                        Selection   Set
 27004              OPS           -     27004                3           1
```

5. Configure Avaya Aura™ SIP Enablement Services

This section describes the steps for creating SIP trunks between SES and Communication Manager, and between SES and Convergys Intervoice Media Server. SIP user accounts are configured in SES and associated with a Communication Manager OPS station extension. Convergys Intervoice Media Server will register with SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

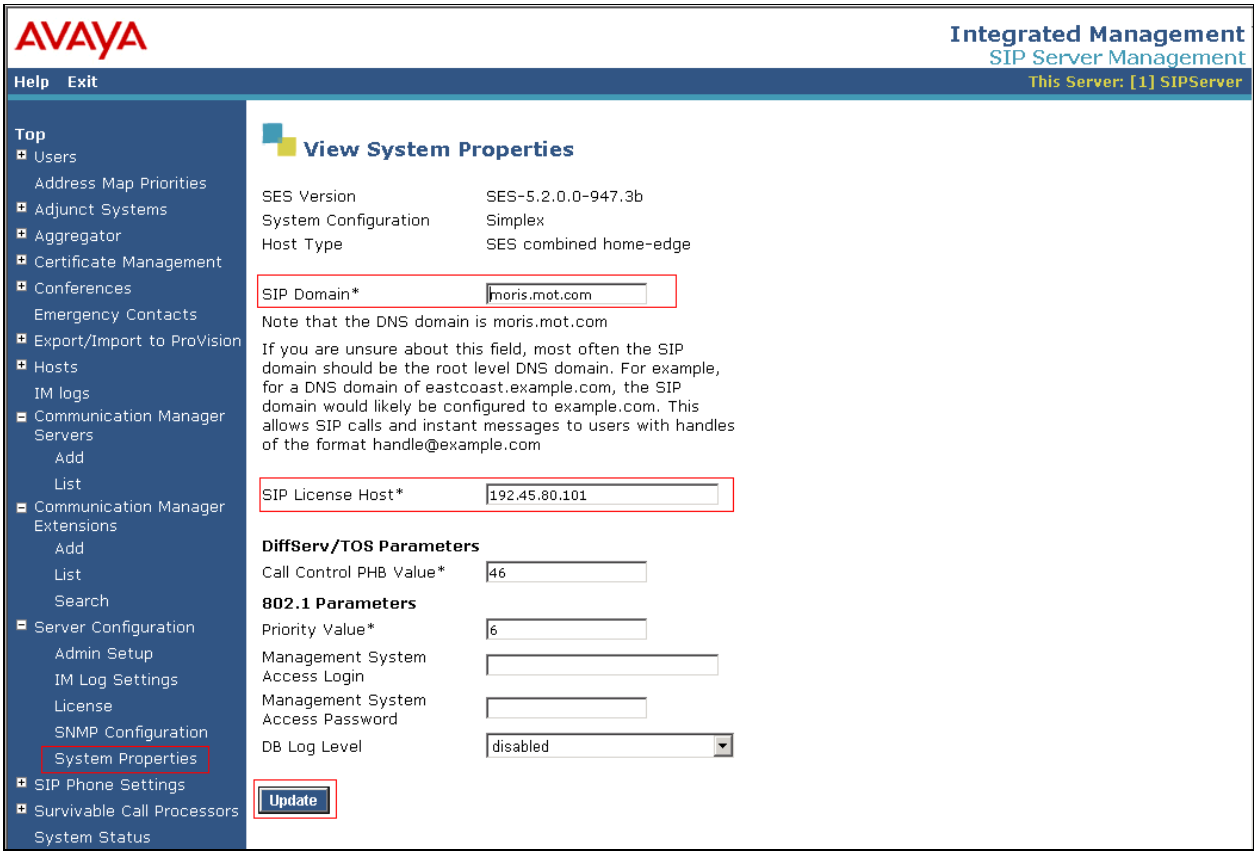
5.1. Configure System Properties

Launch a web browser, enter **https://<IP address of SES server>/admin** in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login. Navigate to **Administration → SIP Enablement Services**.



In the Integrated Management SIP Server Management page, select the **Server Configuration → System Properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Communication Manager in Section 4.3.

Click on the **Update** button, after the completion.

