**Avaya Solution & Interoperability Test Lab**

# Application Notes for Configuring CrystalVoice Remote Extension with Avaya Communication Manager and Avaya SIP Enablement Services[1] Server - Issue 1.0

## Abstract

These Application Notes describe the steps for configuring CrystalVoice Remote Extension to communicate with Avaya Communication Manager and Avaya SIP Enablement Services Server. The CrystalVoice Remote Extension solution consists of the Remote Extension softphone client and the CrystalVoice ISS/IVX Server. The Remote Extension softphone client communicates with the CrystalVoice ISS/IVX Server via a CrystalVoice propriety protocol; the CrystalVoice ISS/IVX Server then registers the Remote Extension client with Avaya Communication Manager and Avaya SIP Enablement Service Server. Emphasis of the testing was placed on verifying good voice quality from CrystalVoice Remote Extension and its ability to operate with the Avaya SIP Enablement Service Server. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

[1] Beginning with release 3.0 software, the Avaya Converged Communication Server (CCS) has been renamed to the Avaya SIP Enablement Services (SES) Server.

# 1. Introduction

Avaya Communication Manager and Avaya SIP Enablement Service (SES) Server has the capability to extend advanced telephony features to SIP stations. This feature set can be extended to non-Avaya SIP phones such as CrystalVoice Remote Extension.

These Application Notes describe a solution for configuring CrystalVoice Remote Extension to operate with Avaya Communication Server and Avaya SIP Enablement Service (SES) Server. The CrystalVoice Remote Extension solution consists of two parts, a softphone client that runs on MS Windows and the CrystalVoice ISS/IVX Server that resides in the core network. CrystalVoice Remote Extension communicates with the CrystalVoice ISS/IVX Server via a propriety protocol. In turn, the CrystalVoice ISS/IVX Server registers softphone client(s) to Avaya Communication Manager with the SIP Enablement Services (SES) Server via the standard SIP protocol.

## 1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. The CrystalVoice ISS/IVX Server has two Ethernet connections, one Ethernet is connected into the private core network, and the other Ethernet is connected to the Internet. Two CrystalVoice Remote Extension clients communicate with the CrystalVoice ISS/IVX Server through a Linksys Router via a common Internet IP address through the Internet. The extension numbers used by CrystalVoice Remote Extensions are registered to Avaya Communication Manager via Avaya SIP Enablement Services Server and are also administered as Off-PBX-Telephones stations in Avaya Communication Manager. As a result, each CrystalVoice Remote Extension softphone has access features available from Avaya Communication Manager.

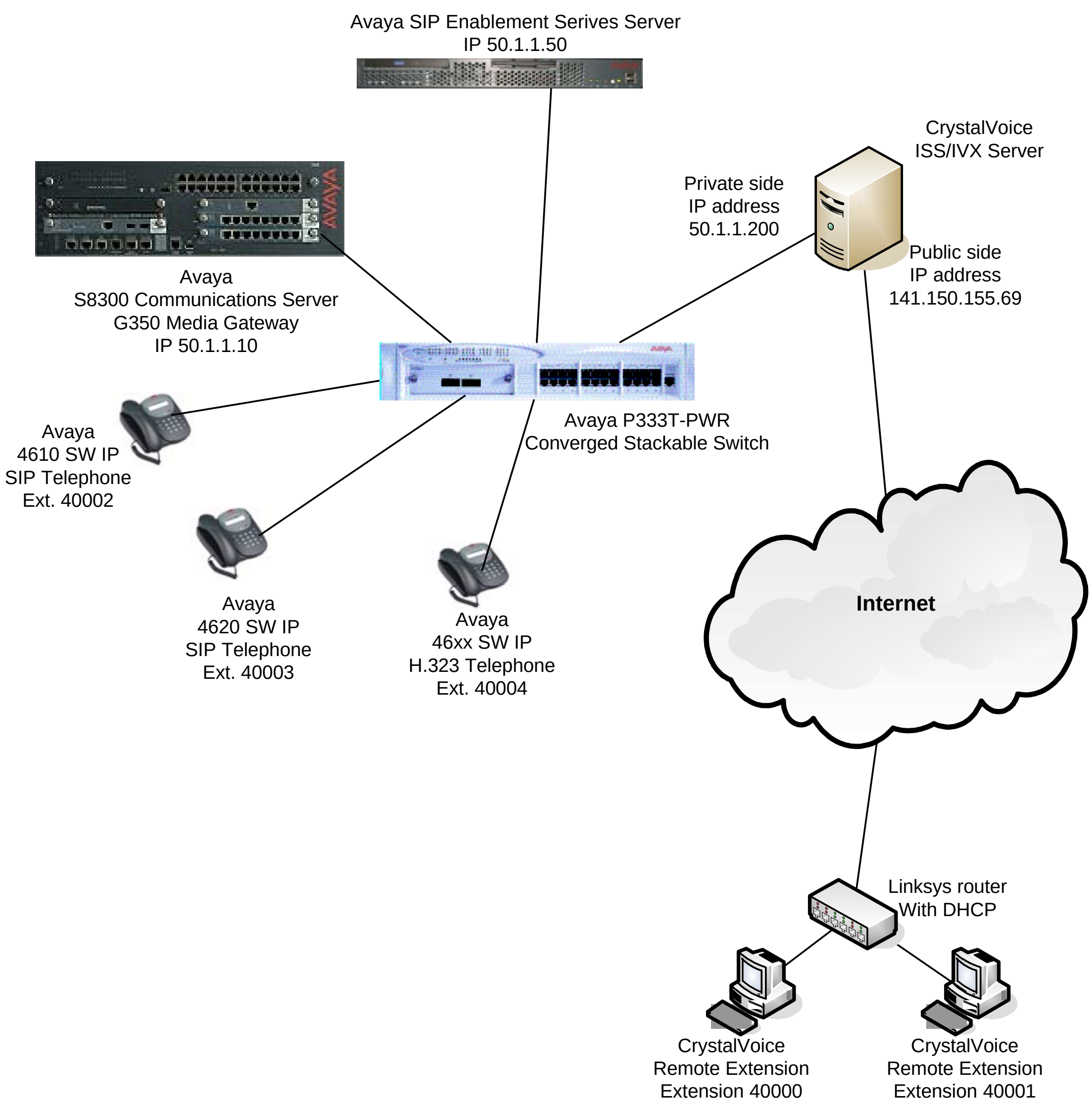



**Figure 1: Sample Network Configuration**

## 2. Equipment and Software Validated

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

| Equipment | Software/Firmware |
|---|---|
| Avaya S8300 Media Server with a G350 Media Gateway | Communication Manager 3.0 (R0.13x.00.0.340.3) with update 00.0.340.3-10458 |
| Avaya SIP Enablement Services Server | 3.0 (build 31) |
| Avaya IA770 INTUITY AUDIX™ | 3.0-1.7 |
| Avaya 4610SW/4620SW IP Telephones | 2.2.3 |
| Avaya P333T-PWR Converged Stackable Switch | 3.12.1 |
| CrystalVoice Integrated System Services (ISS) | 4.2.0.4 |
| CrystalVoice Internet Voice Transcoder for SIP (IVX) | 4.2.0.4 |
| Crystal Voice Remote Extension Client w/ MS Windows 2000 Professional | 4.2.0.3 |

