Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Global IP Solutions Remote Extension with Avaya Communication Manager and Avaya SIP Enablement Services - Issue 1.0

Abstract

These Application Notes describe the steps for configuring Global IP Solutions Remote Extension to communicate with Avaya Communication Manager and Avaya SIP Enablement Services. Global IP Solutions Remote Extension consists of the Remote Extension softphone client and the Global IP Solutions Integrated System Services/Internet Voice Transcoder (ISS/IVX) Server. The Remote Extension softphone client communicates with the Global IP Solutions ISS/IVX Server via a Global IP Solutions propriety protocol; the Global IP Solutions ISS/IVX Server then registers the Remote Extension client with Avaya SIP Enablement Services. Emphasis of the testing was placed on verifying good voice quality from Global IP Solutions Remote Extension and its ability to interoperate with Avaya SIP Enablement Services.

Information in these Application Notes has been obtained through 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 for configuring Global IP Solutions Remote Extension to operate with Avaya Communication Server and Avaya SIP Enablement Services (SES). Global IP Solutions Remote Extension consists of two parts, a softphone client that runs on Microsoft Windows and the Global IP Solutions ISS/IVX Server that resides in the core network. Global IP Solutions Remote Extension communicates with the Global IP Solutions ISS/IVX Server via a propriety protocol. In turn, the Global IP Solutions ISS/IVX Server registers softphone clients to Avaya SES via the standard SIP protocol.

Global IP Solutions Remote Extension clients communicate with the Global IP Solutions ISS/IVX Server via a common IP address through the Internet. The extension numbers used by Global IP Solutions Remote Extensions are registered to Avaya SES and are also administered as Off-PBX-Telephone stations in Avaya Communication Manager. As a result, each Global IP Solutions Remote Extension softphone has access to features available from Avaya Communication Manager.

These Application Notes assume that Avaya Communication Manager and Avaya SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test effort will be described in this document.

Figure 1 illustrates the configuration used in these Application Notes. The Global IP Solutions ISS/IVX Server has two options for Ethernet connections. The first option uses two Ethernet interfaces; one interface is connected into the private core network, and the other interface is connected to the Internet. The second option utilizes a private Ethernet port. A NAT device was used for access from the public network. During the compliance test, the second option is utilized.

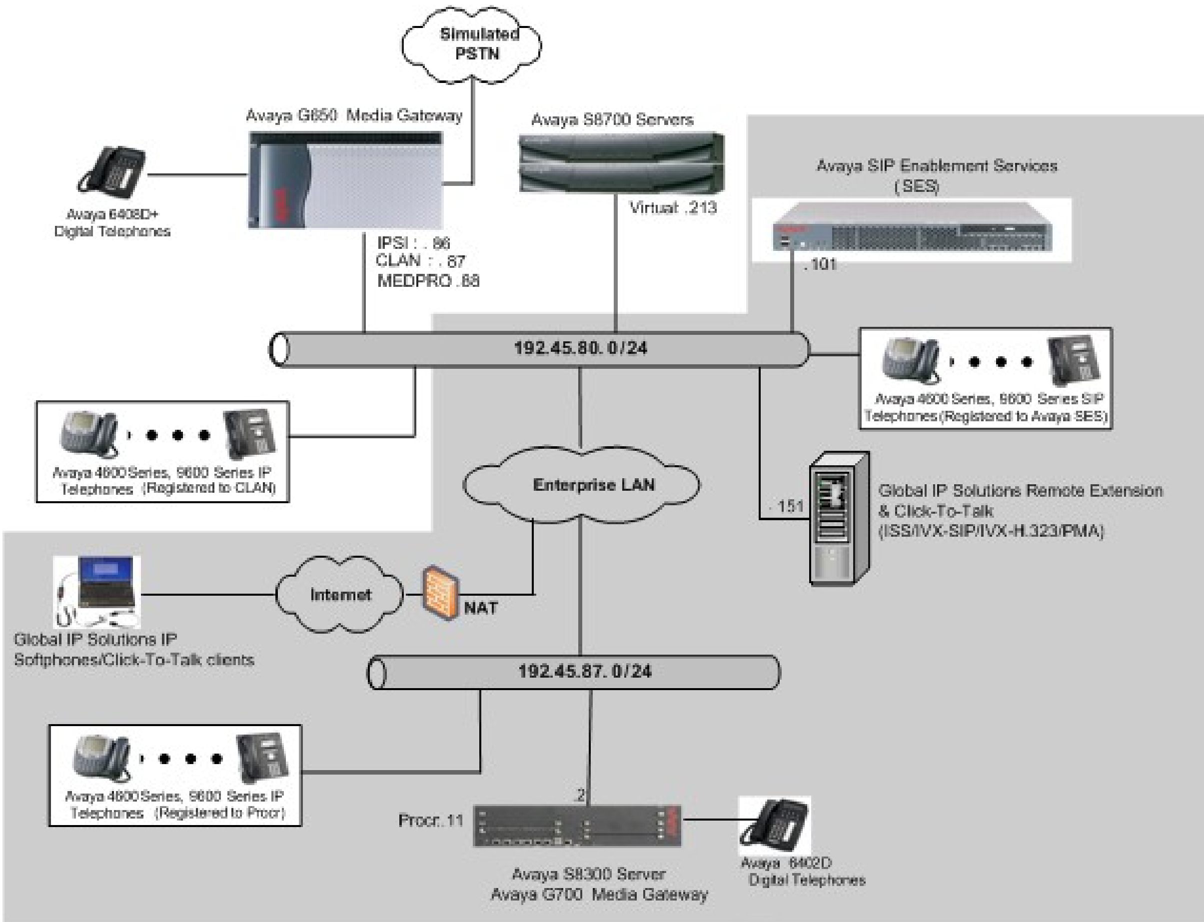


Figure 1: Sample Network Configuration

2. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya S8700 Servers		Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G650 Media Gateway		
	TN2312BP IP Server Interface TN799DP CLAN Interface TN2302AP IP Media Processor	HW11 FW030 HW01 FW017 HW20 FW108
Avaya S8300 Server		Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G700 Media Gateway		25.28.0
Avaya SIP Enablement Services		4.0 (Build 33.6)
Avaya 4600 Series IP Telephones		
	4620 (H.323) 4625 (H.323)	2.8 2.8
Avaya 9600 Series IP Telephones		
	9630 (H.323) 9650 (H.323)	1.5 1.5
Avaya SIP Telephones		-
	4610 (SIP) 9630 (SIP)	2.2.2 1.0.13
Global IP Solutions Integrated System Services (ISS)		5.0.1
Global IP Solutions Internet Voice Transcoder for SIP (IVX)		5.0.1
Global IP Solutions Remote Extension Client on Microsoft Windows XP Professional Version 2002 with Service Pack 2		5.0.1

3. Configure Avaya Communication Manager

This section describes the procedure for setting up a SIP trunk between Avaya Communication Manager and Avaya SES. The steps include setting up an IP codec set, an IP network region, an IP node name, a signaling group, a trunk group, a SIP station, a coverage path, and a hunt group. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. Global IP Solutions Remote Extension Clients and other SIP telephones are configured as off-PBX telephones in Avaya Communication Manager.

3.1. Capacity Verification

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  11
                                OPTIONAL FEATURES

     G3 Version: V14
       Location: 1                             RFA System ID (SID): 1
       Platform: 7                             RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   95
                                      Maximum Stations: 450   17
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    0
                    Maximum Off-PBX Telephones -   OPS: 100   9
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

On **Page 2** of the system-parameters customer-options form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   18
           Maximum Concurrently Registered IP Stations: 50    3
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 5     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 100   50

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
```

3.2. Configure 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. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. An IP codec set is used in **Section 3.3** when configuring the IP network region to specify which audio codec may be used within and between network regions. For the compliance testing, G.711MU and G.729AB were used.

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

                          IP Codec Set

    Codec Set: 1

    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. Configure 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. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – This should match the SIP Domain value on Avaya SES, in **Section 4.1**. In the test configuration, **testroom.com** was used.
- 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 – Enter the IP 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 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: testroom.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
```

3.4. Configure IP Node Name

This section describes the steps for setting the IP node name for Avaya SES in Avaya Communication Manager. 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   2
                                  IP NODE NAMES
    Name                 IP Address
CLAN                   192.45.80.87
IA770                  192.45.87.12
S8300CDR               192.45.88.11
S8300G250              192.45.82.11
S8300G350              192.45.81.11
default                0.0.0.0
procr                  192.45.87.11
SIPServer              192.45.80.101
```

3.5. Configure 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. Enter the **add signaling-group** <**s**> command, where **s** is an available signaling group and configure the following:

- Group Type – Set to **sip.**
- Near-end Node Name - Set to **procr** as displayed in **Section 3.4**.
- Far-end Node Name - Set to the Avaya SES name configured in **Section 3.4**.
- Far-end Network Region - Set to the region number configured in **Section 3.3**.
- Far-end Domain - This should match the SIP Domain value in **Section 4.1**. In the test configuration, **testroom.com** was used.

