Spectrum Enterprise SIP Trunking Service
Avaya IPO10
IP PBX Configuration Guide

About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Charter or Charter Business. All references to Charter should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise

Last Update: 2017-06-14

Avaya IPO10 IP PBX Configuration Document

1 Introduction

These Application Notes describe the steps necessary for configuring Session Initiation Protocol (SIP) Trunking service between the legacy Charter Communications Platform and an Avaya SIP-enabled enterprise solution. It does not provide any information how to provision, configure or use the features of the switch. Please refer to the documentation provided with the IP PBX or contact the vendor.

1.1 Service Limitation

Charter Business has conducted thorough testing of the Avaya IP PBX and has determined that the combination of Charter Business SIP Trunks and the Avaya IP-PBX **DOES NOT** support consistent fax receipt or transmission. The customer should make alternative service arrangements in order to support their faxing needs.

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2 Configure IP Office

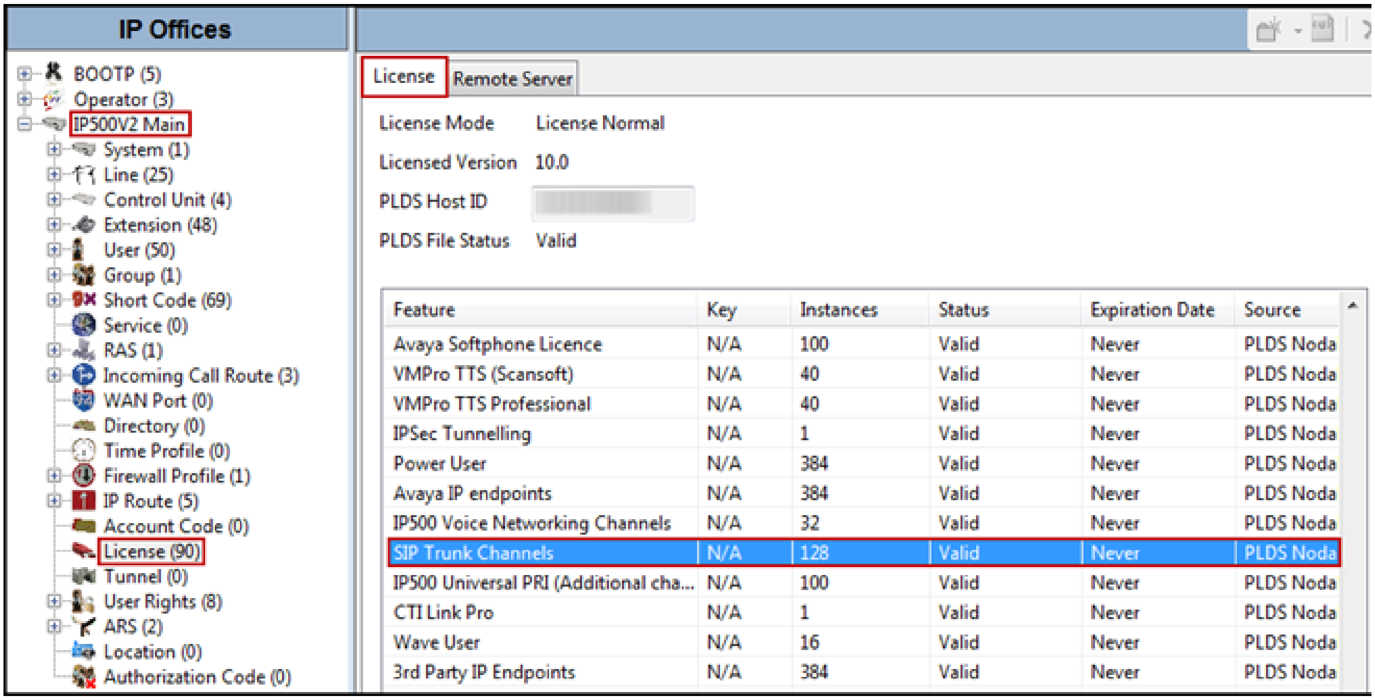
This section describes the IP Office configuration required to interwork with Charter SIP Trunking service. IP Office is configured through Avaya IP Office Manager (IP Office Manager) which is a PC application.

Step	Action	Result
1	Navigate to **Start > Programs > IP Office > Manager**	Launch IP Office Manager
2	Navigate to **File > Open Configuration**	Pop-up Window Appears
3	Select The Proper IP Office	
4	Log In With The Appropriate Credentials	Management Window Appears
5	Go To Next Table	

2.1 Licensing

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative

Step	Action	Result
6	Click: **License** in the **Navigation Pane** and **SIP Trunk Channels** in the **Group Pane**	
7	Confirm There Is A Valid License With Sufficient Instances (Trunk Channels) in the **Details Pane**	
8	Procedure Complete	



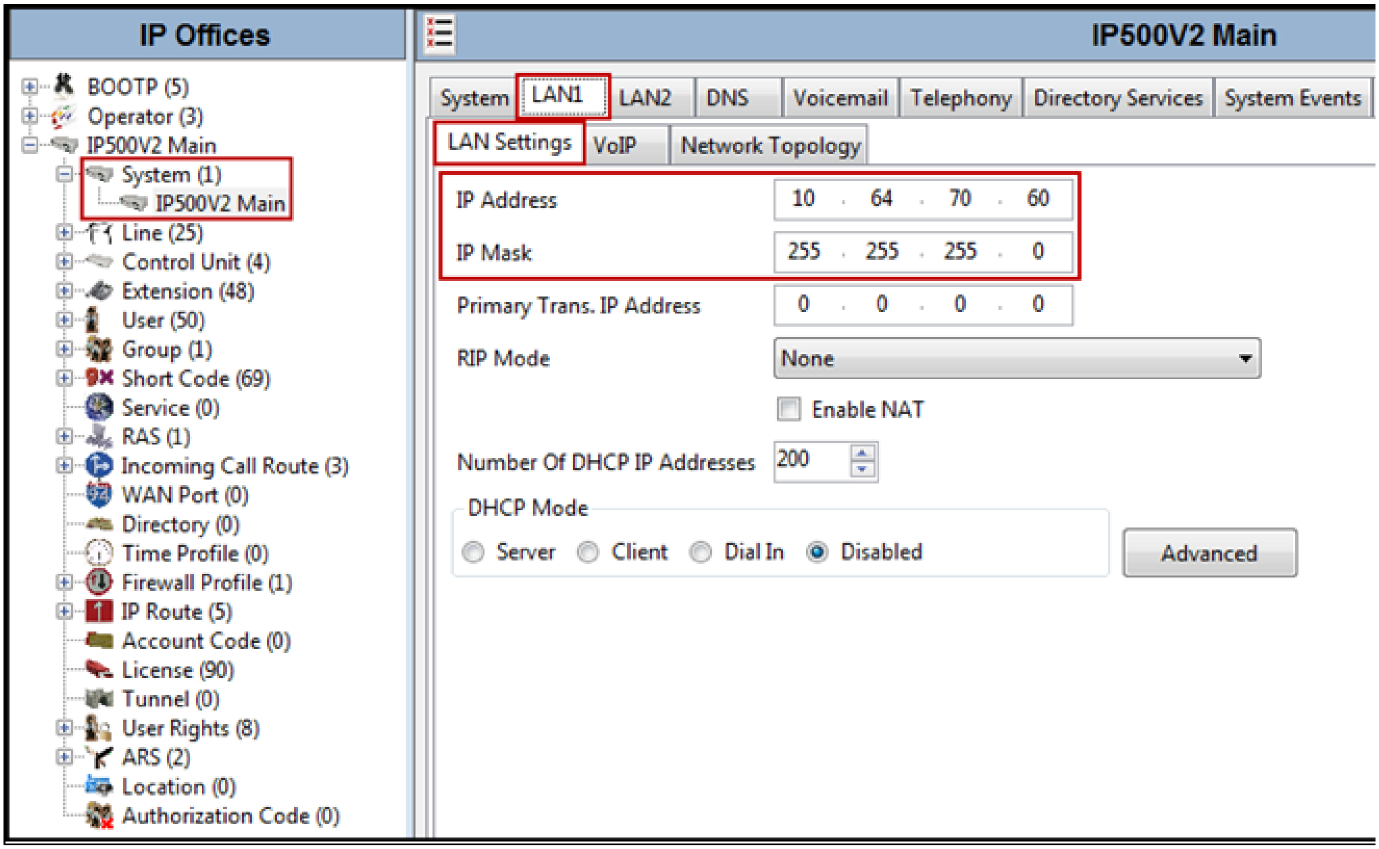
2.2 System

Configure the necessary system settings. In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the LAN1 interface was used to connect Avaya IP Office to the enterprise private network (LAN), LAN2 was not used.

2.2.1 System - LAN1 Tab

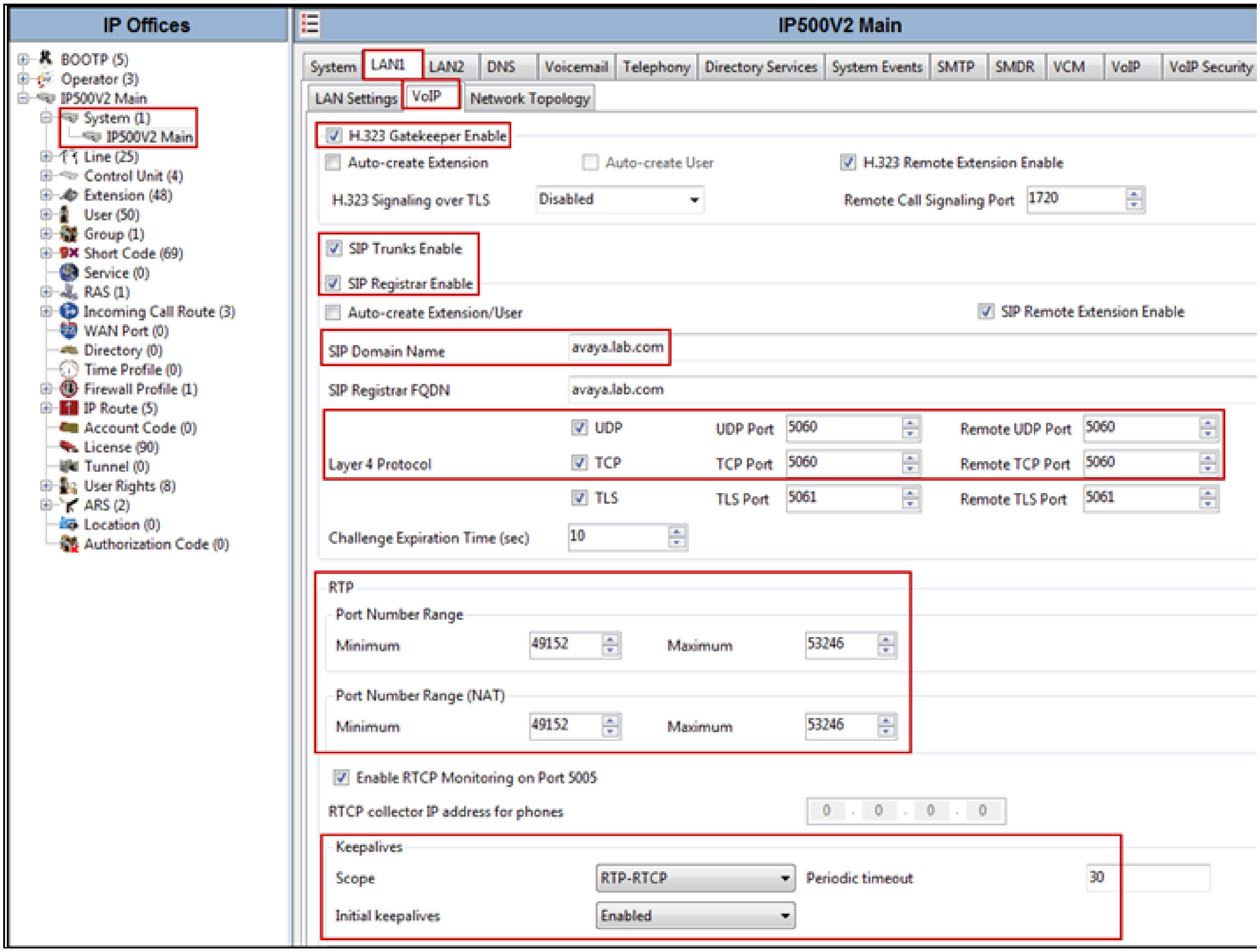
In the sample configuration, I**P500V2 Main** was used as the system name. The **LAN** port connects to Charter's Modular Access Router, across the enterprise LAN (private) network. The **LAN1** settings correspond to the LAN port in IP Office.