## 3. CrystalVoice Remote Extension

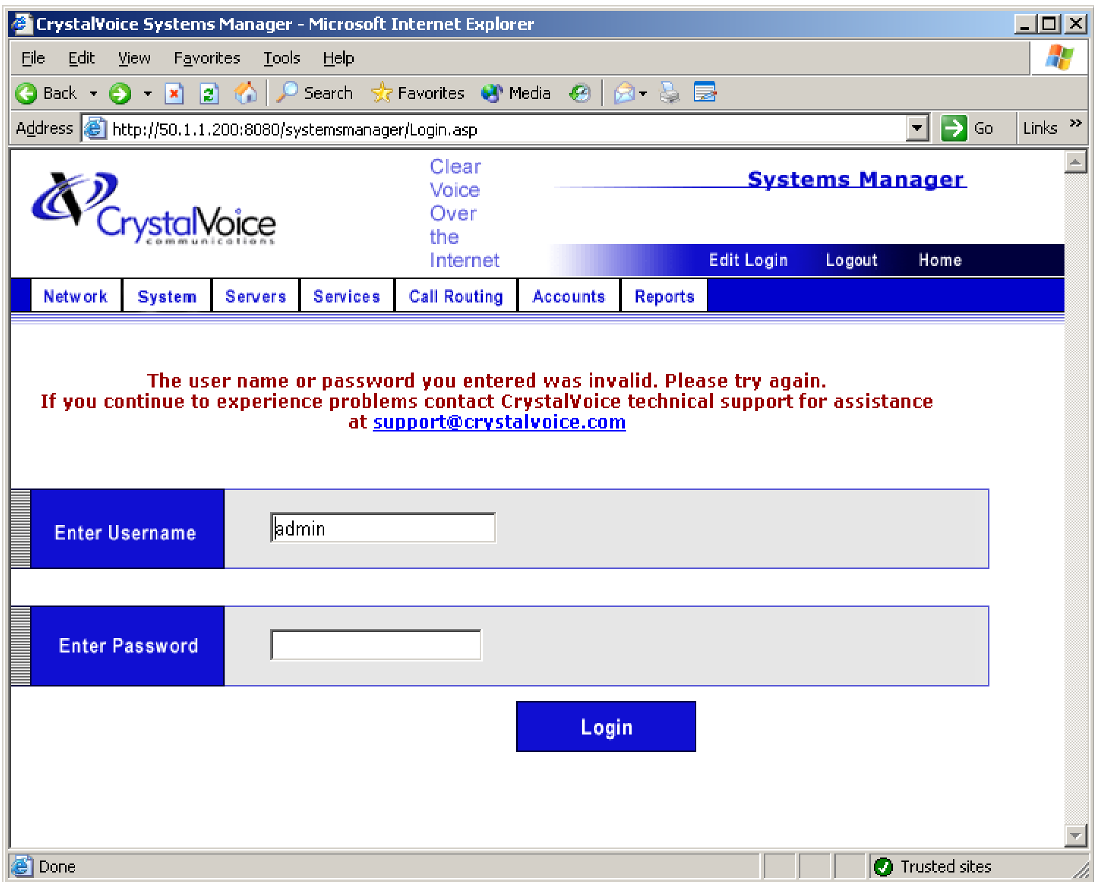
There are two parts to the CrystalVoice Remote Extension solution - the CrystalVoice Remote Extension softphone client and the CrystalVoice ISS/IVX Server. The CrystalVoice ISS/IVX Server transcodes traffic between Remote Extension Clients running CrystalVoice's proprietary protocol and the Avaya SIP Enablement Services Server which communicates via SIP. Although this sample configuration shows a dual Ethernet connected ISS/IVX Server, it is possible to configure the ISS/IVX Server with a single Ethernet connection to support both traffic to and from Avaya Communication Manager and the Internet.

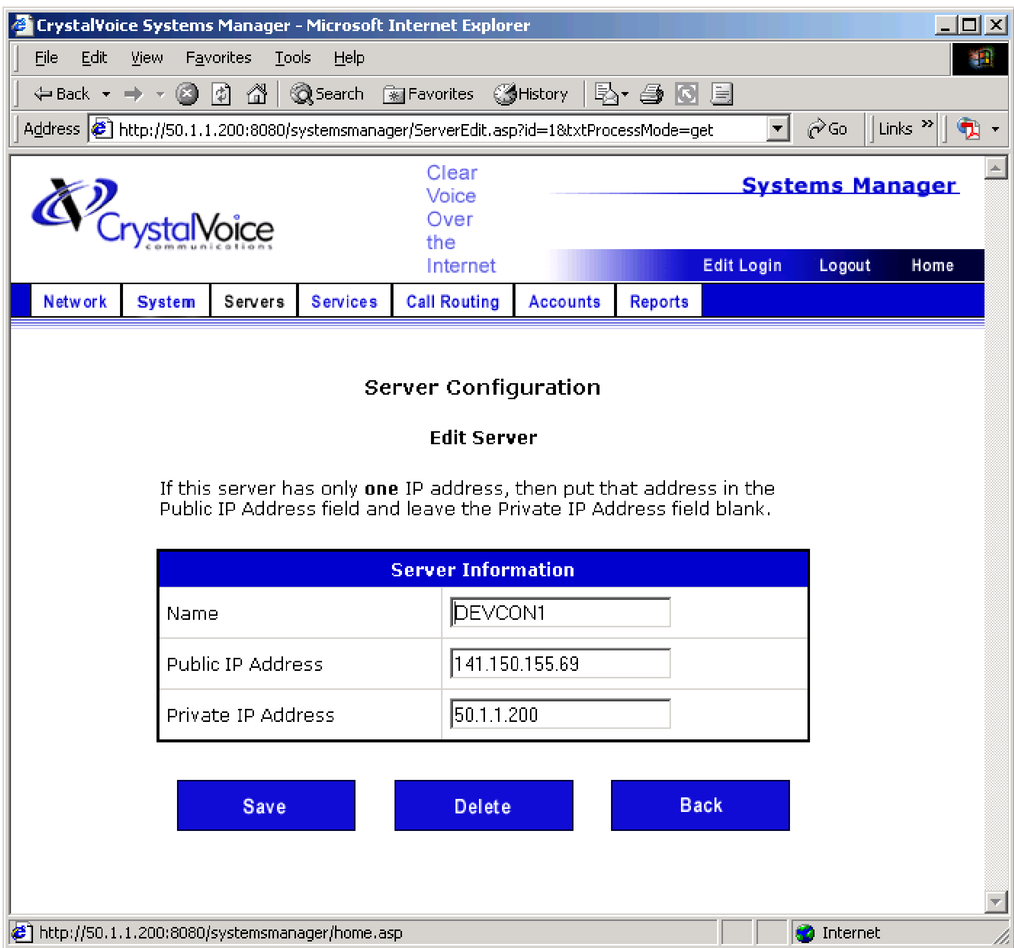
From the perspective of the Avaya SIP Enablement Services Server, CrystalVoice Remote Extension clients look like actual SIP telephones all registered from the IP address of the ISS/IVX Server. Since the call-id, from-Tag, and to-tag in the SIP packet header uniquely identify each SIP dialog, the Avaya SIP Enablement Service Server will be able to treat each as separate SIP telephones.

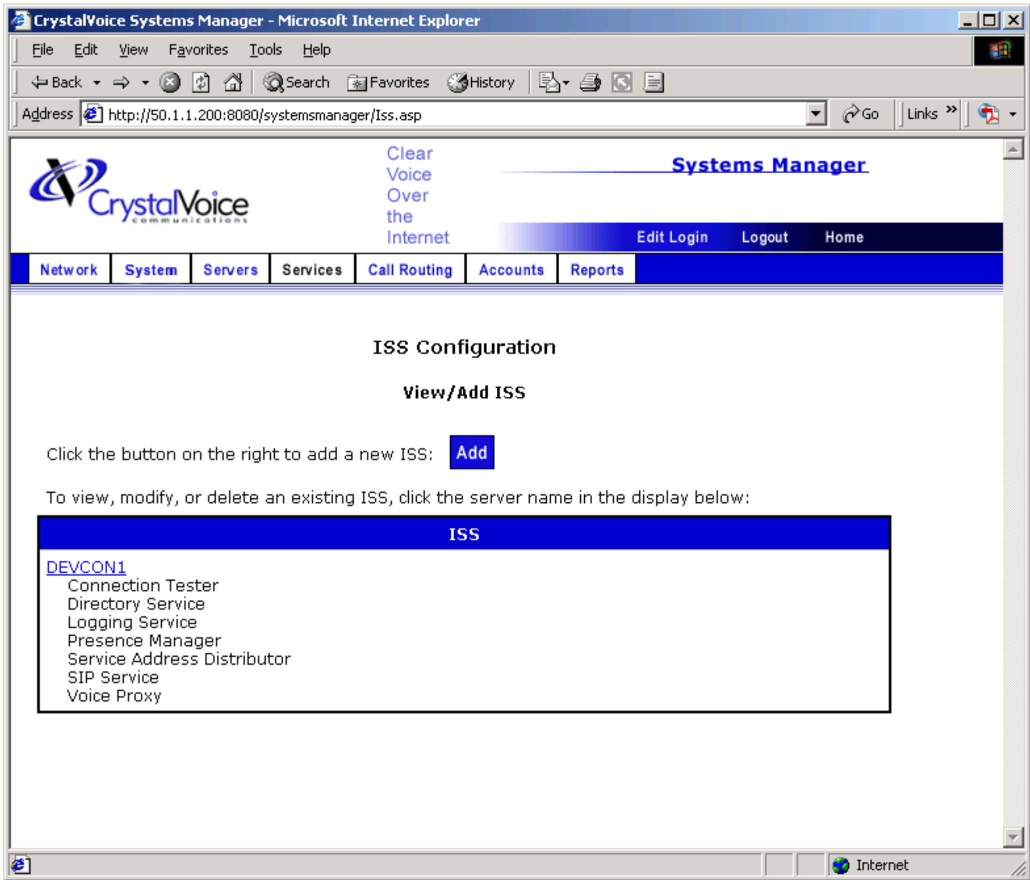
The ISS/IVX Server does not support L2 (802.1Q) or L3 (DiffServ) tagging. Therefore, it is recommended that the Ethernet switch port that the CrystalVoice ISS/IVX Server connects to be configured with the appropriate priority setting, and all intermediate routers to the Avaya SIP Enablement Service Server be configured with the corresponding QoS policy. Both the ISS/IVX Server and the Avaya SIP Enablement Service Server reside on the same network in the sample configuration.