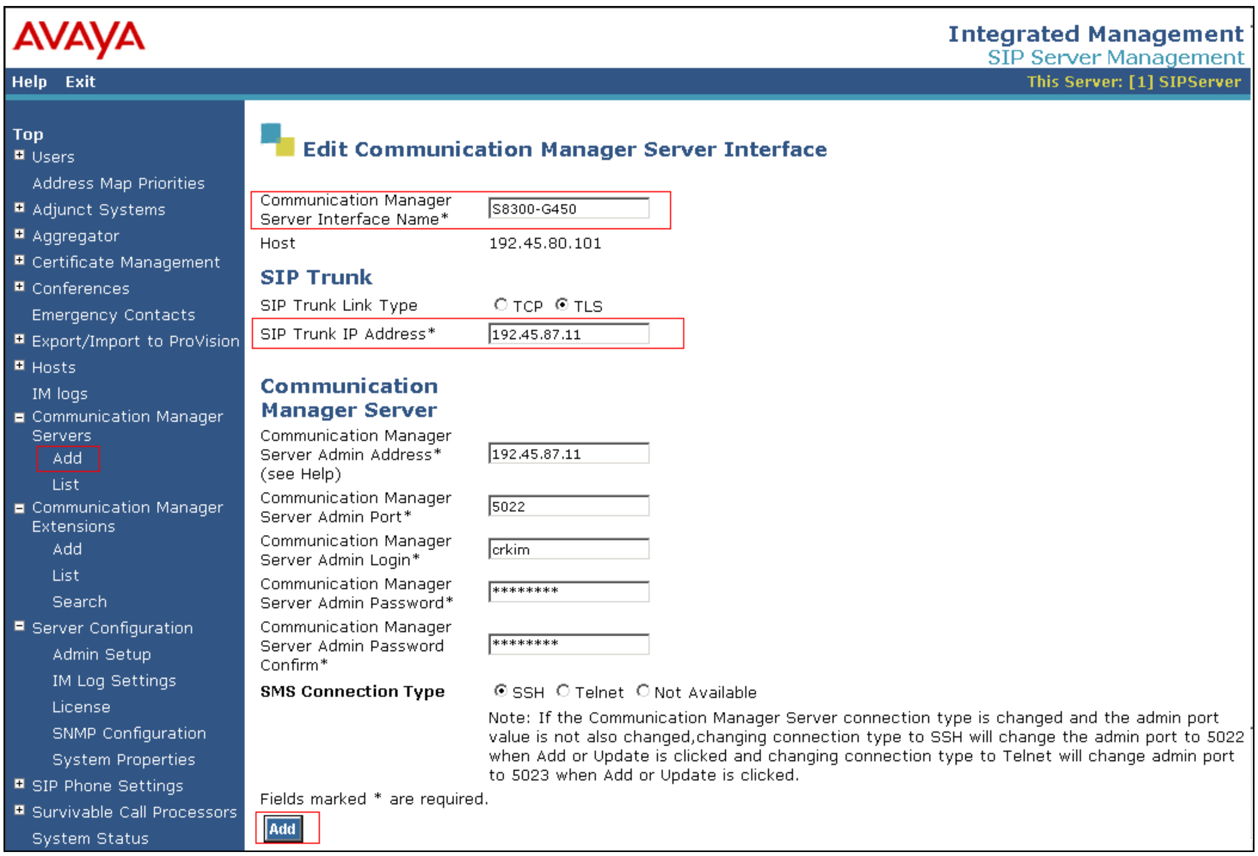


5.2. Configure Communication Manager

This section provides steps to add SIP-enabled media servers to the SIP domain. In the Integrated Management SIP Server Management page, select the **Communication Manager Servers → Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

- Communication Manager Server Interface Name – Enter a descriptive name for the communication manager server interface.
- SIP Trunk IP Address – Enter the IP address for the media server's procr (or CLAN) IP interface that terminates the SIP link from SES.

Click **Add** when finished.

