Avaya Solution & Interoperability Test Lab

Application Notes for Eircom SIP Trunk Service with Avaya IP Office Release 9.0 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Eircom SIP Trunk service and Avaya IP Office.

The Eircom SIP Trunk service provides PSTN access via a SIP trunk connected to the Eircom Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or Digital trunks. Eircom is a member of the Avaya DevConnect Service Provider program.

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.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Eircom SIP Trunk service and Avaya IP Office. Customers using this Avaya SIP-enabled enterprise solution with Eircom's SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to the Eircom SIP Trunk service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

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. Interoperability Compliance Testing

To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types including H.323, SIP, Digital and Analogue telephones at the enterprise
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider
- Outgoing PSTN calls from various phone types including H.323, SIP, Digital, and Analogue telephones at the enterprise
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider
- Calls using G.711A and G.729A codecs
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- Inbound and outbound PSTN calls to/from an IP Office Softphone clients
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance
- Caller ID presentation and Caller ID restriction
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Eircom's SIP Trunk service with the following observations:

- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked with the Emergency Services Operator.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

For technical support on Eircom products please contact Eircom Customer Care at:

- Telephone: 1800 255 255
- Telephone: +353 1 4688530
- Email: servicedesk@eircom.ie

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Eircom SIP Trunk service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 9600 Series IP Telephones (with SIP firmware), Avaya 1140e SIP Telephones, Avaya 2420 Digital Telephone, Avaya Analogue Telephone and fax machine. The site also has a Windows XP PC running Avaya IP Office Manager to configure the Avaya IP Office as well as an IP Office Softphone client and Flare Experience for Windows for mobility testing. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been changed to a private format and all phone numbers have been obscured beyond the city code.

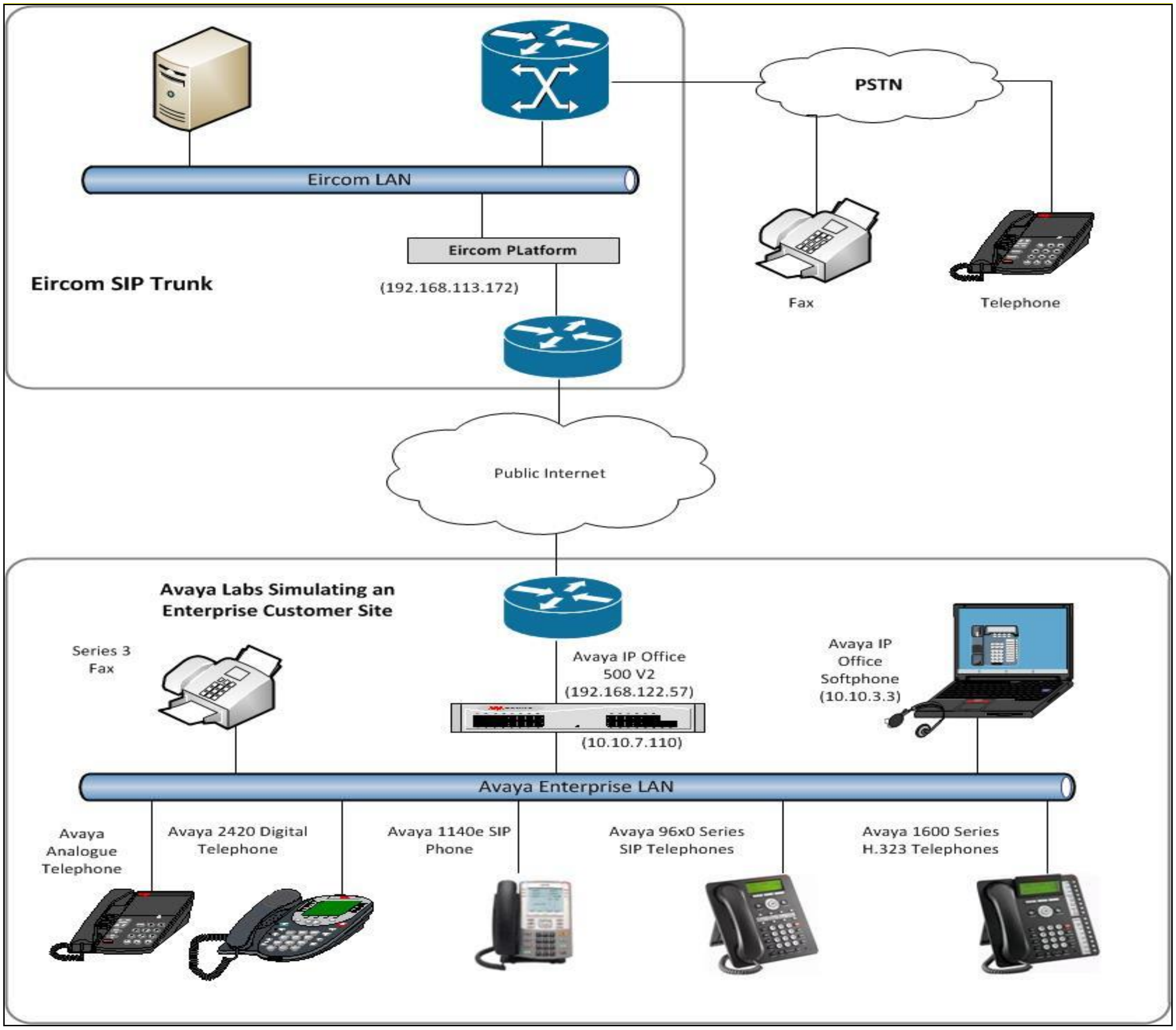


Figure 1: Test Setup Eircom SIP Trunk service to Simulated Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Version 9.0.3.0 build 941
Avaya 1608 Phone (H.323)	1.3.5
Avaya 9600 Series Phone (SIP)	6.3.0
Avaya SoftPhone (SIP)	3.056516
Avaya Flare Experience for Windows (SIP)	1.1.3.14
Avaya 1140e (SIP)	FW: 04.04.10.00.bin
Avaya 2420 Digital Phone	R6.0
Avaya 98390 Analogue Phone	N/A
Eircom	
Eircom SIP Trunk Service	Broadsoft Broadworks rel 19SP1 Ericsson IMS rel 13A AcmePacket SD running on 4500 platform, software release 6.4

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Eircom's SIP Trunk service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the 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. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider (such as twinning) is assumed to already be in place.

5.1. Verify System Capacity

Navigate to **License → SIP Trunk Channels** in the Navigation Pane. In the Details Pane verify that the **License Status** is Valid and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by Eircom.

License | Remote Server

Feature	License Key	Instances	Status	Expiry Date	Source
Office Worker	WtX400gbtXrB3O3Wqt_3rRbPd9_rdrx5	255	Valid	Never	ADI Nodal
one-X Portal for IP Office	dTna3Bo4dSj@qq6q3fmc5uwyLMdWv1L9	255	Valid	Never	ADI Nodal
Phone Manager Pro IP Audio Enabled…	MnzlkOhxVs50htJyK5pkoxglYLMpYm21	255	Valid	Never	ADI Nodal
Phone Manager Pro	JaBHzCy0Xd__hXk@gNmvxBhf1GMYg1xH	255	Valid	Never	ADI Nodal
Phone Manager Pro (per seat)	bt9Sih5rLV8F3OQBaUopgCiVNgLGYpiz	255	Valid	Never	ADI Nodal
IP Office Dealer Support - Profession…	kIHr2EoSvO233UeiKhmjBQdnA9_nS82z	255	Valid	Never	ADI Nodal
IP Office Dealer Support - Standard E…	jKM9f_d7SGlfjNWFKB79xwSFLGxwv1VM	255	Valid	Never	ADI Nodal
RAS LRQ Support (Rapid Response)	GKOlzgBLDdPdt0_pGqmf9JkZQ5N7LEse	255	Valid	Never	ADI Nodal
Report Viewer	eveekN6mEIO0ZzeiBuLjcN8uSlrYEErM	255	Valid	Never	ADI Nodal
CCC Agent Rostering	PIn_3dL@tvY@yYWuBU74rjsDp1dnwLbC	255	Valid	Never	ADI Nodal
Customer Service Agent	rU02FO5PVX8DNfCjfmue5@j61M7lY6s9	255	Valid	Never	ADI Nodal
CCR CCC UPG	sTJh@vdhQS8c@HYkg@_35IRb9LLjGJjB	255	Valid	Never	ADI Nodal
CCR Designer	faW7akviXUIz37rsADc799SAIQuIsp0c	255	Valid	Never	ADI Nodal
CCR SUP	8U288A6iXXTlFrRh32pRLJhhIZMWE7x5	255	Valid	Never	ADI Nodal
Advanced Small Community Networking	eT@t6I5TtO942yxYwI7gBIG8A0oIw_8B	255	Obsolete	Never	ADI Nodal
SIP Trunk Channels	unXMbE6x9dJKGKJ73uEpof7JrpF4smme	255	Valid	Never	ADI Nodal
Small Office Edition VCM (channels)	eABRzdgr9vhDAe9YGOuwrpqHEGuLjueM	255	Obsolete	Never	ADI Nodal
Small Office Edition WiFi	2XQ66z5GdVEFD6e61deEHwEu9LNiSQuW	255	Obsolete	Never	ADI Nodal
Small Site Software Upgrade 255	lGXCiQ5LMtVPa4TK6H7qwwReU0EJAWMJ	1	Valid	Never	ADI Nodal
Receptionist	wGM8o16nddQm2mr@UMm9LwA@XRFhgM0e	255	Valid	Never	ADI Nodal
Software Upgrade 255	wqxeh6@XLO1ZV@MFkjo6BCY@IdFyggZu	1	Valid	Never	ADI Nodal
CCC Spectrum Wallboards	MAXOapb3VvFZdn614ep_kkR2GEZVSw@C	255	Valid	Never	ADI Nodal

