Avaya Solution & Interoperability Test Lab

Application Notes for the Intervoice IVR MediaServer Endpoint 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, and Intervoice MediaServer. The Intervoice MediaServer (MS) is SIP based VoIP software which provides an IVR driven-menu for executing Voice Extensible Markup Language (VXML) based applications. For the purpose of compliance testing, several demo VXML IVR applications were provided by Intervoice Inc. to exercise SIP call flows between Intervoice MS ports registered with Avaya SES and SIP and non-SIP telephones.

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 a solution comprised of Avaya Communication Manager 4.0, Avaya SIP Enablement Services 3.1.2, and Intervoice MediaServer (MS) 3.5. The Intervoice MS is a SIP based VoIP software which provides an IVR driven-menu for executing Voice Extensible Markup Language (VXML) based applications. For the purpose of compliance testing, several demo VXML IVR applications were provided by Intervoice Inc. to exercise SIP call flows between Intervoice MS ports registered with Avaya SES and SIP and non-SIP telephones.

Figure 1 illustrates a sample configuration consisting of Avaya S8710 Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and Intervoice MS. Avaya Communication Manager was installed on S8710 Servers. The solution described herein is also extensible to other Avaya Servers and Media Gateways. For completeness Avaya 4600 Series SIP IP Telephones, Avaya one-X™ Desktop Edition, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls with the SIP-based Intervoice MS 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 Intervoice MS.

The Intervoice MS **registers** t with Avaya SES for a pre-determined number of ports. Using this mechanism keeps track of the number of Intervoice MS ports in service. The Intervoice MS is configured with G711 and G729 using RFC2823 for DTMF.

Typical call flows in this configuration between Avaya Communication Manager and Avaya SES, and the Intervoice MS are as follows:

- Calls originate from PSTN, H.323 or SIP trunks/endpoints to a destination number associated with the Intervoice MS.
- Avaya Communication Manager uses Automatic Alternate Routing (AAR) to route the call to Avaya SES over SIP trunks.
- Avaya SES looks up its registration table to route the call to the Intervoice MS.

The Intervoice MS either completes the call or transfers the call to an extension in Avaya Communication Manager by sending a REFER message to Avaya SES.

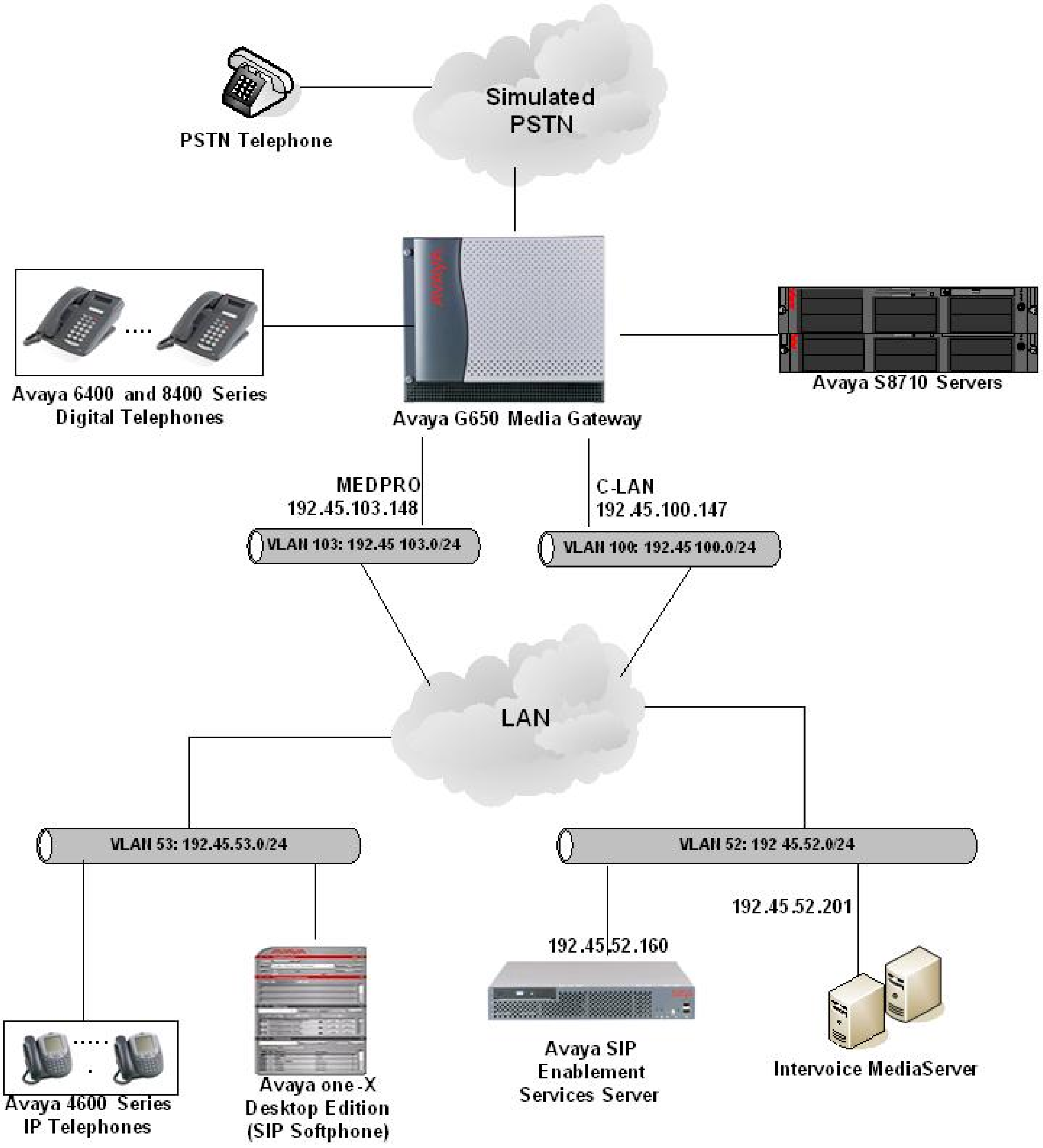


Figure 1: Sample configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Server	Avaya Communication Manager 4.0.0 (R014x.00.0.730.5)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12, FW039
TN799DP C-LAN Interface	HW01, FW024
TN2302AP IP Media Processor	HW20, FW116
Avaya SIP Enablement Services Server	3.1.2 (SES-3.1.2.0-309.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.3 (4610SW SIP)
Avaya one-X Desktop IP Phone	R2.1 SP1
Avaya 6400 and 8400 Series Digital Telephones	-
Analog Telephone	-
Intervoice MediaServer	3.5

3. Configure Avaya Communication Manager

This section describes a procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES which includes steps for setting up an IP codec set, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Refer to [1] for additional details.

3.1. Capacity Verification

Step	Description
1.	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.

```
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 1194
                                      Maximum Stations: 36000 446
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones -   OPS: 200   70
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones – PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
         (NOTE: You must logoff & login to effect the permission changes.)
```

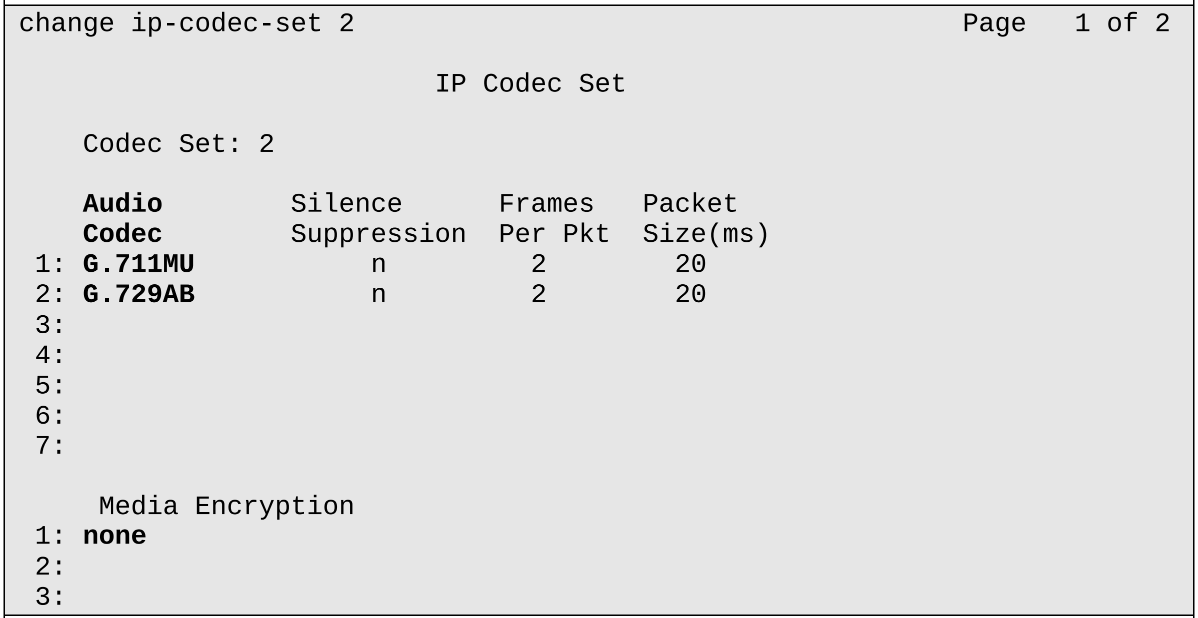
Step	Description
2.	Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses. **Note:** *Each 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.*

```
display system-parameters customer-options                      Page   2  of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 400   298
           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: 100   0
                  Maximum Video Capable H.323 Stations: 100   12
                   Maximum Video Capable IP Softphones: 100   6
                       Maximum Administered SIP Trunks: 5000  253

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                    Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

3.2. IP Codec Set

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

Step	Description
1.	Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 3.3** for configuring IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, **G.711MU** and **G.729AB** were used and **Media Encryption** was set to **none** as encryption is currently not supported for SIP telephony.



```
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.729AB           n         2        20
 3:
 4:
 5:
 6:
 7:

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

3.3. IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive and configure as follows: • **Authoritative Domain –**This should match the **SIP Domain** value in **Section 4**, **Step 2**. • **Intra-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 SES in the same IP network region. • **Codec Set –** Set the codec set number as provisioned in **Section 3.2**. • **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 SES in different IP network regions.

```
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                RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled: y
   Keep-Alive Interval (sec): 5                  RSVP Profile: guaranteed-service
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 46
```

Step	Description
2.	Proceed to **Page 3** of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, **codec set** was set to the IP codec set configured in **Section 3.2**.

```
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
```

3.4. IP Node Names

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

Step	Description
1.	Enter the **change node-names ip** command and add a node name for Avaya SES along with its IP address.

```
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
```

3.5. 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.

Step	Description
1.	Issue the command **add signaling-group <s>**, where **s** is an available signaling group and configure the following: • **Group Type –** Set to **sip.** • **Transport Method –** Set to **tls**. • **Near-end Node Name** - Set to CLAN name as displayed in **Section 3.4**. • **Far-end Node Name** - Set to Avaya SES name configured in **Section 3.4**. • **Far-end Network Region** - Set to the region configured in **Section 3.3**. • **Far-end Domain** - This should match the **SIP Domain** value in **Section 4**, **Step 2**. • **DTMF over IP** - Set to **rtp-payload** (RFC2833).

```
add signaling-group 10                                               Page 1 of 1
                                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
 Enable Layer 3 Test? n
 Session Establishment Timer(min): 120
```

3.6. 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 **add trunk-group <t>**, where **t** is an unallocated trunk group and configure the following: • **Group Type** – Set to the **Group Type** field value configured in **Section 3.5**. • **Group Name** – Enter any descriptive name. • **TAC** (Trunk Access Code) – Set to any available trunk access code. • **Signaling Group** – Set to the **Group Number** field value configured in **Section 3.5**. • **Number of Members –** Allowed values are between **0** and **255**. Set to a value large enough to accommodate the number of SIP telephone extensions being used. **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.*

```
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
```

3.7. 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 Intervoice MS.