## 3.1. Configuring the ISS/IVX Server

The following steps describe the configuration for the ISS/IVX Server to communicate with the Avaya Enablement Services Server. This configuration allows for calls to and from Remote Extension clients. A Web Server must be installed on the same machine where the CrystalVoice ISS/IVX server software is running. The sample configuration uses Microsoft IIS.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Log in to the CrystalVoice ISS/IVX Server via its IP address using the Web browser. The CrystalVoice ISS/IVX installation process automatically changes the Web Server port to 8080.<br><br>http://50.1.1.200:8080/systemsmanager/home.asp<br><br><br></td></tr>
</table>

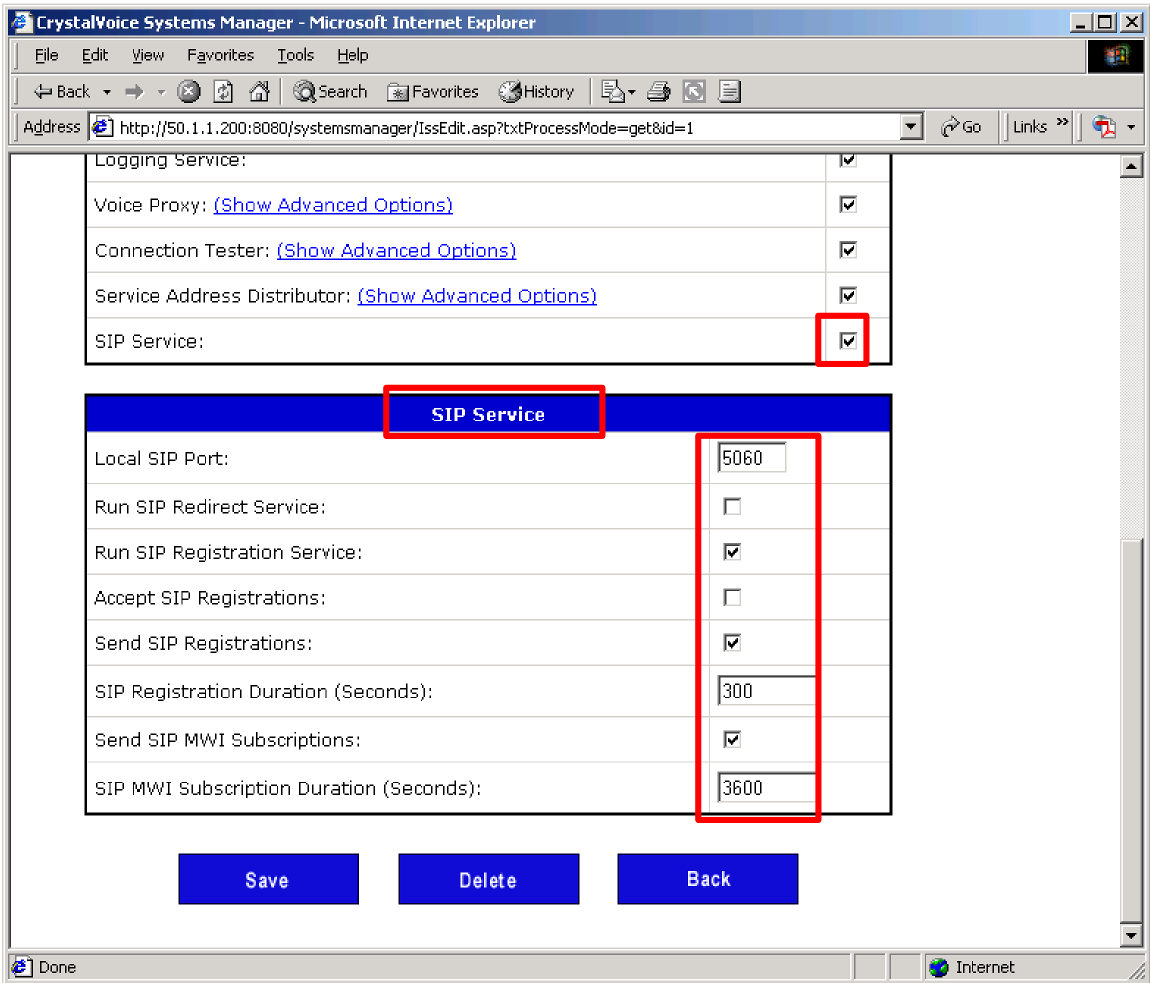
| Step | Description |
|---|---|
| **2.** | Click **Servers** on the main menu bar and select **Add** Server.<br>Note:<br>• For a single Ethernet connection scenario, the IP address must be entered into the Public IP Address field<br>• For the Private IP address field, the CrystalVoice software is designed to recognize private IP addresses as defined in RFC 1918, Address Allocation for Private Internets.  The CrystalVoice software is designed to handle calls from both internal/LAN connected and external/Internet connected softphones when the private/LAN IP addresses fall within the recommendations of RFC 1918.  In general, this should be the case for most companies.  If public IP addresses are used in the private network and there is a public Internet connection with IP addressing in a different public network, for example the LAN is addressed in the 50.1.1.x network and the public Internet connection in the 141.150.155.x network, then contact CrystalVoice to discuss the setup.<br> |

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Select Services→ ISS from main menu. Click on the server created in Step 2. The sample configuration used DEVCON1.<br>


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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>

Scroll down to the **SIP Service** portion. Make sure the red highlighted fields are set as shown. Click **Save** to complete.



