



## Avaya Solution & Interoperability Test Lab

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# Application Notes for 911 ETC CrisisConnect<sup>®</sup> for Softphones and CrisisConnect<sup>®</sup> for VoIP with Avaya IP Office – Issue 1.0

### Abstract

These Application Notes describe the procedures for configuring the 911 ETC CrisisConnect<sup>®</sup> for Softphones and CrisisConnect<sup>®</sup> for VoIP with Avaya IP Office.

911 ETCs' CrisisConnect<sup>®</sup> for VoIP solution enables E911 call routing to the correct Public Safety Answering Point (PSAP) and delivers the caller's address directly to the PSAP operator's panel in order to provide immediate emergency assistance.

911 ETCs' CrisisConnect<sup>®</sup> for Softphones forces Avaya one-X<sup>®</sup> Communicator users to provision their current location. Location information provisioned by users was stored in the 911 ETC VoIP Positioning Center through the SoftLoc server for Automation Location Identification (ALI) use, if users were to make an Emergency Call.

Readers should pay attention to section 2, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Testing was performed using Avaya IP Office 500 V2 R9.1, but it also applies to Avaya IP Office Server Edition R9.1 (single site configuration only).

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

911 ETC provides a VoIP Positioning Center (VPC) Service that is able to deliver 911 calls to U.S. and Canada PSAPs independent of the region the call originates from; via a SIP trunk from Avaya IP Office (IP Office), 911 ETC provides SIP specifications for a primary and secondary Session Border Controller (SBC). 911 ETC configures SBC(s) for all customer SIP switches or SBCs that will be connecting to 911 ETC for E911 purposes.

CrisisConnect<sup>®</sup> for Softphones uses the 911 ETC VoIP Positioning Center (VPC) service to allow Avaya one-X<sup>®</sup> Communicator users to provision a location in near real-time. CrisisConnect<sup>®</sup> for VoIP is a required service. 911 ETC provides the SoftLoc server software and a distributable client software package to be installed on computers where the Avaya one-X<sup>®</sup> Communicator is installed. The suggested work flow for this solution is as follows:

911 ETC provides the SoftLoc Server software package along with requirements. 911 ETC will also aid in the installation and configuration. 911 ETC provides the SoftLoc Client software package. The software package can be distributed using most distribution methods that support MSI files (Active Directory Domain Policy, Windows scripting, etc.).

SoftLoc Client assists/requires users of soft phones to provision their current location to ensure accurate routing of an outgoing 911 call. It was developed because of concerns by 911 ETC's customers that soft phone users will ignore critical location information when logging onto their soft phones.

SoftLoc Client runs as a Windows system-tray application and quietly waits for the user to launch a configured soft phone application. Upon launch, SoftLoc will appear above all other applications and remind the user to provision an emergency location. Up to three frequently-used locations can be saved to the remote emergency server and quickly provisioned with just a few mouse clicks. If the user chooses not to provision an emergency location, the soft phone application will be forcibly closed. Responsibility, and therefore liability, is placed back upon the user and accurate location information is ensured in the event of an emergency.

## 2. General Test Approach and Test Results

The compliance test focused on verifying that 911 ETC CrisisConnect® for VoIP ability to route emergency call and 911 ETC CrisisConnect® for Softphone to update addresses.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Test Results

All test cases were successful.

### 2.2. Interoperability Compliance Testing

The compliance test validated the ability of 911 ETC CrisisConnect® for Softphone and CrisisConnect® for VoIP to update users' address information in near real time, route emergency calls and provide ALI information to the PSAP. Feature tests also included the following:

- Call setup using SIP (UDP).
- Codec verification using G.711.
- Call routing based on Locations configured in IP Office.
- Calls from Analog, Digital, Avaya one-X Communicator®, Avaya 1100 Series IP Endpoints, and Avaya 9600 Series IP Endpoints.
- Mis-provision of ANI in 911 ETC database, which resulted in call getting routed to Emergency Calls Relay Center (ECRC).
- Verification of alerts generated when dialing emergency number from all types of endpoints.

Failover tests were also performed for the cases where the SIP trunk to 911 ETC is down (SIP 408) and a negative response from 911 ETC (SIP 503), which resulted in alternate routing to secondary route.

For this test effort, only calls related to audio, and PSAP ALI, were placed by dialing 911. The rest of the test calls, due to the nature of emergency calling, were placed to 933. 933 is an Address Verification Service provided by 911 ETC.

### 2.3. Support

Technical support for 911 ETC can be obtained through the following:

- Web: <http://www.911etc.com/contact-us>
- E-mail: [support@911etc.com](mailto:support@911etc.com)
- Phone: (480) 719-8559

### 3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the 911 ETC CrisisConnect® for VoIP and 911 ETC SoftLoc Server and Client.

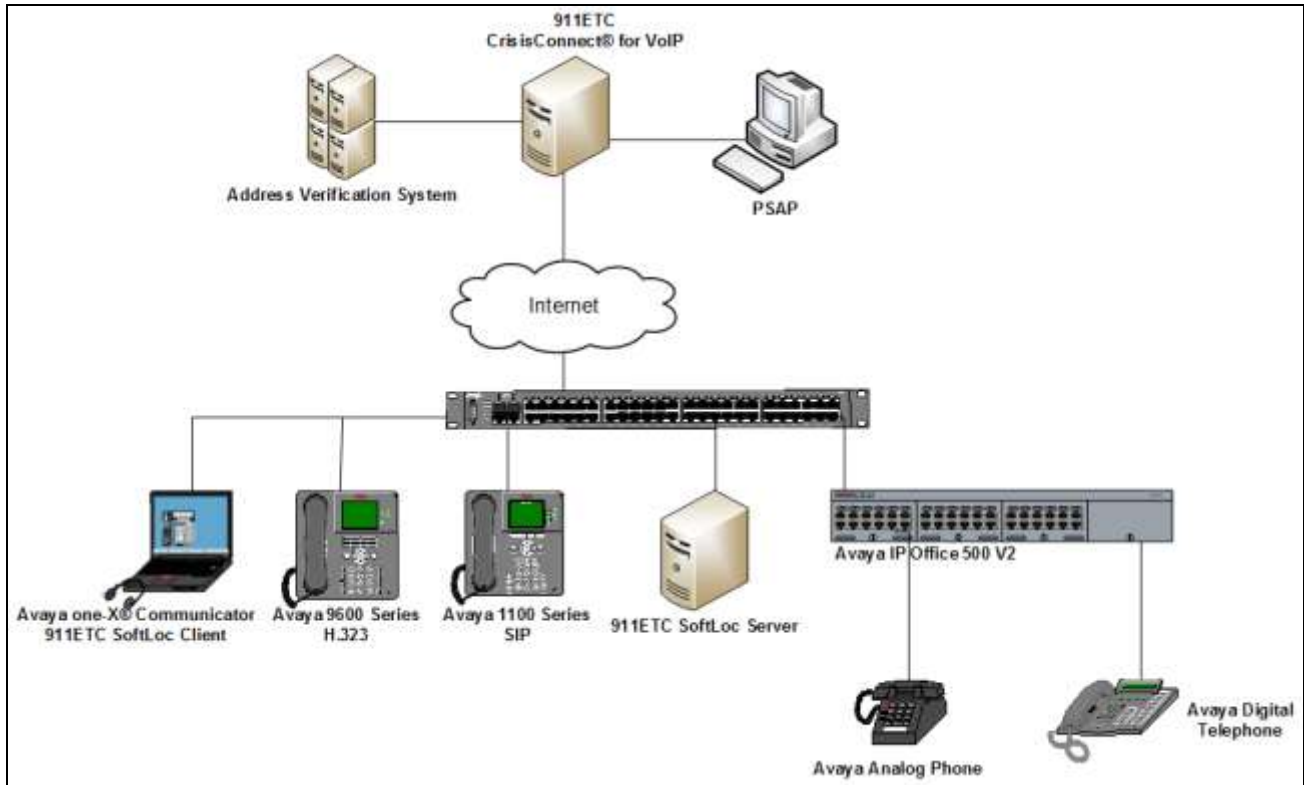


Figure 1: Reference Configuration

## 4. Equipment and Software Validated

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

<b>Equipment</b>	<b>Release</b>
Avaya IP Office 500 V2 with Digital Expansion Module	R9.1
Avaya IP Office Manager	9.1.0.437
Avaya 9600 Series IP Deskphone (H.323)	3.2.4
Avaya 1100 Series IP Deskphone (SIP)	4.4.18
Avaya one-X® Communicator	6.2001
Avaya 5420 Digital Telephone	N/A
Avaya 6211 Analog Telephone	N/A
911 ETC CrisisConnect®	5.2.3
911 ETC SoftLoc Server	2.0
911 ETC SoftLoc Client	2.0.5.0

## 5. Configure Avaya IP Office

This section describes Avaya IP Office configuration to support connectivity to the 911 ETC. Avaya IP Office is configured through the Avaya IP Office Manager, a PC desktop application. From a PC running the Avaya IP Office Manager application, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Navigate to **File** → **Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section, showing all the Avaya IP Office configurable components in a configuration tree in the left pane.

### 5.1. Licenses

From the configuration tree in the left pane, select **License**. Verify the **License Status** for **SIP Trunk Channels** are **Valid**.