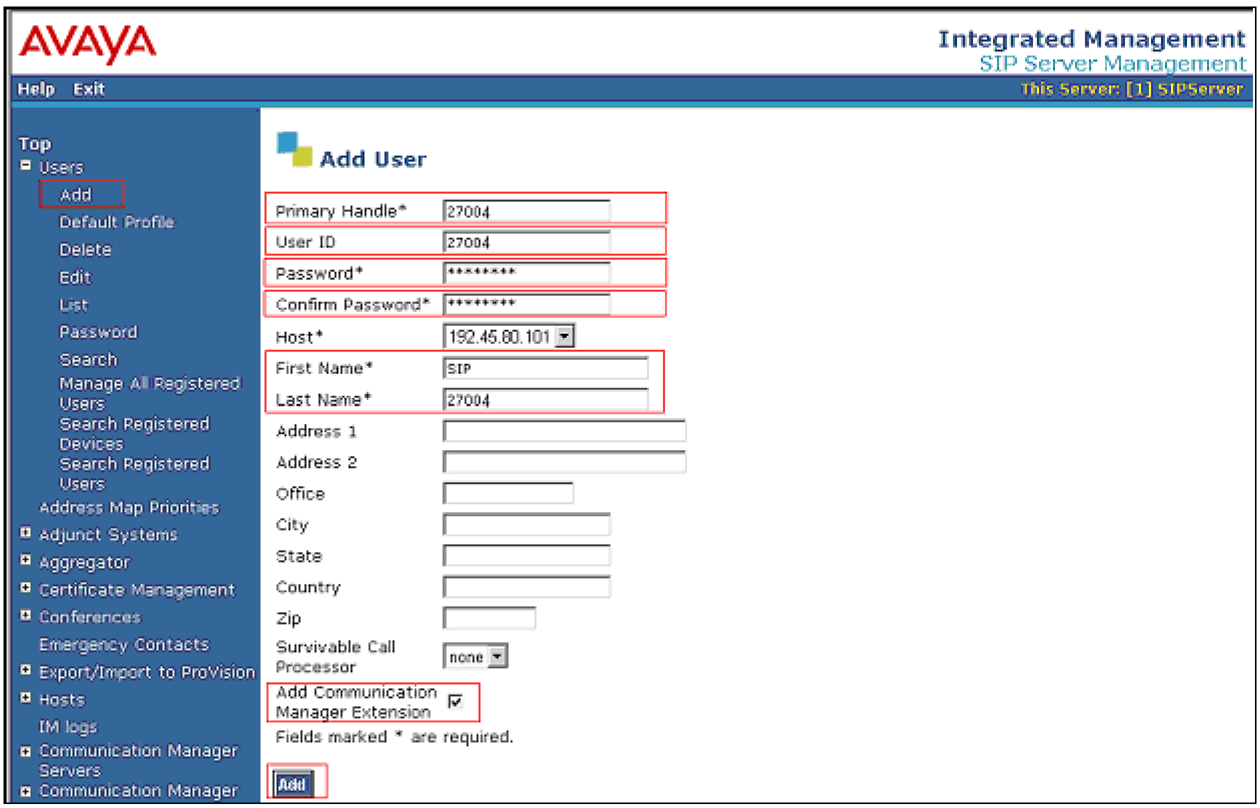


5.3. Configure Users

This section provides steps to add users to be administered in the SES database. In the Integrated Management SIP Server Management page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test

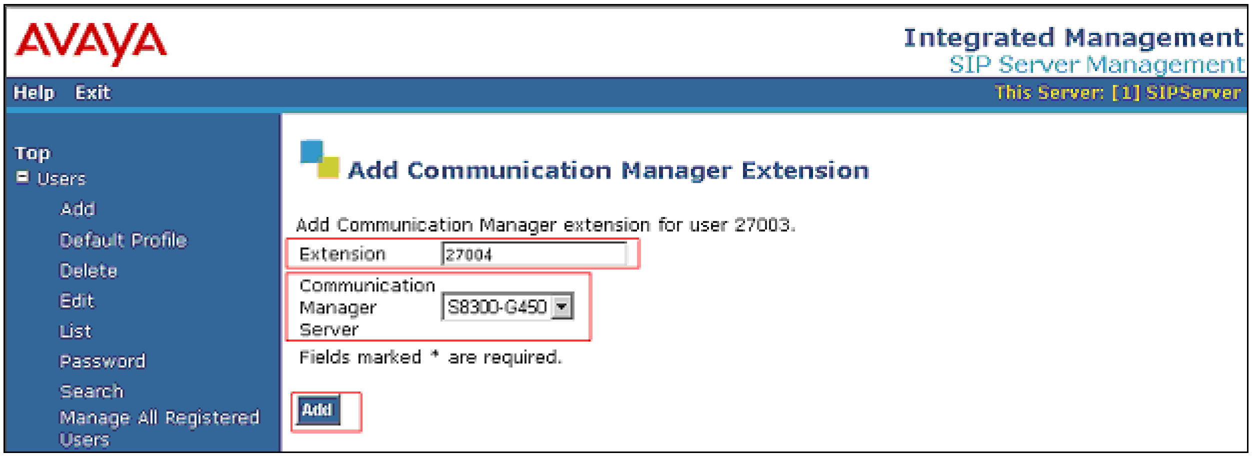
- Primary Handle – Enter the phone number of Convergys Intervoice Media Server. This number was configured in Section 4.5.
- Password / Confirm Password – Enter a password; both field entries must match exactly.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Check the **Add Communication Manager Extension** check box if you want to associate a new extension number with this user in the database now. If so, the **Add MS Extension** screen will be displayed next, after this user profile has been added. If not, in the future you may choose to associate extensions with the user.

Click **Add** when finished.



From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's Communication Manager Server from the drop-down list.

Click on the **Add** button.