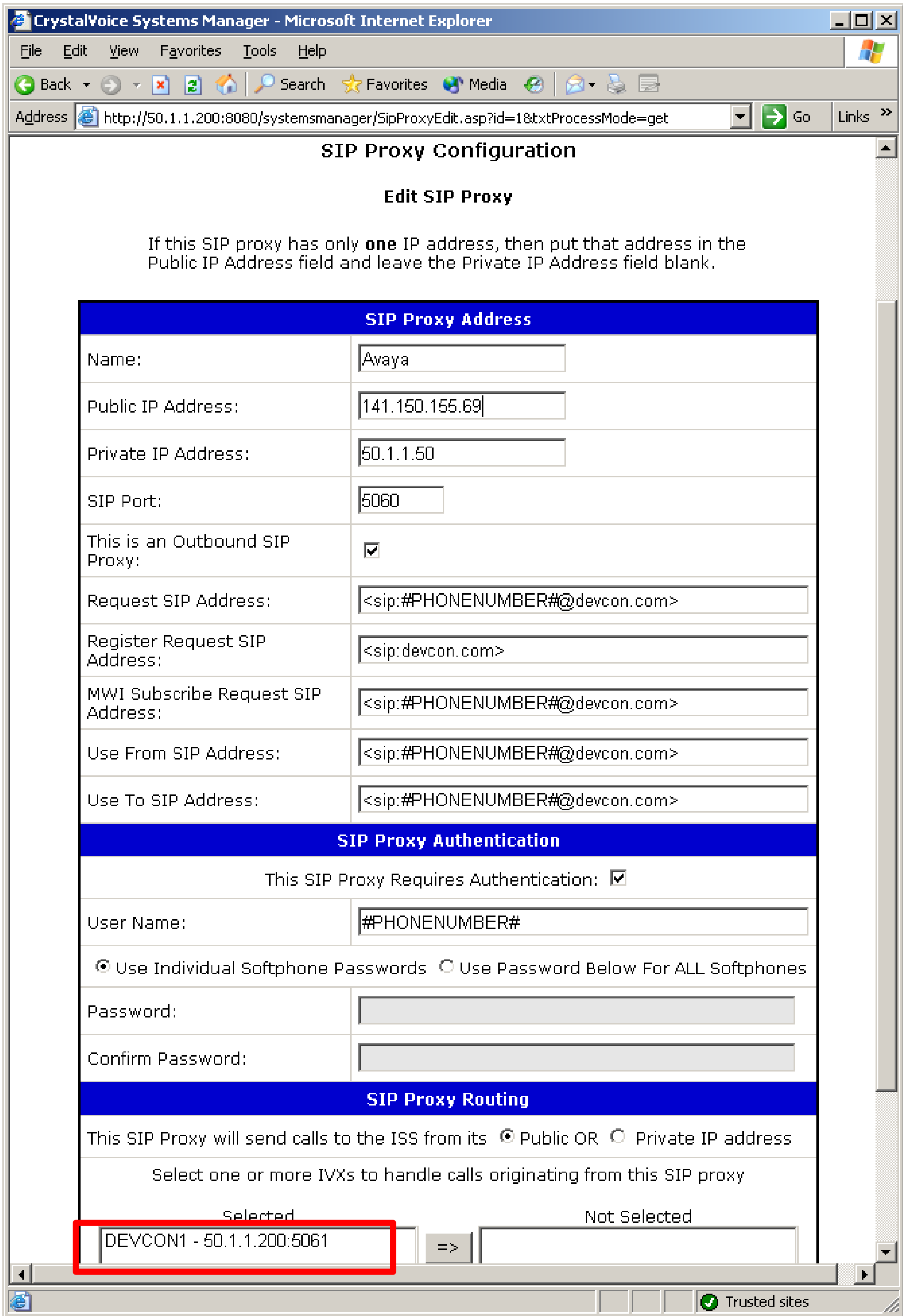

</td></tr>
<tr><td>5.</td><td>

After clicking on call **Call Routing → SIP Proxies** from the main menu. The following **SIP Proxy Configuration** page will be displayed. Enter all fields as shown. Make sure the server ***DEVCON1*** entered in step 2 is selected. Click **Save** to complete.

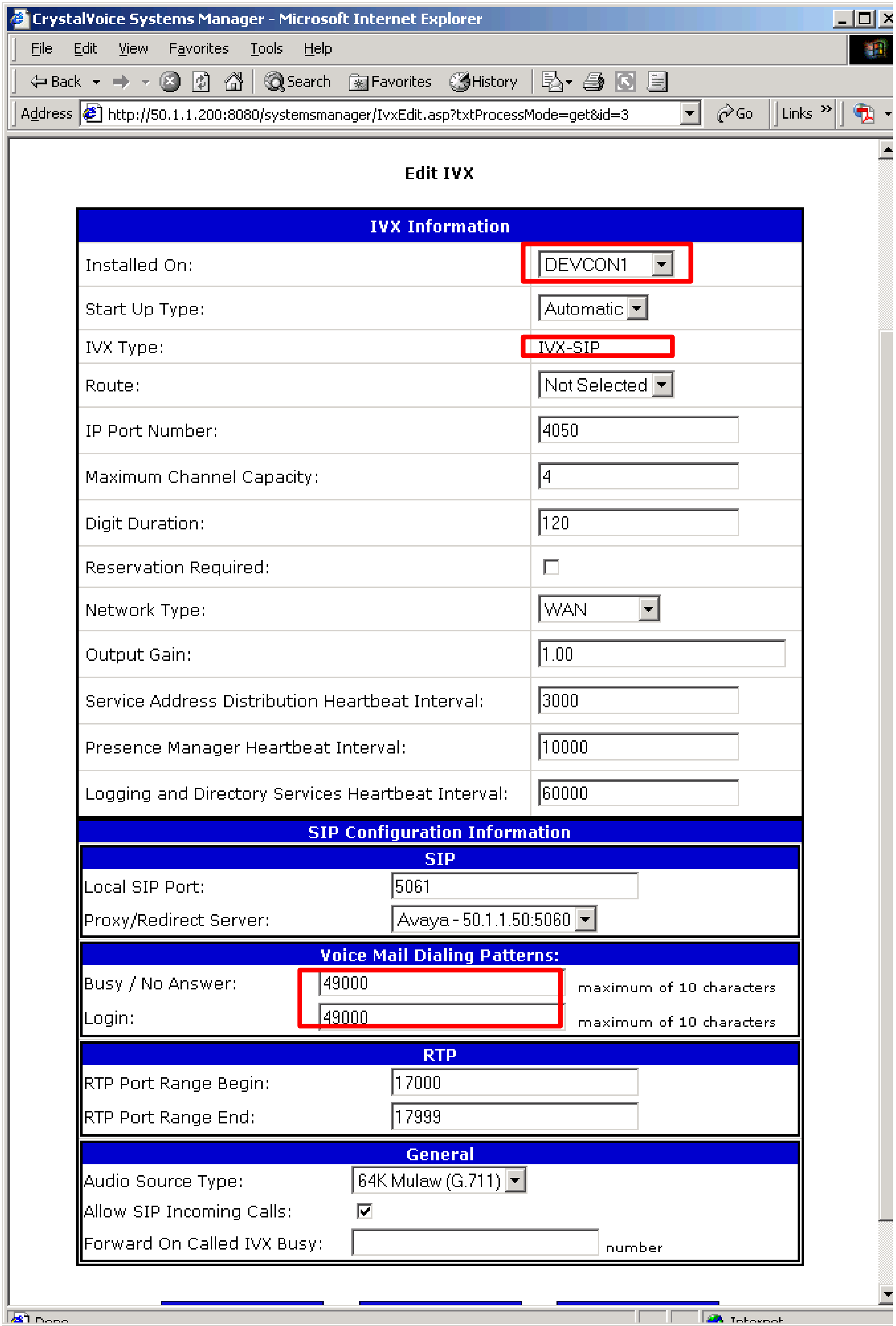
**Public IP Address**: Public side IP address of the CrystalVoice ISS/IVX Server<br>
**Private IP Address**: IP address of the Avaya SIP Enablement Services Server.<br>
**SIP Port**: This is the port number used to communicate with the Avaya SIP Enablement Services Server<br>
***devcon.com*** is the domain name used in the sample configuration. Appropriate domain name should be entered in place of ***devcon.com.***