Step	Description
1.	Issue the command **change uniform-dialplan <dialstring>** where dialstring is the string to match for the dialed number, and configure as follows: • **Matching Pattern –** Set it to a value for routing calls to Avaya SES for proper AAR digit analysis • **Len –** The dialed string length to be analyzed. • **Del** – Set to **0**. • **Net** – Set to **aar**.

```
change uniform-dialplan 5                                          Page 1 of 2
                             UNIFORM DIAL PLAN TABLE
                                                           Percent Full:    02


 Matching                Inserted                 Node
 Pattern   Len  Del      Digits      Net  Conv    Num
 54000      5    0                   aar   n
                                           n
                                           n
```

Step	Description
2.	Issue the command **change public-unknown-numbering <e>**, where **e** is extension code to be administered. • **Ext Len –** Set to the length of calling party number. • **Ext Code** – Extension Code to be administered. Set to **5** in this example. • **Trk Grp<s>** - Trunk Group/s where the call is coming on. • **Total CPN Len** – Length of the dialed number.

```
change public-unknown-numbering 5                              Page   1  of  2
                    NUMBERING – PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext          Trk       CPN             CPN
Len Code         Grp<s>    Prefix          Len

5   5            21                         5
```

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 3.6**.
- **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
```

4. Issue the command **change aar analysis 5** and configure as follows:

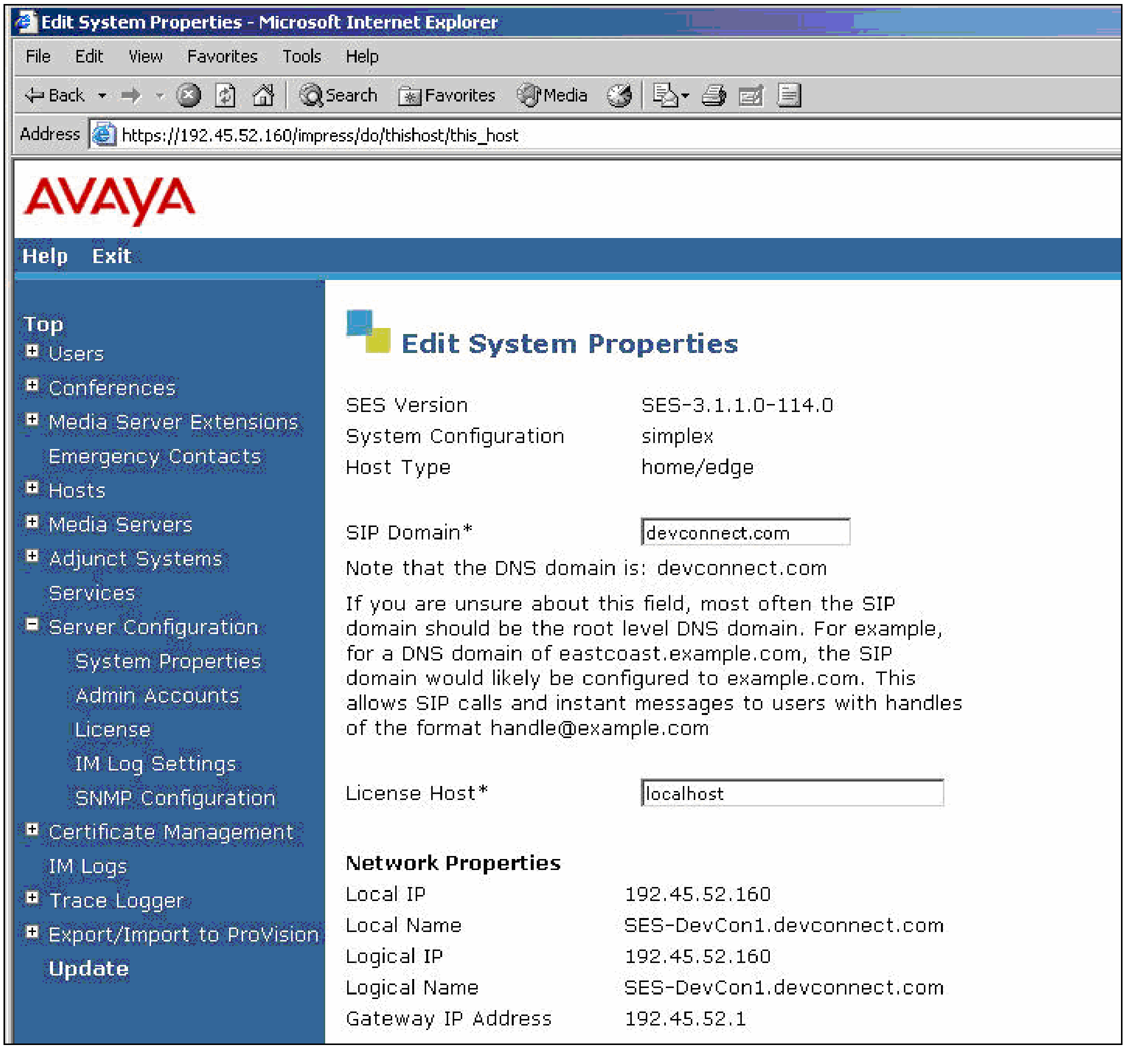
- **Dialed String –** Set it to same value as **Matching Pattern** in **Step 1**.
- **Total Min and Max –** Set it to same value as **Len** in **Step 1.**
- **Route Pattern –** Set a value for a route pattern defined in **Step 3**.
- **Call Type –** Set to **aar**
- **ANI Reqd –** Set to **n**

