Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.0 with HIPCOM SIP Trunk – Issue 1.0

Abstract

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

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

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 HIPCOM SIP Trunk and Avaya IP Office. HIPCOM SIP Trunk provides PSTN access via a SIP trunk connected to the HIPCOM network as an alternative to legacy Analogue or Digital trunks. This approach generally results in lower cost for customers.

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 HIPCOM SIP Trunk. 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.

Testing was performed with IP Office 500 v2 R9.0, but it also applies to IP Office Server Edition R9.0. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R9.0 to support analog or digital endpoints or trunks. IP Office Server Edition does not support TAPI Wave or Group Voicemail

2.1. Interoperability Compliance Testing

Avaya IP Office was connected to the HIPCOM SIP Trunk. 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, Digital and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from HIPCOM
- Outgoing PSTN calls from various phone types including H.323, Digital, and Analogue telephones at the enterprise.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to HIPCOM
- Inbound and outbound PSTN calls to/from an IP Office Softphone client
- Various call types including: local, international, toll free (outbound) and directory assistance
- Codecs G.711A, G.729A and G.711MU
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833

- 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 the HIPCOM SIP Trunk with the following observations:

- Outbound call to unallocated numbers and no replywere treated by playing announcements from the network. This worked well, but gave no indication in signalling as to the cause of the failure.
- Inbound Call attempts with no matching codecs in the SDP resulted in "488 Not Acceptable Here" being sent from the IP Office. When it received this, the network re-attempted the call several times before playing a tone to the caller. This resulted in a period of silence of approximately 15 seconds.
- Outbound call attempts with no matching codecs received a response from the network of "480 Temporarily Unavailable". A more commonly used response is "488 Not Acceptable Here"
- During DTMF testing, the Avaya Galway voicemail system was dialled to test menu navigation using DTMF digit. It was found that when a hash (#) was pressed on a SIP phone, it was not recognized by the system. All other digits were recognized, however, and hash was recognized by two other IVR systems dialled. This was taken to be an issue on local test systems and set-up.
- Toll Free access was not available for incoming calls and was not tested.
- Access to Emergency Services was not tested as no test call was booked with the Emergency Services Operator.
- The network call hold test was not a true test of SIP trunk functionality as it was handled in the network and there was nothing in signalling to indicate the call was on hold. Similarly, IP Office provides no indication in signalling to the network that a call is on hold.
- Outbound fax was found during test to be unreliable with the first attempt failing after 6 pages of a 10 page fax. This was taken to be an issue specific to the test network.

2.3. Support

For technical support on HIPCOM products please contact an authorized HIPCOM representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to HIPCOM SIP Trunk. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 9600 Series IP Telephone (with SIP firmware), an 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 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.

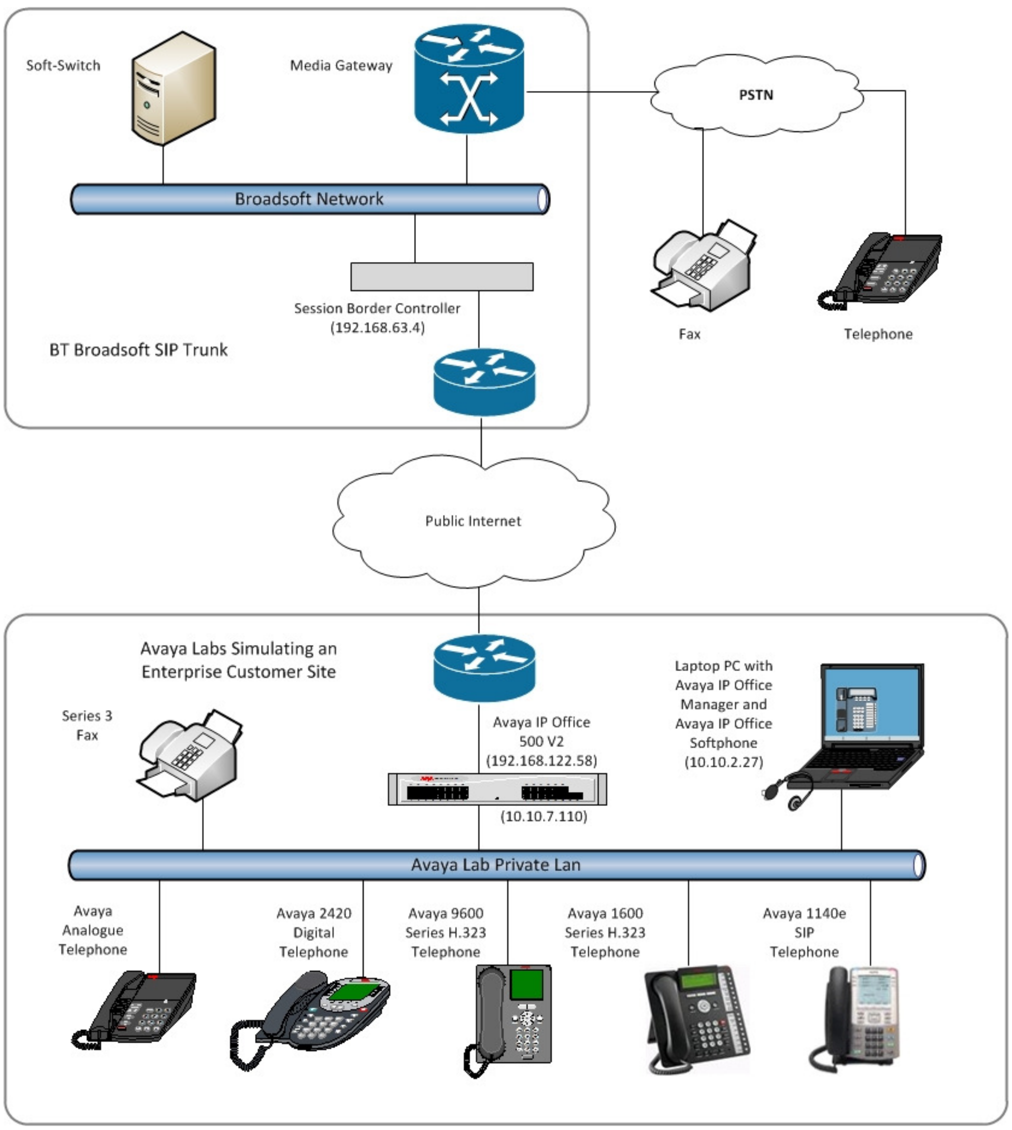


Figure 1: HIPCOM SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

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

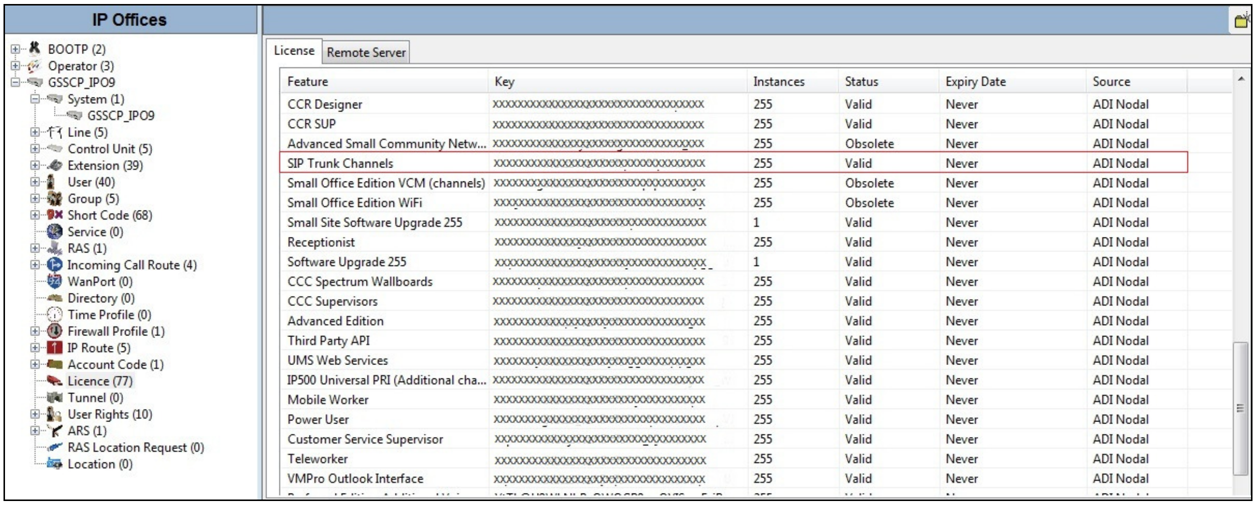
Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Avaya IP Office R9.0 (829)
Avaya 1140e SIP Telephone	04.03.09.00
Avaya 1603 Phone (H.323)	1.3.3
Avaya 9620 Phone (H.323)	3.2.0
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Avaya Softphone	3.2.3.48 (67009)
HIPCOM	
ACME Net-Net 4500 SBC	v6.1
BroadWorks	v17sp4

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the HIPCOM SIP Trunk. 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.