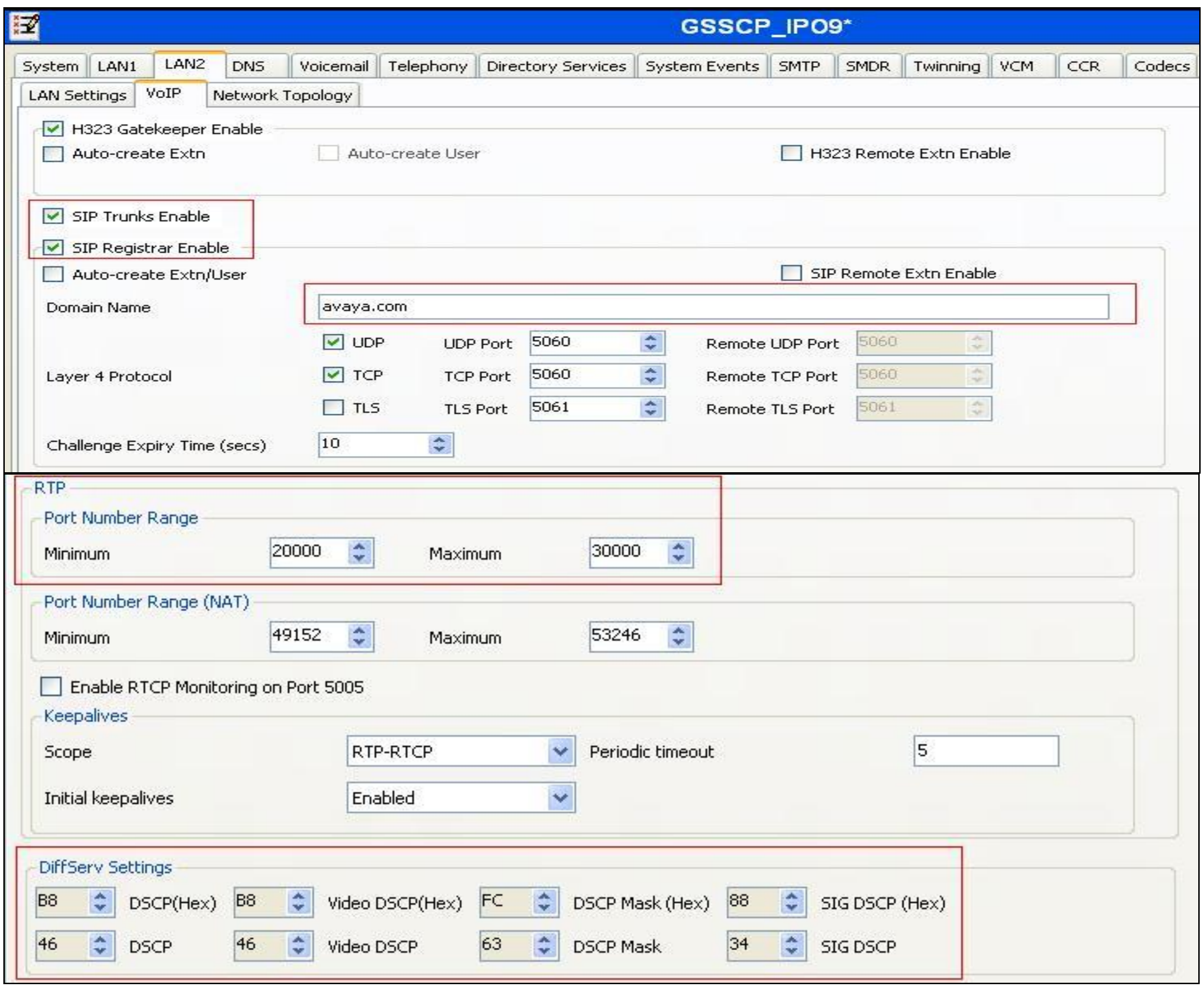
5.2. LAN2 Settings

In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO2** in the Navigation Pane where GSSCP_IPO9 is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).

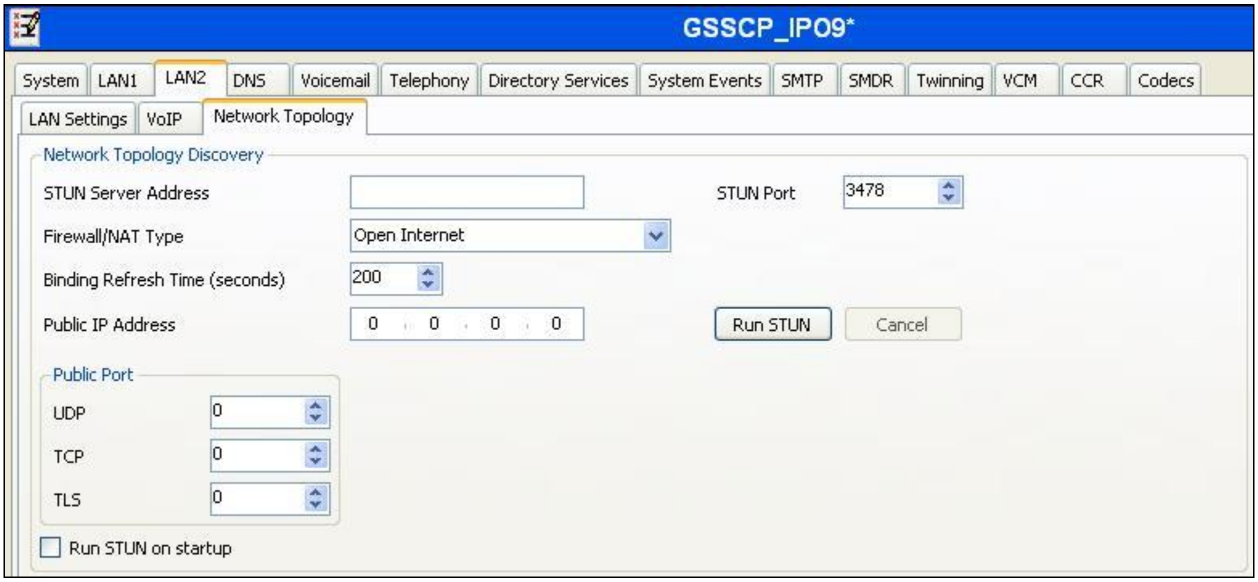
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **Domain Name** has been set to the customer premises equipment domain "**avaya.com**". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN 2 IP Address. All other parameters shown are default values.

The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signalling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signalling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

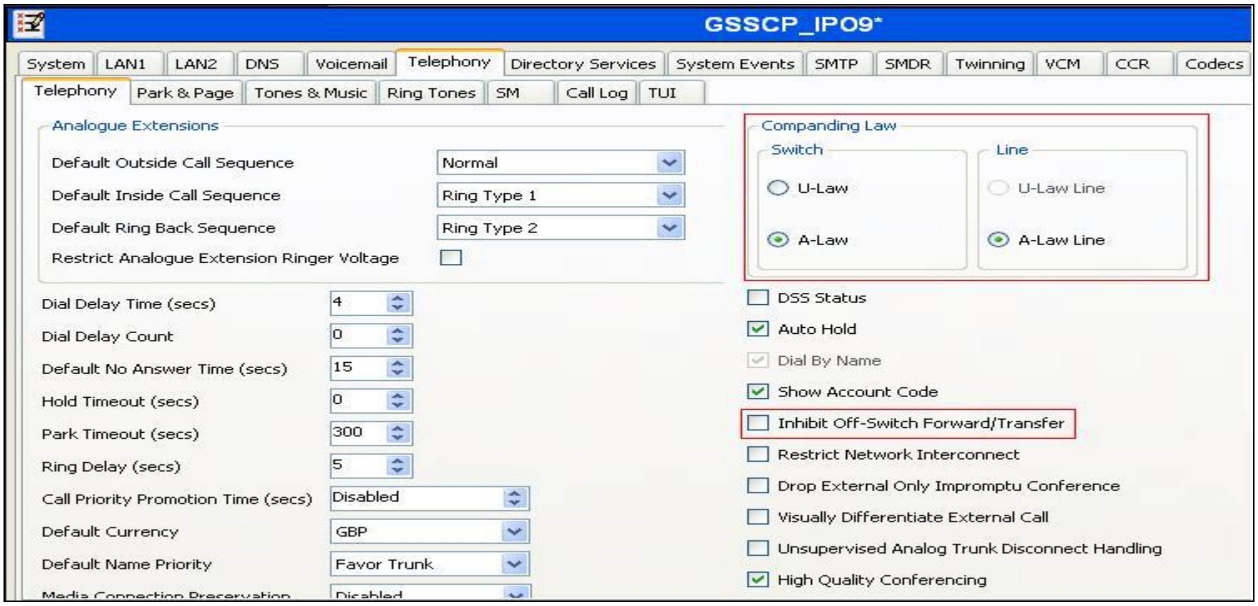


Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to "None" in **Section 5.6.2**. It is important that the **Binding Refresh Time** is set to the correct value. Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. Below is a sample configuration. On completion, click the **OK** button (not shown).