```
Change aar analysis 5                                               Page 1 of 2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    54000                 5    5     10       aar          n
    2                     5    5     15       aar          n
    2                     7    7     999      aar          n
    245                   5    5     33       aar          n
```

4. Configure Avaya SIP Enablement Services

This section describes the steps for configuring Avaya SES to communicate with Avaya Communication Manager and the Intervoice MS. The Intervoice MS will register with Avaya SES as an endpoint using the SIP user accounts. Refer to [3, 4] for additional details.

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.

Step	Description
2.	On the **SIP Server Management** page: • Click the **+** sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** field value configured for the signaling group on Avaya Communication Manager in **Section 3.5**. 

Edit System Properties

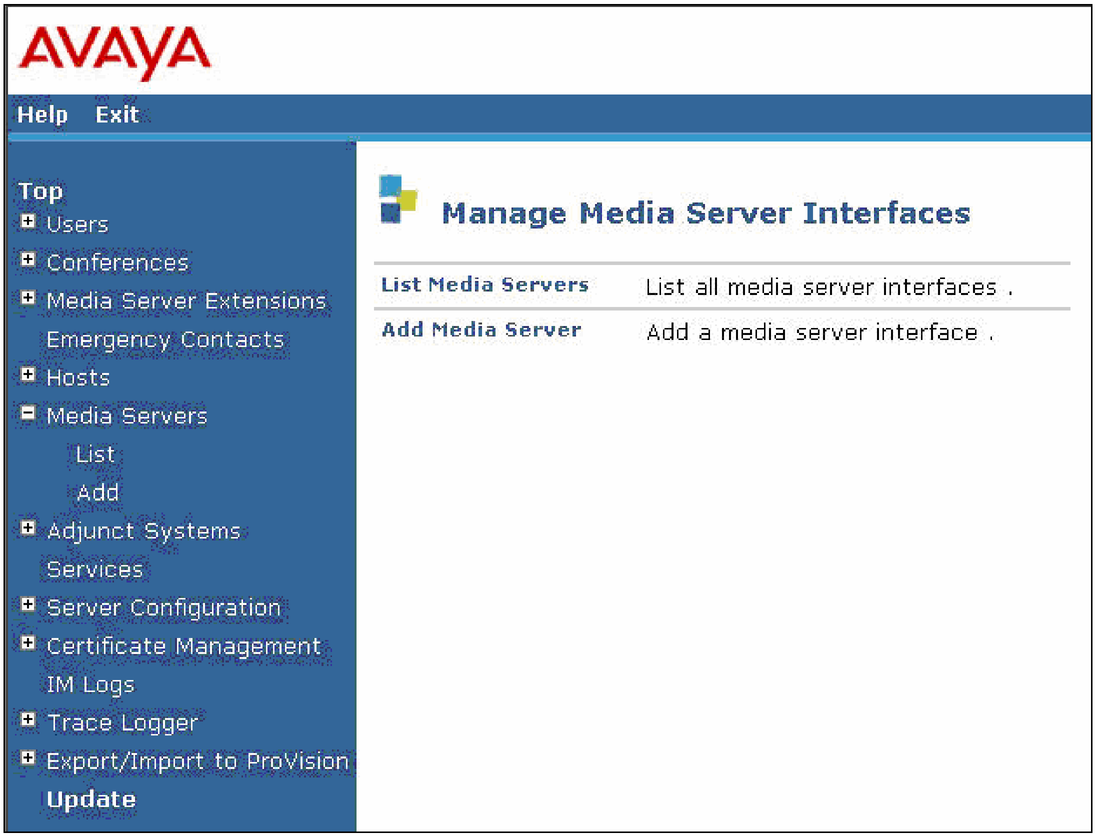
SES Version SES-3.1.1.0-114.0
System Configuration simplex
Host Type home/edge

SIP Domain* devconnect.com
Note that the DNS domain is: devconnect.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

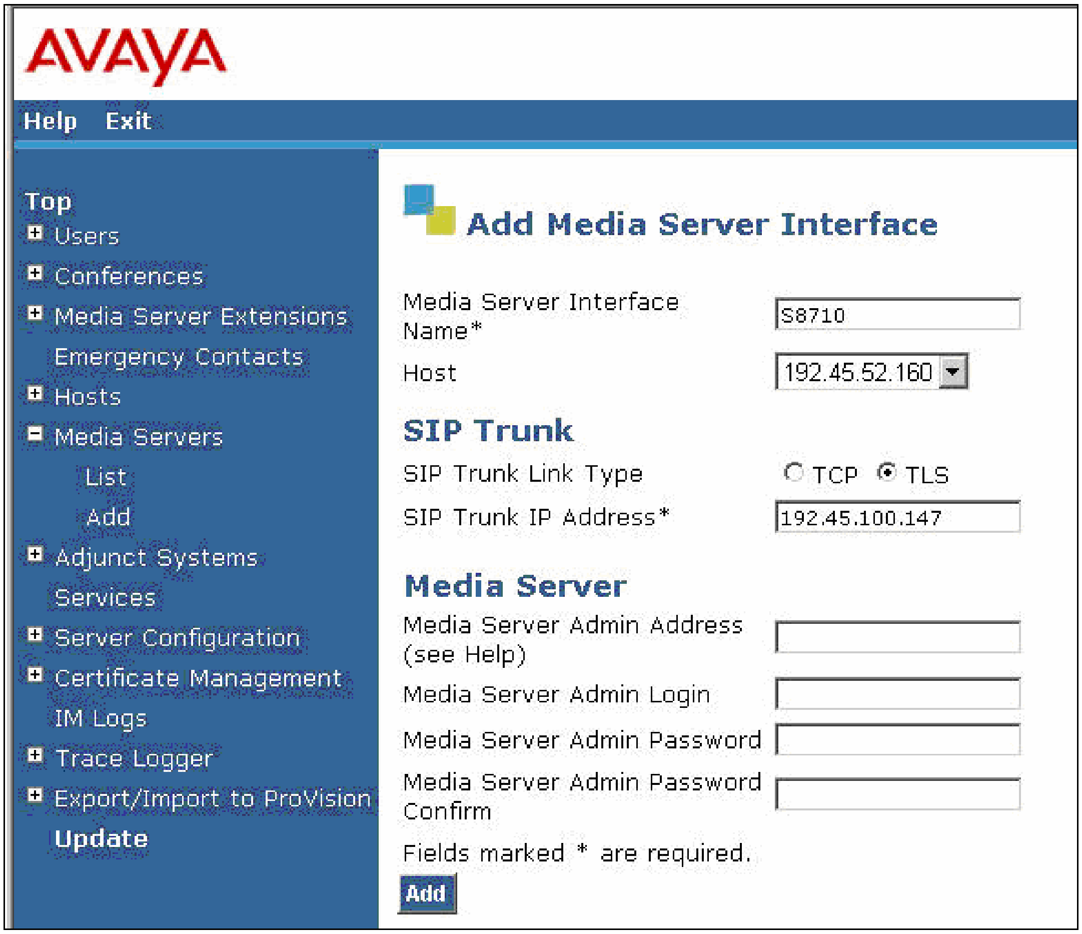
License Host* localhost

Network Properties
Local IP 192.45.52.160
Local Name SES-DevCon1.devconnect.com
Logical IP 192.45.52.160
Logical Name SES-DevCon1.devconnect.com
Gateway IP Address 192.45.52.1

Step	Description
3.	To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a media server corresponding to Avaya Communication Manager from the **SIP Server Management** page: • Click the + sign to expand the options under **Media Servers**. • Click **Add**. 

Manage Media Server Interfaces

List Media Servers	List all media server interfaces .
Add Media Server	Add a media server interface .

Step	Description
4.	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 the **Transport Method** field value in **Section 3.5**. • **SIP Trunk IP Address -** Set to the CLAN IP address as displayed in **Section 3.4**. • Click **Add** when finished and then click **Continue** on the confirmation page [not shown]. 

Add Media Server Interface

Media Server Interface Name* S8710

Host 192.45.52.160

SIP Trunk

SIP Trunk Link Type ○ TCP ● TLS

SIP Trunk IP Address* 192.45.100.147

Media Server

Media Server Admin Address (see Help)

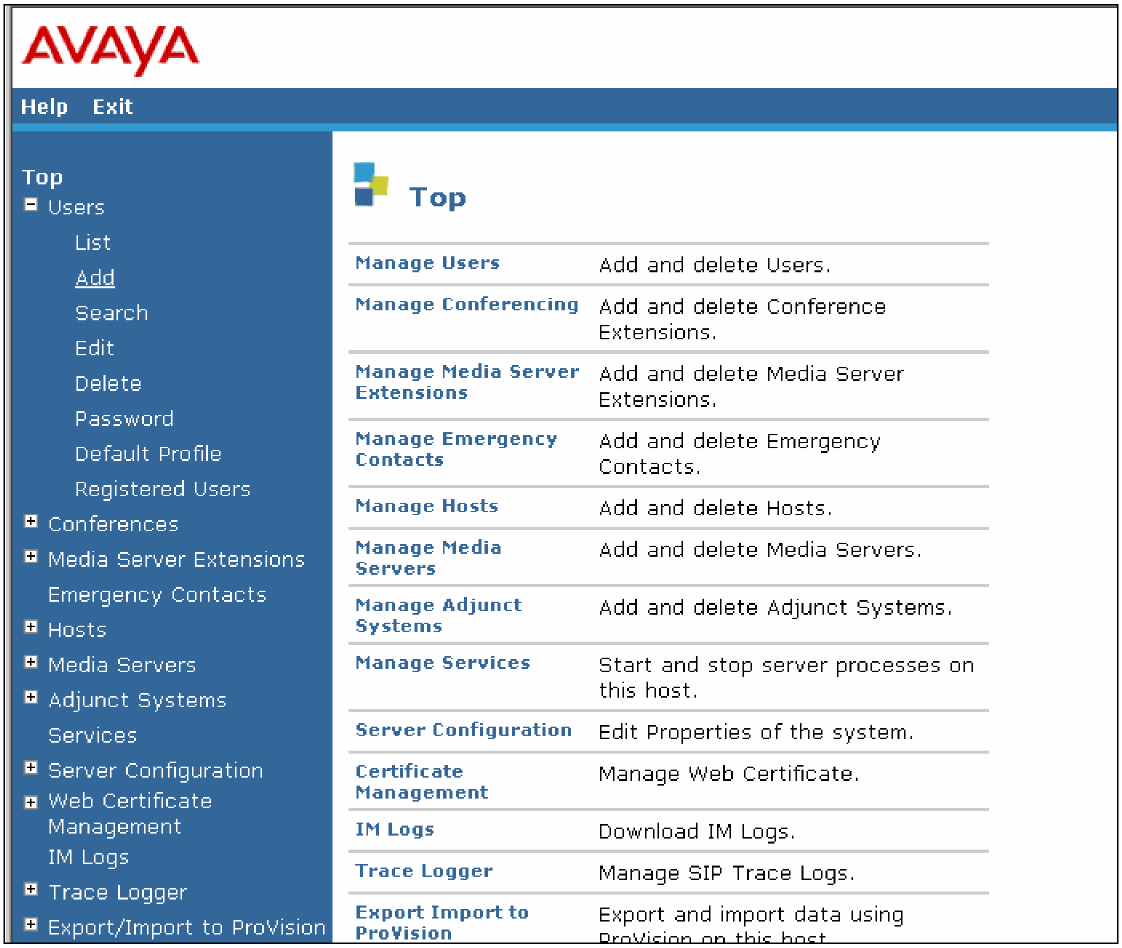
Media Server Admin Login

Media Server Admin Password

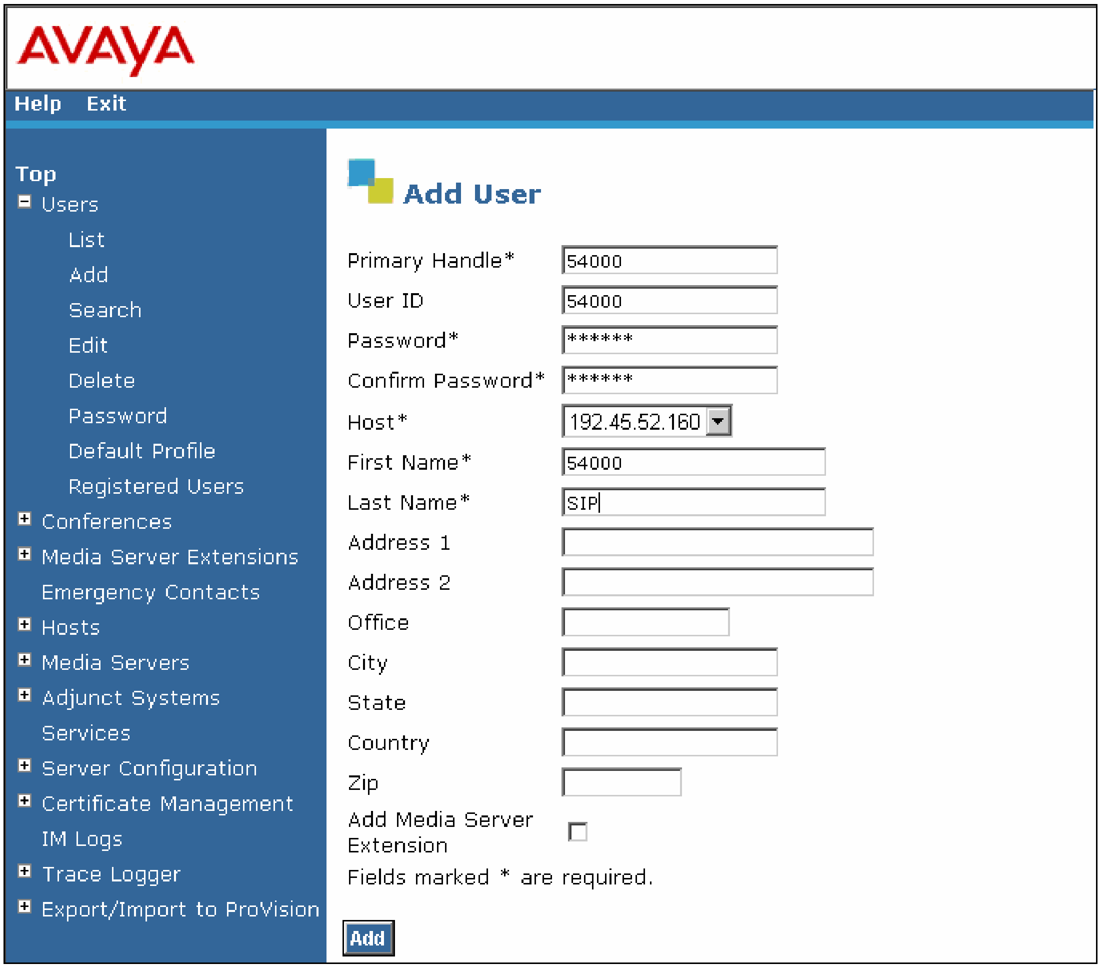
Media Server Admin Password Confirm

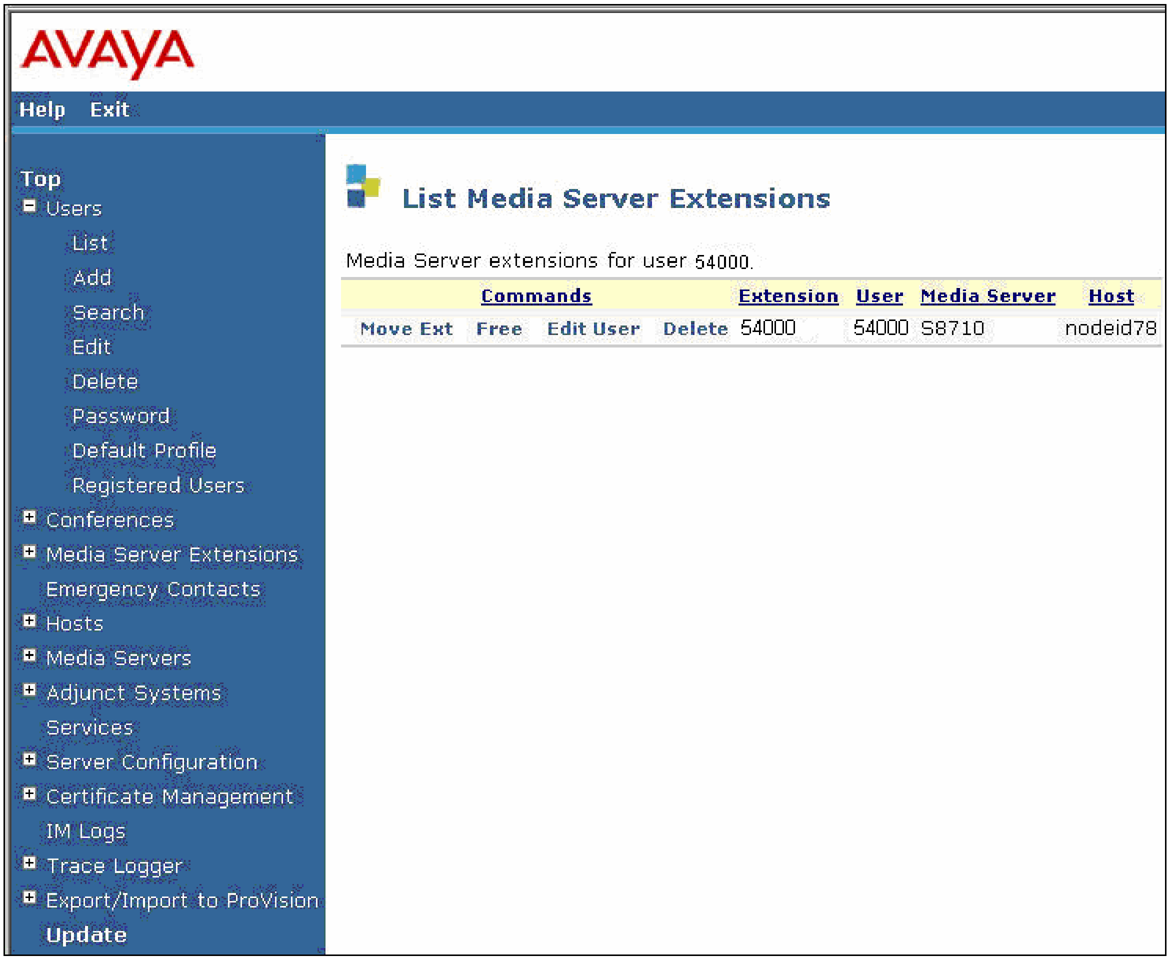
Fields marked * are required.

Step	Description
5.	In the left pane of the **SIP Server Management** page, expand **Users** and click **Add**.



Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Web Certificate.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host

Step	Description