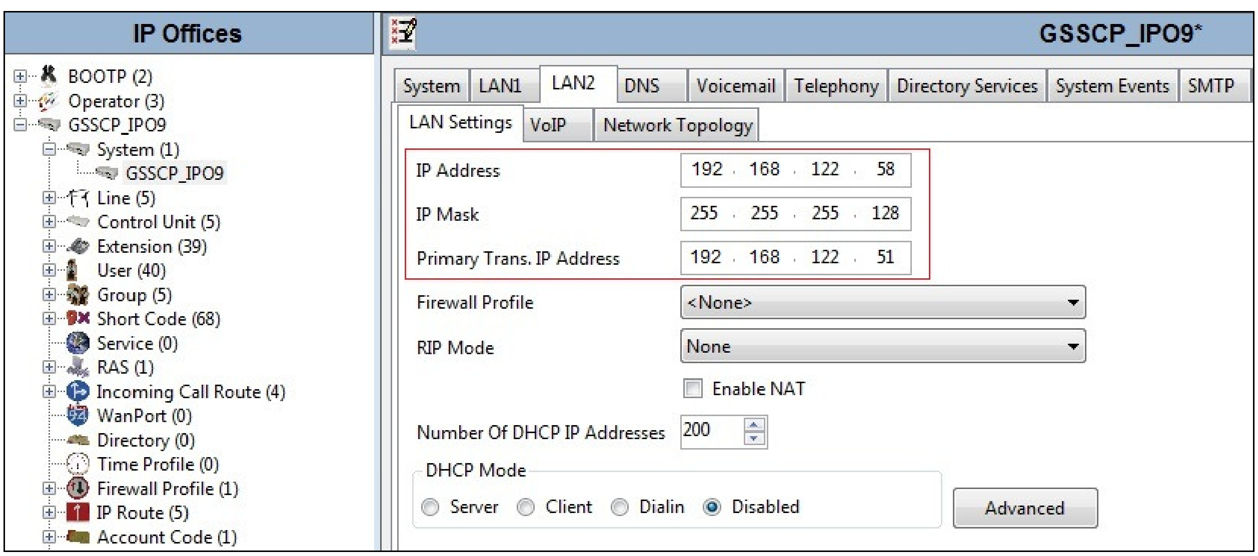
5.1. Verify System Capacity

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

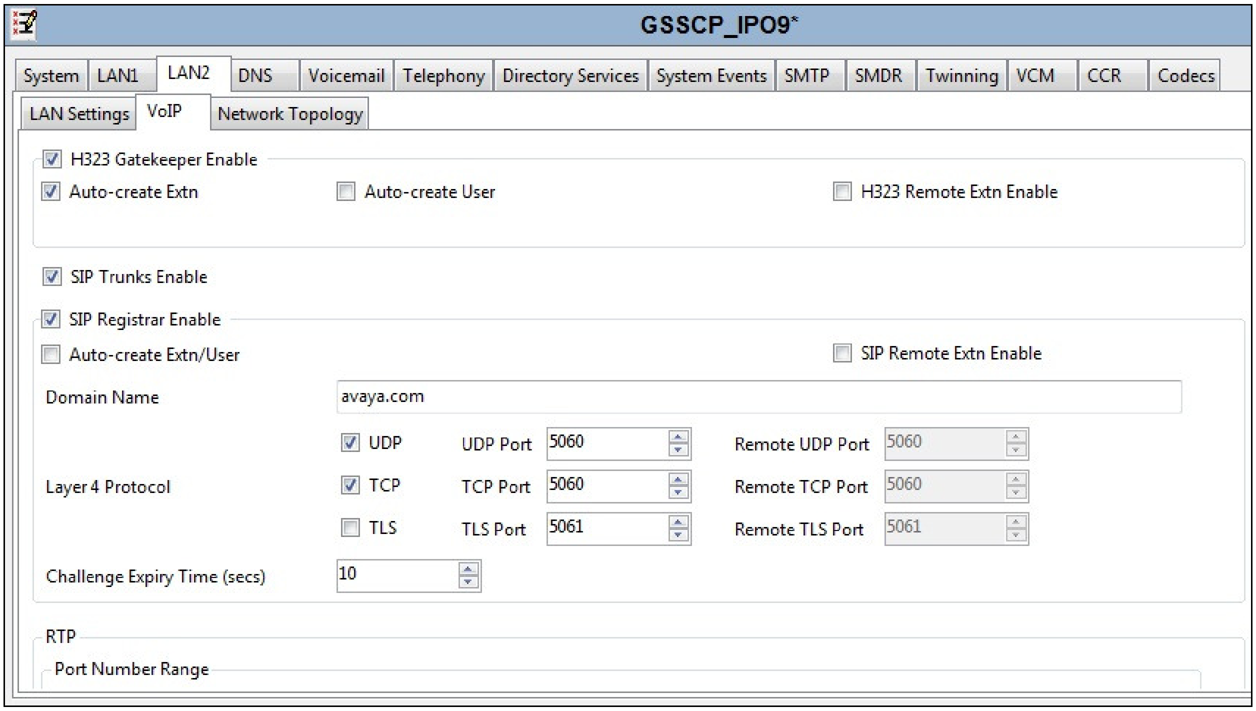


5.2. LAN2 Settings

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 → <IP Office Name>** in the Navigation Pane where IP Office Name is the name of the IP Office. This is **GSSCP_IPO9** in the GSSCP test environment. 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, **Primary Trans. IP Address** is the next hop, usually the default gateway address. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).



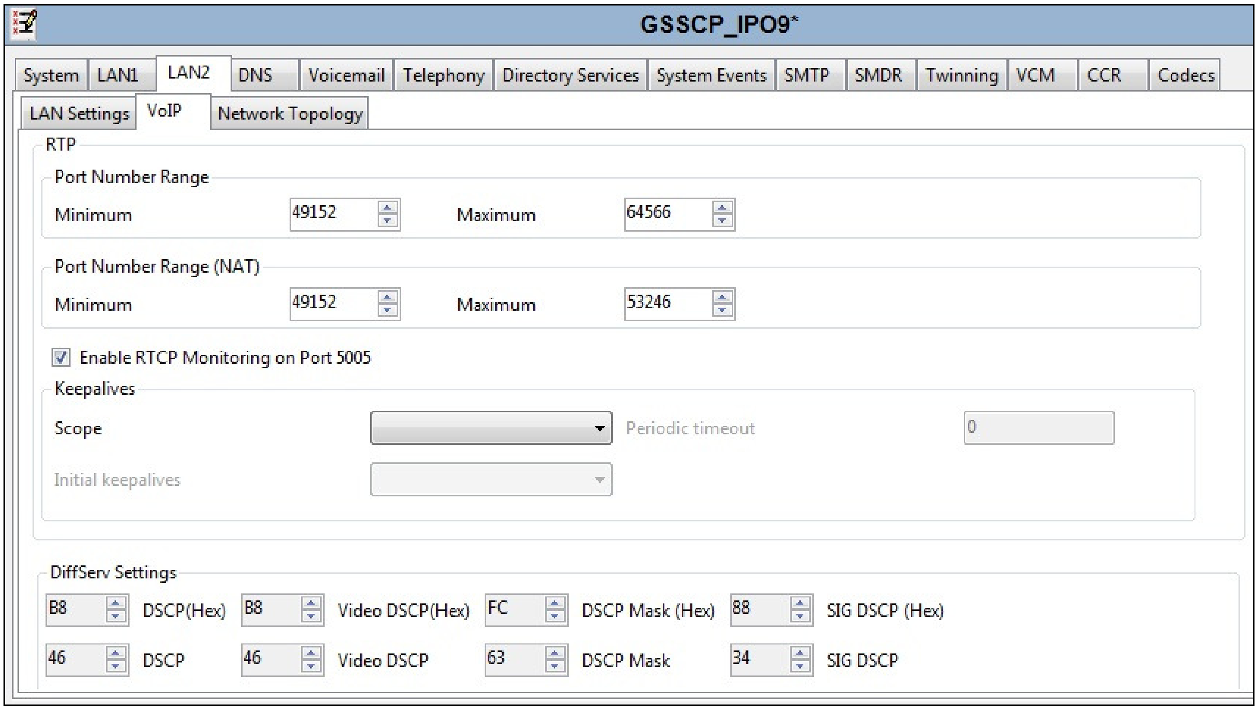
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. Define the port to be used for the signalling transport, in the test environment **UDP** was used and it the port number was left at the default value of **5060**.



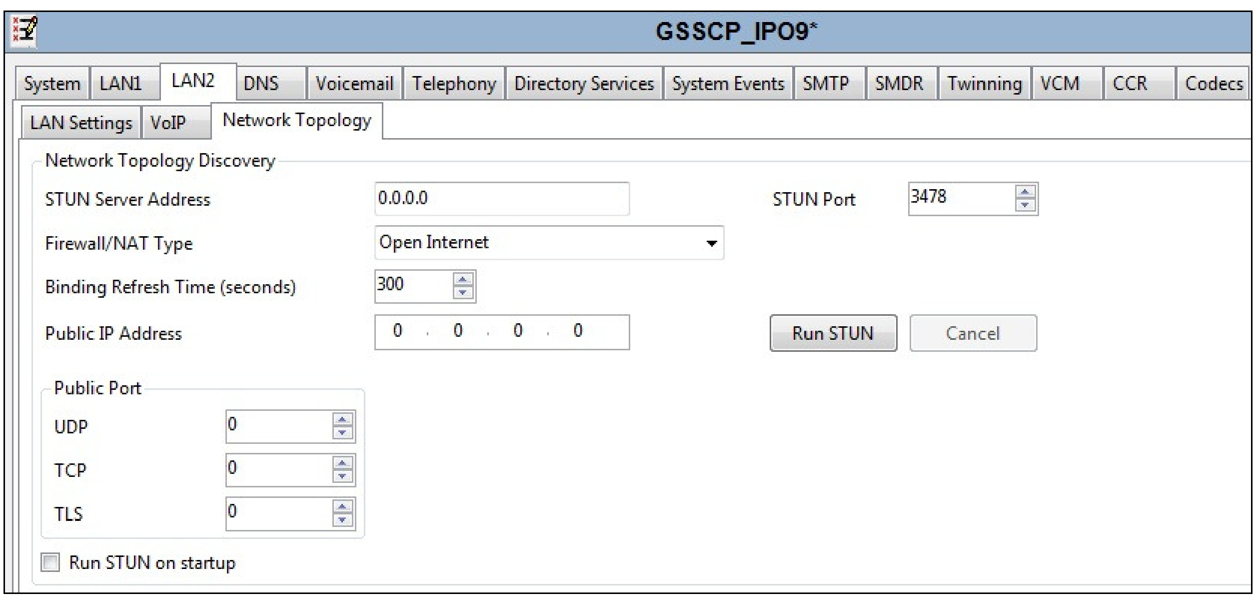
Scroll down for further configuration. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

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. DSCP for media can be set for both voice and video. The **DSCP** field is the value used for voice, **Video DSCP** is the value used for video and the **SIG DSCP** is the value used for signalling. For the compliance test, the DSCP values were left at their default values which are shown in the example over the page.