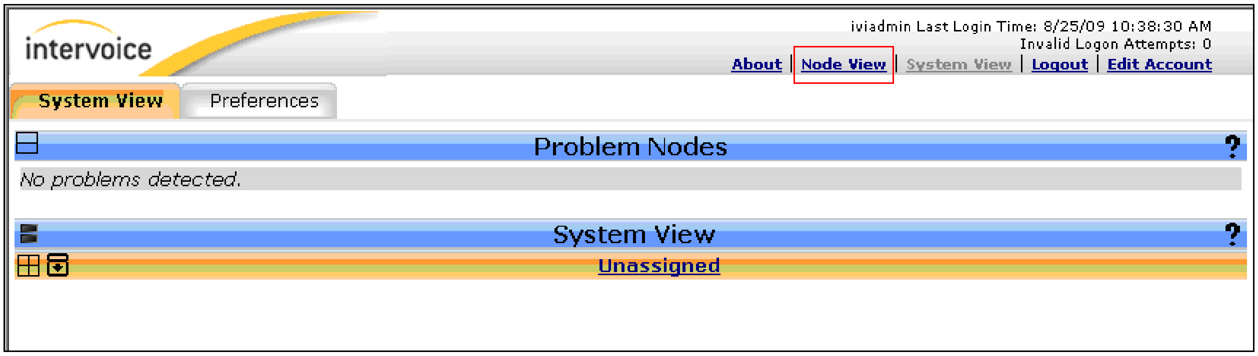


6. Configure Convergys Intervoice Media Server

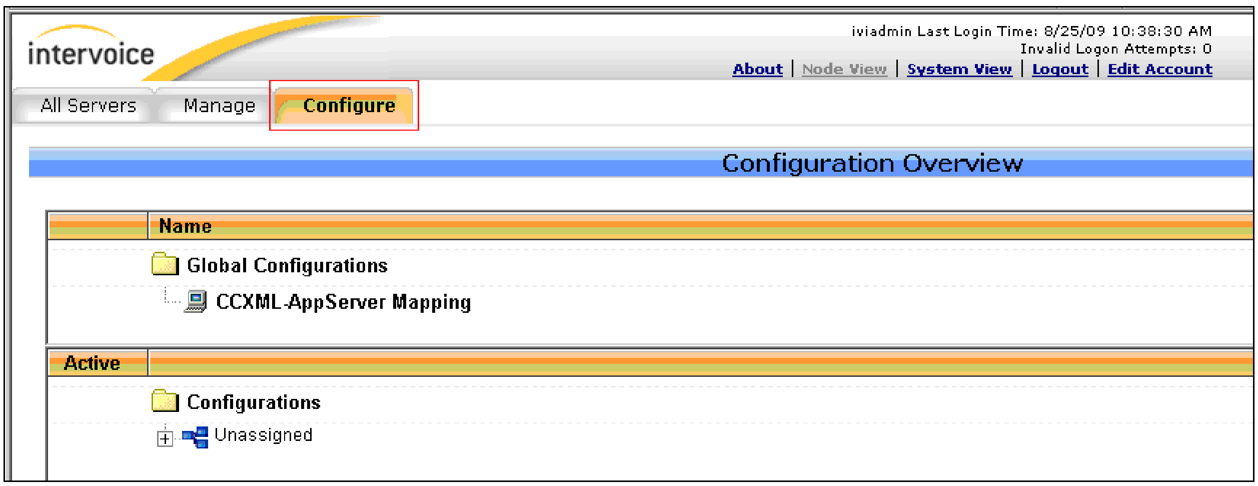
This section provides steps to configure Convergys Intervoice Media Server. Convergys installs, configures, and customizes the Intervoice Media Server application for end customers. This section describes the initial Intervoice Media Server configuration.

Launch a web browser, enter http://<IP address of the Convergys Intervoice Media Server > in the URL, and log in with the appropriate credentials to access the **System View** page.

