



Avaya Solution & Interoperability Test Lab

Application Notes for Convergys Interservice Media Server with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services - Issue 1.0

Abstract

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

The overall objective of the interoperability compliance testing is to verify the SIP protocol messages between Convergys Interservice Media Server and Avaya Aura™ 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 Interoice 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 Interoice Media Server can interoperate with Communication Manager and SIP Enablement Services (here on, refers to as SES) in an Avaya IP Telephony environment.

Convergys Interoice 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 Interoice Media Server. Convergys Interoice Media Server operations such as inbound calls, DTMF tone, blind transfer, and Convergys Interoice 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 Interoice 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.interoice.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 Interservice 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.

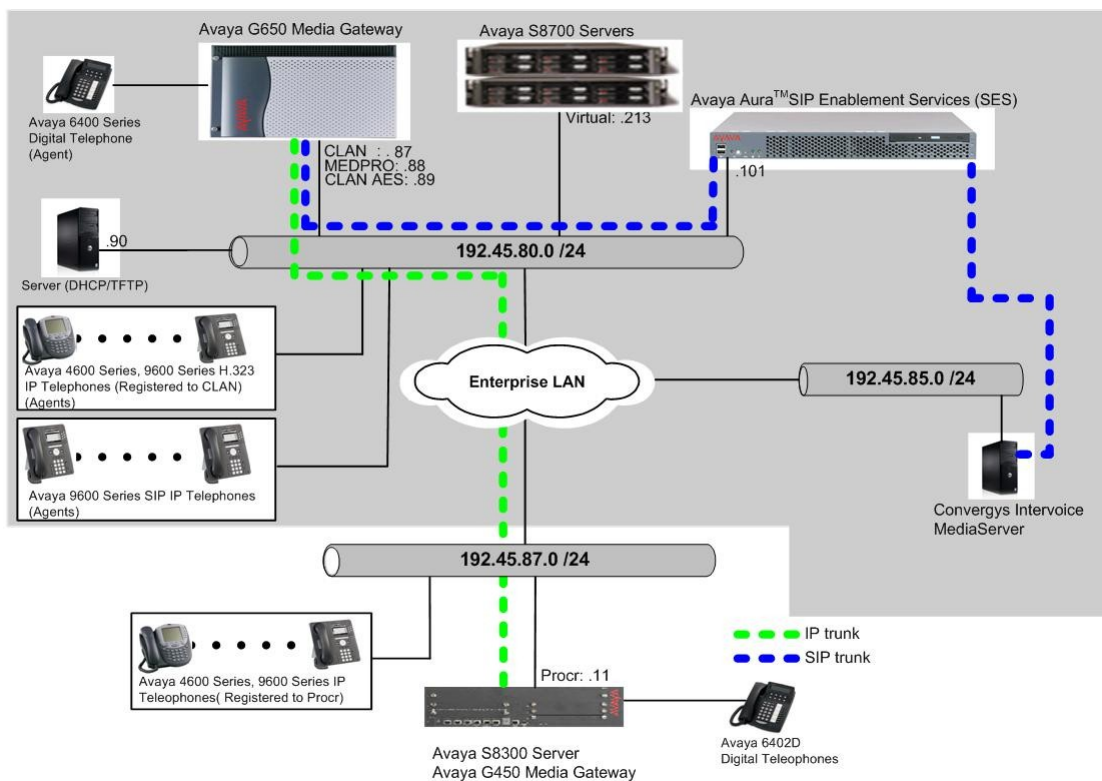


Figure 1. Test configuration of Convergys Interservice Media Server with Avaya Aura™ SIP Enablement Services

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 Intervice 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 Intervice 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
Loss Group: 2                                         Time of Day Lock Table:
Data Module? n                                       Personalized Ringing Pattern: 1
Speakerphone: 2-way                                  Message Lamp Ext: 27004
Display Language: english                            Mute Button Enabled? y
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 Intervice 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.

