



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.

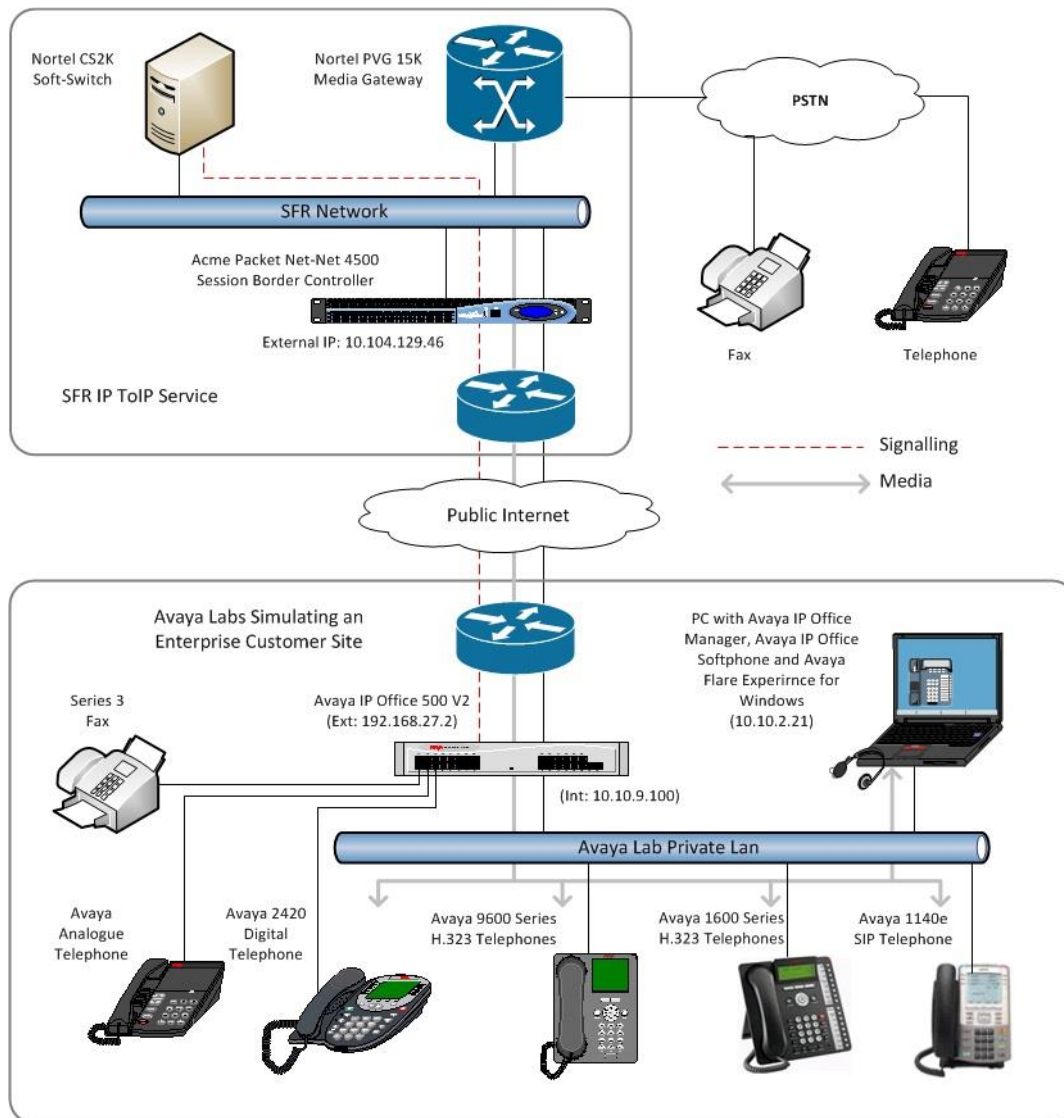


Figure 1: SFR Collecte SIP to Avaya IP Office Topology

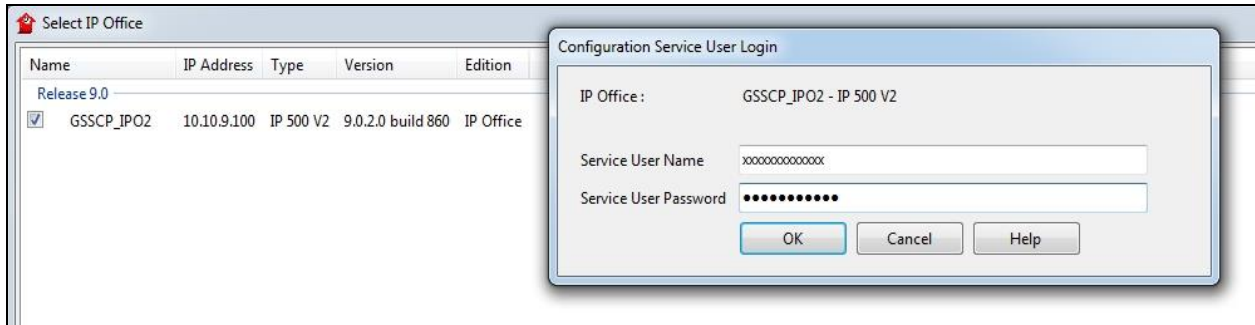
4. Equipment and Software Validated

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

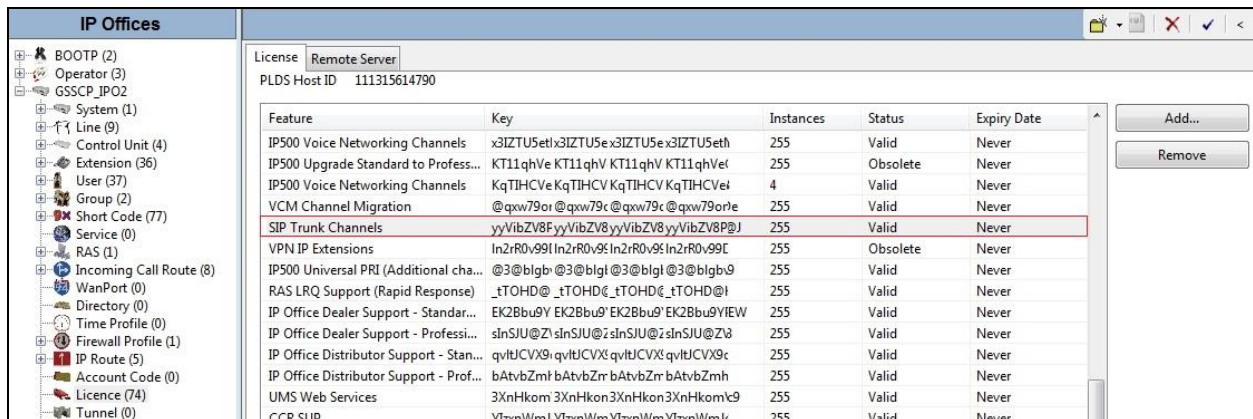