6.	In the **Add User** page, configure the following: • **Primary Handle –** Enter the Intervoice MS extension. This must match the **phone number** entered in **Section 3.7 Step 3**. • **Password** and **Confirm Password –** Specify a password that the Intervoice MS must use to successfully register with Avaya SES. • **Host –** Select the IP address or Fully Qualified Domain Name (FQDN) of the Avaya SES server. • **First Name** and **Last Name –** Enter descriptive names. • Click **Add** when finished and then click **Continue** on the next page [not shown].  Add User Primary Handle* 54000 User ID 54000 Password* ****** Confirm Password* ****** Host* 192.45.52.160 First Name* 54000 Last Name* SIP Address 1 Address 2 Office City State Country Zip Add Media Server Extension ☐ Fields marked * are required. Add

Step	Description
7.	Click **Update** at the bottom of the left pane and then click **Continue** at the bottom of the next screen [not shown]. 

List Media Server Extensions

Media Server extensions for user 54000.

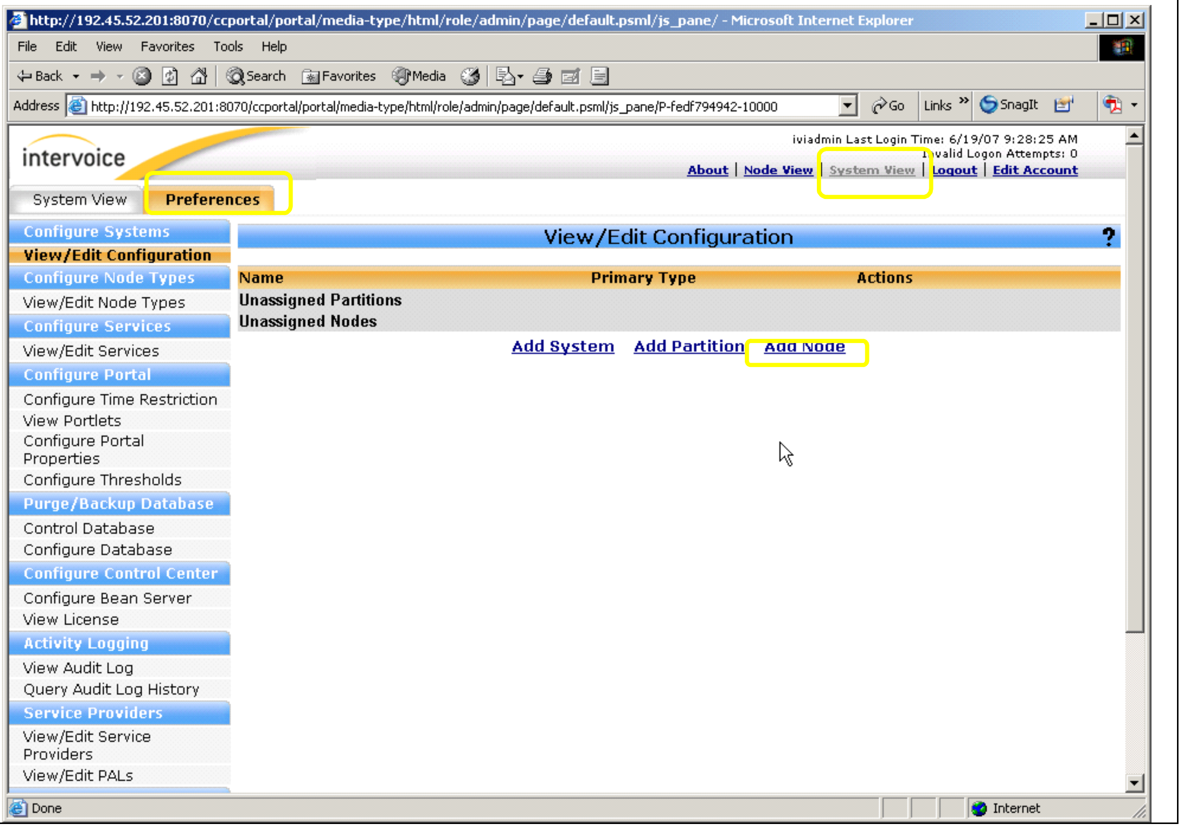
Commands				Extension	User	Media Server	Host
Move Ext	Free	Edit User	Delete	54000	54000	S8710	nodeid78

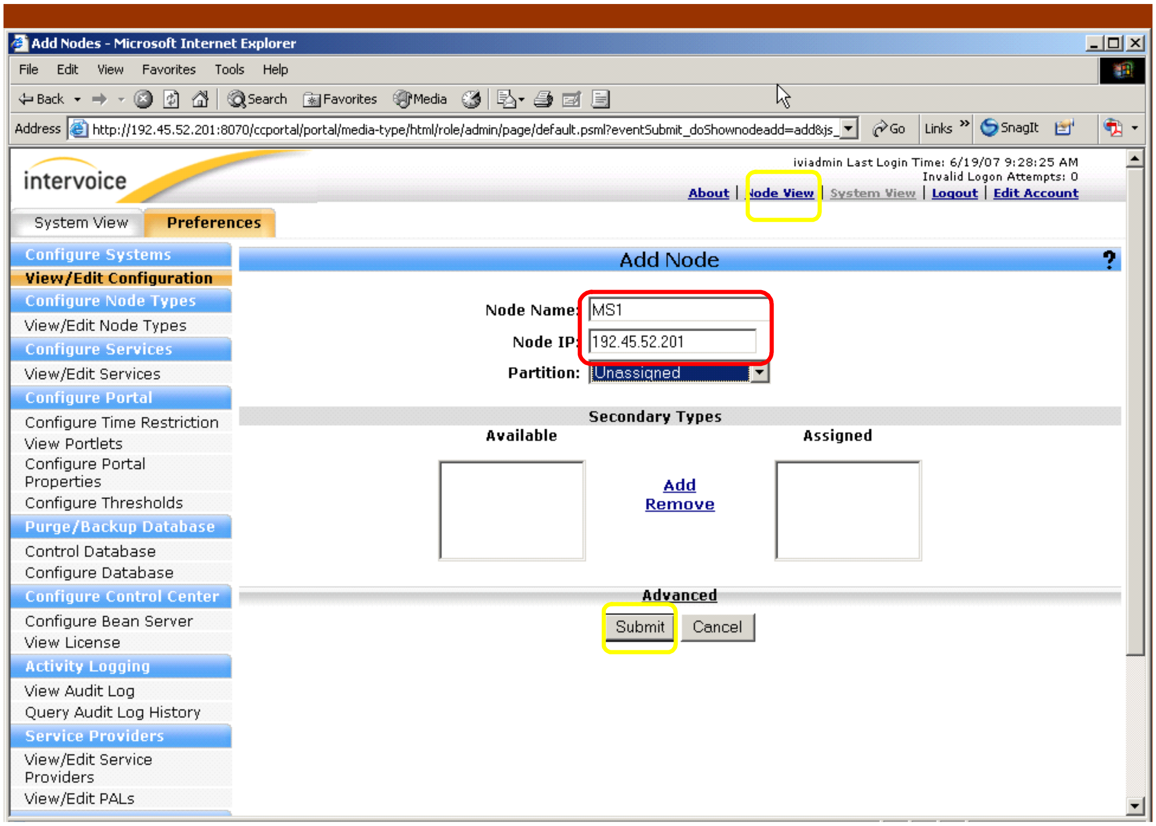
5. Configure the Intervoice MediaServer

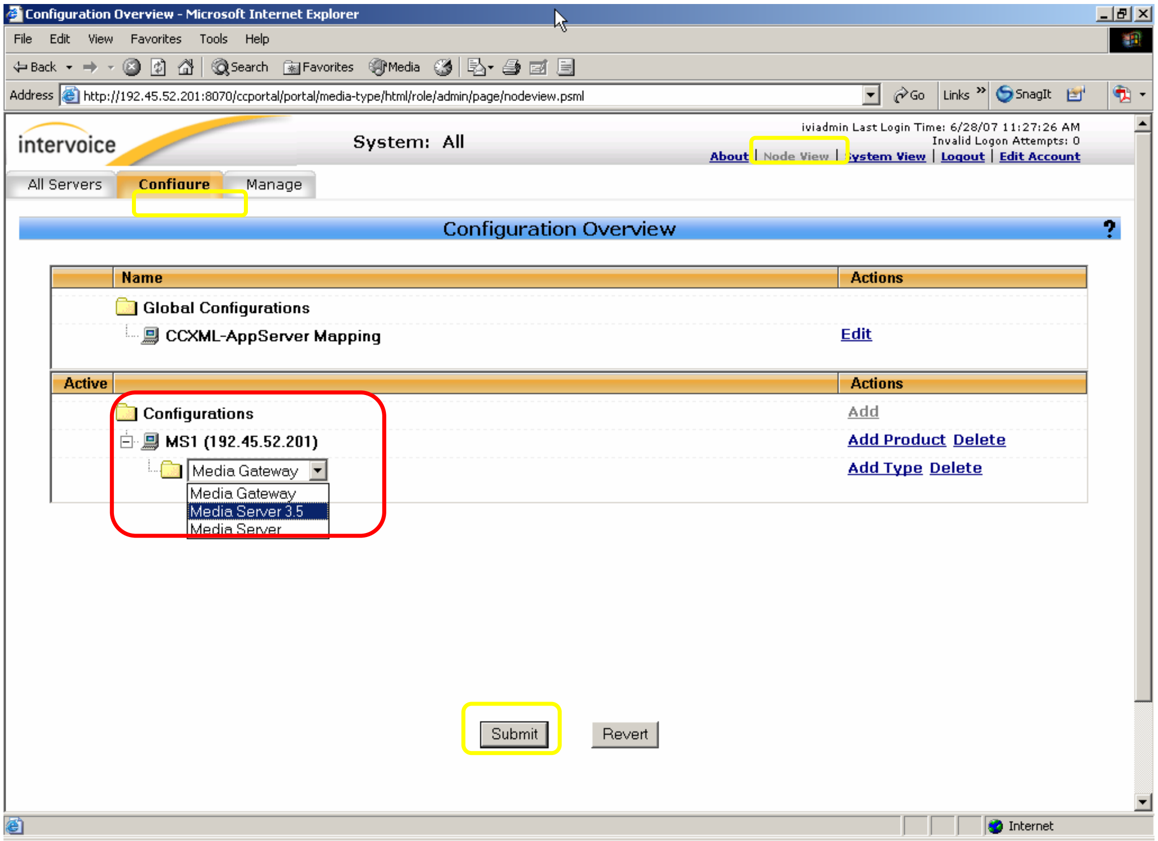
This section describes the steps for configuring the Intervoice MS to register endpoints with Avaya SES. This section assumes that the Intervoice MS software and Media Control Center configuration utility are already installed and IP addresses are set. Configuration steps described in this section apply only to the fields where a value needs to be modified or entered. Default values are used for all other fields. Only relevant screens and configuration steps are shown here. For further details refer to Intervoice Inc. documentation.

Note: Due to the page size, only the most relevant fields have been included in the screen shots.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Open a web browser, enter http://<ip address of MediaServer>:8074/ccportal/ for the URL, and log in with the appropriate credentials.</td></tr>
<tr><td>2.</td><td>Select the System View button on the top, click Preferences and click Add Node.


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

Step	Description
3.	On the **Add Node** screen, configure as follows: • **Node Name** – Enter any descriptive name. • **Node IP** – Set to the IP address configured for Intervoice MS**.** • Click **Submit**.  Node Name: MS1 Node IP: 192.45.52.201 Partition: Unassigned

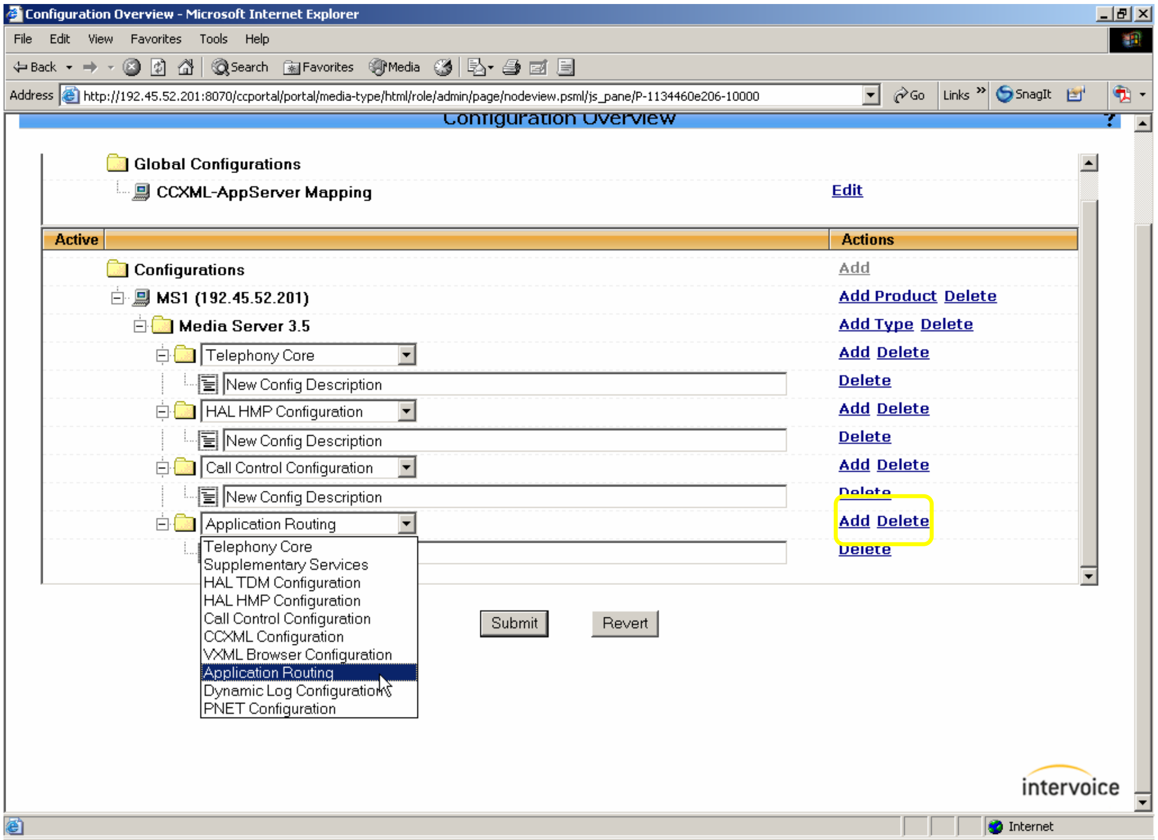
Step	Description
4.	At the **System: All** screen, configure as follows: • Select the **Node View** on the top and then select **Configure**. • Click **Add** on the **Configurations** line to display the node configured in **Step 3**. • Click **Add Product** and select the **Media Server 3.5** from the list of choices. • Click **Submit**. 

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

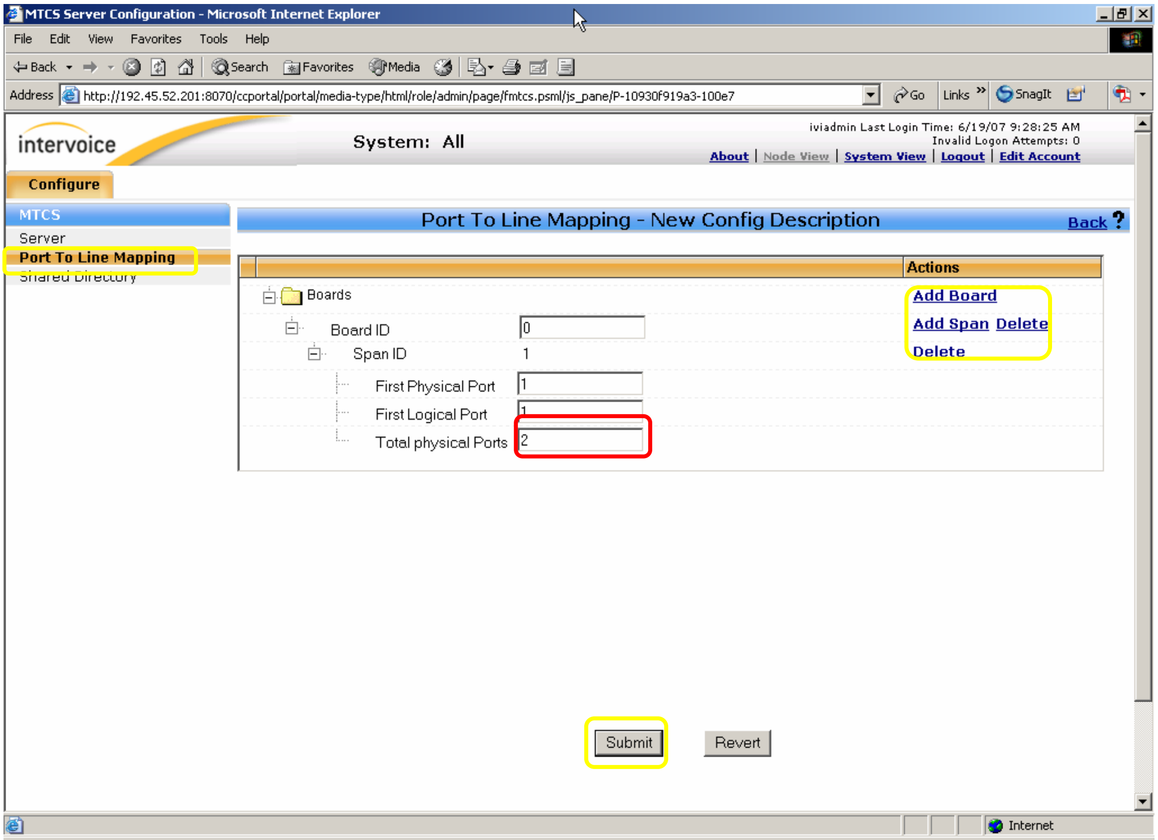
Only one screen shown for following steps:

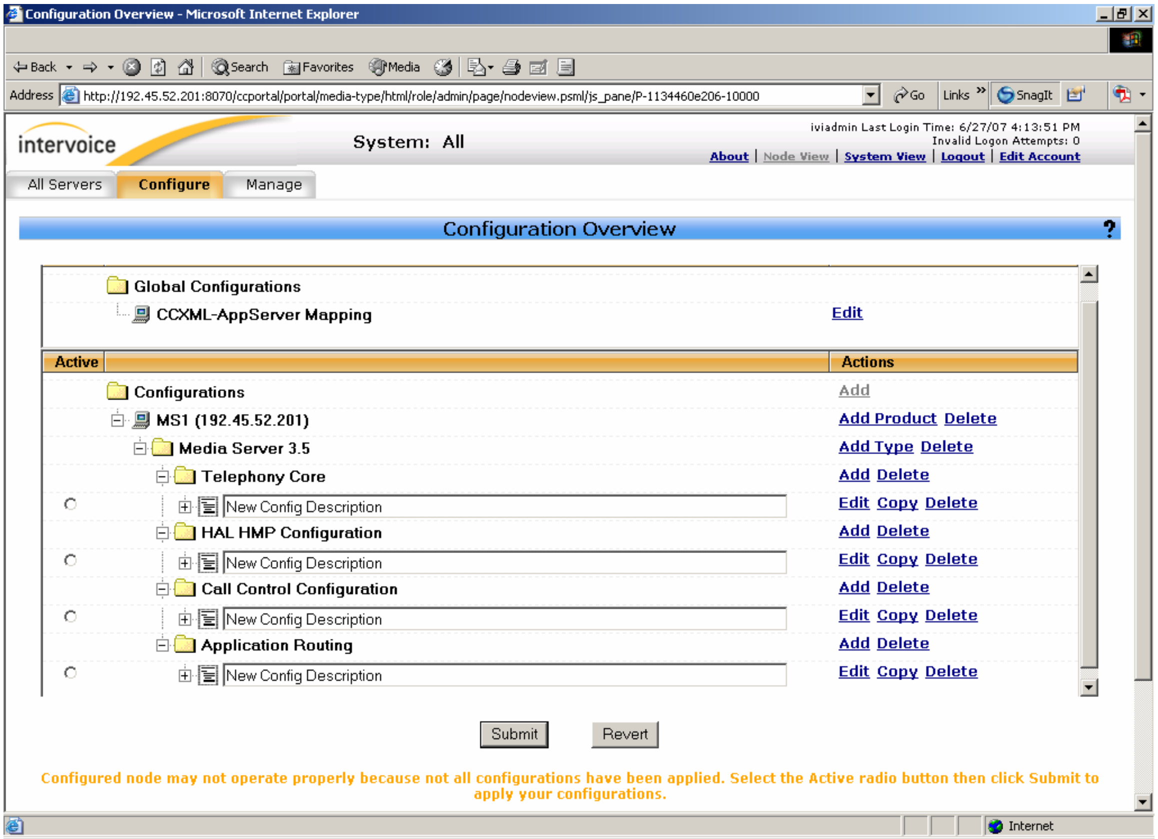
- Click **+** in the **MS1** line to expose the **Media Server 3.5** line.
- Click **+** in the **Media Server 3.5** line to expose **Telephony Core** line.
- Click **Add Type** and select **Telephony Core** from the list of choices.