Step	Action	Result
1	To Access **LAN1** Settings, Navigate to **System(1) > IP5000V2 Main** in the **Navigation Pane**	
2	In the **Details Pane** Navigate to the **LAN1 > LAN Settings Tab**	
3	Set IP Address: **LAN IP address, e.g. 10.64.70.60.**	
4	Set IP Mask: **LAN subnet mask, e.g. 255.255.255.0.**	
5	All other parameeters should be set according to customer requirements	
6	Click: **OK** to commit	
7	Go to Next Table	



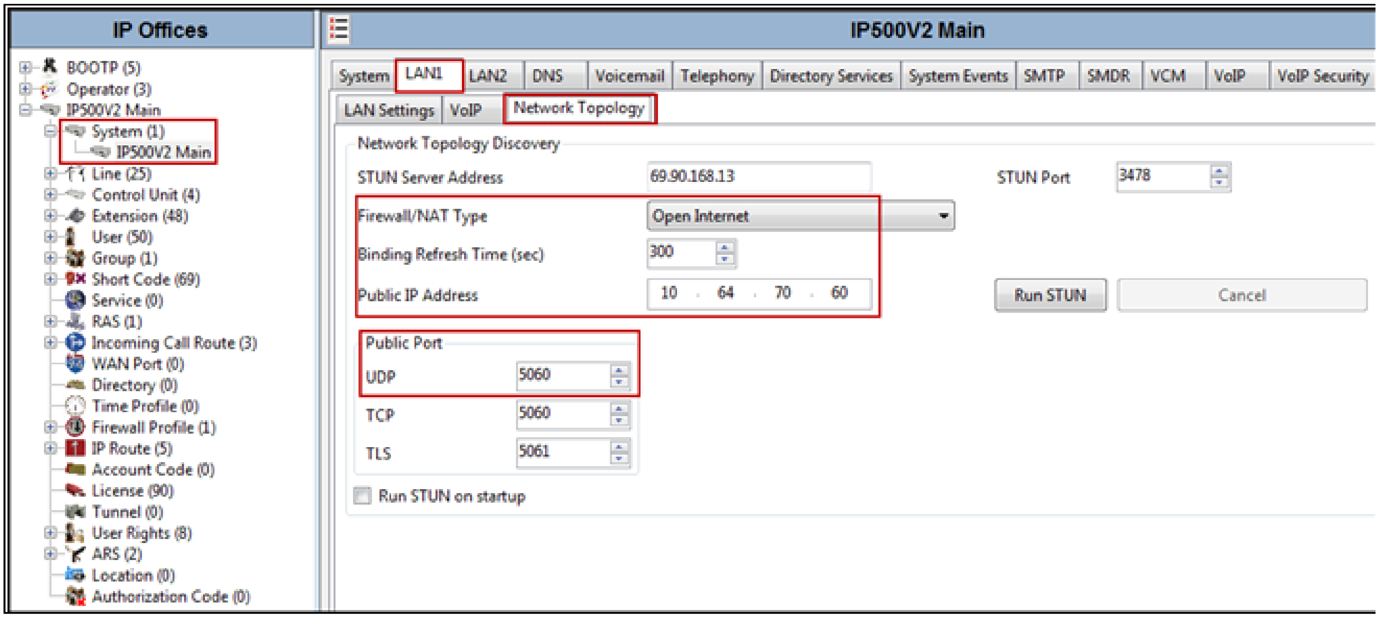
Step	Action	Result
8	Click On: **VoIP Tab**	
9	Check: **H323 Gatekeeper Enable** to allow Avaya IP Telephones/Softphone using the H.323 protocol to register.	
10	Check: **SIP Trunks Enable** to enable the configuration of SIP Trunk connecting to Charter.	
11	Check: **SIP Registrar Enable** to allow Avaya IP Telephones/Softphone to register using the SIP protocol.	
12	Enter: the Domain Name of the enterprise under **Doma in Name**.	
13	Verify: the **UDP Port** and **TCP Port** numbers under **La yer 4 Protocol** are set to **5060**.	
14	Verify: the **RTP Port Number Range** settings for a specific range for the RTP traffic. The **Port Range (Minimum)** and **Port Range (Maximum)** values were kept as default.	
15	In the Keepalives Section, Set Scope: **RTP-RTCP**	This will cause the IP Office to send RTP keepalive packets at the beginning of the calls and every 30 seconds thereafter if no other RTP traffic is present.
16	Set Periodic Timeout: **30**	
17	Set Initial Keepalives: **Enabled**	

18	All other parameters should be set according to customer requirements.	
19	Click: **OK** to commit	
20	Go to Next Table	



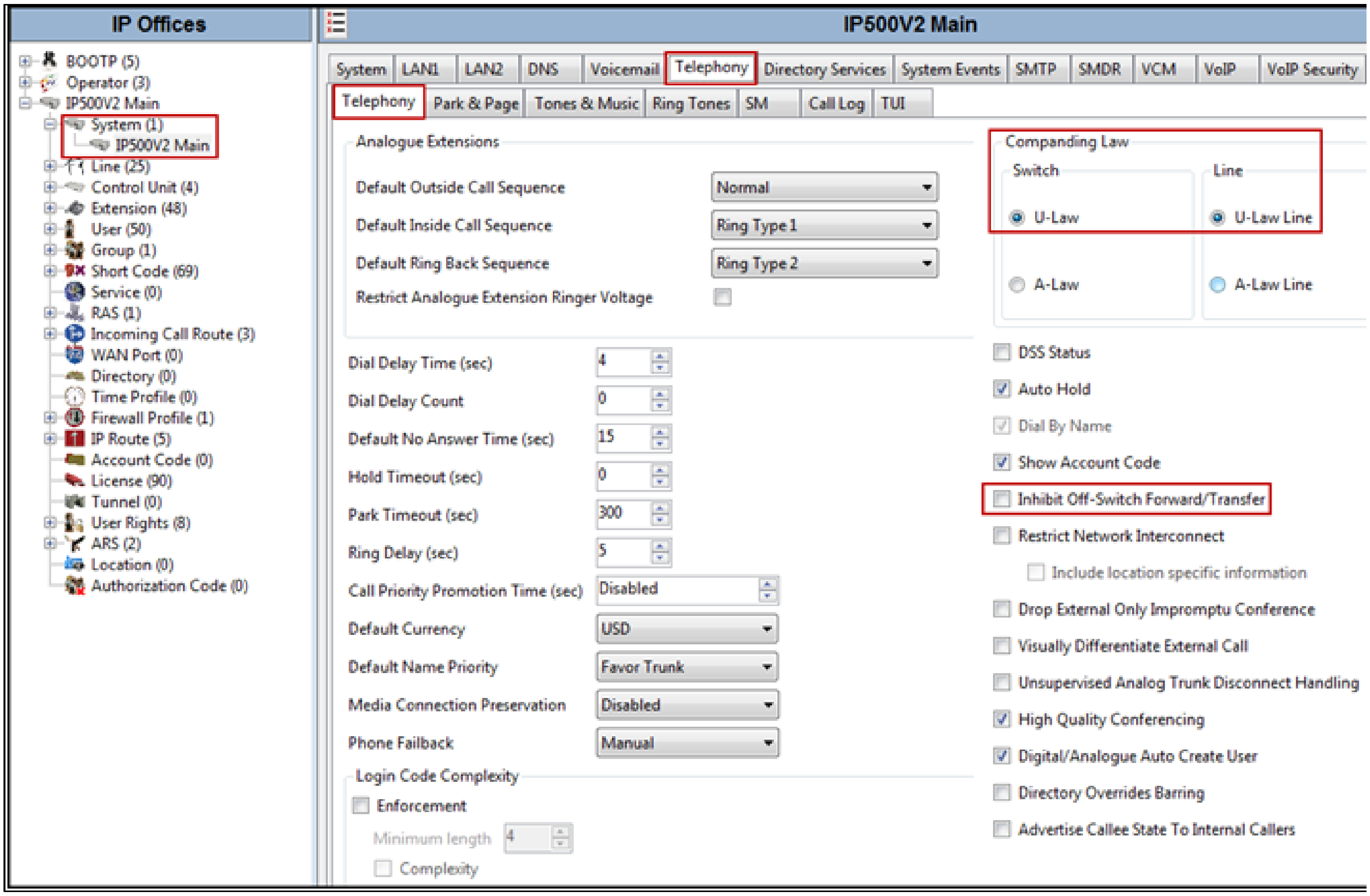
Step	Action	Result
21	Click On: **Network Topology Tab**	
22	Select: **Firewall/NAT Type** from the pull-down menu to the option that matches the network configuration. In the compliance testing, it was set to **Op en Internet**.	With this configuration, even though the default STUN settings are populated, they will not be used.
23	Set Binding Refresh Time (seconds): **Desired Value**, the value of **300 (or every 5 minutes)** was used during the compliance testing.	This value is used to determine the **frequency** that IP Office will send OPTIONS heartbeat to the service provider.
23	Set Public IP address: **The IP Address assigned under the LAN Settings tab, e.g., 10.64.70.60**.	
24	Set Public Port: **5060** for **UDP**	

25	All other parameters should be set according to customer requirements.	
26	Click: **OK** to commit	
27	Go to Next Table	



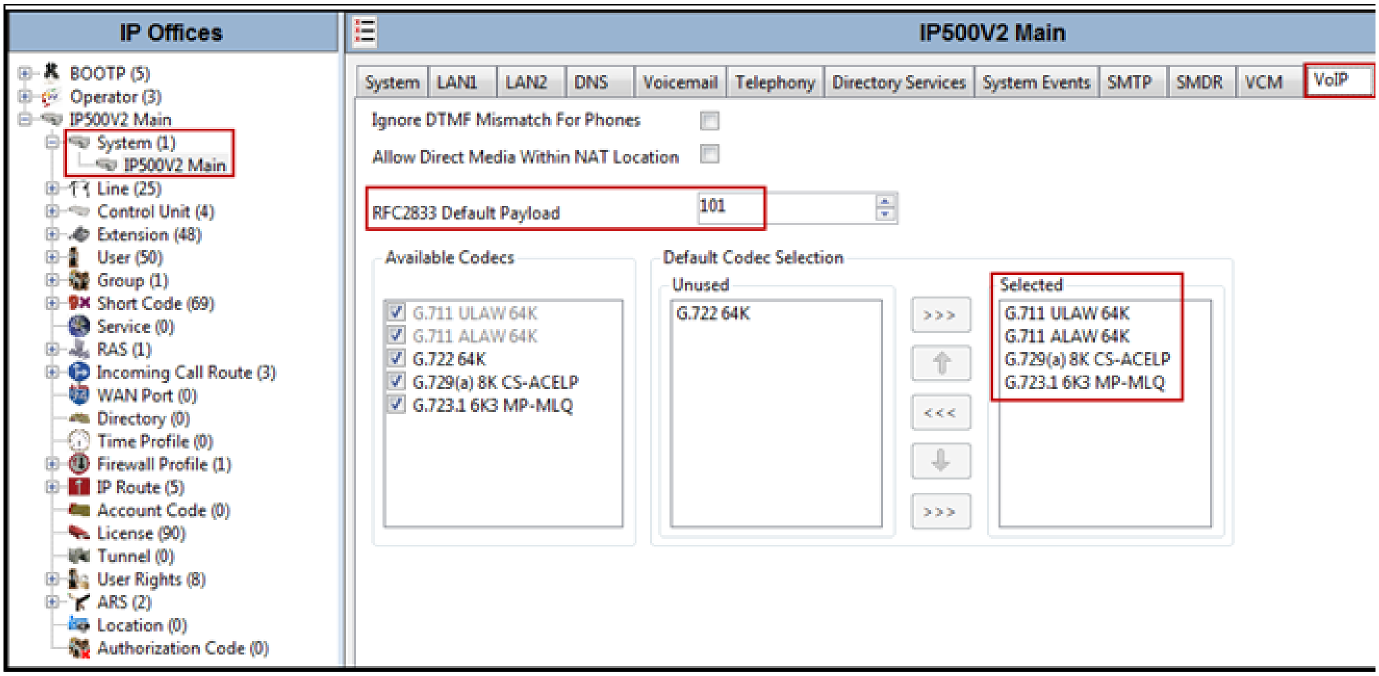
2.2.2 - Telephony Tab

Step	Action	Result
28	Navigate to: **Telephony > Telephony Tab in the Details Pane**	
29	Choose: the **Companding Law** typical for the enterprise location, **U-Law** was used.	
30	Uncheck: **Inhibit Off-Switch Forward/Transfer box**	Allow call forwarding and call transfers to the PSTN via the SIP trunk to the service provider.
31	All other parameters should be set according to customer requirements.	
32	Click: **OK** to commit	
33	Go to Next Table	



2.2.3 System - VoIP TAB

Step	Action	Result
34	Navigate To: **System (1) > IP500V2 Main in the Navigation Pane**	
35	Select: **VoIP tab**	
36	The **RFC2833 Default Payload** field is new in IP Office release 10. The default value **101** was used.	It allows the manual configuration of the payload type used on SIP calls that are initiated by the IP Office.
37	For **Codec Selection:** select the codecs and codec order of preference on the right, under the **Selected** column. The **Default Codec Selection** area enables the codec preference order to be configured on a system-wide basis. The buttons between the two lists can be used to move codecs between the **Unused** and **Selected** lists, and to change the order of the codecs in the **Selected** codecs list. The example below shows the codecs used for IP phones (SIP and H.323), the system's default codecs and order was used.	By default, all IP lines and phones (SIP and H.323) will use the system default codec selection shown here, unless configured otherwise for a specific line or extension.
38	Click: **OK** to commit	
39	Procedure Completed	