5. Configure Avaya IP Office

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

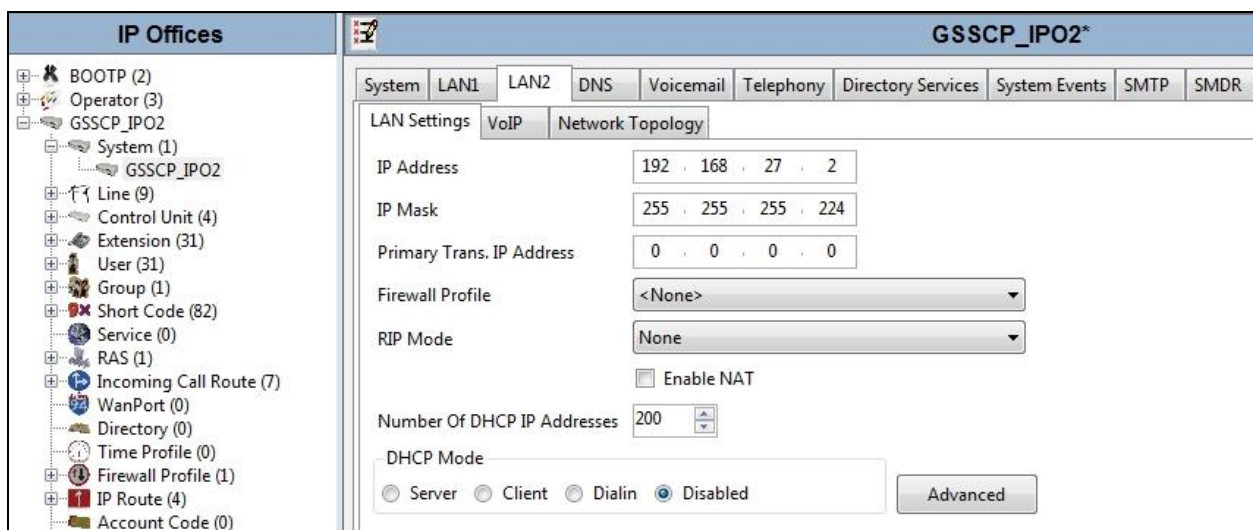
Navigate to **License** in the Navigation Pane. In the Details Pane verify that the **License Status** for **SIP Trunk Channels** is Valid and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by SFR.