```
add signaling-group 1                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: tls




   Near-end Node Name: procr               Far-end Node Name: SIPServer
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 1
       Far-end Domain: testroom.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
```

3.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. Enter the **add trunk-group <t>** command, 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 a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code that is valid in the provisioned dial plan.
- Signaling Group – Set to the Group Number field value configured in **Section 3.5**.
- 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.

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 1                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: to SES                     COR: 1        TN: 1        TAC: 115
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 1
                                                    Number of Members: 50
```

3.7. Configure SIP Endpoint

This section describes the steps for administering Off PBX Stations (OPS) stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of Global IP Solutions Remote Extension clients. For each Global IP Solutions Remote Extension client to be supported, 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 **6408D+**.
- Port – Set to **X**.
- Coverage Path 1 – Set to **99**. This feature is needed to test the MWI feature.
- Name – Enter a descriptive name.

```
add station 27001                                               Page   1 of   5
                                     STATION

Extension: 27001                         Lock Messages? n               BCC: 0
     Type: 6408D+                        Security Code:                  TN: 1
     Port: X                           Coverage Path 1: 99              COR: 1
     Name: SIP 27001                   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: 27001
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english

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

On Page 2, set the MWI Served User Type field to **qsig-mwi**.

```
add station 27001                                                Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe               Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: none
            CDR Privacy? n                                 Data Restriction? n
   Redirect Notification? y                      Idle Appearance Preference? n
 Per Button Ring Control? n                    Bridged Idle Line Preference? n
   Bridged Call Alerting? n                        Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: qsig-mwi               Display Client Redirection? n
                                                Select Last Used Appearance? n
                                                  Coverage After Forwarding? s

                                            Direct IP-IP Audio Connections? y
  Emergency Location Ext: 27001                        IP Audio Hairpinning? n
```

Enter the **add off-pbx-telephone station-mapping** command and configure the following for each of the extension administered above:

- Station Extension – Enter the extension configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that Global IP Solutions Remote Extension clients will use for registration on Avaya SES. 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 3.6**.
- Config Set – Set to 1, which contains the default values.

```
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
 27001              OPS          -       27001               1            1
```

3.8. Configure the Coverage Path

Enter **add coverage path c**, where **c** is an unused coverage path number. In the compliance test, the Point1 field under the COVERAGE POINTS section is set to **h99** (i.e. the hunt group configured in **Section 3.9**).

```
add coverage path 99                                            Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 99
                                                  Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?            n               n
               Busy?            y               y
       Don't Answer?            y               y        Number of Rings: 2
                All?            n               n
 DND/SAC/Goto Cover?            y               y
   Holiday Coverage?            n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h99            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

3.9. Configure Hunt Group

Enter **add hunt-group h**, where **h** is an unused hunt group number. The following fields were configured for the compliance test:

- Group Name – Provide a descriptive name of the group
- Group Extension – Provide the hunt group extension

```
add hunt-group 99                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: Audix                           Queue? n
         Group Extension: 70000                          Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1           Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:
```

On **Page 2**, the following fields were configured for the compliance test.

- Message Center – Set to **qsig-mwi**.
- Voice Mail Number – Set to **70000**.
- Routing Digits (e.g. AAR/ARS Access Code) - Enter the AAR Access Code. In the test configuration, **8** was used.

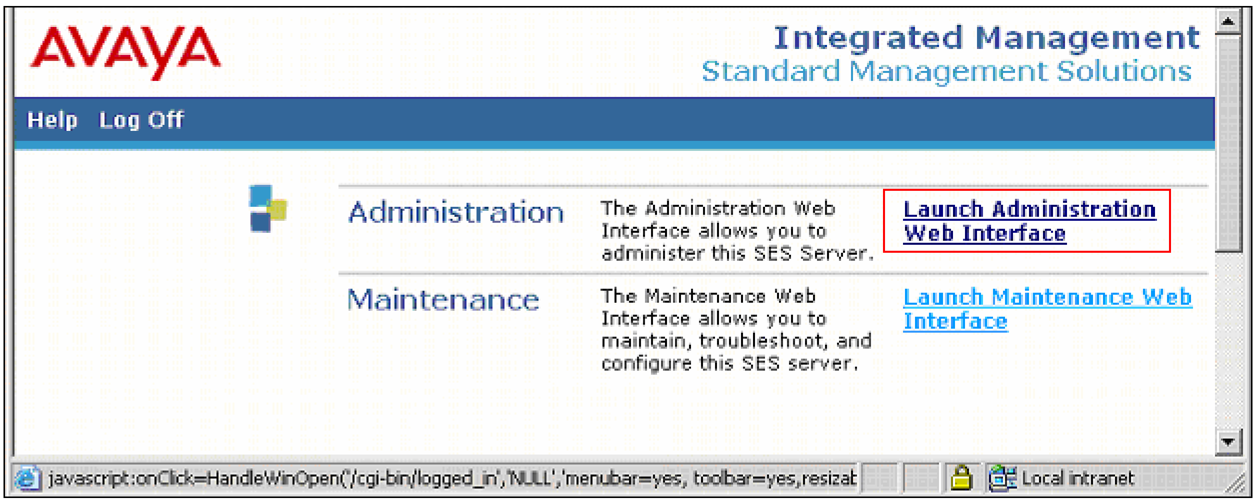
```
add hunt-group 99                                               Page   2 of  60
                                  HUNT GROUP

                       LWC Reception: none           AUDIX Name:



                       Message Center: qsig-mwi
                      Send Reroute Request: y
                     Voice Mail Number: 70000
Routing Digits (e.g. AAR/ARS Access Code): 8           Provide Ringback? n
                 TSC per MWI Interrogation? n
```