<note>

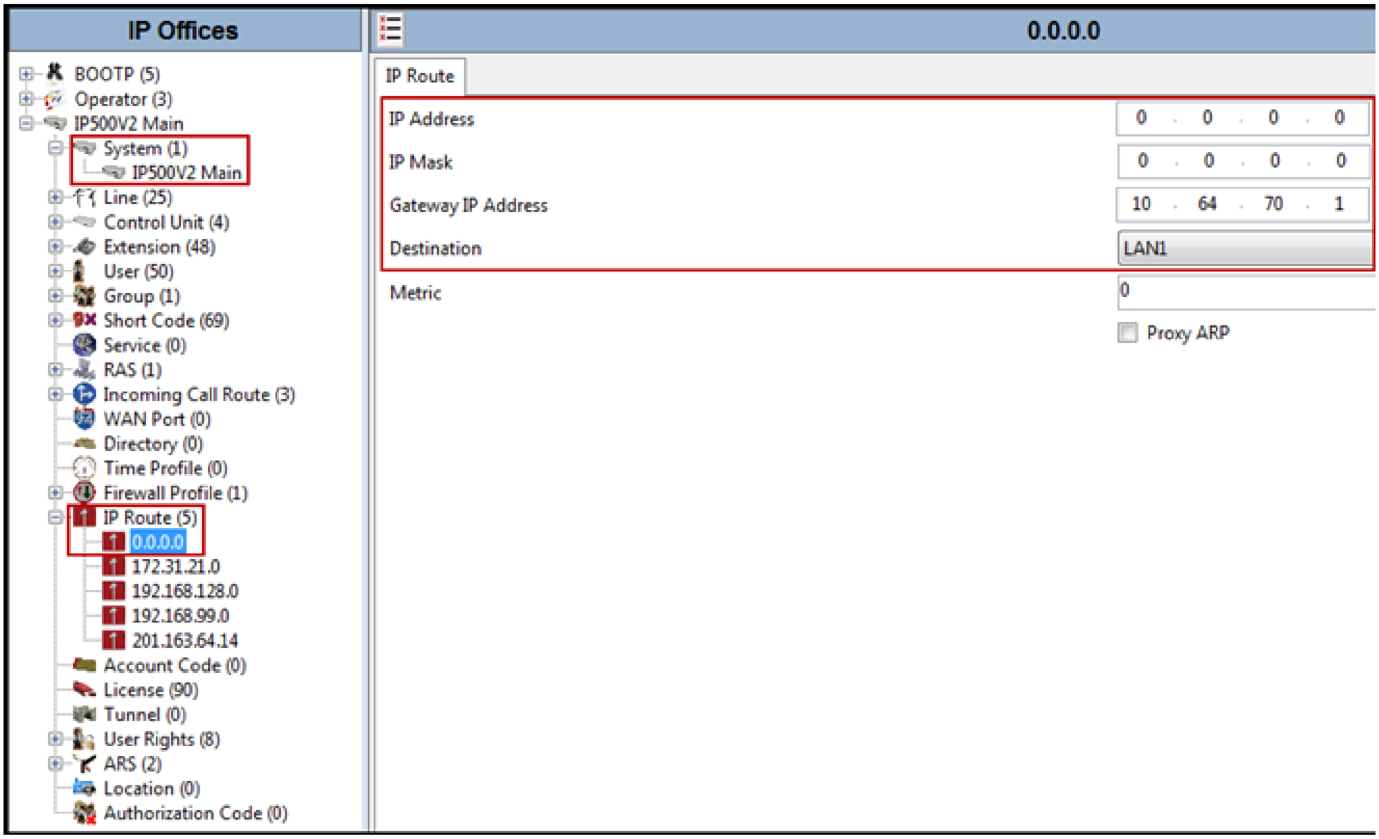
Note: The codec selections defined under this section (System – VoIP Tab) are the codecs selected for the IP phones/extensions. The codec selections defined under **Section 2.4.6** (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk).

2.3 IP Route

In the reference configuration, the IP Office LAN1 interface and the private interface of the Charter's Modular Access Router resided on the same IP subnet, so an IP route was not necessary. In an actual customer configuration, these two interfaces may be in different IP subnets, and in that case an IP route would have to be created to specify the IP address of the gateway or router where the IP Office needs to send the packets, in order to reach the IP subnet where the Charter Modular Access Router resides.

To create an IP route to specify the IP address of the gateway or router where the IP Office needs to send the packets in order to reach the IP subnet where the Charter's Modular Access Router resides (if located in different subnets).

Step	Action	Result
1	In The Left **Navigation Pane**, Right click on **IP Route** > Select **New**	
2	Set the **IP Address** and **IP Mask** to **0.0.0.0** to make this the default route.	
3	Set **Gateway IP Address** to the IP Address of the gateway/route used to route calls to the public network (to the network where Charter's Modular Access Router resides, if located in different subnet), e.g., **10.64.70.1**.	
4	Set **Destination** to **LAN1** from the pull-down menu	
5	Click: **OK** to commit	
6	Procedure Completed	



2.4 SIP Line

A SIP Line is needed to establish the SIP connection between IP Office and the Charter SIP Trunking Service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a SIP Line. Follow the steps in Sections 2.4.1 and 2.4.2 to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses.
- SIP trunk Registration Credentials.
- SIP URI entries.
- Setting of the Use Network Topology Info field on the Transport tab.

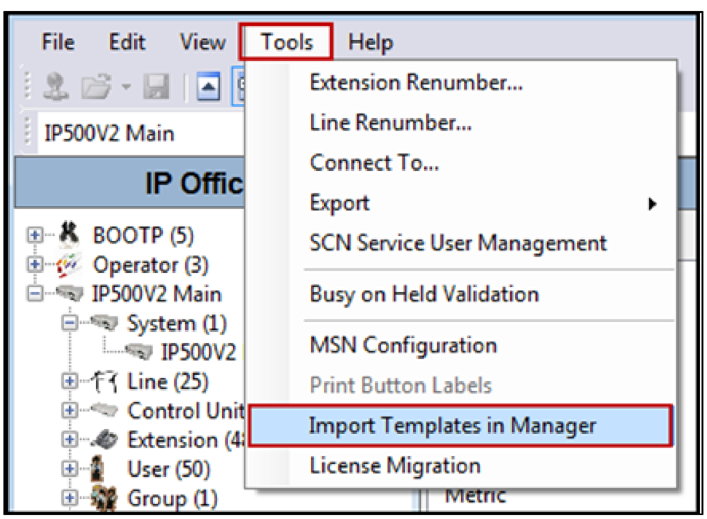
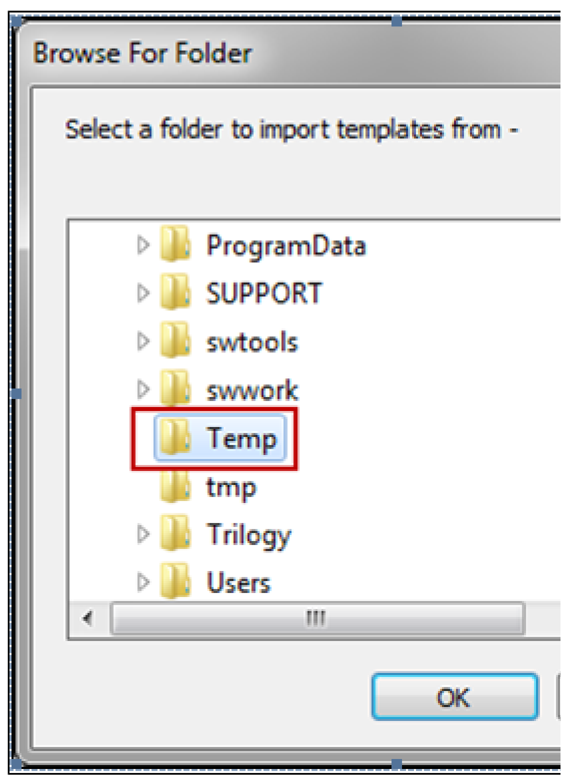
Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 2.4.3** to **2.4.7.**

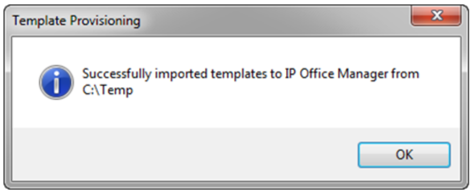
Alternatively, a SIP Line can be created manually. To do so, right-click on **Line** in the **Navigation** pane and select **New à SIP Line**. Then, follow the steps outlined in **Sections 2.4.3** to **2.4.7**

2.4.1 Importing a SIP Line Template

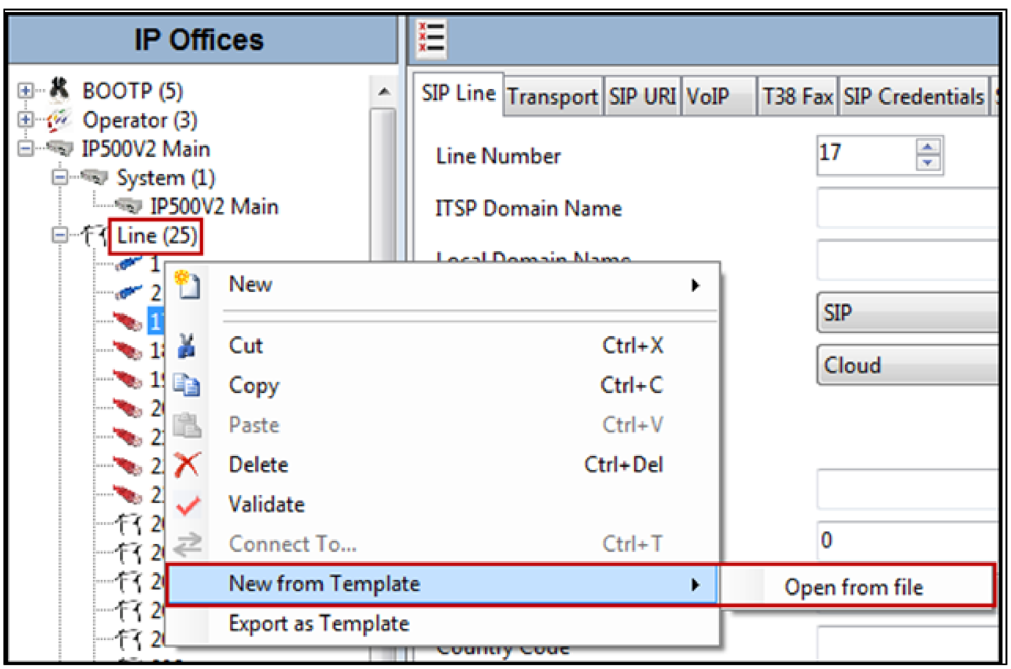
<note>

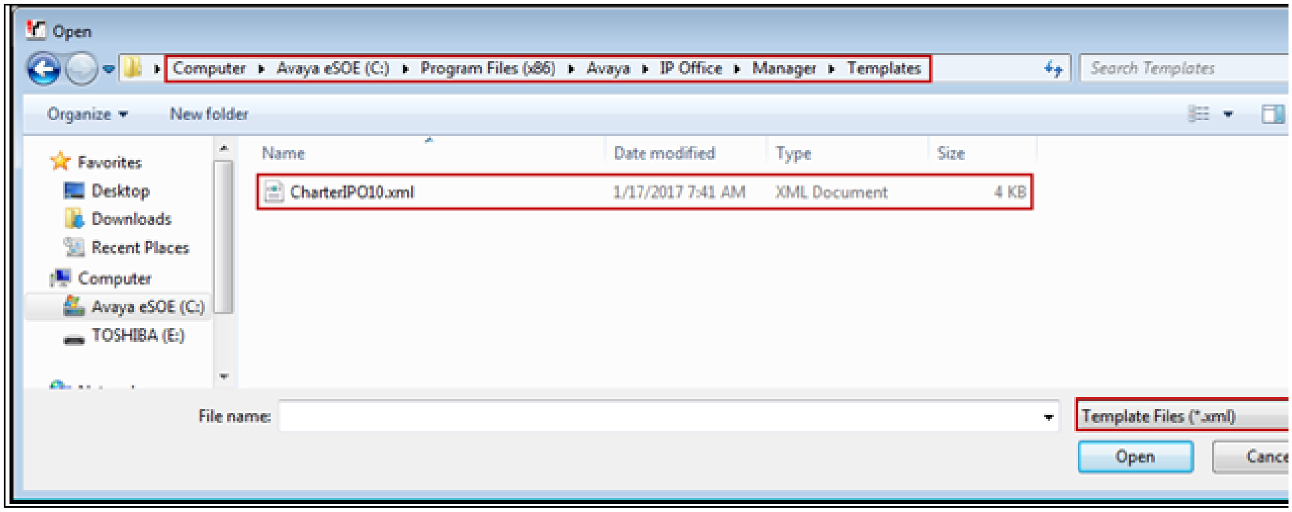
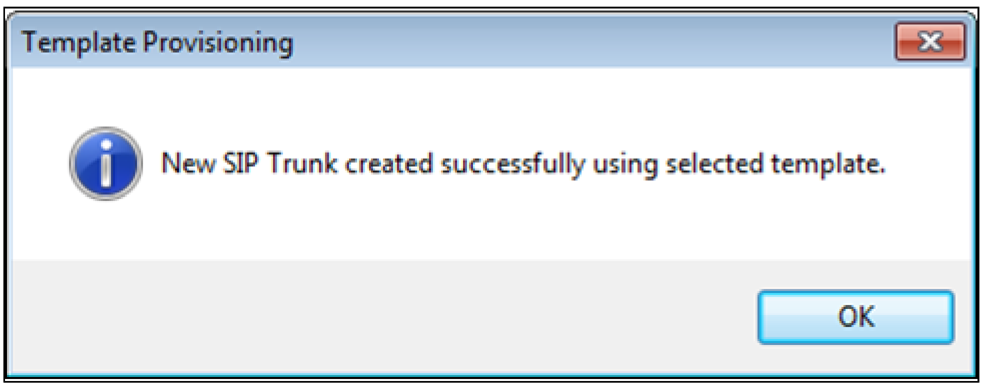
Note: DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

Step	Action	Results
1	Copy a previously created template file to a location (e.g., C:*Temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **<user supplied text>.xml**, where the **<user supplied text>** portion is entered during template file creation.	
2	**<note>** **Note:** If necessary, the ***<user supplied text>*** portion of the template file name may be modified, however the ***<user supplied text>*.xml** format of the file name must be maintained. For example, an original template file ***Test*.xml** could be changed to ***Test1*.xml**. The template file name is selected in **Section 5.4.2**, **step 1**, to create a new SIP Line.	
3	Import the template into IP Office Manager. From IP Office Manager, select **Tools > Import Templates in Manager**.  	A folder browser will open. Select the directory used to stor C:*Temp*). 

