Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 9.1 with Avaya Session Border Controller for Enterprise Release 6.3 to support Vodafone Germany SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office R9.1 and the Avaya Session Border Controller for Enterprise 6.3 to support Vodafone Germany SIP Trunk Service.

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

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Vodafone Germany SIP Trunk service and Avaya IP Office. In the test configuration, the Avaya IP Office solution consists of an Avaya Session Border Controller for Enterprise Release 6.3, and Avaya IP Office 500 v2 Release 9.1 Preferred Edition, Avaya Voicemail Pro, Avaya Communicator for Windows, and Avaya H.323, SIP, digital, and analogue endpoints.

Avaya IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

The Avaya Session Border Controller for Enterprise (Avaya SBCE) is the point of connection between Avaya IP Office and Vodafone Germany SIP Trunk service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

Vodafone Germany SIP Trunk service provides PSTN access via a SIP trunk connected to the Vodafone Germany 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 and Avaya SBCE to connect to the Vodafone Germany SIP Trunk service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

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

- Incoming PSTN calls to various phone types including H.323, SIP, Digital and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider.
- Outgoing PSTN calls from various phone types including H.323, SIP, Digital, and Analogue telephones at the enterprise.

- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Inbound and outbound PSTN calls to/from an IP Office Softphone client.
- Calls using G.711A and G.729 codec's.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711A.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance.
- Caller ID presentation and Caller ID restriction.
- 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 test configuration was completed with successful results for Vodafone Germany's SIP Trunk service with the following observations:

- T.38 Fax transmission is not supported by Vodafone Germany.
- When there were no matching codecs in the SDP offer of an outbound call, "503 Service Unavailable" response was returned from the Vodafone Germany network. The more commonly used response is "488 Not Acceptable Here".
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked with the Emergency Services Operator.

2.3. Support

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

For technical support on Vodafone Germany products please visit the website at www.vodafone.de or contact an authorized Vodafone representative.

3. Reference Configuration

Figure 1 below illustrates the test configuration. The test configuration shows an enterprise site connected to the Vodafone Germany SIP Trunk service. Located at the enterprise site is an Avaya IP Office 500v2 with Avaya SBCE. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140e SIP Telephones, Avaya Analogue Telephone and fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as Avaya Communicator for Windows Softphone client. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

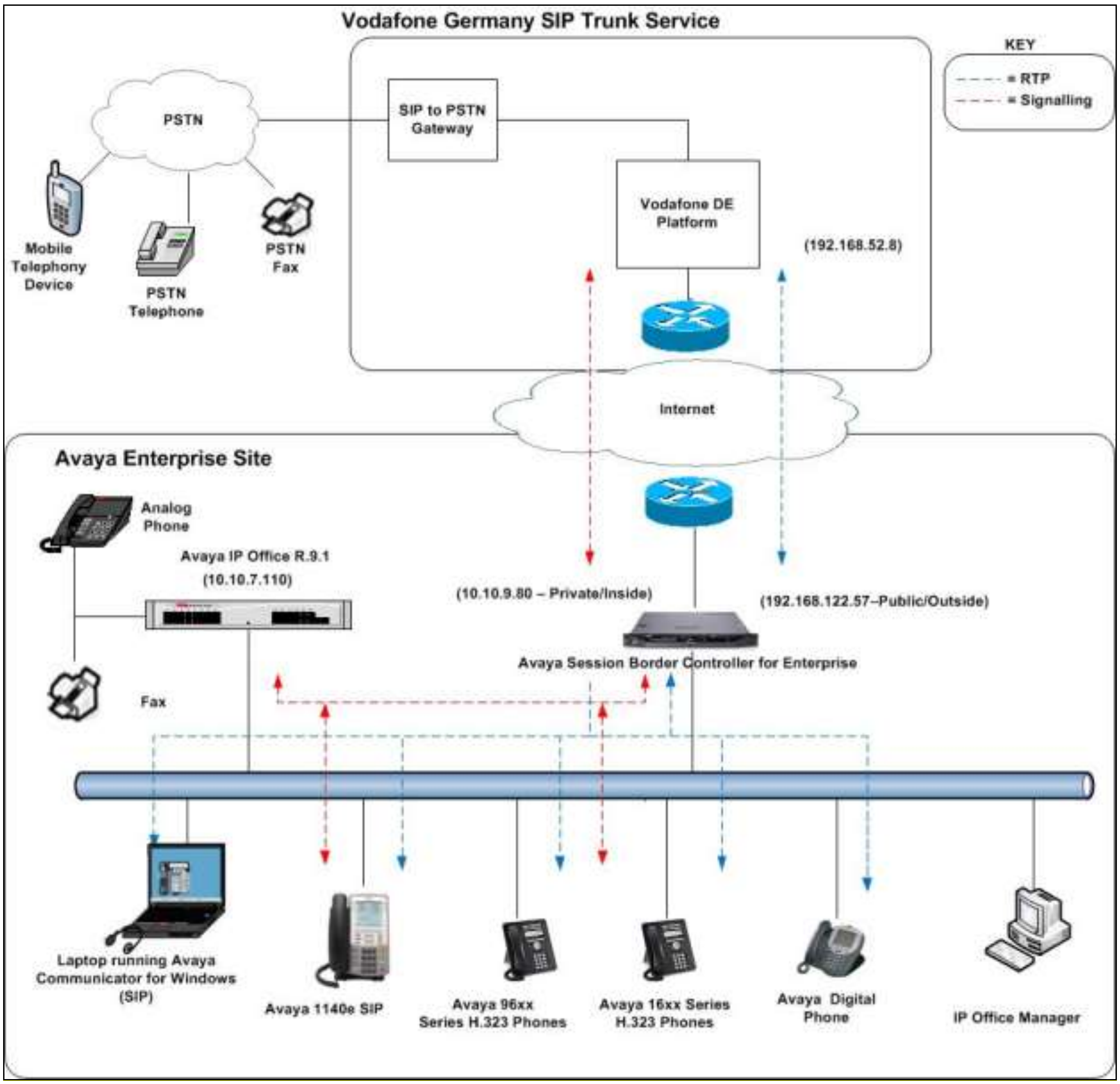


Figure 1: Test Setup Vodafone Germany SIP Trunk service to simulated Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya Session Border Controller for Enterprise	Version 6.3.000-19-4338
Avaya IP Office 500 V2	9.1.1.0 build 10
Avaya Voicemail Pro Client	9.1.1.0 build 3
Avaya 1603 Phone (H.323)	1.3.5
Avaya 9600 Series Phone (H.323)	6.3.0
Avaya Communicator for Windows (SIP)	2.0.3.30
Avaya 1140e (SIP)	FW: 04.04.18.00.bin
Avaya 98390 Analogue Phone	N/A
Vodafone Germany	
ACME Net-Net 4500 SBC	SCX6.4.0 MR-5 Patch 4 (build 482)
Italtel Softswitch	Ver.20.50.50 (VTCHX, VSSX, BSHX, OMS). Ver.20.60.10 (Loadbalancer, MRF IBCF).

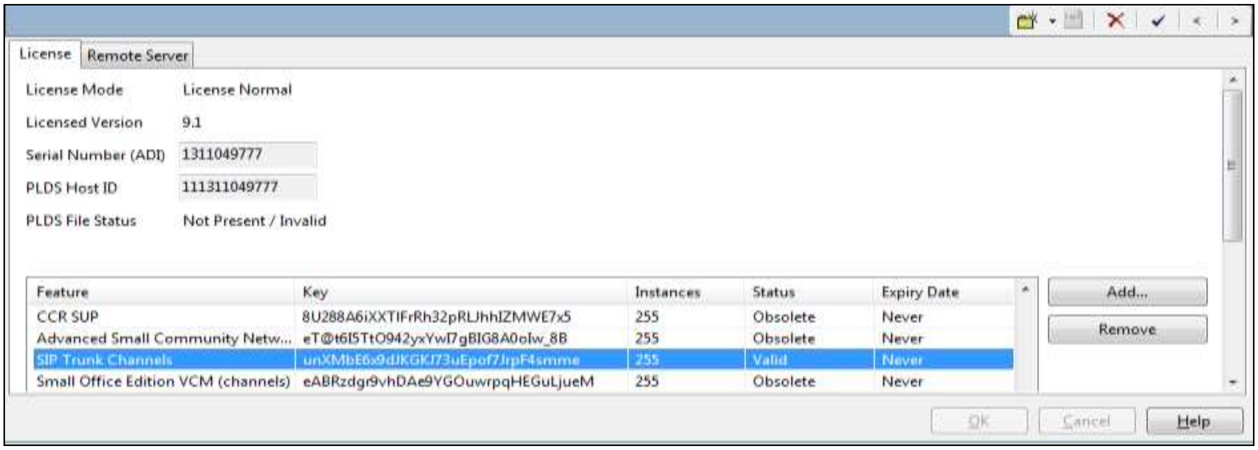
Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition **without T.38 Fax Service**.

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the Vodafone Germany SIP Trunk service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider (such as twinning) is assumed to already be in place.

5.1. Verify System Capacity

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



5.2. LAN Settings

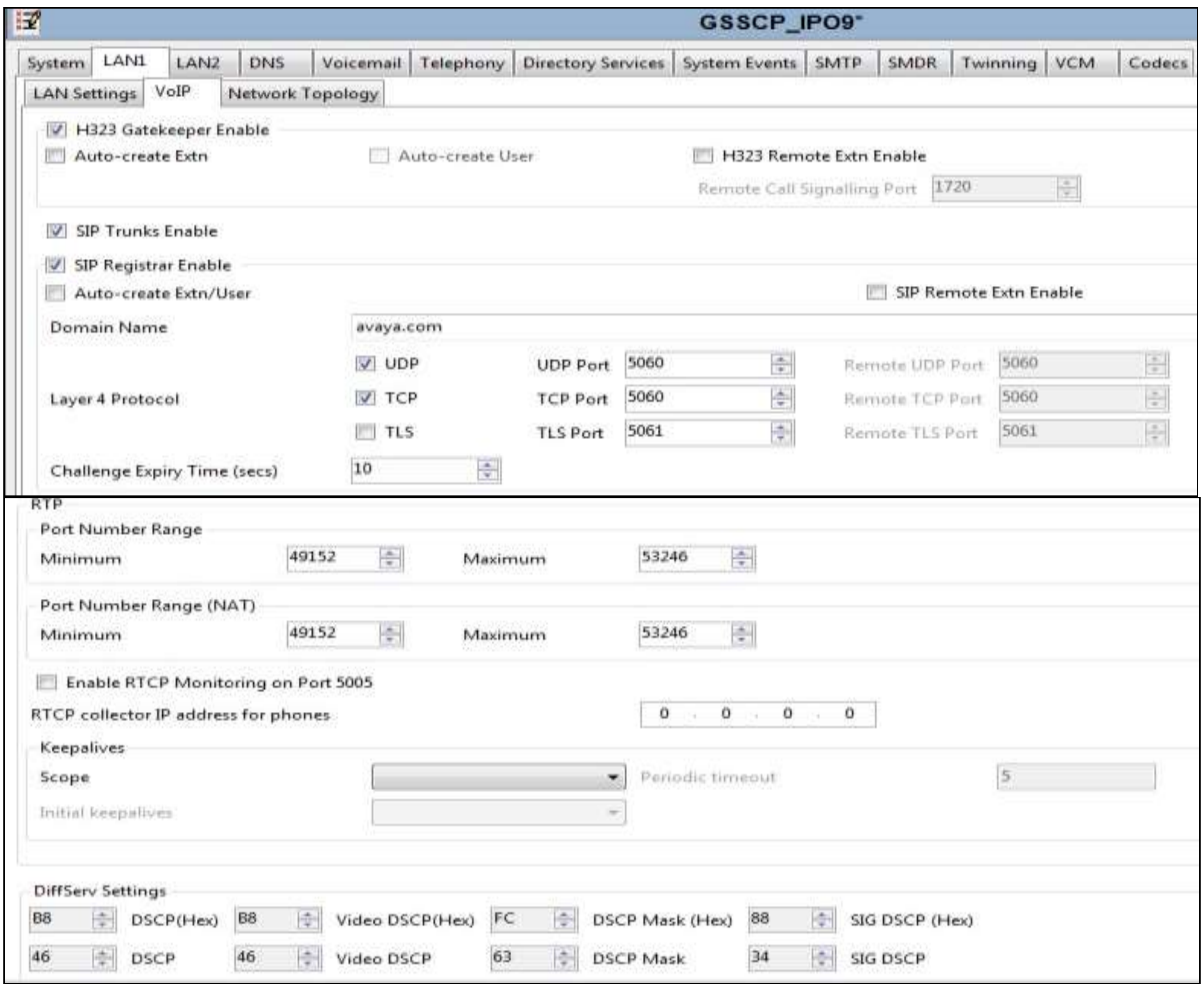
The IP500/IP500 V2 control units have 2 RJ45 Ethernet ports, physically marked as LAN and WAN. Within the system configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.

In the test configuration, the LAN1 port was used to connect the Avaya IP Office to the enterprise network. To access the LAN1 settings, first navigate to **System → GSSCP_IPO9** in the Navigation Pane where **GSSCP_IPO9** is the name of the IP Office. Navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the management interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

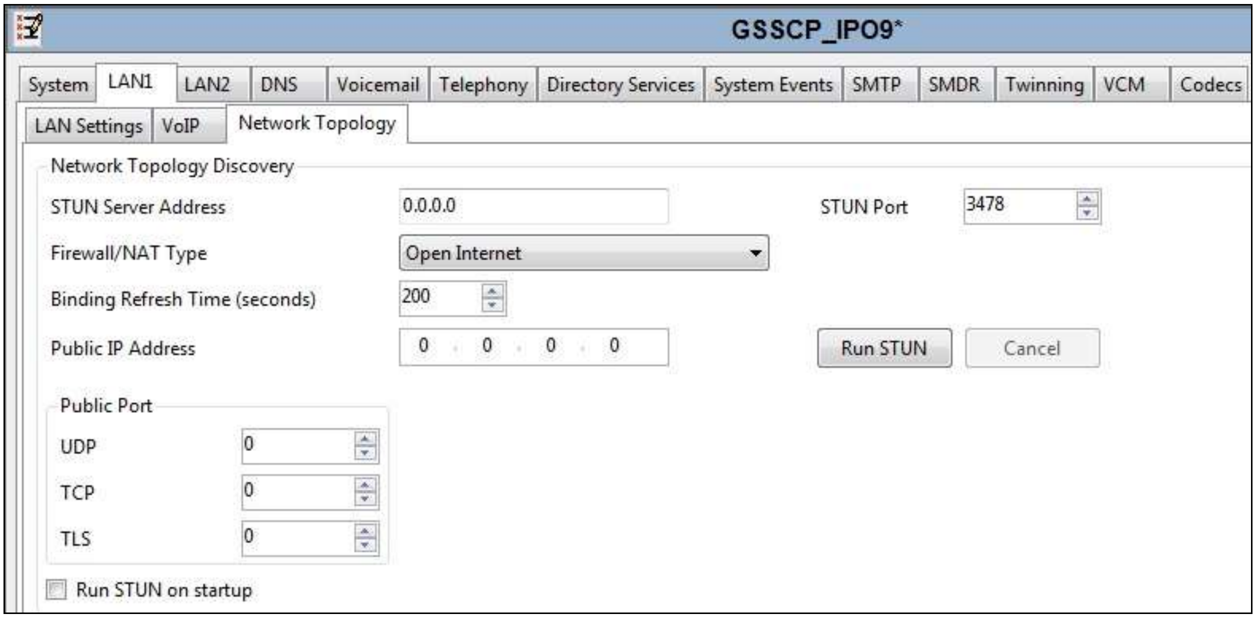
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The Avaya Communicator uses SIP. If Avaya Communicator along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **Domain Name** has been set to the customer premises equipment domain "**avaya.com**". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN 1 IP Address. All other parameters shown are default values.