The screenshot shows the Avaya IP Office Manager application window. The left pane displays a configuration tree with the following items:

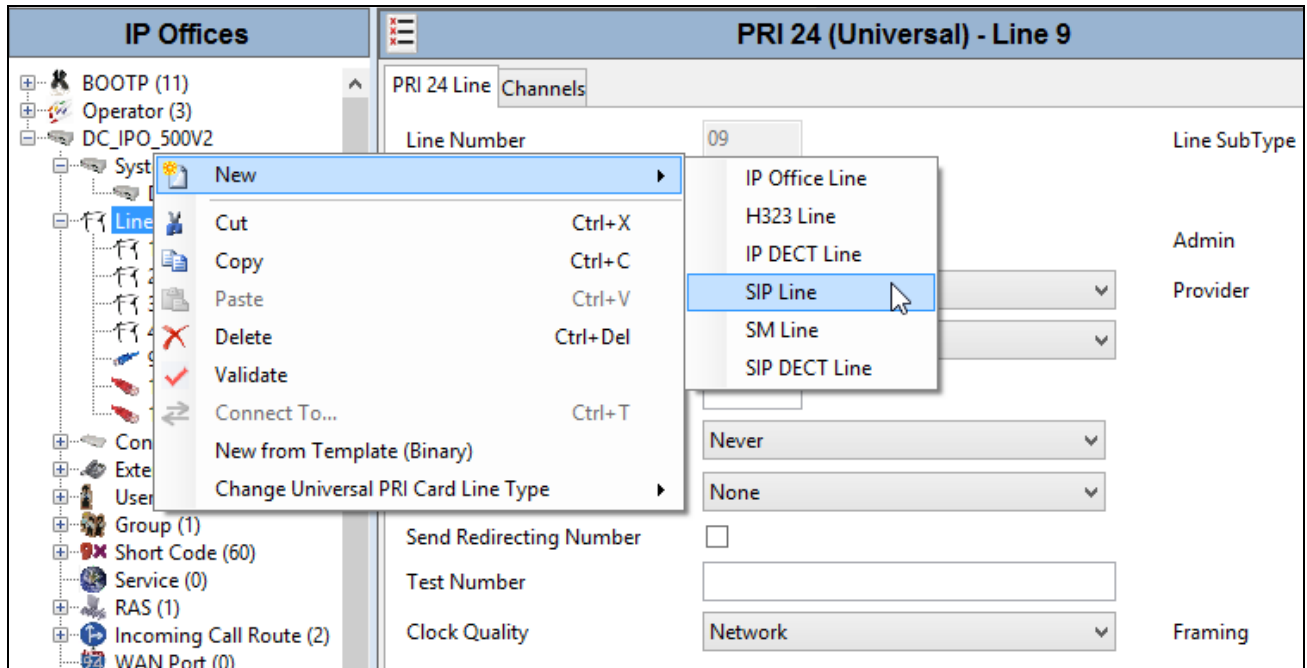
- BOOTP (11)
- Operator (3)
- DC\_IPO\_500V2
  - System (1)
    - DC\_IPO\_500V2
      - Line (7)
        - Control Unit (3)
        - Extension (19)
          - User (21)
          - Group (1)
          - Short Code (60)
          - Service (0)
          - RAS (1)
          - Incoming Call Route (2)
          - WAN Port (0)
          - Directory (0)
          - Time Profile (0)
          - Firewall Profile (1)
          - IP Route (3)
          - Account Code (0)
          - License (12)
          - Tunnel (0)
          - User Rights (9)
          - ARS (2)
          - Location (2)
          - Authorization Code (0)

The right pane shows the 'License Remote Server' configuration page. The 'PLDS Host ID' is 111308609925 and the 'PLDS File Status' is 'Not Present / Invalid'. A table lists various features and their license keys, instances, and status. The 'SIP Trunk Channels' feature is highlighted in blue and has a status of 'Valid'.

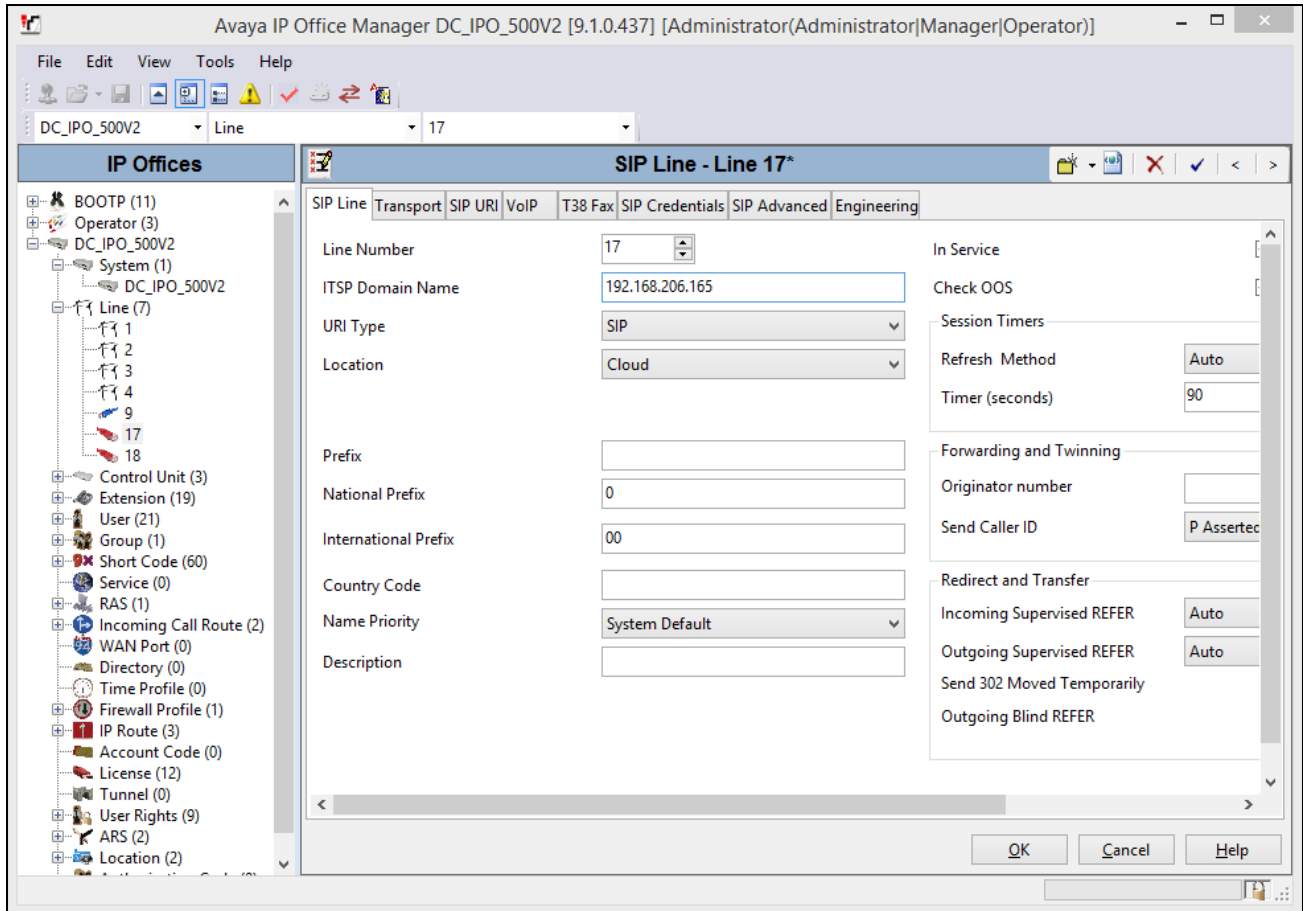
Feature	License Key	Instances	Status
IP500 Voice Networking Channels	y2la6e6bANpeUg16VrBaELrWkdm4gX_2	4	Valid
Small Site Software Upgrade 8 (R9.1)	WVJBHryCLS4dl0h1kr5dWt0SgYcjEluy	1	Valid
IP500 Universal PRI (Additional cha...	MbCK0N52MGFr4kLg3AWqJXV@hpSKEZa	8	Valid
SIP Trunk Channels	JyDNF3@PAs8W2hJbRLYspI54o@EbGDjO	10	Valid
Avaya IP endpoints	vb15brylDNG3vWQdzXYR7W6U2Bc1j60n	20	Valid
3rd Party IP Endpoints	_jNhkcoQdDgG0io00Dkb7_vxoOZjSmF2	20	Valid
Mobile Worker	zdHjaqdm5tZQIKjaMoRgPr6c247hjiK	20	Valid
Office Worker	usnAT0gL9SpFh5@V0eYtey3ABq_aGcZD	20	Valid
Power User	nNnmbCgZLSUG2mNOOpRipFGTHAZLj7NN	20	Valid
Teleworker	mD9dyg5VXv@JX1leEsFTQS_yacrusQbl	20	Valid
Essential Edition	RITp@7tmSKXBh3Nun2EcBYjOUaSJSPLC	255	Valid
R8+ Preferred Edition (VM Pro)	U31Ynh@ySINZjLyzyEEQg5_2jmE0LREM	255	Valid