All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

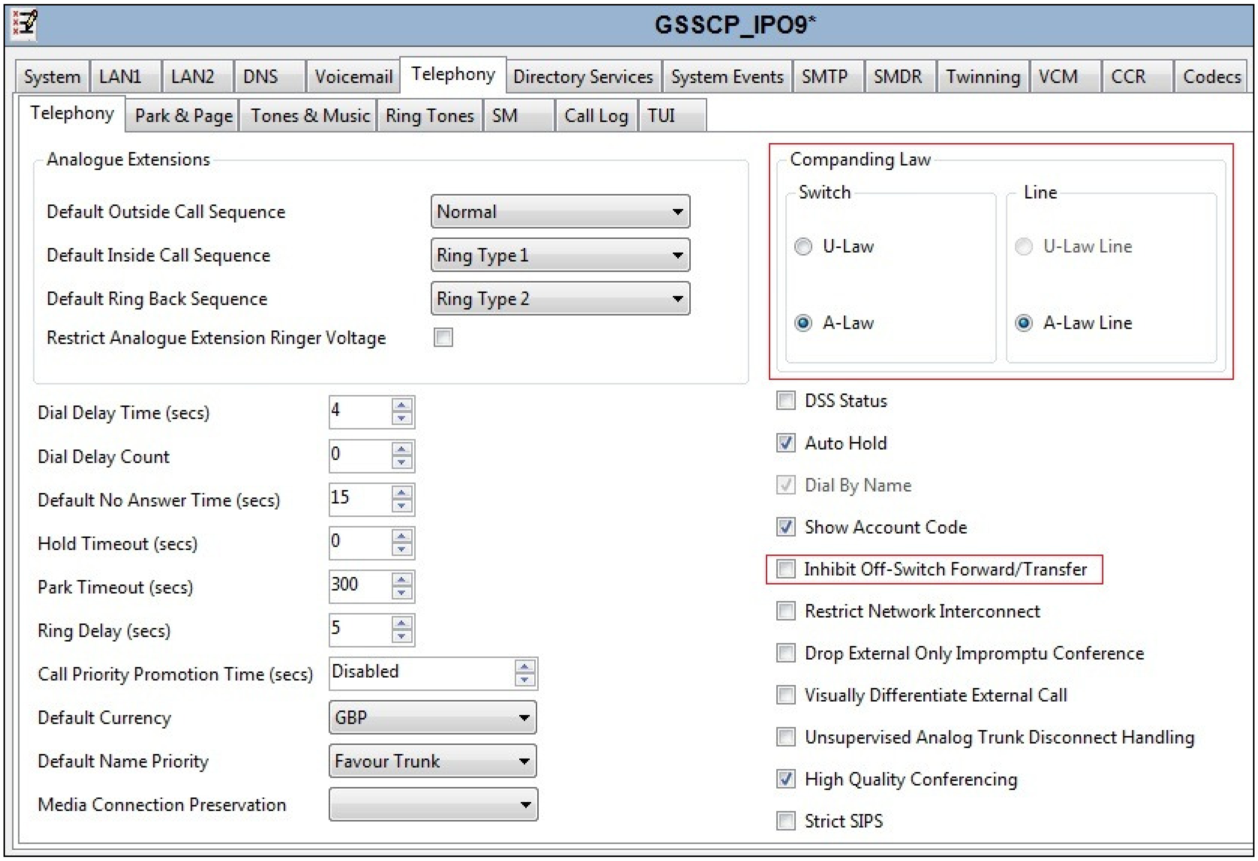


On the **Network Topology** tab in the Details Pane enter the information required if NAT is to be used. During test, NAT was not required and there was no requirement for a STUN server. To disable this facility, 0.0.0.0 is entered in the **STUN Server IP Address** and **Public IP Address** fields. If NAT is to be used, this tab can also be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS



5.3. System Telephony Settings

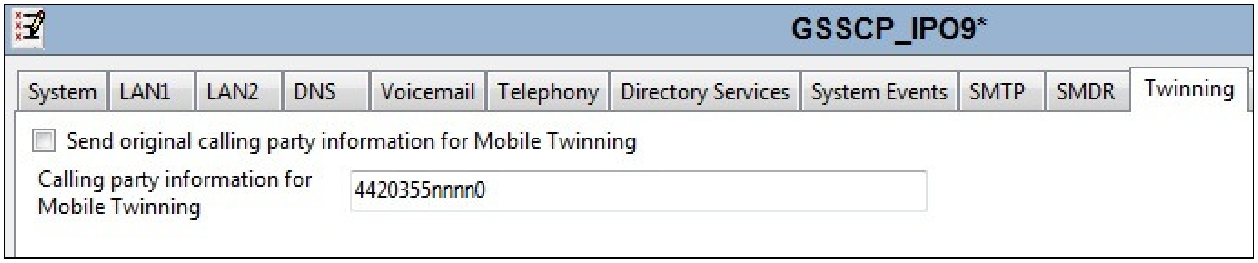
Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **ALAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across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



5.4. System Twinning Settings

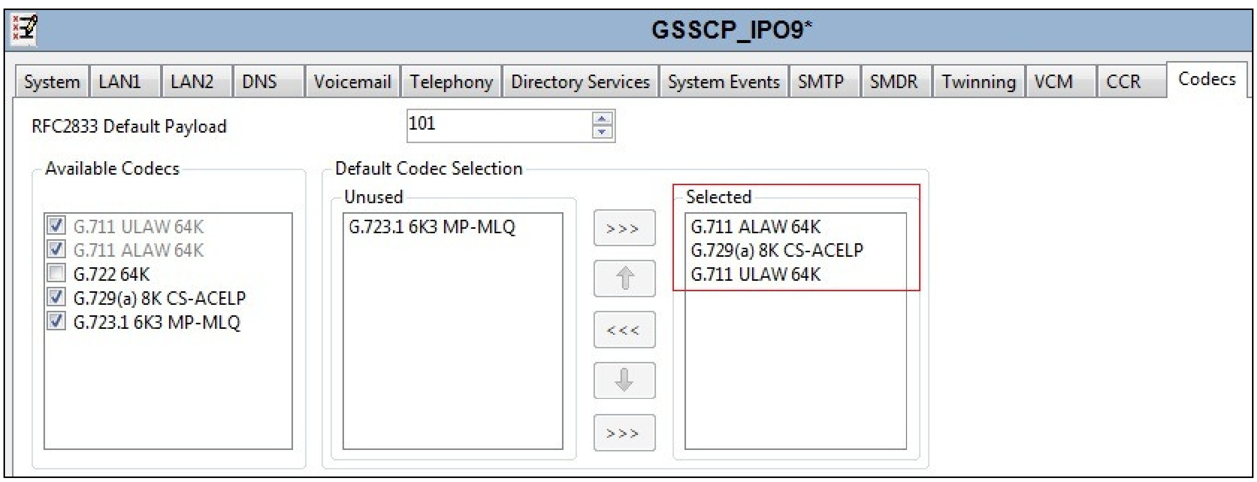
Navigate to the **Twinning** tab, ensure that the box labeled **Send original calling party information for Mobile Twinning** is not checked. HIPCOM verifies the calling number in the SIP From header, so this has to be a number in the DDI range assigned to the IP Office. During test, the number assigned as the calling number was the same as that used for SIP registration messages. To assign a number, enter it in the **Calling party information for Mobile Twinning** field.

These settings only affects twinning and do not impact the messaging of other redirected calls such as forwarded calls. If a number other than the **Calling party information for Mobile Twinning** field is to be used, set the **Send Caller ID** parameter on the SIP line (**Section 5.6**). On completion, click the **OK** button (not shown).



5.5. Codec Settings

Navigate to the **Codecs** tab (not shown) 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.711 ALAW 64K**, **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** were used. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the HIPCOM SIP Trunk. 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:

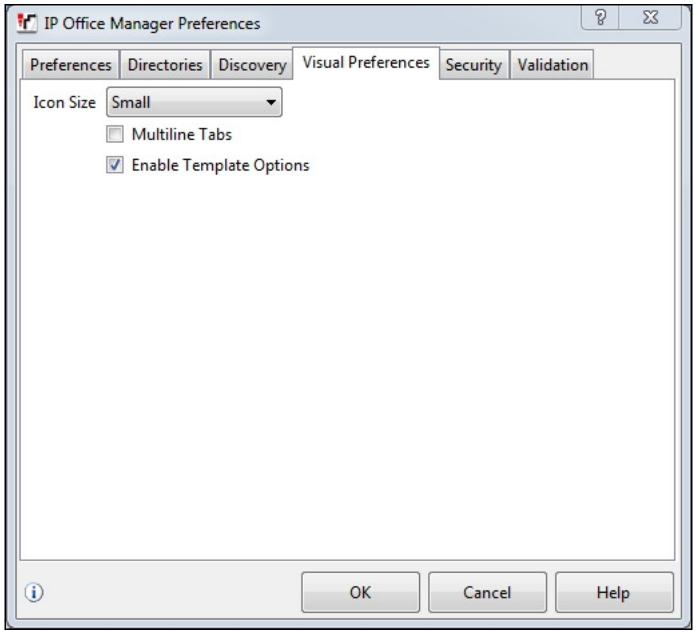
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

Alternatively, a SIP Line can be created manually. To do so, right-click Line in the Navigation Pane and select **New→SIP Line** (not shown). Then, follow the steps outlined in **Section 5.6.2**.

