Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.0 with SFR SIP Trunk (Collecte SIP) – Issue 1.0

Abstract

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

The SFR SIP Trunk (Collecte SIP) provides PSTN access via a SIP trunk connected to the SFR Voice Over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. SFR is 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 SFR Collecte SIP and Avaya IP Office. SFR Collecte SIP provides PSTN access via a SIP trunk connected to the SFR 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 SFR Collecte SIP. 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

Avaya IP Office was connected to SFR Collecte SIP. 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. Calls were routed to the enterprise across the SIP trunk from SFR.
- Outgoing PSTN calls from various phone types including H.323, SIP, digital, and analogue telephones at the enterprise. Calls were routed from the enterprise across the SIP trunk to SFR.
- Inbound and outbound PSTN calls to/from an IP Office Softphone client and Avaya Flare® Experience for Windows.
- Various call types including: local, international, toll free (outbound) and directory assistance.
- Codecs G.729A and G.711A.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38
- 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 mobile twinning.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the SFR Collecte SIP with the following observations:

- When a call comes in from the PSTN and is not answered on IP Office, the SFR network clears the call on the PSTN side after 3 minutes. It does not, however, clear the call on the IP Office side and the called party continues to ring but can't be answered. IP Office has a default setting of 5 minutes before it clears the call and at release 9.0 this can't be changed. This issue is resolved in IP Office release 9.1 as the No Answer timeout is configurable. Release 9.1 became Generally Available subsequent to completion of testing.
- There was no Toll Free access available for testing.
- Incoming T.38 fax was unreliable from the Lab in Galway. It was consistently successful from SFR premises so the fault was considered to be network related.
- When the SIP Trunk was out of service, the SFR network did not provide any indication to the caller. Instead the caller heard silence for a number of minutes.

2.3. Support

Le Service Technique SFR Business Team est joignable 24H/24, 7J/7 par un numéro gratuit pour signalisation des incidents techniques sur le service Collecte SIP.

CENTRE SERVICE CLIENT SFR Business Team
0 800 950 920

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to SFR Collecte SIP. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), an Avaya 1140e SIP Telephone, an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as an IP Office Softphone client and Avaya Flare® Experience for Windows for mobility testing. For security purposes, any PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead the phone numbers have been obscured beyond the city code.

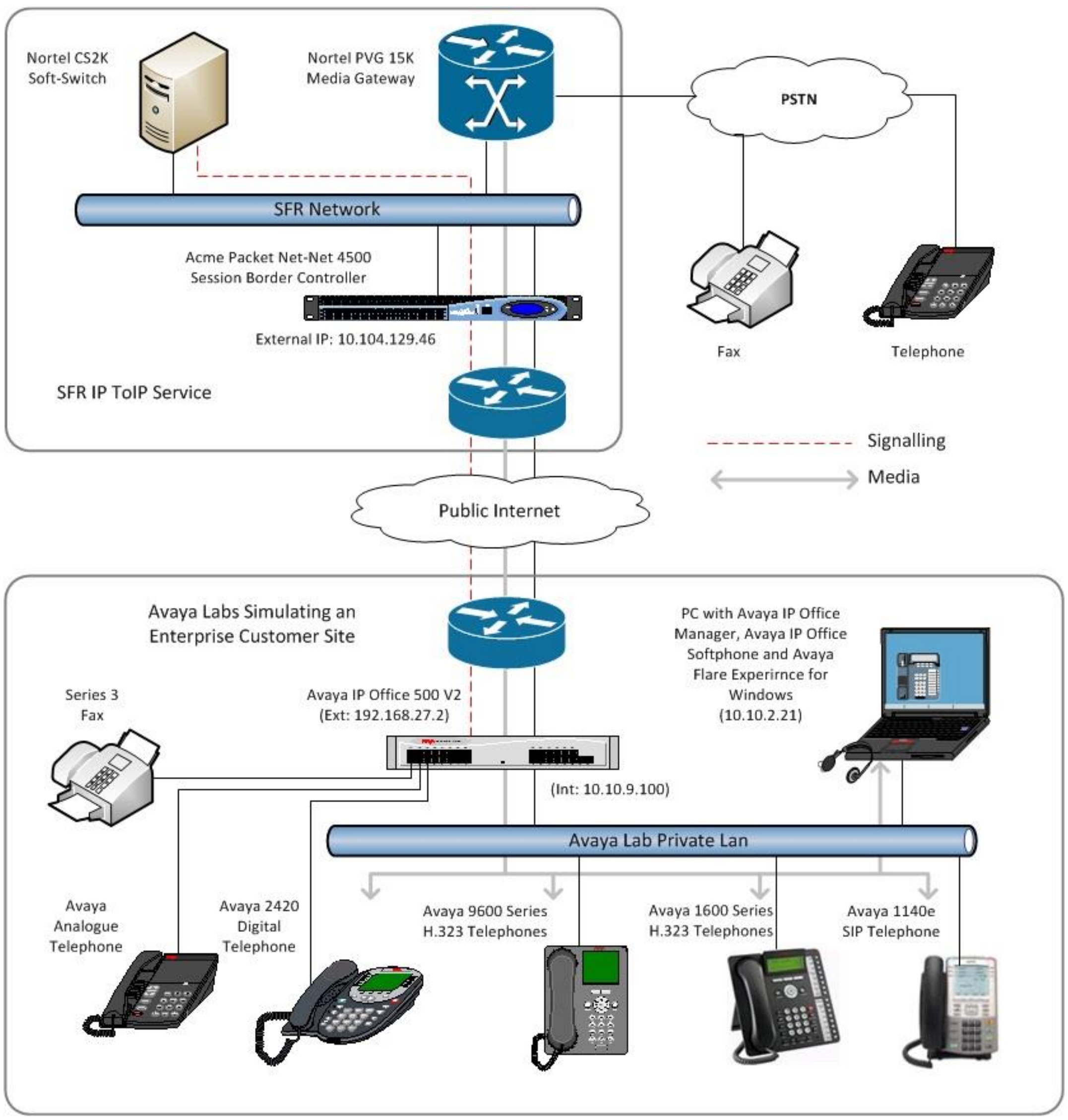


Figure 1: SFR Collecte SIP to Avaya IP Office Topology

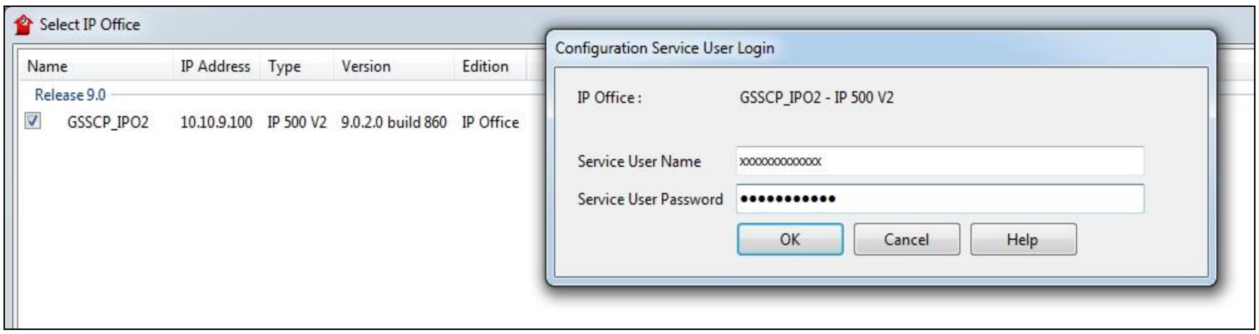
4. Equipment and Software Validated

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 R9.0.400.965
Avaya 1140e IP SIP Telephone	04.04.10.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.3.1.16
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Avaya Softphone	3.2.3.49
Avaya Flare® Experience for Windows	1.1.3.14
Avaya IP Office Manager	Version 9.0.4.0 build 965
SFR	
Nortel Media Server	Communication Server 2000 (CS2K) CVM16
Nortel PSTN gateway	PVG 15k PCR 8.2
Acme Packet Net-Net 4500 SBC	SCX6.2.0 MR-6 GA