5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **ALAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk. On completion, click the **OK** button (not shown).



5.4. System Twinning Settings

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked, and the **Calling party information for Mobile Twinning** is left blank in the reference configuration. With this configuration, the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the Eircom SIP Trunk.

5.5. Codec Settings

Navigate to the **Codecs** tab on the Details Pane. Check the Available Codecs boxes as required. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test, **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** were the supported codecs used for testing.

5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Eircom 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 IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** 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 Credentials (if applicable)
- 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 5.6.2**.

Also, the following SIP Line settings are not supported on Basic Edition:

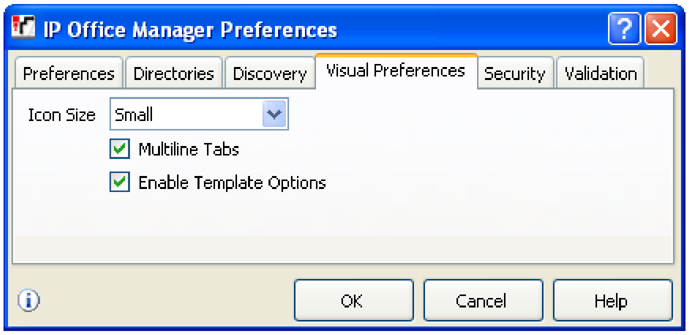
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

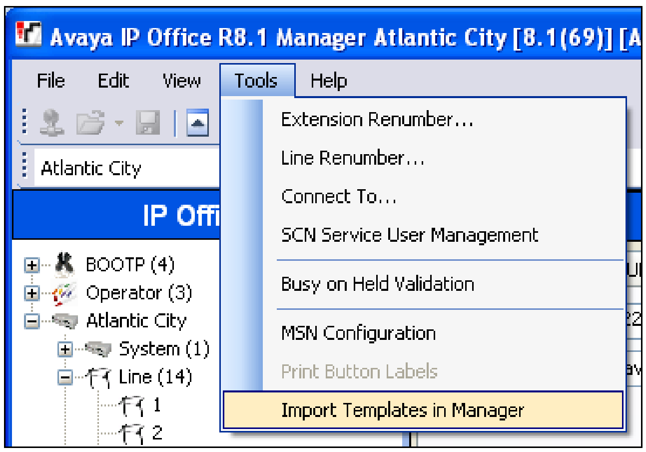
5.6.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **IE_Eircom_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

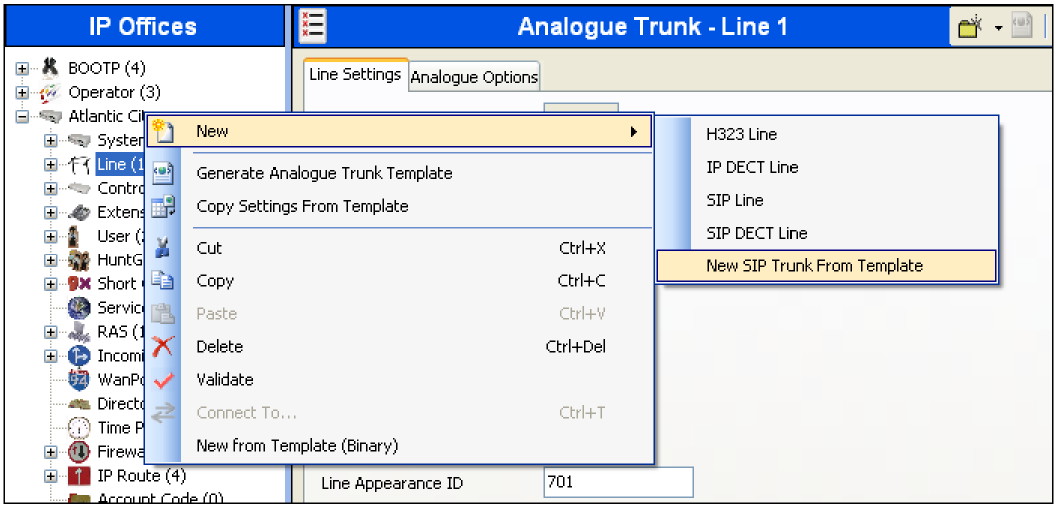


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



5. In the subsequent Template Type Selection pop-up window, select **Ireland** from the **Country** pull-down menu and select **Eircom** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name **(IE_Eircom_SIPTrunk.xml)** created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.

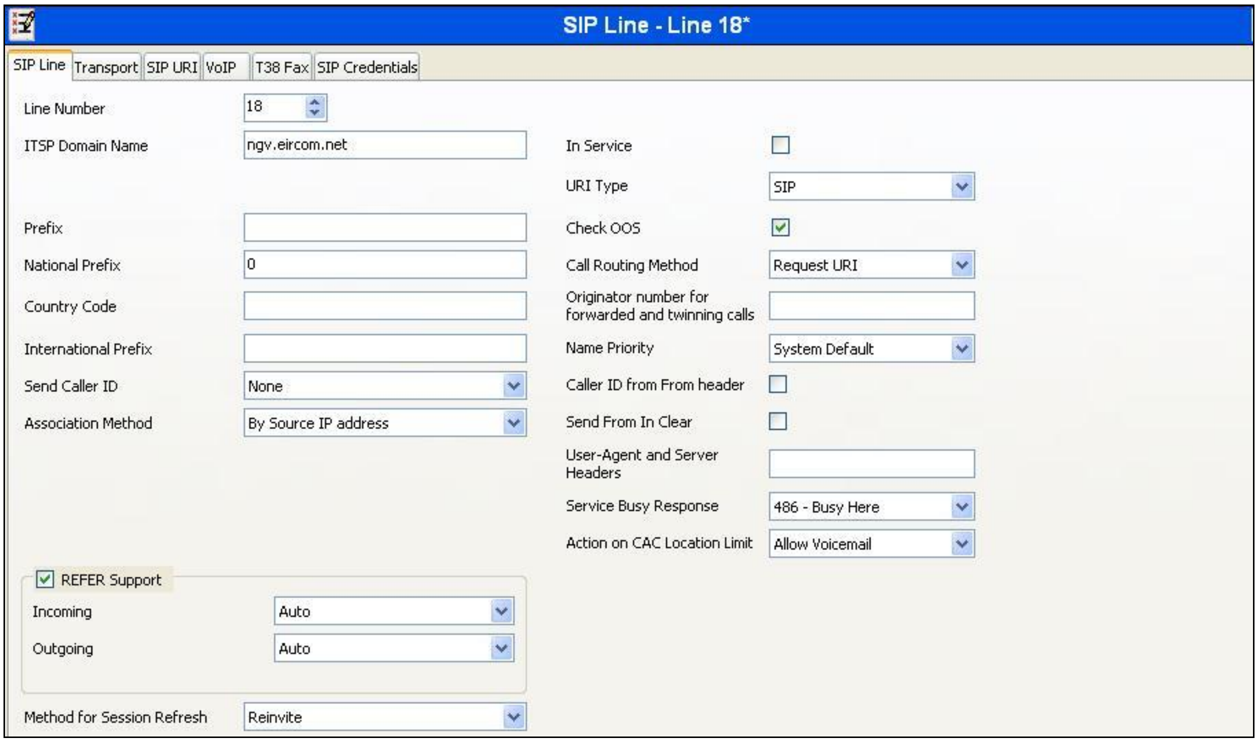
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

5.6.2. SIP Line – SIP Line Tab

On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to the SIP Trunking service.

- Set **ITSP Domain Name** to **ngv.eircom.net.**
- Ensure the **In Service** box is checked.
- Set **REFER Supported** to Auto.
- Set **Method for Session Refresh** to **Reinvite**.
- Ensure the **Check OOS** box is checked.
- Default values may be used for all other parameters.

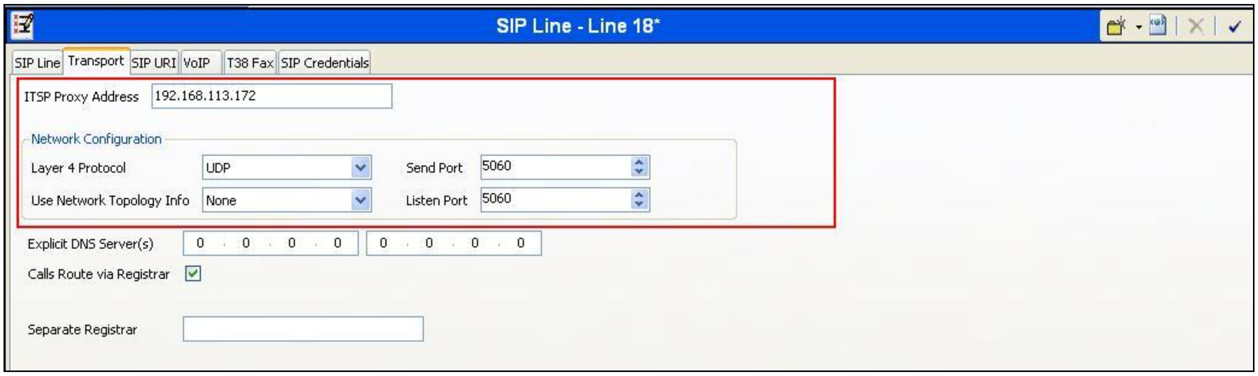
On completion, click the **OK** button (not shown).