- Click **Add** and next line with **New Config Description** appears.

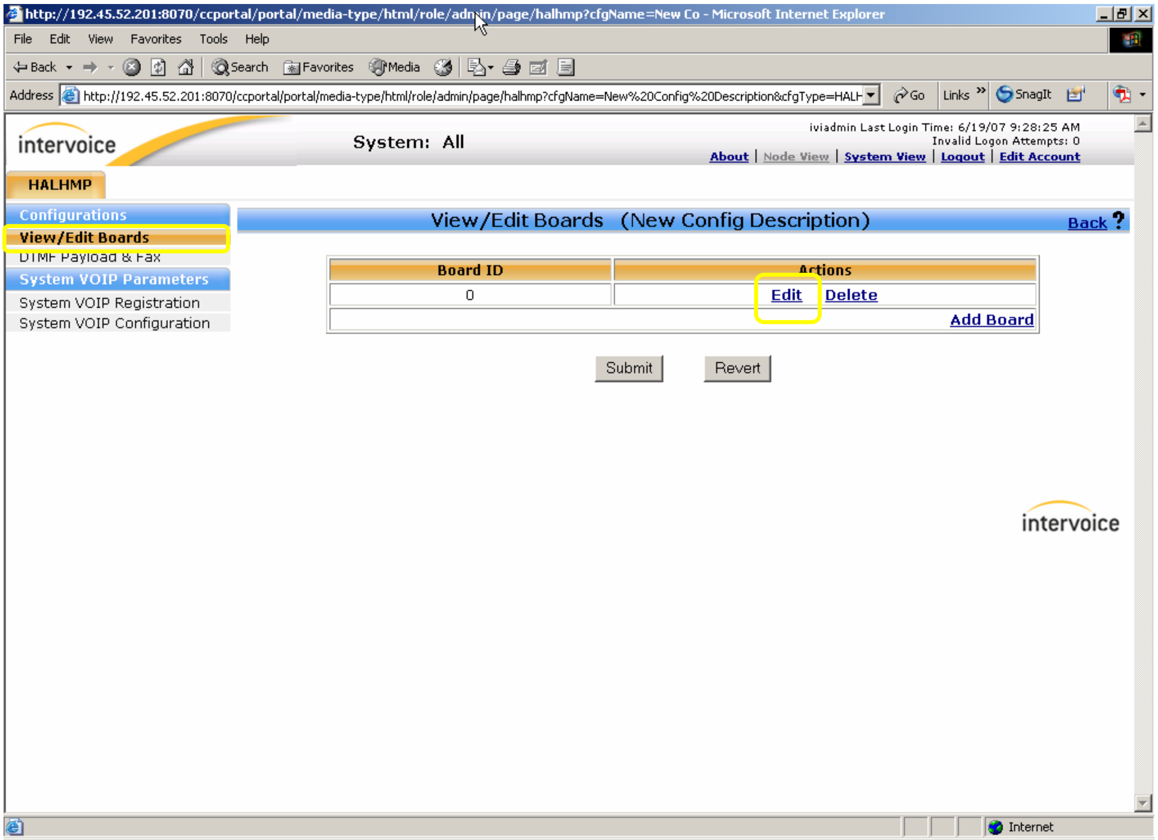
Note: This step is repeated for all the configured features like Telephony Core, HAL HMP configuration, etc.



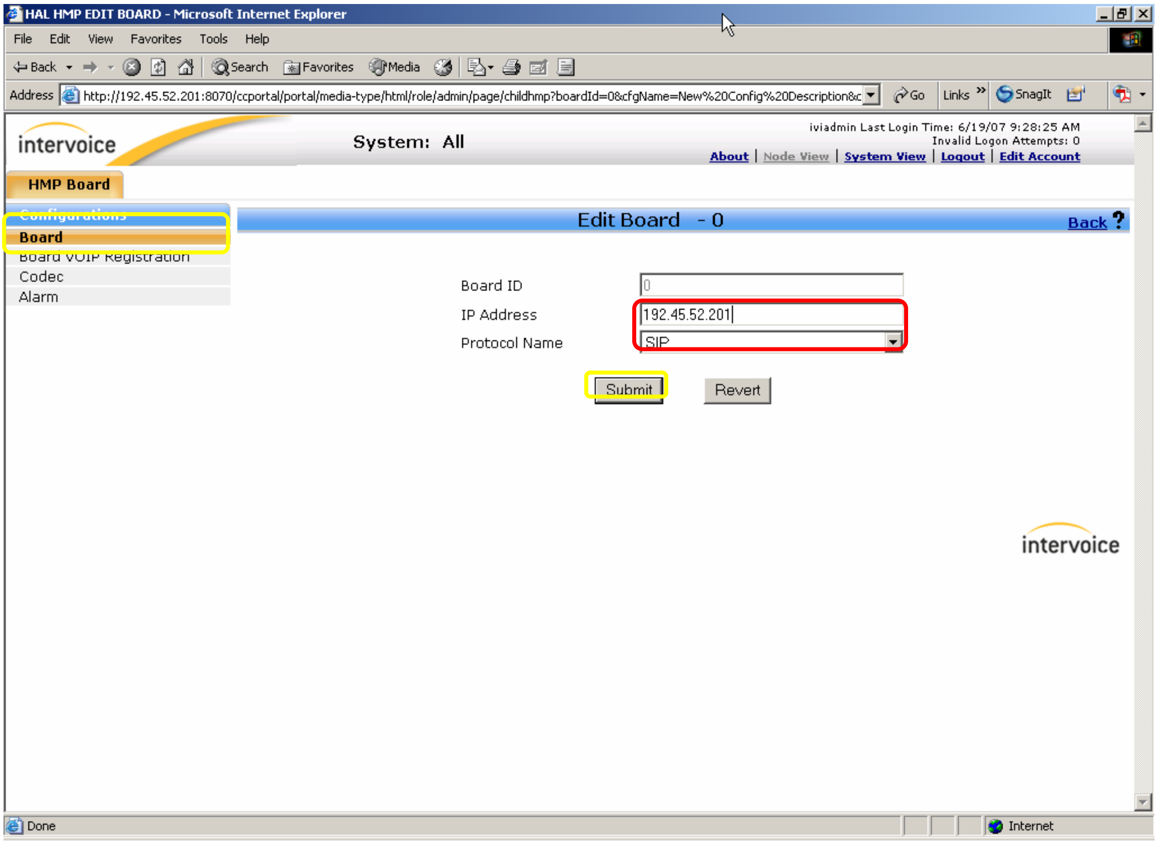
</td></tr>
</table>

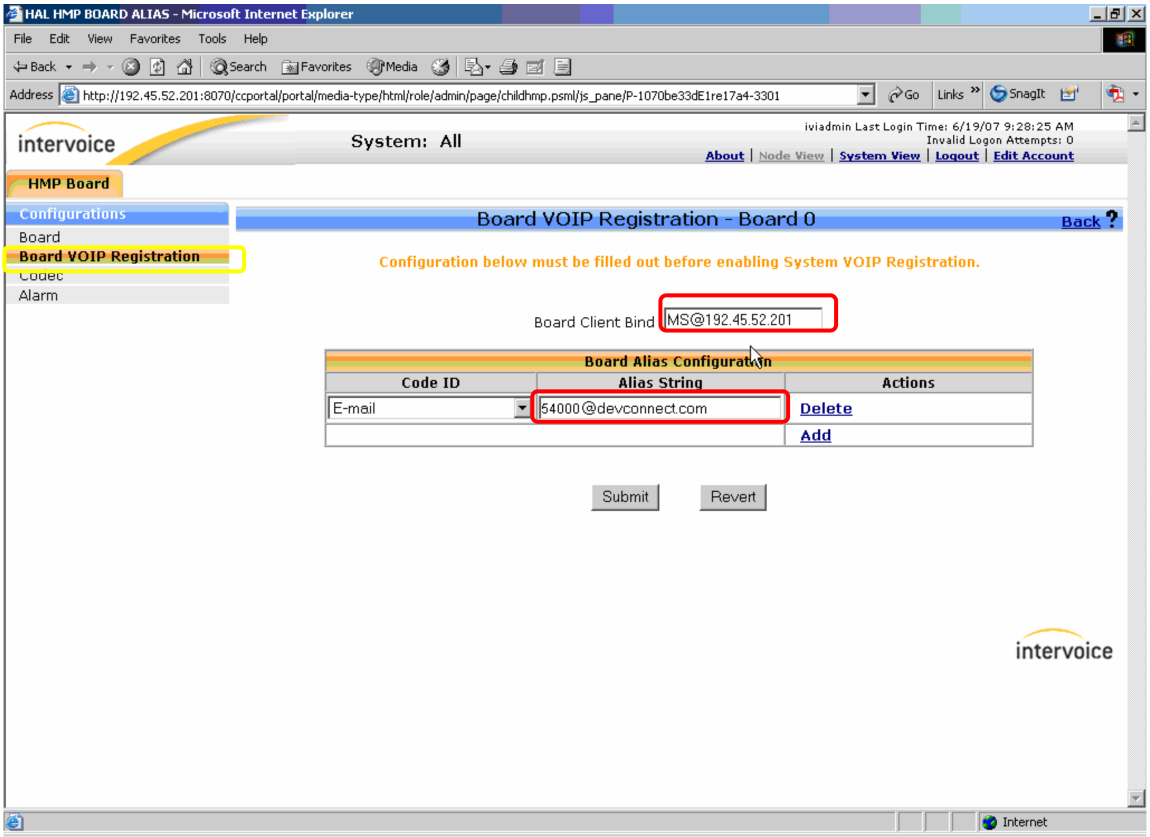
Step	Description
6.	Select **Port to Line Mapping** in the left pane of the **Configure** screen and configure as follows: • At the **Boards** line, click **Add Board** and **BoardID** line appears. • At the **BoardID** line, click **Add Span** and the next three fields appear. • **Total physical Ports** – Set to a value depending upon available licenses. • Click **Submit**.  MTCS Server Configuration - Microsoft Internet Explorer Address http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/fmtcs.psml/js_pane/P-10930f919a3-100e7 iviadmin Last Login Time: 6/19/07 9:28:25 AM Invalid Logon Attempts: 0 System: All Port To Line Mapping - New Config Description Board ID: 0 Span ID: 1 First Physical Port: 1 First Logical Port: 1 Total physical Ports: 2

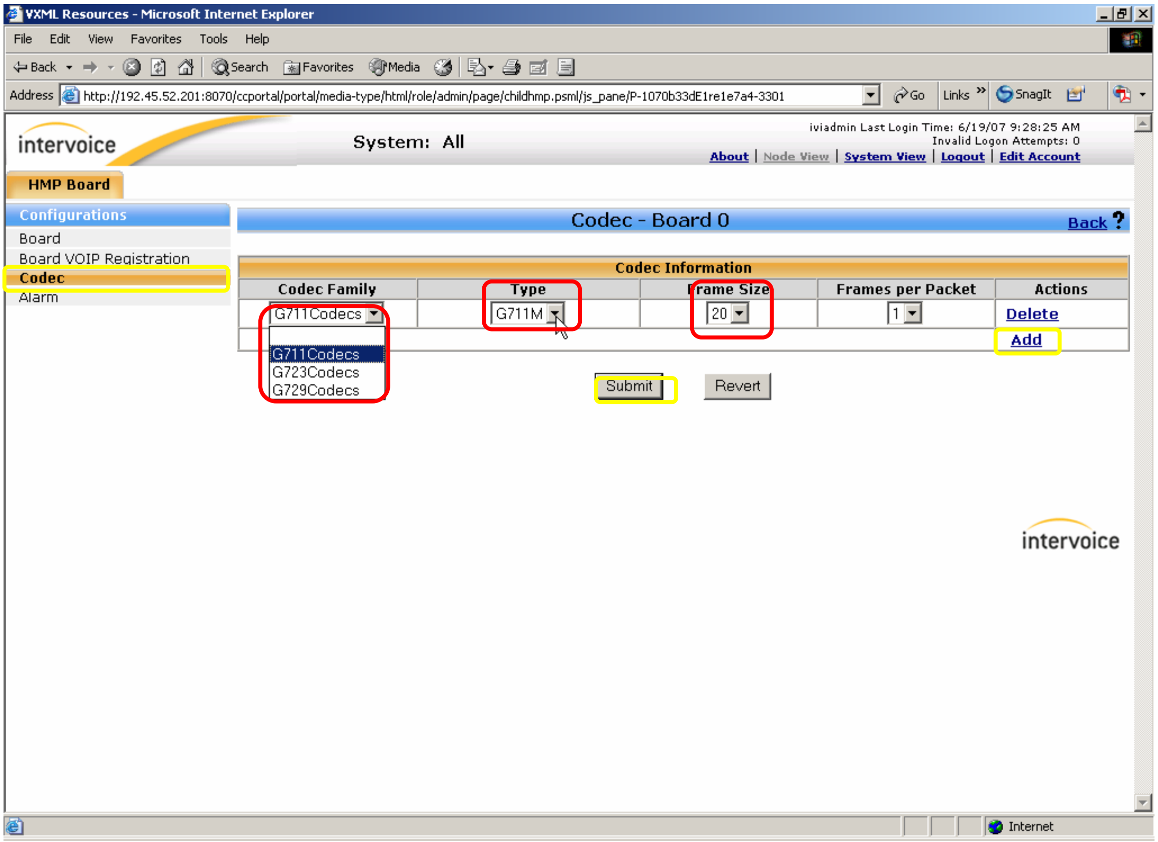
Step	Description
7.	Select the **Active** radio button associated with to activate the **Telephony Core** feature and click **Submit** to activate. Note: This step is repeated for all the configured features like Telephony Core, HAL HMP configuration, etc.  Configuration Overview - Microsoft Internet Explorer Address http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/nodeview.psml/js_pane/P-1134460e206-10000 iviadmin Last Login Time: 6/27/07 4:13:51 PM Invalid Logon Attempts: 0 Configured node may not operate properly because not all configurations have been applied. Select the Active radio button then click Submit to apply your configurations.
8.	Repeat **Step 5** but select **HAL HMP Configuration** instead of **Telephony Core**.

Step	Description
9.	Select **View/Edit Boards** in the left pane of the **HALHMP** screen and click **Edit**. 

Board ID	Actions
0	Edit Delete
	Add Board

Step	Description
10.	Select **Board** in the left pane of the **HMP Board** screen and configure as follows: • **IP Address** – Set to the IP Address of the Intervoice MS • **Protocol Name** – Set to **SIP**. • Click **Submit**.  Address: http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/childhmp?boardId=0&cfgName=New%20Config%20Description&c iviadmin Last Login Time: 6/19/07 9:28:25 AM Invalid Logon Attempts: 0 System: All Edit Board - 0 Board ID: 0 IP Address: 192.45.52.201 Protocol Name: SIP

Step	Description
11.	Select **Board VOIP Registration** in the left pane of the **HMP Board** screen and configure as follows: • Click **Add** at the initial **Board VOIP Registration** screen [not shown] • **Board Client Bind** – Set to any ***name@MediaServerIPAddress***. • **Alias String** – Set to the ***UserId@domainname.com***. ***UserId*** should match **User Id** field value configured in **Section 4, Step 6** and ***domainname*** should match the **SIP Domain** field value shown in **Section 4, Step 2**.  Address: http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/childhmp.psml/js_pane/P-1070be33dE1re17a4-3301 iviadmin Last Login Time: 6/19/07 9:28:25 AM Invalid Logon Attempts: 0 System: All Board VOIP Registration - Board 0 Configuration below must be filled out before enabling System VOIP Registration. Board Client Bind: MS@192.45.52.201 Board Alias Configuration Code ID: E-mail Alias String: 54000@devconnect.com