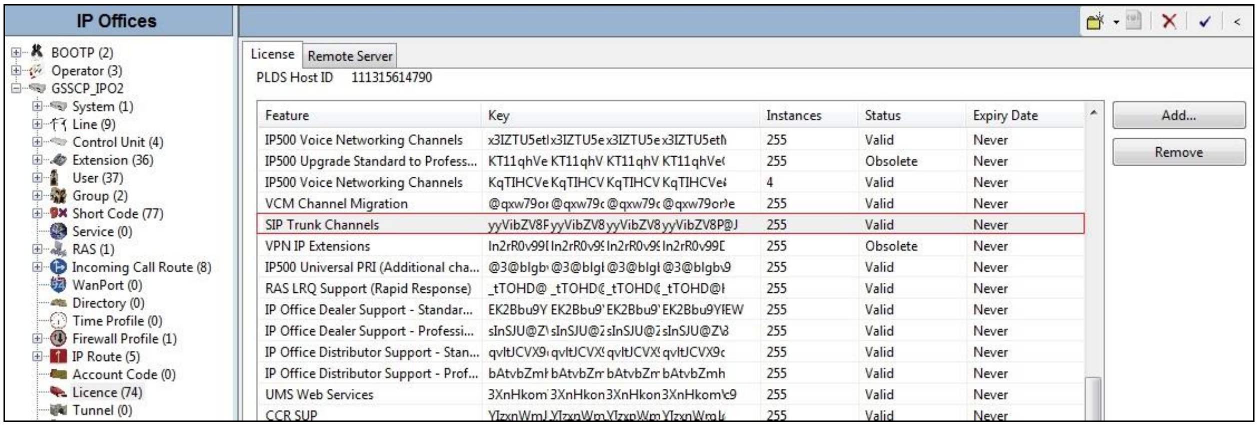
5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the SFR Collecte SIP. 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** (not shown), 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 mobile twinning) is assumed to already be in place.



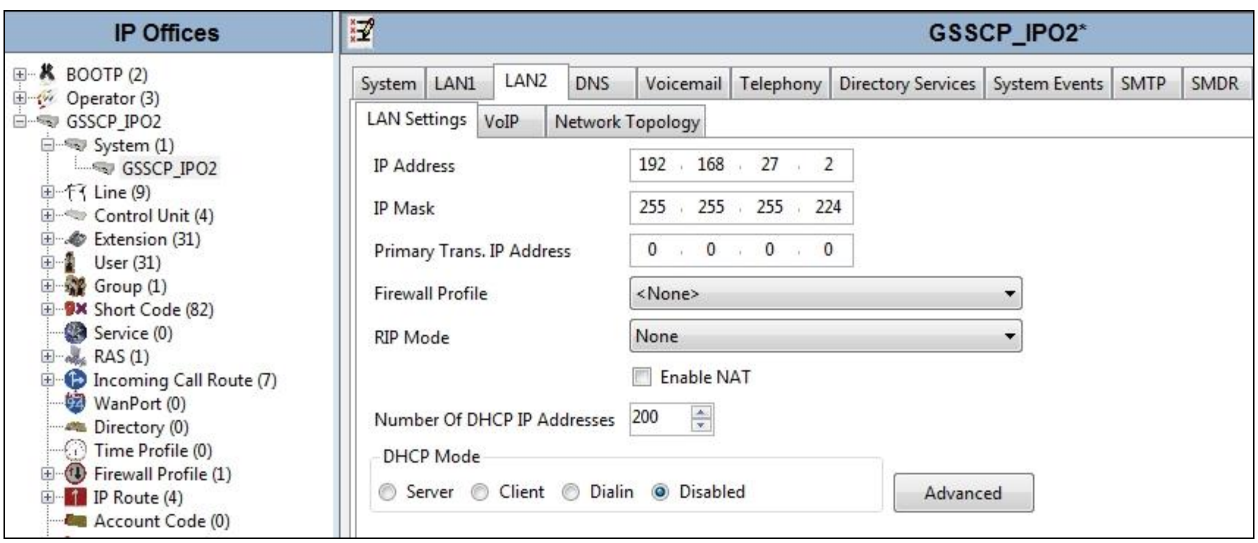
5.1. Verify System Capacity

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 SFR.



5.2. LAN2 Settings

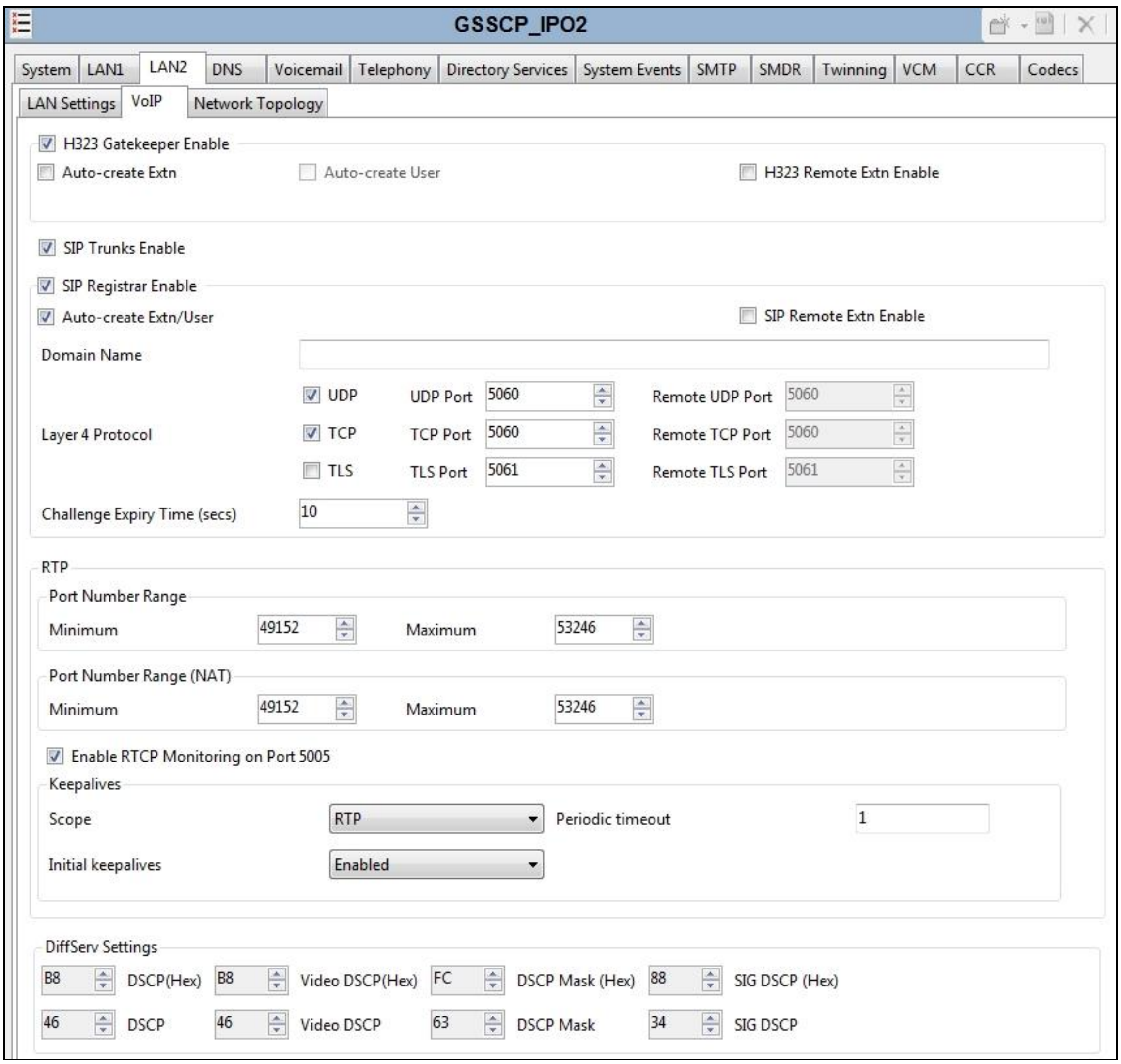
In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → <IP Office Name>** in the Navigation Pane where <IP Office Name> is the name of the IP Office. This is **GSSCP_IPO2** in the GSSCP test 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. If SIP Endpoints are to be used such as the IP Office Softphone, Avaya Flare® Experience for Windows and the Avaya 1140e, 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 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 requests RTP media to be sent to a UDP port in the configurable range for calls using LAN2.