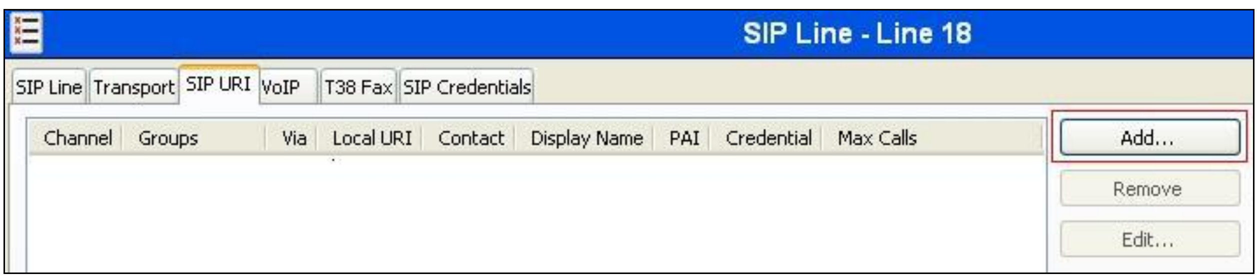
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the Eircom SIP proxy
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** to **5060** and **Listen Port** to **5060**
- Set **Network Topology Info** to **None**

On completion, click the OK button (not shown).

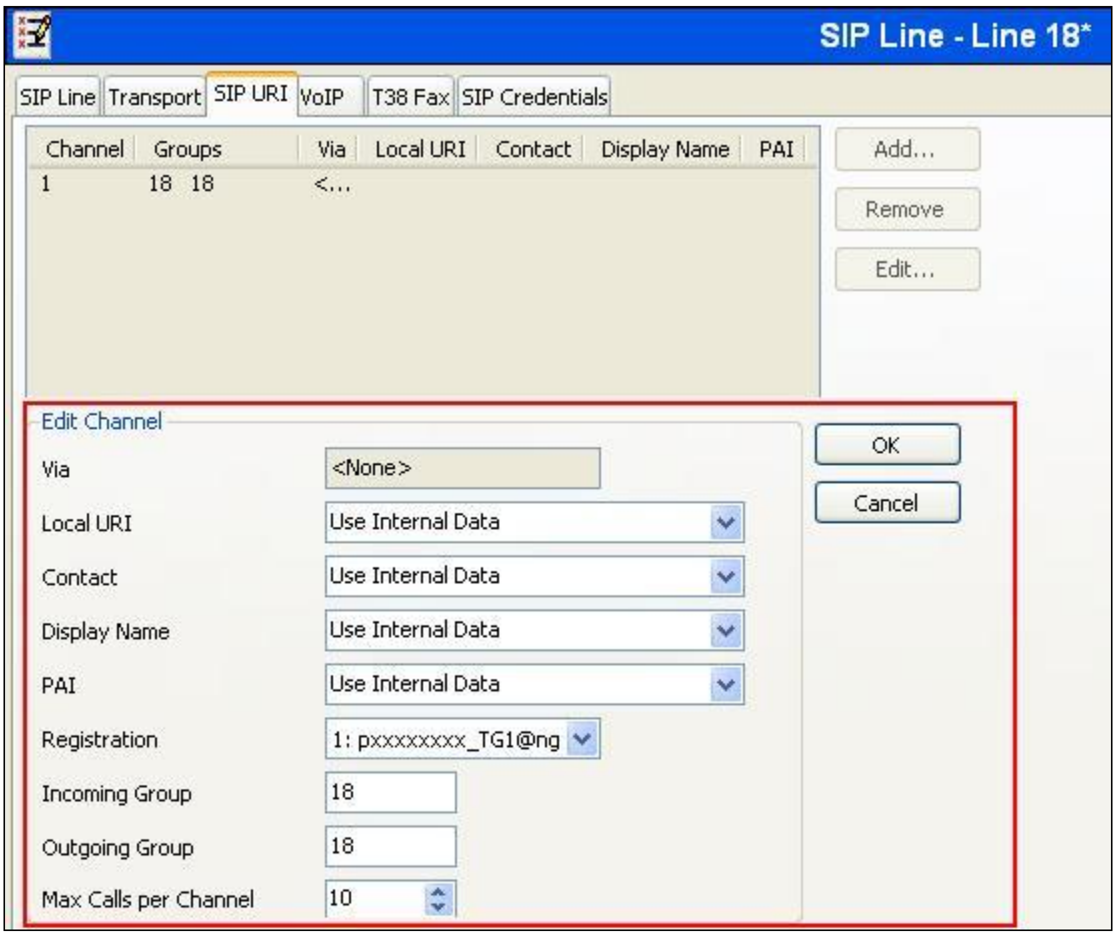


After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

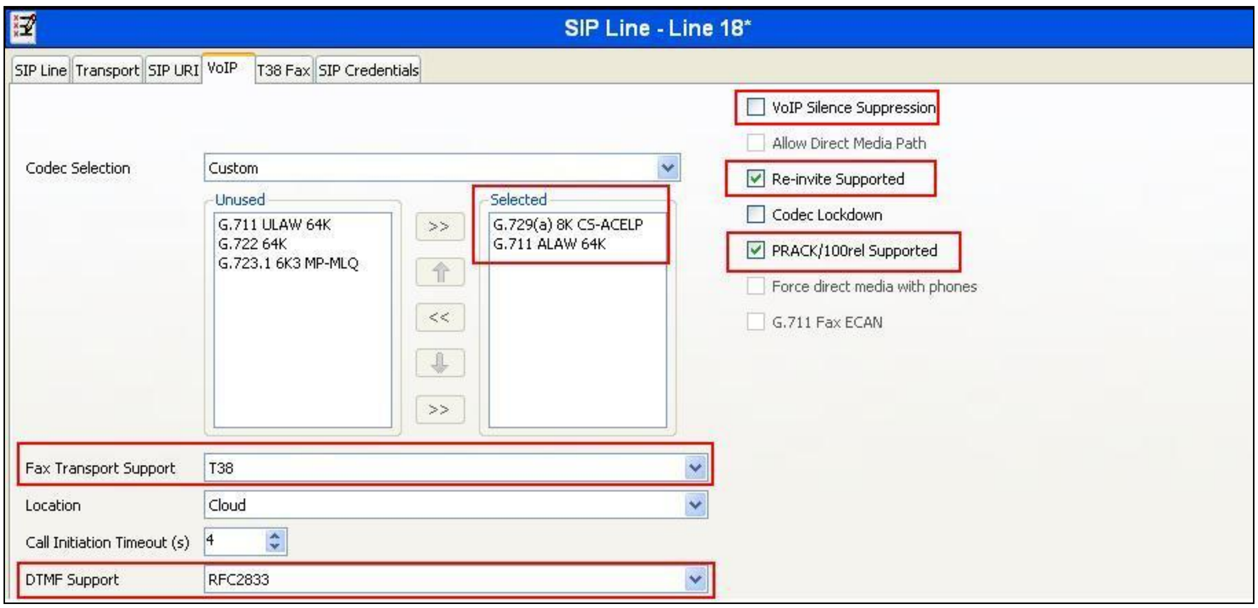
- Set **Local URI, Contact, Display Name and PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.8**.
- For **Registration**, select **1: <pxxxxxxxx@ngv.eircom.net>** from the pull-down menu since this configuration uses SIP registration.
- 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. Similarly, 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 **18** was defined that was associated to a single line (line 18).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.



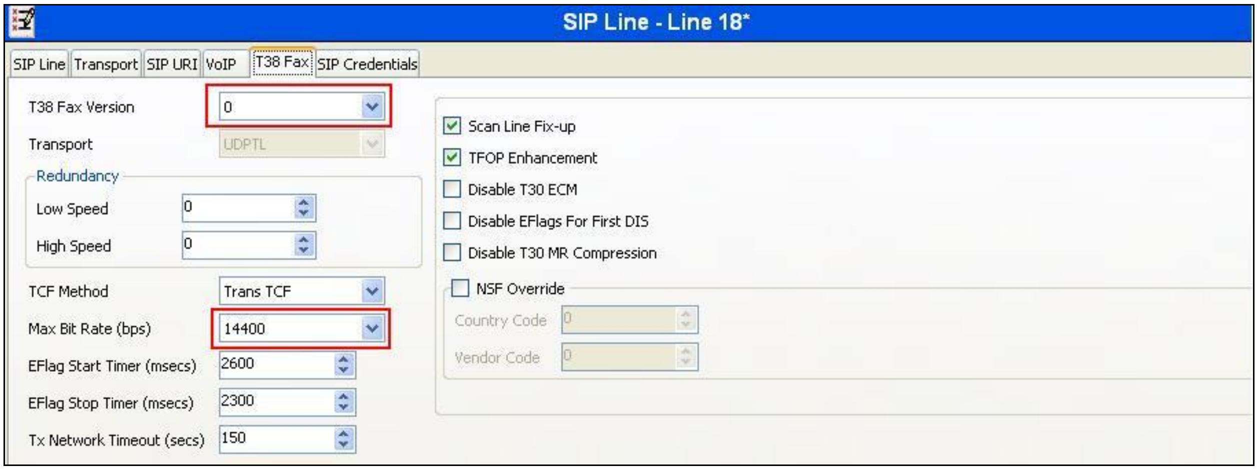
Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- Select **Custom** from the drop-down menu.
- Select **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** codecs.
- Select the **Fax Transport Support** box to **T.38**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Uncheck the **VoIP Silence Suppression** box.
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check **PRACK/100rel Supported** to advertise the support for provisional responses and Early Media to the Eircom network.