The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN1.

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

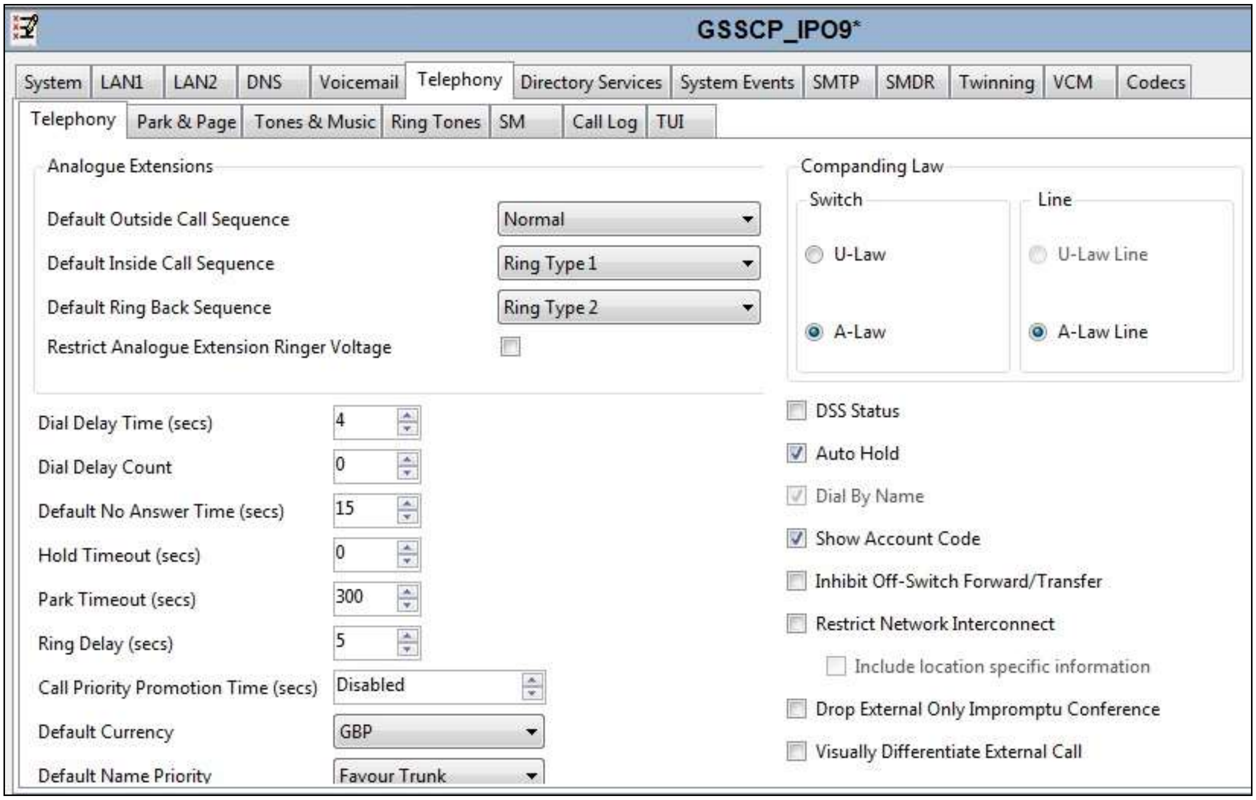


On the **Network Topology** tab, select the **Firewall/NAT Type** from the pulldown menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used as NAT was not required for this configuration, therefore resulting in no requirement for a STUN server. The **Use Network Topology Info** in the **SIP Line** was set to **None** in **Section** Error! Reference source not found.. Set **Binding Refresh Time (seconds)** to **200**. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. Default values were used for all other parameters. On completion, click the **OK** button (not shown).



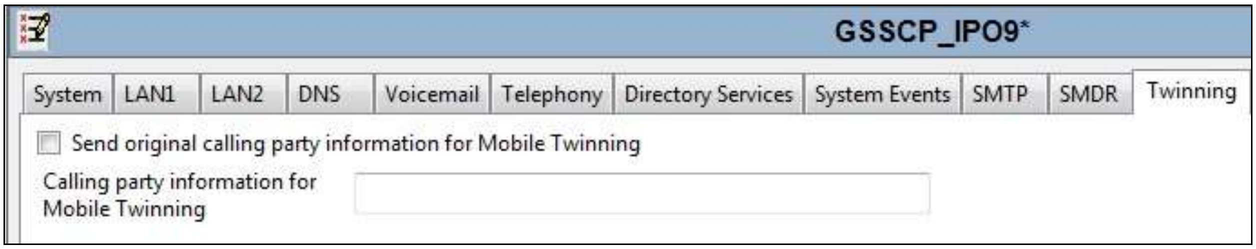
5.3. System Telephony Settings

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



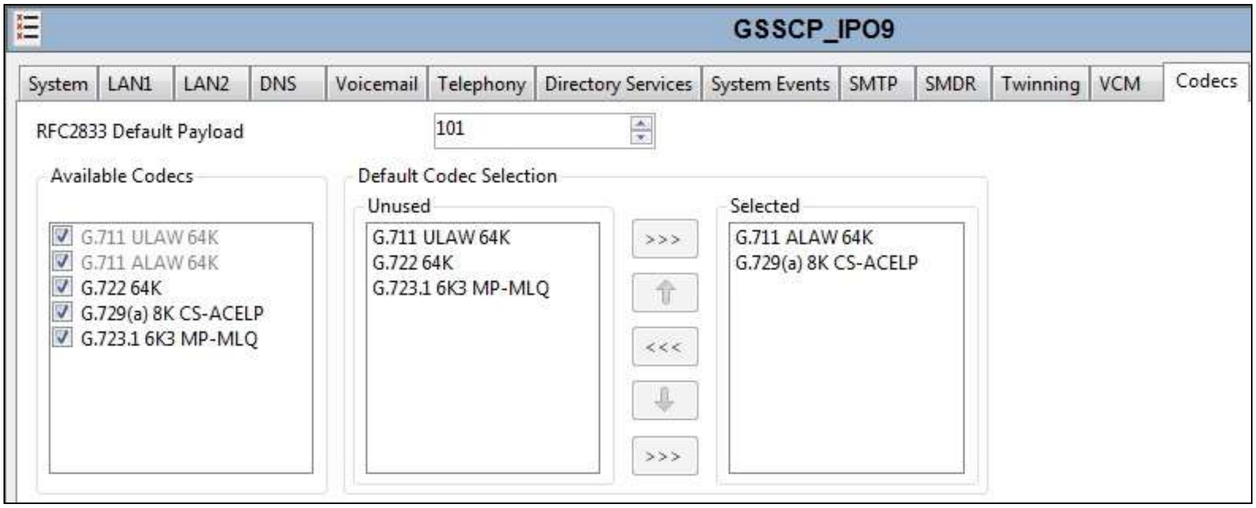
5.4. System Twinning Settings

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



5.5. Codec Settings

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



5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Vodafone Germany SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** to create the SIP Line from the template.

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

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.6.2**.

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

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

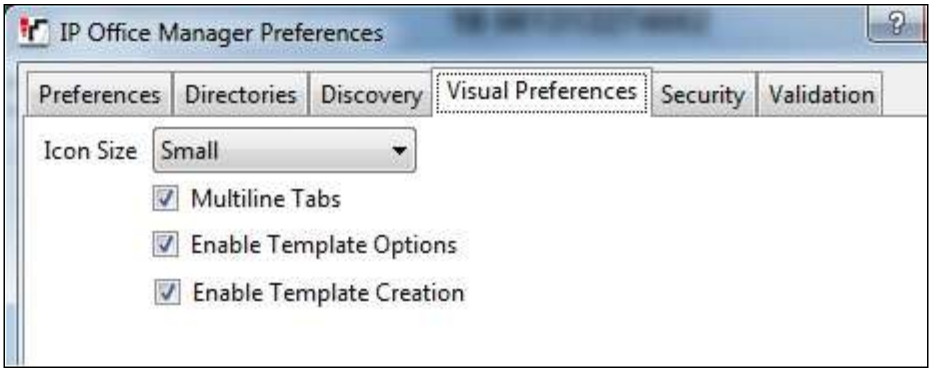
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

5.6.1. SIP Line From Template

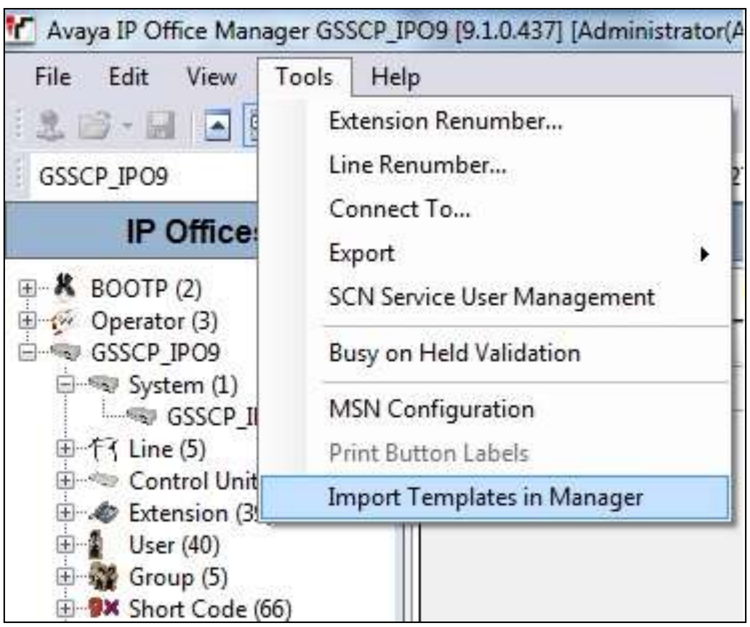
DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500v2) and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

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed. Rename the template file to **AF_Vodafone Germany_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

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

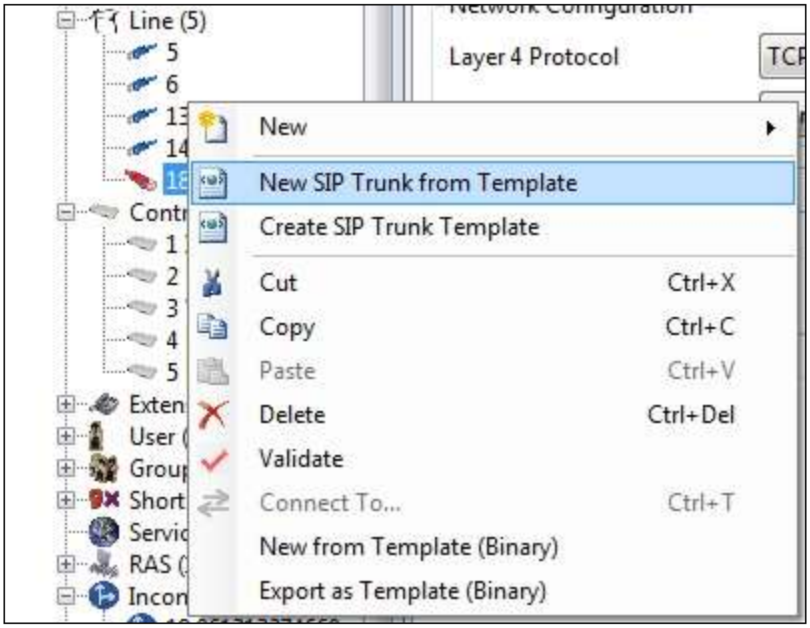


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**.



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

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



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

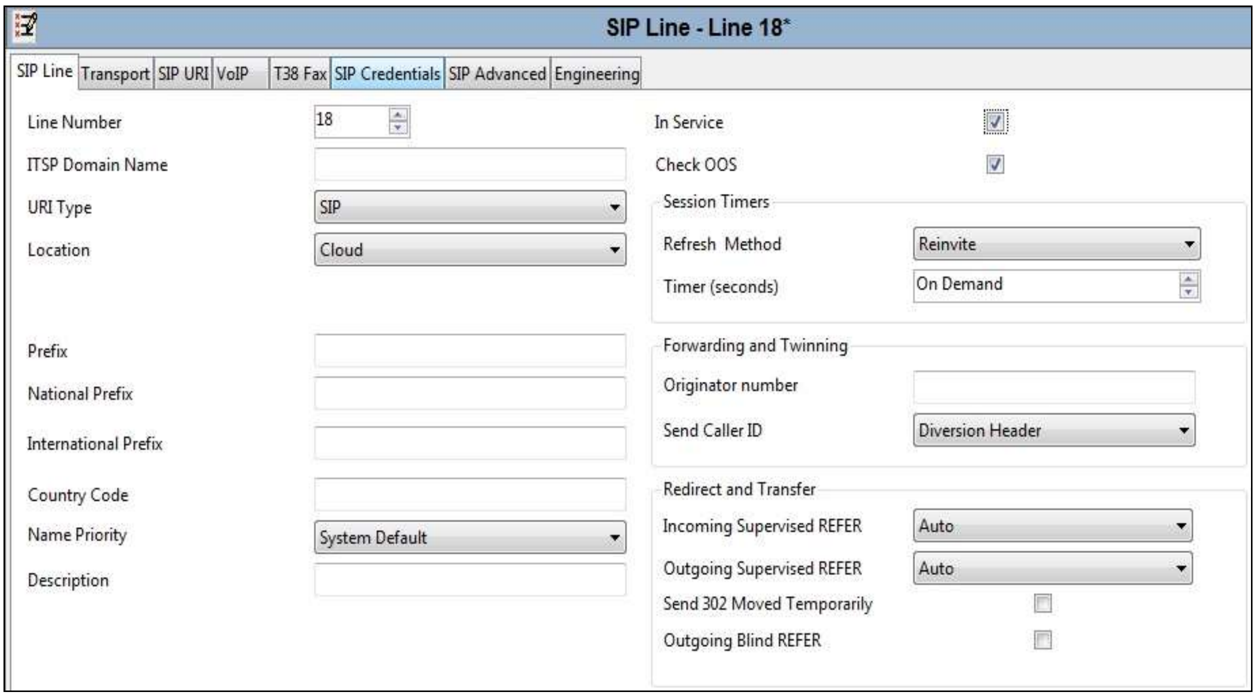
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

5.6.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** to a domain name provider by the Service Provider if required, however no ITSP Domain Name was used in this configuration.
- Ensure the **In Service** box is checked.
- Set **Refresh Method** to **Reinvite**.
- Set **Send Caller ID** to **Diversion Header**.
- Default values may be used for all other parameters.