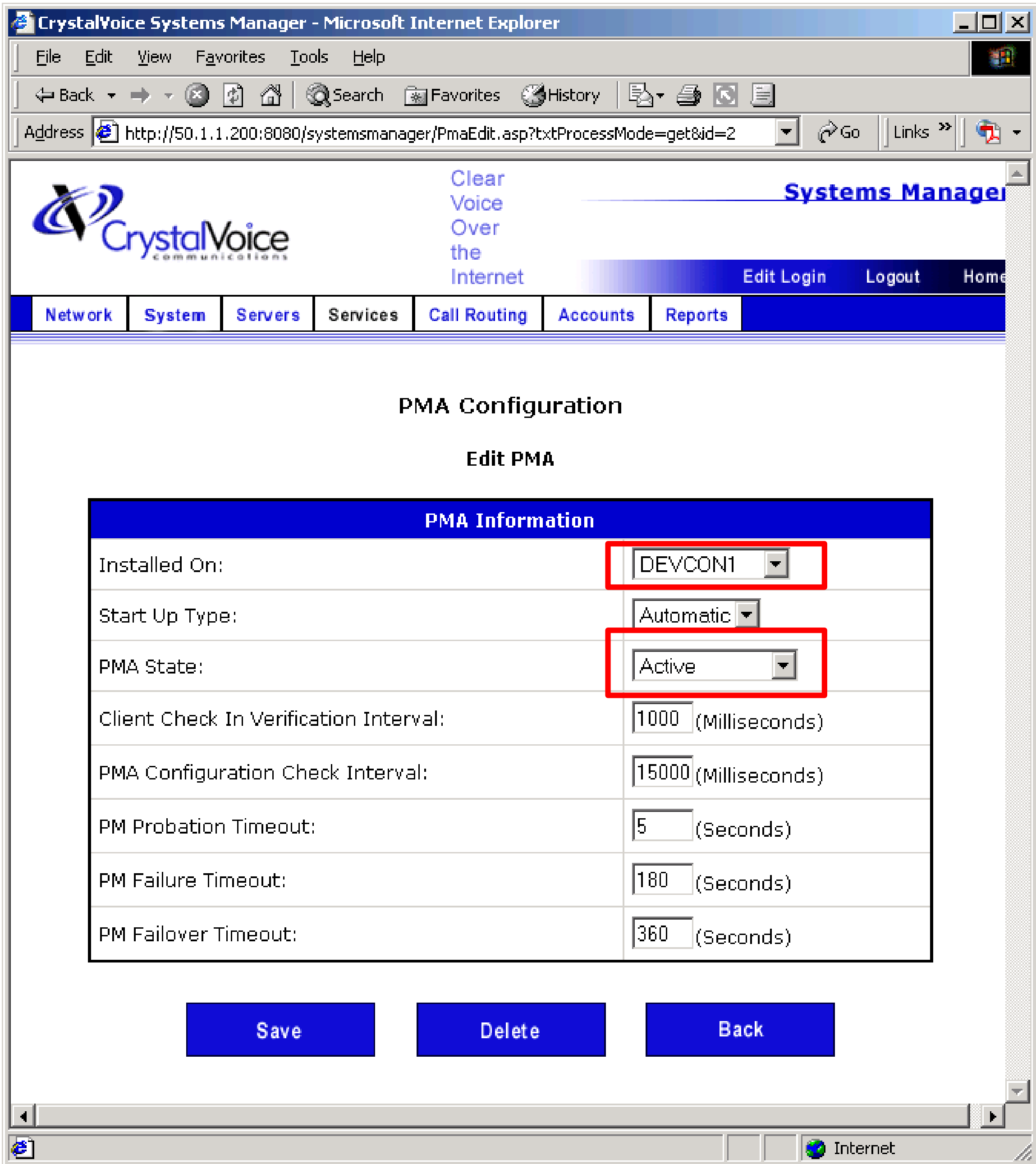
</td></tr>
</table>

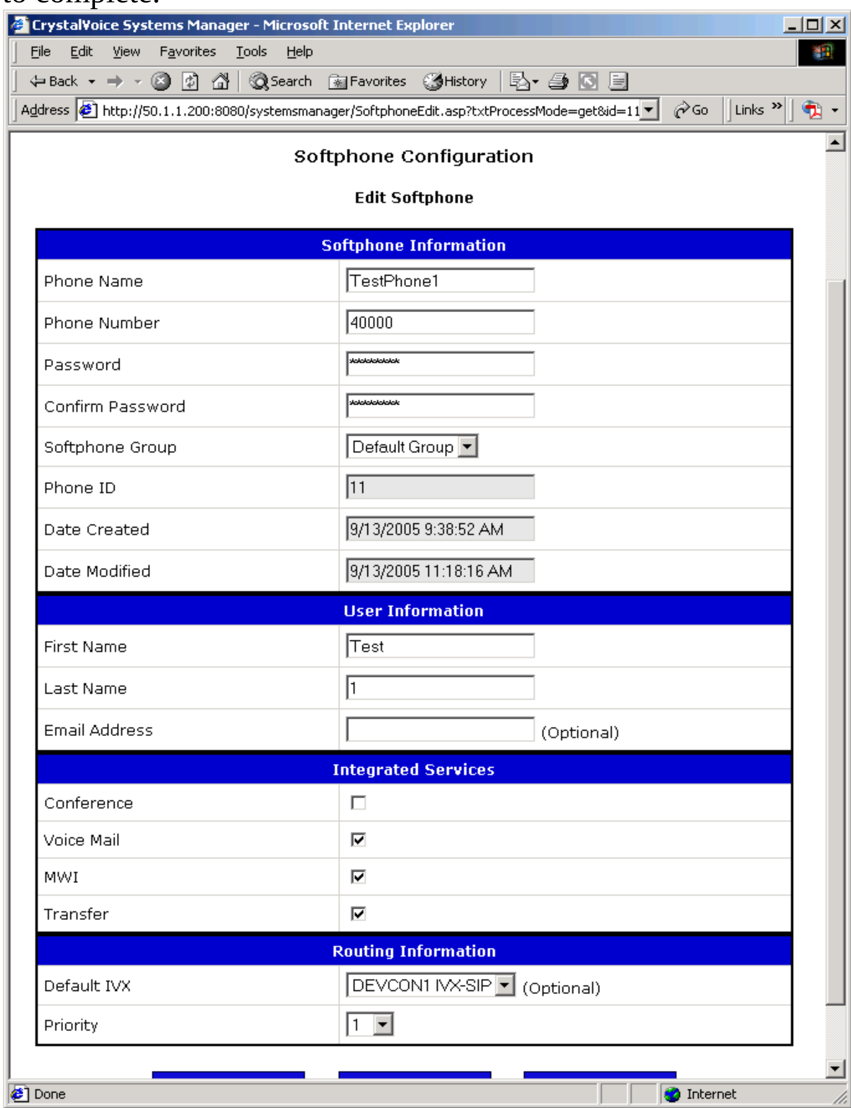
| Step | Description |
| --- | --- |
| | In the case of a single Ethernet connection, enter the IP address in the **Public IP Address** field. Scroll down and click **Save** to complete. |



| Step | Description |
|---|---|
| 6. | After clicking on **Services → IVX** from the main menu and selecting **Add** a new **IVX** service, the following screen will appear. The default code set for CrystalVoice ISS/IVX Server is G.711. Click **Save** to complete. |

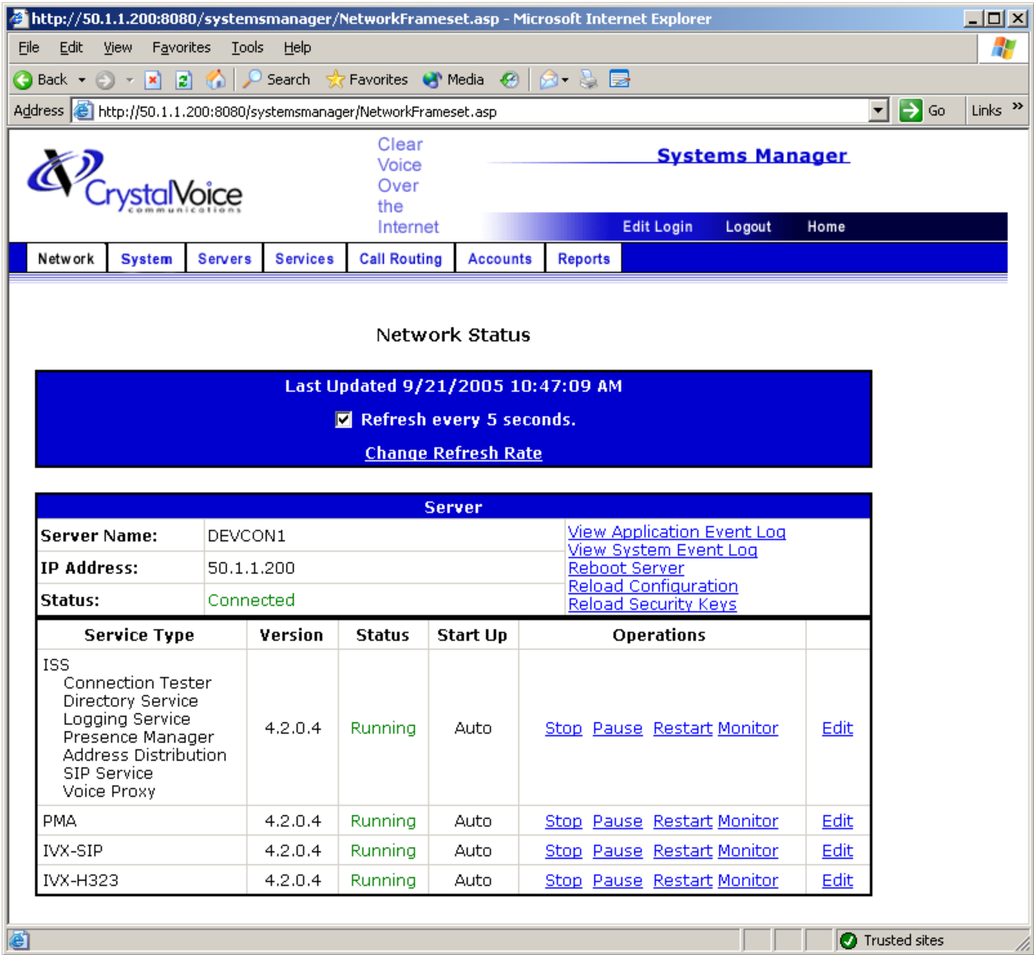


<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>Click on Services → PMA from the main menu and Add a new PMA configuration. Select the Server, DEVCON1 that was created in Step 2 for the Installed On field. Set PMA state as Active. Leave all other field as default. Click Save to complete.<br>
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>8.</td><td>
Click on Accounts → Softphones from the main menu and Add a new softphone account into the system.<br>
Phone Name is the Remote Extension client’s log in name.<br>
Phone Number is 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 40000 as the Phone Number.<br>
Password is the same as the password set in the Avaya SIP Enablement Services Server.<br>
Voice Mail, MWI, & Transfer are checked.<br>
Default IVX is the server created under services IVX earlier.<br><br>
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.<br><br>
Click Save to complete.<br>