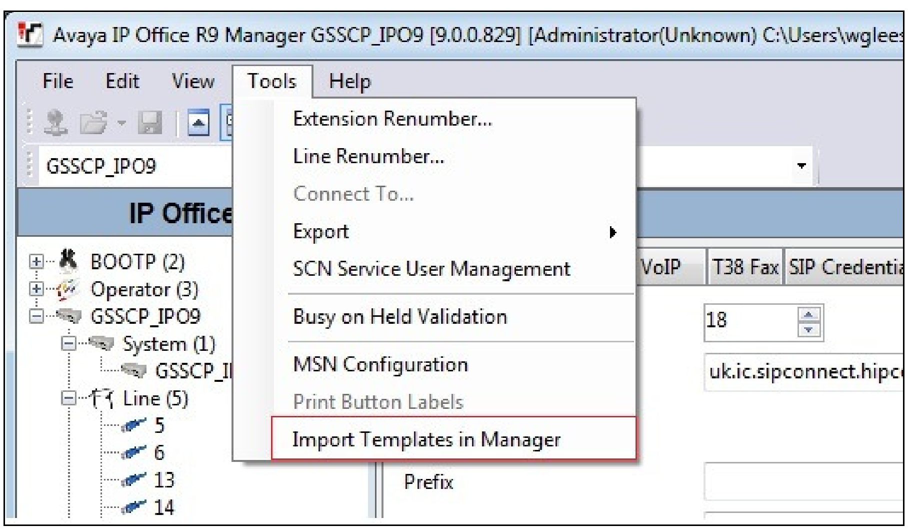
5.6.1. SIP Line From Template

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

Verify that template options are enabled 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**.

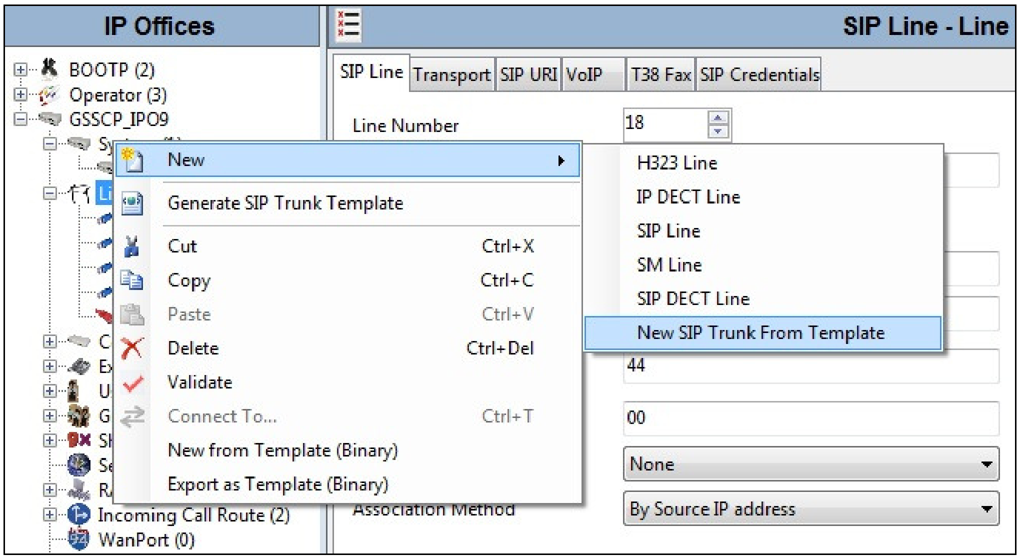


Import the template into IP Office Manager. Select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus. The default template location is **C:\Program Files\Avaya\IP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



In the pop-up window (not shown) that appears, select the directory where the template file was copied. 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.

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**.



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

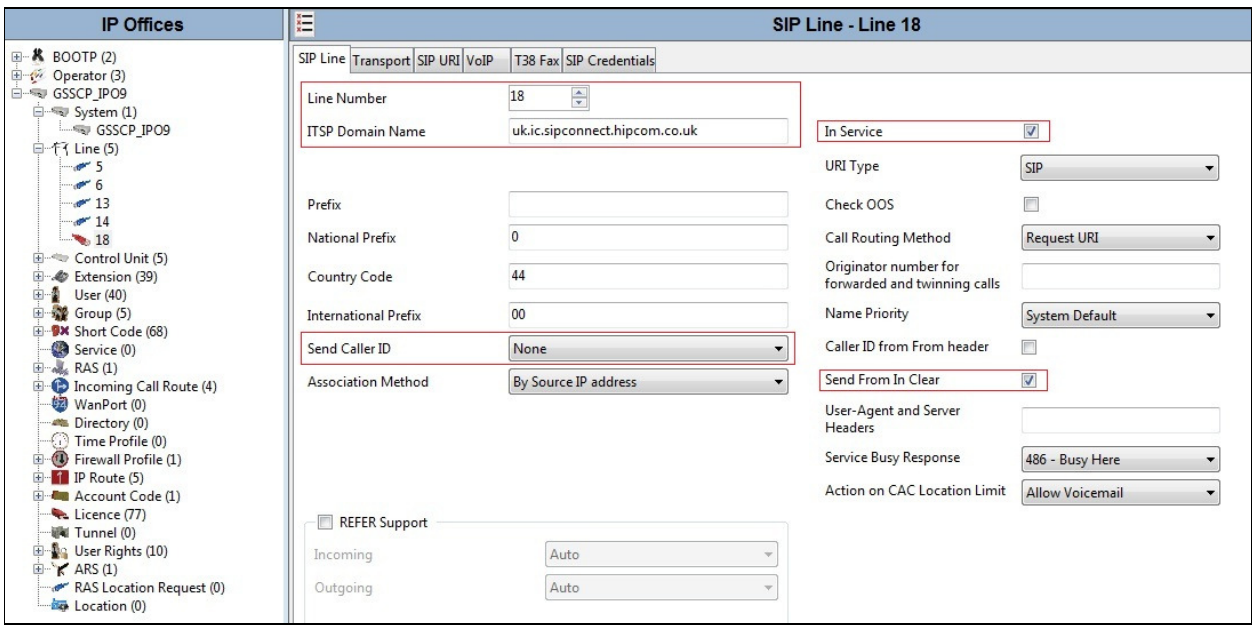
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

5.6.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to the domain name used by HIPCOM. In test this was **uk.ic.sipconnect.hipcom.co.uk**.
- Set **Send Caller ID** to **None** as the **Calling party information for Mobile Twinning** field in **Section 5.4** is populated with a number for the DDI range assigned to the IP Office. If a number is required that is representative of the extension making the call, this field can be used.
- Ensure the **In Service** box is checked.
- Check the **Send From In Clear** box to ensure that the calling number is inserted into the SIP From header even when the calling number is restricted.
- Default values may be used for all other parameters.

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

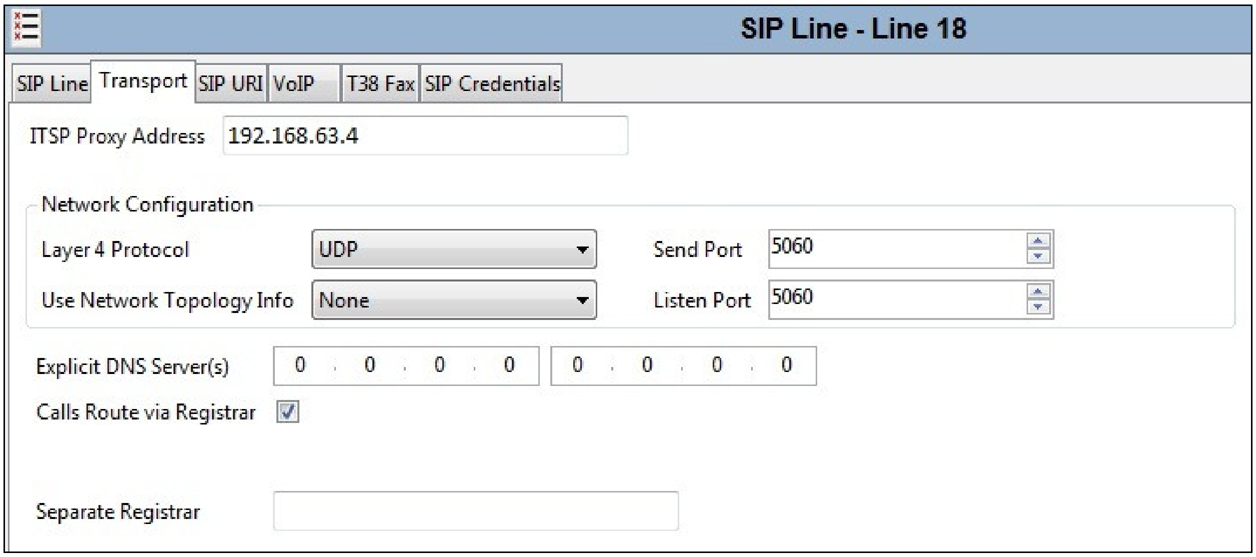


Note: The **Send From In Clear** box must be checked so that the calling number is inserted in the From header as it is required for verification by the HIPCOM network. The network uses the SIP Privacy header to identify where the calling number is restricted and does not require “anonymous@anonymous.invalid” in the From header.