## 5.2. Administer SIP Line

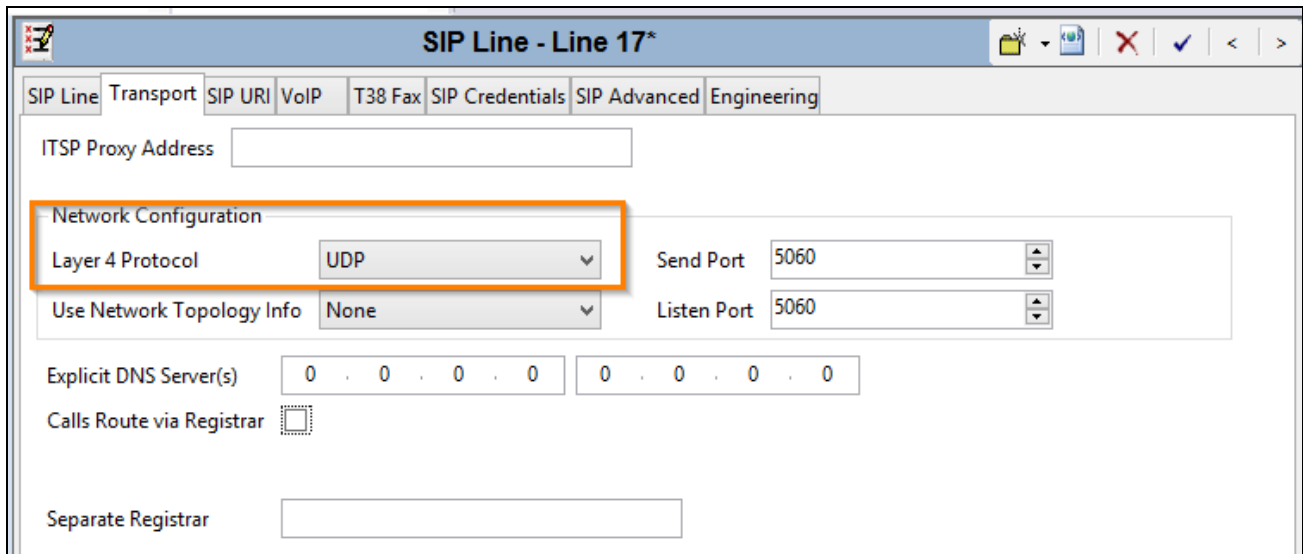
From the configuration tree in the left pane, select **Line**. Right click on **Line** → **New** → **SIP Line**.



In the **ITSP Domain Name**, type in the IP Address of 911 ETC SBC.



Select **Transport** tab and set **Layer 4 Protocol** to **UDP**



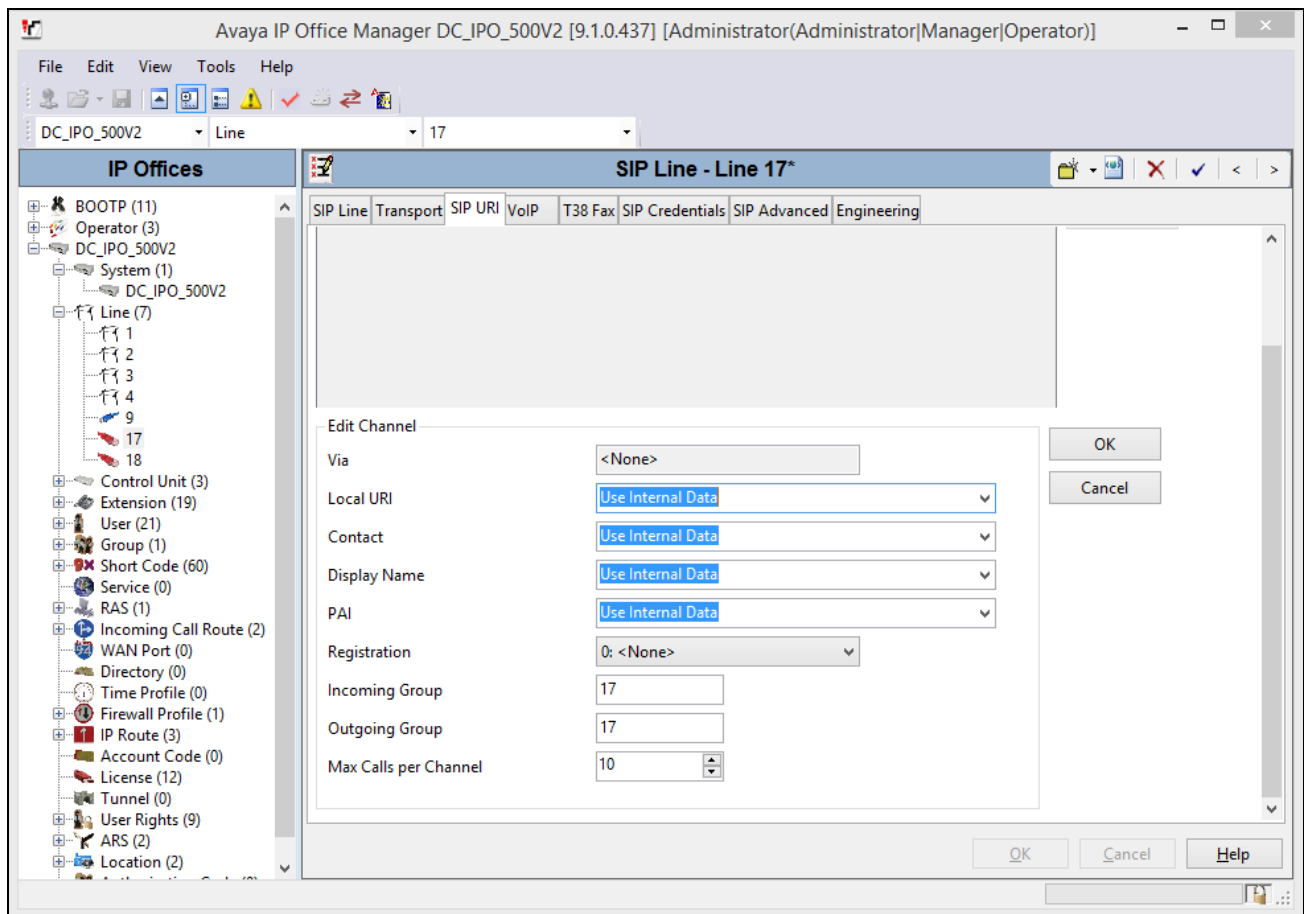


Select **SIP URI** tab and click **Add**.

- Type in the SIP Line number of the line that is being added in **Incoming Group** and **Outgoing Group**, i.e. 17 in this case.
- Type in a value in **Max Calls per Channel**.
- Select **OK**.

At the bottom of the window select **OK** to save configuration.

For Compliance, another SIP line – Line 18 was added for failover testing. Repeat this section to add another SIP Line.

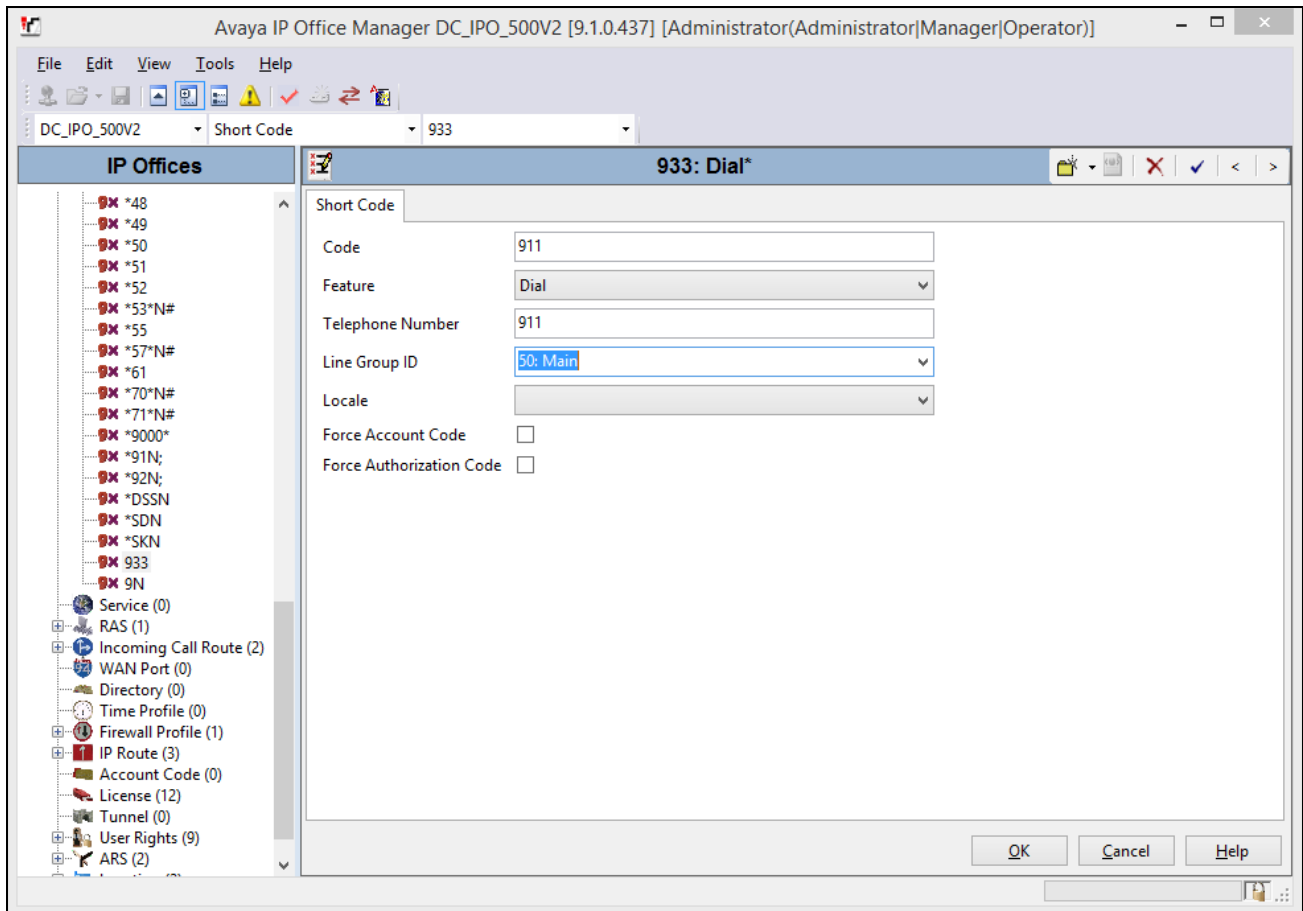


### 5.3. Administer System Short Code For 911

In times of emergency, users will expect to dial a well known number to contact emergency services. In the United States, 911 is used for this purpose.