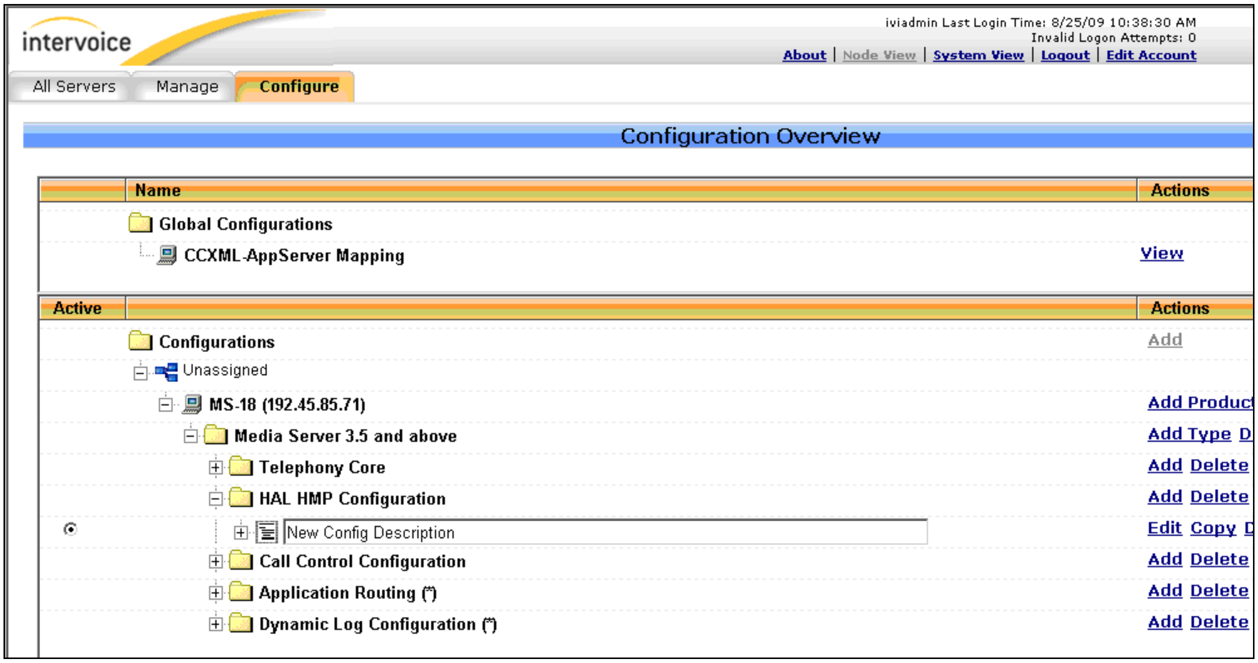
Select the **Node View** link at the top right.



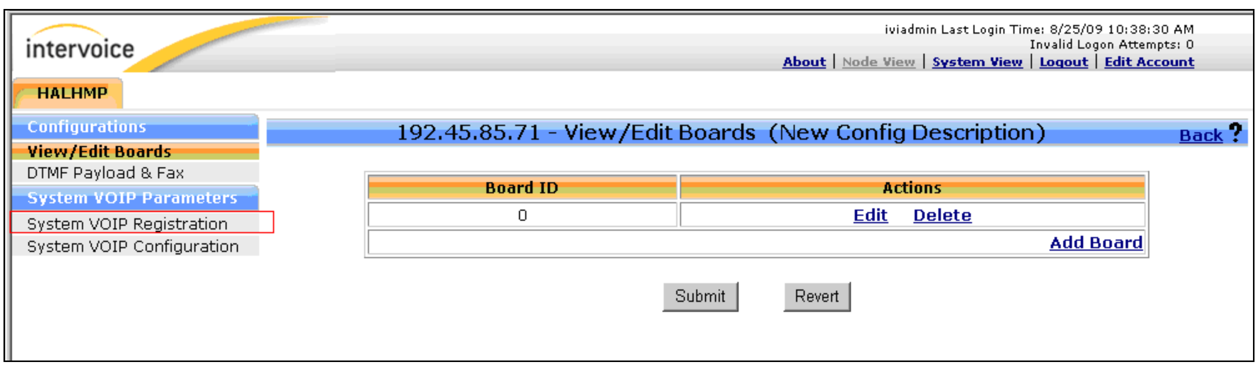
Click the **Configure** tab to start configuring the Convergys Intervoice Media Server.



Expand and navigate to **Unassigned → MS-18 (192.45.85.71) → Media Server 3.5 and above → HAL HMP Configuration → New Config Description.** Click the **Edit** button next to the New Config Description field.

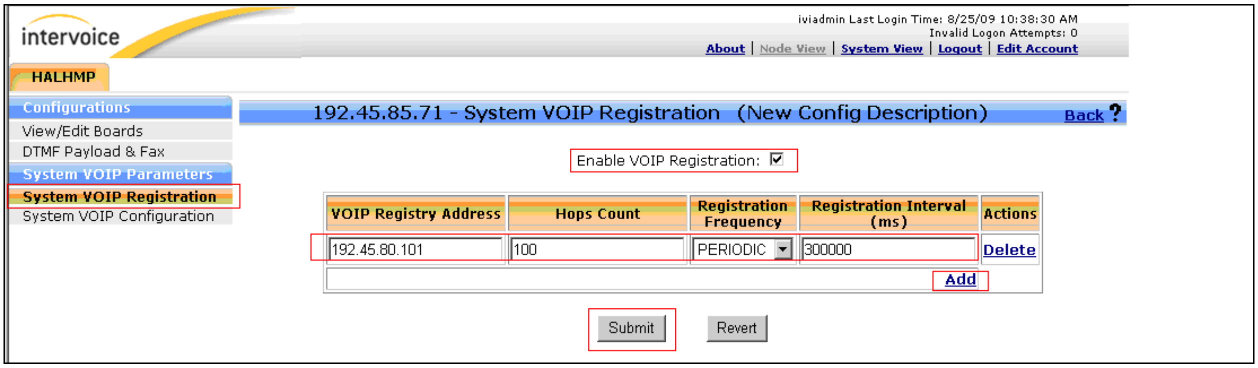


Select **System VOIP Registration** under the System VOIP Parameters menu.