Feature	Key	Instances	Status	Expiry Date
IP500 Voice Networking Channels	x3IZTU5ethx3IZTU5ex3IZTU5ex3IZTU5eth	255	Valid	Never
IP500 Upgrade Standard to Profess...	KT11qhVe KT11qhV KT11qhV KT11qhVeI	255	Obsolete	Never
IP500 Voice Networking Channels	KqTlHCVe KqTlHCV KqTlHCV KqTlHCVeI	4	Valid	Never
VCM Channel Migration	@qxw79or @qxw79c @qxw79c @qxw79orle	255	Valid	Never
SIP Trunk Channels	yyVibZv8FyyVibZv8yyVibZv8yyVibZv8P@J	255	Valid	Never
VPN IP Extensions	In2rR0v99I In2rR0v9s In2rR0v9s In2rR0v99I	255	Obsolete	Never
IP500 Universal PRI (Additional cha...	@3@blgb @3@blgl @3@blgl @3@blgb9	255	Valid	Never
RAS LRQ Support (Rapid Response)	_tTOHD@ _tTOHD@ _tTOHD@ _tTOHD@I	255	Valid	Never
IP Office Dealer Support - Standar...	EK2Bbu9Y EK2Bbu9 EK2Bbu9 EK2Bbu9YIEW	255	Valid	Never
IP Office Dealer Support - Profess...	sInSIU@Zi sInSIU@i sInSIU@i sInSIU@Zi	255	Valid	Never
IP Office Distributor Support - Stan...	qvtUCVX9 qvtUCVX9 qvtUCVX9 qvtUCVX9c	255	Valid	Never
IP Office Distributor Support - Prof...	bAtvbZmI bAtvbZm bAtvbZm bAtvbZmh	255	Valid	Never
UMS Web Services	3XnHkom 3XnHkon 3XnHkon 3XnHkomc9	255	Valid	Never
CCR SUP	YlznWm YlznWm YlznWm YlznWmk	255	Valid	Never

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).



System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR
LAN Settings									
IP Address: 192 . 168 . 27 . 2									
IP Mask: 255 . 255 . 255 . 224									
Primary Trans. IP Address: 0 . 0 . 0 . 0									
Firewall Profile: <None>									
RIP Mode: None									
<input type="checkbox"/> Enable NAT									
Number Of DHCP IP Addresses: 200									
DHCP Mode: <input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled									
Advanced									

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.

The screenshot shows the 'GSSCP_IPO2' configuration window with the 'VoIP' tab selected. The 'SIP Trunks Enable' and 'SIP Registrar Enable' checkboxes are checked. The 'Domain Name' field is empty. The 'Layer 4 Protocol' section shows 'UDP' selected with a port of 5060, and 'TCP' also selected with a port of 5060. The 'Challenge Expiry Time (secs)' is set to 10. The 'RTP' section shows 'Port Number Range' with a minimum of 49152 and a maximum of 53246. The 'Port Number Range (NAT)' section also shows a minimum of 49152 and a maximum of 53246. The 'Enable RTCP Monitoring on Port 5005' checkbox is checked. The 'Keepalives' section shows 'Scope' set to 'RTP' and 'Periodic timeout' set to 1. The 'Initial keepalives' dropdown is set to 'Enabled'. The 'DiffServ Settings' section shows 'DSCP (Hex)' set to B8, 'Video DSCP (Hex)' set to B8, 'DSCP Mask (Hex)' set to FC, and 'SIG DSCP (Hex)' set to 88. Below these, the 'DSCP' is set to 46, 'Video DSCP' is set to 46, 'DSCP Mask' is set to 63, and 'SIG DSCP' is set to 34.