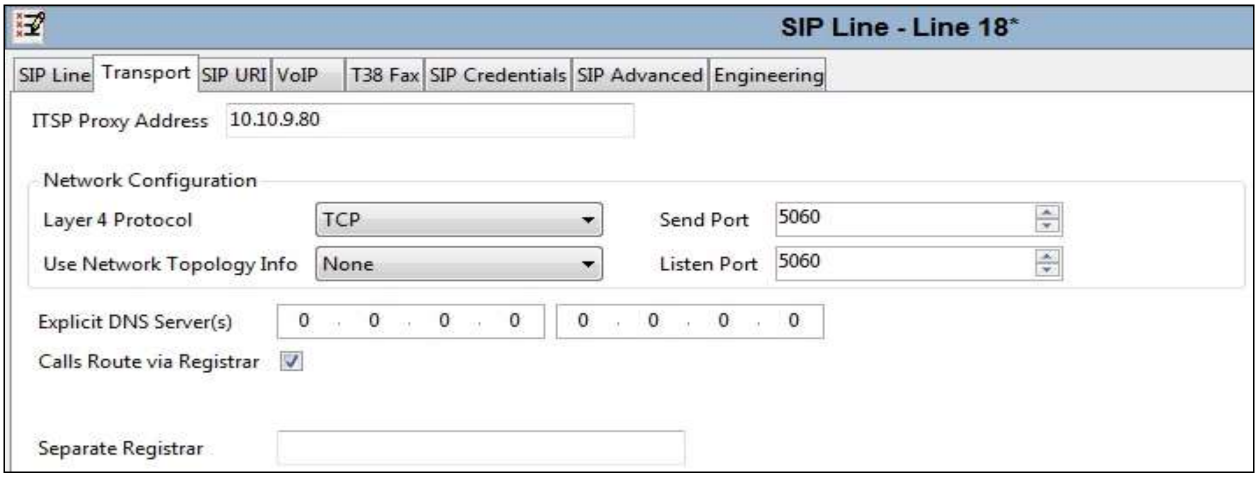
On completion, click the **OK** button (not shown).



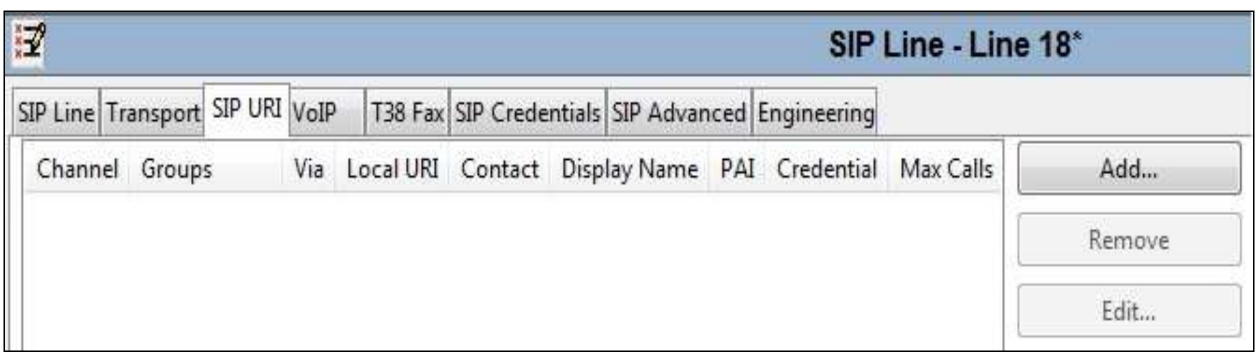
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the inside interface IP address's of the Avaya SBCE as shown in **Figure 1**.
- Set **Layer 4 Protocol** to **TCP**.
- Set **Send Port** to **5060** and **Listen Port** to **5060**.
- Set **Use Network Topology Info** to **None**.

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

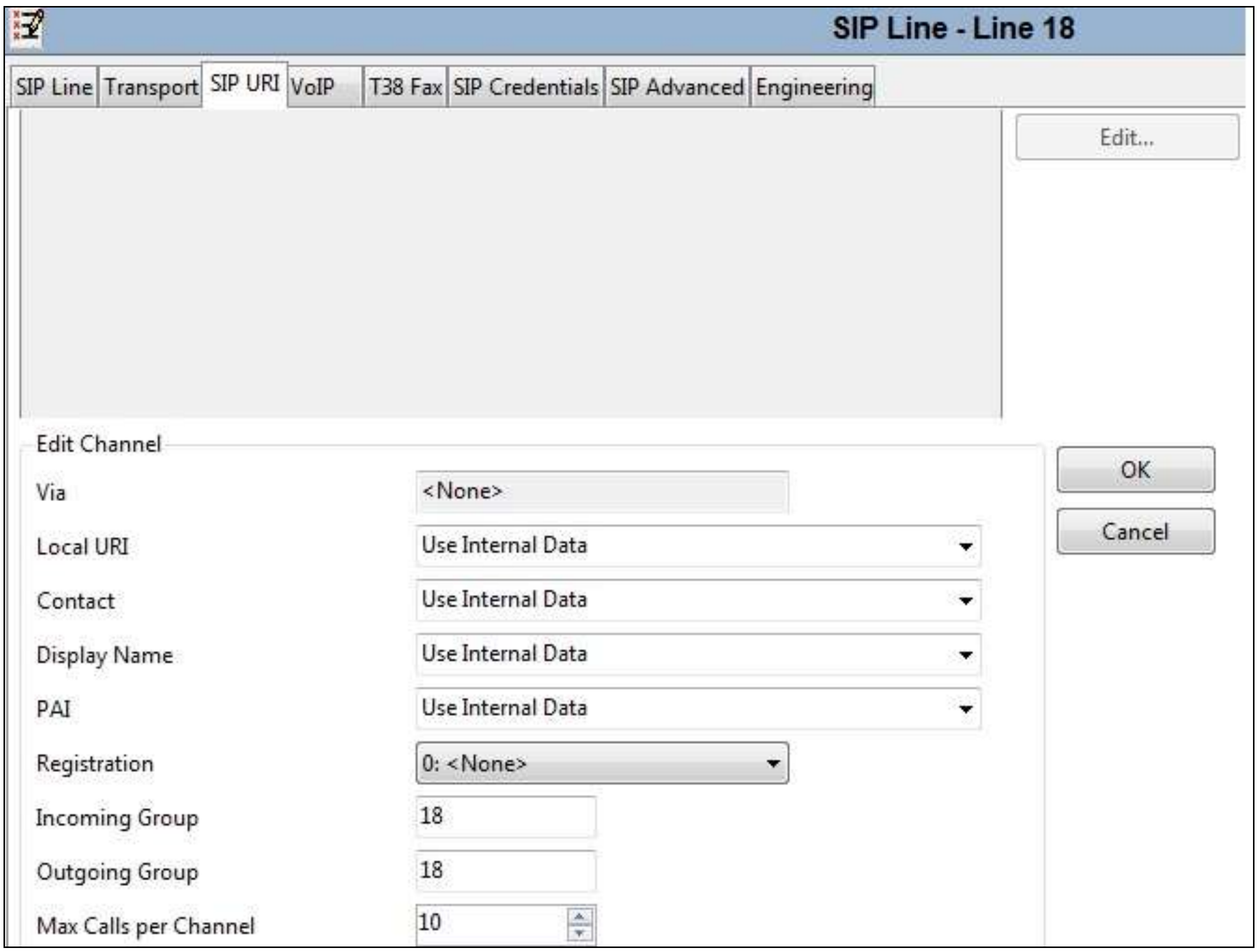


After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



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

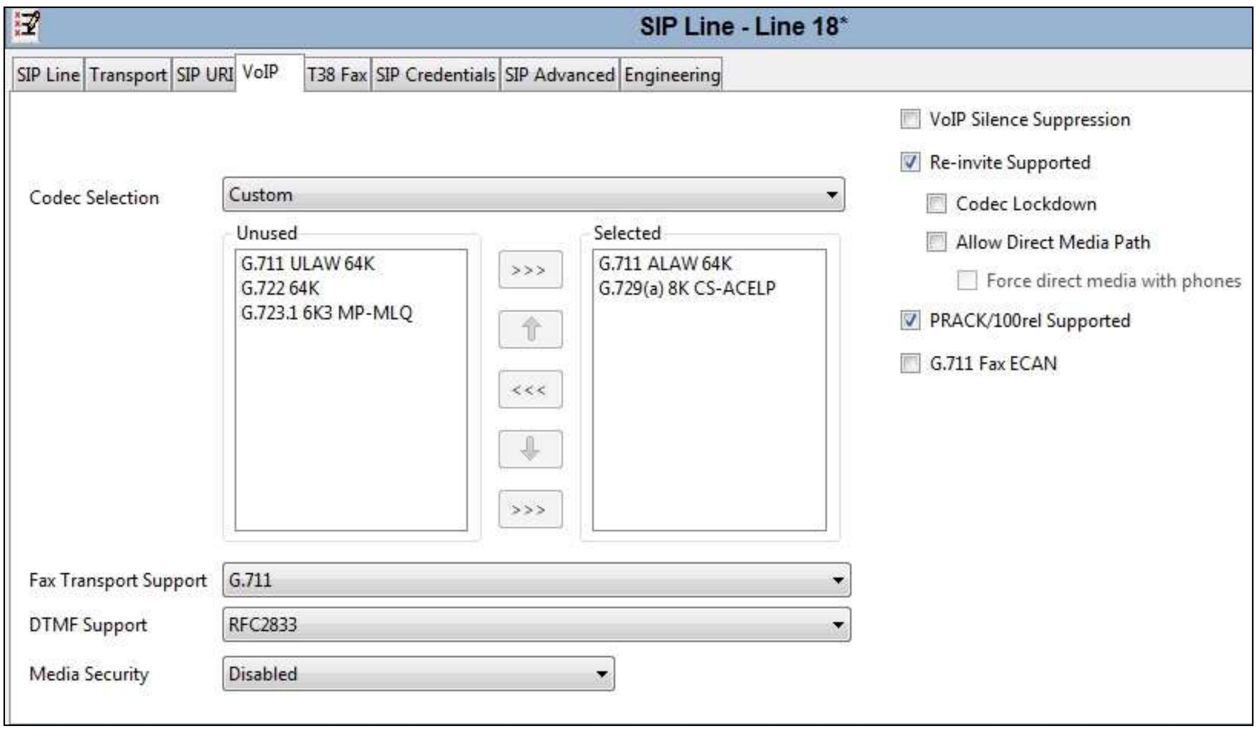
- Set **Local URI, Contact, Display Name and PAI** to **Use Internal Data**. This setting allows calls on this line who's SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.8**.
- For **Registration**, select **0: <None>** from the pull-down menu.
- Associate this 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.



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

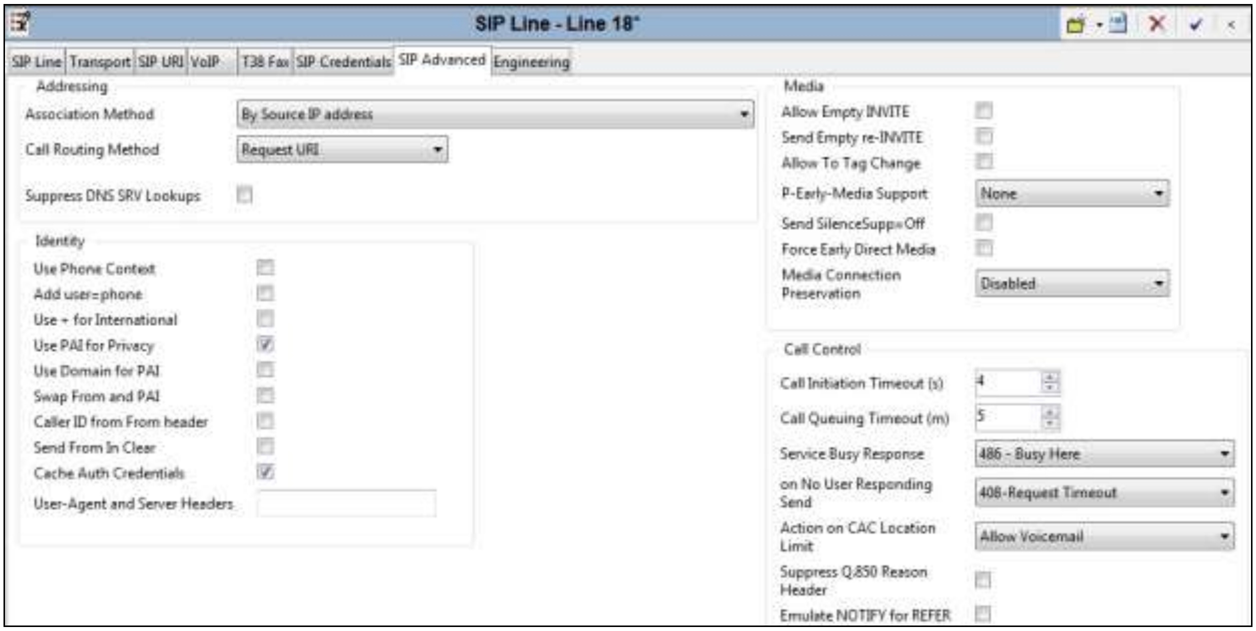
- Select **Custom** from the drop-down menu.
- Select **G.711 ALAW 64K** and **G.729(a) 8K CS-ACELP** codecs.
- Select the **Fax Transport Support** box to **G.711**.
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined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

Default values may be used for all other parameters.



Select the **SIP Advanced** tab. For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “anonymous”. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, Avaya IP Office will use the PPI header for privacy. For the compliance test, PAI was used for the purposes of privacy.

To configure Avaya IP Office to use the PAI header for privacy calls, on the **SIP Advanced** tab, check **Use PAI for Privacy**. All other fields retained their default values.



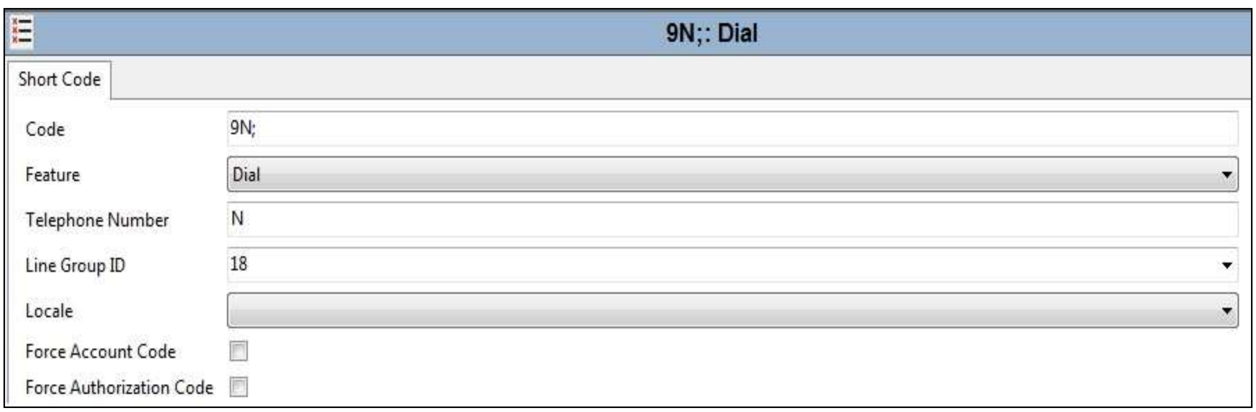
Note: It is advisable at this stage to save the configuration as described in **Section 5.10**.