4. Configure Avaya SES

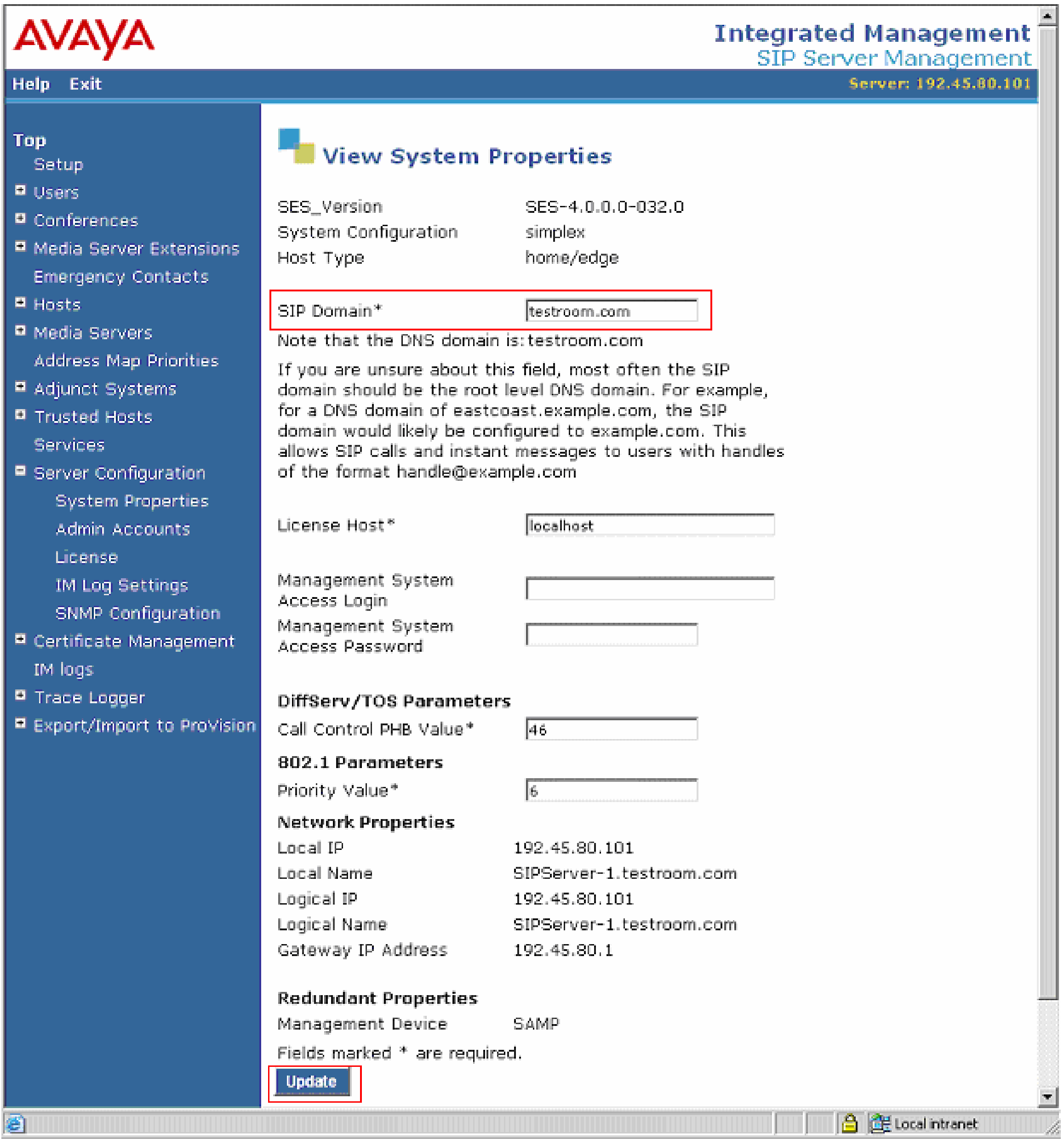
This section describes the steps for creating a SIP trunk between Avaya SES and Avaya Communication Manager. SIP user accounts are configured in Avaya SES and associated with an Avaya Communication Manager OPS station extension. Global IP Solutions Remote Extension Clients will register with Avaya SES 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.

4.1. Configure SES Server 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 Administration Web Interface** link upon successful login.



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 Avaya Communication Manager in **Section 3.5**. Click on the **Update** button if a field change was necessary.

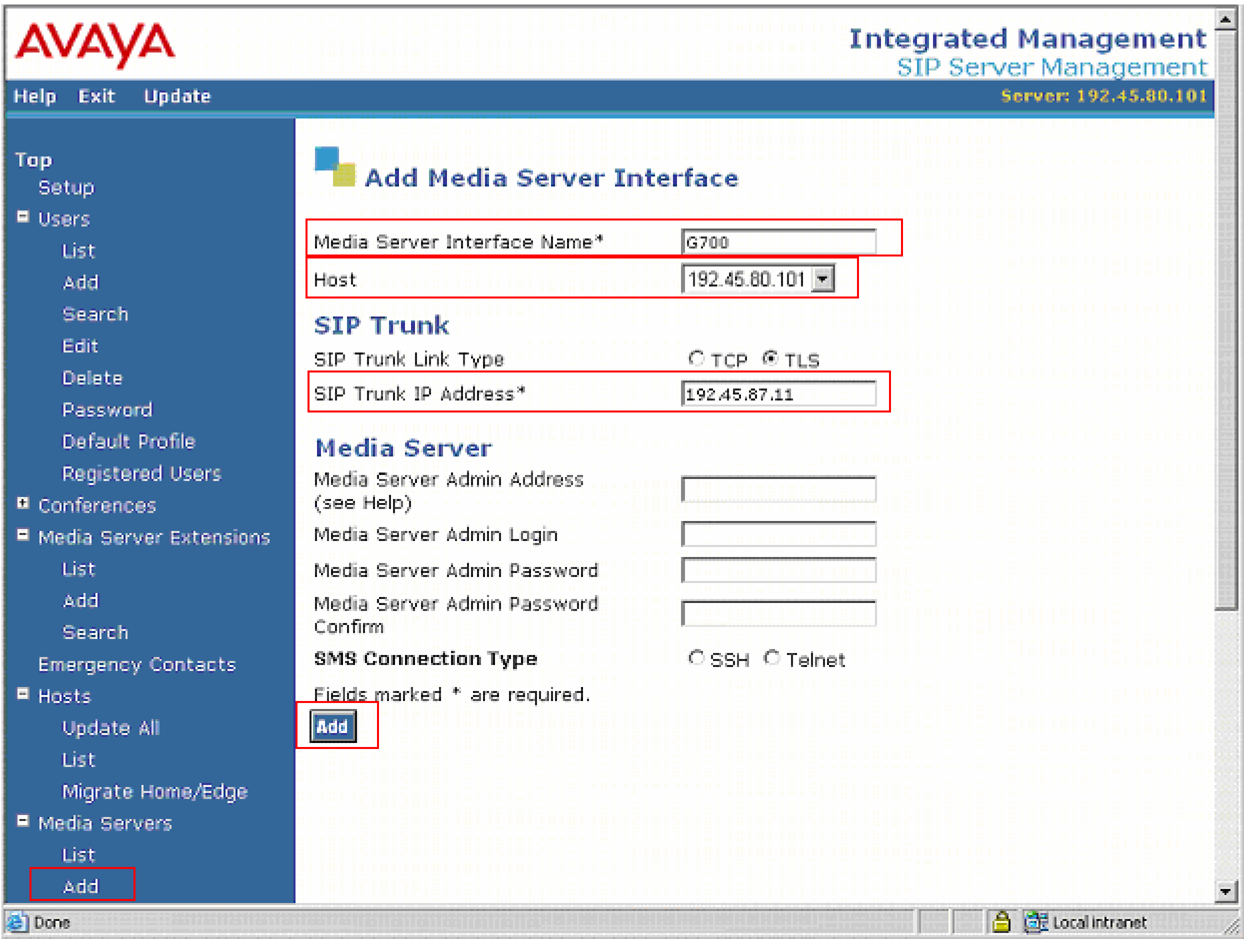


4.2. Configure Media Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the **Integrated Management SIP Server Management** page, select the **Media 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:

- Media Server Interface Name – Enter a descriptive name for the media server interface.
- Host – From the alphabetized drop-down list of names, select the name of the Avaya SES server to be associated with the Media 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 Avaya SES.

Click **Add** when finished.