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**.

The screenshot shows the 'GSSCP_IPO2*' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and settings:

- STUN Server Address:** 0.0.0.0
- STUN Port:** 3478
- Firewall/NAT Type:** Open Internet
- Binding Refresh Time (seconds):** 300
- Public IP Address:** 0 . 0 . 0 . 0
- Public Port:**
 - UDP: 570
 - TCP: 0
 - TLS: 0
- ☐ Run STUN on startup

Buttons for 'Run STUN' and 'Cancel' are located to the right of the Public IP Address field.

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).

The screenshot displays the 'GSSCP_IPO2' configuration window, specifically the 'Telephony' tab. The interface is divided into several sections:

- Analogue Extensions:** Includes dropdowns for 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type1), and 'Default Ring Back Sequence' (Ring Type2). There is also a checkbox for 'Restrict Analogue Extension Ringer Voltage'.
- Dial Delay Time (secs):** A numeric input field set to 4.
- Dial Delay Count:** A numeric input field set to 0.
- Default No Answer Time (secs):** A numeric input field set to 15.
- Hold Timeout (secs):** A numeric input field set to 0.
- Park Timeout (secs):** A numeric input field set to 300.
- Ring Delay (secs):** A numeric input field set to 5.
- Call Priority Promotion Time (secs):** A dropdown menu set to 'Disabled'.
- Default Currency:** A dropdown menu set to 'EUR'.
- Default Name Priority:** A dropdown menu set to 'Favour Trunk'.
- Media Connection Preservation:** A dropdown menu set to 'Disabled'.
- Companding Law:** A section with two sub-sections: 'Switch' and 'Line'. Both have radio buttons for 'U-Law' and 'A-Law'. 'A-Law' is selected for both.
- Other Settings:** A list of checkboxes including 'DSS Status', 'Auto Hold', 'Dial By Name', 'Show Account Code', 'Inhibit Off-Switch Forward/Transfer' (unchecked), 'Restrict Network Interconnect', 'Drop External Only Impromptu Conference', 'Visually Differentiate External Call', 'Unsupervised Analog Trunk Disconnect Handling', 'High Quality Conferencing', 'Strict SIPs', and 'Digital/Analogue Auto Create User'.

5.4. System Twinning Settings

Navigate to the **Twining** 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).

The screenshot shows the 'GSSCP_IP02*' configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is checked. Below it, there is a text field labeled 'Calling party information for Mobile Twinning' which is currently empty.

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).

The screenshot shows the 'GSSCP_IP02' configuration window with the 'Codecs' tab selected. The 'RFC2833 Default Payload' is set to '101'. The 'Available Codecs' list includes G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' area shows 'Unused' codecs (G.711 ULAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ) and 'Selected' codecs (G.711 ALAW 64K, G.729(a) 8K CS-ACELP). Horizontal arrows are used to move codecs between the 'Unused' and 'Selected' lists.

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:

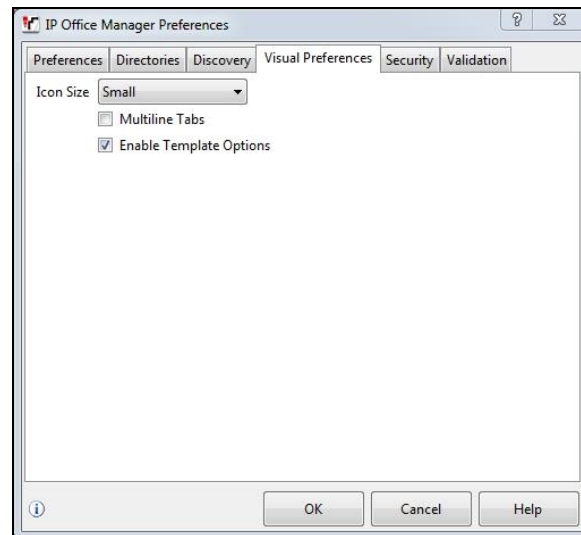
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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**.