From the configuration tree in the left pane, right-click on **Short Code** and select **New** to add a new short code. In the right pane that appears, configure the following:

- In the **Code** field, enter the dial string which will trigger this short code. In this case, **911**.
- Set the **Feature** field to **Dial** since the purpose of this short code is to dial a number.
- In the **Telephone Number** field, enter the number the system should dial when the user dials 911.
- Set the **Line Group Id** select ARS route that will be used to route 911 calls.



## 5.4. Administer ARS Routing for 911 Calls

Before configuring the primary route, create a failover route. From the configuration tree on the left pane, right-click on **ARS** and select **New**.

- In the **Route Name** field, type in a name, i.e Failover.
- Edit the short code for **911**, by double clicking on it. In the **Telephone Number** field, type in **911**.
- Select a SIP line that was added as a secondary route, **Line Group ID 18**

The screenshot shows the 'Failover' configuration window in a software interface. The window title is 'Failover'. The main configuration area includes fields for 'Route Name' (set to 'Failover'), 'Dial Delay Time' (set to 'System Default (4)'), and 'Description'. There is a 'System Tone' dropdown menu and a 'Check User Call Barring' checkbox. Below these are 'In Service' and 'Out of Service Route' sections, both with '<None>' dropdowns. A 'Time Profile' section is also visible. An 'Edit Short Code' dialog box is overlaid on the main window, showing fields for 'Code' (911), 'Feature' (Dial Emergency), 'Telephone Number' (911), 'Line Group ID' (18), and 'Locale'. There are 'OK' and 'Cancel' buttons in the dialog box. At the bottom of the main window are 'OK', 'Cancel', and 'Help' buttons.

Now configure the primary route and associate failover route to the **Main** ARS. From the configuration tree on the left pane, select **ARS** → **Main**. Select **Alternate Route** as Failover. Edit the short code for **911**, by double clicking on it; in the **Telephone Number** field, type in **911** and set **Line Group ID** to primary SIP Line.

Also please note that a code of **11** was also added for access to emergency calls.

ARS

ARS Route Id: 50

Route Name: Main

Dial Delay Time: System Default (4)

Description:

In Service:  → Out of Service Route: <None>

Time Profile: <None> → Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
11	911	Dial Emergency	17
933	933	Dial Emergency	17
0N;	0N	Dial 3K1	9
1N;	1N	Dial 3K1	9
XN;	N	Dial 3K1	9
XXXXXXXXXXN	N	Dial 3K1	9
911	911	Dial Emergency	17

Buttons: Add..., Remove, Edit...

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

Alternate Route: 51: Failover

Buttons: OK, Cancel, Help

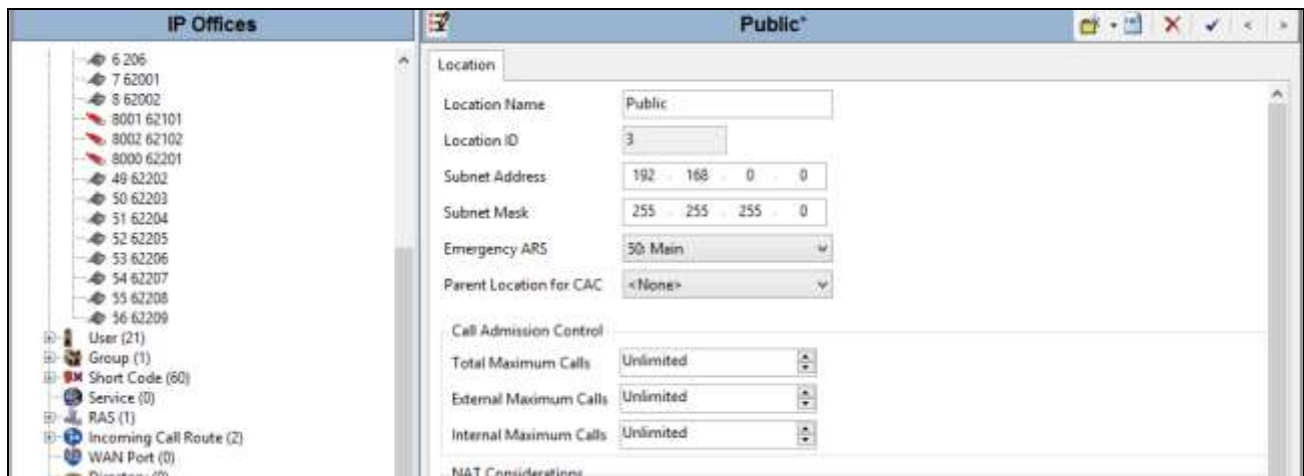
## 5.5. Configure Locations

From the configuration tree on the left, select **Location**. Right click **Location** and select **New** to add a new location, (not shown). Configure the **Subnet Address** and **Subnet Mask** of the network region where the phones will reside. Select **Emergency ARS** of **Main** as configured in **Section 5.4**.

Configuring locations allows you to specify named locations for groups of phones, IP Office systems, or IP Trunks. The IP Office system must also be assigned a location. Multiple systems in a Small Community Network (SCN) or Server Edition group of systems may reside in the same location. In an SCN environment, locations must be configured at the top level and therefore, all systems must be configured with the same settings, except when the emergency ARS needs to be set at the system level.

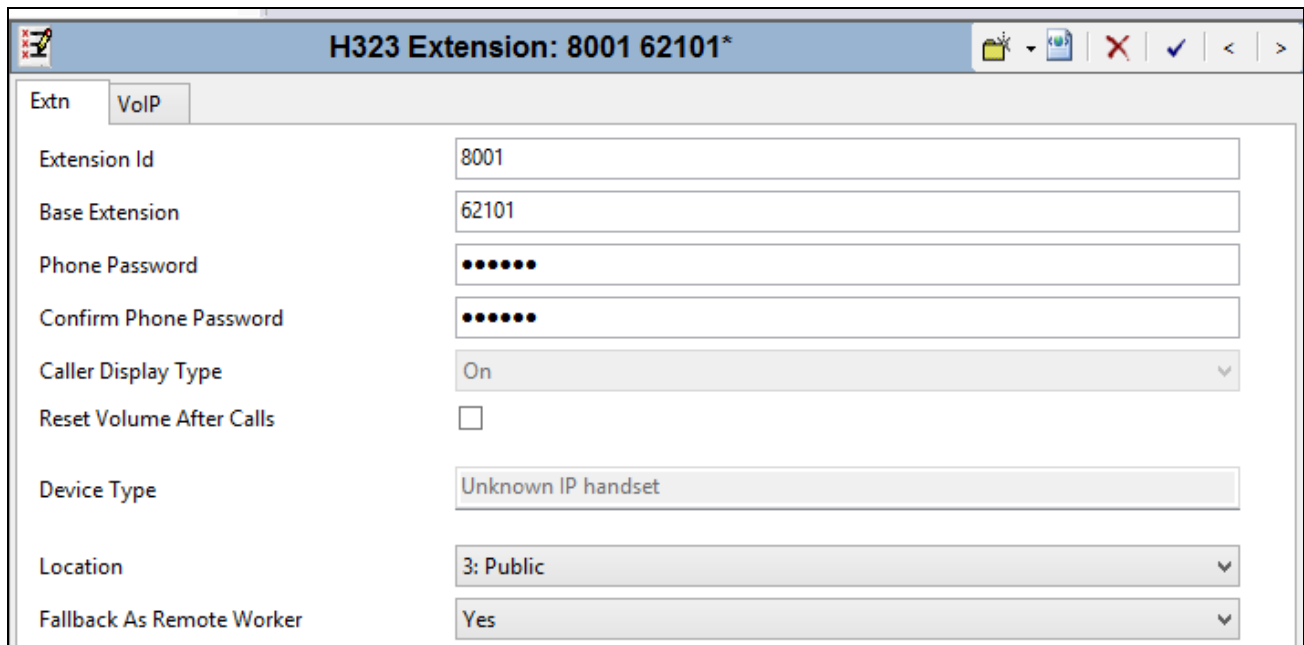
Once locations have been defined, extensions can be allocated to them in the extension configuration. IP phones can be identified by the IP address that they register from. Each location can have only one subnet defined, but phones outside that subnet can be explicitly assigned that location. During compliance testing, extensions were configured to use the location as mentioned in this section.

For more information regarding locations, please refer to the **Help** section.



## 5.6. Configure Extensions

From the configuration tree on the left, select **Extension**. Select an extension and under the **Extn** tab, select the location configured in previous section from the **Location** drop down menu.