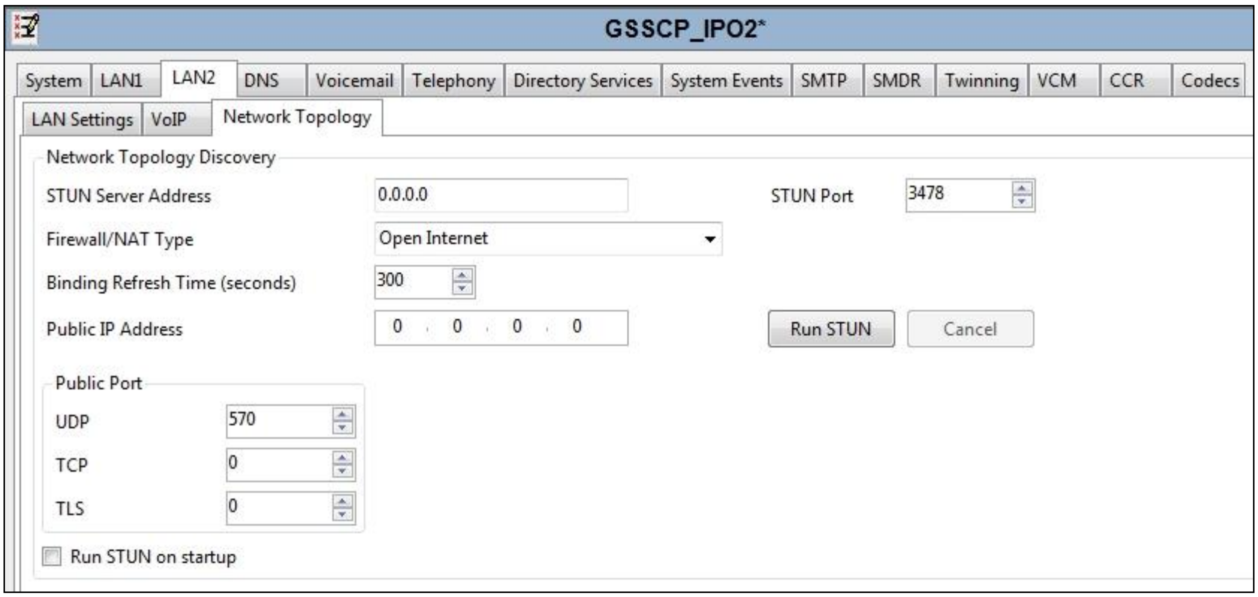
Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signalling and media. DSCP for media can be set for both voice and video. The **DSCP** field is the value used for voice 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 below.



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

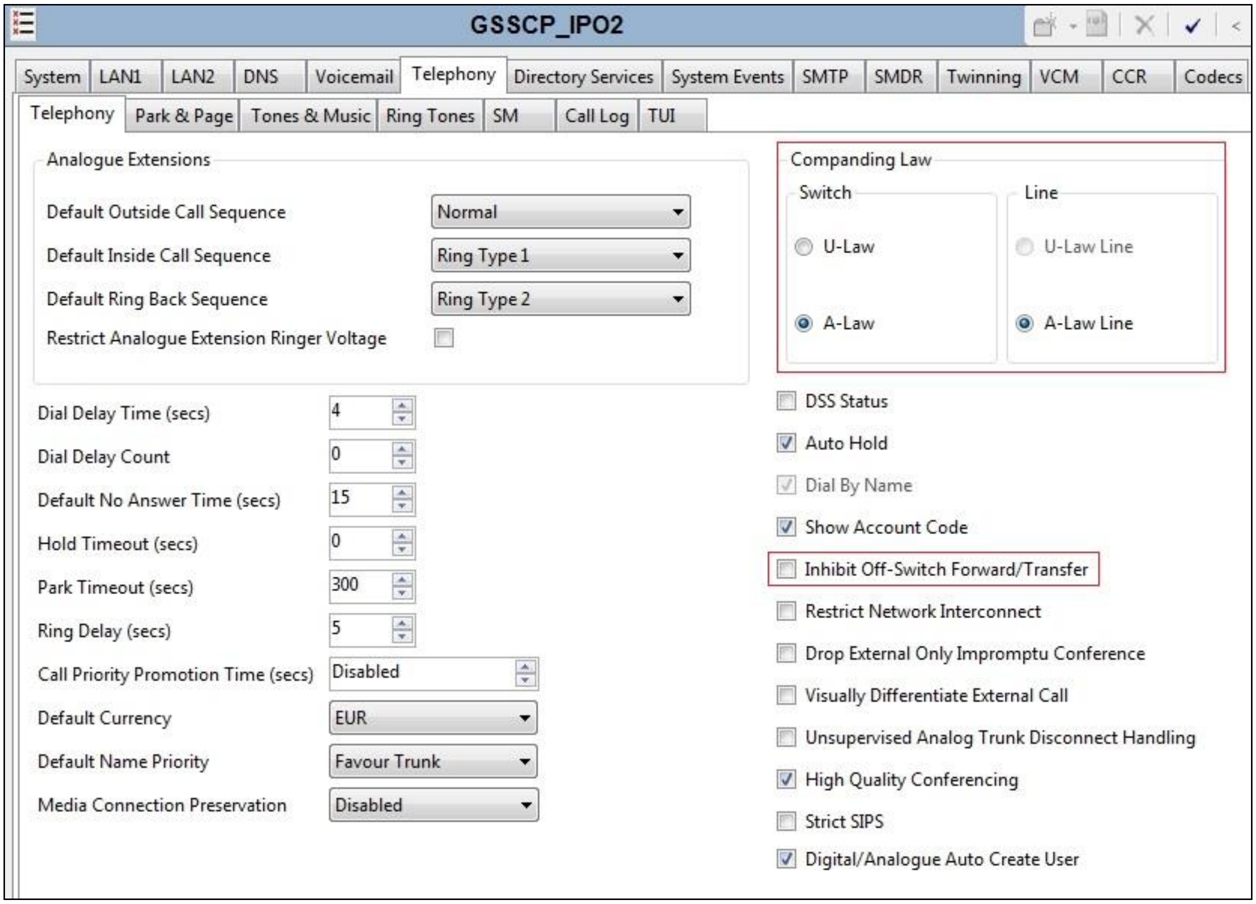
On the **Network Topology** tab in the Details Pane enter the information required if Network Address Translation (NAT) is to be used. During test, NAT was not required and there was no requirement for a STUN server. To disable this facility, 0.0.0.0 is entered in the **STUN Server IP Address** and **Public IP Address** fields.

The Network Topology tab can also be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. During testing, IP Office sent OPTIONS messages at an interval of 5 minutes. This was achieved by setting the **Binding Refresh Time** to **300**.



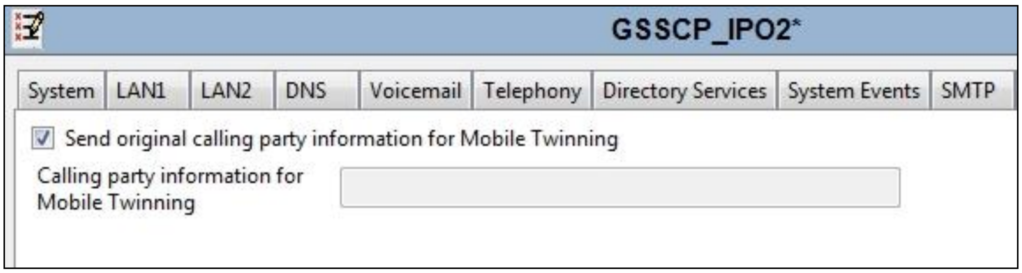
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Choose the **Companding Law** typical for the enterprise location. For Europe, **A-LAW** 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).



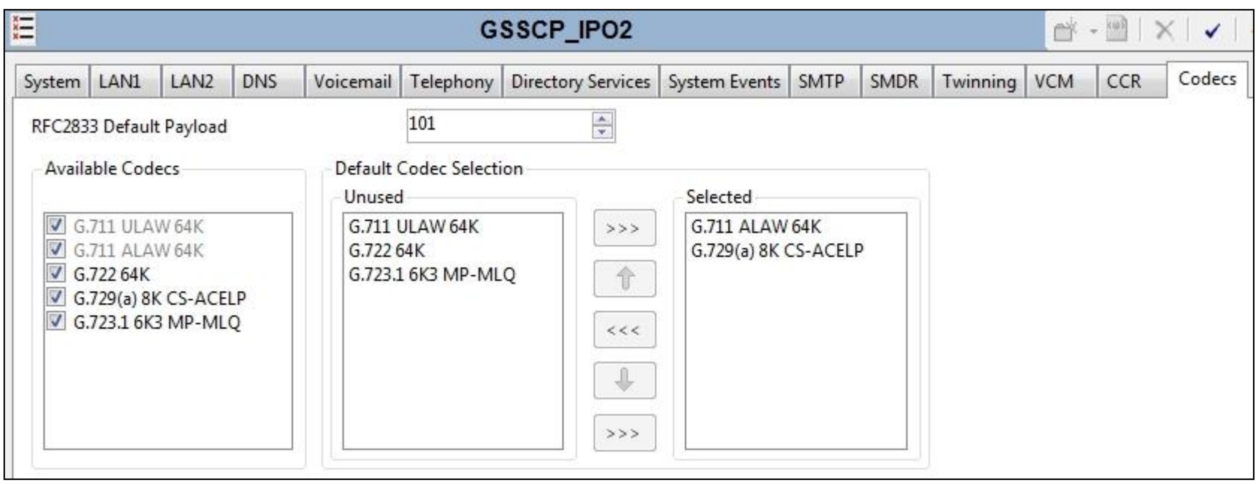
5.4. System Twinning Settings

Navigate to the **Twinning** tab (not shown) and check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a "twinned enabled" phone, Avaya IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of 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 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** and **G.729(a) 8K CS-ACELP** were used as preferred codecs. 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 SFR Collecte SIP. 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:

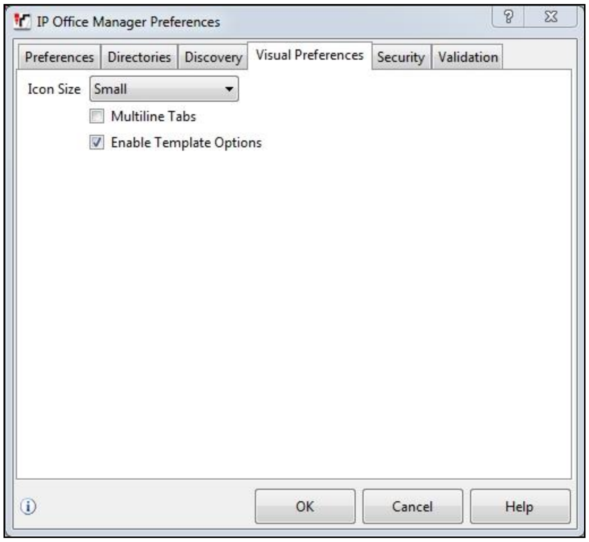
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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**.

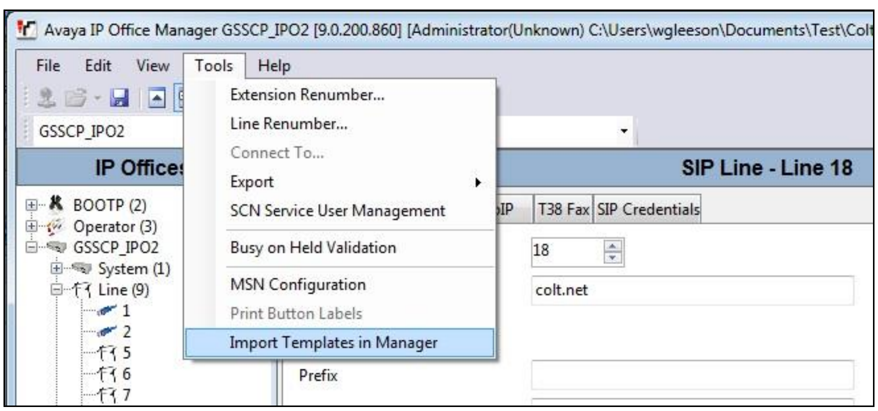
5.6.1. SIP Line From Template

Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **FR_SFR_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**.

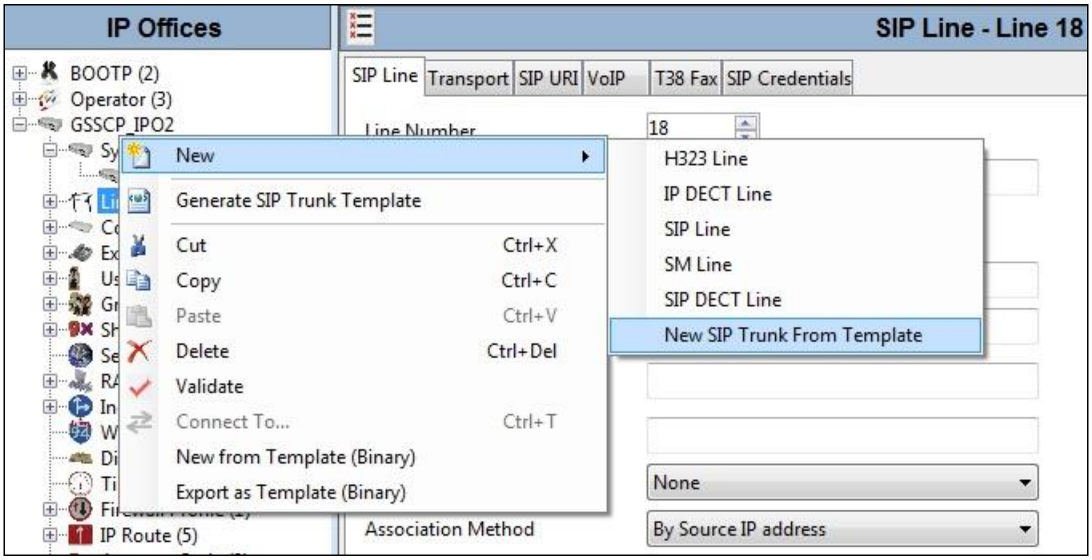


Import the template into IP Office Manager. Select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus. The default template location is **C:\Program Files\Avaya\IP 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 **France** from the **Country** pull-down menu and select **SFR** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**FR_SFR_SIPTrunk.xml**) created earlier. Click **Create new SIP Trunk** to finish creating the trunk.

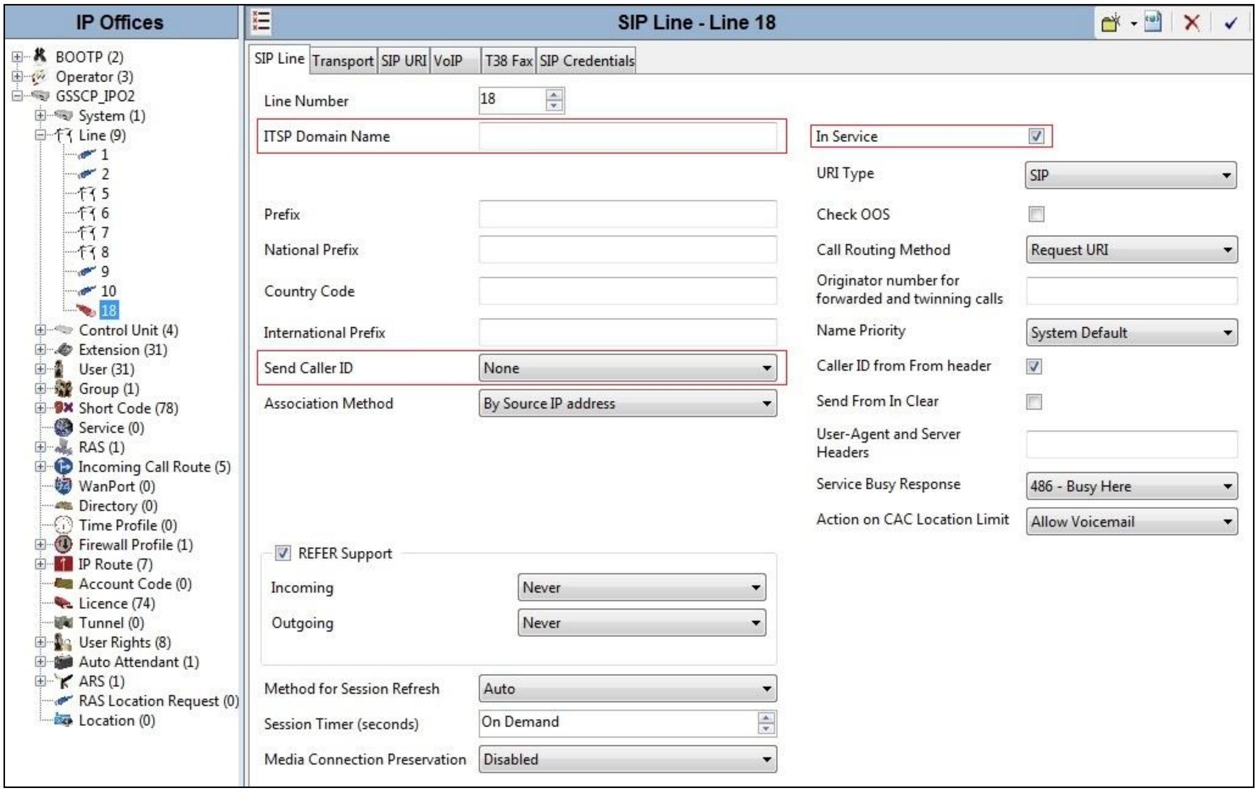
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

5.6.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to the domain name used by SFR. In test no domain name was provided, though **sfr.fr** was entered in the SIP Line template.
- Set **Send Caller ID** to **None** as it is only required if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4**.
- Ensure the **In Service** box is checked.
- Default values may be used for all other parameters.