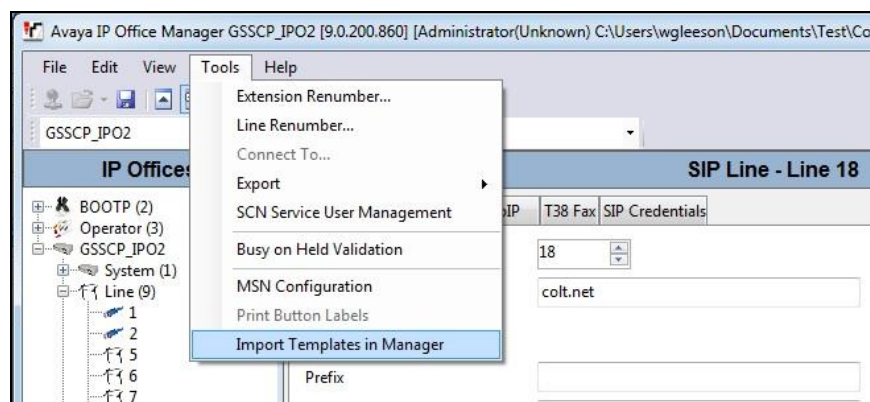
5.6.1. SIP Line From Template

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**.

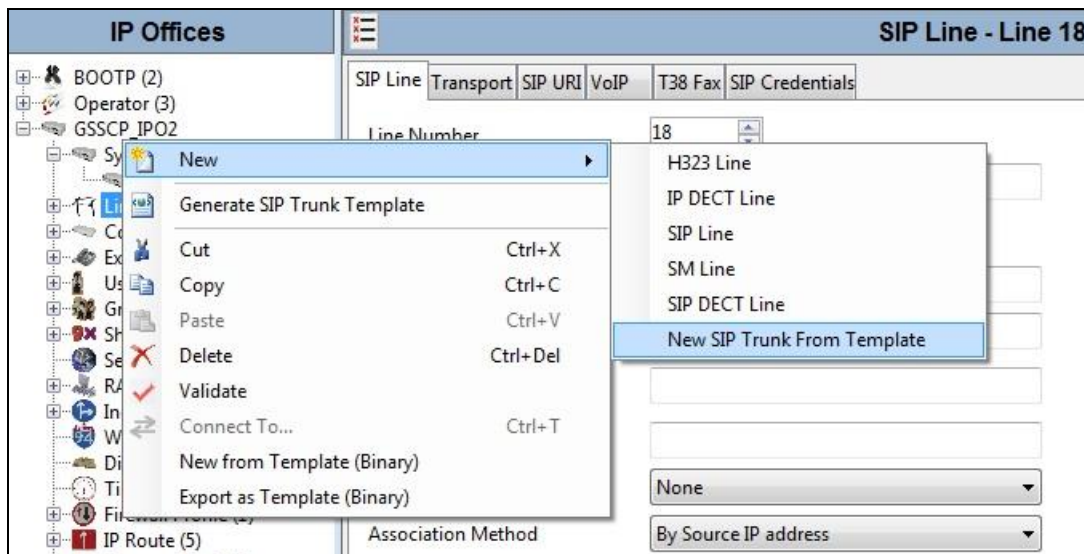


Import the template into IP Office Manager. Select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus. The default template location is **C:\Program Files\Avaya\IP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

The screenshot shows the 'SIP Line - Line 18' configuration window. On the left is a tree view under 'IP Offices' with various system components like BOOTP, Operator, GSSCP, System, Line, Control Unit, Extension, User, Group, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, Licence, Tunnel, User Rights, Auto Attendant, ARS, RAS Location Request, and Location. The main area is the 'SIP Line' tab, which has sub-tabs for Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'SIP Line' sub-tab is active. It contains the following fields and settings:

- Line Number: 18
- ITSP Domain Name: (empty field)
- In Service: ☒
- URI Type: SIP
- Prefix: (empty field)
- Check OOS: ☐
- National Prefix: (empty field)
- Call Routing Method: Request URI
- Country Code: (empty field)
- Originator number for forwarded and twinning calls: (empty field)
- International Prefix: (empty field)
- Name Priority: System Default
- Send Caller ID: None
- Caller ID from From header: ☒
- Association Method: By Source IP address
- Send From In Clear: ☐
- User-Agent and Server Headers: (empty field)
- Service Busy Response: 486 - Busy Here
- Action on CAC Location Limit: Allow Voicemail
- REFER Support: ☒
 - Incoming: Never
 - Outgoing: Never
- Method for Session Refresh: Auto
- Session Timer (seconds): On Demand
- Media Connection Preservation: Disabled

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).