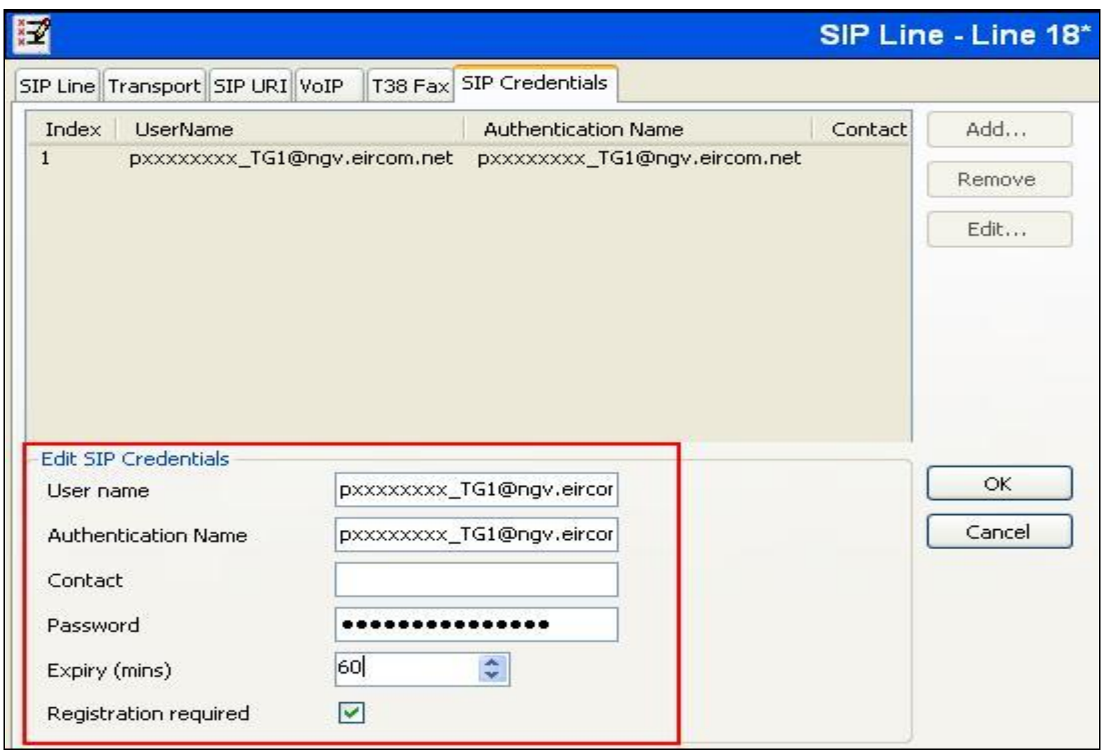
Default values may be used for all other parameters.



Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the Use Default Values box (not shown) and select **0** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate (bps)** to **14400**. All other field may retain their default values. On completion, click the **OK** button (not shown).



Select the **SIP Credentials** tab to administer registration details provided by Eircom. This allows the SIP Trunk to authenticate to the Eircom SIP Trunk service. Choose **Add** (not shown) and enter the registration credentials provided by Eircom as shown below. Click the **OK** button to complete the SIP line administration.



Note: It is advisable at this stage to save the configuration as described in **Section 5.11**.

5.7. Short Codes

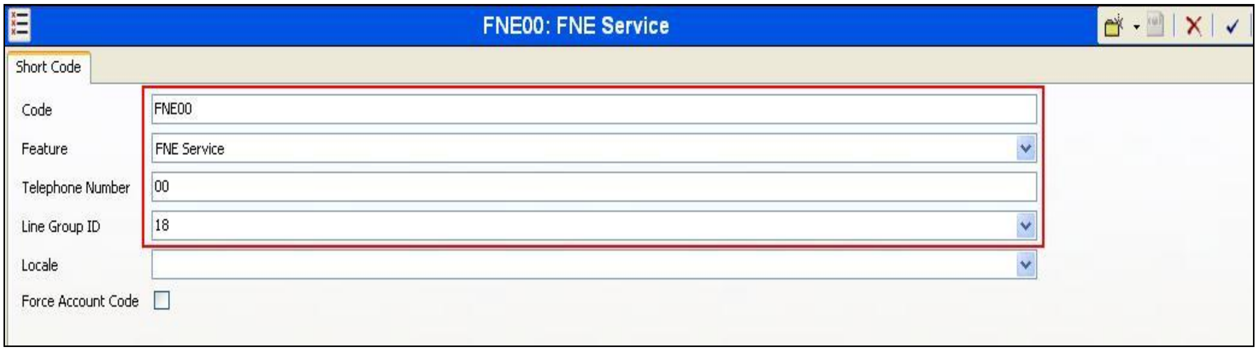
Define a short code to route outbound traffic to the SIP line and route incoming calls from mobility extensions to access Feature Name Extensions (FNE) hosted on IP Office. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The example shows **9N**; which will be invoked when the user dials 9 followed by the dialed number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **0N** which will insert the prefix **0** before a dialled number **N** which is a requirement by Eircom for all outbound calls to terminate successfully. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6**

On completion, click the **OK** button (not shown).

For incoming calls from mobility extension to FNE features hosted by IP Office to provide dial tone or mobile callback functionalities, Short Code **FNE00** was created.

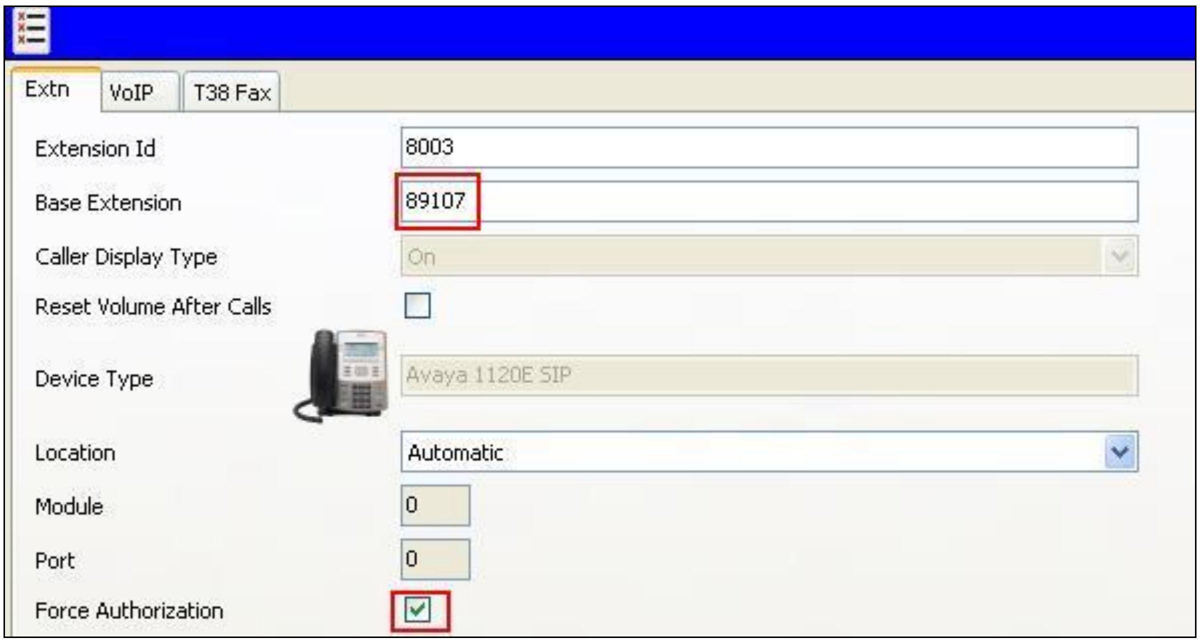
- In the **Code** field, enter the FNE feature code as **FNE00**.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00** for **FNE00**.
- Set the **Line Group Id** to **18** which is the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6**.



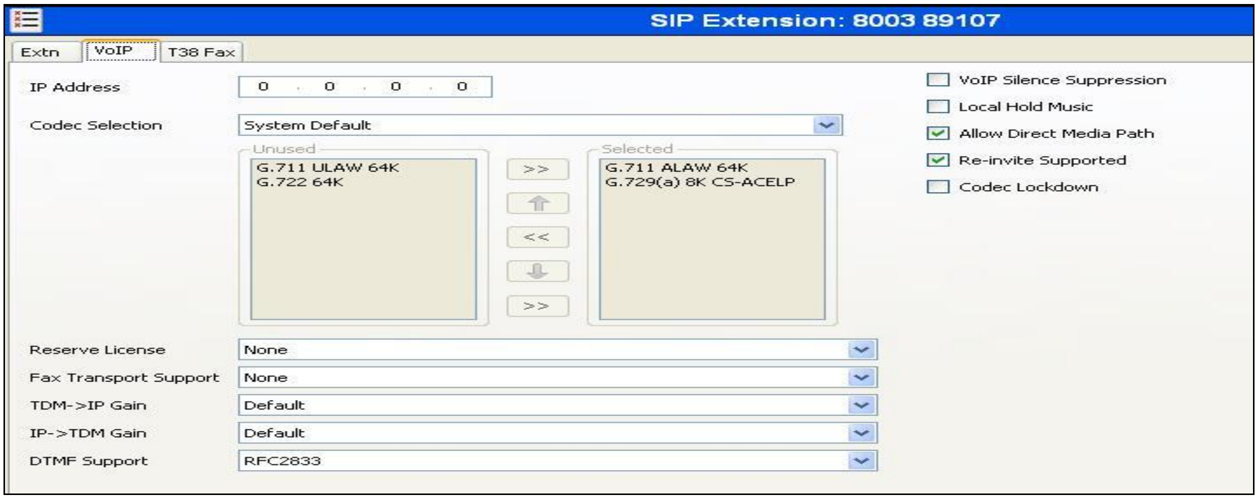
5.8. User and Extensions

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users.