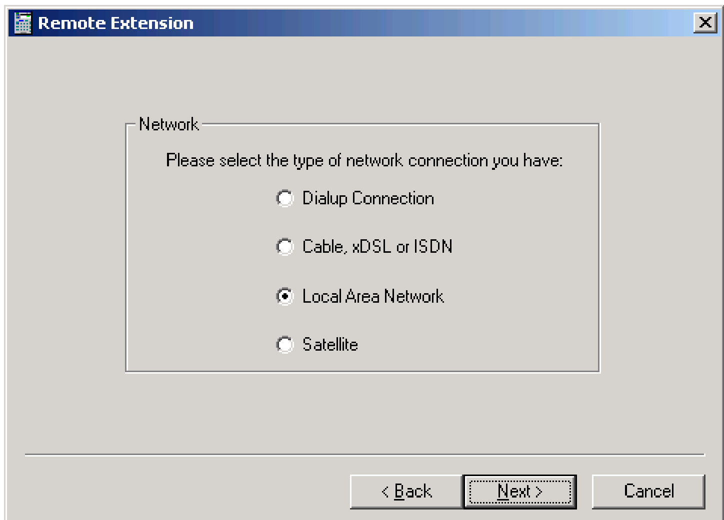
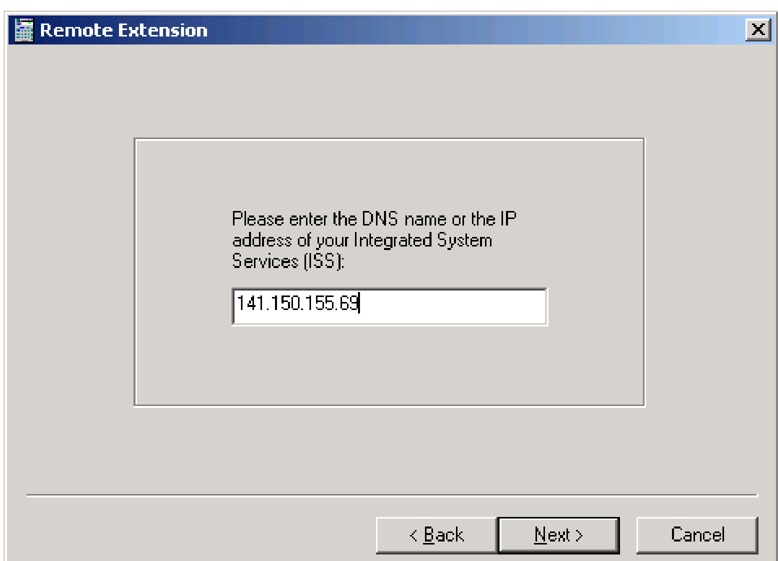
</td></tr>
</table>

| Step | Description |
|---|---|
| **9.** | To verify if the CrystalVoice ISS/IVX is running, click on **Network** from the main menu. This will display a list of all the services and the current status. **ISS**, **PMA**, and **IVX-SIP** should all be listed as *Running*. |



## 3.2. Configuring Remote Extension Client

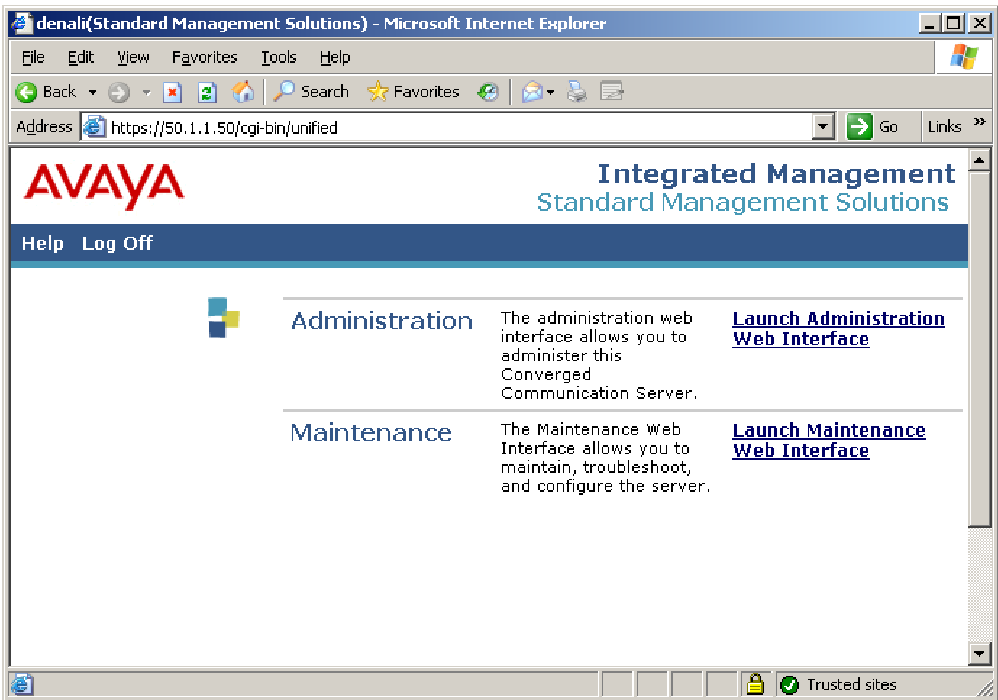
The following screens highlight the areas that must be set during the installation of CrystalVoice Remote Extension Client. Begin installation of the CrystalVoice Remote Extension Client by running the client installation software “SoftphoneInstall.RE.SIP.exe”.

| Step | Description |
| --- | --- |
| 1. | Select the appropriate type of network connection. The sample configuration used ***Local Area Network***.<br> |
| 2. | Enter the **DNS name** or **IP address** of the ISS. In the sample configuration, the CrystalVoice ISS/IVX Server’s IP address was used.<br> |

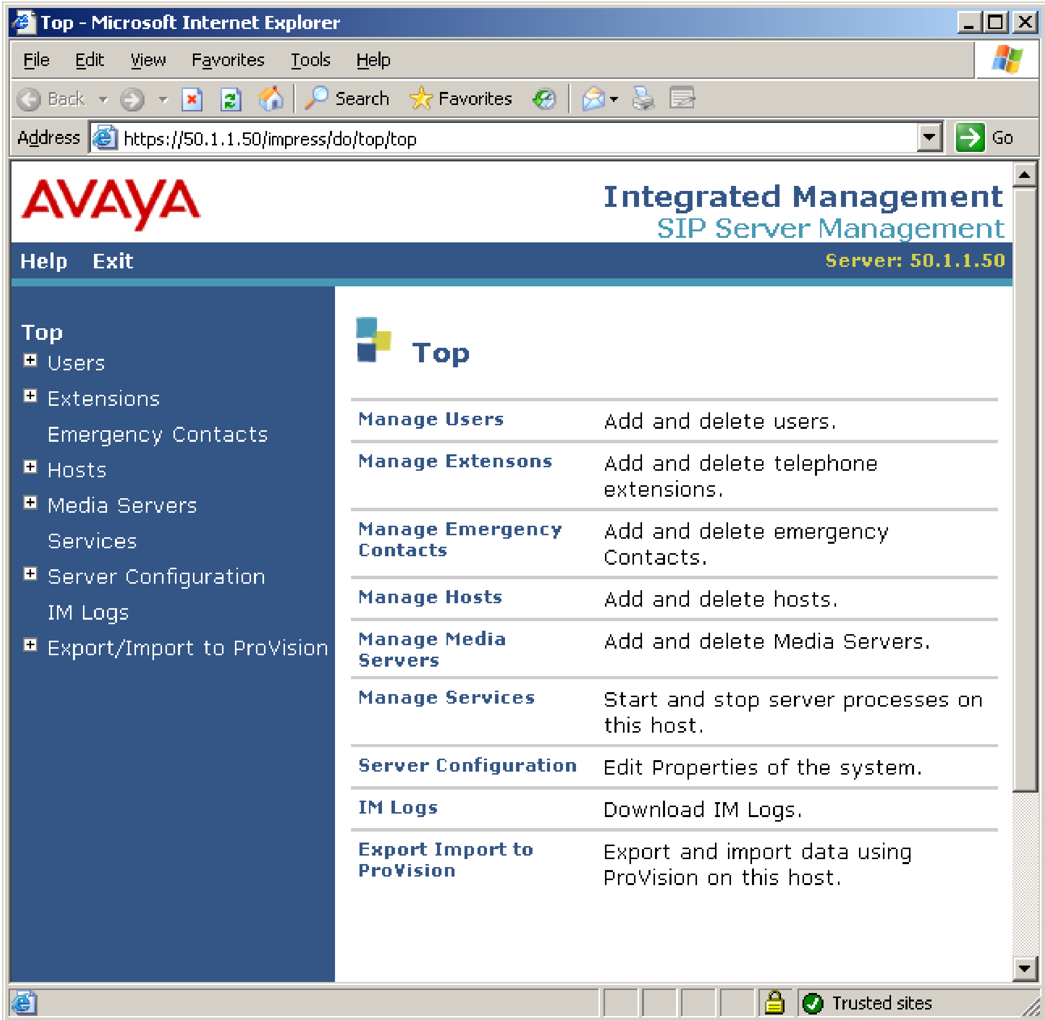
| Step | Description |
|---|---|
| **3.** | Log in to the CrystalVoice Remote Extension client.<br>**Phone name** The Softphone account name configured on the CrystalVoice ISS/IVX Server.<br>**Password**: The password for the Phone name above, which is the same as configured on the Avaya SIP Enablement Services Server for the extension that corresponds to the Phone name.<br><br>Below displays the login window of the Remote Extension Client. Phone name *testphone2* was created following step 8 in section 3.1.<br><br><br>CrystalVoice Remote Extension client screen is displayed below. Both L1 and L2 should display **Ready**.<br><br> |