5.7. Short Codes

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

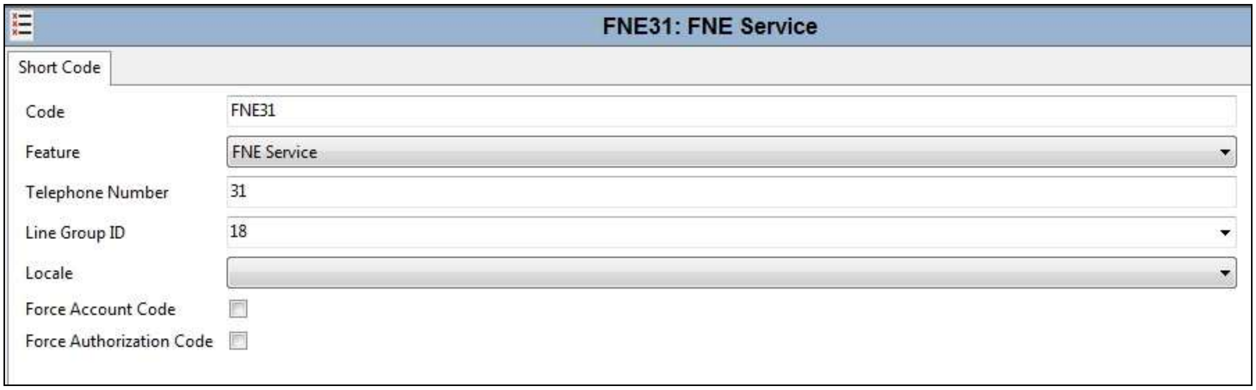
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon. The example shows **9N**; which will be invoked when the user dials 9 followed by the dialed number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N** which will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter N. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6.2**.

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



For incoming calls from mobility extension to FNE features hosted by IP Office to provide dial tone or mobile callback functionalities, Short Code **FNE31** was created.

- In the **Code** field, enter the FNE feature code as **FNE31**.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **31** for **FNE31**.
- Set the **Line Group Id** to **18** which is the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6.2**.

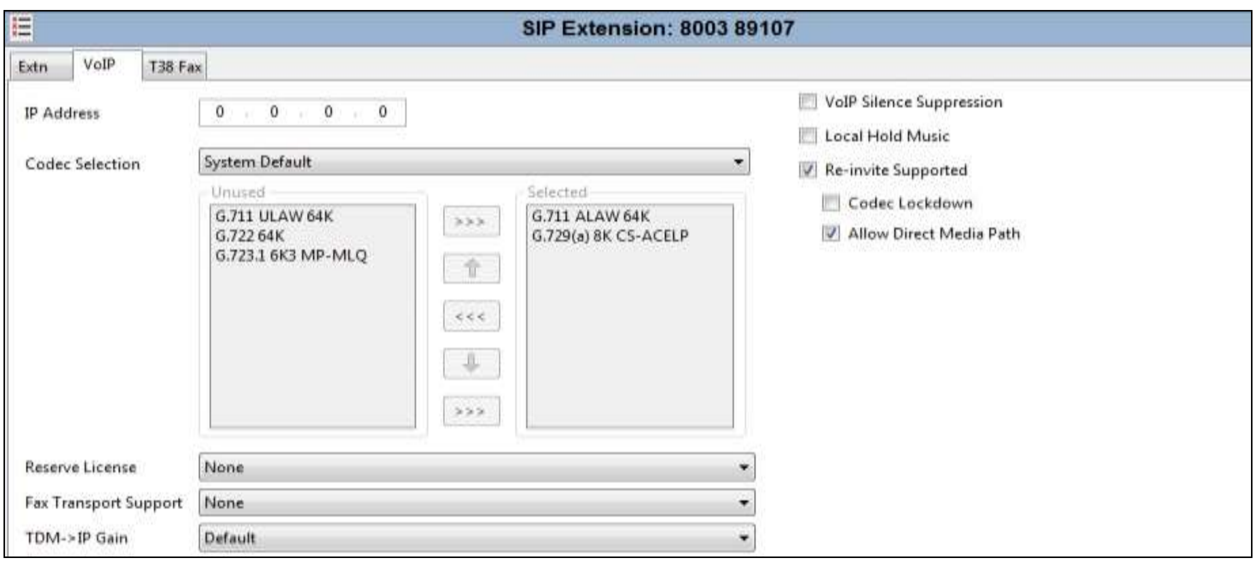


5.8. Users and Extensions

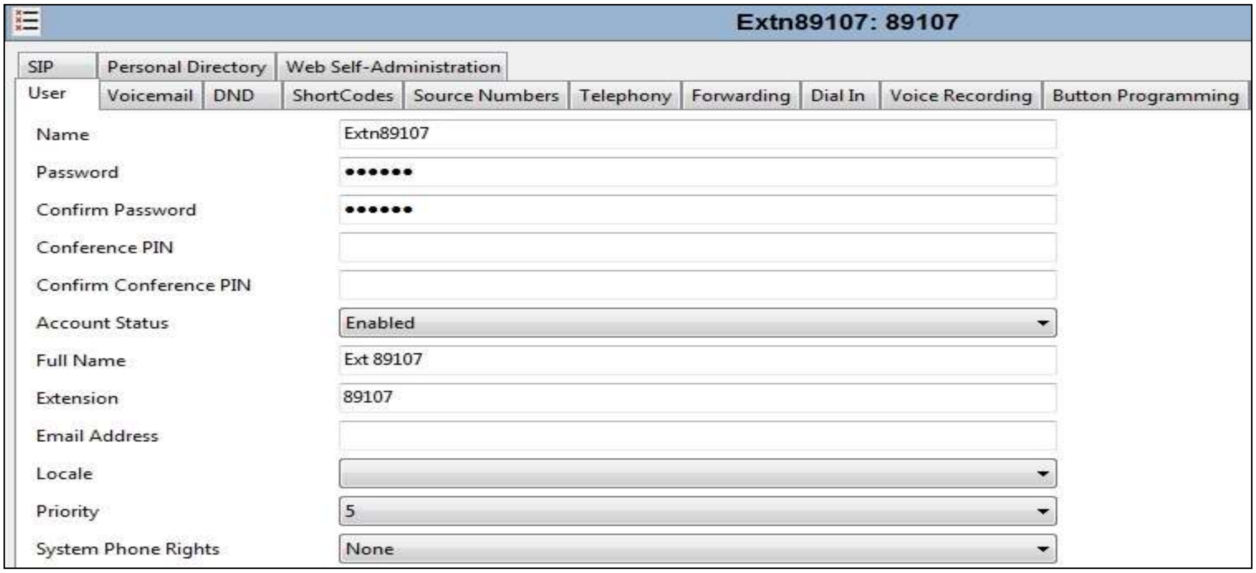
In this section, examples of IP Office Users and Extensions will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users.

A new SIP extension may be added by right-clicking on **Extension** (not shown) in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with **89107**, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.

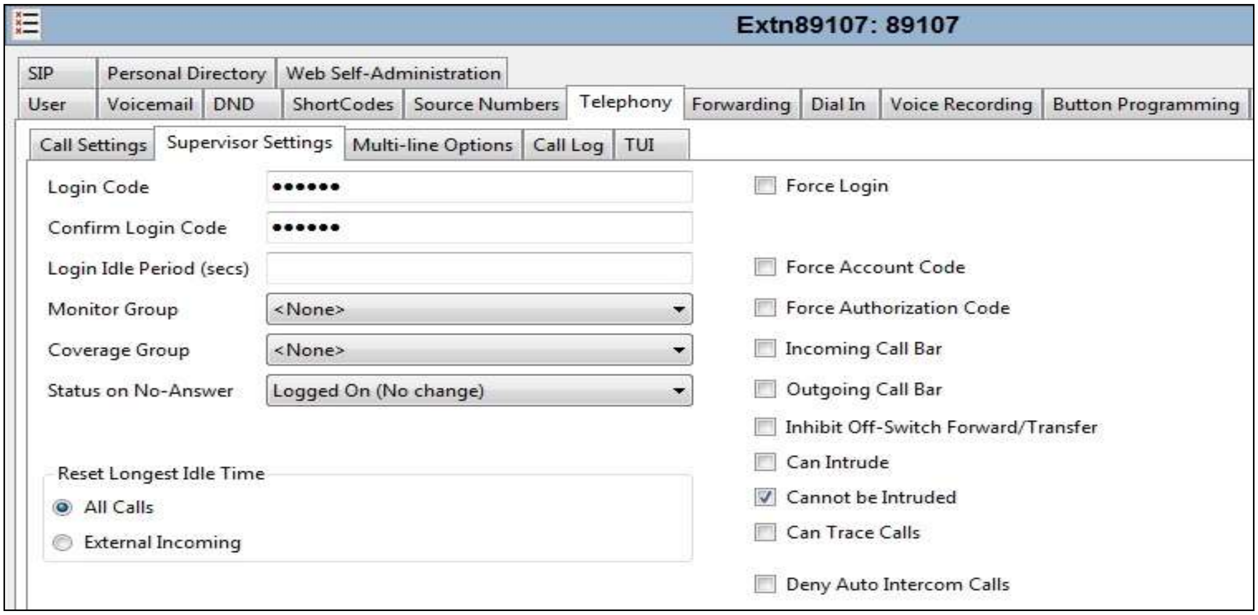
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank or populated with a static IP address. The new **Codec Selection** parameter may retain the default setting **System Default** to follow the system configuration shown in **Section 5.5**. Alternatively,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



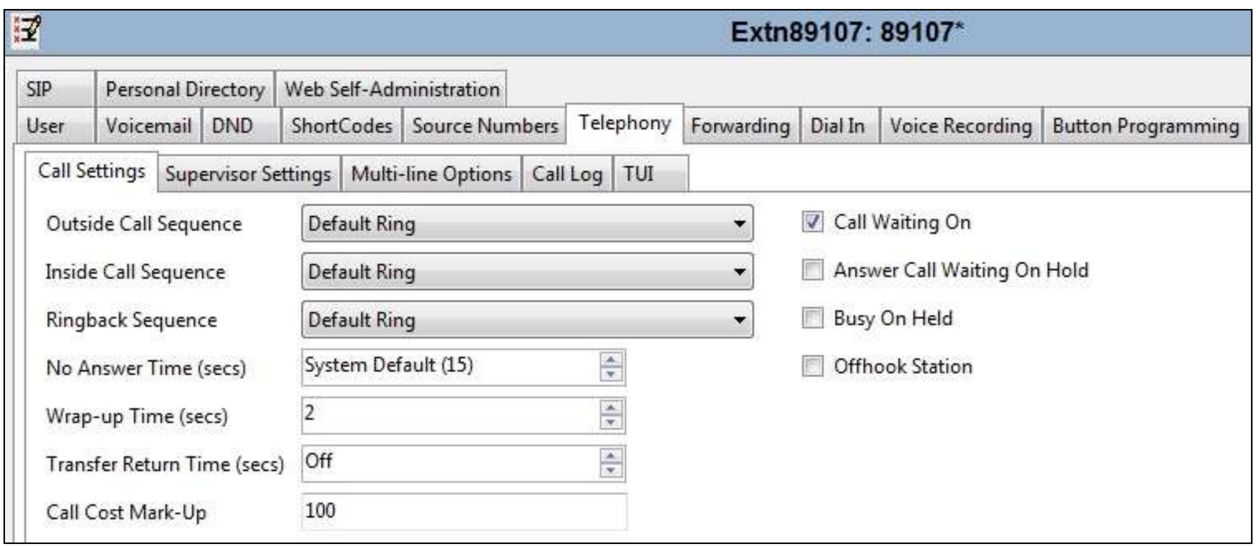
To add a User, right click on **User** in the Navigation pane, and select **New.** To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane. Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.6.2**. To configure these settings, select the **User** tab if any changes are required. The example below shows the changes required to use Avaya 1140E which was used in test.



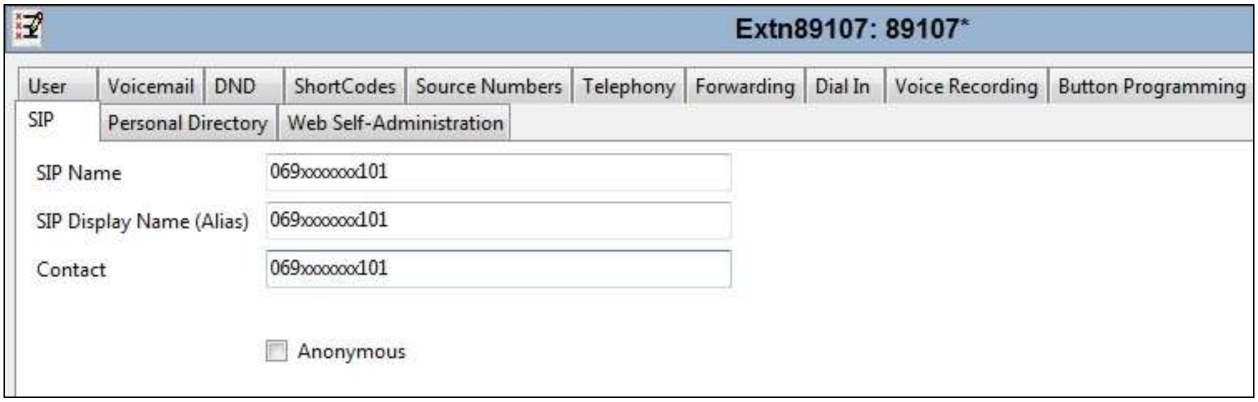
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



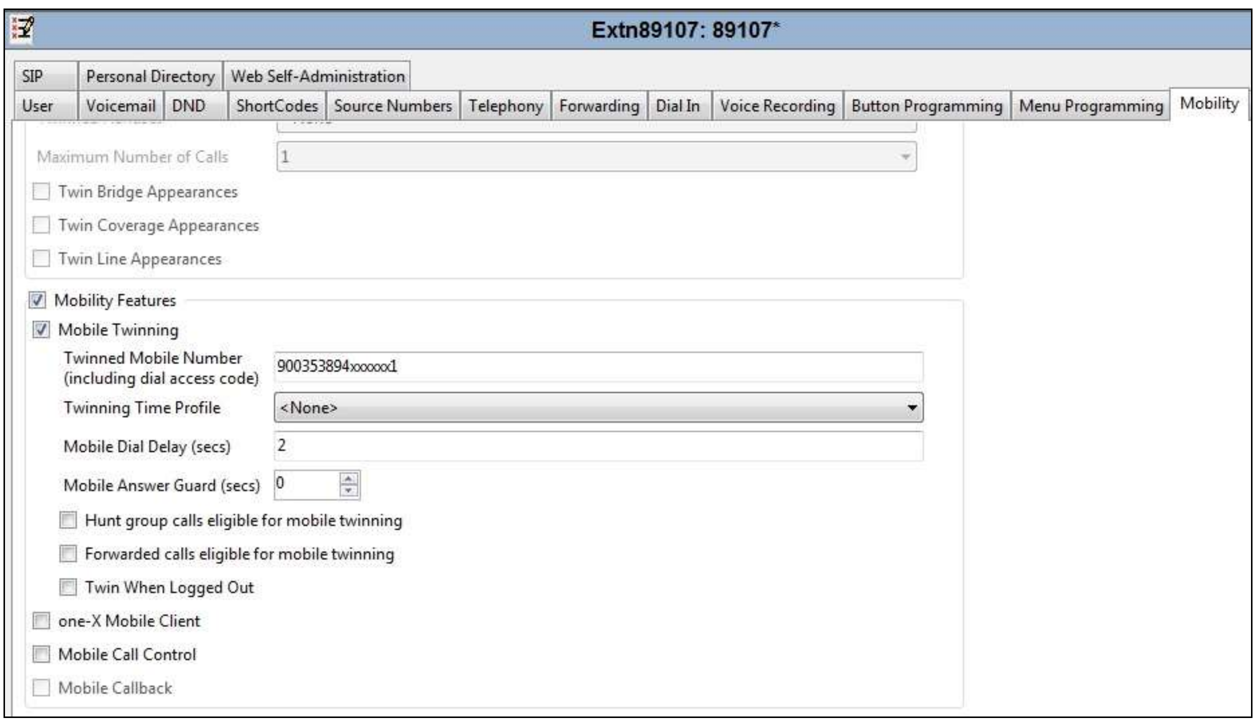
Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.