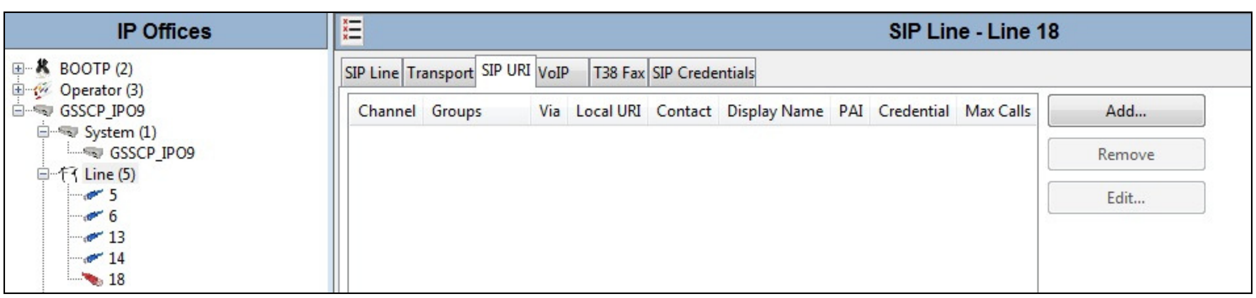
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the HIPCOM SIP proxy.
- Set **Use Network Topology Info** to **None** if NAT is not to be used and the Network Topology settings defined in **Section 5.2** are not required. This was the case during test.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.

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



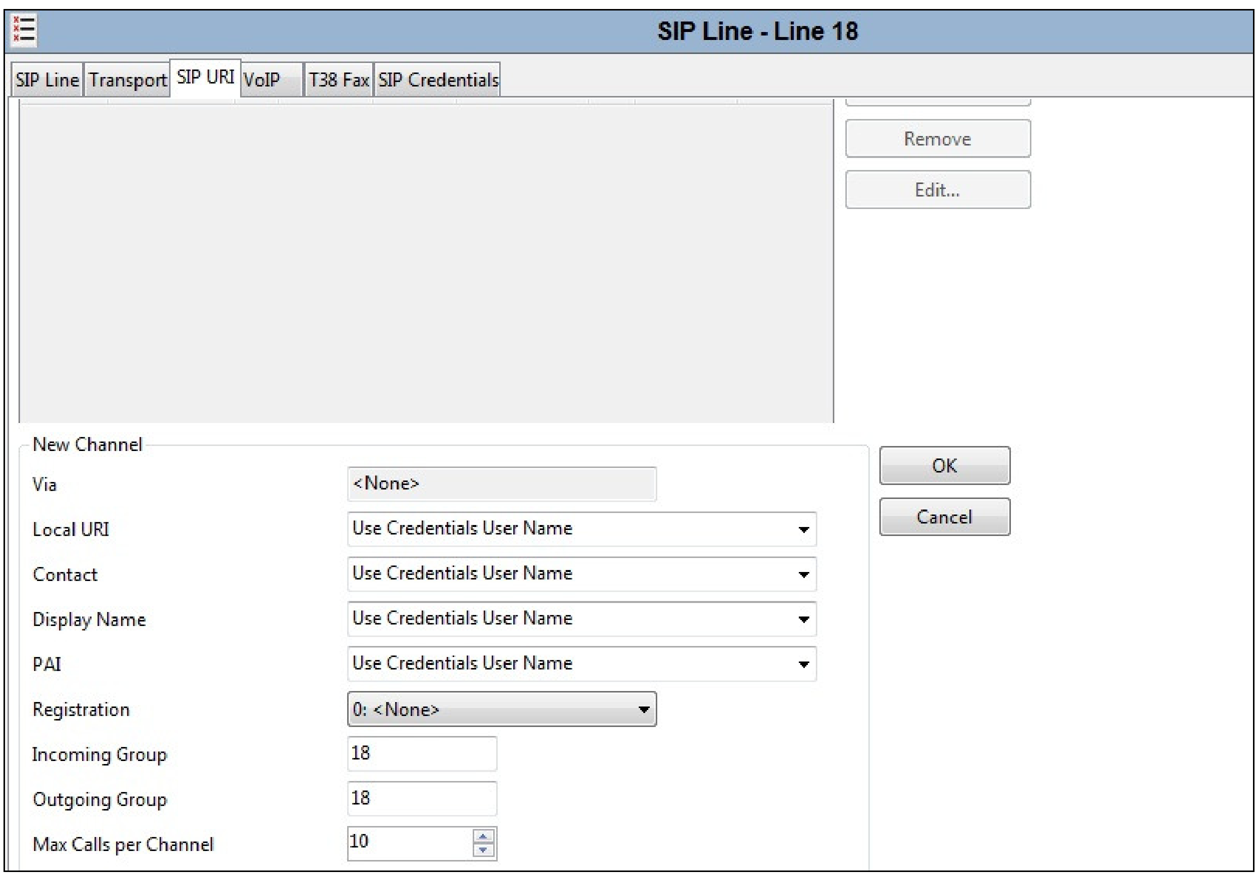
After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



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 Credentials User Name**. This will use the number in the DDI range that was supplied for SIP Registration. If the DDI number applied to the specific extension is required, this should be set to **Use Internal Data**.
- For **Registration**, only **0: <None>** is available at this point. Registration details are defined under the **SIP Credentials** tab and, once defined, are available in this drop down menu.
- 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.

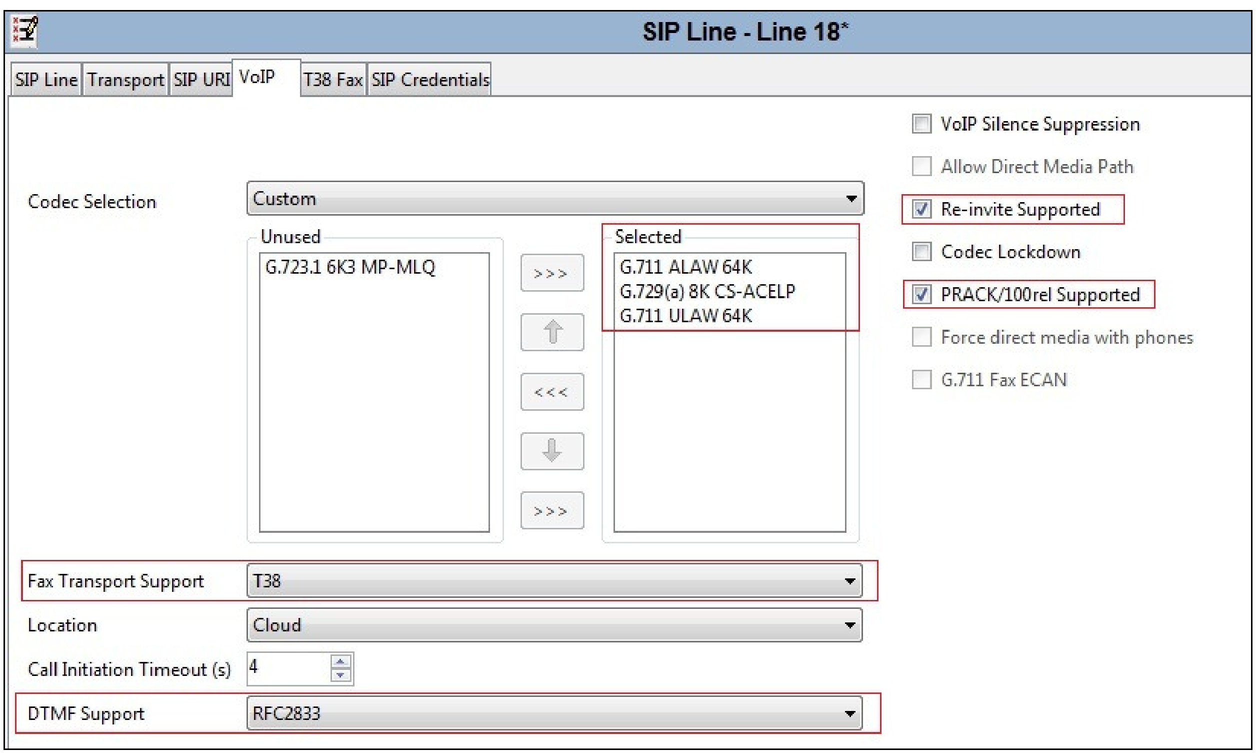
On completion, click the **OK** button.