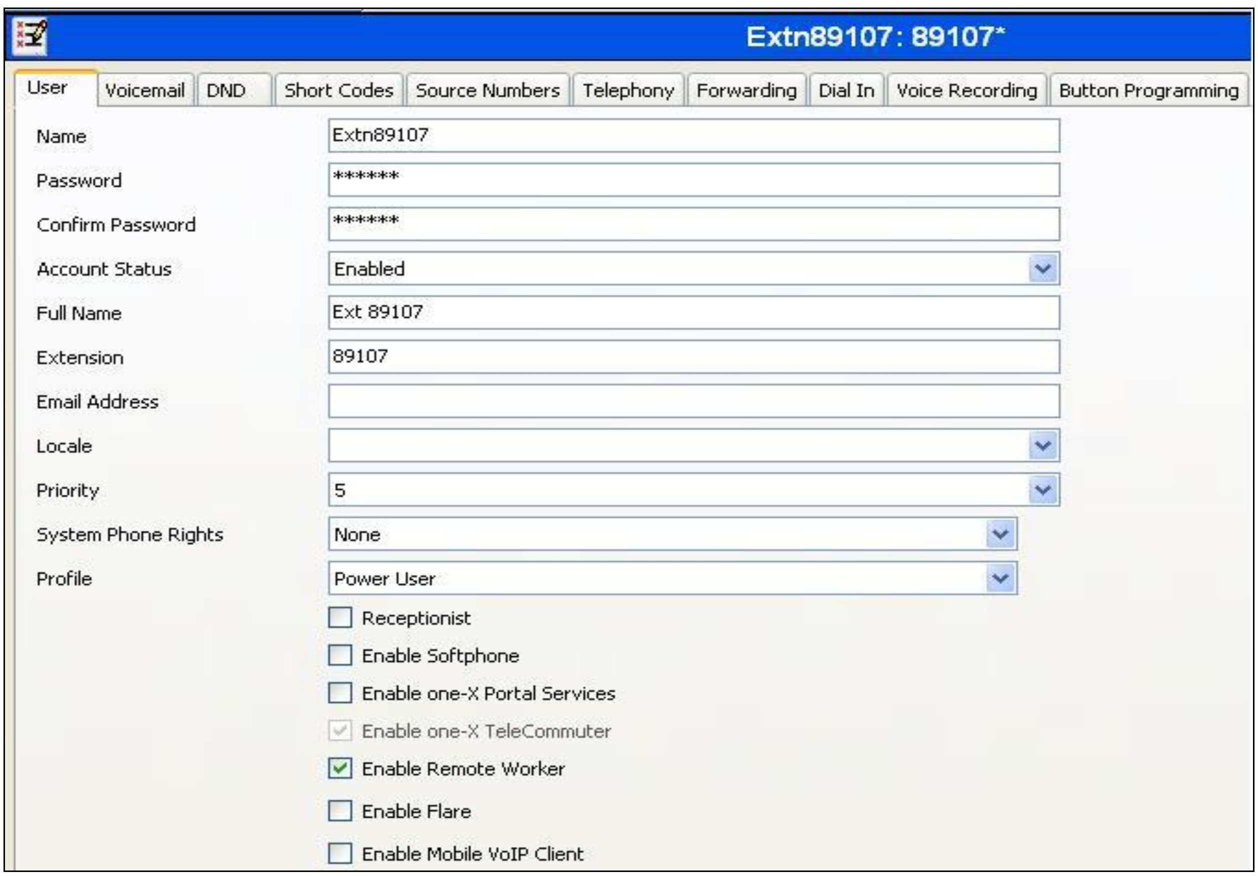
A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with 89107, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.



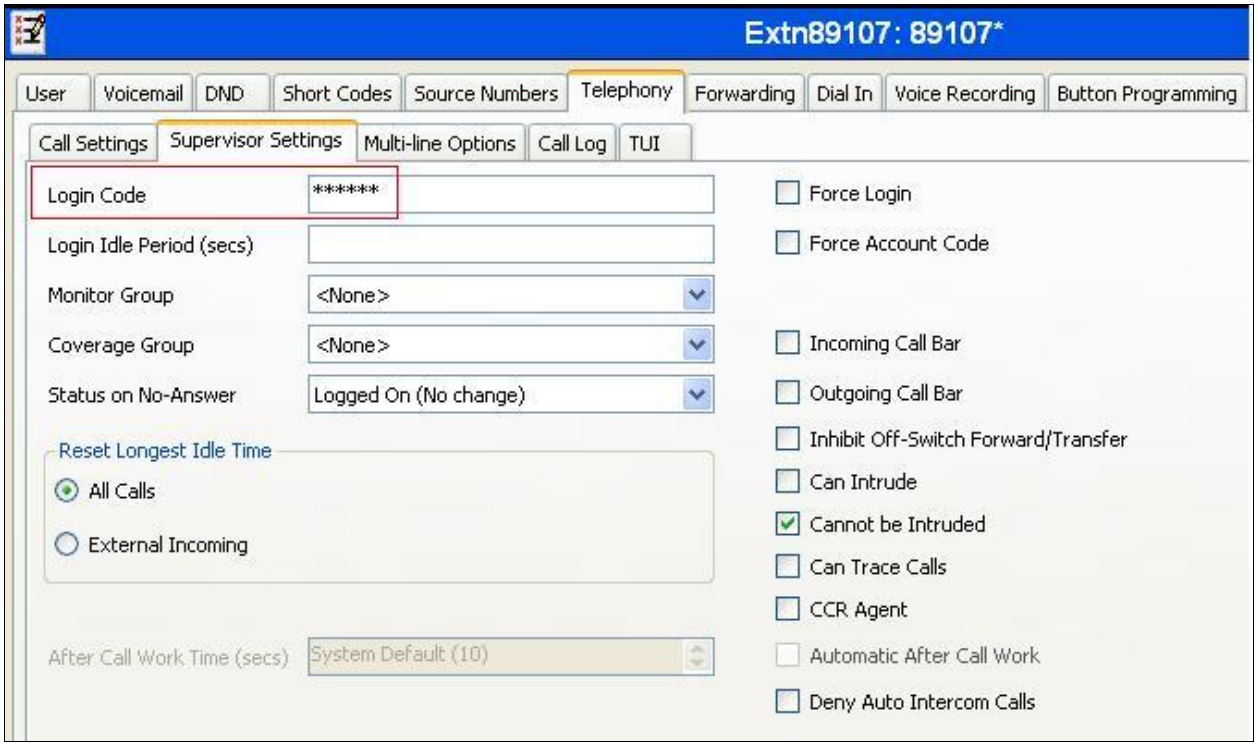
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank or populated with a static IP address. Check the **Reserve Avaya IP endpoint license** box. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in **Section 5.5**. Alternatively, “Custom” may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.



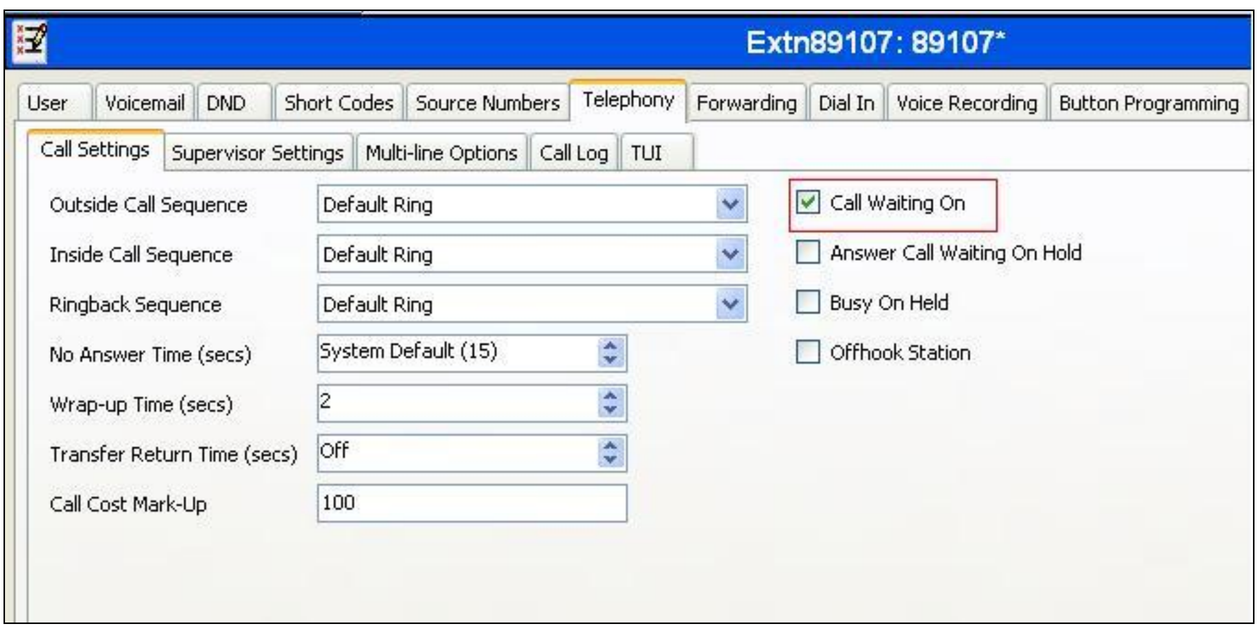
To add a User, right click on **User** in the Navigation pane, and select **New**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane. Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, select the **User** tab if any changes are required. The example below shows the changes required to use Avaya 1140E which was used in test.