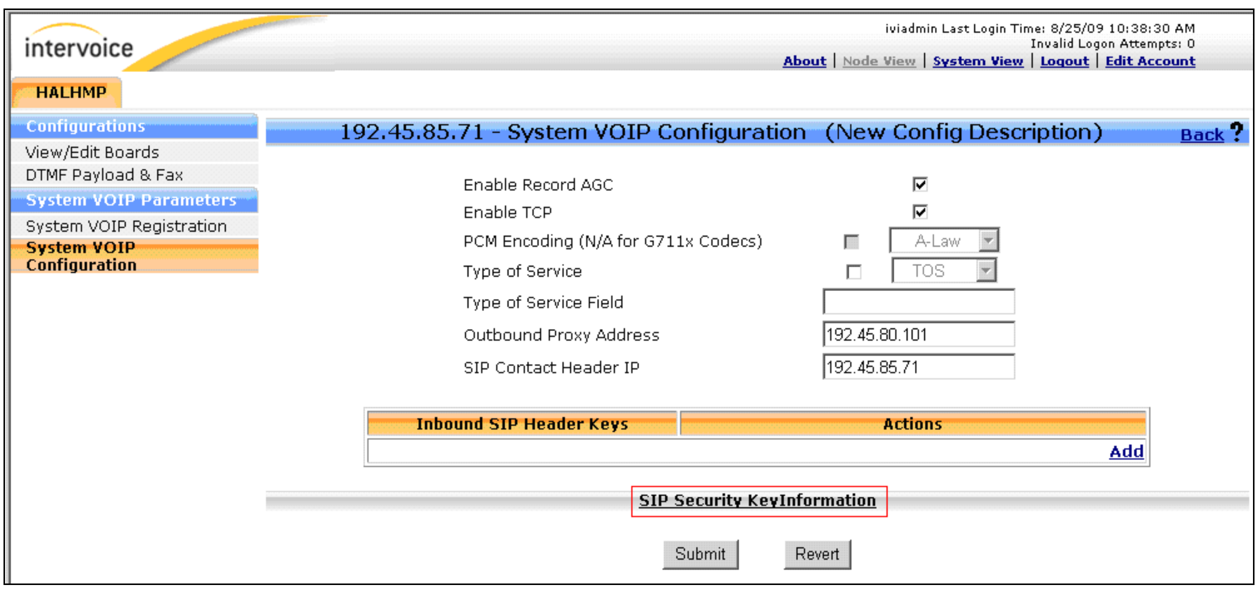


On the **System VOIP Registration** page, select the **Enable VOIP registration** check box. Select **Add** and provide the VOIP Registry Address by entering the SES IP address.

Click the **Submit** button to submit changes.



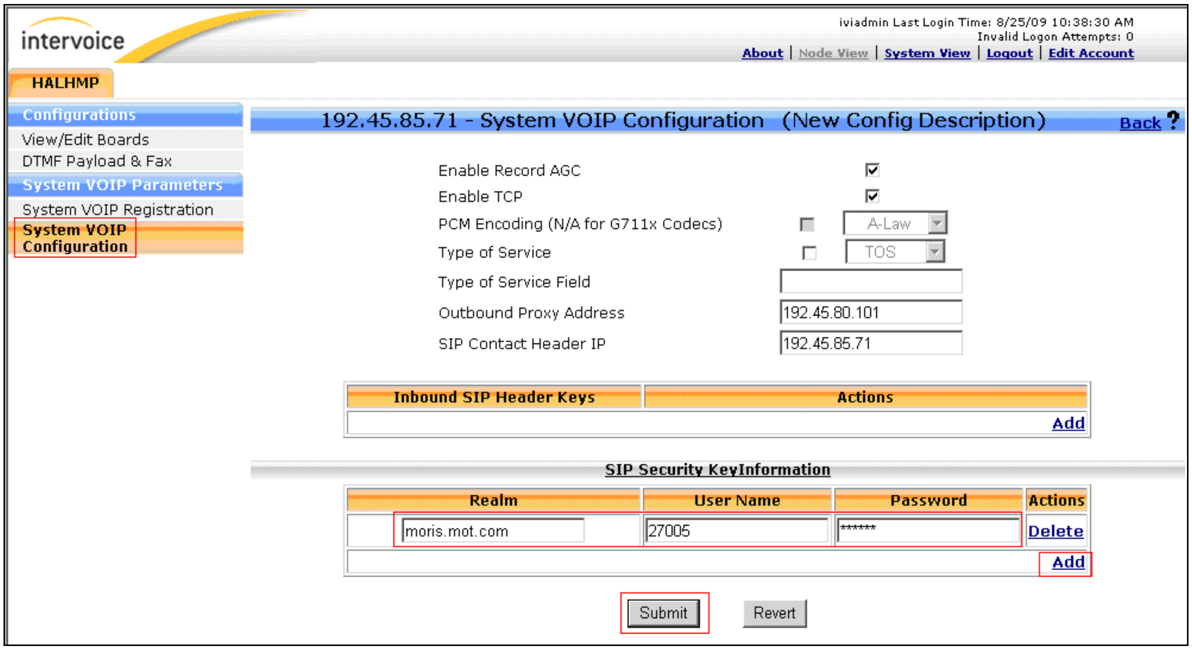
Select **System VOIP Configuration** under the System VOIP Parameters menu. Select **SIP security Information** link.