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to '10.104.129.46'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Use Network Topology Info' is set to 'None', 'Send Port' is '5060', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '8 . 8 . 8 . 8' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

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.

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'SIP URI' tab selected. The 'Channel' column is highlighted. On the right, there are buttons for 'Add...', 'Remove', and 'Edit...'. The table below shows columns for Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls.

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.

The 'Edit Channel' dialog box contains the following fields and values:

Field	Value
Via	<None>
Local URI	Use Internal Data
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	Use Internal Data
Registration	1: 00000000018
Incoming Group	18
Outgoing Group	18
Max Calls per Channel	10

Buttons: OK, Cancel

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:

The 'Edit Channel' dialog box contains the following fields and values:

Field	Value
Via	<None>
Local URI	04274nnnn5
Contact	Use Internal Data
Display Name	Use Internal Data
PAI	None
Registration	0: <None>
Incoming Group	18
Outgoing Group	18
Max Calls per Channel	10

Buttons: OK, Cancel

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).

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'VoIP' tab selected. The 'Codec Selection' section has a dropdown set to 'Custom'. Below it, the 'Unused' box contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' box contains 'G.729(a) 8K CS-ACELP' and 'G.711 ALAW 64K'. Between the boxes are arrows for moving items and a double arrow for priority. On the right, checkboxes for 'VoIP Silence Suppression', 'Allow Direct Media Path', 'Re-invite Supported' (checked), 'Codec Lockdown', 'PRACK/100rel Supported' (checked), 'Force direct media with phones', and 'G.711 Fax ECAN' are shown. At the bottom, 'Fax Transport Support' is set to 'T38 Fallback', 'Location' is 'Cloud', 'Call Initiation Timeout (s)' is '4', and 'DTMF Support' is 'RFC2833'.

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).

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'T38 Fax' tab selected. The window has a tabbed interface with 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'T38 Fax' tab contains the following settings:

- T38 Fax Version: 3 (dropdown)
- Transport: UDPTL (dropdown)
- Redundancy section:
 - Low Speed: 0 (spinner)
 - High Speed: 0 (spinner)
- TCF Method: Trans TCF (dropdown)
- Max Bit Rate (bps): 14400 (dropdown)
- EFlag Start Timer (msecs): 2600 (spinner)
- EFlag Stop Timer (msecs): 2300 (spinner)
- Tx Network Timeout (secs): 150 (spinner)
- Checkboxes on the right:
 - ☒ Scan Line Fix-up
 - ☒ TFO Enhancement
 - ☐ Disable T30 ECM
 - ☐ Disable EFlags For First DIS
 - ☐ Disable T30 MR Compression
 - ☐ NSF Override
- Country Code: 0 (spinner)
- Vendor Code: 0 (spinner)
- ☒ Use Default Values (checkbox at the bottom left)

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).

The screenshot shows the 'Short Code' configuration window. On the left, the 'IP Offices' list includes various codes, with '90N;' selected. The main configuration area on the right is titled '90N;: Dial' and contains the following fields:

Code	90N;
Feature	Dial
Telephone Number	0N
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

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.

The screenshot shows the 'Short Code' configuration window for the code '112'. The window title is '112: Dial'. The configuration fields are as follows:

Code	112
Feature	Dial
Telephone Number	112
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

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.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' navigation pane is expanded, showing a tree structure with 'User (31)' selected. The main area displays the configuration for user 'SIP89060: 89060'. The 'User' tab is active, showing fields for Name, Password, Confirm Password, Account Status, Full Name, Extension, Email Address, Locale, Priority, System Phone Rights, Profile, and checkboxes for various services. The 'Profile' dropdown is set to 'Power User', and 'Enable Softphone' and 'Enable Flare' are checked. The 'Device Type' is 'Unknown SIP device'.