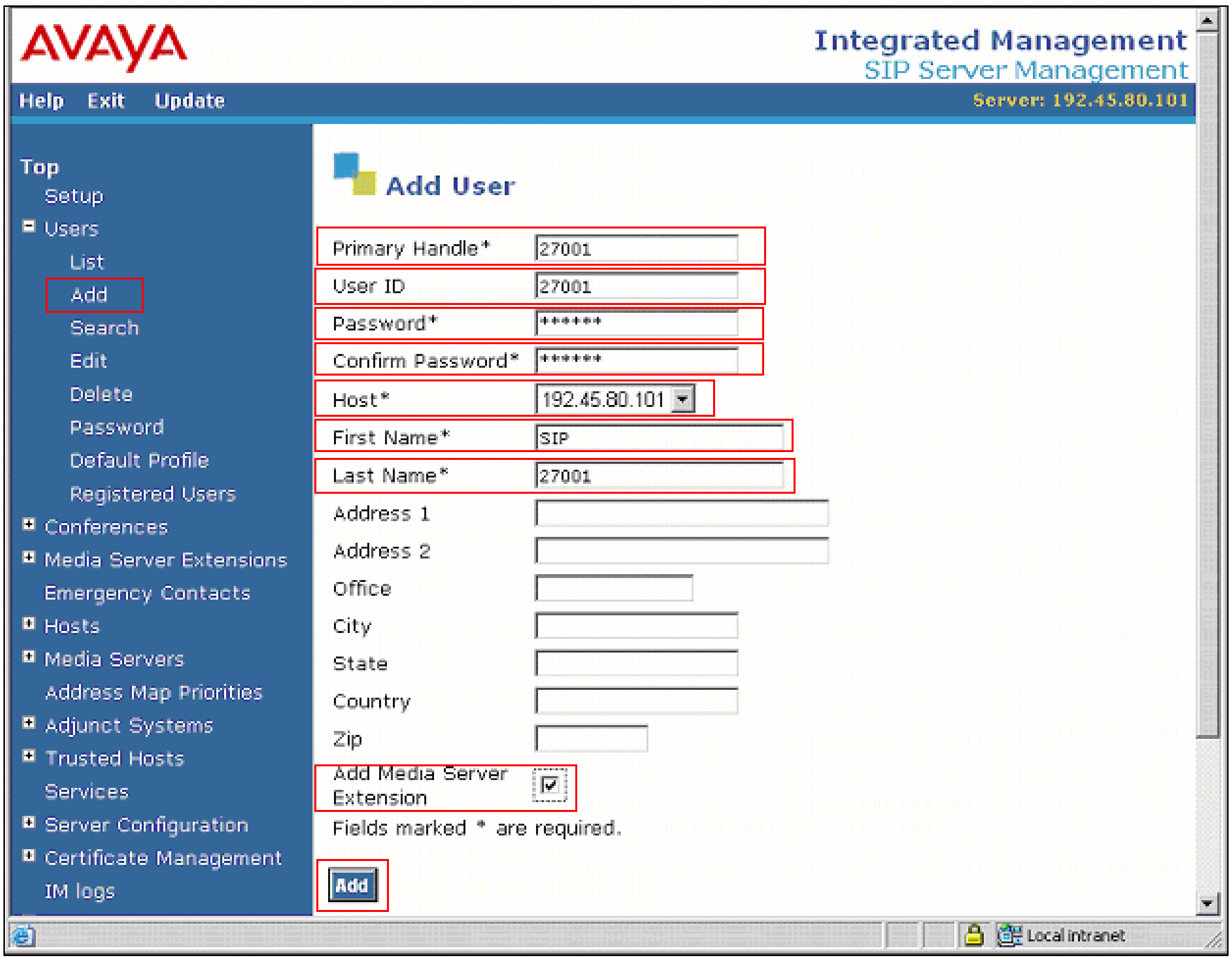
4.3. Configure Users

This section provides steps to add users to be administered in the Avaya 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

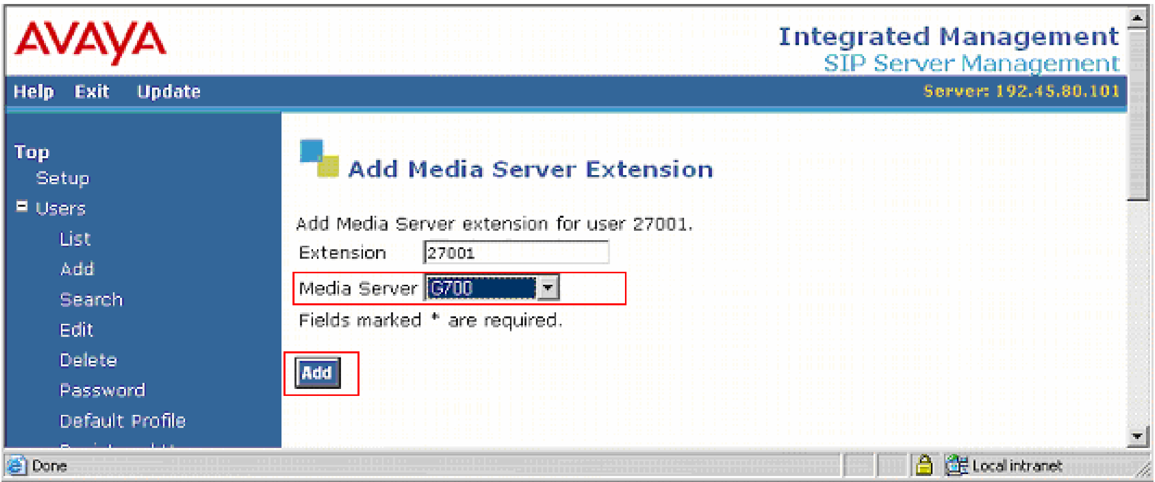
- Primary Handle – Enter the phone number of Global IP Solutions Remote Extension Client. This number was configured in **Section 3.7**.
- User ID – Set to any descriptive name.
- Password / Confirm Password – Enter a password of at least 6 alphanumeric characters; both field entries must match exactly.

- Host – From the drop-down list of names, select the host serving the domain for this user. The host name of the current server is selected by default.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Add Media Server Extension - Select this field to associate a new extension number with this user in the database. The Add Media Server Extension screen will be displayed next, after this user profile has been added.

Click **Add** when finished.



From the next screen, enter the numeric telephone extension you want to create in the database. This should match the phone number entry on the off-pbx-telephone station-mapping form in **Section 3.7**. Select the extension's media server from the drop-down list. Click on the **Add** button.