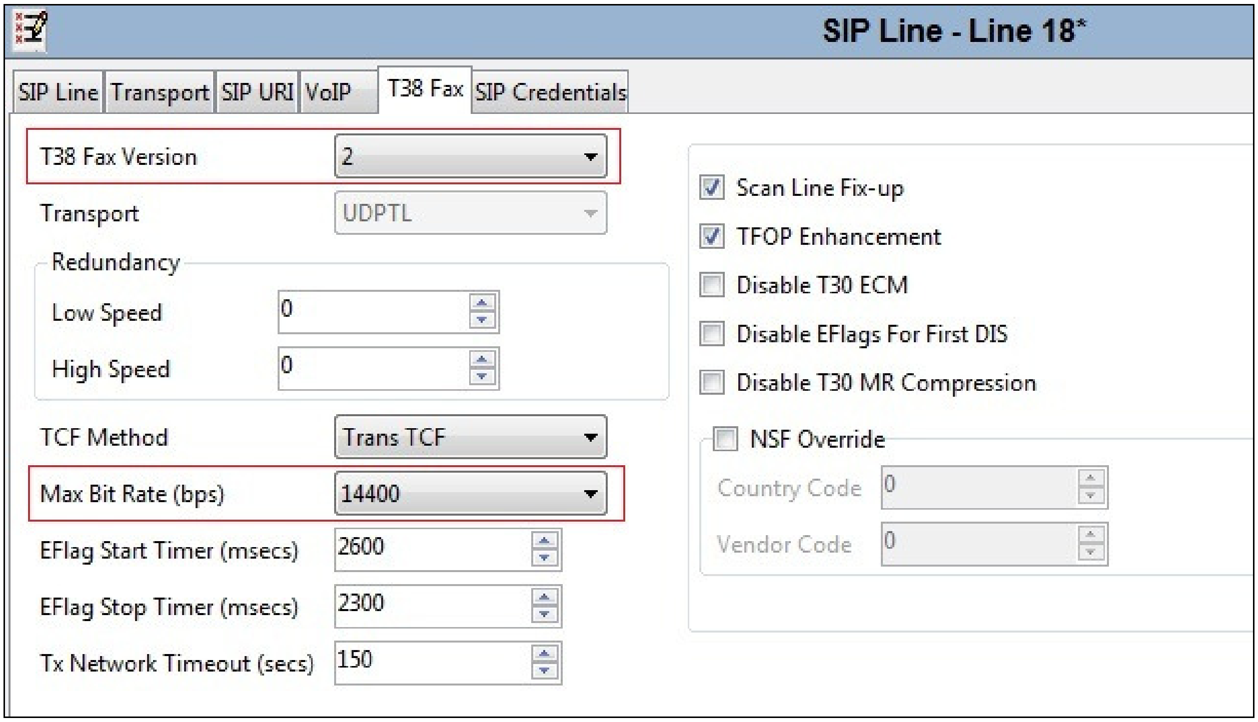
Note: It will be necessary to return to this screen after the SIP Credentials have been defined to select them in the **Registration** drop down menu.

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

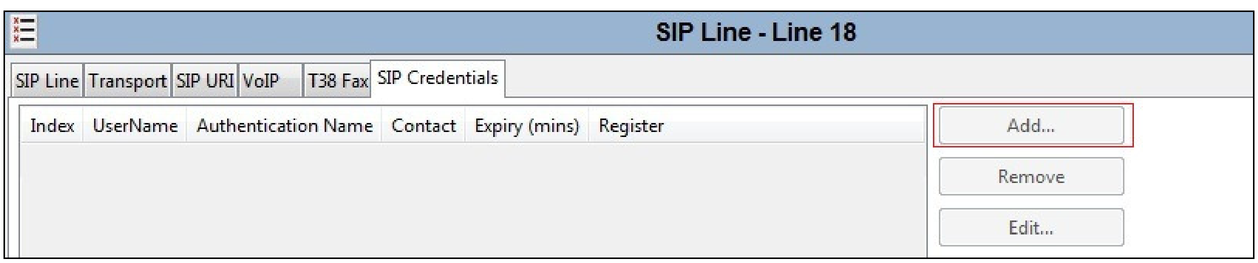
- Select **Custom** in the **Codec Selection** drop down menu to specify the preferred codecs
- Highlight codecs in the **Unused** box that are to be used on this line and click on the right arrows to move them to the **Selected** box
- Highlight codecs in the **Selected** box that are not to be used and click on the left arrows to move them to the **Unused** box
- Highlight codecs in the **Selected** box and use the up and down arrows to change the priority order of the offered codecs, for testing with HIPCOM this was **G.711 ALAW 64K**, **G.729(a) 8K CS-ACELP** and **G.711 ULAW 64K** in priority order from the highest to the lowest.
- Select **T38** in the **Fax Transport Support** drop down menu to allow T.38 fax operation
- Select **RFC2833** in the **DTMF Support** drop down menu. 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.
- Check the **PRACK/100rel Supported** box to allow for reliable responses to provisional call set-up messages such as 183 Session progress and 180 Ringing.
- Default values may be used for all other parameters.
- On completion, click the **OK** button (not shown).



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 **2** 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 configure the authentication parameters for the HIPCOM SIP Trunk. Click on Add (not shown).



The registration details are entered at the bottom of the screen.

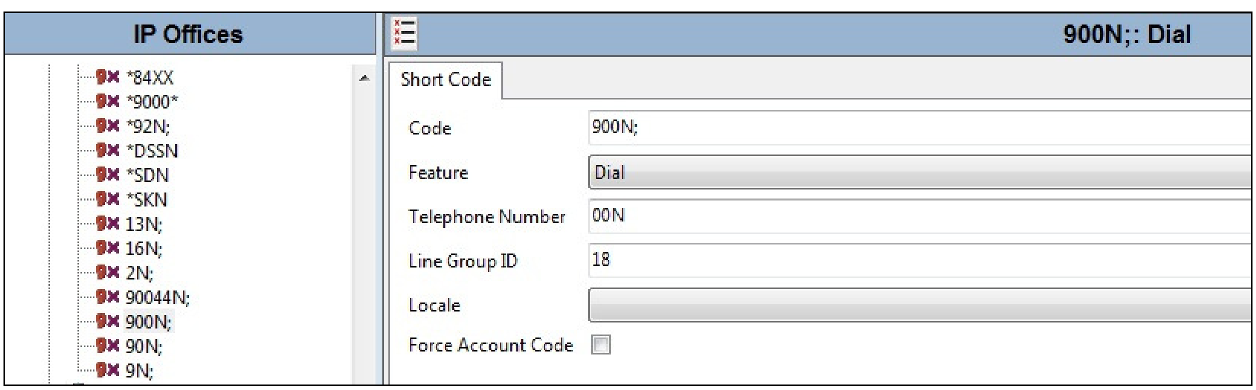
The authentication takes placed on the SIP registration. Define the login details in the **User name** and **Authentication Name** fields. Define the domain in the **Contact** field and the password in the **Password** field. Check the **Registration required** box

Note: At this stage, return to the SIP URI tab and select the registration details defined above in the **Registration** drop down menu. It is advisable to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.6** available.

5.7. Short Codes

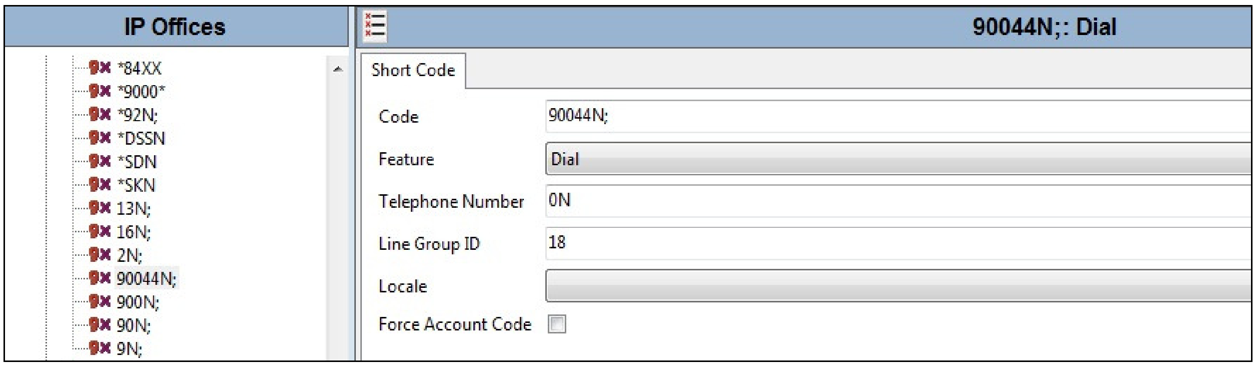
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown in the example below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon
- The example shows **900N**; which will be invoked when the user dials 9 followed by an international number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **00N** which inserts the international number with international dialling prefix into the Request URI and To headers 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).



Short codes are also used for routing of national calls and Operator calls. An example for national calls where the full number with country code has been dialled is shown below.

- The example shows **90044N**; which will be invoked when the user dials 9 followed by a UK number in international number.
- Set **Telephone Number** to **0N** which removes the country code and inserts the national number with leading 0 into the Request URI and To headers in the outgoing SIP INVITE message
- Set other parameters as shown in the first example.



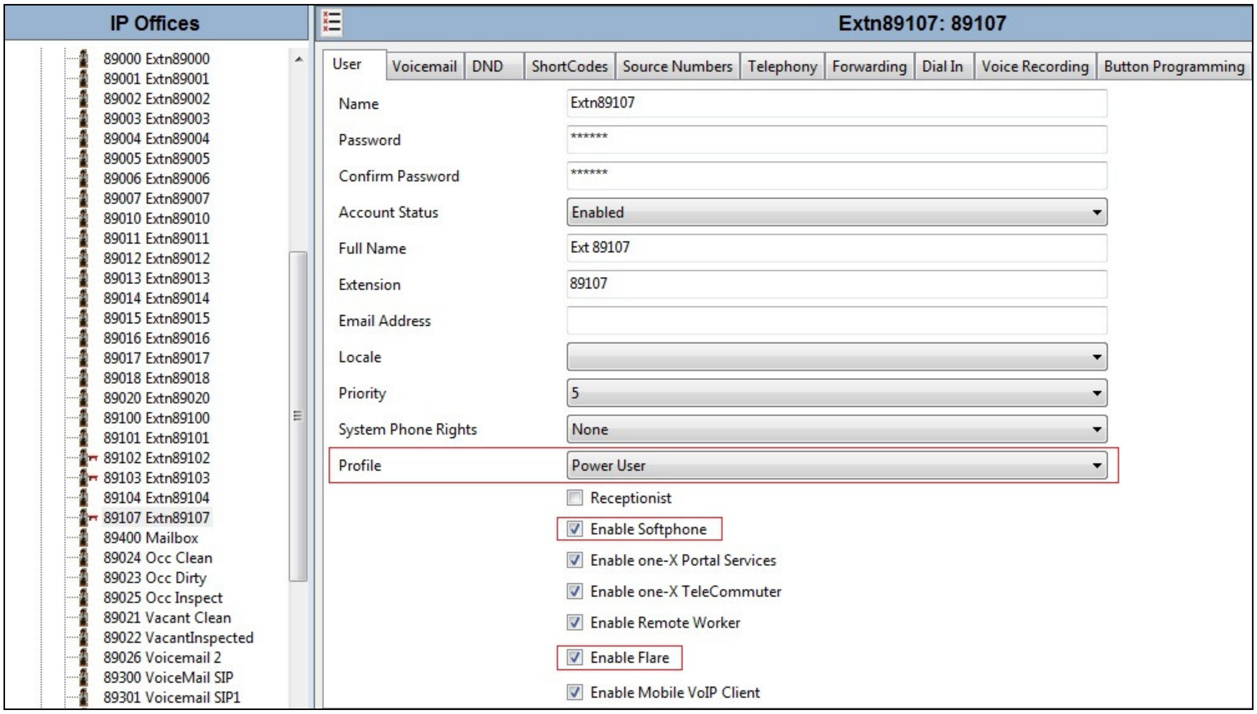
5.8. User

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, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required.