Next select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details Pane until it becomes visible. The values entered for the SIP **Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6.2**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Vodafone Germany.



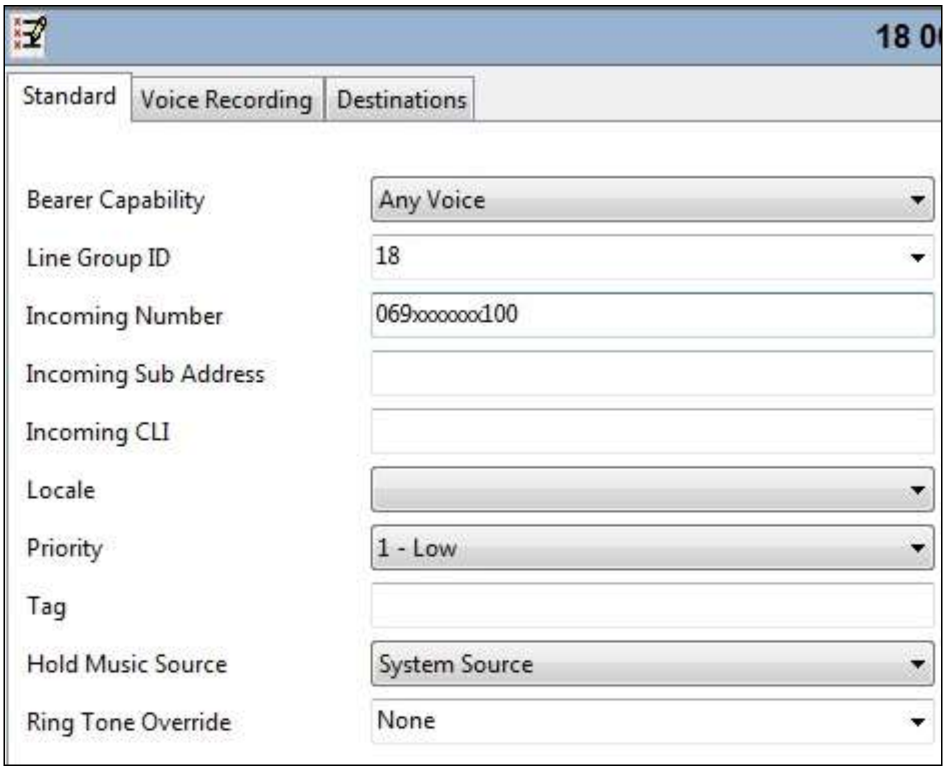
The following screen shows the Mobility tab for user 89107. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.



5.9. Incoming Call Routing

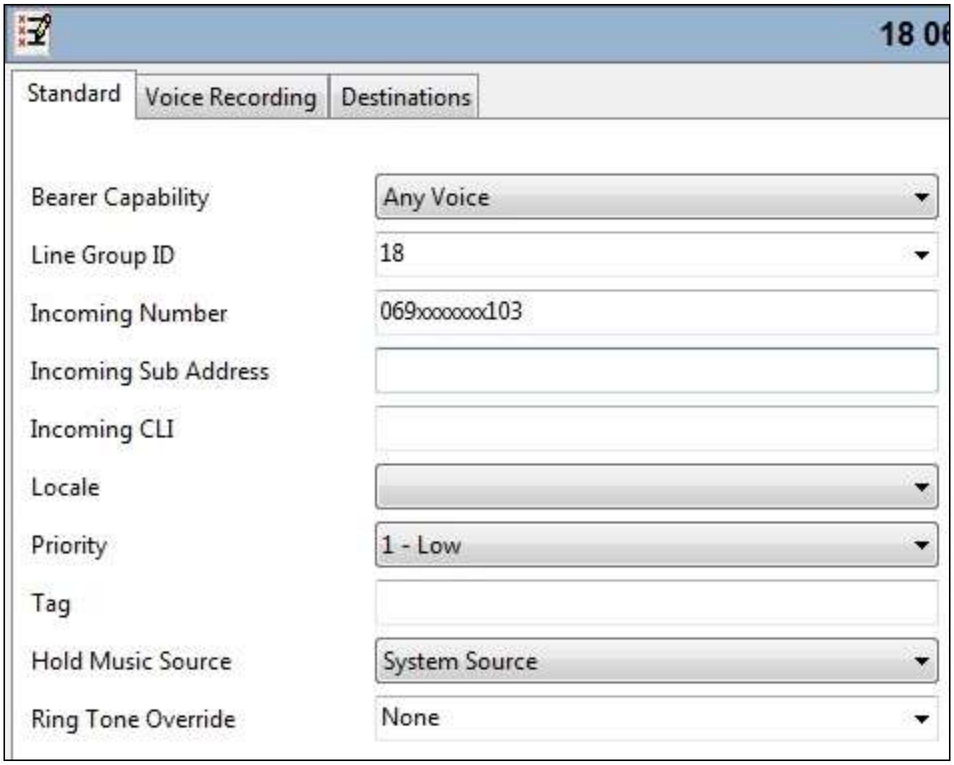
An incoming call route maps an inbound DDI number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Route**s in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.2**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.

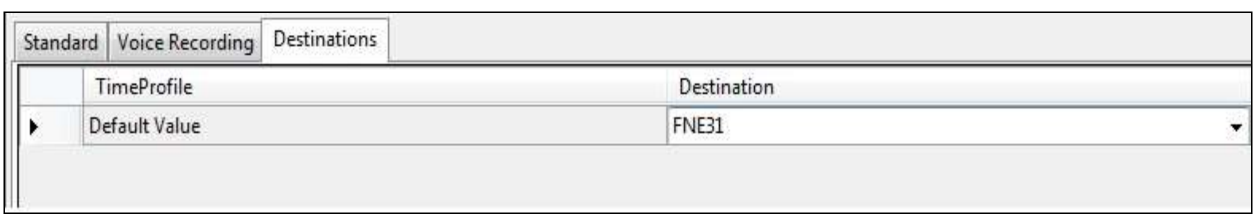


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **069xxxxxx100** on line 18 are routed to extension 89107.

Incoming Call Routes for other direct mappings of DDI numbers to IP Office users or FNE short codes etc. can be configured in the same fashion. In the screenshot below, the incoming call route for **069xxxxxx103** mapped to a shortcode **FNE** is illustrated.



The **Destination** tab contains the Destination field **FNE31** which was entered manually. An incoming call to **069xxxxxx103** will be delivered directly to internal dial tone allowing the caller to perform dialing actions to both internally and external calls. The incoming caller ID must match the Twinned Mobile Number entered in User Mobility tab (**Section 5.8**) otherwise IP Office responds with a 486 Busy Here and busy tone.



5.10. 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.

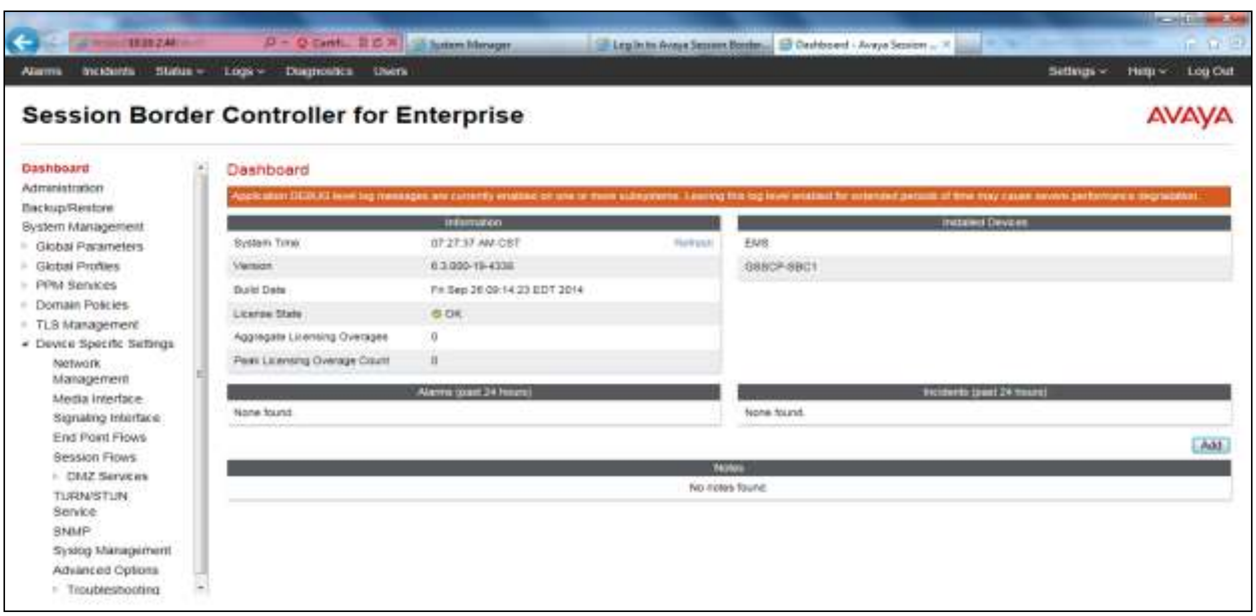
6. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary.

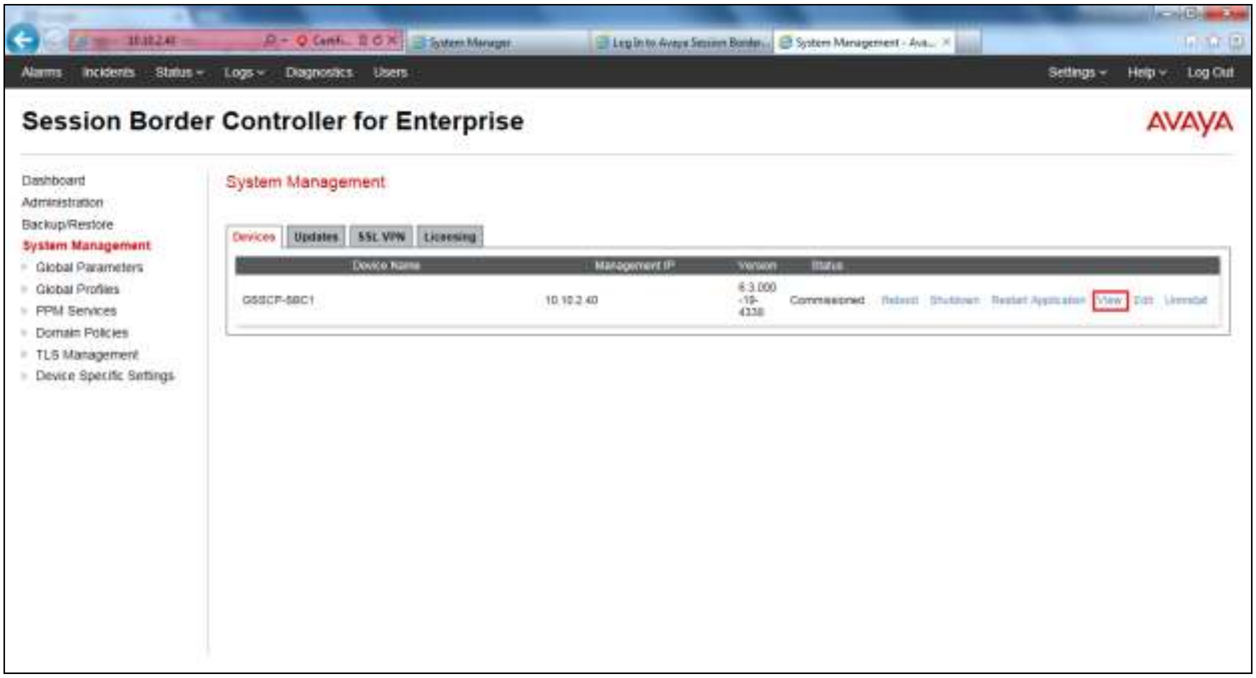
6.1. Accessing Avaya Session Border Controller for Enterprise

Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.

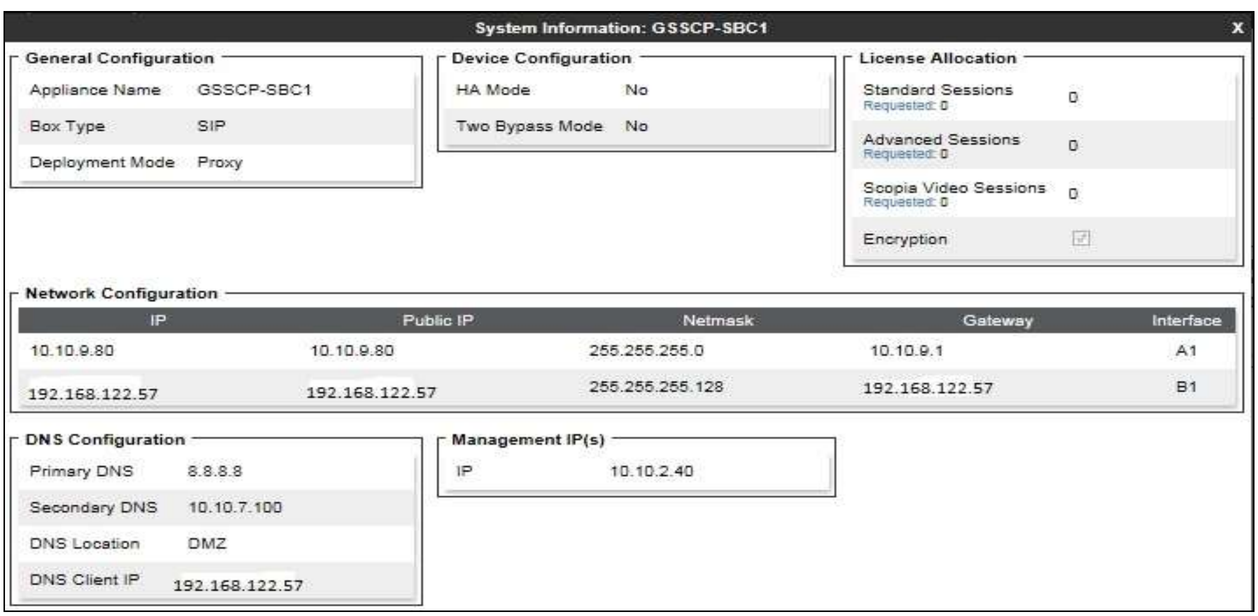
Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.