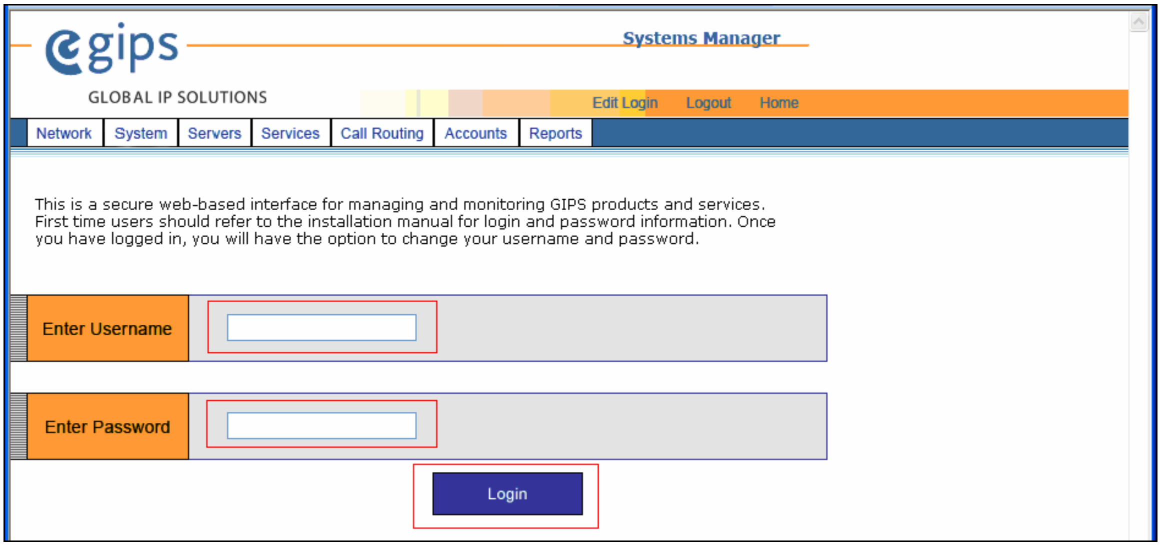


5. Global IP Solutions Remote Extension

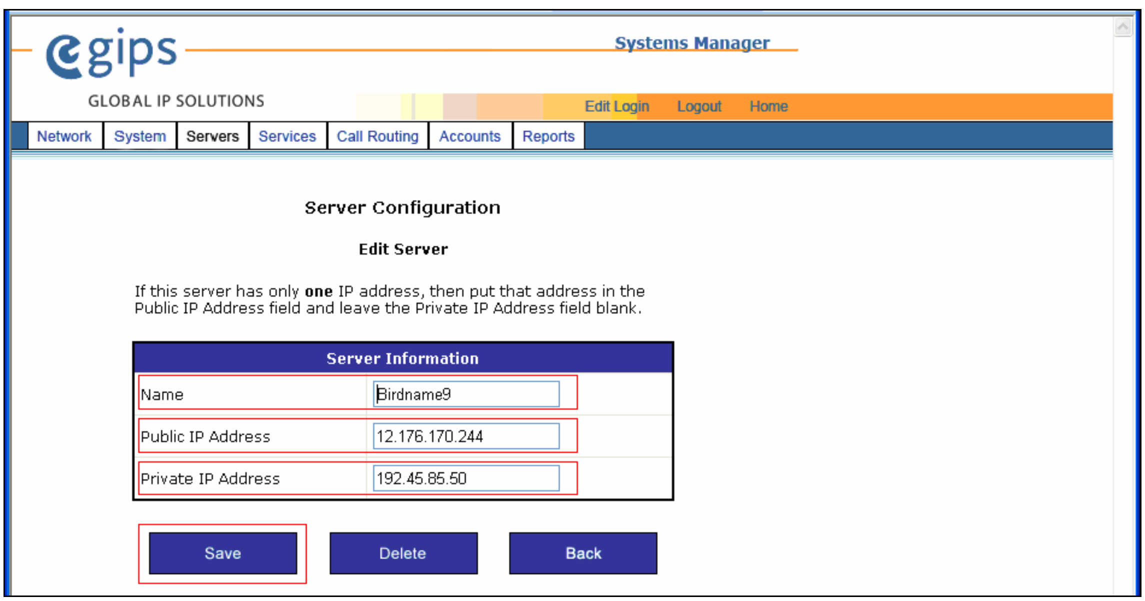
This section describes the configuration for the ISS/IVX Server to communicate with Avaya SES. This configuration allows for calls to and from Remote Extension clients. From the perspective of the Avaya SIP Enablement Services Server, Global IP Solutions Remote Extension clients look like actual SIP telephones, registered from the IP address of the ISS/IVX Server.

5.1. Configuring the ISS/IVX Server

Launch a web browser, enter **http://<IP address of Server >:8080/systemsmanager/home.asp** in the URL, and log in with the appropriate credentials for accessing the Systems Manager page.

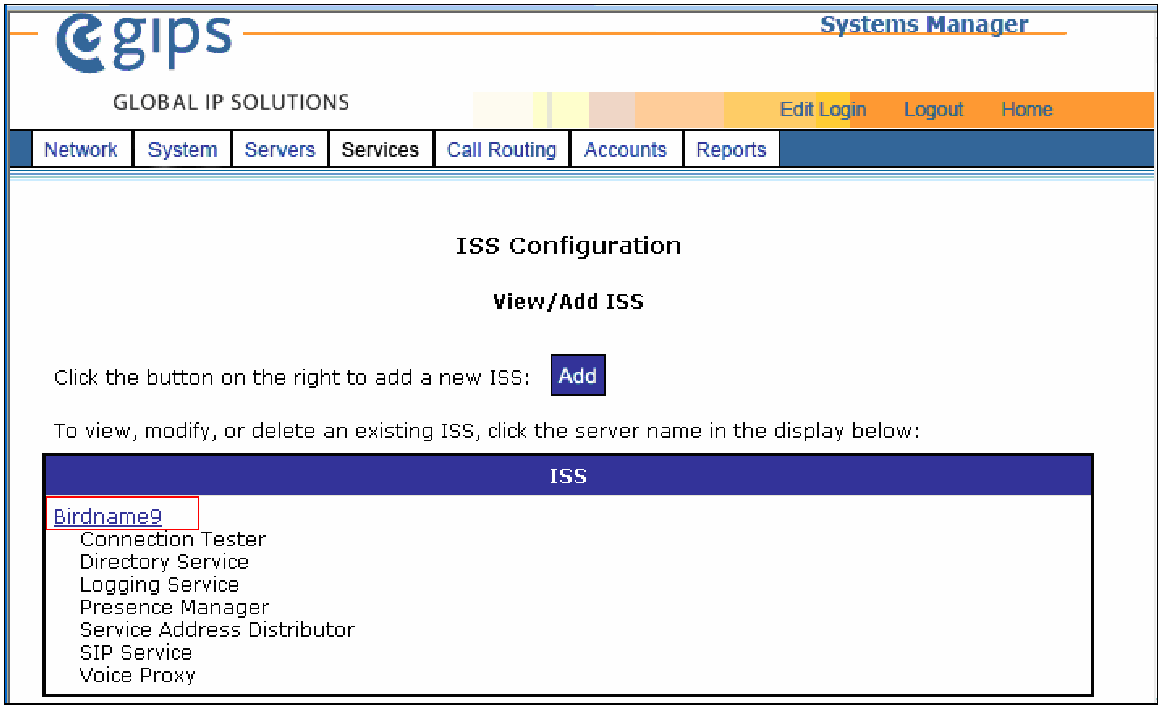


Navigate to **Servers** on the main menu bar and select the **Add** button (not shown). For a single Ethernet connection scenario, the IP address must be entered into the Public IP Address field. For the Private IP address field, the Global IP Solutions software is designed to recognize private IP addresses as defined in RFC 1918, *Address Allocation for Private Internets*. As a softphone connects to the server, the public IP address is NATed to the private IP address.

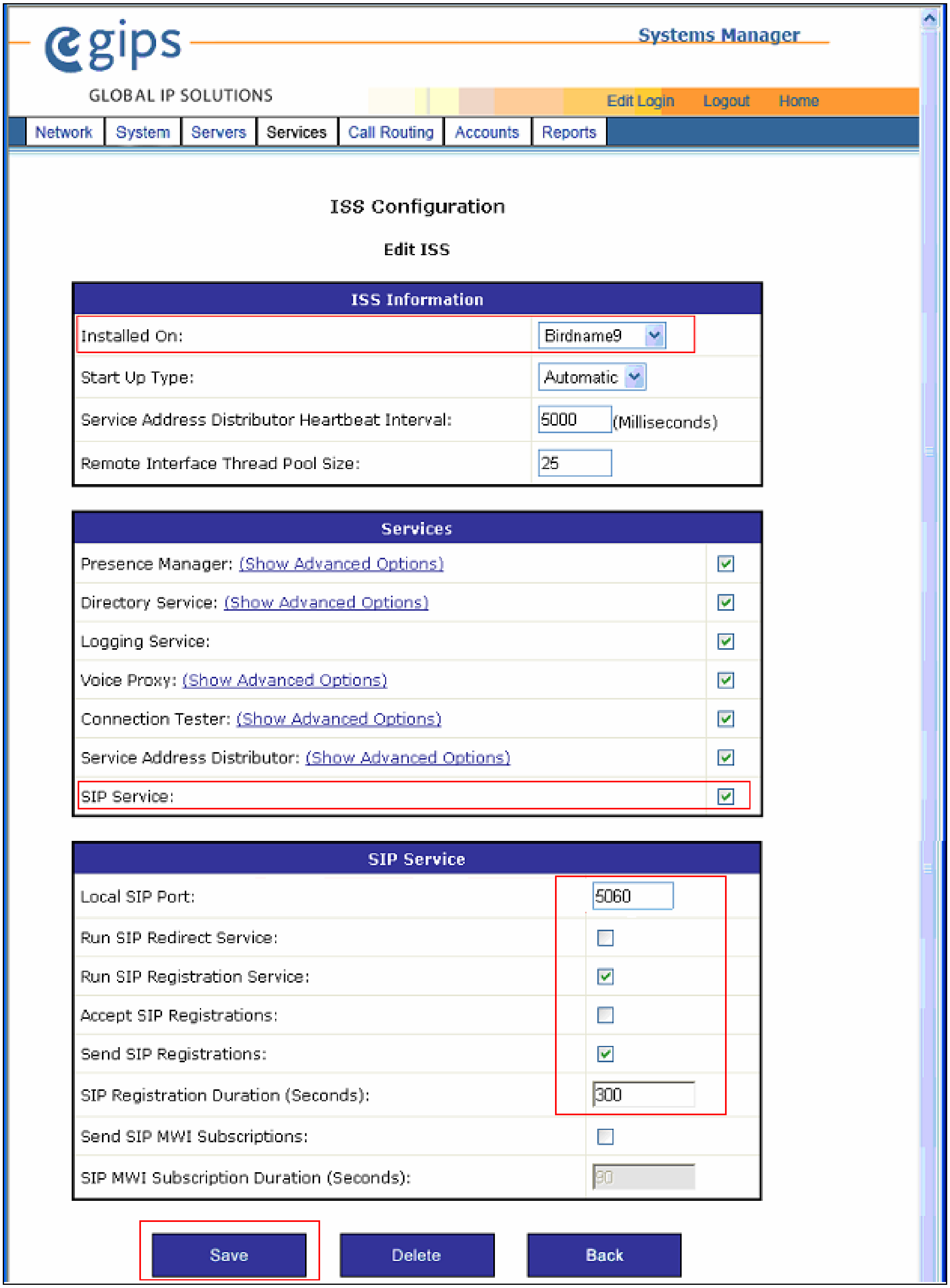


5.2. Configuring the ISS Service

Navigate to **Services→ ISS** from main menu. Click on the server created in **Section 5.1**. The sample configuration used **Birdname9**.