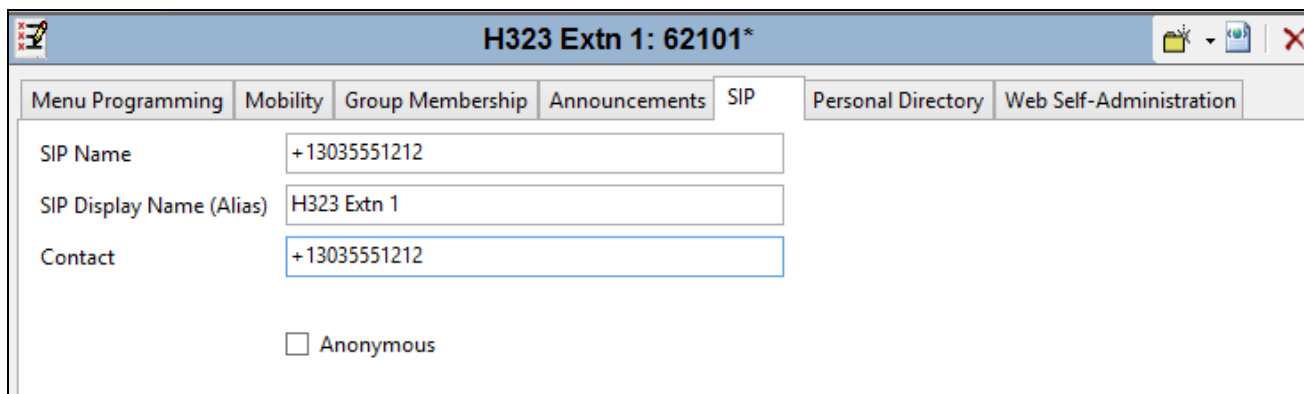


The screenshot shows the configuration page for an H323 Extension. The title bar reads "H323 Extension: 8001 62101\*". Below the title bar, there are two tabs: "Extn" (selected) and "VoIP". The configuration fields are as follows:

Extension Id	8001
Base Extension	62101
Phone Password	••••••
Confirm Phone Password	••••••
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Unknown IP handset
Location	3: Public
Fallback As Remote Worker	Yes

## 5.7. Configure User

From the configuration tree on the left, select **User**. Select a user and click **SIP** tab. Type in the a 10 digit number in +CCNPANXXXXX format in **SIP Name** and **Contact** fields. Type in a name in **SIP Display Name (Alias)**. Please note that the number configured in SIP Name and Contact will be used by 911 ETC to provision a location against it.



The screenshot shows the configuration page for H323 Extn 1: 62101\*. The title bar reads "H323 Extn 1: 62101\*". Below the title bar, there are several tabs: "Menu Programming", "Mobility", "Group Membership", "Announcements", "SIP" (selected), "Personal Directory", and "Web Self-Administration". The configuration fields are as follows:

SIP Name	+13035551212
SIP Display Name (Alias)	H323 Extn 1
Contact	+13035551212

Anonymous

## 5.8. Save Configuration

Navigate to **File** → **Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

## 6. Configure 911 ETC CrisisConnect® for VoIP

The customer and 911 ETC need to exchange SIP peering information. 911 ETC will configure their Session Border Controllers based on peering information provided by the customer. 911 ETC can provide dashboard access to the customer on request. Data needs to be provisioned prior to testing. Below are the steps to provision data via 911 ETC dashboard. The 911 ETC dashboard is accessed via a browser. For security purposes, the web address is not shown in the configuration below.

1. 911 ETC will setup customer and dashboard.
2. Configure endpoint: Select **Endpoints** → **Create Endpoint**; Type in **Telephone No** and **Caller Name** and click **Save and Add Address**.

The screenshot shows a web application interface with a navigation bar at the top containing tabs: Customer Management, User Management, Dashboard, SIP Peer, User Request, Endpoints, Notification, Batches, Summary, and Reports. The 'Endpoints' tab is selected, and a dropdown menu is open, showing options: Create Endpoint, List/Edit Endpoint, and Delete Endpoint. Below the navigation bar, the breadcrumb 'Endpoints > Create Endpoint' is visible. The main content area is titled 'Create Endpoint' and contains a sub-header 'Create new endpoint on selected dashboard'. The form includes a 'Dashboard Name' dropdown menu set to 'Demo', a 'Telephone No \*' field with a '1-' prefix, and a 'Caller Name \*' field. At the bottom of the form are two buttons: 'Save' and 'Save and Add Address'.

3. Enter **Address Line1** and **Address Line2**, **Community**, **State** and **Postal Code** and click **Submit**.

Note: Address Line2 contains all the additional information pertaining to an address, i.e., Suite 109. Address Line2 is an optional parameter.

Customer Management	User Management	Dashboard	SIP Peer	User Request	Endpoints	Notification	Batches	Summary	Reports
Endpoints > Endpoint Detail > Create Address									
<b>Create Address</b>									
Address for Endpoint (Telephone No: 1-562-985-4333, Caller Name: TEST)									
Address Line1 *	15655 W Roosevelt St *								
Address Line2	Suite# 109								
Community *	GOODYEAR								
State *	ARIZONA								
Postal Code *	85338								
<input type="button" value="Submit"/> <input type="button" value="Cancel"/>									

- In order to create a recipient for Text and Email notification, select **Notification** → **Create Recipient**. Provision **First and Last Name, Email, Notification Type, Mobile Number and Carrier**.

Customer Management	User Management	Dashboard	SIP Peer	User Request	Endpoints	Notification	Batches	Summary	Reports
Notification > Edit Recipient						<ul style="list-style-type: none"> <li>Create Recipient</li> <li>Manage Recipient</li> <li>Configure Endpoints</li> <li>Delete Recipient</li> </ul>			
<b>Edit Recipient</b>									
Recipient Details									
First Name *	[Redacted]								
Last Name	[Redacted]								
Email *	[Redacted]								
Notification Type	<input type="checkbox"/> Network <input checked="" type="checkbox"/> Emergency (911) Calls <input checked="" type="checkbox"/> Test (933) Calls <input type="checkbox"/> Unprovisioned Calls <input type="checkbox"/> Dashboard								
Mobile Number	[Redacted]								
Carrier	[Redacted]								

**Note:**

- Notifications may be truncated when using SMS as carriers generally limit SMS messages to 160 characters. If possible, select an MMS enabled carrier.
- SMS and MMS notifications make use of the carrier's email-to-SMS gateway. Carriers may limit usage or place other restrictions on messages.



- Carriers may apply a fee for received SMS/MMS messages. Consult your carrier for fees associated with received SMS/MMS messages.

5. To link a recipient to specific endpoints in the dashboard, so that the recipient receives notifications only when specific endpoints makes an emergency call, select **Notification** → **Configure Endpoints** and then click **Link** at the bottom.



6. Select the endpoints that need to be configured for receiving notifications; click **Save**.

**Note:** If the recipient is not linked to an endpoint or endpoints, it will receive notification for every endpoint in the dashboard that makes an emergency call.

Customer Management | User Management | Dashboard | SIP Peer | User Request | Endpoints | Notification | Batches | Summary | Reports

Notification > Configure Recipient > Link Endpoints

### Link Endpoints

Recipient Name:

**Search Criteria:**

Telephone No:  Caller Name:  Status Type: All

**Endpoints List:**

<input type="checkbox"/> Select All	Telephone Number	Caller Name	Status Type
<input type="checkbox"/>	[REDACTED]	[REDACTED]	PROVISIONED
<input type="checkbox"/>	[REDACTED]	[REDACTED]	PROVISIONED
<input type="checkbox"/>	[REDACTED]	[REDACTED]	PROVISIONED
<input checked="" type="checkbox"/>	[REDACTED]	[REDACTED]	PROVISIONED
<input checked="" type="checkbox"/>	[REDACTED]	[REDACTED]	PROVISIONED

7. Select all the endpoints and click **Link** at the bottom.

Customer Management | User Management | Dashboard | SIP Peer | User Request | Endpoints | Notification | Batches | Summary | Reports

Notification > Configure Recipient

### Configure Endpoints with Recipient

**Search Criteria:**

Recipient List:


**Endpoints linked to recipient:**

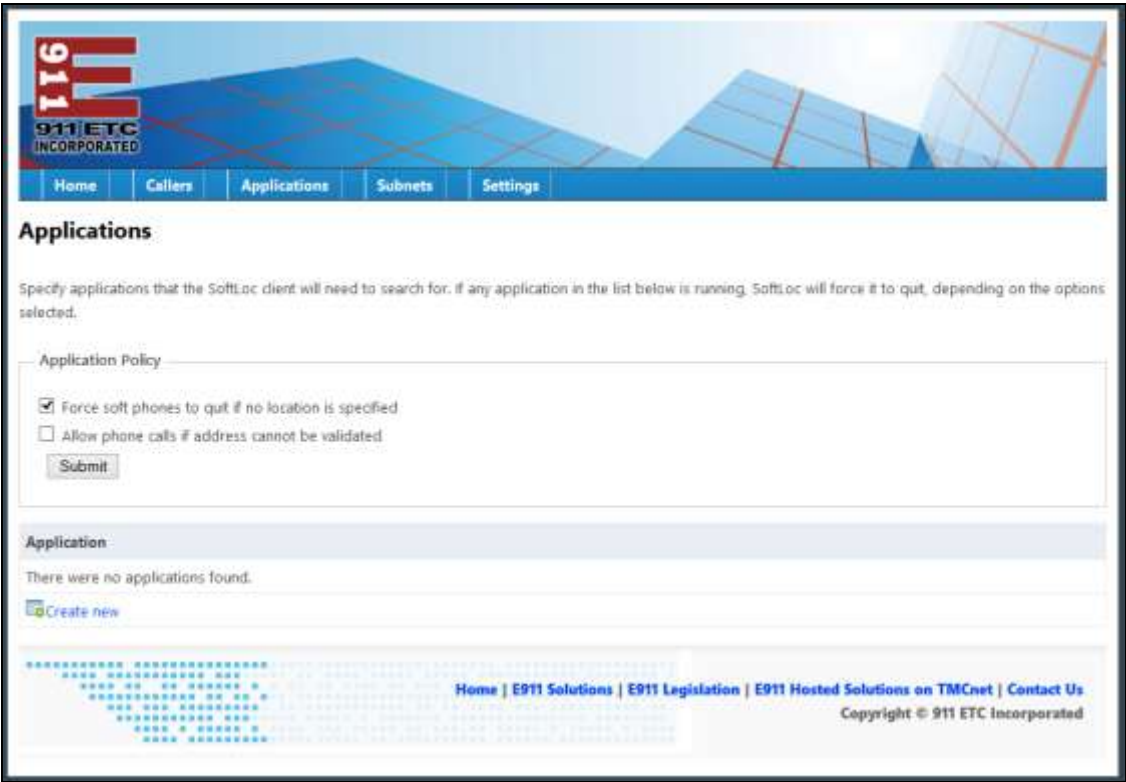

<input checked="" type="checkbox"/> Select All	Telephone Number	Caller Name	Status Type
<input checked="" type="checkbox"/>	[Redacted]	[Redacted]	PROVISIONED
<input checked="" type="checkbox"/>	[Redacted]	[Redacted]	PROVISIONED