To view system information that was configured during installation, navigate to **System Management**. A list of installed devices is shown in the right pane. In the case of the test configuration, a single device named **GSSCP-SBC1** is shown. To view the configuration of this device, click **View** (the third option from the right).



The **System Information** screen shows the **General Configuration**, **Device Configuration**, **License Allocation**, **Network Configuration**, **DNS Configuration** and **Management IP** information.



6.2. Global Profiles

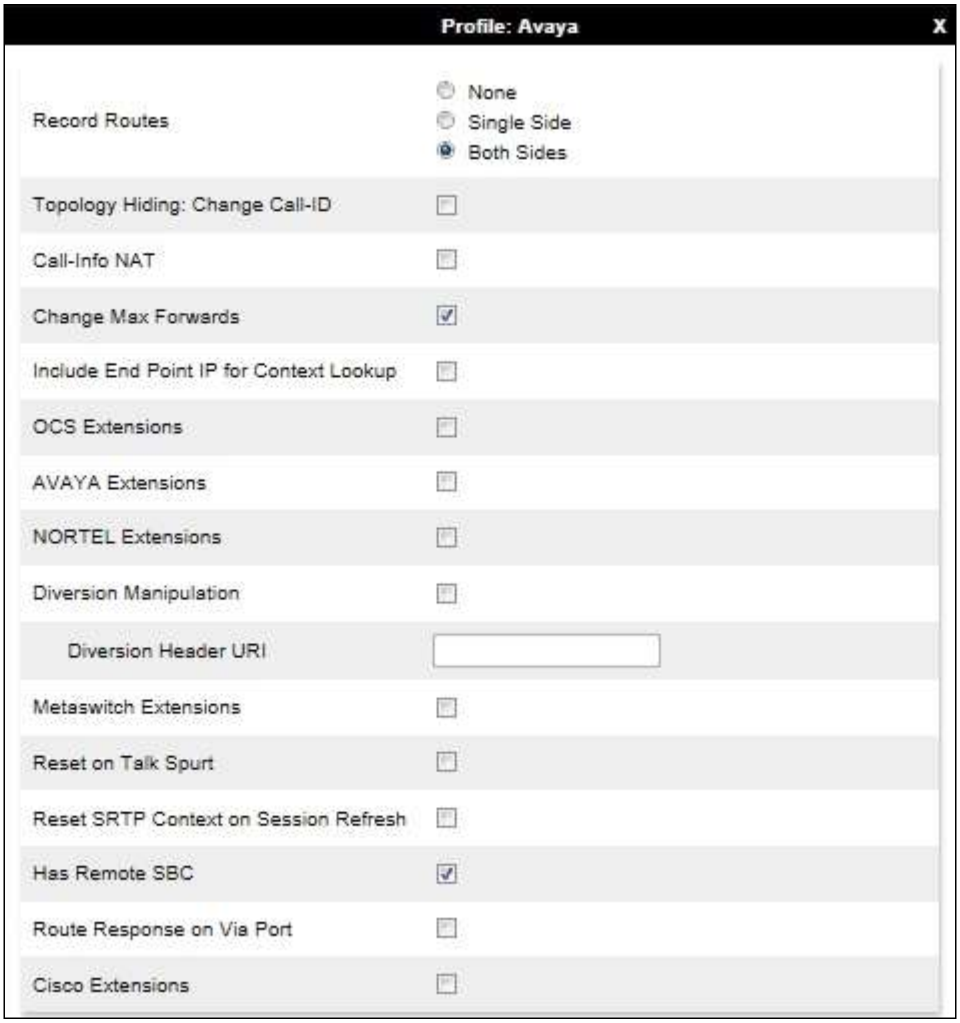
When selected, Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

6.2.1. Server Interworking Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles → Server Interworking** and click on **Add**.

- Enter profile name such as **Avaya** and click **Next** (Not Shown)
- Check **Hold Support=None**
- Check **T.38 Support**
- All other options on the **General** Tab can be left at default

Default values can be used for the **Advanced Settings** window. Click **Finish**

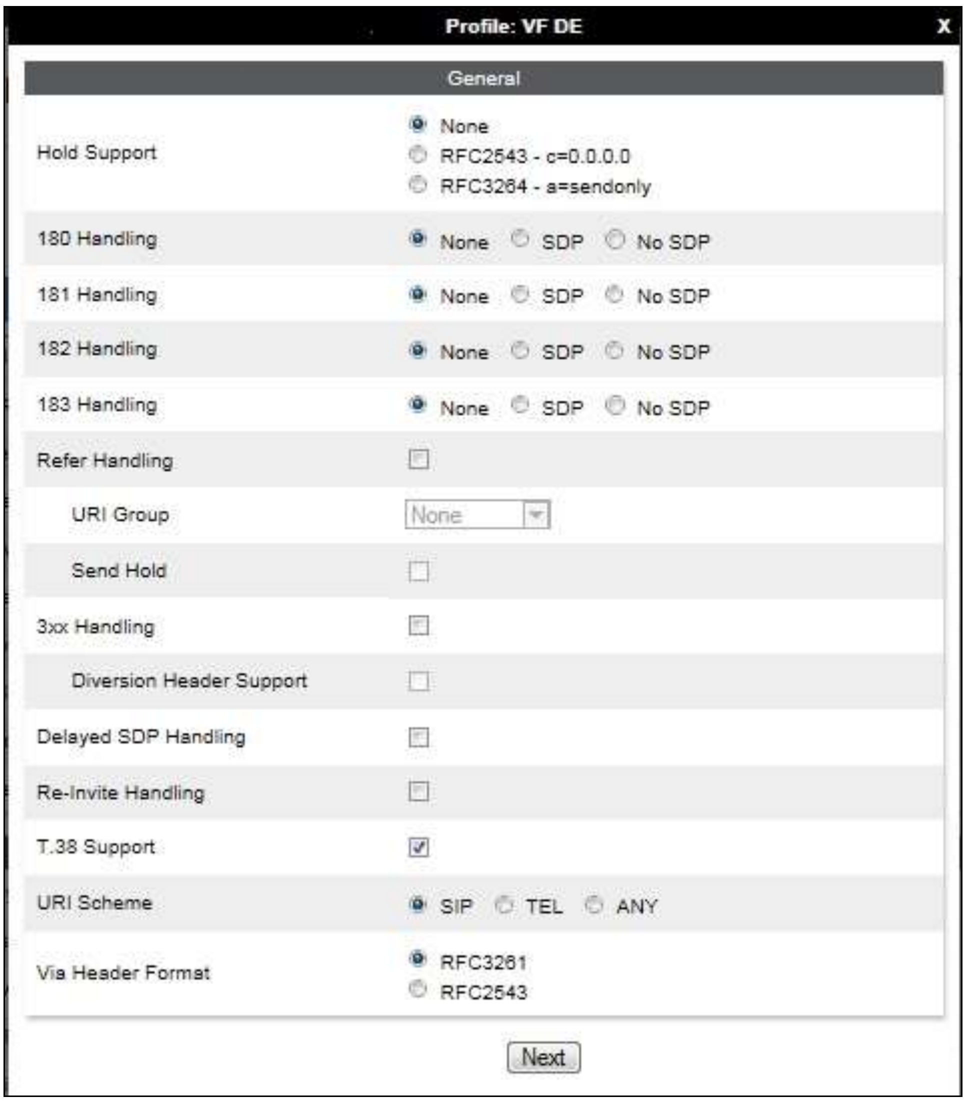


6.2.2. Server Interworking – Vodafone Germany

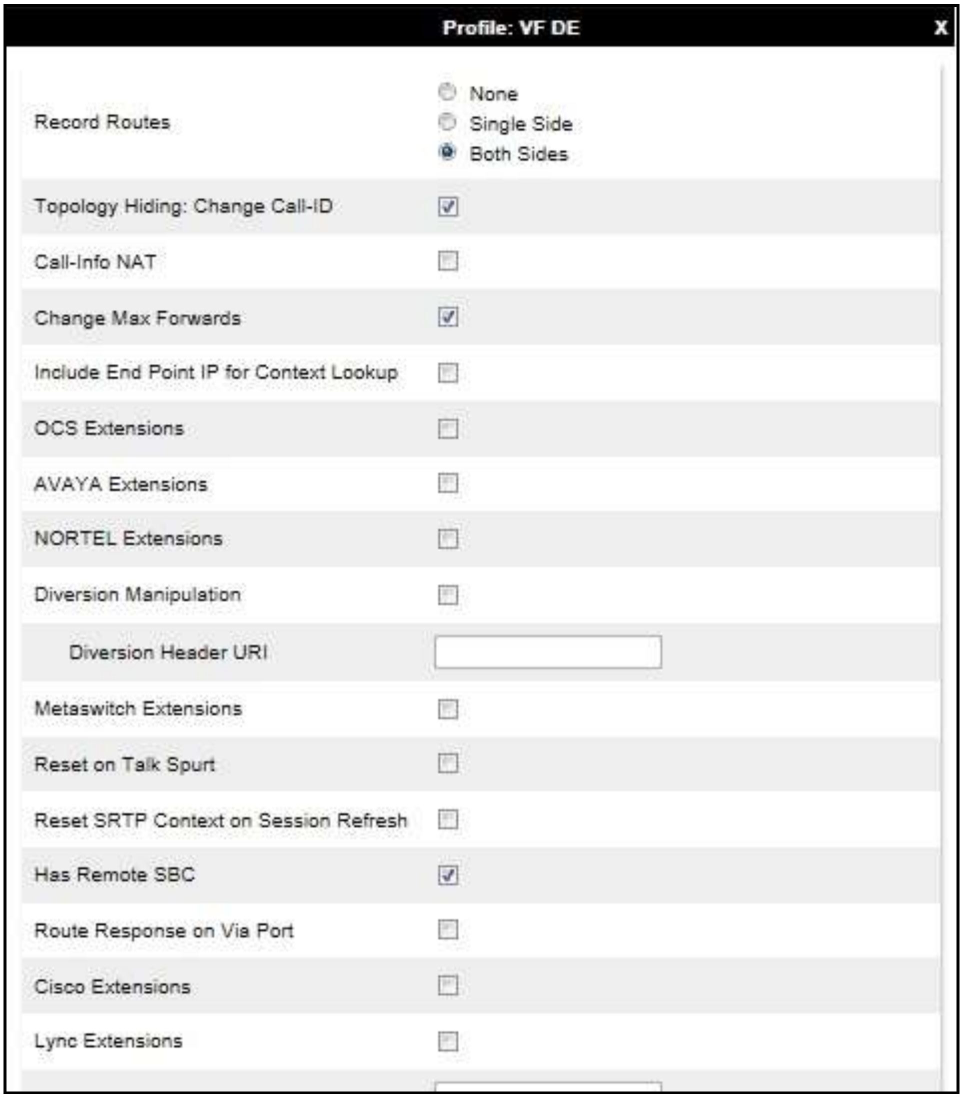
Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Global Profiles → Server Interworking** and click on **Add**.

- Enter profile name such as **VF DE** and click **Next** (Not Shown)
- Check **Hold Support = None**
- Check **T.38 Support**
- All other options on the **General** Tab can be left at default

Click on **Next** on the following screens and then **Finish**.



Default values can be used for the **Advanced Settings** window. Click **Finish**.



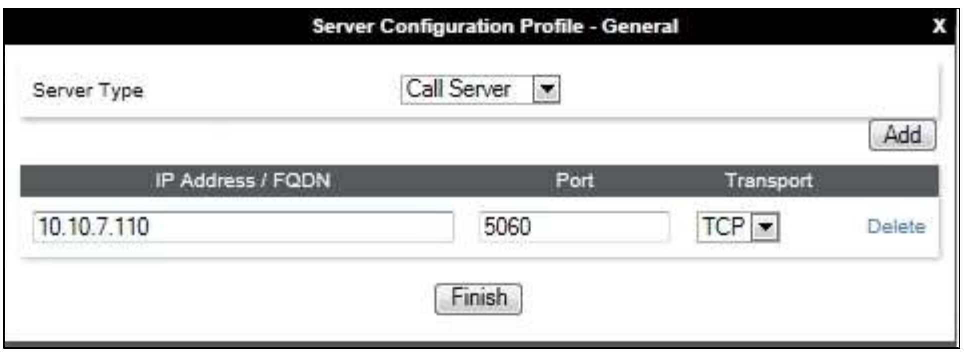
6.2.3. Server Configuration– Avaya IP Office

Servers are defined for each server connected to the Avaya SBCE. In this case, Vodafone Germany is connected as the Trunk Server and IP Office is connected as the Call Server.

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow the configuration and management of various SIP call server-specific parameters such as TCP and UDP port assignments, IP Server type, heartbeat signaling parameters and some advanced options.

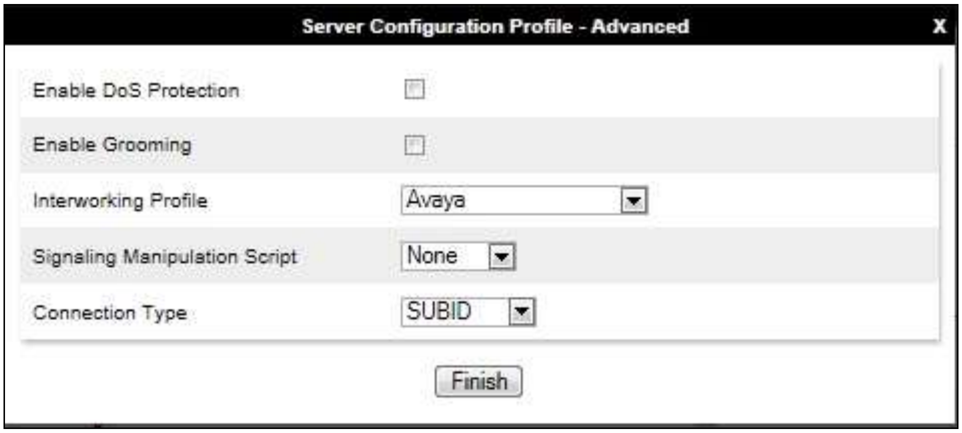
From the left-hand menu select **Global Profiles → Server Configuration** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profile** (not shown) tab, set the following:

- Select **Server Type** to be **Call Server**.
- Enter **IP Address / FQDN** to **10.10.7.110** (IP Office LAN1 IP Address).
- For **Port**, enter **5060**.
- For **Transport,** select **TCP**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



On the **Advanced** tab:

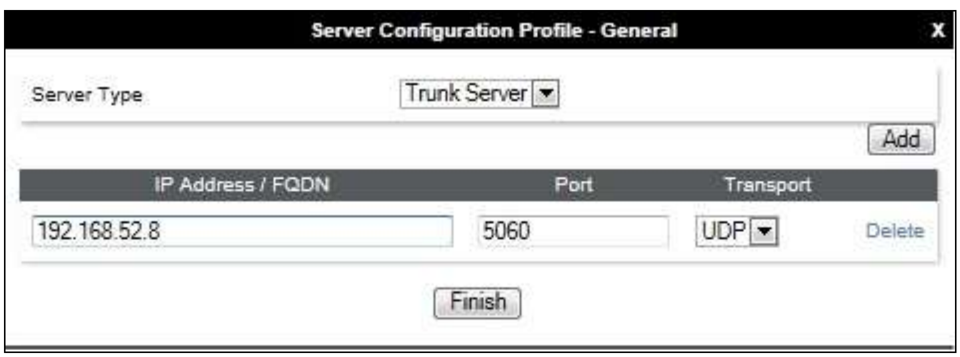
- Select **Avaya** for **Interworking Profile**.
- Click **Finish**.