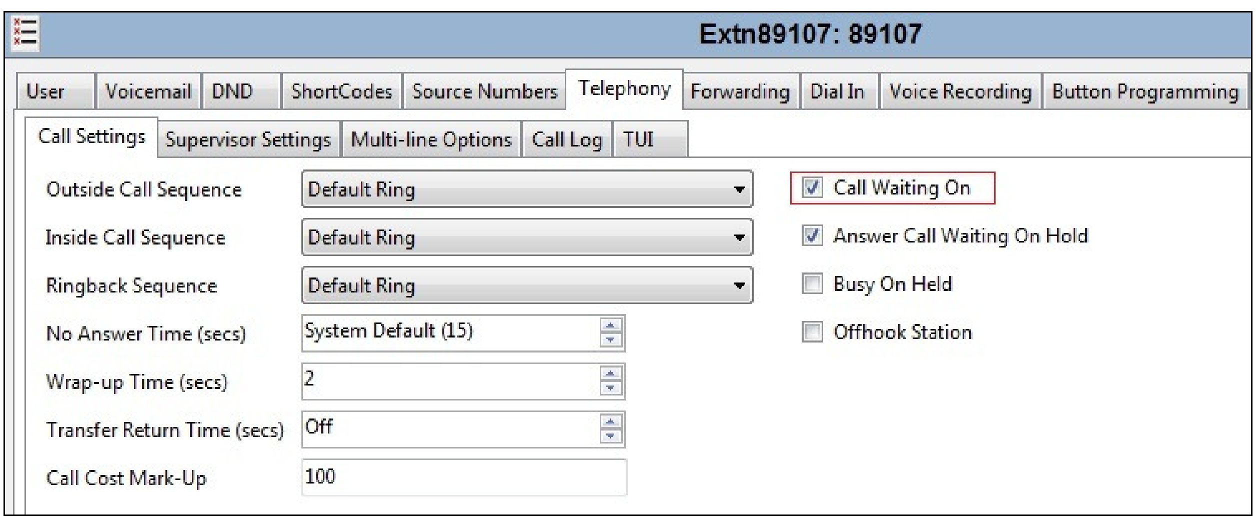
The example over the page shows the configuration required for a SIP endpoint that can be used for a SIP phone or Softphone. Two types of Softphone were successfully tested; these were IP Office Softphone and Avaya Flare Experience for Windows. To configure a SIP User that can accommodate the different types of phone, it has to be given a profile that supports them. In test, **Power User** was used.

Change the **Name** of the User if required, this will be used for login to the IP Office Softphone

- Select **Power User** from the Profile drop down menu
- Check the **Enable Softphone** box
- Check the **Enable Flare** box

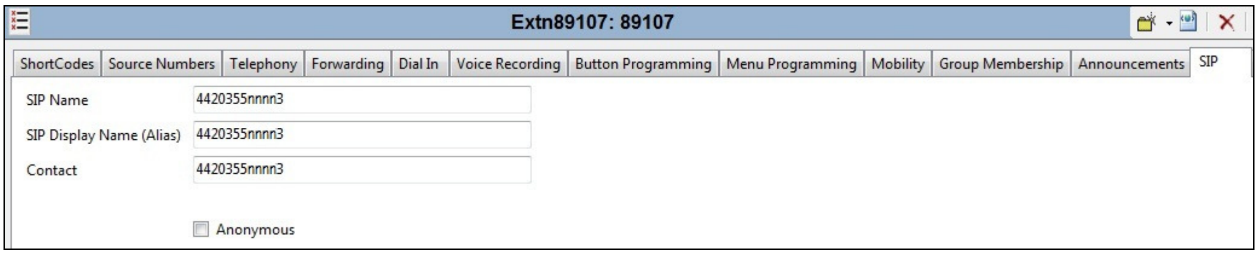


SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**. Call forwarding and transfer make use of the SIP REFER message. To handle SIP REFER on IP Office, the Call waiting function is used. To turn on Call Waiting, navigate to **Telephony→Call Settings**. Check the **Call Waiting On** box.



Next Select the **SIP** (not shown) 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 HIPCOM.

In the example below, one of the DDI numbers in the test range is used, though some of the digits have been obscured. The **SIP Display Name (Alias)** parameter must also be a DDI number as this is used as part of the network verification of the outgoing calls. On completion, click the **OK** button (not shown).



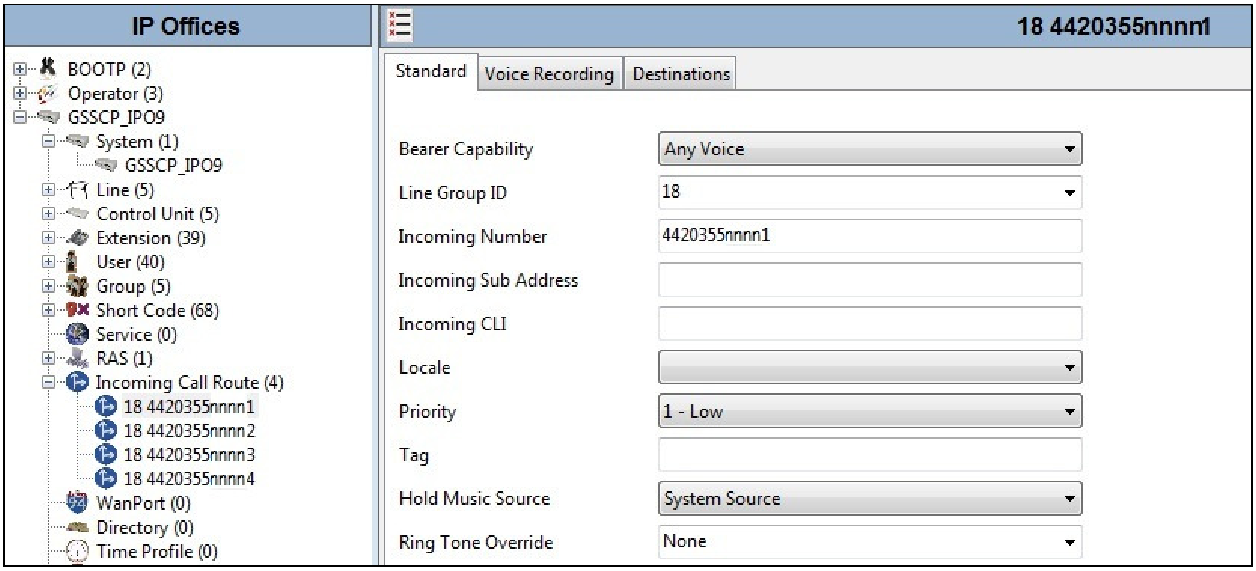
Note: The **Anonymous** box can be checked if The Calling Line Identity is to be Restricted (CLIR).

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**.

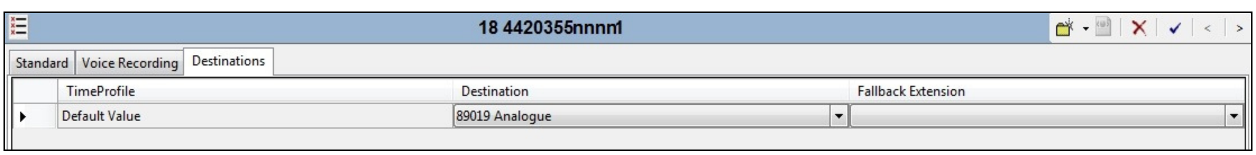
On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- 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



Note: A number of digits of the DDI have been obscured. Number format is international.

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 89019.

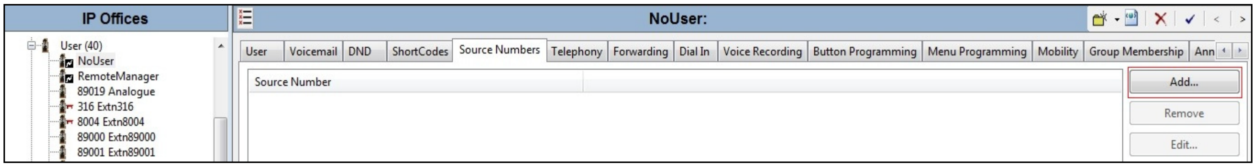


5.10. SIP Options

Avaya IP Office can be configured to send SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the lower value of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP_OPTIONS_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **NoUser** user. During test, the **Network Topology** information was not used as there was no requirement for NAT and a STUN server. Also, OPTIONS was not required as SIP registration is used and REGISTER messages were sent every 2 minutes.