4	In the reference configuration, template files **CharterIPO10.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.	
5	After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.  	
6	Go To Next Table	

2.4.2 Creating a SIP Trunk From An XML Template

Step	Action	
7	To create the SIP Trunk from a template, right-click on **Line** in the **Navigation** pane, and select **New from Template > Open from file**.  	

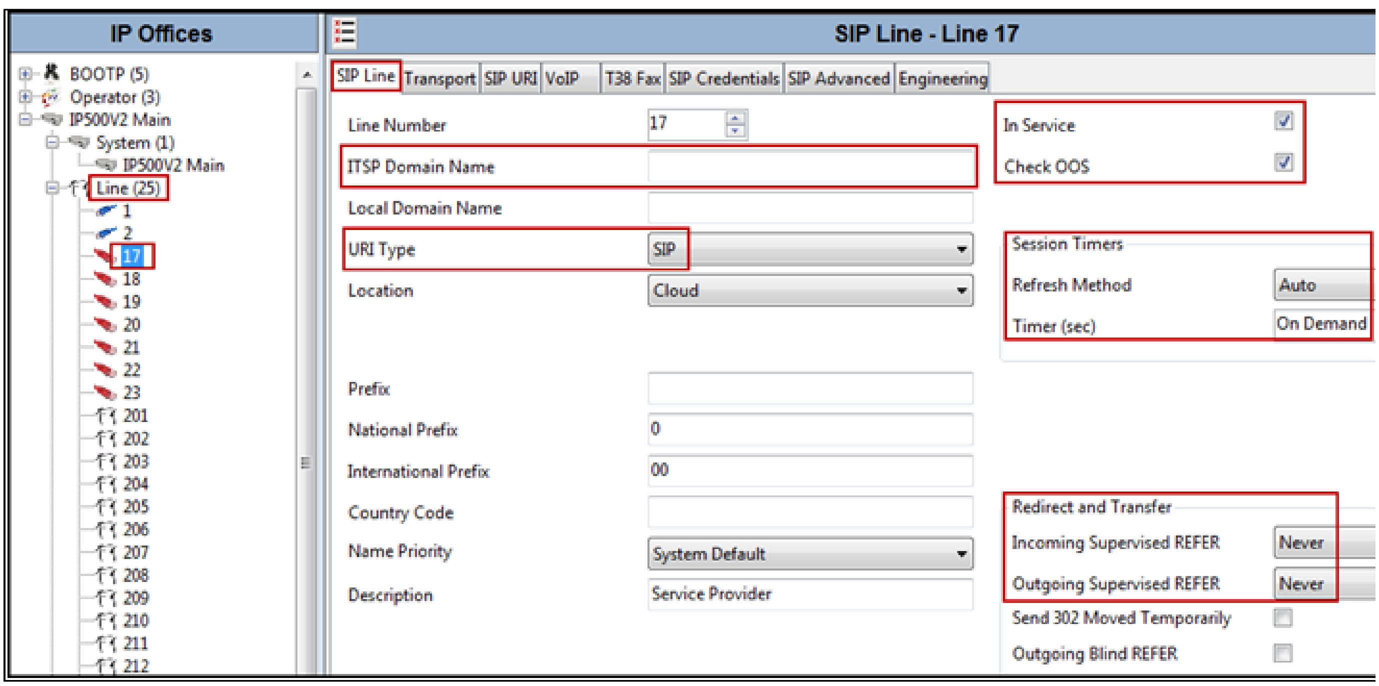
8	Navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates** (or C:*Program Files (x86)*\Avaya\IP Office\Manager\Templates), or bottom right hand side chose **Template Files (*.xml)** format and select the template, in this case **CharterIPO10.xml** was selected. 
9	After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK** 
10	Go To Next Table

<note>

Note: It is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Sections 2.4.3** to **2.4.7**.

2.4.3 SIP Line - SIP Line Tab

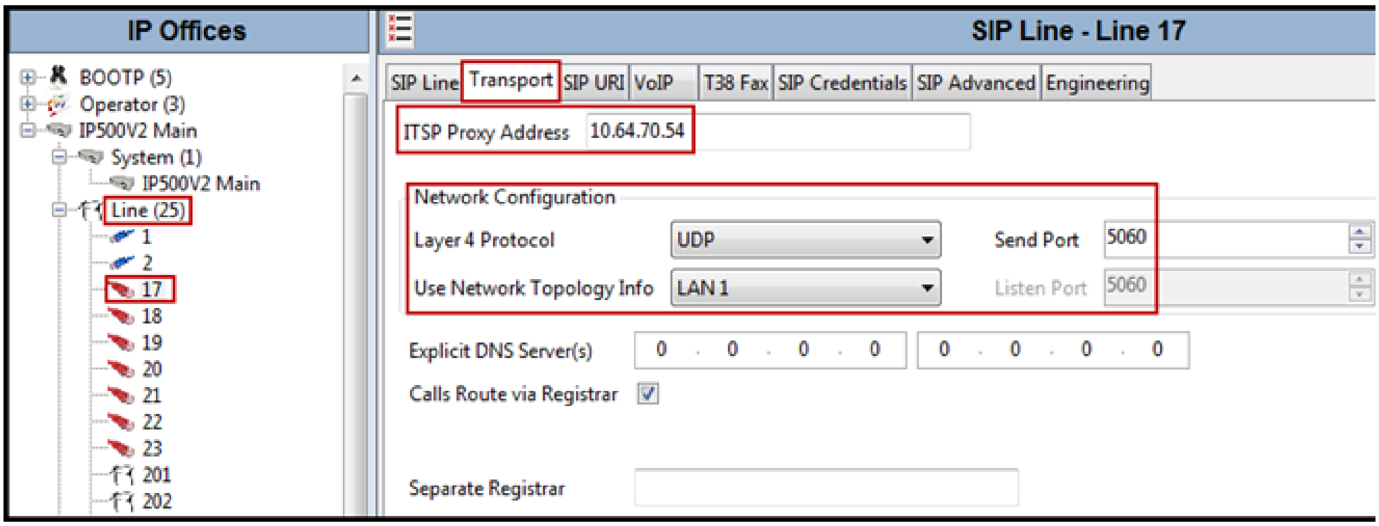
Step	Action	Results
11	Click on the **SIP Line** tab in the **Details** pane	
12	Leave the **ITSP Domain Name** blank	If this field is left blank, then IP Office inserts the ITSP Proxy Address from the Transport tab as the ITSP Domain in the SIP messaging.
13	Verify that **URI Type** is set to **SIP**	
14	Verify that **In Service** box is checked, which is the default value	This makes the trunk available to incoming and outgoing calls.
15	Verify that **Check OOS** box is checked, the default value.	IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office will use the Binding Refresh Time for LAN1, as shown in **Section 2.2.1**.
16	Verify that **Refresh Method** is set to **Auto**.	
17	Verify that **Timer (seconds)** is set to **On Demand**	
18	Under **Redirect and Transfer**, set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Never** (see **Sections 2.1**).	
19	All other parameters should be set to default or according to customer requirements.	
20	Click **OK** to commit	
21	Go to Next Table	



2.4.4 SIP Line - Transport Tab

Step	Action	Results
22	Click on the **Transport** tab in the **Details** pane	

23	Set the **ITSP Proxy Address** to the IP address of the inside interface (or private side) assigned to Charter's Modular Access Router.	
24	Set the **Layer 4 Protocol** to **UDP**.	
25	Set **Use Network Topology Info** to **LAN1** as configured in **Section 2.2**.	
26	Set the **Send Port** to **5060**.	
27	All other parameters should be set to default or according to customer requirements.	
28	Click **OK** to commit	
29	Go to Next Table	



2.4.5 SIP Line - SIP URI Tab

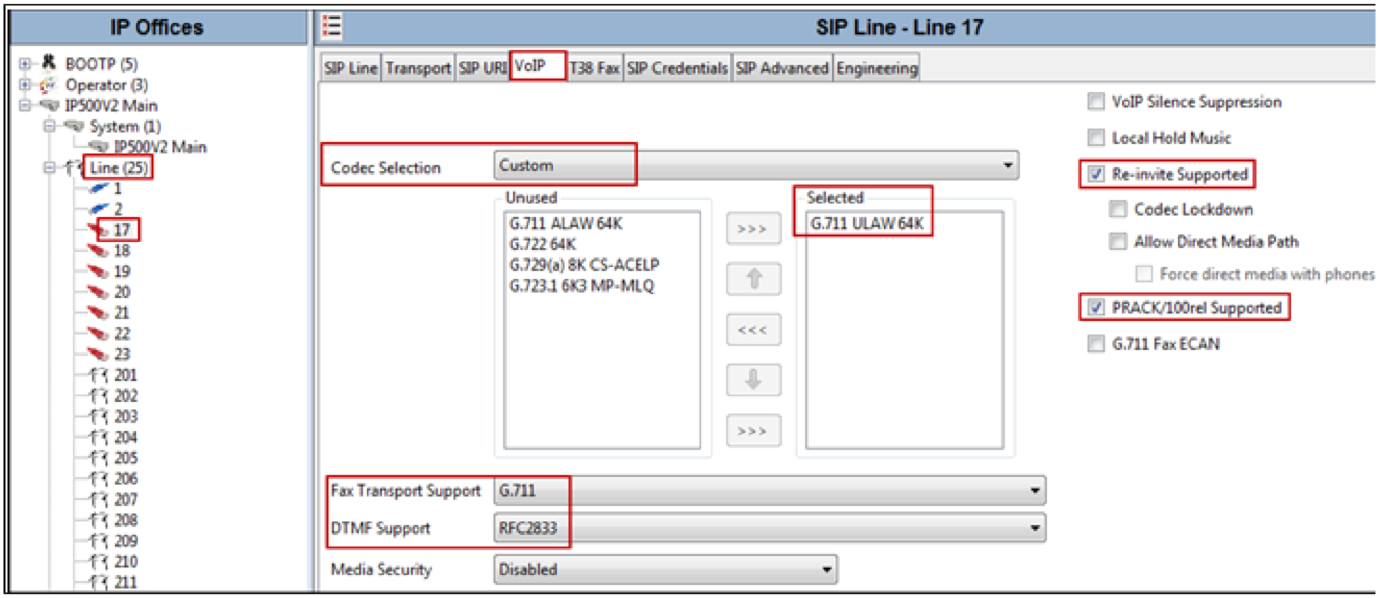
A SIP URI entry needs to be created to match each incoming number that IP Office will accept on this line.

Step	Action	Results
30	In the SIP URI Tab click **Add**	The **New Channel** area will appear at the bottom of the pane.
31	To edit an existing entry, click an entry in the list at the top, and click the **Edit** button.	
32	Set **Local URI**, **Contact**, **Display Name** to **Use Internal Data**.	
33	Set **Identity** under **Identity** to **Auto**.	
34	Set **Header** under **Identity** to **P Asserted ID**	
35	Set **Send Caller ID** under **Forwarding and Twinning** to **Diversi on Header**.	
36	Set **Diversion Header** to **Auto**.	
37	Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field.	This line group number will be used in defining incoming call routes for this line.