Step	Description
12.	Select **Codec** in the left pane of the **HMP Board** screen and configure as follows: • Click **Add** at the next screen [not shown]. • **Codec Family –** Set to the **Audio Codecs** field value configured in **Section 3.2**. • **Type –** Set to the **Audio Codecs** field value configured in **Section 3.2**. • **Frame Size –** Set the **Packet Size** field value to match the value configured in **Section 3.2**. • Repeat above steps to add additional codecs. • Click **Submit**. 

Address: http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/childhmp.psml/js_pane/P-1070b33dE1re1e7a4-3301

System: All

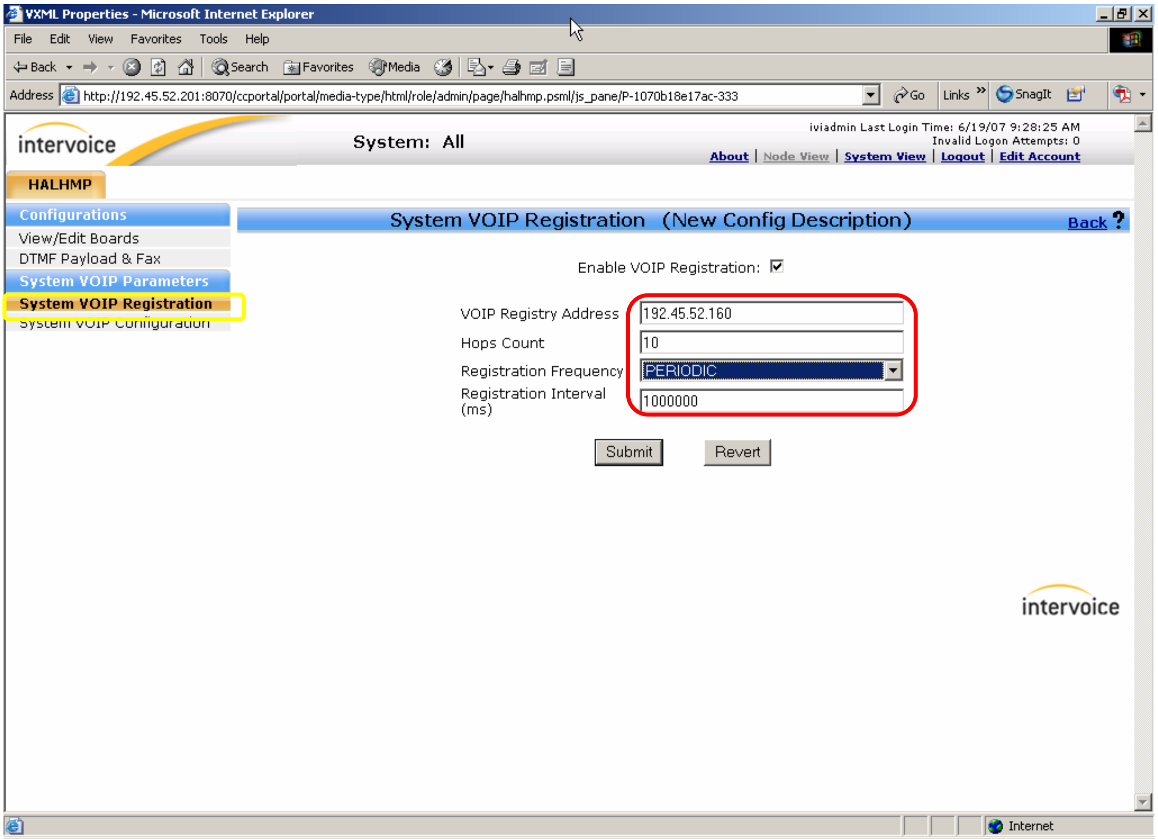
iviadmin Last Login Time: 6/19/07 9:28:25 AM
Invalid Logon Attempts: 0

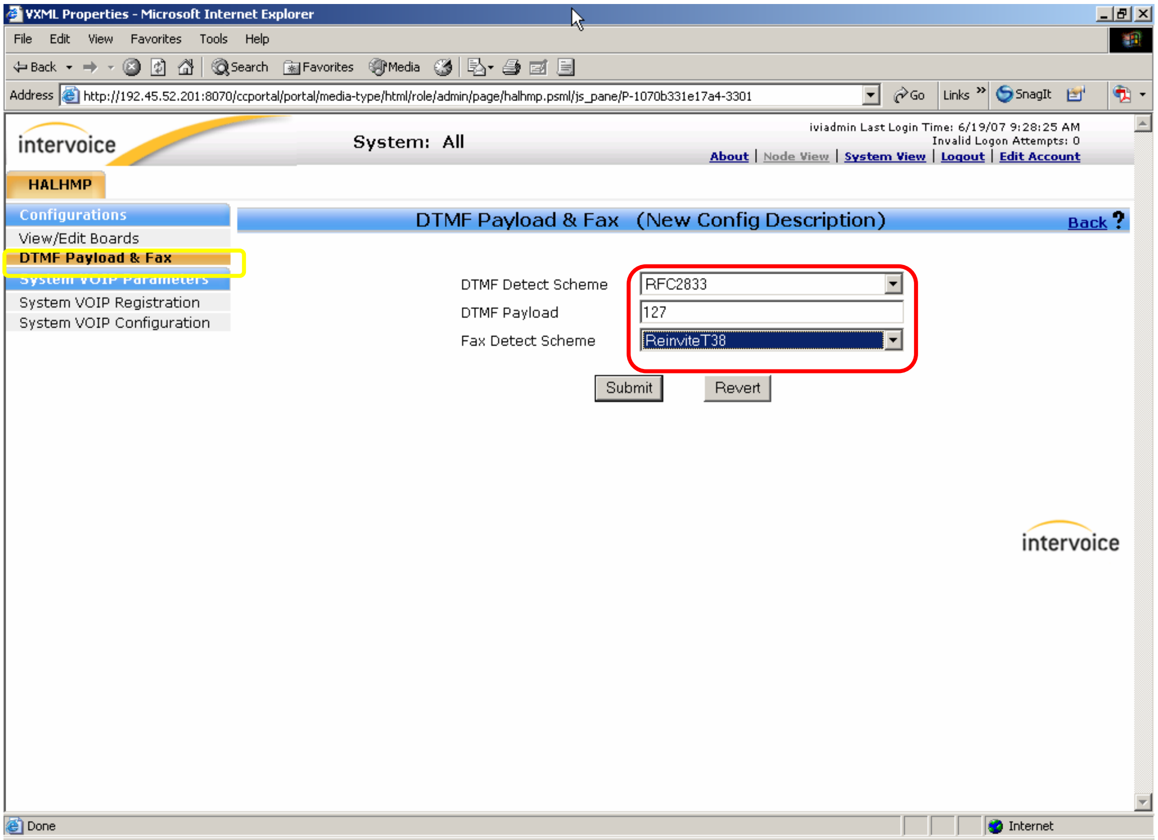
Codec - Board 0

Codec Information

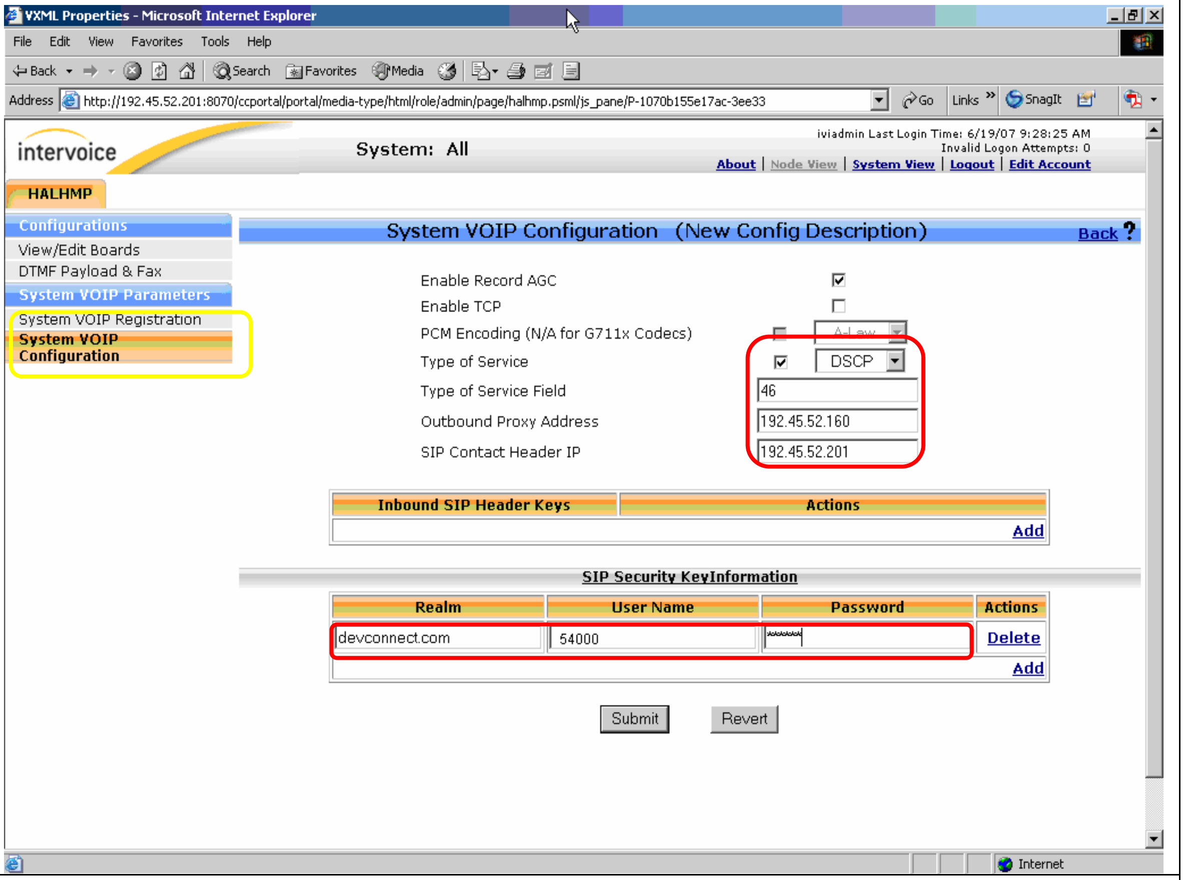
Codec Family	Type	Frame Size	Frames per Packet	Actions
G711Codecs	G711M	20	1	Delete
				Add

G711Codecs
G723Codecs
G729Codecs

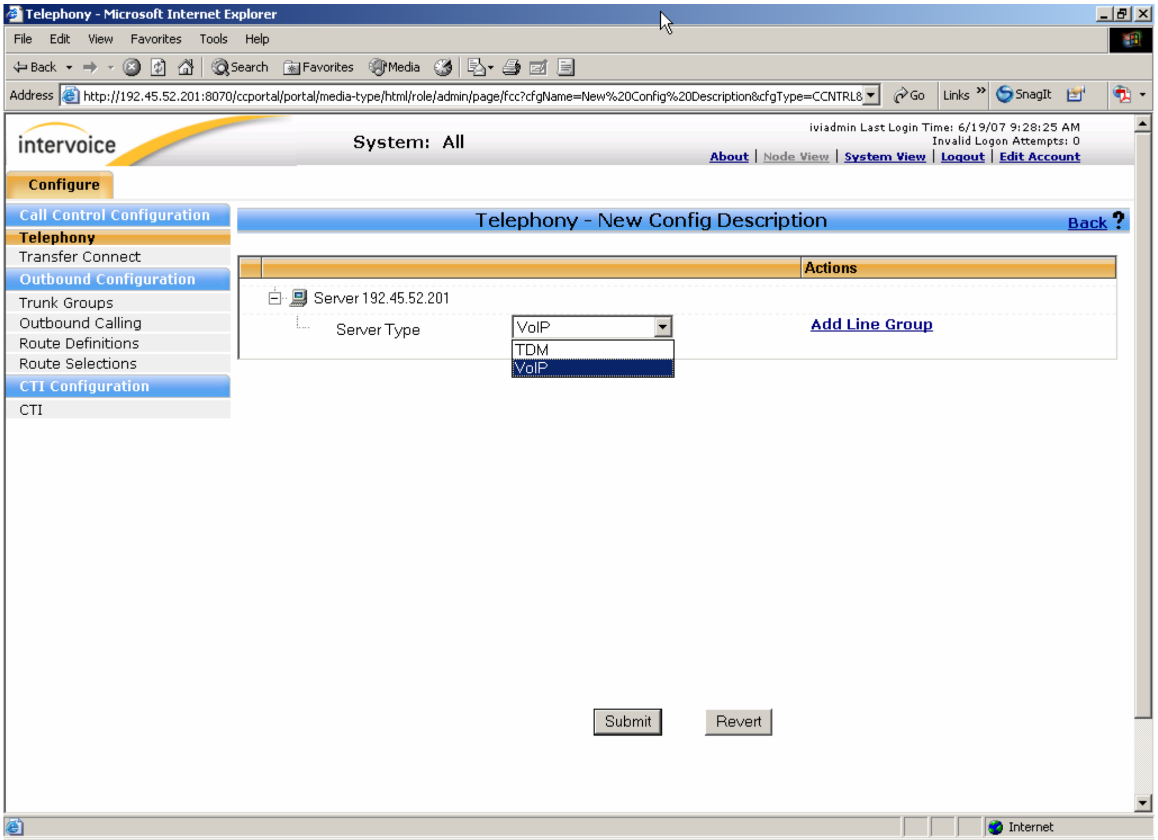
Step	Description
13.	Select **System VOIP Registration** from the left pane of the **HALHMP** screen and configure as follows: - **VOIP Registry Address –** Set to the **Local IP** field value displayed in **Section 4, Step 2**. - **Hops Count –** Set to a value between 1 and 100. - **Registration Frequency –** Set to **PERIODIC**. - **Registration Interval –** Set to a value between 300 and 1000000.  Enable VOIP Registration: ☑ VOIP Registry Address 192.45.52.160 Hops Count 10 Registration Frequency PERIODIC Registration Interval (ms) 1000000

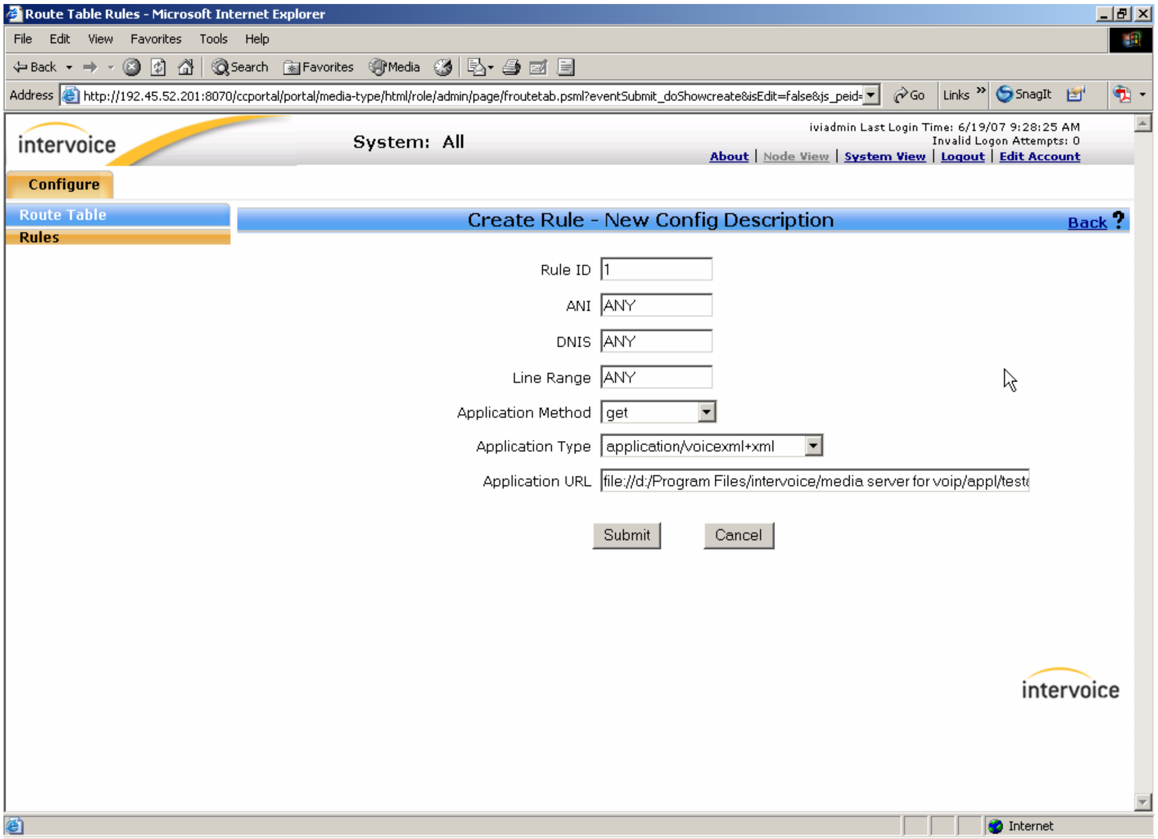
Step	Description
14.	Select **DTMF Payload and Fax** from the left pane of the **HALHMP** screen and configure as follows: - **DTMF Detect Scheme** – Set to match the DTMF scheme used in **Section 3.5**. - **DTMF Payload** – Set to **127**. - **Fax Detect Scheme** – Set to **Reinvite T38**.  VXML Properties - Microsoft Internet Explorer Address http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/halhmp.psml/js_pane/P-1070b331e17a4-3301 iviadmin Last Login Time: 6/19/07 9:28:25 AM Invalid Logon Attempts: 0 System: All DTMF Payload & Fax (New Config Description) DTMF Detect Scheme: RFC2833 DTMF Payload: 127 Fax Detect Scheme: ReinviteT38

Step	Description

15.	Select **System VOIP Configuration** from left pane of the **HALHMP** screen and configure as follows:
	- **Type of Service** – Set to **DSCP**.
	- **Type of Service Field** – Set to a value between 0 and 63.
	- **Outbound Proxy Address** – Set to match the **Local IP** address displayed in **Section 4, Step 2**.
	- **SIP Contact Header IP** – Set to the IP address of the Intervoice MS **Node IP** configured in **Section 5, Step 3**.
	- Click **SIP Security KeyInformation** and configure as follows:
	  o **Realm** – Set to match **SIP Domain** field value in **Section 4, Step 2**.
	  o **User Name** – Set to match **User ID** field value in **Section 4, Step 6**.
	  o **Password** – Set to match the **Password** field value in **Section 4, Step 6**.
	  o Repeat this step for configuring additional ports to be configured.
	- Click **Submit**.
	
	Address: http://192.45.52.201:8070/ccportal/portal/media-type/html/role/admin/page/halhmp.psml/js_pane/P-1070b155e17ac-3ee33
	System: All
	iviadmin Last Login Time: 6/19/07 9:28:25 AM
	Invalid Logon Attempts: 0
	System VOIP Configuration (New Config Description)
	Enable Record AGC ☑
	Enable TCP ☐
	PCM Encoding (N/A for G711x Codecs) ☐ A-Law
	Type of Service ☑ DSCP
	Type of Service Field 46
	Outbound Proxy Address 192.45.52.160
	SIP Contact Header IP 192.45.52.201
	Inbound SIP Header Keys / Actions: Add
	SIP Security KeyInformation
	Realm / User Name / Password / Actions
	devconnect.com / 54000 / ***** / Delete
	Add

16.	Repeat **Step 7** and but select **HAL HMP Configuration**. Repeat **Step 5** but select **Call Control Configuration**.

Step	Description
17.	Select **Telephony** from left pane of the **Call Control Configuration** screen and configure as follows: • Click **Delete** at the next screen. [Not Shown] • Select **VoIP** for the **Server Type** field and click **Submit**.  Telephony - New Config Description Server 192.45.52.201 Server Type VoIP
18.	Repeat **Step 8** and but select **Call Control Configuration**. Repeat **Step 5** but select **Application Routing**.

Step	Description
19.	Select **Rules** from left pane of the **Route Table** screen and configure as follows: • Click **Create Rule** on the next screen [Not Shown]. • **Application Type** – Set to the type of Application to be used. Set to **application/voicexml+xml**, a VXML application in this example. • **Application URL** – Physical location of the VXML application on the Intervoice MS. • Click **Submit**.  Rule ID 1 ANI ANY DNIS ANY Line Range ANY Application Method get Application Type application/voicexml+xml Application URL file://d:/Program Files/intervoice/media server for voip/appl/testa
20.	Repeat **Step 7** but select **Application Routing**.

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily to exercise Intervoice MS IVR solution using DTMF. Tests were done to verify that Intervoice MS is able to recognize DTMF commands and take appropriate action for calls originating from SIP, H.323, digital, analog and PSTN phones using Avaya SES and Avaya Communication Manager.

6.1. General Test Approach

The general test approach was to place calls from any phone to establish a call into Intervoice MS and exercise the supported features. The main objectives were to verify that:

- The Intervoice MS successfully registers with Avaya SES.
- The Intervoice MS successfully initiates and terminates calls to IP and non-IP telephones.
- The Intervoice MS successfully executes a blind transfer.
- The Intervoice MS successfully shuffles for VoIP calls.
- The Intervoice MS successfully handles DTMF.
- The Intervoice MS successfully handles T.38 fax.