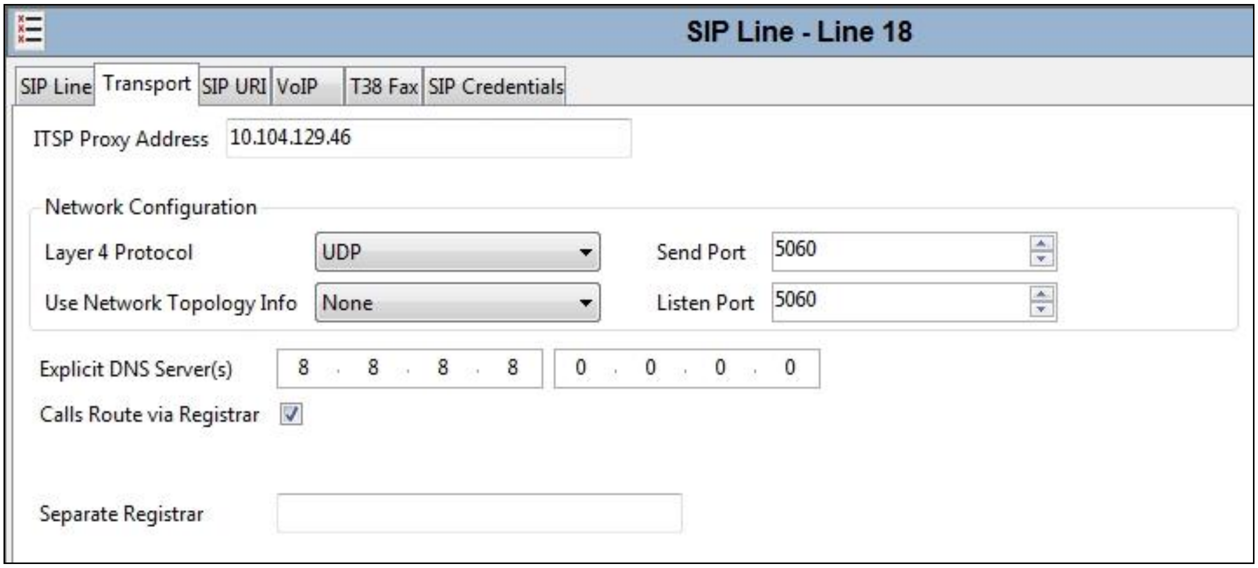
On completion, click the **OK** button (not shown).



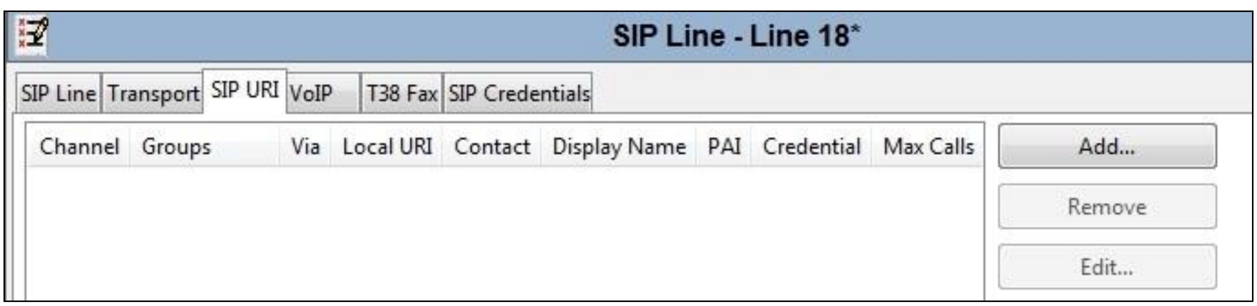
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the external IP address of the SFR SBC.
- 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.

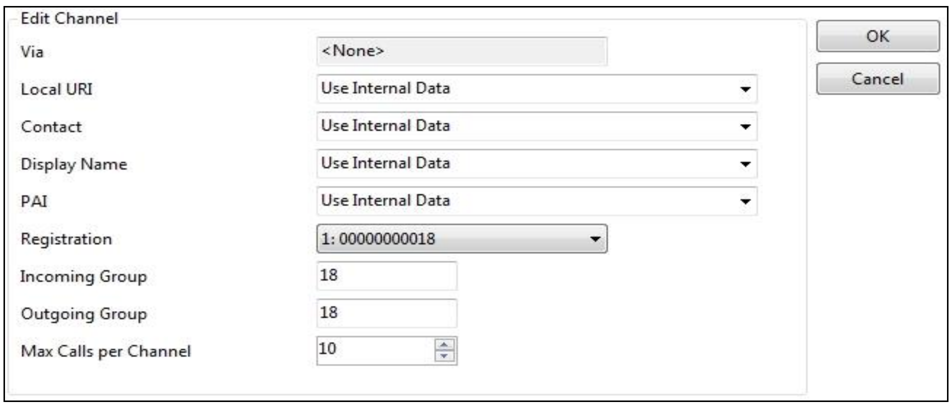


For the compliance test, a generic SIP URI entry was created that matched any number assigned to an Avaya IP Office user. There were additional entries to match specific numbers, for example Auto Attendant and a Feature Name Extension (FNE) service.

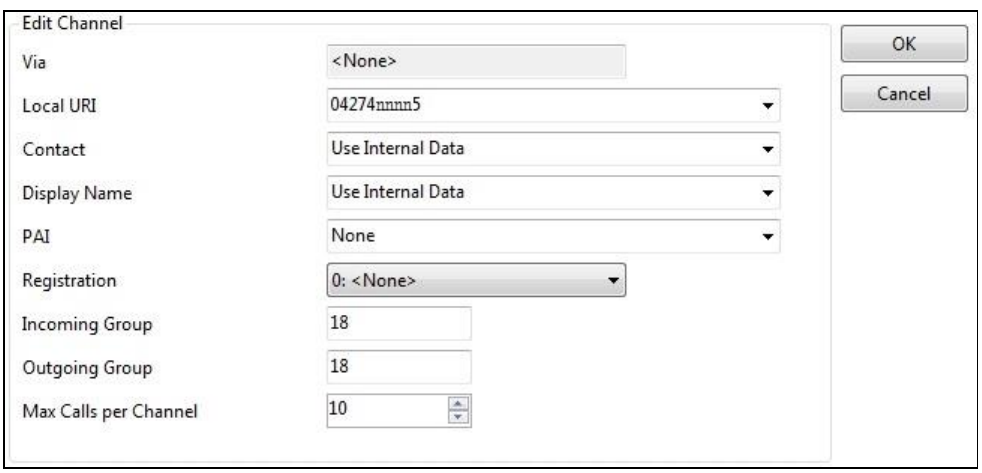
The entry for DDI numbers was created with the parameters shown below.

- Set **Local URI**, **Contact**, **Display Name** and **PAI** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.8**.
- The **Registration** field is not required as registration is not defined for SFR Collecte SIP.
- 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.