38	Associate the line to an outgoing line group using the **Outgoing Group** field.	The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **17** wa s defined that only contains this line (line 17).
39	Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.	
40	Click **OK** to commit	
41	Click **OK** to commit again	
42	Go to Next Table	

2.4.6 SIP Line -VoIP Tab

Step	Action	Results
43	Select the **VoIP tab** to set the Voice over Internet Protocol Parameters of the SIP Line	The **New Channel** area will appear at the bottom of the pane.
44	In the sample configuration, the **Codec Selection** was configured using the **Custom** option , allowing an explicit order of codecs to be specified for the SIP Line. The buttons allow setting the specific order of preference for the codecs to be used on the SIP Line, as shown. Charter only supports codec G.711ULAW for audio.	
45	Select G.711 for **Fax Transport Support** (Refer to **Section 2.1**).	
46	Set the **DTMF Support** field to **RFC2833.**	This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
47	Check the **Re-invite Supported** box	Allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
48	Check the **PRACK/100rel Supported** box	Advertise the support for reliable provisional responses and Early Media to Charter.
49	Set **Diversion Header** to **Auto**.	
50	Default values may be used for all other parameters.	
51	Click **OK** to commit	
52	Go to Next Table	

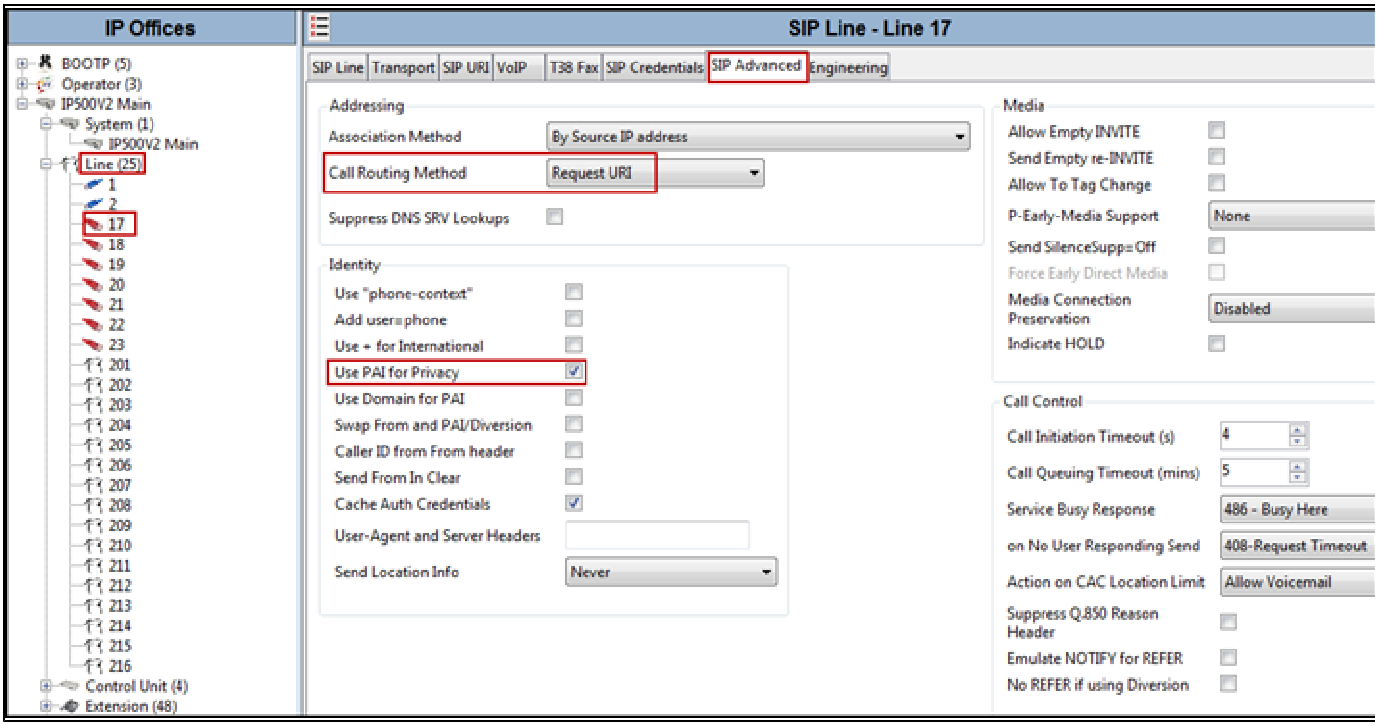


<note>

Note: The codec selections defined under this section (SIP Line – VoIP tab) are the codecs selected for the SIP Line (Trunk). The codec selections defined under Section 2.2.3 (System – VoIP tab) are the codecs selected for the IP phones/extension (H.323 and SIP).

2.4.7 SIP Line - SIP Advanced Tab

Step	Action	Results
53	Select the **SIP Advanced tab** to configure IP Office to use the PAI header for privacy calls	For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “anonymous”. IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, IP Office will use the PPI header for privacy.
54	Verify the **Call Routing Method** is set to **Request URI**.	
55	Check the box for **Use PAI for Privacy**.	
56	Default values may be used for all other parameters.	
57	Click **OK** to commit	
58	Procedure Completed	

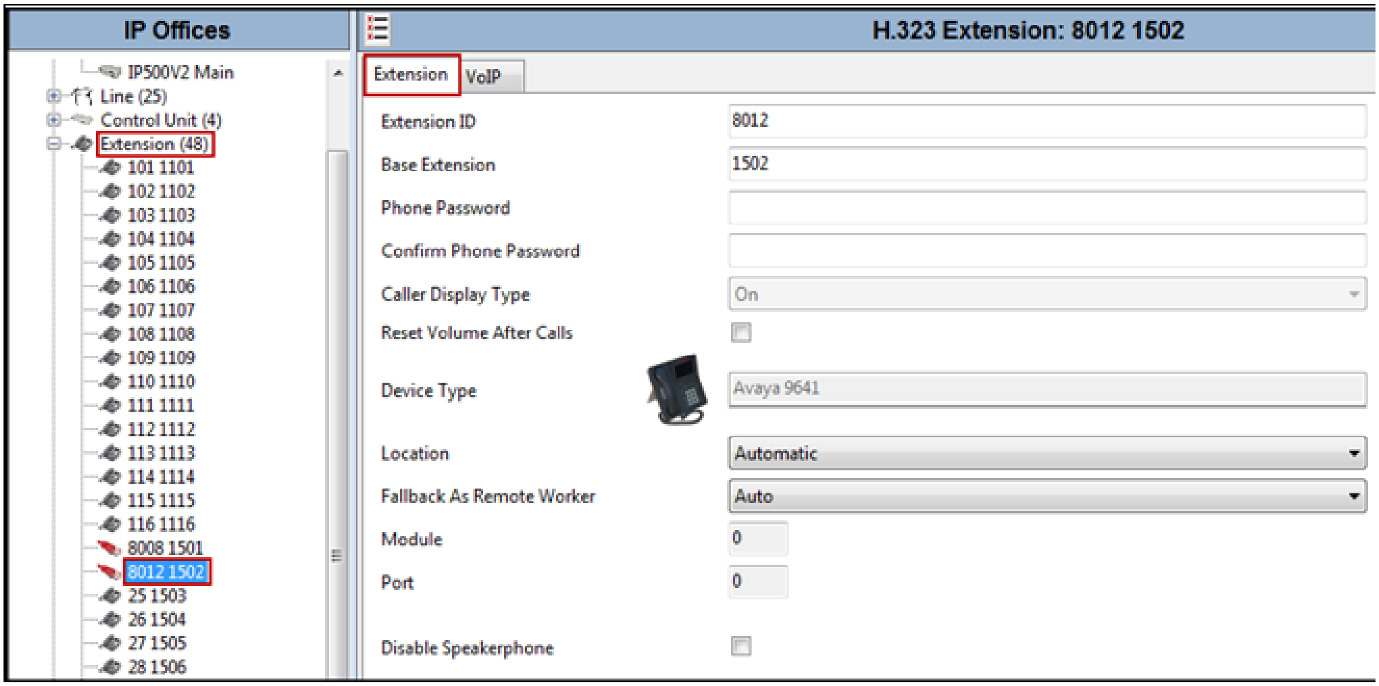


2.5 Extension

In this section, an example of an Avaya IP Office Extension will be illustrated. In the interest of brevity, not all users and extensions will be presented, since the configuration can be easily extrapolated to other users and extensions.

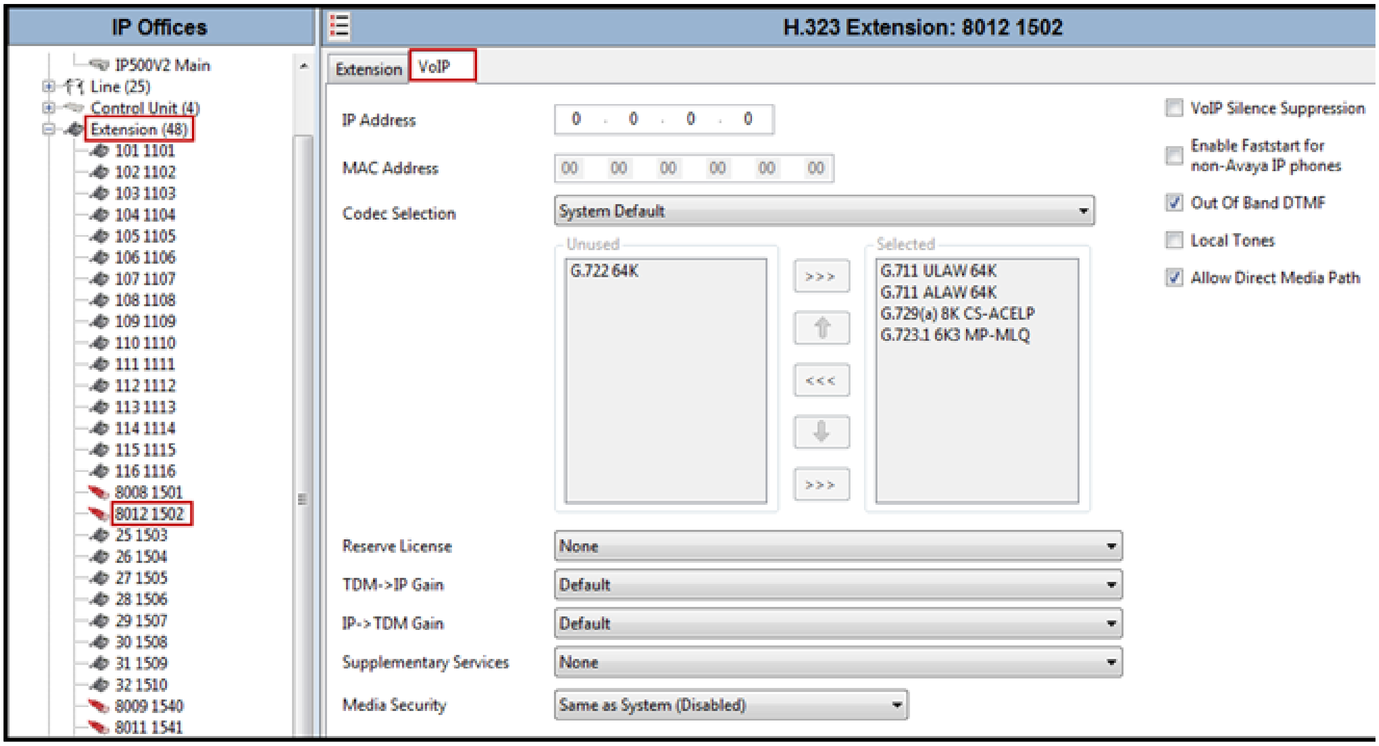
Step	Action	Results
1	Right click on **Extension**	
2	Select **New >** Select **H323** or **SIP**	
3	Select the **Extension** Tab	
4	Go to Next Table	

Following is an example of extension 1502; this extension corresponds to an H.323 extension:



Step	Action	Results
5	Select the **V OIP** tab	
6	Use default values on **V oIP** tab	By default, all IP phones (SIP and H.323) will use the system default codec selection configured under the System VoIP tab (**Section 2.2.3**), unless configured otherwise for a specific extension by selecting **Custom** under **Codec Selection** o n the screenshot shown below.
7	Procedure Completed	

The example below shows the codecs used for IP phones (SIP and H.323).

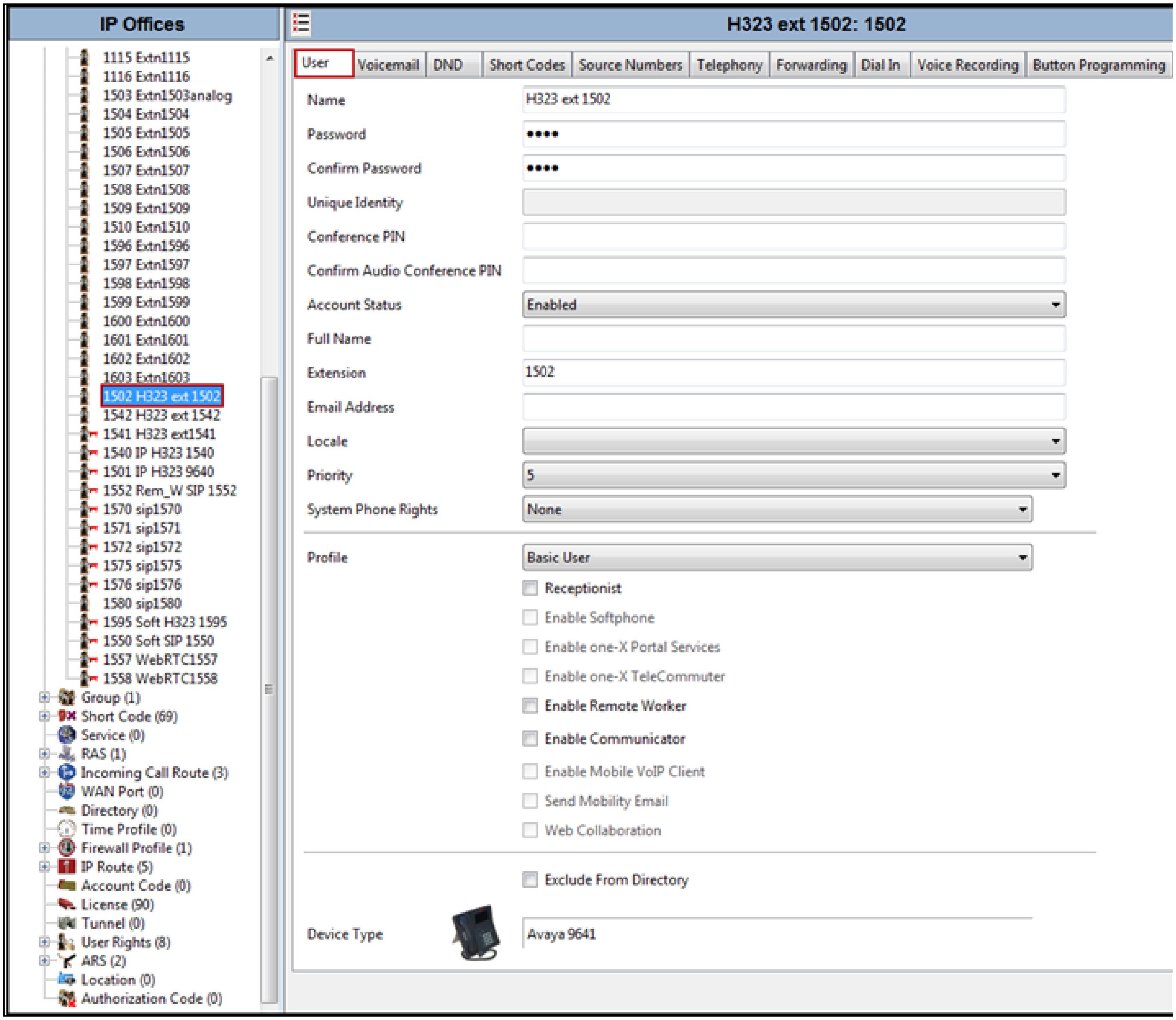


2.6 Users

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 2.4**.