6.2.4. Server Configuration – Vodafone Germany

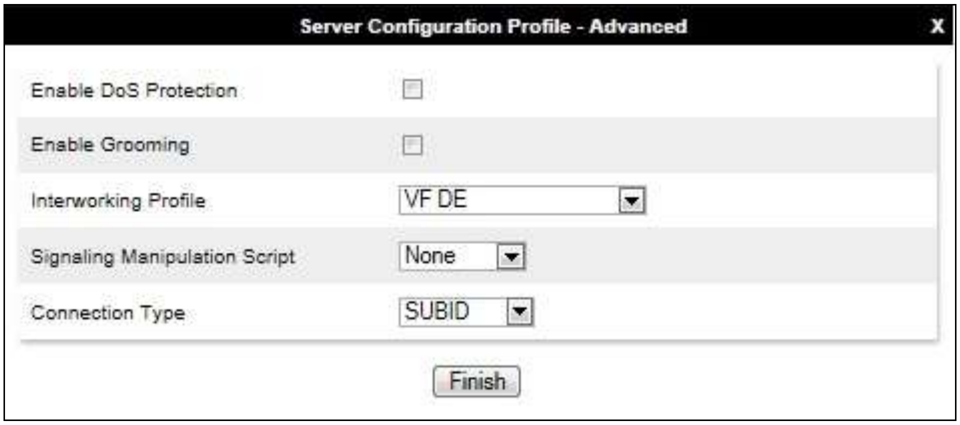
To define the Vodafone Germany SBC as a Trunk Server, navigate to **Global Profiles → Server Configuration** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Trunk Server**.
- Enter **IP Address / FQDN** to **192.168.52.8** (Vodafone Germany SBC IP Address).
- For **Port**, enter **5060**.
- For **Transport,** select **UDP**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



On the Advanced tab:

- Select **VF DE** for Interworking Profile.
- Click **Finish**.


Server Configuration Profile - Advanced
X
Enable DoS Protection
Enable Grooming
Interworking Profile
VF DE
Signaling Manipulation Script
None
Connection Type
SUBID
Finish

6.2.5. Routing

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

Routing information is required for routing to IP Office on the internal side and Vodafone Germany addresses on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used.

6.2.5.1 Routing – Avaya

Create a Routing Profile for IP Office.

- Navigate to **Global Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

The Routing Profile window will open. Use the default values displayed and click **Add**.

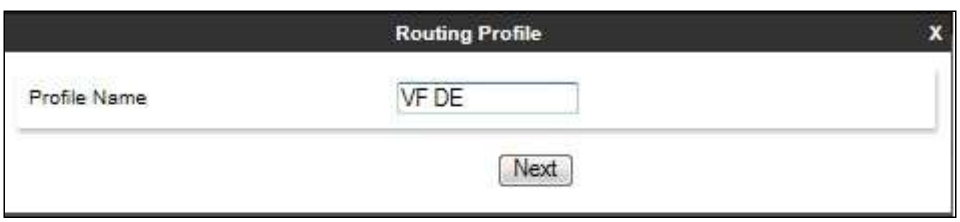
On the **Next Hop Address** window, set the following:

- **Priority/Weight** = **1**.
- **Server Configuration** = **Avaya** (**Section 6.2.3**) from drop down menu.
- **Next Hop Address** = Select **10.10.7.110:5060 TCP** from drop down menu.
- Click **Finish**.

6.2.5.2 Routing – Vodafone Germany

Create a Routing Profile for Vodafone Germany.

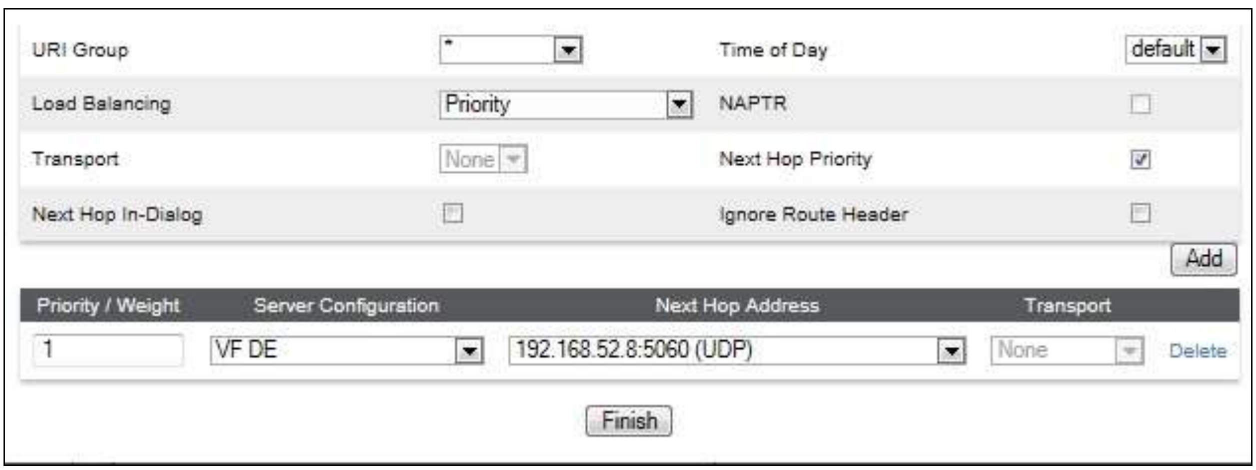
- Navigate to **Global Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.



The Routing Profile window will open. Use the default values displayed and click **Add**.

On the **Next Hop Address** window, set the following:

- **Priority/Weight** = **1**.
- **Server Configuration** = **VF DE** (**Section 6.2.4**) from drop down menu.
- **Next Hop Address** = Select **192.168.52.8:5060 UDP** from drop down menu.
- Click **Finish**.

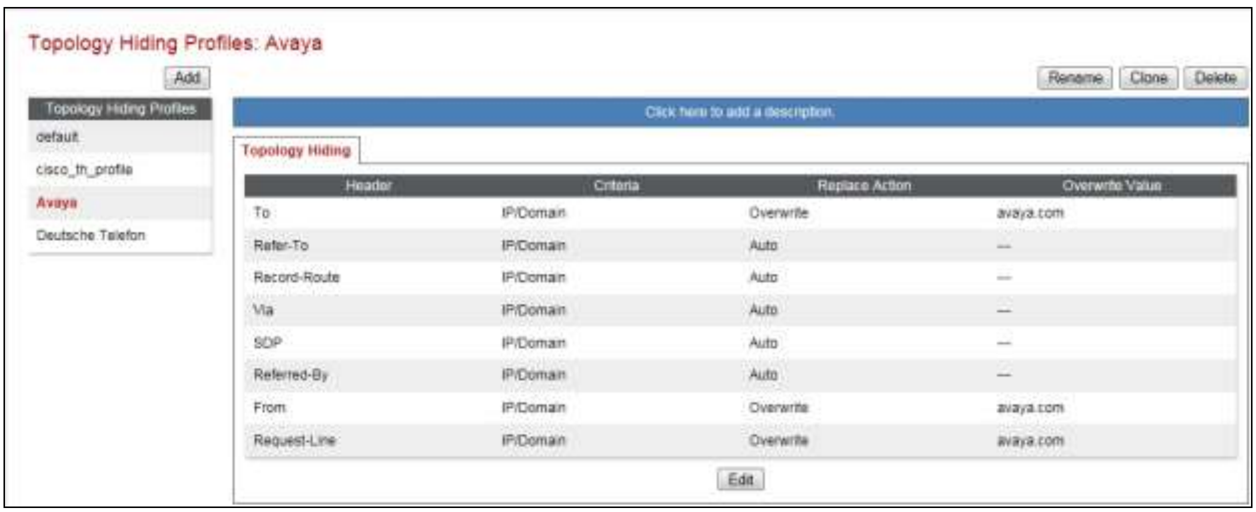


6.2.6. Topology Hiding

Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

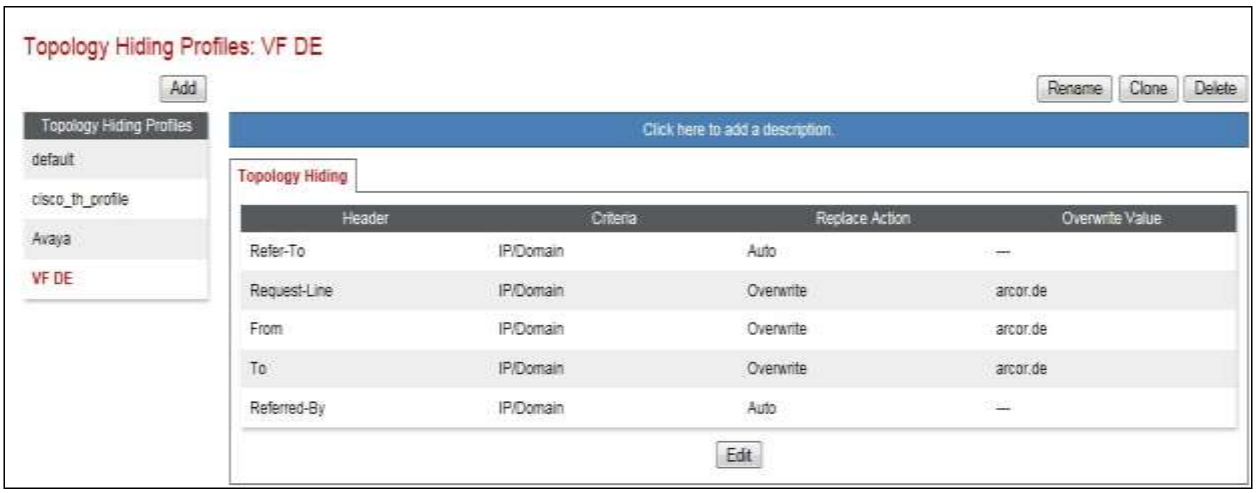
To define Topology Hiding for IP Office, navigate to **Global Profiles → Topology Hiding** from menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya**
- If the required Header is not shown, click on **Add Header**
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Overwrite Value**, insert **avaya.com**. Note: **avaya.com** is the domain used by IP Office in **Section 5.2**.
- Click **Finish** (not shown)



To define Topology Hiding for Vodafone Germany, navigate to **Global Profiles → Topology Hiding** from the menu on the left hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for Vodafone Germany and click **Next**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For **Overwrite Value**, insert **arcor.de**. Note: **arcor.de** is the FQDN provided by Vodafone Germany to be used in the test configuration.
- Click **Finish** (not shown).



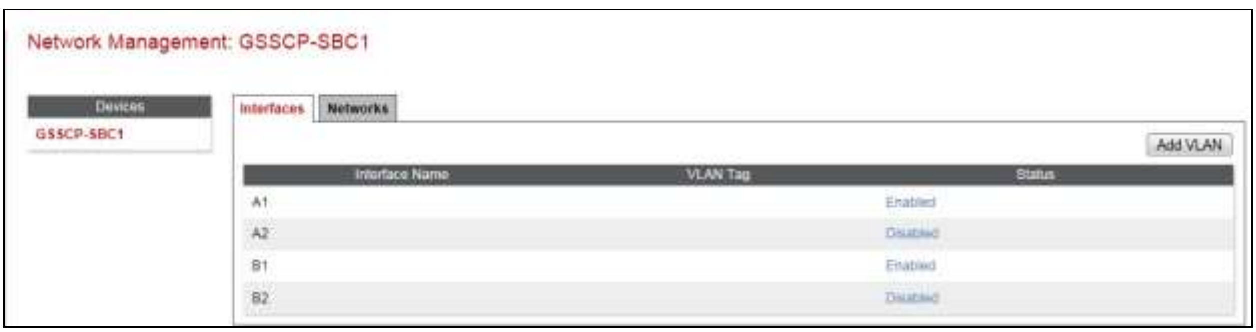
6.3. Define Network Information

Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one interface assigned.

To define the network information, navigate to **Device Specific Settings → Network Management** from the menu on the left-hand side and click on **Add IP** (not shown). Enter details in the blank box that appears at the end of the list

- Define the internal IP address with screening mask and assign to interface **A1**
- Select **Save** to save the information
- Click on **Add IP**
- Define the external IP address with screening mask and assign to interface **B1**
- Select **Save** to save the information
- Click on **System Management** in the main menu
- Select **Restart Application** indicated by an icon in the status bar (not shown)

Select the **Interface Configuration** Tab and use the **Toggle** button to enable the interfaces.



6.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

6.4.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** from the menu on the left hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

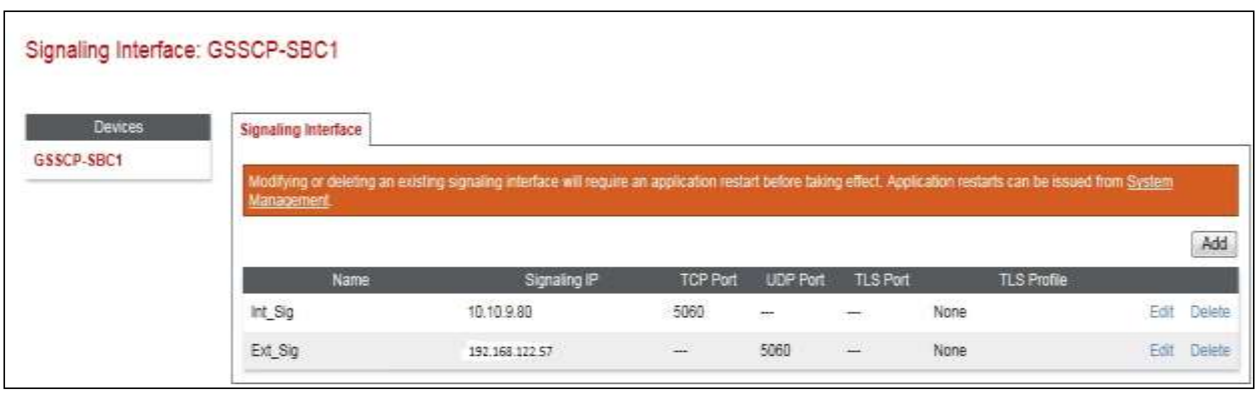
To enter details of transport protocol and ports for the SIP signalling on the internal interface:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the interface
- For **Signaling IP**, select the **internal** signalling interface IP addresses defined in **Section 6.3**
- Select **TCP** port number, **5060** is used for IP Office

To enter details of transport protocol and ports for the SIP signalling on the external interface:

- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown)
- In the **Name** field enter a descriptive name for the external signalling interface
- For **Signaling IP**, select the **external** signalling interface IP address defined in **Section 6.3**
- Select **UDP** port number, **5060** is used for the Vodafone Germany SIP Trunk

The following screen shows the Signalling Interfaces created in the test configuration for the inside and outside IP interfaces.