User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming
Name	SIP89060								
Password	****								
Confirm Password	****								
Account Status	Enabled								
Full Name	SIP89060								
Extension	89060								
Email Address									
Locale									
Priority	5								
System Phone Rights	None								
Profile	Power User								
<input type="checkbox"/> Receptionist									
<input checked="" type="checkbox"/> Enable Softphone									
<input checked="" type="checkbox"/> Enable one-X Portal Services									
<input checked="" type="checkbox"/> Enable one-X TeleCommuter									
<input checked="" type="checkbox"/> Enable Remote Worker									
<input checked="" type="checkbox"/> Enable Flare									
<input checked="" type="checkbox"/> Enable Mobile VoIP Client									
<input type="checkbox"/> Send Mobility Email									
<input type="checkbox"/> Ex Directory									
Device Type	Unknown SIP device								
User Rights									
User Rights view: User data									
Working hours time profile: <None>									
Working hours User Rights:									
Out of hours User Rights:									

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.

The screenshot shows the 'Call Settings' tab for 'SIP89060: 89060*'. The 'Call Waiting On' checkbox is checked. Other settings include: Outside Call Sequence (Default Ring), Inside Call Sequence (Default Ring), Ringback Sequence (Default Ring), No Answer Time (secs) (System Default (15)), Wrap-up Time (secs) (2), Transfer Return Time (secs) (Off), and Call Cost Mark-Up (100).

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).

The screenshot shows the 'SIP' tab for 'SIP89060: 89060'. The 'SIP Name' and 'Contact' fields are both populated with '04274nnnn4'. The 'SIP Display Name (Alias)' field is populated with 'Extn89060'. The 'Anonymous' checkbox is unchecked.

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 Routes** 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.

The screenshot shows the 'IP Offices' configuration window. On the left is a navigation tree with 'Incoming Call Route (7)' expanded, showing several routes. The route '18 04274nnnn4' is selected. The main pane shows the 'Standard' tab for this route. The fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	04274nnnn4
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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

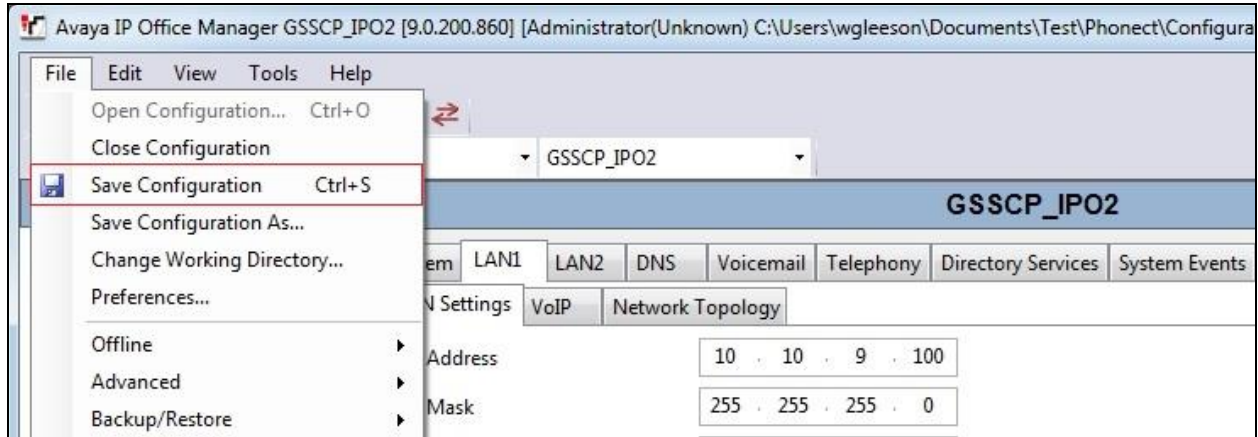
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89060.

The screenshot shows the 'Destinations' tab for the selected route. It displays a table with two columns: 'TimeProfile' and 'Destination'.