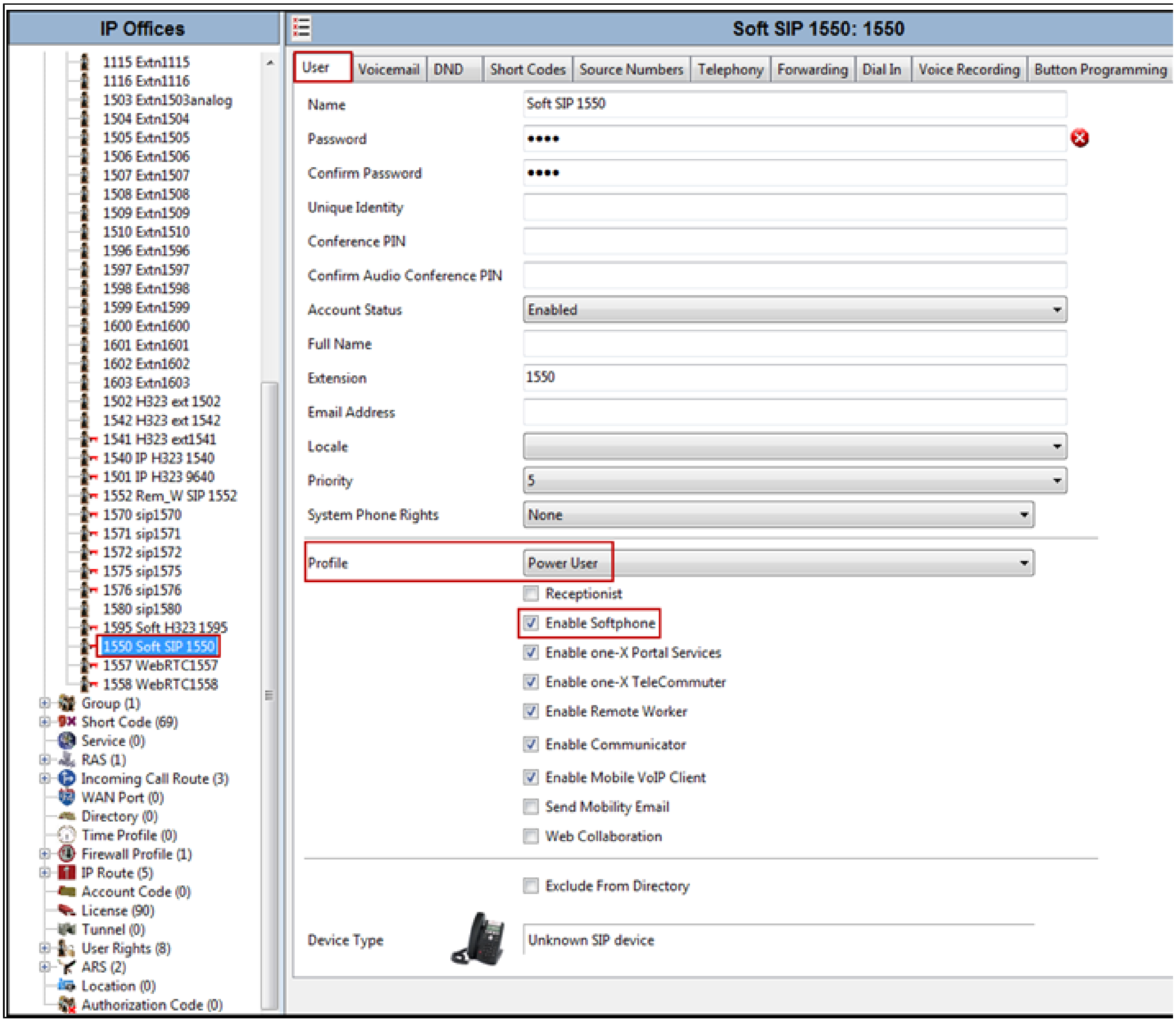
Step	Action	Results
1	Navigate to **User** in the left **Navigation** pane	
2	Select the name of the user to be modified in the center **Group** pane	
3	Go to Next Table	

In the example below, the name of the user is “H323 Ext 1502”:

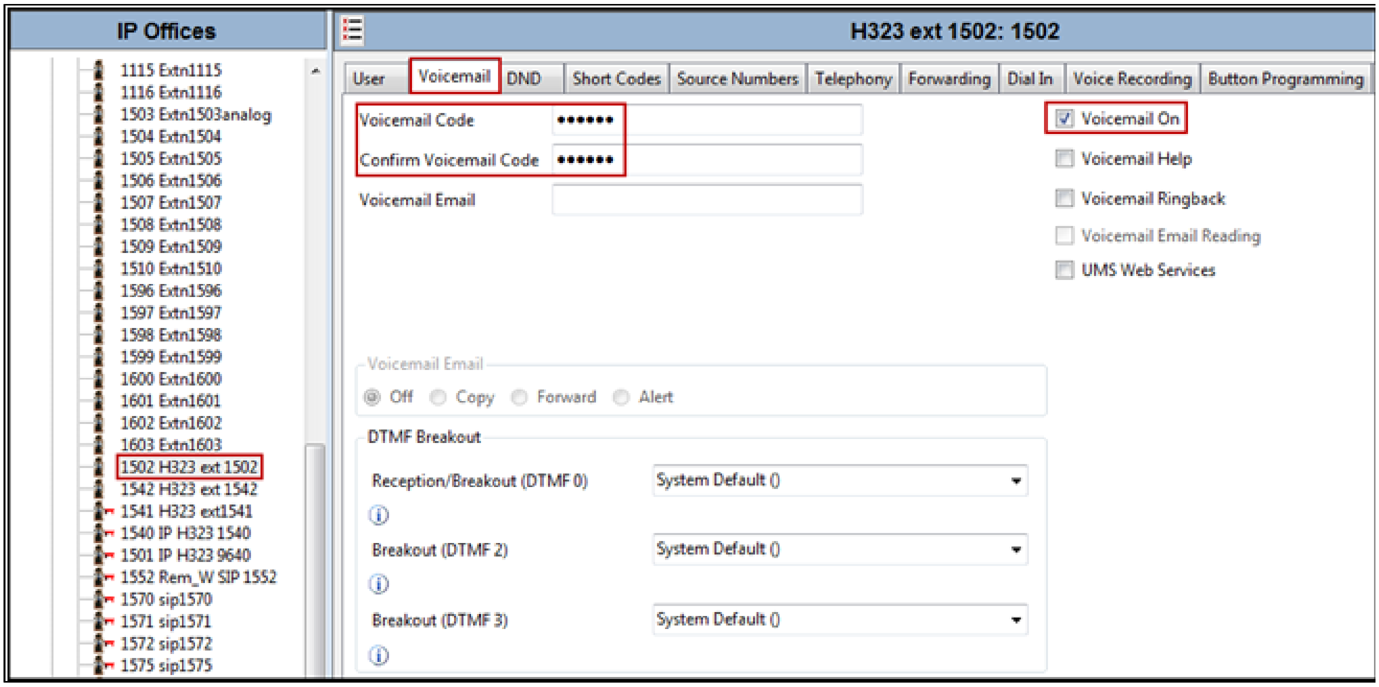


In the example below, the name of the user is “Soft SIP 1550”.

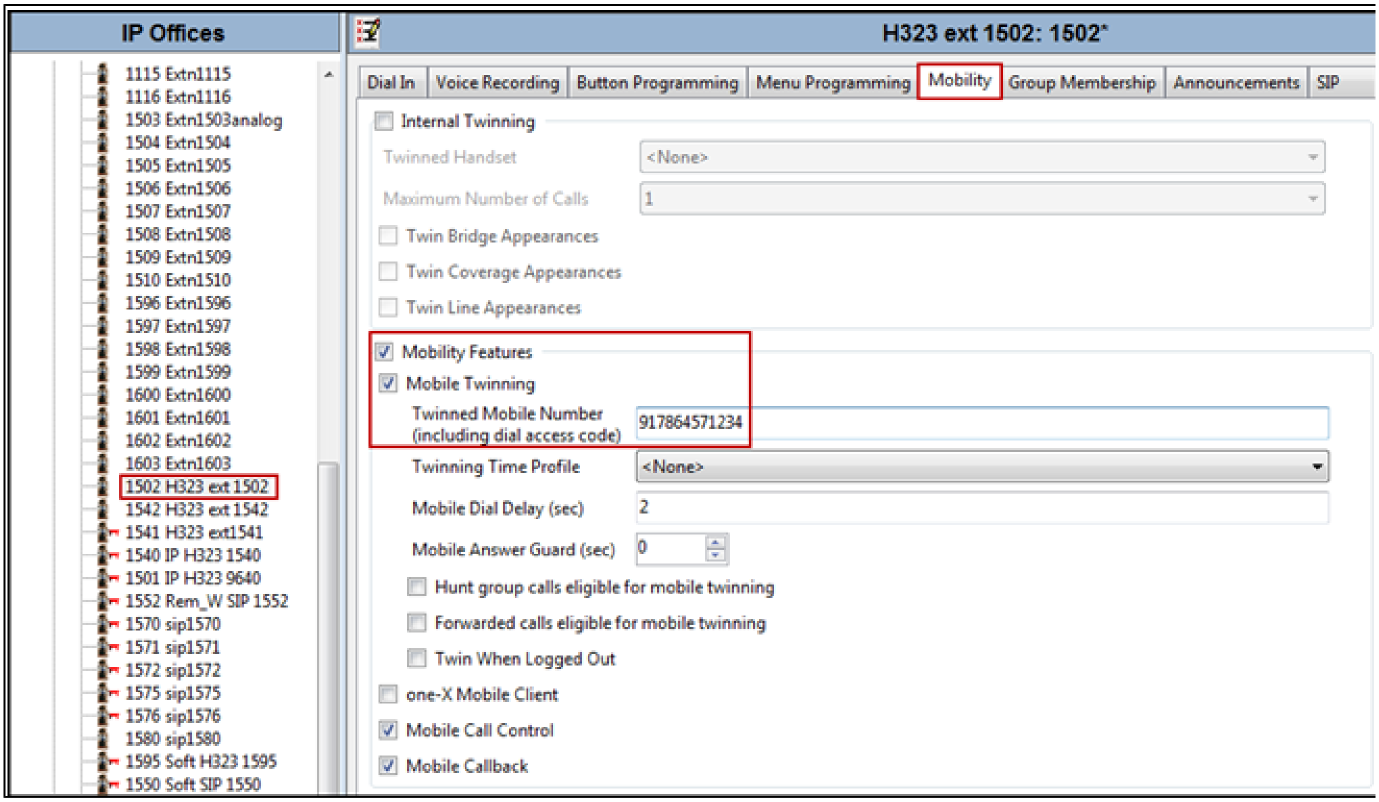
Step	Action	Results
4	This is an Avaya IP Office Softphone user, set the Profile to **Power User**	
5	Ceck **Enable Softphone**	
6	Go to Next Table	