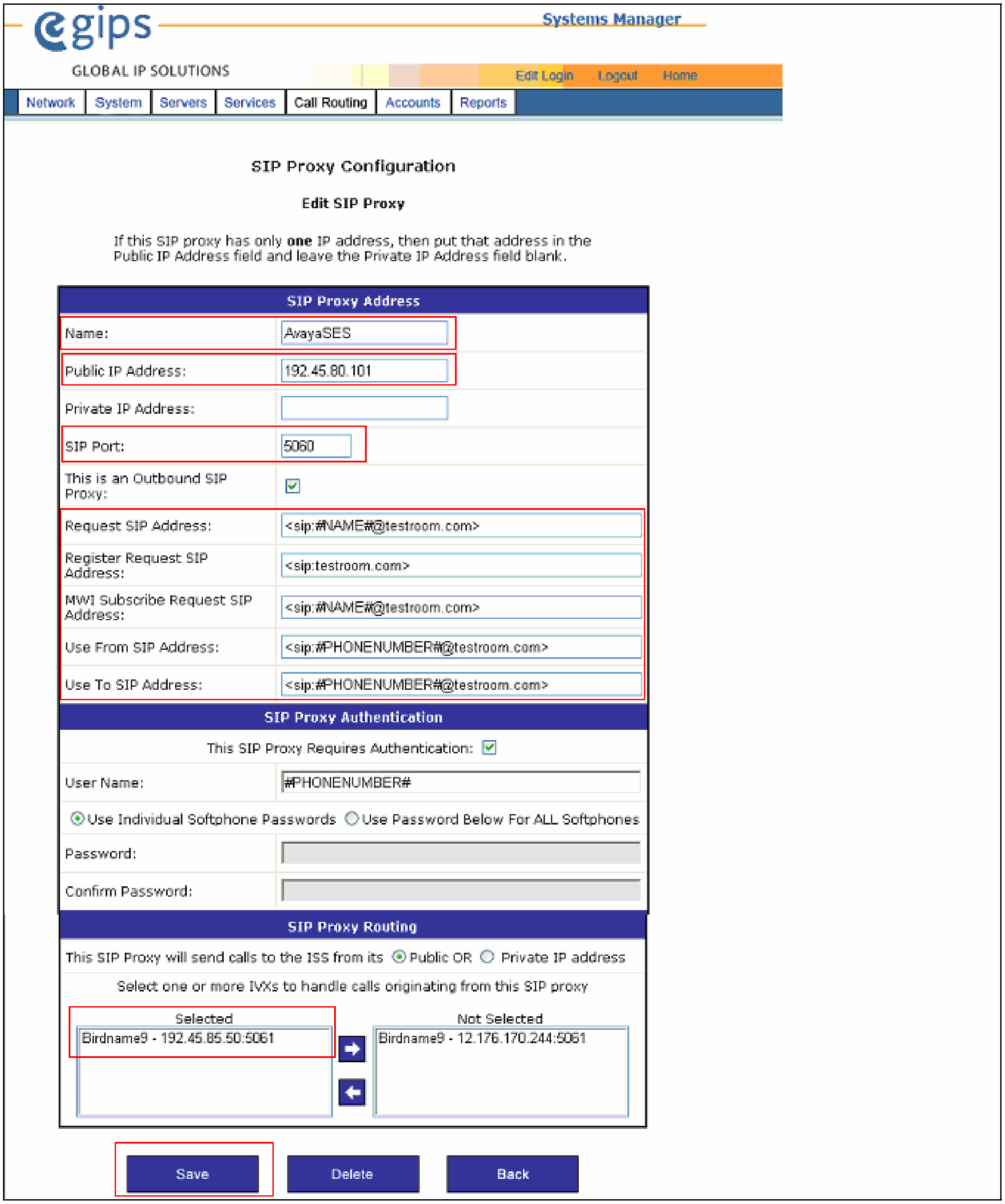


Make sure the highlighted fields are set as shown. Click **Save** to complete.


gips
GLOBAL IP SOLUTIONS
Systems Manager
Edit Login
Logout
Home
Network
System
Servers
Services
Call Routing
Accounts
Reports
ISS Configuration
Edit ISS
ISS Information
Installed On:
Birdname9
Start Up Type:
Automatic
Service Address Distributor Heartbeat Interval:
5000
(Milliseconds)
Remote Interface Thread Pool Size:
25
Services
Presence Manager: (Show Advanced Options)
Directory Service: (Show Advanced Options)
Logging Service:
Voice Proxy: (Show Advanced Options)
Connection Tester: (Show Advanced Options)
Service Address Distributor: (Show Advanced Options)
SIP Service:
SIP Service
Local SIP Port:
5060
Run SIP Redirect Service:
Run SIP Registration Service:
Accept SIP Registrations:
Send SIP Registrations:
SIP Registration Duration (Seconds):
300
Send SIP MWI Subscriptions:
SIP MWI Subscription Duration (Seconds):
90
Save
Delete
Back