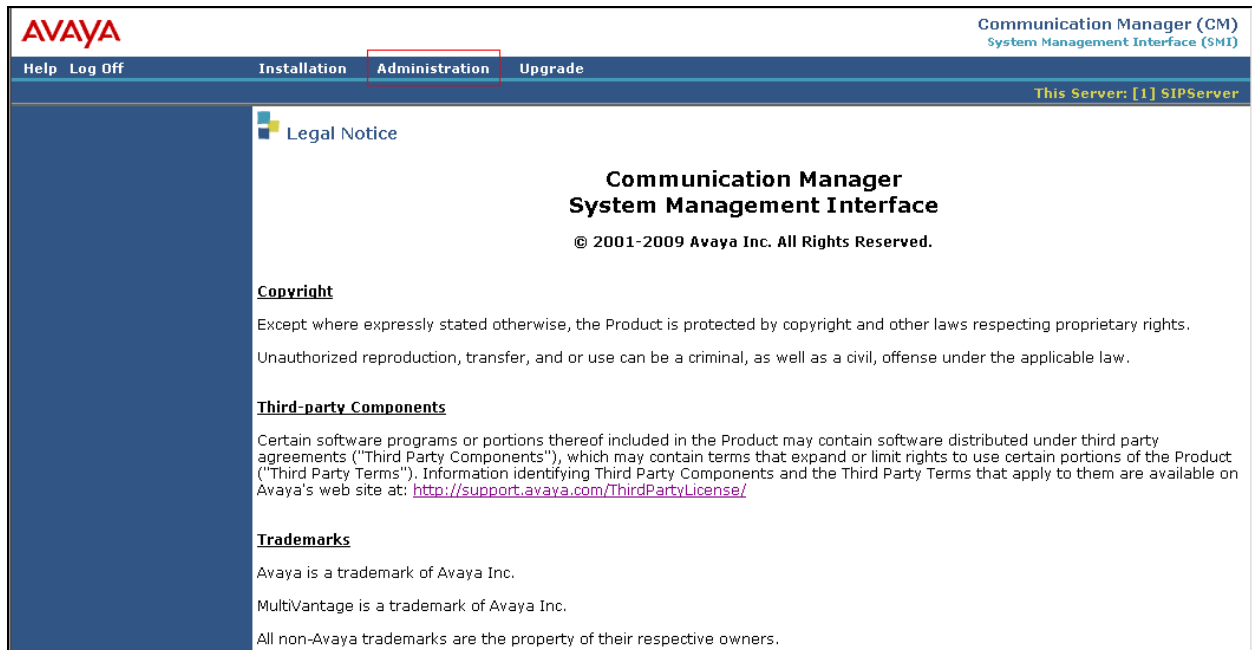
```
add off-pbx-telephone station-mapping               Page 1 of 2
                                                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station      Application Dial  CC  Phone Number      Trunk      Config
Extension    Prefix          Selection      Set
27004       OPS            -              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 Convergy's InterVoice Media Server. SIP user accounts are configured in SES and associated with a Communication Manager OPS station extension. Convergy's 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**.



The screenshot shows the Avaya Communication Manager (CM) System Management Interface (SMI) Administration page. The top navigation bar includes 'Help', 'Log Off', 'Installation', 'Administration' (highlighted with a red box), and 'Upgrade'. The page title is 'Communication Manager (CM) System Management Interface (SMI)' and it indicates 'This Server: [1] SIPServer'. The main content area displays the 'Legal Notice' section, which includes the title 'Communication Manager System Management Interface', the copyright notice '© 2001-2009 Avaya Inc. All Rights Reserved.', and sections for 'Copyright', 'Third-party Components', and 'Trademarks'.

AVAYA Communication Manager (CM)
System Management Interface (SMI)

Help Log Off Installation **Administration** Upgrade This Server: [1] SIPServer

Legal Notice

**Communication Manager
System Management Interface**

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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.

AVAYA Integrated Management SIP Server Management
This Server: [1] SIPServer

Help Exit

Top

- Users
 - Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
 - Emergency Contacts
- Export/Import to ProVision
- Hosts
 - IM logs
- Communication Manager Servers
 - Add
 - List
- Communication Manager Extensions
 - Add
 - List
 - Search
- Server Configuration
 - Admin Setup
 - IM Log Settings
 - License
 - SNMP Configuration
 - System Properties**
- SIP Phone Settings
- Survivable Call Processors
- System Status

View System Properties

SES Version SES-5.2.0.0-947.3b
System Configuration Simplex
Host Type SES combined home-edge

SIP Domain* moris.mot.com

Note that the DNS domain is moris.mot.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host* 192.45.80.101

DiffServ/TOS Parameters

Call Control PHB Value* 46

802.1 Parameters

Priority Value* 6

Management System
Access Login
Management System
Access Password
DB Log Level disabled

Update

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.

AVAYA Integrated Management SIP Server Management
This Server: [1] SIPServer

Help Exit

Top

- Users
- Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
- IM logs
- Communication Manager Servers
 - Add**
 - List
- Communication Manager Extensions
 - Add
 - List
 - Search
- Server Configuration
 - Admin Setup
 - IM Log Settings
 - License
 - SNMP Configuration