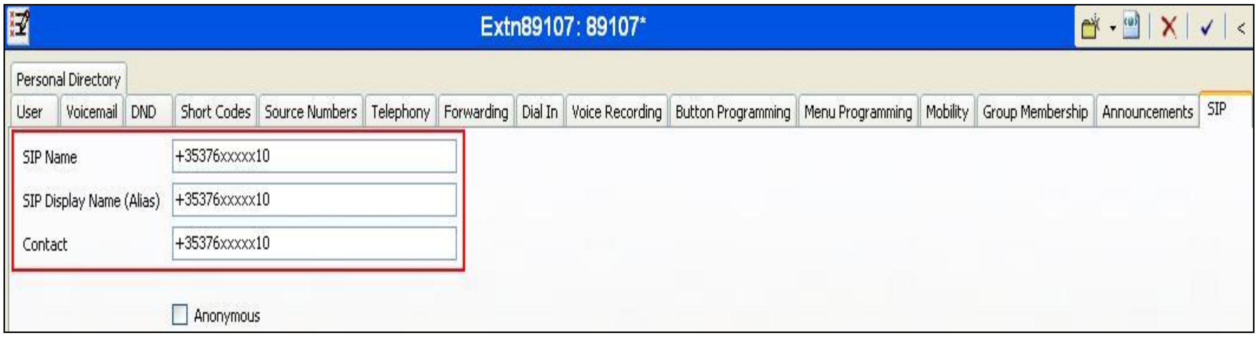
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



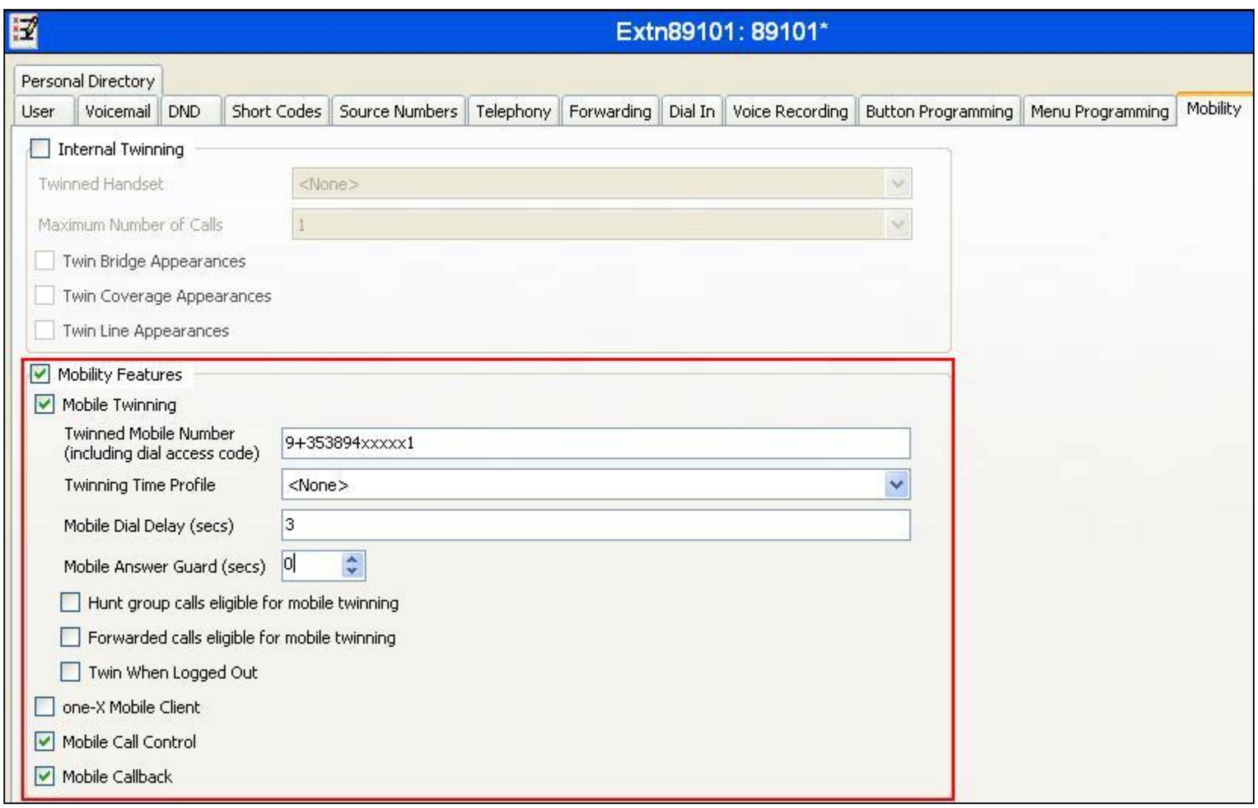
Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow incoming notifications and transfer operations.



Next, select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details Pane until it becomes visible. The values entered for the SIP **Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Eircom.



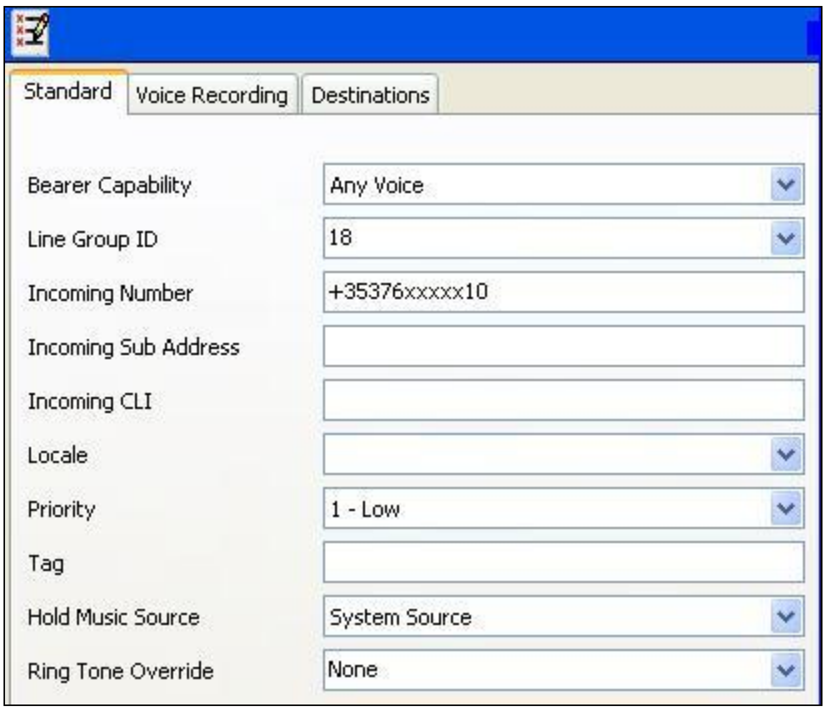
The following screen shows the Mobility tab for user 89101. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.



5.9. Incoming Call Routing

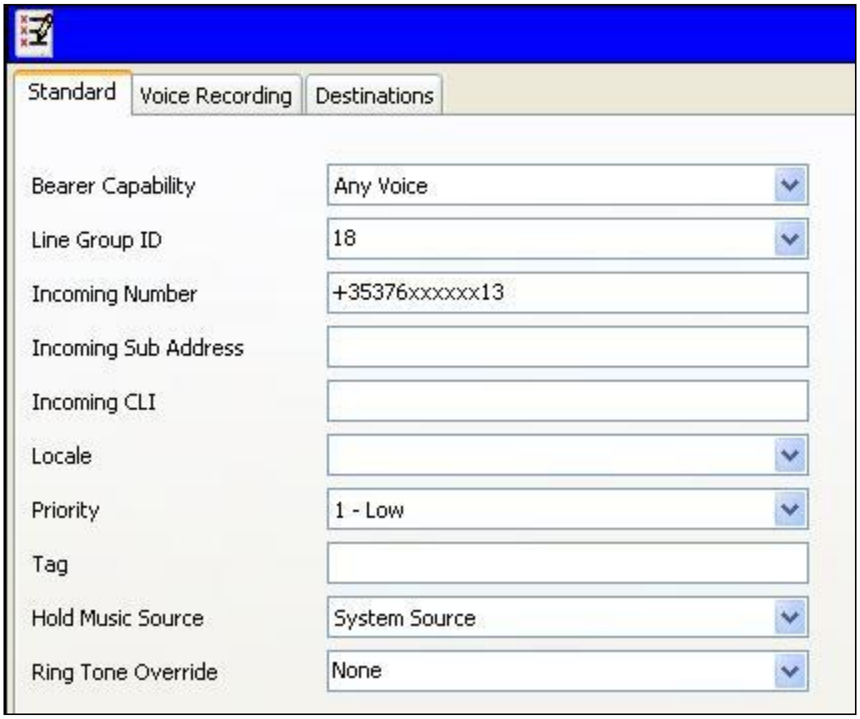
An incoming call route maps an inbound DDI number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Route**s in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to **Any Voice.**
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.**
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89107.