TimeProfile	Destination
Default Value	89060 SIP89060

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:

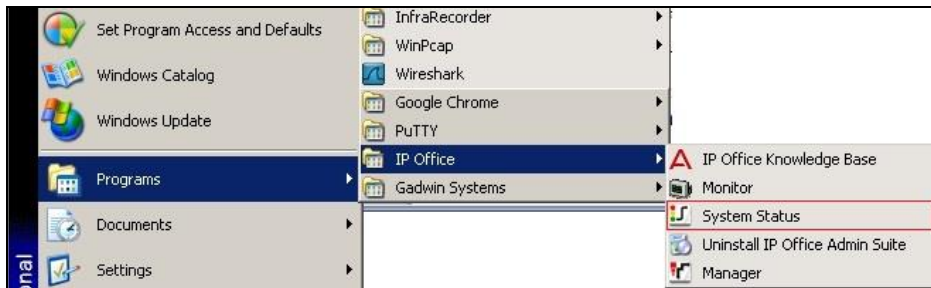
- Domain Name or IP address of SFR SIP proxy.
- Network SIP Domain.
- Supported codecs.
- DDI numbers.
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

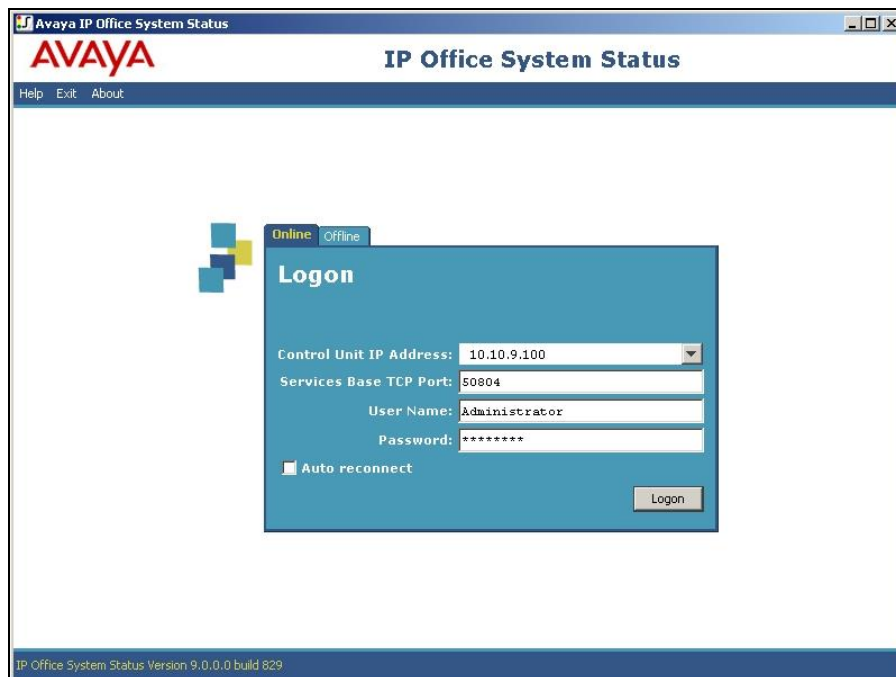
This section includes steps that can be used to verify that the configuration has been done correctly.

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.

The screenshot shows the Avaya IP Office System Status window. The left-hand menu is expanded to 'Trunks (9)', and 'Line: 18' is selected. The main window displays the 'SIP Trunk Summary' for Line 18. The summary includes the following information:

- Peer Domain Name: sip://10.104.129.46
- Resolved Address: 10.104.129.46
- Line Number: 18
- Number of Administered Channels: 30
- Number of Channels in Use: 1
- Administered Compression: G729 A, G711 A
- Silence Suppression: Off
- Layer 4 Protocol: UDP
- SIP Trunk Channel Licenses: Unlimited
- SIP Trunk Channel Licenses in Use: 1
- SIP Device Features: 0%

Below the summary is a table showing call details for Line 18. The table has columns for Channel, U..., Call Ref, Curr..., Time in State, Remote Media, Co..., Conn..., Caller ID or..., Other Party on Call, Direc..., Round Trip, Recei..., Recei..., Trans..., and Tran... The table shows 7 rows of data, with the first row indicating a call in progress (Conn... 00:0..., Idle 00:4..., 10.104..., T3..., VCM 0139..., Extn 89022, Inco..., 0ms, 0ms, 0%, ...).

At the bottom of the window, there are buttons for Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time 7:17:04 AM and the status Online.

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.

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