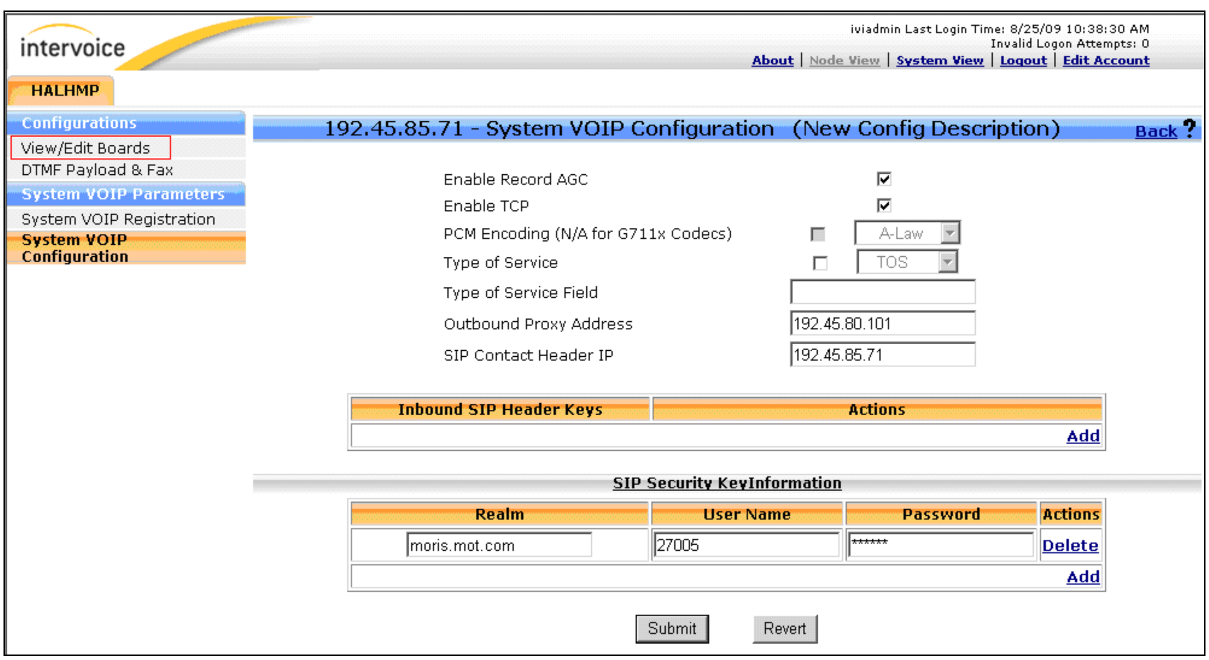
Select **Add** and enter the following information:

- SIP Domain
- User extension
- User Password

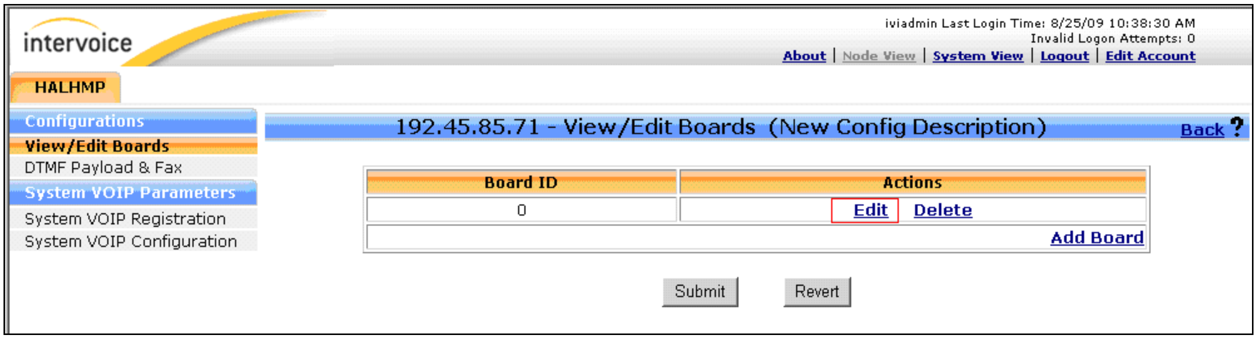
Click the **Submit** button to submit changes.