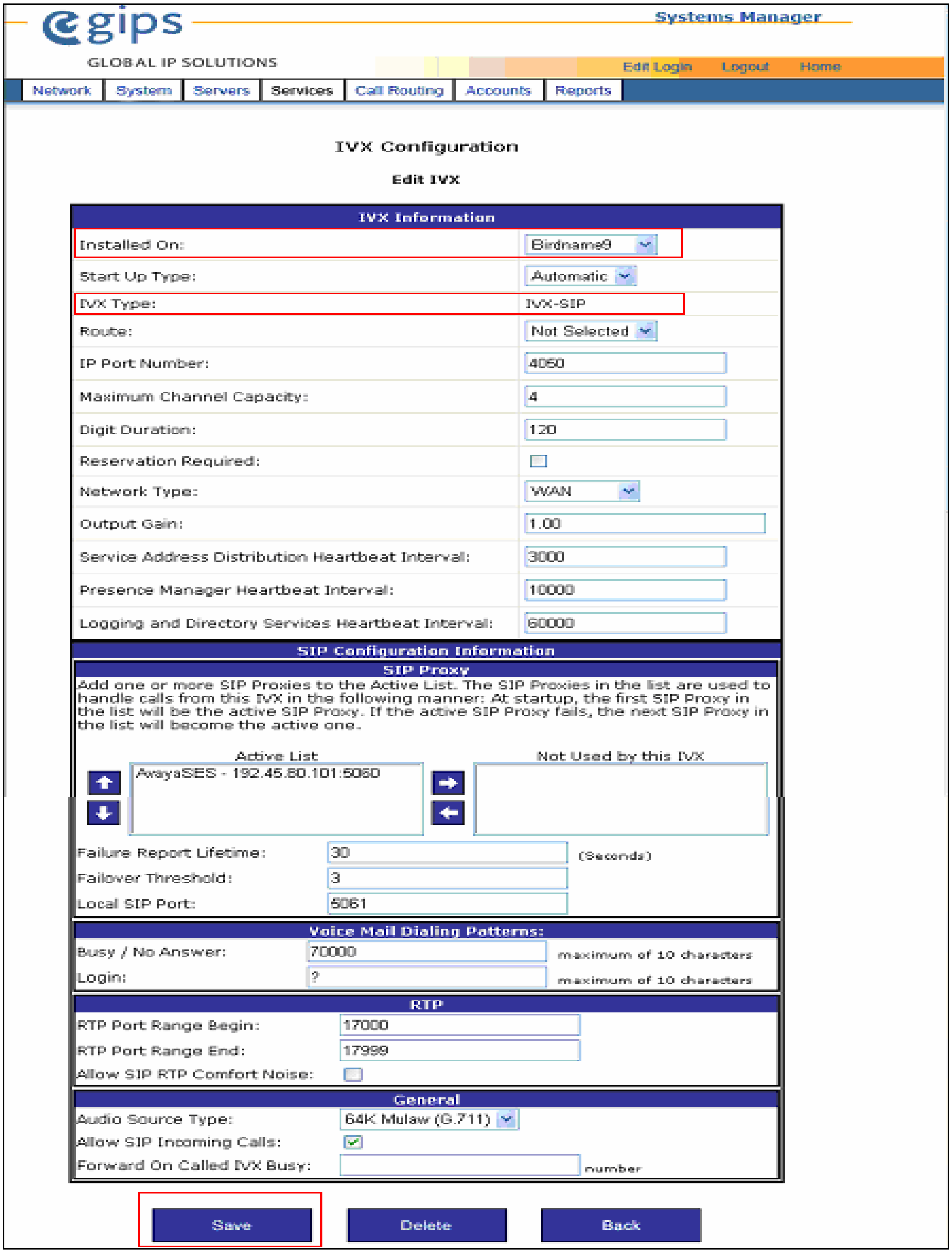
5.3. Configuring the SIP Proxies

Navigate to **Call Routing → SIP Proxies** from the main menu. The following shows the SIP Proxy Configuration page. Enter highlighted fields as shown. The domain name, **testroom.com**, was used in the sample configuration. An appropriate domain name should be entered in place of testroom.com**.** In the case of a single Ethernet connection, enter the IP address in the **Public IP Address** field. Scroll down and click **Save** to complete.



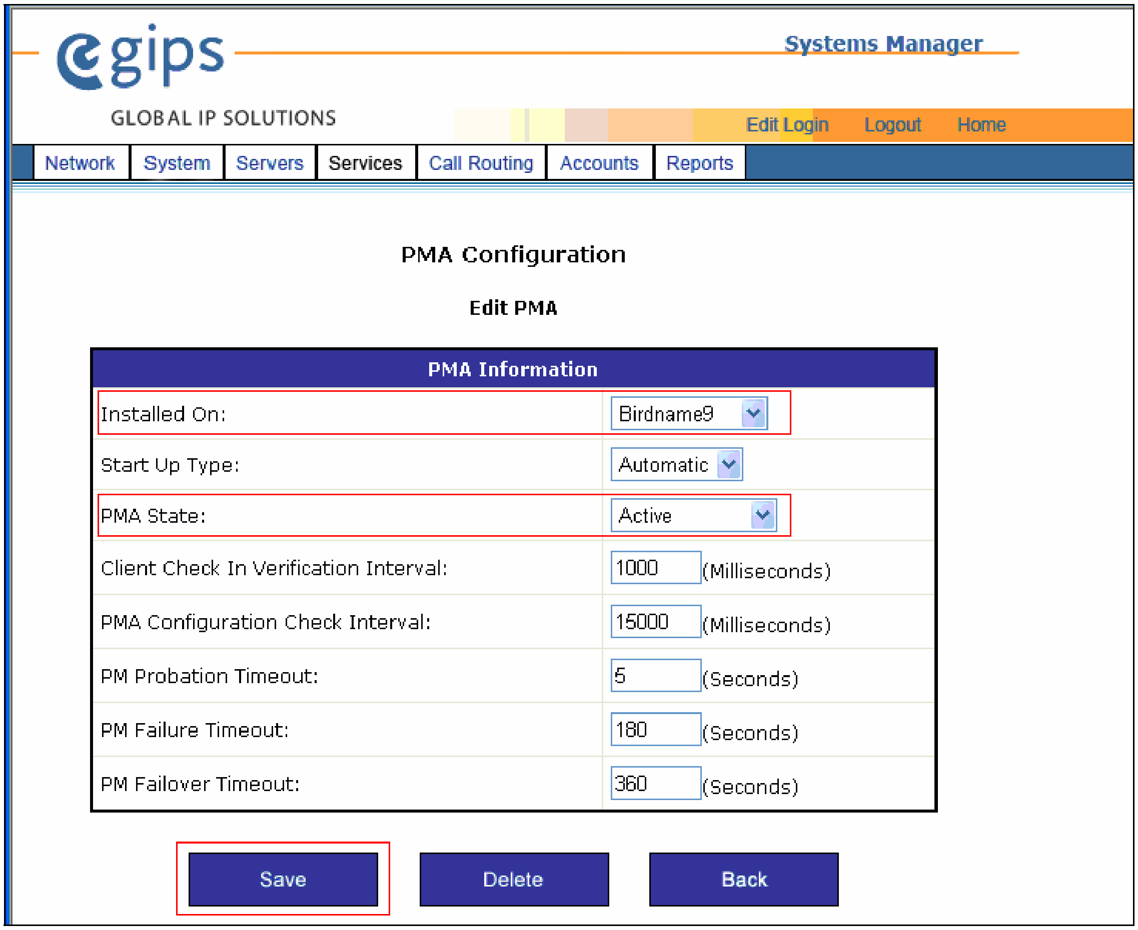
5.4. Configuring IVX Service

Navigate to **Services → IVX** from the main menu and select the **Add** button to add a new **IVX** service. The following shows the IVX Configuration page. Enter highlighted fields as shown. The default code set for Global IP Solutions ISS/IVX Server is G.711. Click **Save** to complete.



5.5. Configuring the Presence Manager Aggregator (PMA) Server

Click on **Services → PMA** from the main menu and select the **Add** button to add a new PMA configuration. Select the server, **Birdname9**, for the Installed On field. Set PMA state as **Active**. Leave all other field as default. Click **Save** to complete.