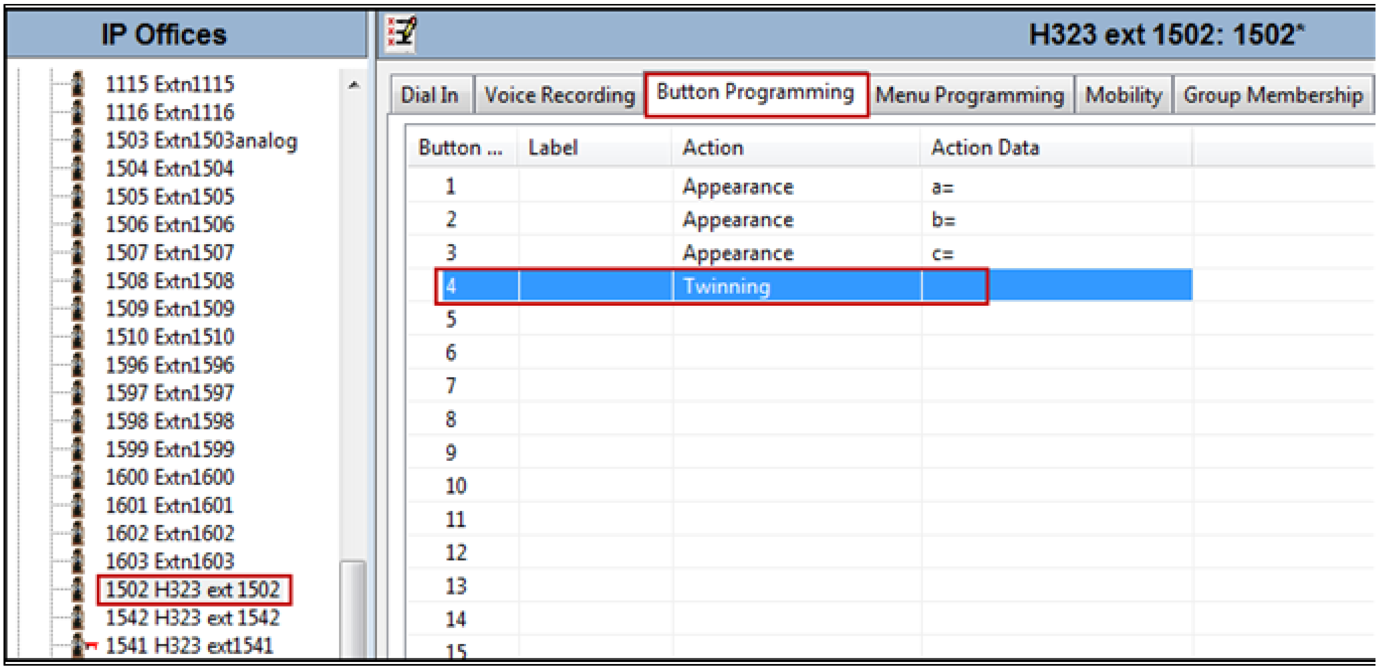
Step	Action	Results
7	Select the **Voicemail** Tab	The screen shows the **Voicemail** tab for the user with extension 1502. Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters. In the verification of these Application Notes, incoming calls from Charter to this user were redirected to Voicemail Pro after no answer. Voicemail messages were recorded and retrieved successfully. Voice mail navigation and retrieval were performed locally and from PSTN telephones to test DTMF using RFC 2833.
8	Go to Next Table	



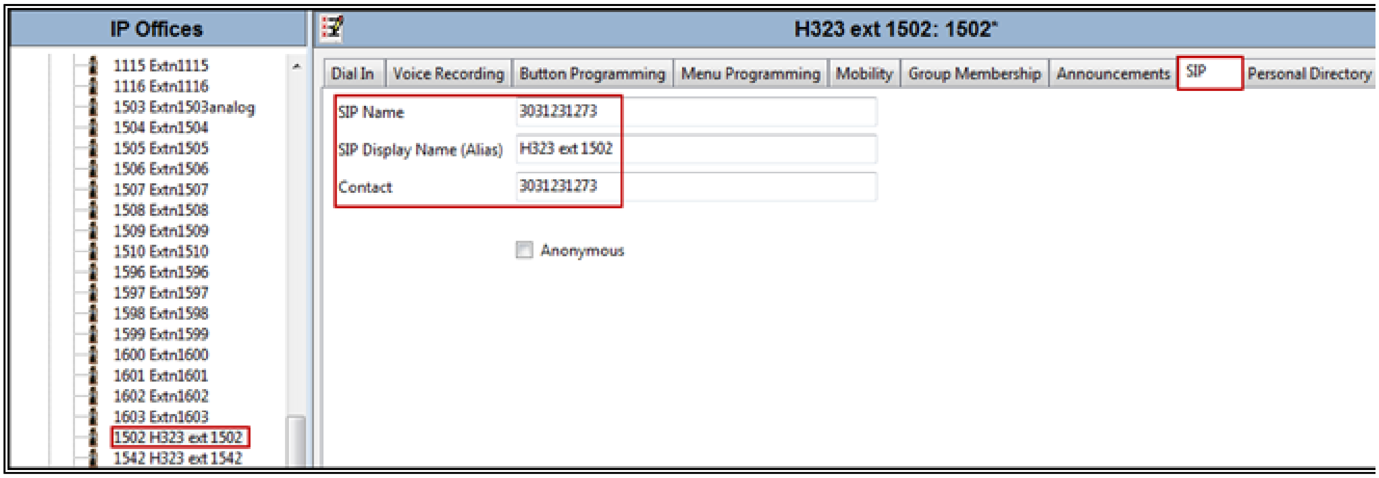
Step	Action	Results
9	Select the **Mobility** tab	
10	Check **Mobility Features** box	
11	Check **Mobile Twinning** box	
12	Configure **Twinned Mobile Number** field with the number to dial to reach the twinned telephone, including the dial access code “9”, in this case **917864571234**.	
13	Other options can be set according to customer requirements	
14	Go to next Table	



Step	Action	Results
15	Select the **Button Programming** tab on the user	
16	select the **button to program** to turn Mobil Twinning on and off	
17	Click on **Edit > Emulation > Twinning**	In the sample below, button **4** was programmed to turn Mobil Twinning on and off on user 1502.
18	Go to next Table	



Step	Action	Results
19	Select **SIP** tab	the values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the "From" and "Contact" headers for outgoing SIP trunk calls. In addition, these settings are used to match against the SIP URI of incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 2.4**)
20	The **SIP Name** and **Contact** are set to one of the DID numbers assigned to the enterprise by Charter.	In the example, DID number **3031231273** was used.
21	The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name.	
22	If all calls involving this user should be considered private, then the **Anonymous** box may be checked to withhold the Caller ID information from the network.	
23	Procedure Completed	

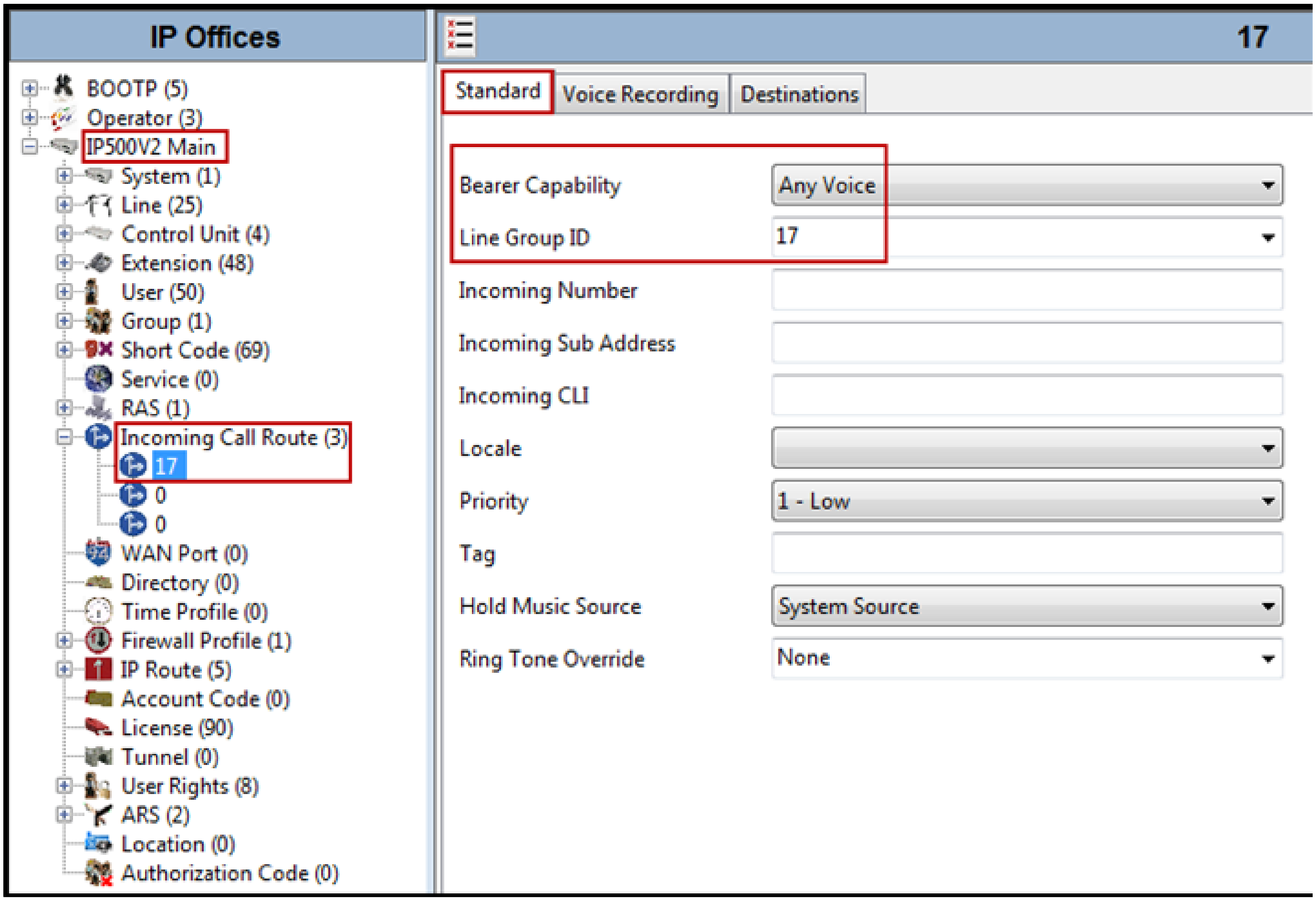


2.7 Incoming Call Route

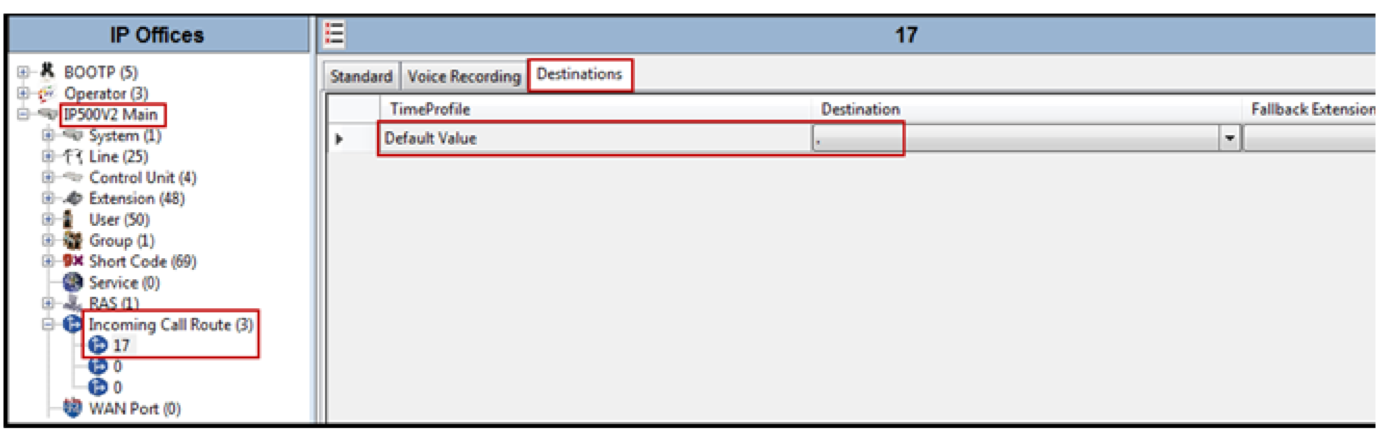
An incoming call route maps inbound DID numbers on a specific line to internal extensions, hunt groups, short codes, etc., within the IP Office system.

In a scenario like the one used for the compliance test, only one incoming route is needed, which allows any incoming number arriving on the SIP trunk to reach any predefined extension in IP Office. The routing decision for the call is based on the parameters previously configured for **Call Routing Method** and **SIP URI (Section 2.4.5)** and the users **SIP Name** and **Contact**, already populated with the assigned Charter DID numbers **(Section 2.6)**.