Incoming Call Routes for other direct mappings of DDI numbers to IP Office users or FNE short codes etc. can be configured in the same fashion. In the screenshot below, the incoming call route for **+35376xxxxxx13** mapped to a shortcode **FNE** is illustrated.



The **Destinations** tab contains the Destination field FNE00 which was entered manually. An incoming call to **+35376xxxxxx13** will be delivered directly to internal dial tone allowing the caller to perform dialing actions to both internally and external calls. The incoming caller ID must match the Twinned Mobile Number entered in User Mobility tab (**Section 5.8**); otherwise, IP Office responds with a 486 Busy Here and busy tone.

5.10. Privacy / Anonymous Calls

There are multiple methods for a user to withhold outgoing identification:

- Dialing the short code *67 to access the SIP Line.
- Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user.
- Avaya Telephones equipped with a “Features” button can also request privacy for a specific call, without dialing a unique short code, using **Features → Call Settings → Withhold Number**, on the phone itself.

To configure IP Office to include the caller’s DID number in the P-Asserted-Identity SIP header, required to admit an otherwise anonymous caller to the network, the following procedure may be used.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field, type **SIP_USE_PAL_FOR_PRIVACY**. Click **OK**.

The source number **SIP_USE_PAI_FOR_PRIVACY** should now appear in the list of Source Numbers as shown below.

5.11. 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. Eircom SIP Trunk Service Configuration

Eircom is responsible for the configuration of the SIP Trunk service. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. Eircom will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

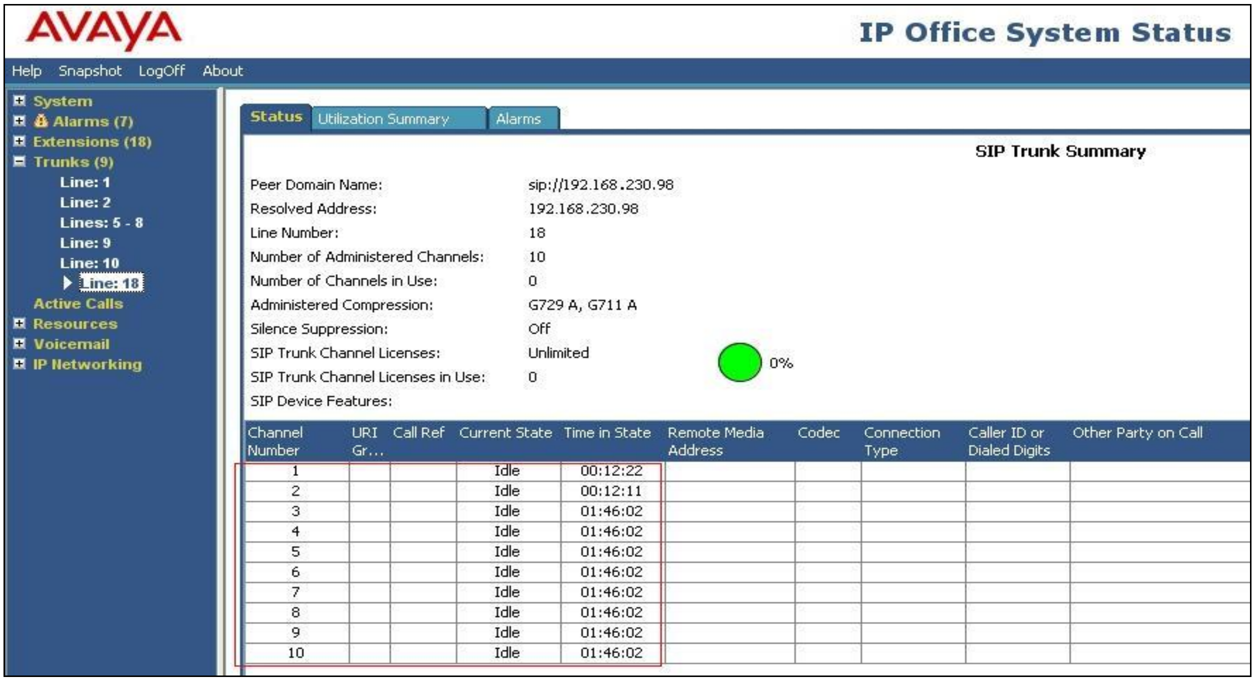
7. Verification Steps

This section includes steps that can be used to verify that the configuration has been done correctly.

7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found on the PC where IP Office Manager is installed in PC programs under **Start →All Programs →IP Office →System Status** (not shown).
Log into IP Office System Status at the prompt using the **Control Unit IP Address** for the IP office. The **User Name** and **Password** are the same as those used for IP Office Manager.

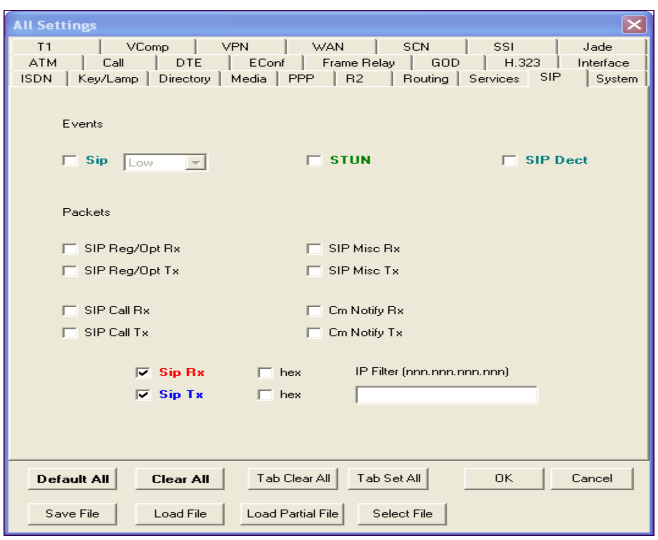
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational. IP address has been changed.



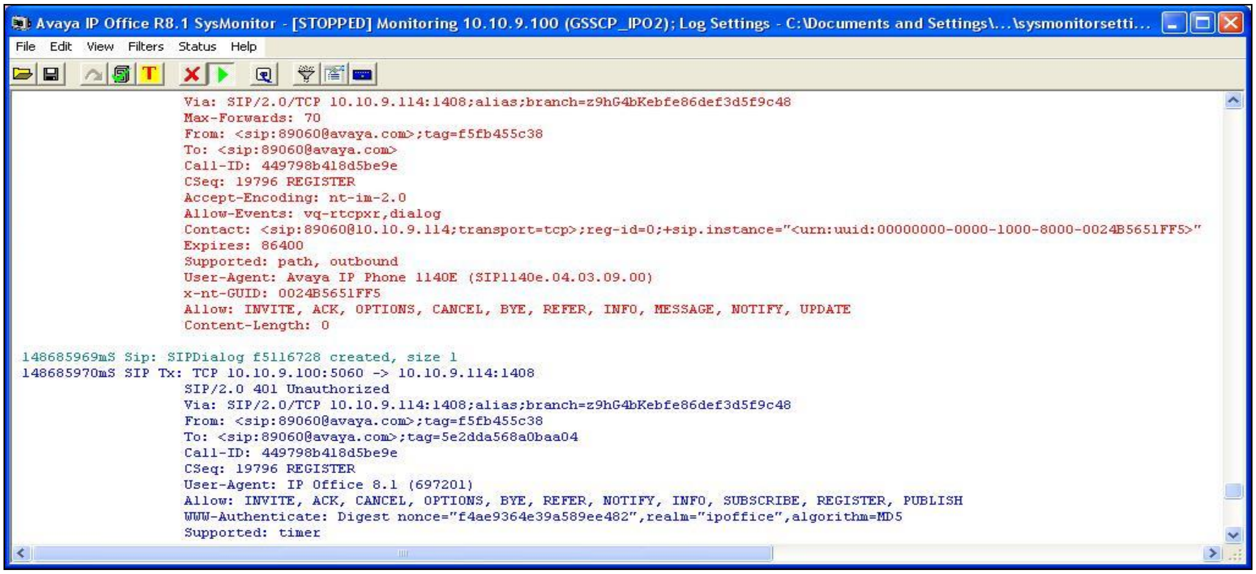
7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters →Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window for a Registration attempt to the SIP trunk.



8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and Eircom SIP Trunk solution as shown in **Figure 1**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya IP Office can be configured to interoperate successfully with Eircom's SIP Trunk service. Eircom's SIP Trunk service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

9. Additional References

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

[1] *Avaya IP Office 9.0* Documentation CD, Jan 2014.
[2] *IP Office 9.0 Installation Manual,* Document Number 15-601042, Jan 2014.
[3] *IP Office Manager Manual 10.0,* Document Number 15-601011, Jan 2014
[4] *IP Office Release 9.0 Implementing Voicemail Pro*, Dec 2013
[5] *System Status Application*, November 2013
[6] *IP Office Softphone Installation*, September 2014
[7] *IP Office SIP Extension Installation*, October 2014
[8] *Avaya IP Office Knowledgebase*, http://marketingtools.avaya.com/knowledgebase
[9] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.