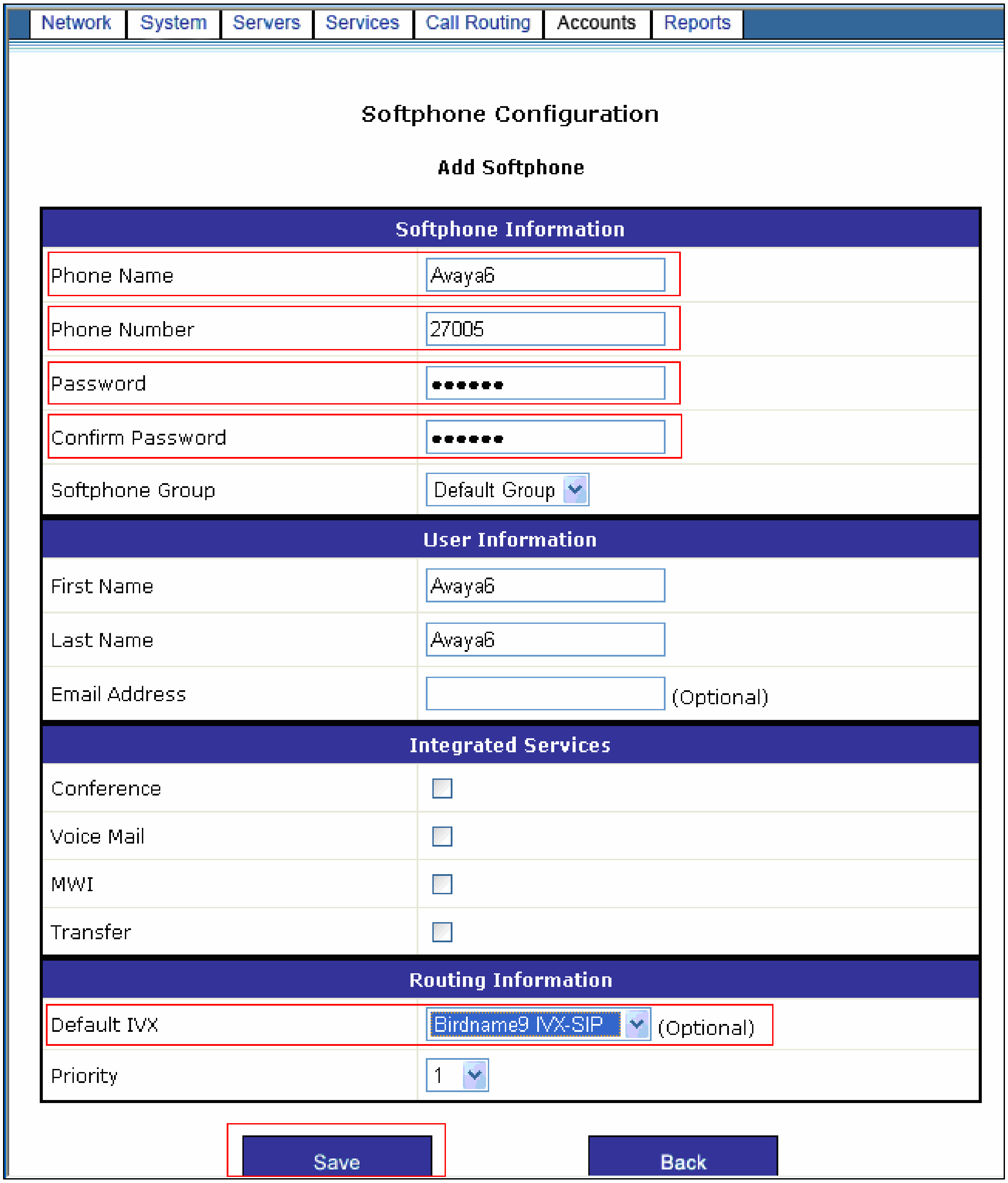
5.6. Configuring Softphones

Select **Accounts → Softphones** from the main menu and click the **Add** button to add a new softphone account into the system. Enter the following:

- Phone Name - Enter the Remote Extension client's log in name.
- Phone Number - Enter the phone number assigned to the softphone client. This phone number should be the same as phone number assigned for this softphone in the Avaya SIP Enablement Services Server. This sample configuration used **27005** as one of the Phone Numbers.
- Password - Enter the password set in Avaya SES from **Section 4.3**.
- Default IVX - Enter the server created in **Section 5.1**.

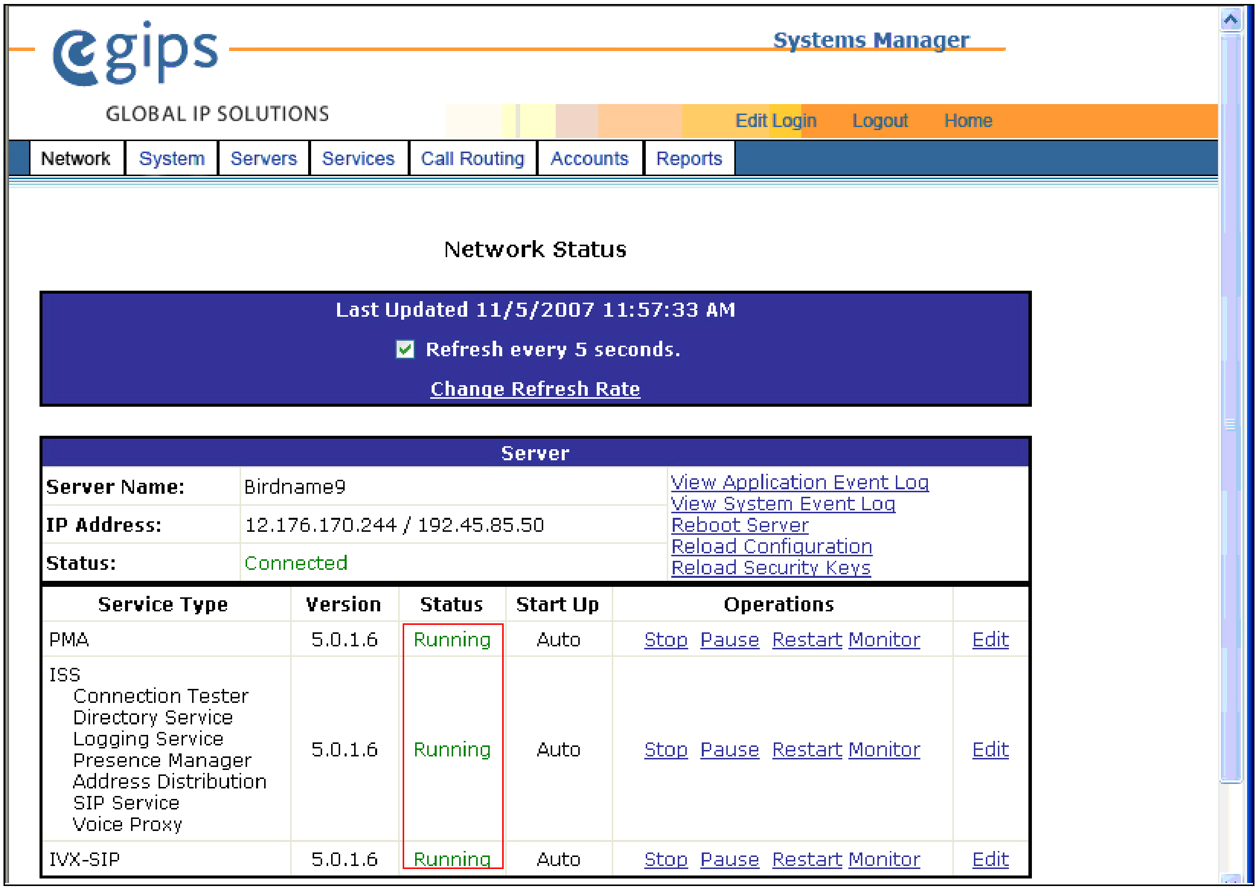
Repeat this step for each Remote Extension softphone client that needs to log in to the system. Enter a unique Phone Name and Phone Number for each client.

Click **Save** to complete.



5.7. Services Verification

To verify that the Global IP Solutions ISS/IVX is running, click on **Network** from the main menu. This will display a list of all the services and the current status of each. ISS, PMA, and IVX-SIP services should all be listed as **"**Running."



6. General Test Approach

The general test approach was to attempt calls between Avaya IP telephones (SIP and H.323), and Global IP Solutions Remote Extension softphones while exercising features of the telephones such as Hold, Transfer and Conference. Additionally, Remote Extension was tested against many of the Avaya Feature Name Extensions such as Call Park, Call Pick-up, Find-me, and Call Forwarding. Both G.711 and G.729AB audio codecs were exercised.

6.1. Test Results

Remote Extension successfully completed test cases for all supported features with the exception of the following:

- Global IP Solutions Remote Extension does not support the Consult (or Attended) Transfer feature.
- During a blind transfer, when the "xfer" button is pushed the first time, the voice can be heard on the call established.

Please contact Global IP Solution for further updates on these observations.

7. Verification Steps

The following steps may be used to verify the configuration:

- Log in to the Global IP Solutions Systems Manager via the web browser and select **Network** from the main menu. The following three (3) services should be in a running state, as shown in **Section 5.7**.
  i) ISS
  ii) PMA
  iii) IVX-SIP
- Log in to Avaya SES via the Web browser. The Registered Users field under the **Users** section will also show all registered SIP users including all the registered Remote Extension users.
- Place a call from a Global IP Solutions Remote Extension client to an Avaya IP Telephone. Verify that the call can be completed with a talk path established, and that the call can be disconnected.

8. Support

For technical support on the Global IP Solutions product line, contact Global IP Solutions at SolutionsSupport@GIPScorp.com and 1-805-899-4260.

9. Conclusion

These Application Notes describe the administration steps required to support Global IP Solutions Remote Extension on Avaya Communication Manager with the Avaya Enablement Service (SES) Server. With the exception of the issues identified in **Section 6.1**, the Global IP Solutions Remote Extension supported all basic and extended features that were tested and can interoperate successfully with Avaya Communication Manager and Avaya SES.

10. Additional References

[1] Administrator Guide for Avaya Communication Manager, 03-300509, Issue 3.1, February 2007
[2] Installing and Administering SIP Enablement Services, 03-600768, May 2007
[3] Global IP Solutions Remote Extension for SIP Install Notes, Doc #5900-1045
[4] Global IP Solutions Systems Manager Reference Guide, Doc #5900-1029

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

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