 - System Properties
- SIP Phone Settings
- Survivable Call Processors
- System Status

Edit Communication Manager Server Interface

Communication Manager Server Interface Name* S8300-G450

Host 192.45.80.101

SIP Trunk

SIP Trunk Link Type TCP TLS

SIP Trunk IP Address* 192.45.87.11

Communication Manager Server

Communication Manager Server Admin Address* 192.45.87.11 (see Help)

Communication Manager Server Admin Port* 5022

Communication Manager Server Admin Login* crkim

Communication Manager Server Admin Password* *****

Communication Manager Server Admin Password Confirm* *****

SMS Connection Type SSH Telnet Not Available

Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.

Fields marked * are required.

Add

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 Intervice 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.

The screenshot shows the 'Add User' form in the Avaya Integrated Management SIP Server Management interface. The form is titled 'Add User' and contains the following fields and options:

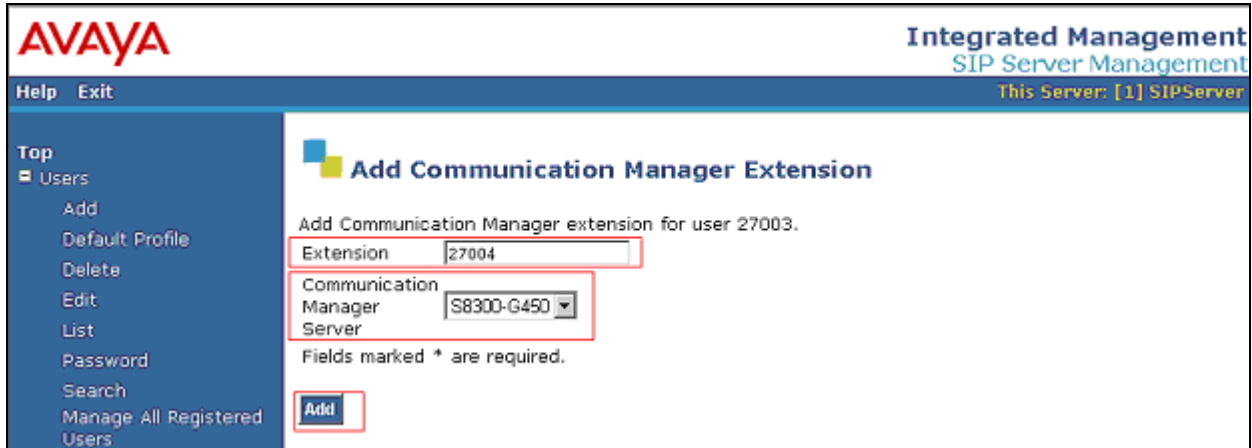
- Primary Handle*: 27004
- User ID: 27004
- Password*: *****
- Confirm Password*: *****
- Host*: 192.45.80.101
- First Name*: SIP
- Last Name*: 27004
- Address 1: [Empty]
- Address 2: [Empty]
- Office: [Empty]
- City: [Empty]
- State: [Empty]
- Country: [Empty]
- Zip: [Empty]
- Survivable Call Processor: none
- Add Communication Manager Extension:

Fields marked * are required.

The 'Add' button is located at the bottom left of the form.

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.



The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top left corner features the Avaya logo. The top right corner shows the text "Integrated Management SIP Server Management" and "This Server: [1] SIPServer". Below the header is a navigation bar with "Help" and "Exit" links. A left-hand menu lists various user management options: "Top", "Users", "Add", "Default Profile", "Delete", "Edit", "List", "Password", "Search", and "Manage All Registered Users". The main content area is titled "Add Communication Manager Extension" and contains the following text: "Add Communication Manager extension for user 27003." Below this, there are two input fields: "Extension" with the value "27004" and "Communication Manager Server" with a dropdown menu showing "S8300-G450". A note below the fields states "Fields marked * are required." At the bottom of the form is an "Add" button.

6. Configure Convergys Intervoice Media Server

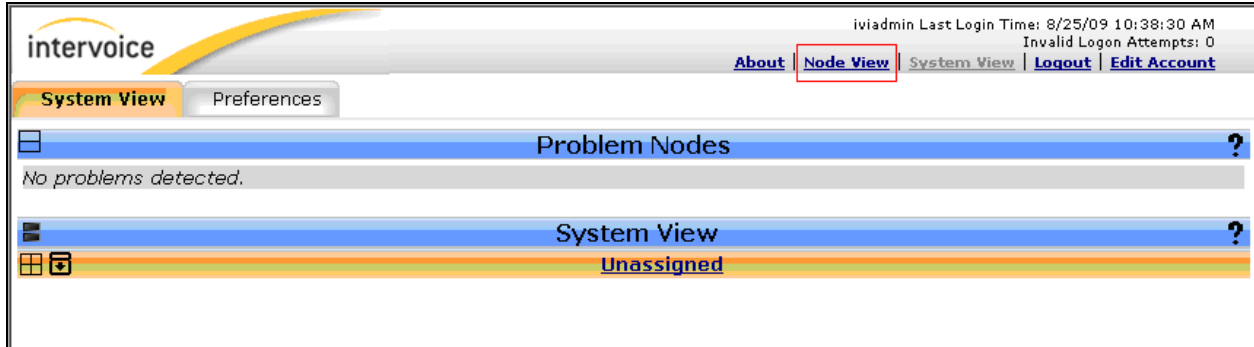
This section provides steps to configure Convergys Intervoice Media Server. Convergys installs, configures, and customizes the Intervoice Media Server application for end customers. This section describes the initial Intervoice Media Server configuration.

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.

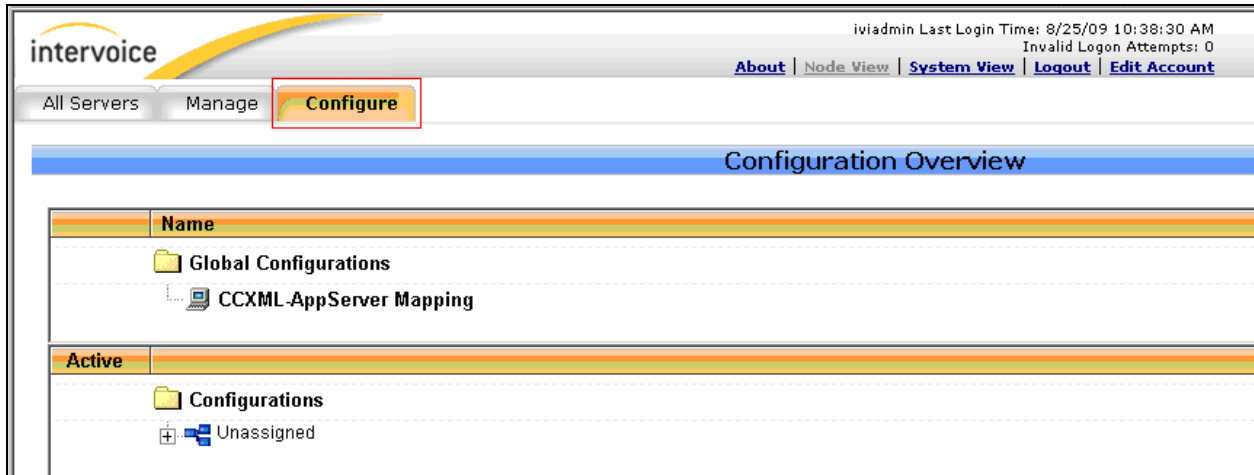