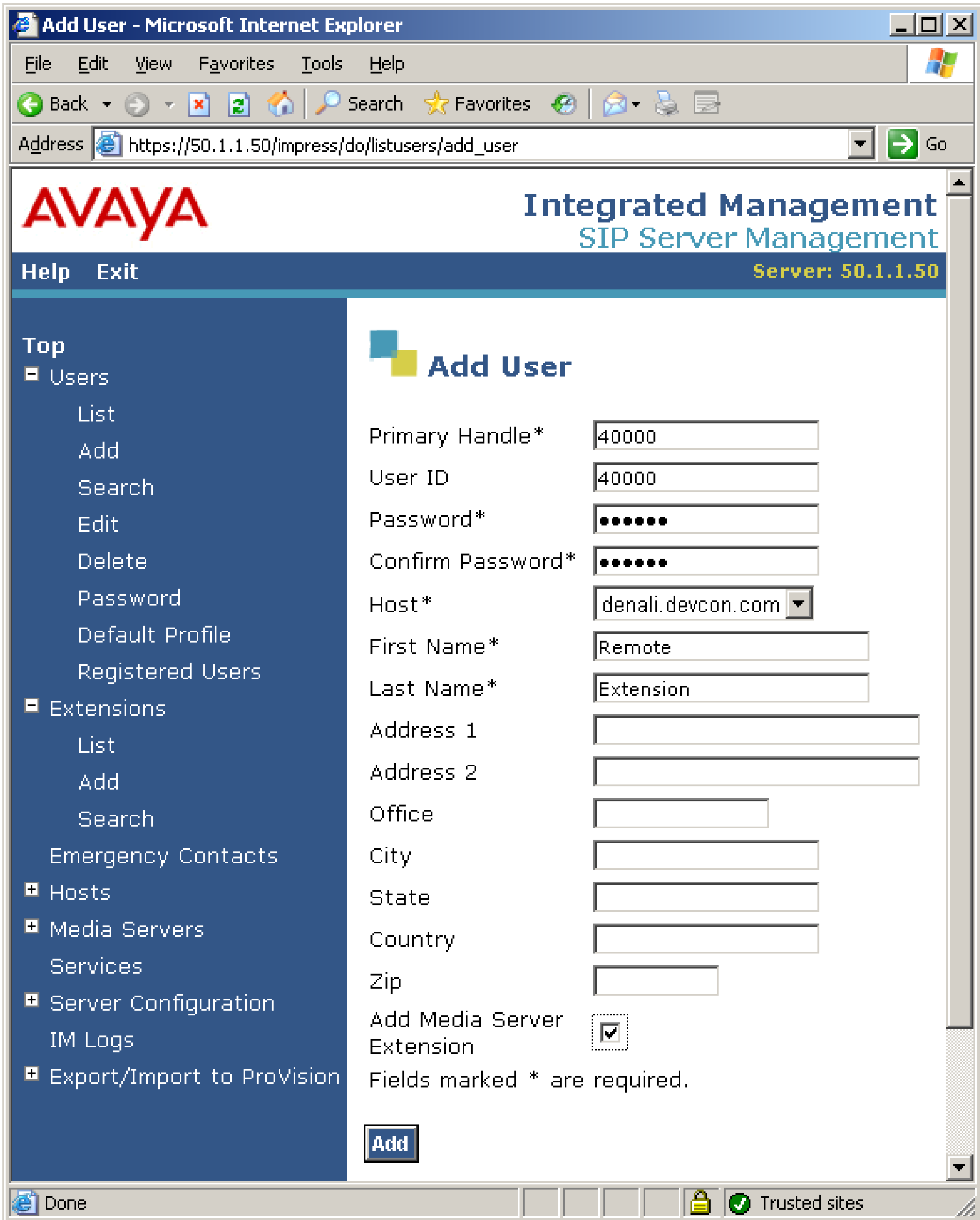
# 4. Configuring the Avaya SIP Enablement Services Server

The following steps describe the configuration of the Avaya SIP Enablement Services server to support Remote Extension Clients.

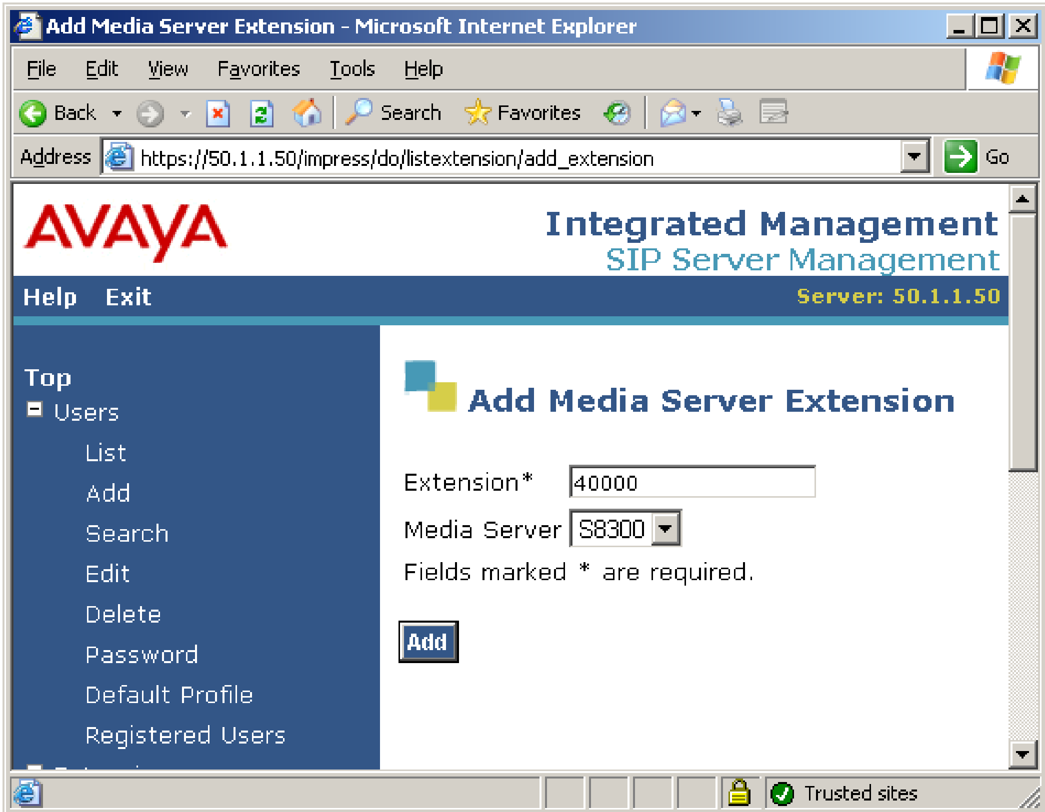
| Step | Description |
|---|---|
| 1. | The Avaya SIP Enablement Services (SES) Server is configured using a web browser. Set the URL to the IP address of the SIP Enablement Service (SES) Server, and log in using appropriate user name and password. The URL in the sample configuration is **https://50.1.1.50/admin**. Select **Launch Administration Web Interface** to continue.<br> |

| Step | Description |
|---|---|
| 2. | Click on the "+" sign next to **Users** on the left side to expand the selection. Select **Add** from the list under **Users** to add a new user. |



| Step | Description |
|---|---|
| 3. | Enter the phone number in both the **Primary Handle** and **User ID**. Enter the password for this extension. This password is the same as that was used in step 8 of section 3.1. Select the appropriate Avaya SIP Enablement Services Server for the **Host** field. The **First Name** and **Last Name** are for informational purpose only, but are required fields. Check the **Add Media Server Extension** field at the bottom. Click **Add** to continue.  |

| Step | Description |
|---|---|
| **4.** | Type in the phone number in the **Extension** field. Click **Add** to complete. |