Once, the System VOIP Configuration is completed, select **View/Edit Boards** under the Configurations menu.



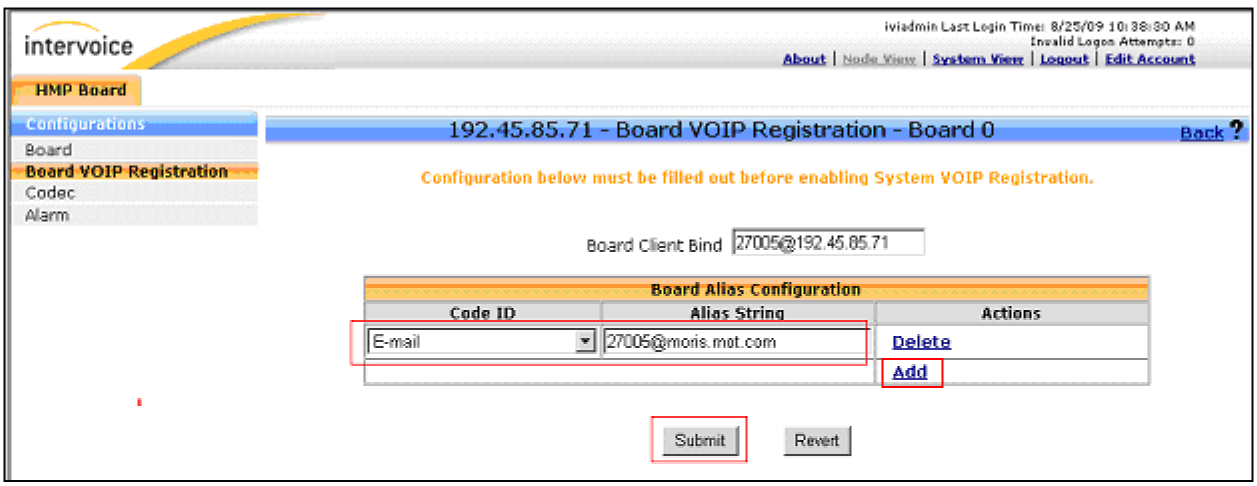
Select **Edit** under the Action menu on the right pane of the window.



Select **Add** and enter the following information:

- Code ID – Select E-mail using the drop-down menu.
- Alias String – Type in the fully qualified domain name of the Convergys Intervoice Media Server.

Click the **Submit** button to submit changes.



7. General Test Approach and Test Results

The general test approach was to verify the SIP protocol messages between SES and Convergys Intervoice Media Server, when calls were manually placed to Convergys Intervoice Media Server. All test cases were successful, and verified utilizing SIP traces.

8. Verification Steps

The following steps may be used to verify the configuration:

- End-to-end verification: When a call was placed to Convergys Intervoice Media Server, the welcome message was heard from the originated extension. The origination telephone types included Avaya H.323 telephones (4625, 9630, and 9650), Avaya SIP telephones (9620 and 9630), and Avaya DCP telephone. Verified correct SIP traces on each call using Wireshark Network Analyzer.
- DTMF tone was verified during each inbound call to Convergys Intervoice Media Server.
- Blind transfer was verified from Convergys Intervoice Media Server to an agent.

9. Conclusion

Convergys Intervoice Media Server was compliance tested with Communication Manager (Version 5.2) and SES (Version 5.2). The correct SIP protocol messages were verified between SES and Convergys Intervoice Media Server when a call was placed to Convergys Intervoice Media Server.

10. References

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

The following Avaya product documentation can be found at http://support.avaya.com .

[1] *Administering Avaya Aura™ Communication Manager* Release 5.2, Issue 5, May 2009, Document Number 03-300509.
[2] *Avaya Aura™ Communication Manager Screen Reference,* Issue 1.0, May 2009, Document Number 03-602878.
[3] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server,* Issue 2.0, May 2008, Document Number 03-602508.

The following document was provided by Convergys.

[4] *Media Server 3.6 (PSTN) Installation Guide,* Document Number 60001489
[5] *Media Server 3.6 (VoIP) Installation Guide,* Document Number 60001490
[6] *Media Server VoiceXML Browser Technical Reference,* Document Number 60001390

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.