The screenshot shows the Intervoice Control Center login page. At the top left is the 'intervoice' logo. Below it is a banner image with the word 'Innovator' and a grid pattern. The main heading is 'Welcome to Control Center'. Below the heading are two input fields: 'Username' and 'Password'. A 'Login' button is positioned below the Password field. At the bottom of the page, there is a link for 'Browser Configuration Information'.

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.

The screenshot shows the 'intervoice' web interface. At the top, there's a navigation bar with 'All Servers', 'Manage', and 'Configure' tabs. Below that is a 'Configuration Overview' section. A tree view shows the following structure:

- Global Configurations
 - CCXML.AppServer Mapping (View)
- Active
 - Configurations
 - Unassigned
 - MS-18 (192.45.85.71)
 - Media Server 3.5 and above
 - Telephony Core (Add, Delete)
 - HAL HMP Configuration
 - New Config Description (Edit, Copy, Delete)
 - Call Control Configuration (Add, Delete)
 - Application Routing (*) (Add, Delete)
 - Dynamic Log Configuration (*) (Add, Delete)

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

The screenshot shows the 'intervoice' web interface for HALHMP configuration. The page title is '192.45.85.71 - View/Edit Boards (New Config Description)'. On the left, there's a sidebar menu with the following items:

- Configurations
- View/Edit Boards
- DTMF Payload & Fax
- System VOIP Parameters
- System VOIP Registration** (highlighted in red)
- System VOIP Configuration

The main content area displays a table with the following data:

Board ID	Actions
0	Edit Delete

Below the table, there is an [Add Board](#) link and two buttons: and .

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.

interviewice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HALHMP

Configurations | 192.45.85.71 - System VOIP Registration (New Config Description) [Back ?](#)

View/Edit Boards
DTMF Payload & Fax

System VOIP Parameters

System VOIP Registration

System VOIP Configuration

Enable VOIP Registration:

VOIP Registry Address	Hops Count	Registration Frequency	Registration Interval (ms)	Actions
192.45.80.101	100	PERIODIC	300000	Delete

[Add](#)

[Submit](#) [Revert](#)

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

interviewice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HALHMP

Configurations | 192.45.85.71 - System VOIP Configuration (New Config Description) [Back ?](#)

View/Edit Boards
DTMF Payload & Fax

System VOIP Parameters

System VOIP Registration

System VOIP Configuration

Enable Record AGC

Enable TCP

PCM Encoding (N/A for G711x Codecs) A-Law

Type of Service TOS

Type of Service Field

Outbound Proxy Address 192.45.80.101

SIP Contact Header IP 192.45.85.71

Inbound SIP Header Keys	Actions
	Add

[SIP Security Key Information](#)

[Submit](#) [Revert](#)

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

- SIP Domain
- User extension
- User Password

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

intervoice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HALHMP