# 5. Avaya Communication Manager

This section highlights the important commands for defining SIP telephones on Avaya Communication Manager. For complete documentation, see Reference[1][2]. Use the System Access Terminal (SAT) interface to perform these steps. Log in with the appropriate credentials.

## 5.1. Adding new stations to Avaya Communication Manager

Using the **add station** command, add a station for each SIP phone to be supported. The sample configuration uses ***6408D+*** for the station type and be sure to include the coverage path for voice mail if it is available. Use the appropriate COS value. Make sure that the station has three (3) ***"call-appr"*** for **Button Assignment**. Set the IP Softphone to ***n***. Repeat the following steps to add additional telephone extensions.

```
add station 40000                                            Page   1 of   4
                                     STATION

Extension: 40000                         Lock Messages? n          BCC: 0
     Type: 6408D+                        Security Code:             TN: 1
     Port: X                           Coverage Path 1: 1          COR: 1
     Name: SIP40000                    Coverage Path 2:            COS: 1
STATION OPTIONS
              Loss Group: 2
             Data Module? n
            Speakerphone: 2-way
        Display Language: english
              Loss Group: 2            Personalized Ringing Pattern: 1
             Data Module? n                        Message Lamp Ext: 40000
            Speakerphone: 2-way                 Mute Button Enabled? y
                                                  Media Complex Ext:
                                                       IP SoftPhone? n
```

```
add station 40000                                            Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: audix            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         Conf/Trans on Primary Appearance? n
        H.320 Conversion? n        Per Station CPN - Send Calling Number? y
                                               Display Client Redirection? n
              AUDIX Name: IA770                Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 40000                          IP Audio Hairpinning? y
```

```
add station 40000                                            Page   3 of   4
                                     STATION
 SITE DATA
      Room:                                           Headset? n
      Jack:                                           Speaker? n
     Cable:                                          Mounting: d
     Floor:                                      Cord Length: 0
  Building:                                         Set Color:
ABBREVIATED DIALING
     List1:                List2:                    List3:
BUTTON ASSIGNMENTS
 1: call-appr                        5:
 2: call-appr                        6:
 3: call-appr                        7:
```

Use the **change off-pbx-telephone station-mapping** command to map Avaya Communication Manager extension (40000) to the Avaya SIP Enablement Service (SES) Server extension (40000). Select the trunk-group number for the **trunk-group** configured between Avaya Communication Manager and Avaya SIP Enablement Services Server. Select the **Configuration Set** number applicable for this configuration. The sample configuration uses **Configuration Set *1***. For additional information related to Avaya Communication Manager and OFF-PBX-EXTENSION support, please refer to Avaya documentation[2][5][6].

```
change off-pbx-telephone station-mapping 40000                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 40000         OPS              - 40000              1          1



change off-pbx-telephone station-mapping 40000                  Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 40000        3         both           all         both
```

Below shows the setting for **configuration-set 1**.

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1
                               CONFIGURATION SET: 1
      Configuration Set Description: Remote Extension
               Calling Number Style: network
                CDR for Origination: phone-number
 CDR for Calls to EC500 Destination? y
        Fast Connect on Origination? n
       Post Connect Dialing Options: dtmf
      Cellular Voice Mail Detection: none
                      Barge-in Tone? n
             Identity When Bridging: principal
```

## 5.2. Verify OPS Capacity

Use the display **system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones –**OPS has been set to a value that will accommodate the number of phones to be supported. Avaya Services has provisioned this during installation according to the system configuration purchased.

```
change system-parameters customer-options                       Page   1 of  10
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1
       Platform: 13
       Location: 1                           RFA System ID (SID): 1
       Platform: 13                          RFA Module ID (MID): 1
                                                                USED
                                Platform Maximum Ports: 900   48
                                      Maximum Stations: 40    20
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    0
                    Maximum Off-PBX Telephones -   OPS: 50    10
                    Maximum Off-PBX Telephones - SCCAN: 0     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

## 5.3. Configuring Audio Codec

In order for calls to be established successfully, during initial call setup the two end points must agree upon a mutually supported codec. Remote Extension supports both the G.711 (default) and G.729 codecs. G.711 codec was used in the sample configuration (see section 3.1 Step 6 for CrystalVoice ISS/IVX Server setting). To use the G.729 codec, use the **change ip-codec-set** command to set the codec setting to G.729AB, and make a corresponding change to the IVX Server in section 3.1 Step 6.

```
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:
```

## 5.4. General Test Approach

The general test approach was to attempt calls between Avaya SIP telephones, CrystalVoice Remote Extension softphones and Avaya H.323 IP telephones 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.729 codecs were exercised as well as shuffling and non-shuffling.

## 5.5. Test Results

Remote Extension successfully completed test cases for all supported features with the exception of Unattended Transfer. Remote Extension must be the call originator and transferring phone in order to execute an Unattended Transfer among SIP telephones. In addition, the CrystalVoice ISS/IVX Server does not perform any L2 (802.1Q) or L3 (DiffServ) tagging for its traffic. Therefore, it is recommended that appropriate priority setting be configured on the Ethernet switch port where the CrystalVoice ISS/IVX Server is connected to and corresponding policy be set on all intermediate routers in the case where the CrystalVoice ISS/IVX Server is situated in a different network from that of Avaya SIP Enablement Service (SES) Server to ensure proper Quality of Service.

# 6. Verification Steps

The following steps may be used to verify the configuration:

- Log in to the CrystalVoice Systems Manager via the web browser and select Network from the main menu. The following three (3) services should be in a running state.
  i) ISS
  ii) PMA
  iii) IVX-SIP
- Log in to the Avaya SIP Enablement Service (SES) Server via the Web browser. The registered users field under Users will also show all registered SIP users including all the registered Remote Extension users.
- Place call from CrystalVoice Remote Extension Client.

# 7. Support

For technical support on the CrystalVoice product line, contact CrystalVoice Communications at support@CrystalVoice.com or 1-805-889-4260

# 8. Conclusion

These Application Notes have described the administration steps required to support CrystalVoice Remote Extension on Avaya Communication Manager with the Avaya Enablement Service (SES) Server. With the exception of Unattended Transfer among SIP telephones, the CrystalVoice Remote Extension supported all basic and extended features that were tested and can interoperate successfully with Avaya Communication Manager SIP solution.

# 9. Additional References


[1] Administrator Guide for Avaya Communication Manager, Doc # 03-300509, Issue 1, June 2005
[2] Avaya Communication Manager Advanced Administration Quick Reference, Doc # 03-300364, Issue 2, June 2005 Release 3.0
[3] Expanded Meet-me Conference (EMMC) version 1.0 Installation and Troubleshooting Guide for the S8500, Doc # 04-300527, Issue 1, June 2005
[4] Avaya IA 770 INTUITY AUDIX Messaging Application, Doc # 585-313-159, Issue 4, December 2003
[5] Converged Communications Server Installation and Administration, Doc # 555-245-705, February, 2004
[6] Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0, Doc # 210-100-500, Issue 9, June 2005
[7] CrystalVoice Remote Extension for SIP Install Notes, Doc #5900-1045
[8] CrystalVoice Systems Manager Reference Guide, Doc #5900-1029


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

Product documentation for CrystalVoice products may be found at
http://www.crystalvoice.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 Developer*Connection* Program at devconnect@avaya.com.