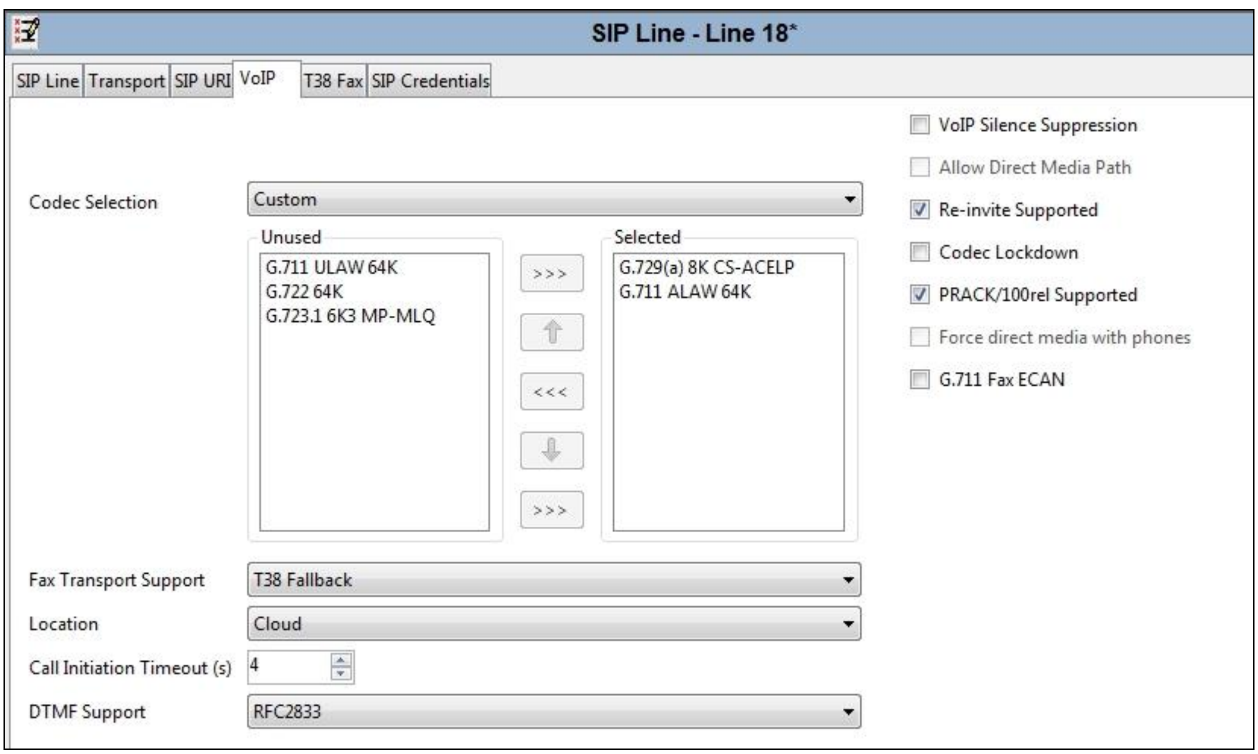


Entries for specific numbers are made in the same way as above, except that the **Local URI** must be set for a specific number. In the following screenshot, some of the DDI digits have been obscured:

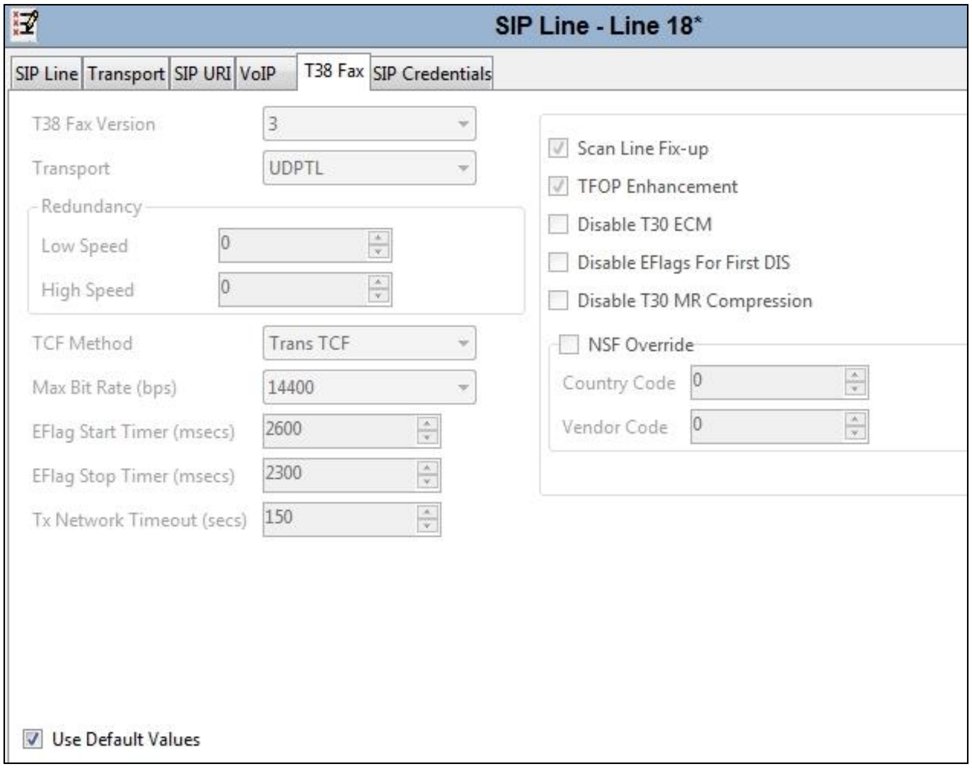


Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- 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 SFR this was **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** in priority order from the highest to the lowest. This reflected the codec list received from the network.
- Select **T38 Fallback** in the **Fax Transport Support** drop down menu to allow both T.38 and G.711 fax operation, though only T.38 was supported by SFR at the time of testing.
- 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. Default values were successfully used during testing. If configuration is required, un-check the **Use Default Values** box and define as required. On completion, click the **OK** button (not shown).

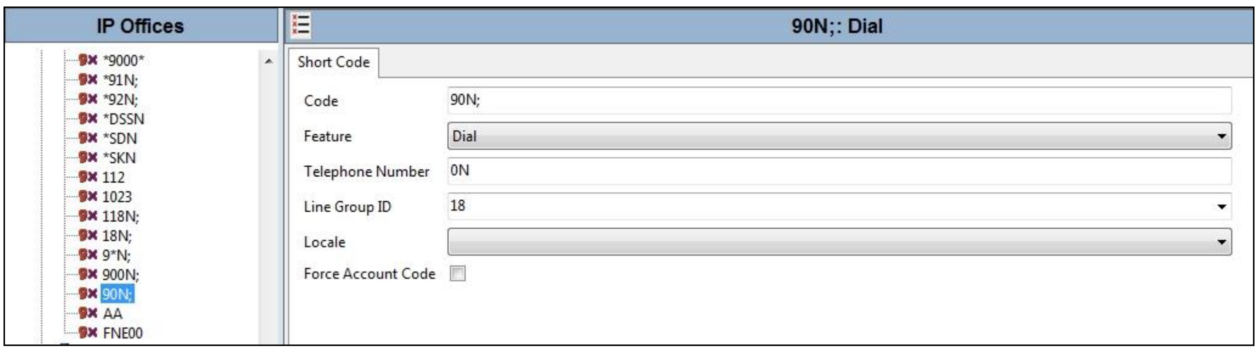


Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.6** available.

5.7. Short Codes

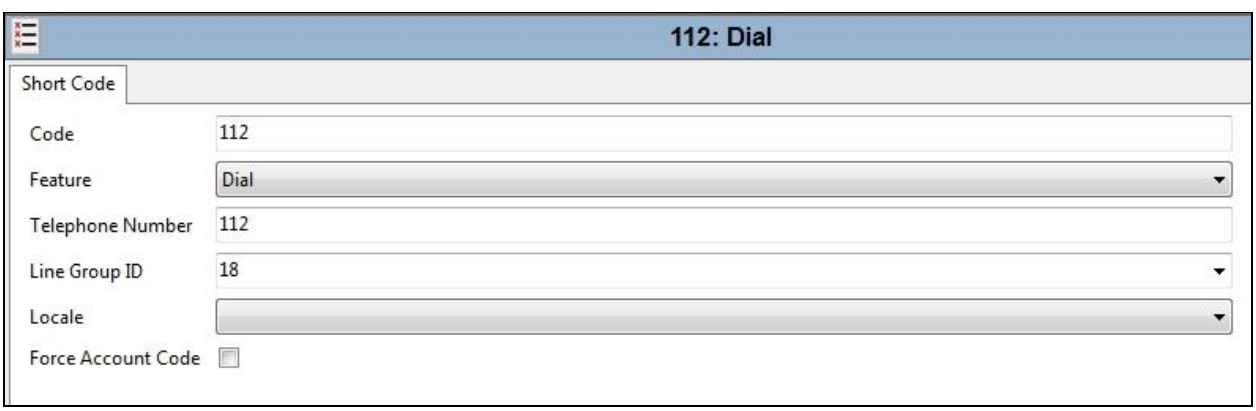
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown in the example for international numbers below.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **90N**; which will be invoked when the user dials 9 followed by a public number
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **0N** which removes the access code and inserts the public number as dialled 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).



Another example shown is Emergency numbers:

- The code is **112** which is the full number for Emergency Services
- Set **Telephone Number** to **112** so that the number is inserted unchanged into the outgoing SIP INVITE message.