Configurations | 192.45.85.71 - System VOIP Configuration (New Config Description) [Back ?](#)

View/Edit Boards
DTMF Payload & Fax
System VOIP Parameters
System VOIP Registration
System VOIP Configuration

Enable Record AGC
 Enable TCP
 PCM Encoding (N/A for G711x Codecs) A-Law
 Type of Service TOS
 Type of Service Field
 Outbound Proxy Address: 192.45.80.101
 SIP Contact Header IP: 192.45.85.71

Inbound SIP Header Keys	Actions
	Add

SIP Security Key Information

Realm	User Name	Password	Actions
moris.mot.com	27005	*****	Delete
			Add

[Submit](#) [Revert](#)

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

intervoice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HALHMP

Configurations | 192.45.85.71 - System VOIP Configuration (New Config Description) [Back ?](#)

[View/Edit Boards](#)
DTMF Payload & Fax
System VOIP Parameters
System VOIP Registration
System VOIP Configuration

Enable Record AGC
 Enable TCP
 PCM Encoding (N/A for G711x Codecs) A-Law
 Type of Service TOS
 Type of Service Field
 Outbound Proxy Address: 192.45.80.101
 SIP Contact Header IP: 192.45.85.71

Inbound SIP Header Keys	Actions
	Add

SIP Security Key Information

Realm	User Name	Password	Actions
moris.mot.com	27005	*****	Delete
			Add

[Submit](#) [Revert](#)

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

interoice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HALHMP

Configurations **192.45.85.71 - View/Edit Boards (New Config Description)** [Back ?](#)

View/Edit Boards
DTMF Payload & Fax
System VOIP Parameters
System VOIP Registration
System VOIP Configuration

Board ID	Actions
0	Edit Delete
Add Board	

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 Interoice Media Server.

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

interoice iviadmin Last Login Time: 8/25/09 10:38:30 AM
Invalid Logon Attempts: 0
[About](#) | [Node View](#) | [System View](#) | [Logout](#) | [Edit Account](#)

HMP Board

Configurations **192.45.85.71 - Board VOIP Registration - Board 0** [Back ?](#)

Board
Board VOIP Registration
Codec
Alarm

Configuration below must be filled out before enabling System VOIP Registration.

Board Client Bind

Board Alias Configuration		
Code ID	Alias String	Actions
E-mail	27005@moris.mot.com	Delete
		Add

7. General Test Approach and Test Results

The general test approach was to verify the SIP protocol messages between SES and Convergys Intervice Media Server, when calls were manually placed to Convergys Intervice 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 Intervice 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 Intervice Media Server.
- Blind transfer was verified from Convergys Intervice Media Server to an agent.

9. Conclusion

Convergys Intervice 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 Intervice Media Server when a call was placed to Convergys Intervice 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

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