- The Intervoice MS successfully handles Signaling and RTP traffic on separate Ethernet cards.
- The Intervoice MS successfully prioritizes traffic.

For serviceability testing, failures such as cable pulls and hardware resets were applied.

6.2. Test Results

The test objectives of Section 6.1 were verified. For serviceability testing, the Intervoice MS operated properly after recovering from failures such as cable disconnects, and resets of the Intervoice MS, the Avaya SES server, and Avaya Communication Manager. Calls placed to the Intervoice MS were successfully shuffled.

The following observations were made during testing:

- Intervoice MS does not support de-registration but it re-registers automatically with Avaya SES once the service is re-started. If the Intervoice MS is manually shut down, then it will de-register by sending **REGISTER 0**.
- Intervoice MS operates only with UDP as the SIP transport protocol.
- Intervoice MS supports only Layer-3 QOS parameters.
- When calls were made using H.323 IP telephones only the following scenarios were supported:
  - In-band DTMF with any codec
  - Rtp-payload DTMF with G729.

Intervoice Inc. expects to resolve the above observations in future releases.

7. Verification Steps

Step	Description
1.	Verify all members for the SIP trunk group provisioned in **Section 3.6** are **in-service/idle**. From a SAT session: • Issue the command "**status trunk 10**". • Verify that all members in Trunk Group 10 are **in-service/idle**.
2.	Verify that the Intervoice MS successfully register with the Avaya SES server by following the **Users -> Registered Users** links on the Avaya SES Administration Web Interface.
3.	Place calls into the Intervoice MS and verify that it responds as expected per the VXML application loaded on the server.

8. Support

For technical support on Intervoice Inc., consult the support pages at http://www.intervoice.com/support or contact Intervoice Inc. technical support at:

- Phone: 1-972-484-1000
- E-mail: support@intervoice.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0, Avaya SIP Enablement Services 3.1.2, and Intervoice MediaServer 3.5. The Intervoice MS is a SIP based VoIP software which provides an IVR driven-menu for executing Voice Portal VXML based applications. For the purpose of compliance testing, several demo VXML IVR applications were provided by Intervoice Inc. to exercise SIP call flows between Intervoice MS ports registered with Avaya SES and SIP and non-SIP telephones. The compliance testing was successful with the exception of the issues noted in **Section 6.2**.

10. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 3.1, February 2007, Document Number 03-300509

[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 12, February 2007, Document Number 555-233-504

[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 7, May 2007, Document Number 555-245-206

[4] *Installing and Administering SIP Enablement Services R3.1.2*, Issue 4, May 2007, Document Number 03-600768

Product documentation for Intervoice Inc. products may be found at http://www.intervoice.com.

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.