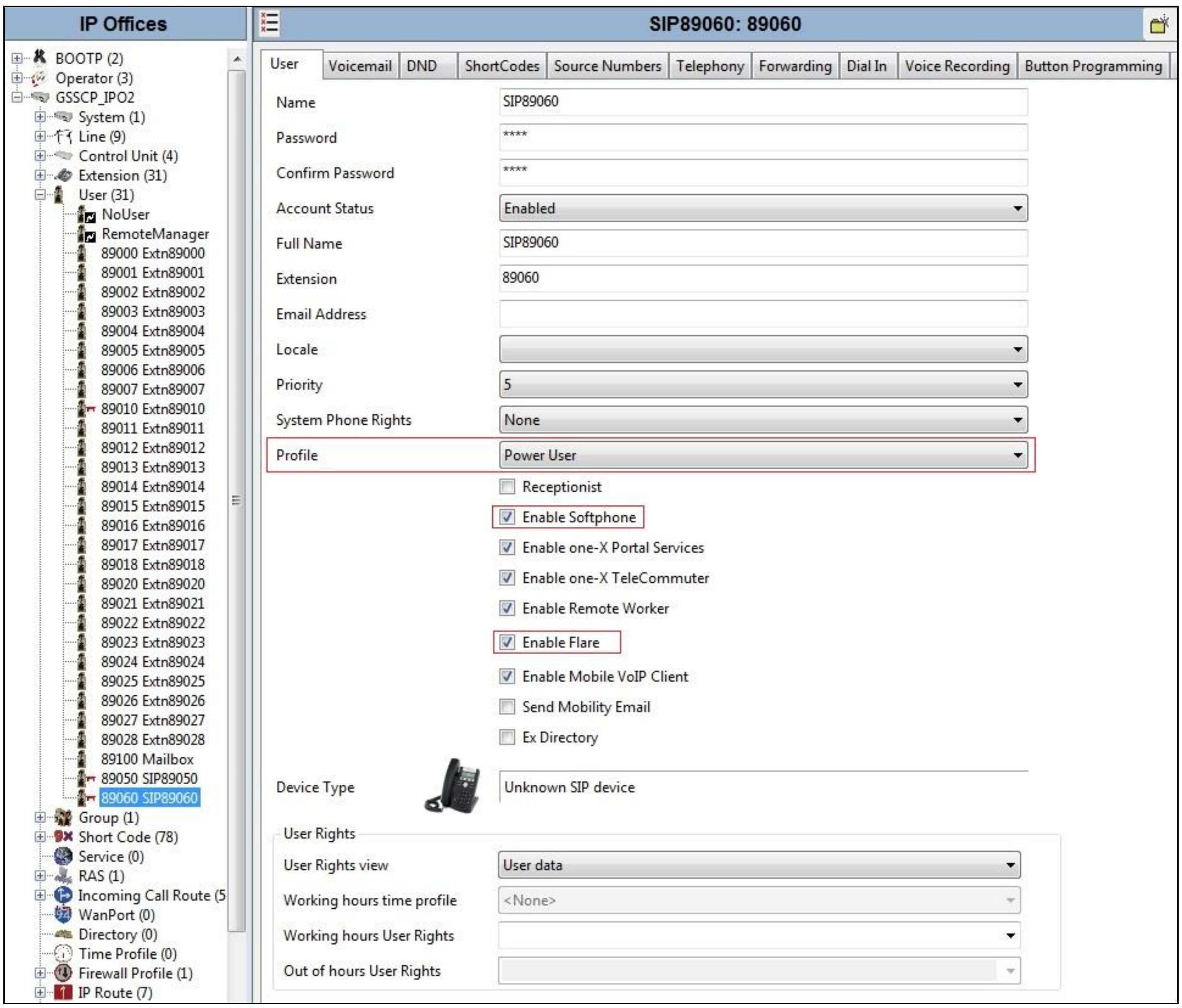
5.8. User

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 selected.

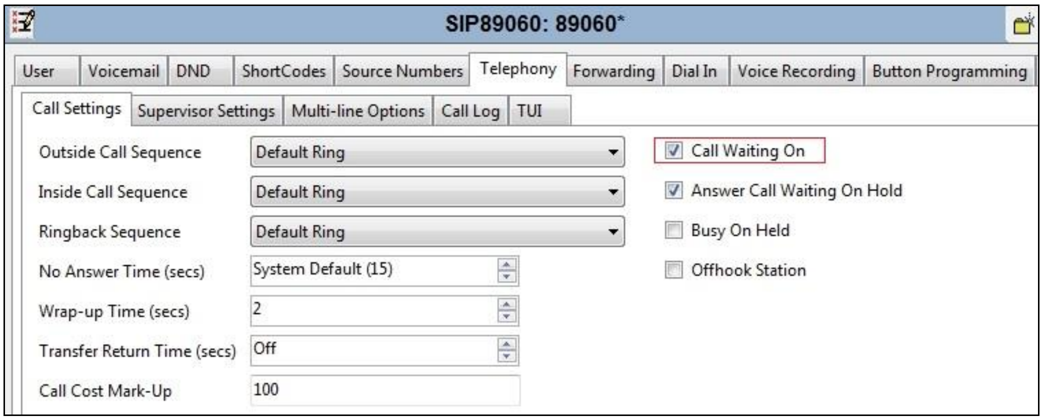
Change the **Name** of the User if required, this will be used for login to the IP Office Softphone

- 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 from SIP endpoints on IP Office, the Call Waiting function is used. The screenshot over the page shows how this is set.

To turn on Call Waiting, navigate to **Telephony→Call Settings**. Check the **Call Waiting On** box.



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 SFR.

In the example below, one of the DDI numbers in the test range is used, though some of the digits have been obscured. On completion, click the **OK** button (not shown).

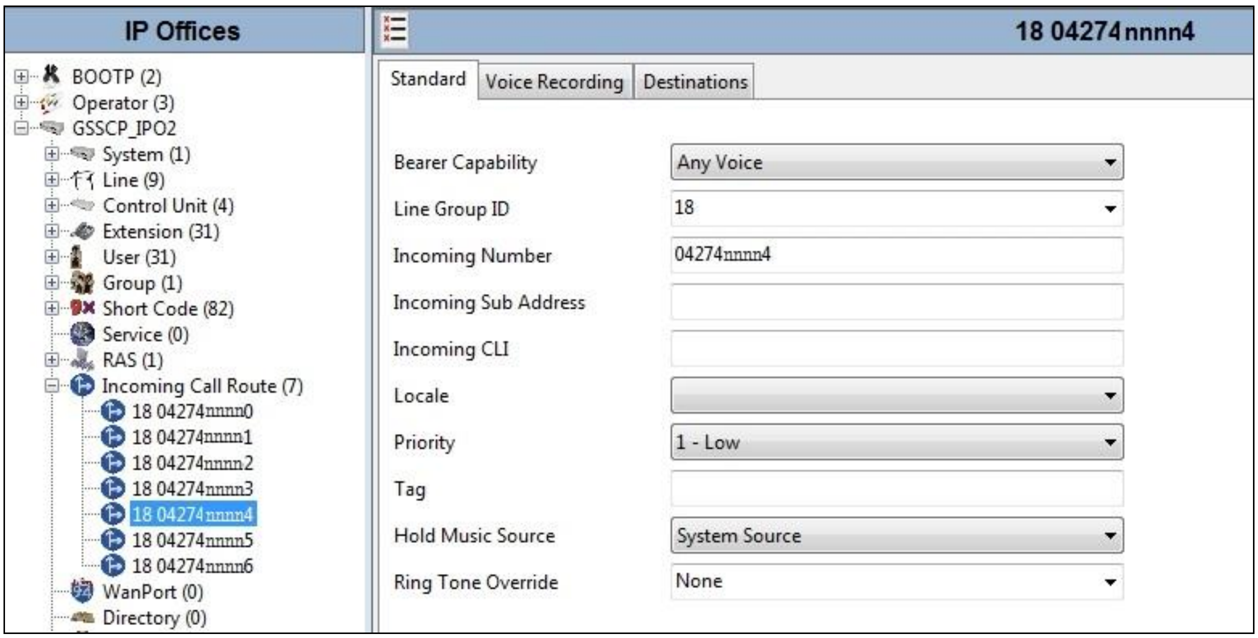
Note: The **Anonymous** box can be used to restrict Calling Line Identity (CLID).

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**, (not shown**)**.

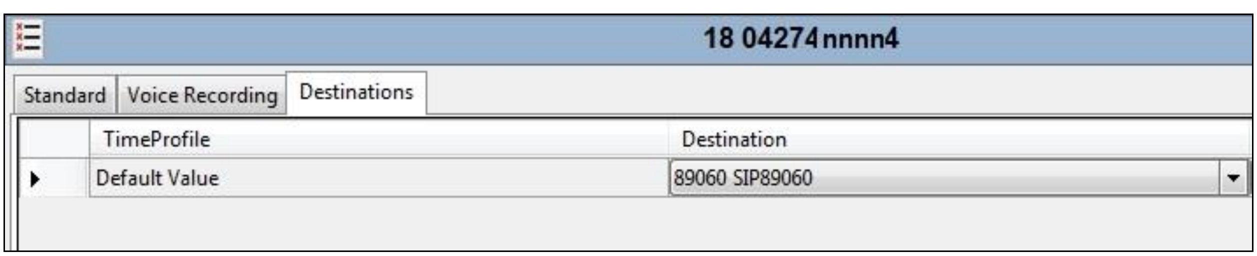
On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** 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.