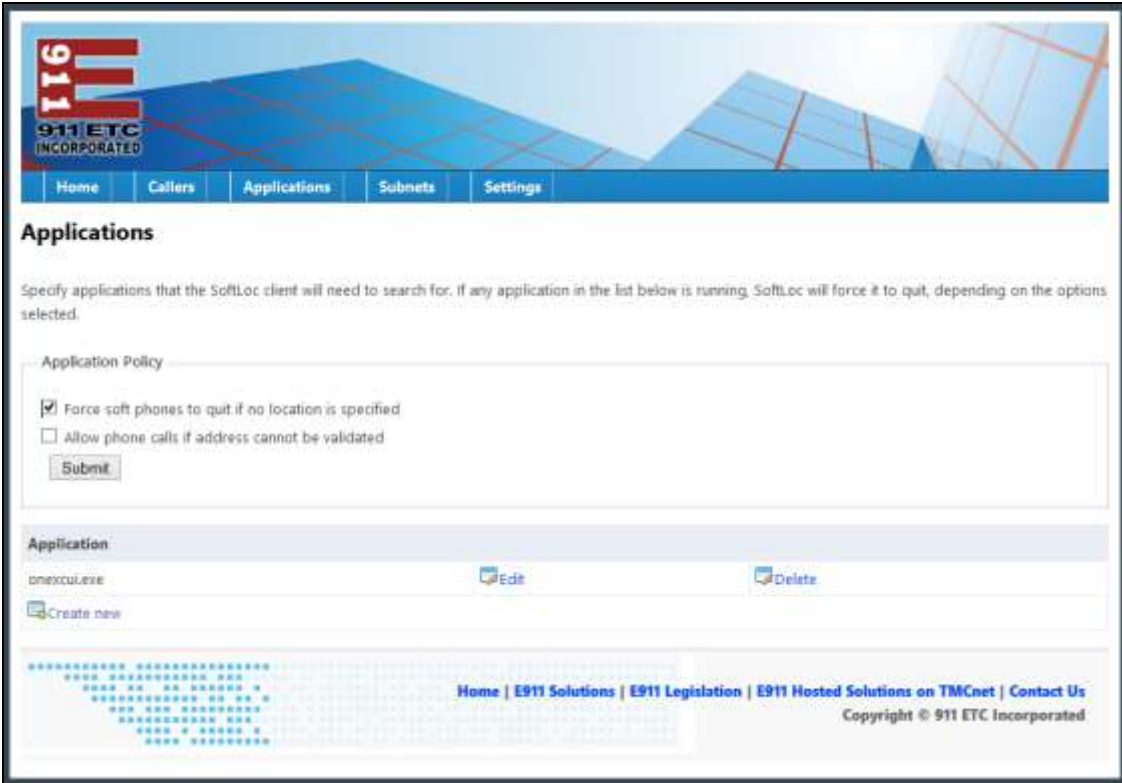
## 7. Configure 911 ETC CrisisConnect<sup>®</sup> for SoftPhones

### 7.1.1. Configure SoftLoc Server


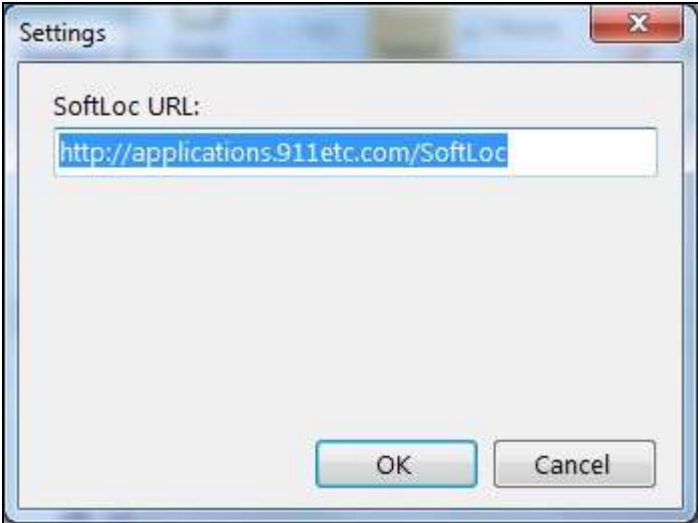
Step	Description
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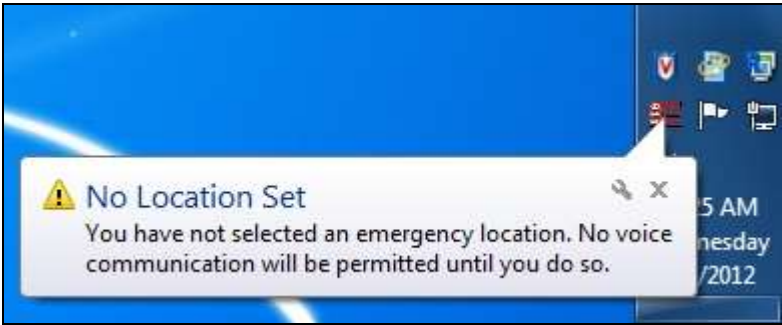
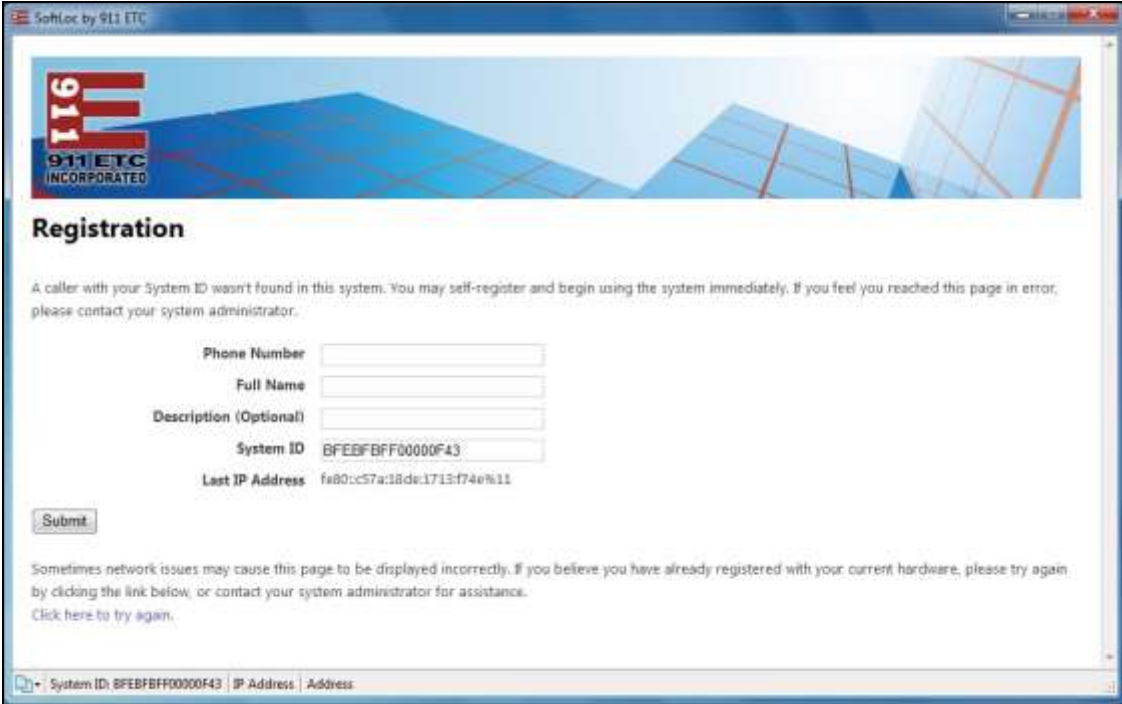
Step	Description
1.	<p>SoftLoc server is configured using browser. Enter the URL of SoftLoc server such as <a href="http://&lt;hostname&gt;/SoftLoc">http://&lt;hostname&gt;/SoftLoc</a> where &lt;hostname&gt; is the IP address or qualified domain name of the SoftLoc server. Login using appropriate credentials.</p>  <p>The screenshot shows the web application interface for 'SoftLoc for Soft Phones'. At the top left is the 911 ETC INCORPORATED logo. A navigation menu includes 'Home', 'Callers', 'Applications', 'Subnets', and 'Settings'. The main content area is titled 'SoftLoc for Soft Phones' and contains two paragraphs of text explaining the application's purpose and functionality. At the bottom, there is a footer with navigation links: 'Home   E911 Solutions   E911 Legislation   E911 Hosted Solutions on TMCnet   Contact Us' and a copyright notice: 'Copyright © 911 ETC Incorporated'.</p>