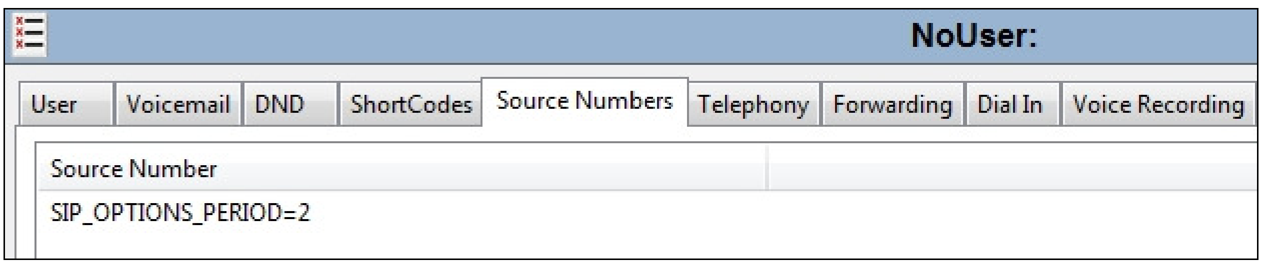
If required, configure the **SIP_OPTIONS_PERIOD** parameter as follows:

- Navigate to **User → NoUser** in the Navigation Pane.
- Select the **Source Numbers** tab in the Details Pane.
- Click the **Add** button



At the bottom of the subsequent Details Pane, the **Source Number** field will appear. Enter **SIP_OPTIONS_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.

The **SIP_OPTIONS_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 2 minutes was used.



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. HIPCOM SIP Trunk Configuration

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

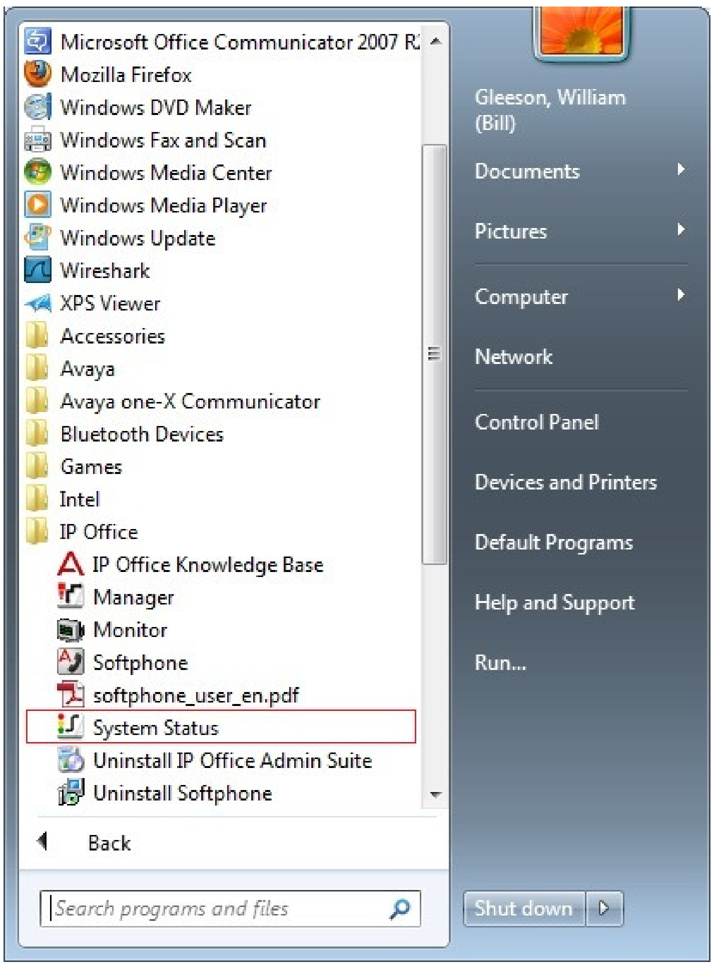
- IP address of HIPCOM SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

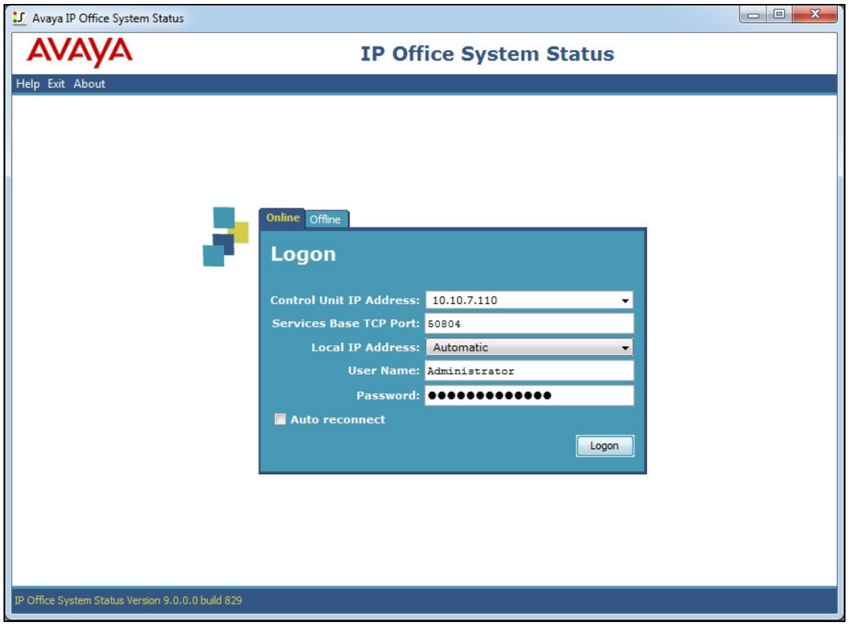
This section includes steps that can be used to verify that the configuration has been done correctly.

7.1. SIP Trunk status

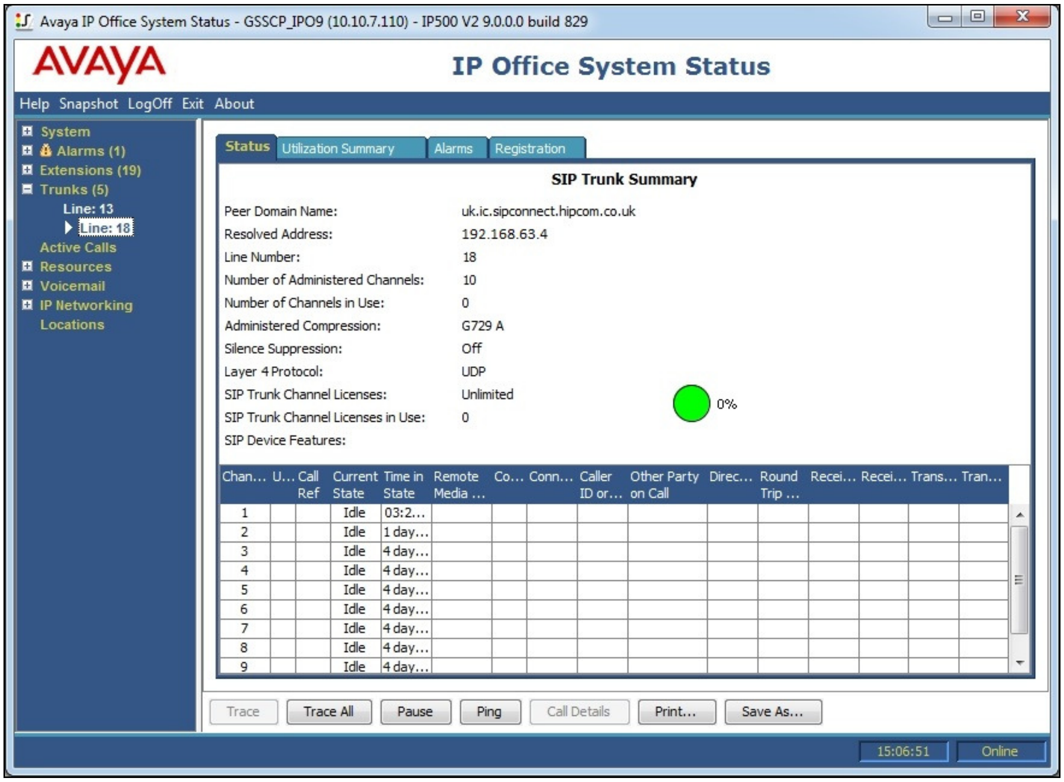
The status of the SIP trunk can be verified by opening the System Status application. A Windows 7 Laptop PC was used for testing and the application was opened by pressing the Start button and selecting **All Programs→ IP Office → System Status**.



Log in to 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.



8. Conclusion

The HIPCOM SIP Trunk passed compliance testing. Interoperability testing of the sample configuration was completed with successful results for the HIPCOM SIP Trunk. Refer to **Section 2.2** for test observations.

9. Additional References

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

[1] *IP Office 9.0 KnowledgeBase Technical Documentation*, Web based.

[2] *IP Office 9.0 Installing IP500/IP500 V2*, Document number15-601042, 11th October 2013.

[3] *IP Office Application Server 9.0 Installation and Maintenance*, Document number15-601011, 29th August 2013.

[4] *IP Office 9.0 Using System Status*, Document number15-601758, 15th August 2013

[5] *IP Office 9.0 Installing IP Office Video Softphone*, Document number 100173998, 21st August 2013

[6] *IP Office 9.0 SIP Extension Installation*, 21st August 2013

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.