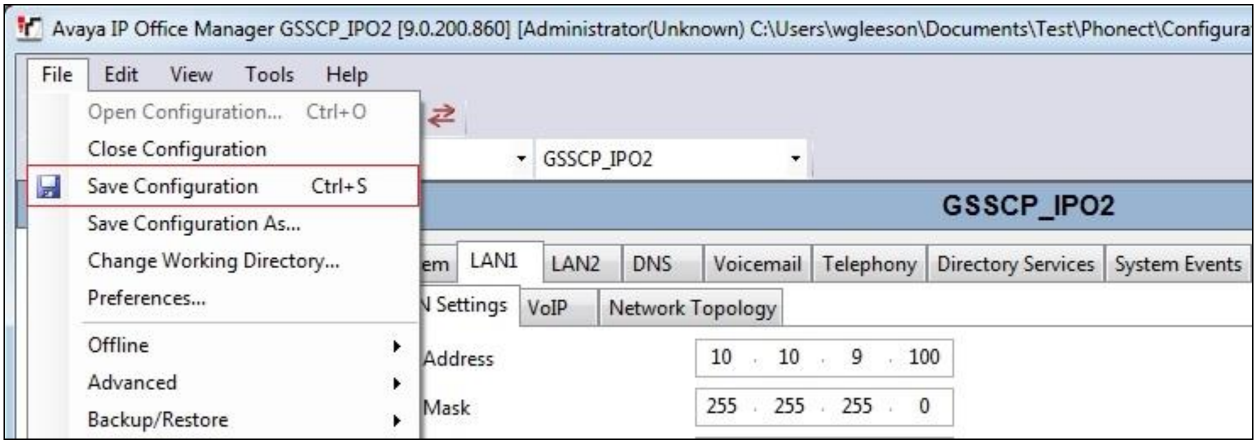
Note: A number of digits of the DDI have been obscured. Number format is international without a "+"prefix.

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 89060.



5.10. 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. SFR Collecte SIP Configuration

SFR 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. SFR will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

- Domain Name or IP address of SFR 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.

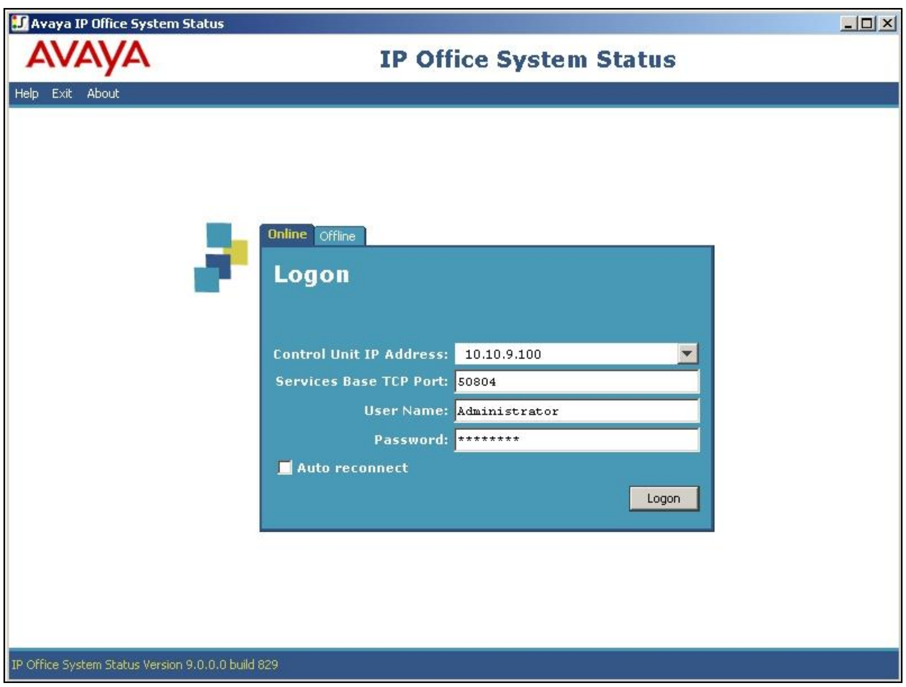
7. Verification Steps

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

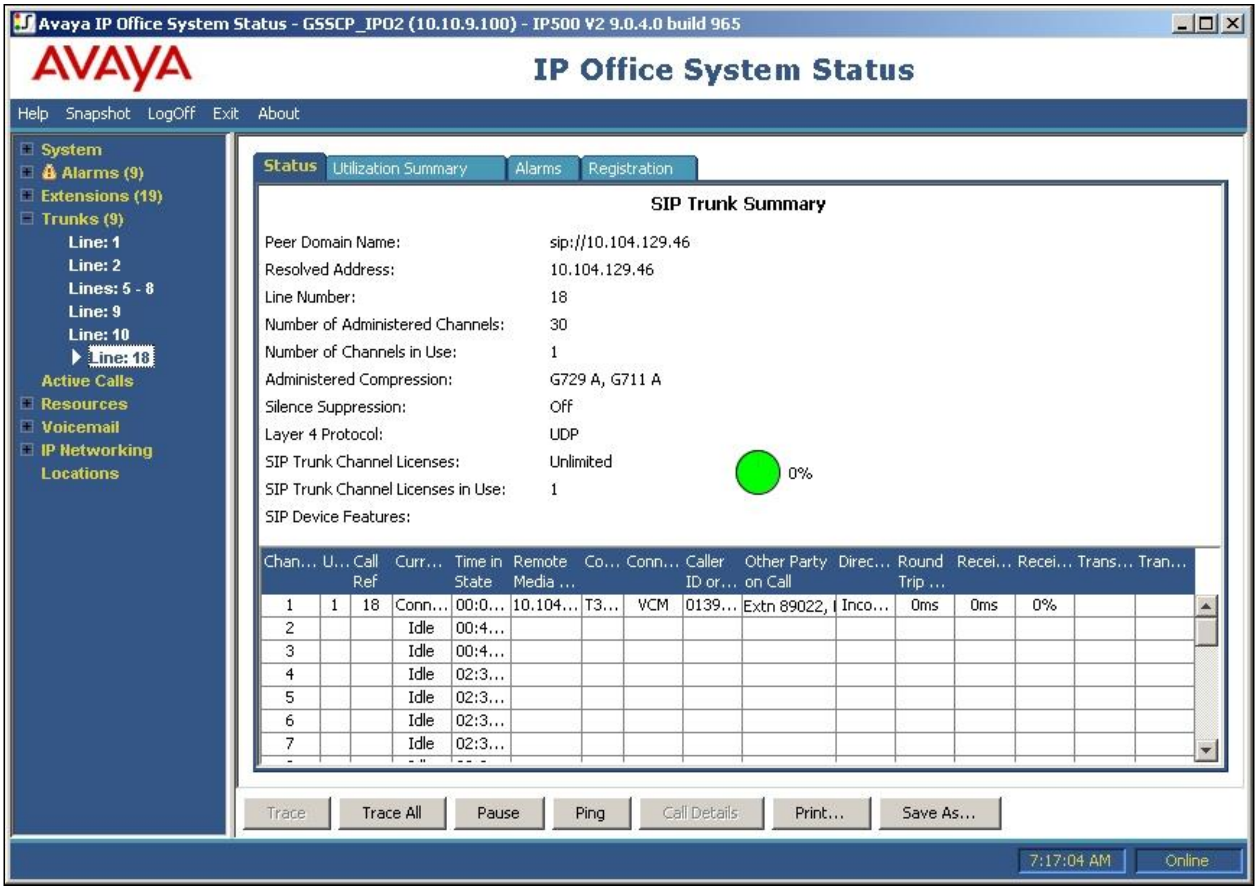
7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. A Windows 7 PC was used for testing and the application was opened by pressing the **Start** button (not shown) and selecting **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.



8. Conclusion

All tests for SFR Collecte SIP were completed. Observations for the testing are listed in **Section 2.2**.

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, 6th November 2014.

[3] *IP Office Application Server 9.0 Installation and Maintenance*, Document number15-601011, 13th November 2014.

[4] *IP Office Manager*, Document number 15-601011, 8th December 2014

[5] *IP Office 9.0 Using System Status*, Document number15-601758, 30th November 2014.

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

[7] *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.