Step	Description
2.	<p>Click on the <b>Applications</b> tab, and ensure that <b>Force soft phone to quit if no location is specified</b> box is checked.</p> 
3.	<p>On the <b>Applications</b> page, click on <b>Create new</b></p> <ul style="list-style-type: none"> <li>Type in <b>onexcul.exe</b> and click on <b>Create new</b></li> </ul> 

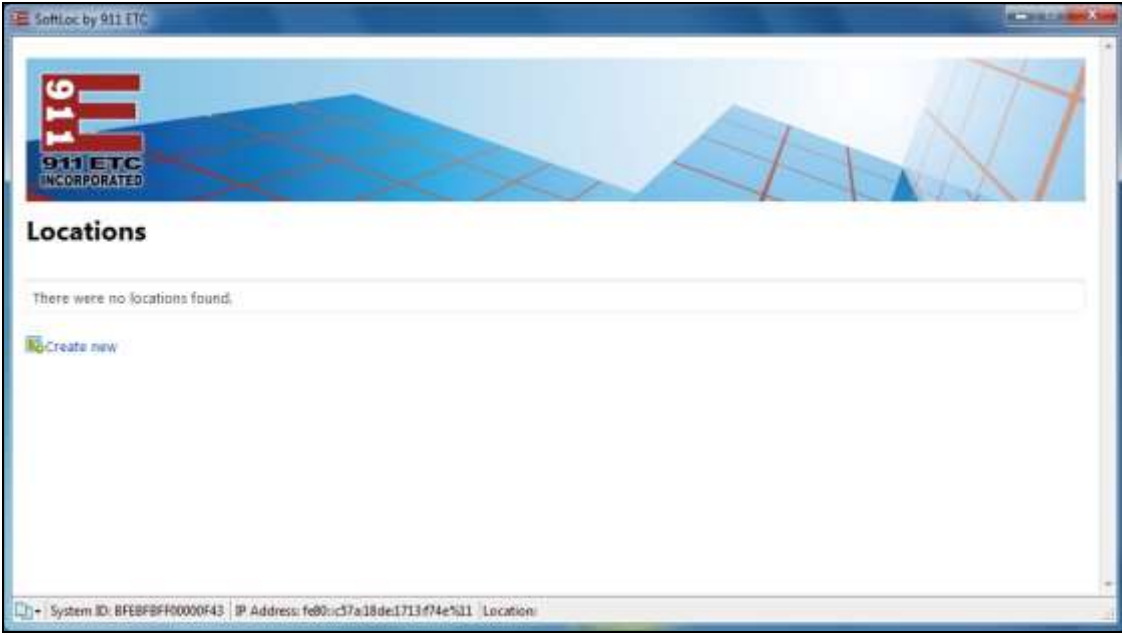
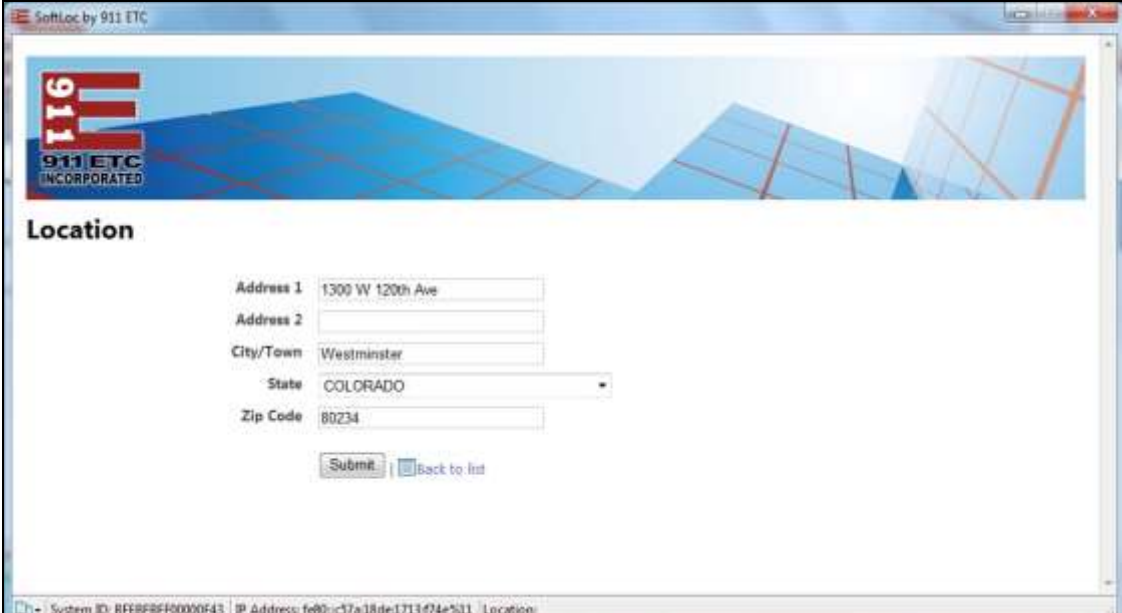
Step	Description
4.	<p>Newly added Application will show on the <b>Application</b> page</p> 

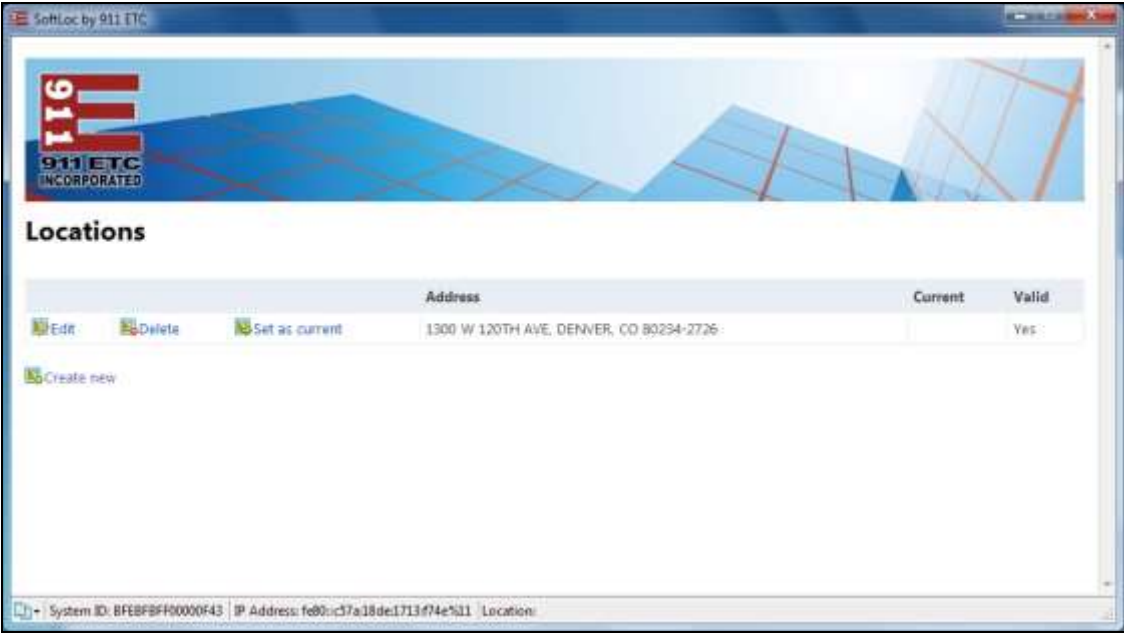
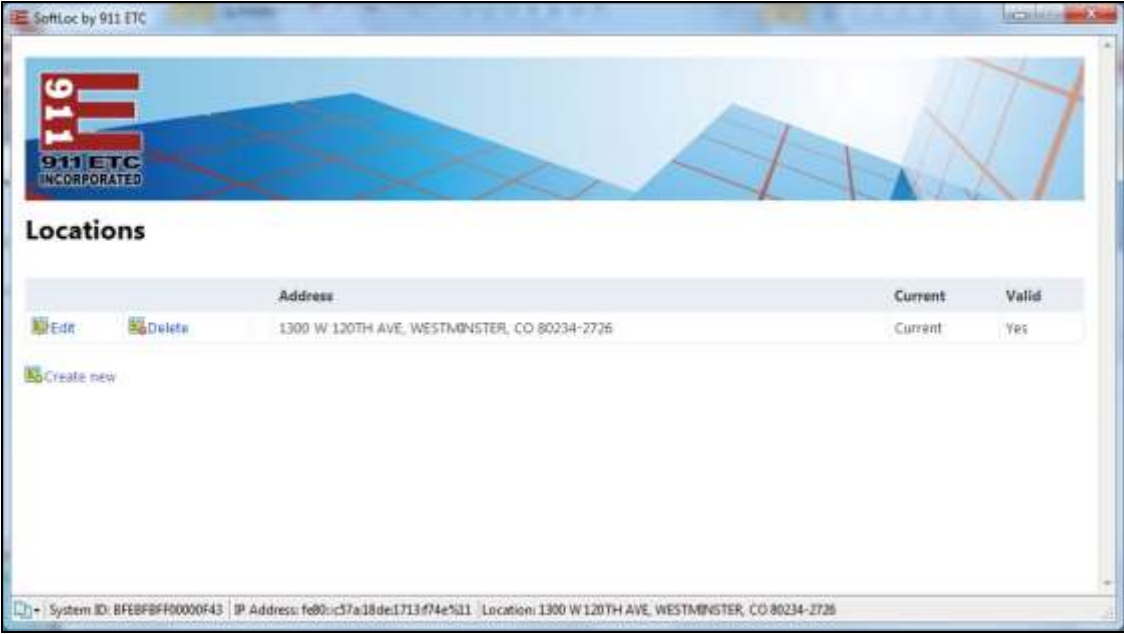
## 7.2. Configure SoftLoc Client