Step	Action	Results
1	From the left **Navigation** pane, right-click on **Incoming Call Route** and select **New**.	
2	On the **Details** pane (not shown), Select the **Standard** tab	
3	Set **Bearer Capacity** to **Any Voice**.	
4	Set the **Line Group ID** to the incoming line group of the SIP line defined in **Section 5.4**.	
5	Default values may be used for all other parameters.	
6	Go to Next Table	



Step	Action	Results
7	Under the **Destinations** tab, enter "**.**" for the **Default Value**.	This setting will allow the call to be routed to any destination with a value on its **SIP Name** f ield, entered on the **SIP** tab of that **User,** which matches the number present on the user part of the incoming Request URI.
8	Click **OK** to commit	
9	Procedure Completed	

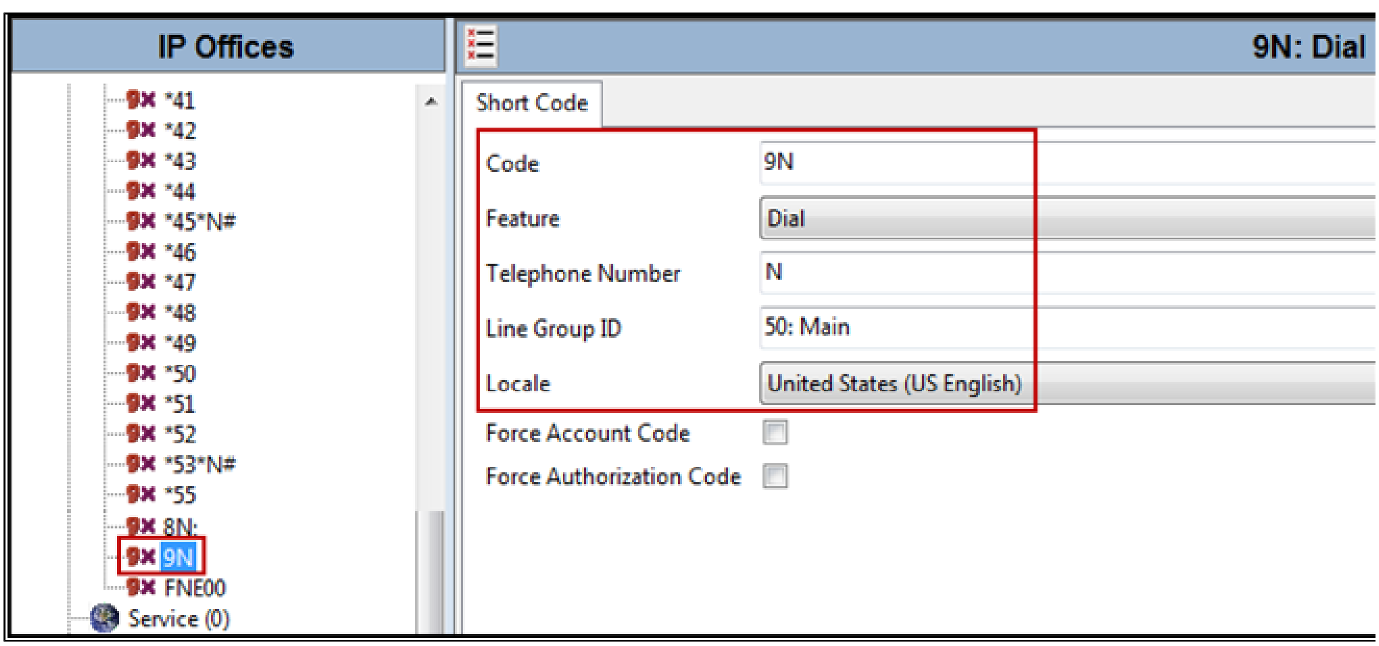


2.8 Outbound Call Routing

For outbound call routing, a combination of system short codes and Automatic Route Selection (ARS) entries are used. With ARS, features like time-based routing criteria and alternate routing can be specified so that a call can re-route automatically if the primary route or outgoing line group is not available. While detailed coverage of ARS is beyond the scope of these Application Notes, and alternate routing was not used in the reference configuration, this section includes some basic screen illustrations of the ARS settings used during the compliance testing.

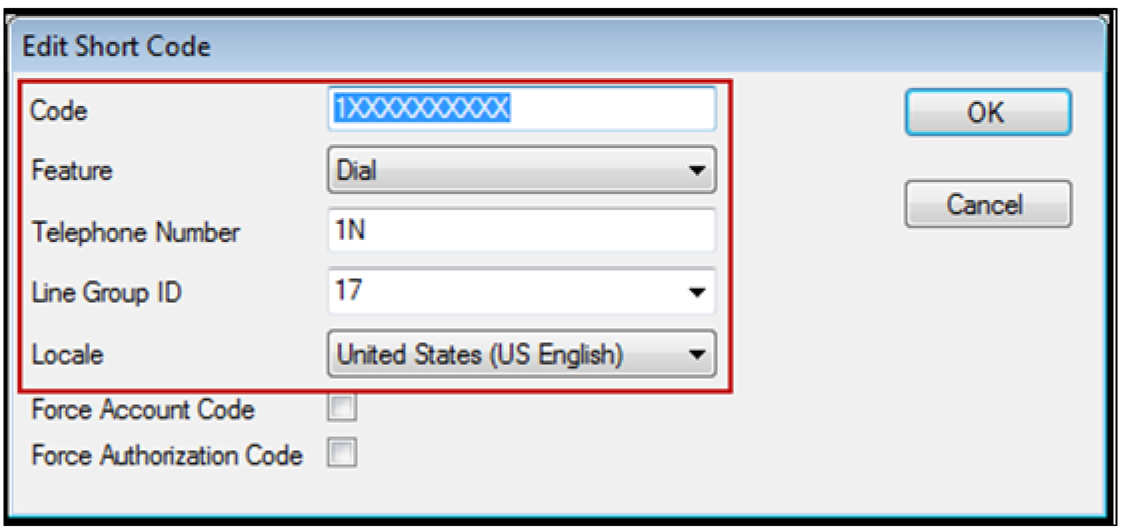
2.8.1 Short Codes and Automatic Route Selection

Step	Action	Results
1	To create the short code used for ARS right-click on **Short Code** in the **Navigation** pane	
2	Select **New**	The screen below shows the creation of the short code **9N** use d in the reference configuration. When the Avaya IP Office users dialed 9 plus any number N, calls were directed to **Line Group 17** which configurable via ARS.
3	In the **Code** field, enter the dial string which will trigger this short code. In this case, **9N** was used (note that the semi-colon is not used here).	
4	Set **Feature** to **Dial**.	This is the action that the short code will perform.
5	Set **Telephone Number** to **N**.	The value **N** represents the number dialed by the user after removing the **9** prefix. This value is passed to ARS.
6	Set the **Line Group ID** to **50: Main** to be directed to **Line Group 17**, which is configurable via ARS.	
7	Set the Locale to **United States (US English)**.	
8	Click the **OK** to commit.	
9	Go to Next Table	

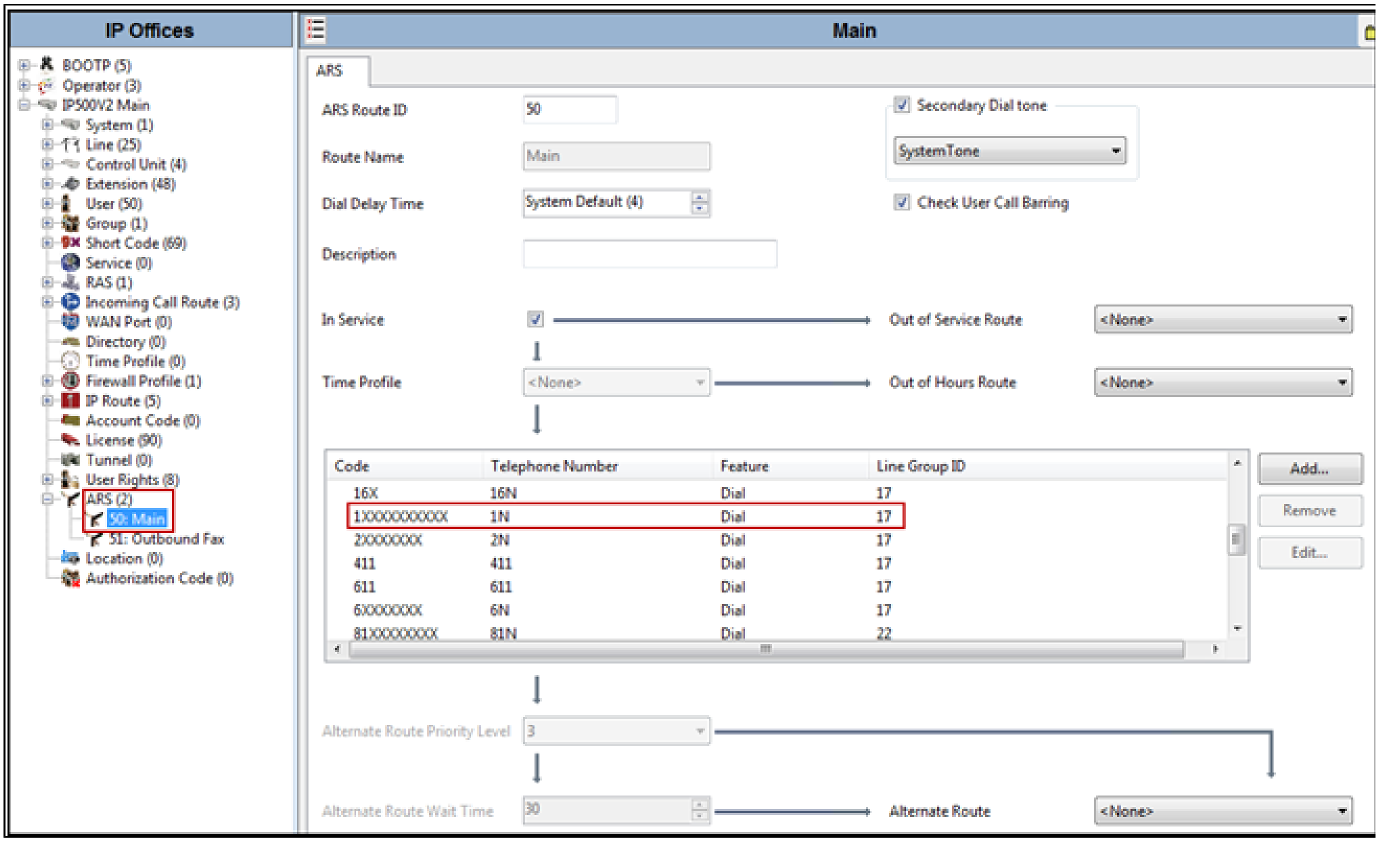


The following screen shows a sample ARS configuration for the route **50: Main**. Note the sequence of **X**'s used in the **Code** field of the entries to specify the exact number of digits to be expected, following the access code and the first set of digits on the string. This type of setting results in a much quicker response in the delivery of the call by IP Office.

Step	Action	Results
10	To create a short code to be used for ARS, select **ARS** > **50: Main** on the **Navigation** pane	
11	Select **New**	
12	In the **Code** field, enter the dial string which will trigger this short code. In this case, **1** followed by **10 X**'s to represent the exact number of digits.	
13	Set **Feature** to **Dial**.	This is the action that the short code will perform.
14	Set **Telephone Number** to **1N**.	The value **N** represents the additional number of digits dialed by the user after dialing **1** (The **9** will be stripped off).
15	Set the **Line Group ID** to the Line Group number being used for the SIP Line, in this case Line **Group ID 17** was used.	
16	Set the Locale to **United States (US English)**.	
17	Click the **OK** to commit.	
18	Procedure Completed	



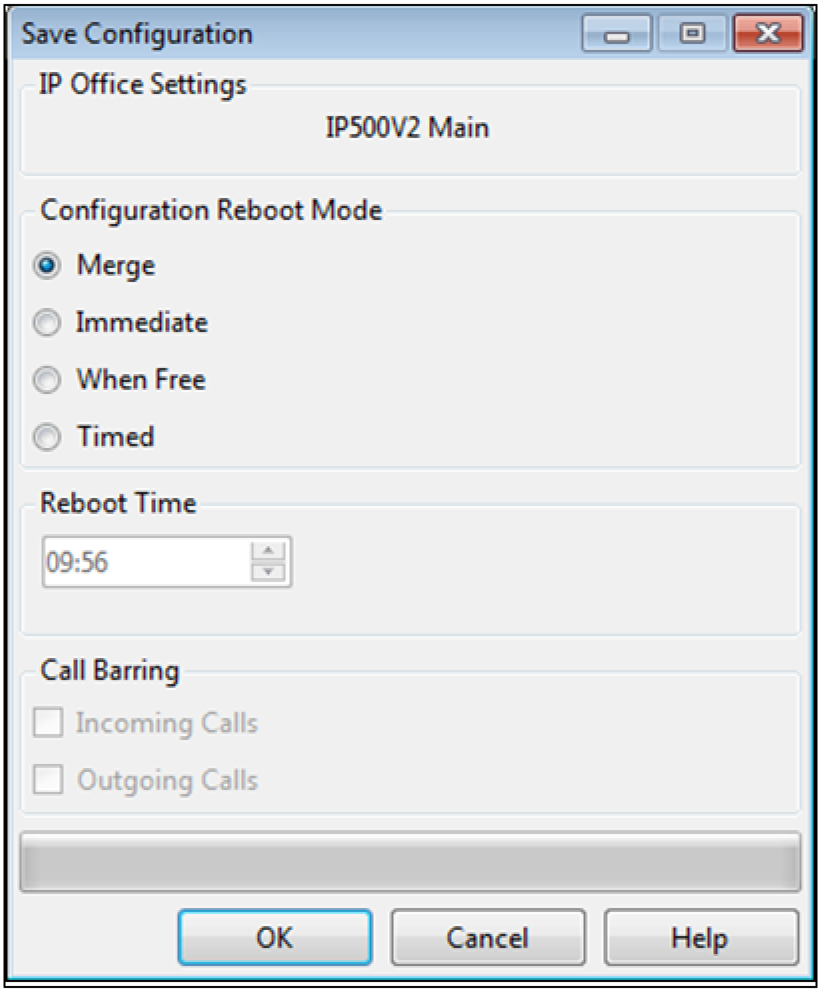
The following screenshot shows the ARS dial pattern entry after it was added.



2.9 Save Configuration

When desired, send the configuration changes made in Avaya IP Office Manager to the Avaya IP Office server in order for the changes to take effect.

Step	Action	Result
1	Navigate to **File > Save Configuration** in the menu bar at the top left of the screen	Save the configuration performed in the preceding sections. Once the configuration is validated, a screen similar to the following will appear, with either the **Merge** or the **Immediate** radio button chosen based on the nature of the configuration changes made since the last save. Note that clicking OK may cause a service disruption due to system reboot.
2	Click **OK** if desired	
3	Procedure Completed	


Save Configuration
IP Office Settings
IP500V2 Main
Configuration Reboot Mode
Merge
Immediate
When Free
Timed
Reboot Time
09:56
Call Barring
Incoming Calls
Outgoing Calls
OK
Cancel
Help