Step	Description
1.	<p>After a SoftLoc Client is installed on a workstation that has Avaya one-X<sup>®</sup> Communicator client installed, 911 ETC icon will appear in the task bar area of Windows desktop</p> <ul style="list-style-type: none"><li>• Right click on the icon, and click on settings</li></ul> 
2.	<p>A pop up window will appear; type in the URL of SoftLoc server. E.g. <code>http://&lt;hostname&gt;</code> where &lt;hostname&gt; is the IP address or qualified domain name of the SoftLoc server</p> 

Step	Description
3.	<p>A notification will pop up in the notification area of windows desktop, alerting user that a Location needs to be set. Click on the Notification.</p> 
4.	<p>A pop up window with a Registration page will appear, prompting the user to register. Fill in the registration information and submit.</p> 



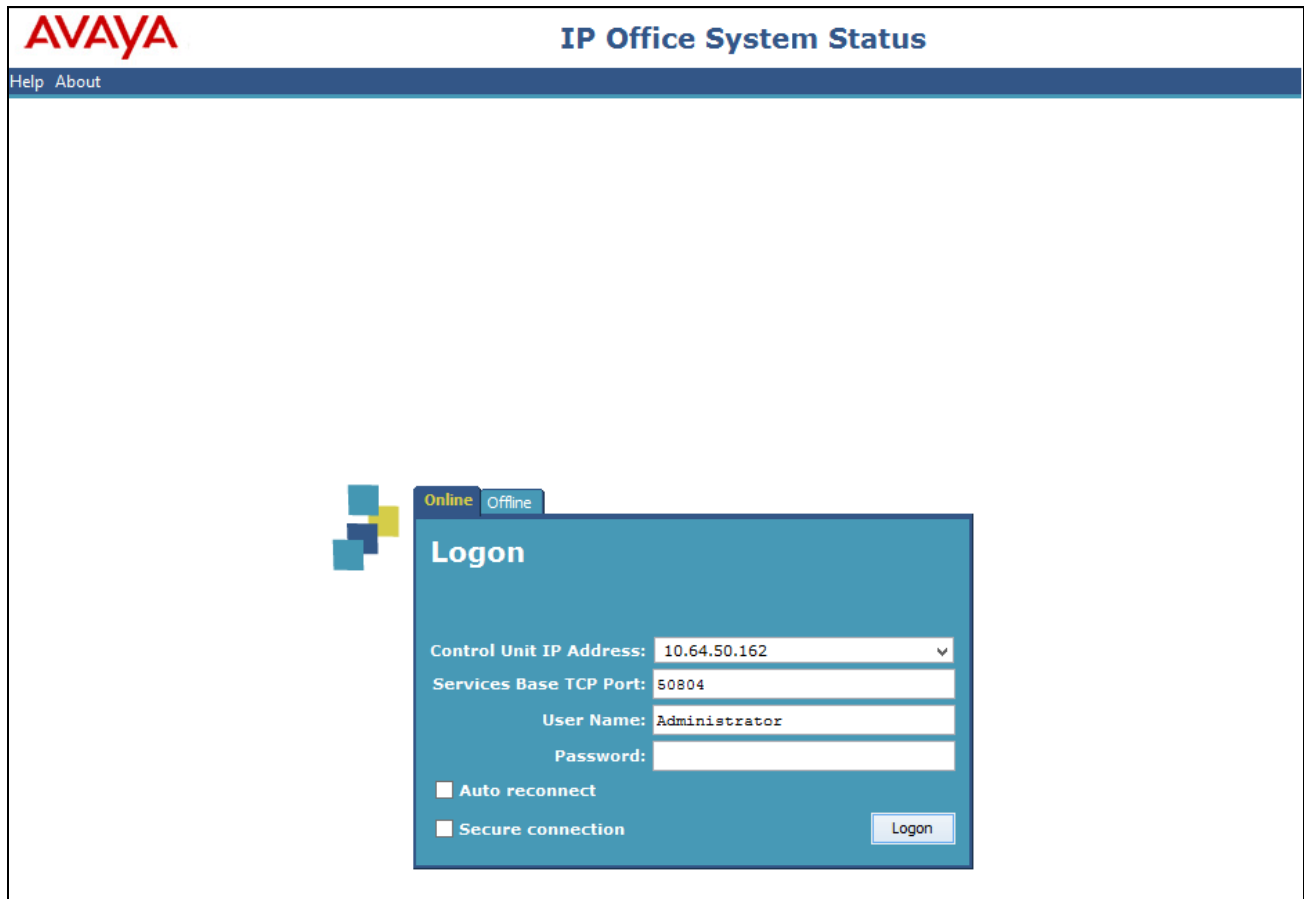
Step	Description
5.	<p>After registration is completed, the Locations page is displayed.</p> 
6.	<p>Click on Create new and fill in users' address information. Submit once done.</p> 

Step	Description
7.	<p>Users' address will now be displayed in Locations page.</p> 
8.	<p>Click on <b>Set as current</b> to make the address as user's current address. <b>Current</b> will show up under <b>Current</b> column confirming that the address has been set as user's current address. A user can add up to 3 addresses.</p> 

## 8. Verification Steps

The following steps may be used to verify the configuration:

- From a web browser go to <http://<IP Office IP Address>/> and select **System Status** (not shown).
- Fill-in Login information and click **Logon**.




The screenshot displays the Avaya IP Office System Status web interface. At the top left is the Avaya logo, and at the top right is the title "IP Office System Status". Below the title is a navigation bar with "Help" and "About" links. The main content area is mostly blank, with a "Logon" dialog box overlaid in the center. The dialog box has a blue header with "Logon" text and two tabs: "Online" (selected) and "Offline". Below the header are four input fields: "Control Unit IP Address" with the value "10.64.50.162", "Services Base TCP Port" with the value "50804", "User Name" with the value "Administrator", and "Password" which is empty. At the bottom of the dialog box are two checkboxes: "Auto reconnect" and "Secure connection", both of which are unchecked. A "Logon" button is located at the bottom right of the dialog box.

To verify the connectivity to 911 ETC for SIP lines added in this document, navigate to **Trunks** → **Line n**, where **n** is the SIP line number that was configured in this document. Verify the **Current State** for all channels is **Idle**.

**Status** Utilization Summary Alarms

### SIP Trunk Summary

Line Service State: In Service  
Peer Domain Name:  
Resolved Address:  
Line Number: 17  
Number of Administered Channels: 10  
Number of Channels in Use: 0  
Administered Compression: G711 A  
Enable Faststart: Off  
Silence Suppression: Off  
Media Stream: RTP  
Layer 4 Protocol: UDP  
SIP Trunk Channel Licenses: 10  0%  
SIP Trunk Channel Licenses in Use: 0  
SIP Device Features: REFER (Incoming and Outgoing), UPDATE (Incoming and Outgoing)

Channel Number	URI G...	Call Ref	Current State	Time in State	Remote Media A...	Co...	Conne...	Caller ID or Dial...	Other Party on Call	Direction of Call	Round Trip D...	Receive Jitter	Receive Packe...	Transmit Jitter	Transmit Packe...
1			Idle	1 day ...											
2			Idle	1 day ...											
3			Idle	1 day ...											
4			Idle	1 day ...											
5			Idle	1 day ...											
6			Idle	1 day ...											
7			Idle	1 day ...											
8			Idle	1 day ...											
9			Idle	1 day ...											
10			Idle	1 day ...											

Once 911 CrisisConnect for VoIP is configured place a test call. Verify that an email or SMS notification is received. Below are the screen captures of Email and SMS notifications.

**Email:**

**911/933 Call Notification**

An emergency call has occurred and you are registered to receive notifications.

Call details:

Subscriber Name: **Keyur Amin**  
Location: **12121 GRANT ST, RM 205, THORNTON, CO 80241**  
Telephone: **13035380123**  
Call Start Time: **6/17/2015 1:50:08 PM MST**  
Call Status: **Started**

Location information was retrieved from the 'Avaya IP Office-IVT' dashboard.

If you believe this notification is in error, please contact customer service at (480)719-8556 or by email at [customerservice@911etc.com](mailto:customerservice@911etc.com) so that we can assist.

Thank you,  
Customer Service  
911 Emergency Telecom Company  
(480)719-8556  
[customerservice@911etc.com](mailto:customerservice@911etc.com)

**SMS:**

911 Emergency Call Notification  
Subscriber Name: Keyur Amin  
Location: 12121 Grant St, RM 205, Thornton, CO 80241  
Telephone: 13035380123  
Call Start Time: 6/17/2015 1:50:08 PM  
Call Status: Started

## 9. Conclusion

911 ETC's CrisisConnect® successfully completed compliance testing. These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the 911 ETC CrisisConnect® as shown in **Figure 1**.

## 10. Additional References

Product documentation for Avaya IP Office may be obtained via the following link.

<http://marketingtools.avaya.com/knowledgebase>

Product documentation for the CrisisConnect® is available from 911 ETC.

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