6.4.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** from the menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** from the menu on the left hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

To enter details of the media IP and RTP port range on the internal interface to be used in the server flow:

- Select **Add Media Interface** and enter details in the pop-up menu
- In the **Name** field enter a descriptive name for the internal media interface
- For **Media IP**, select the **internal** media interface IP address defined in **Section 6.3**
- Select **RTP port** ranges for the media path with the enterprise end-points

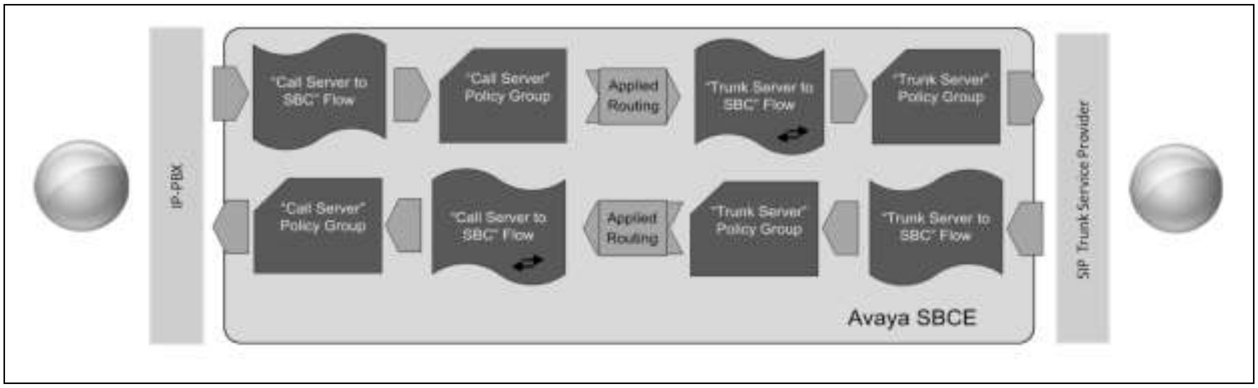
To enter details of the media IP and RTP port range on the external interface to be used in the server flow.

- Select **Add Media Interface** and enter details in the pop-up menu
- In the **Name** field enter a descriptive name for the external media interface
- For **Media IP**, select the **external** media interface IP address defined in **Section 6.3**
- Select **RTP port** ranges for the external media path

The following screen shows the Media Interfaces created in the test configuration for the inside and outside IP interfaces.

6.5. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from IP Office to Vodafone Germany's SIP Trunk and incoming flows from Vodafone Germany s SIP Trunk to IP Office. This configuration ties all the previously entered information together so that signalling can be routed from the IP Office to the PSTN via the Vodafone Germany network and vice versa. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

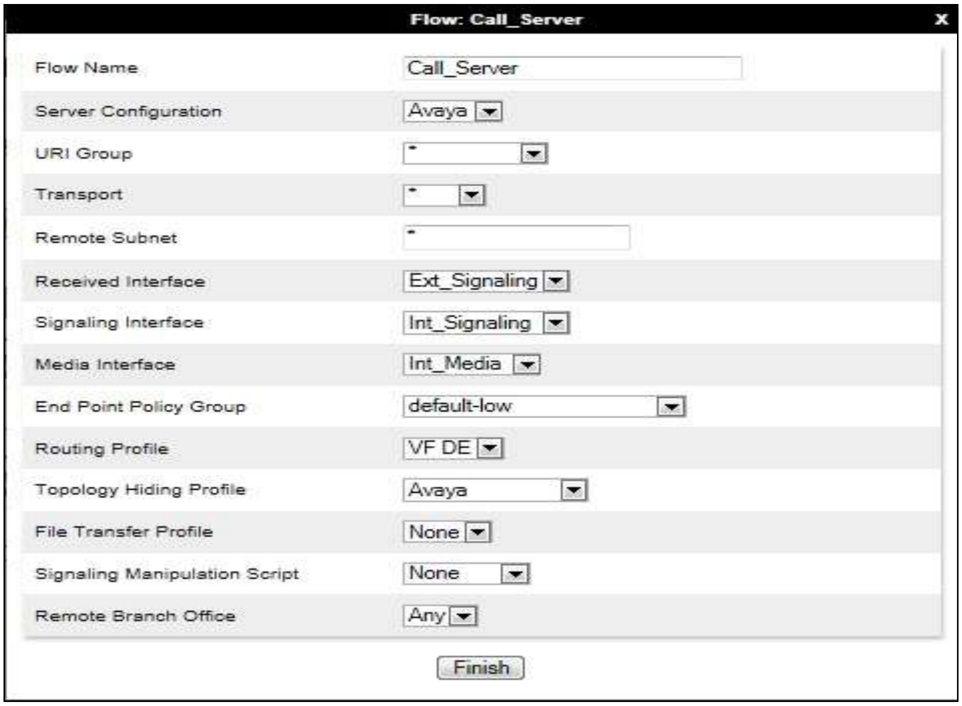


To create a Server Flow, navigate to **Device Specific Settings → End Point Flows**. Select the **Server Flows** tab and click **Add Flow**.

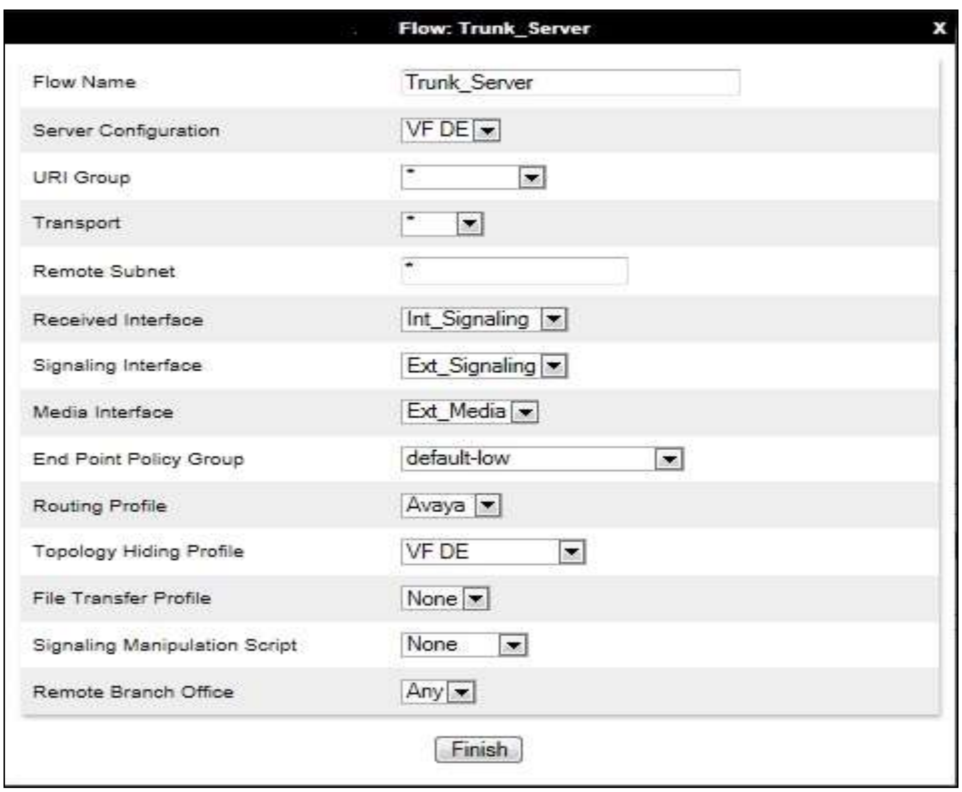
- **Flow Name:** Enter a descriptive name
- **Server Configuration:** Select a Server Configuration created in **Section 6.2.3** and **6.2.4** and assign to the Flow.
- **Received Interface:** Select the Signaling Interface the Server Configuration is allowed to receive SIP messages from.
- **Signaling Interface:** Select the Signaling Interface used to communicate with the Server Configuration.
- **Media Interface:** Select the Media Interface used to communicate with the Server Configuration.
- **End Point Policy Group:** Select the policy assigned to the Server Configuration.
- **Routing Profile:** Select the profile the Server Configuration will use to route SIP messages.
- **Topology Hiding Profile:** Select the profile to apply toward the Server Configuration.

Click **Finish** to save and exit.

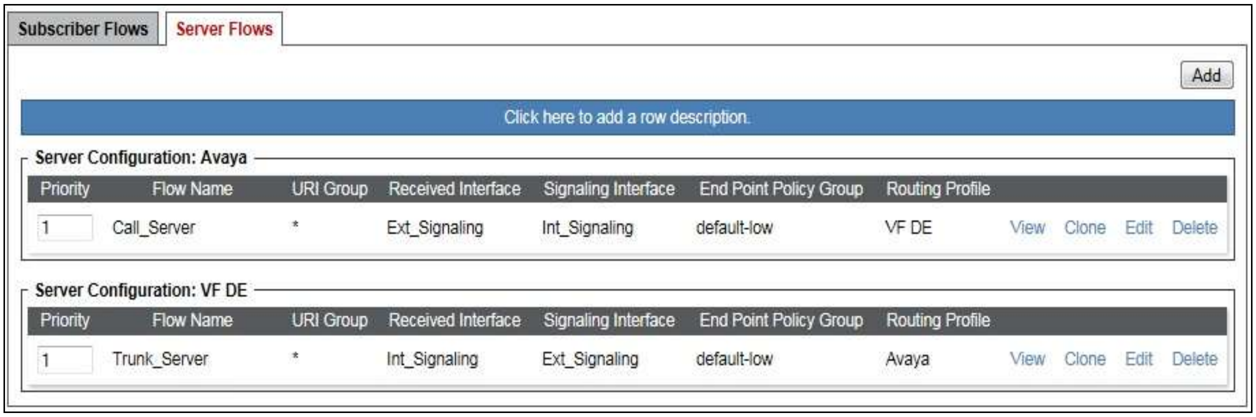
The following screen shows the Server Flow for IP Office.



The following screen shows the Server Flow for Vodafone Germany.



This configuration ties all the previously entered information together so that calls can be routed from IP Office to Vodafone Germany SIP Trunk service and vice versa. The following screenshot shows all configured flows.



7. Vodafone Germany SIP Trunk Service Configuration

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

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

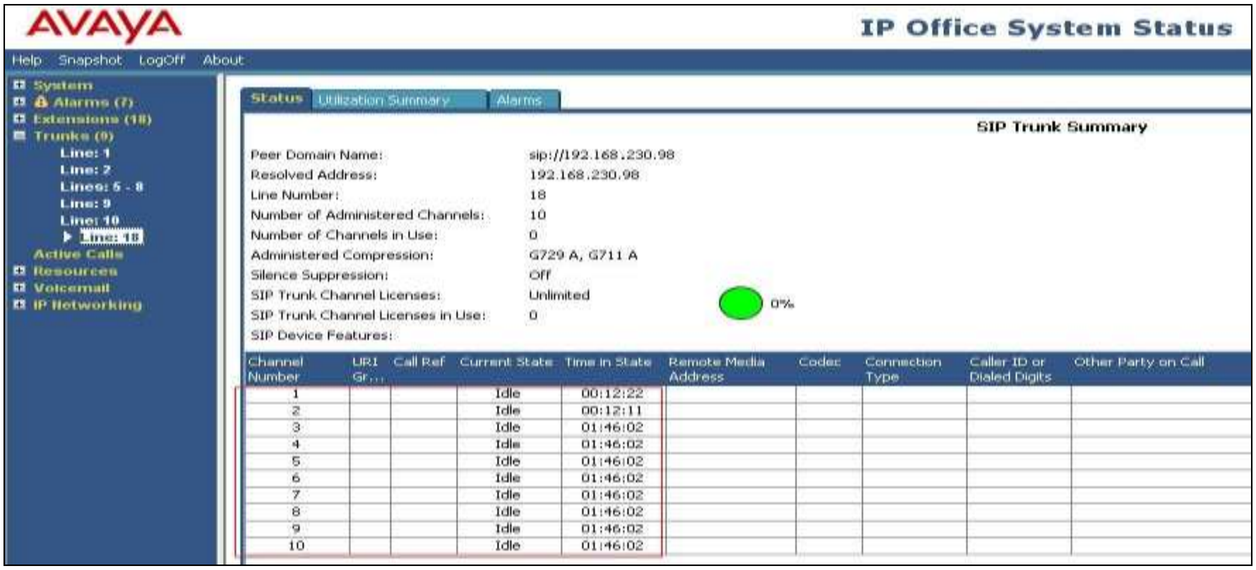
8. Verification Steps

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

8.1. SIP Trunk status

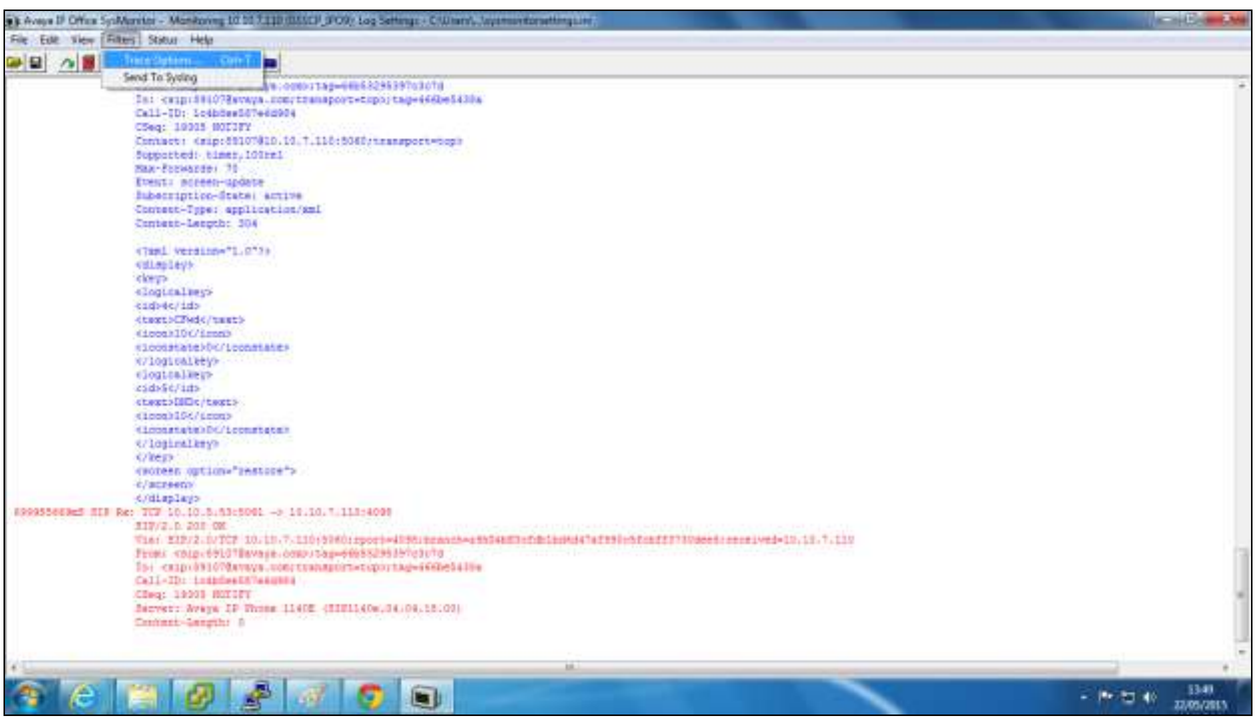
The status of the SIP trunk can be verified by opening the System Status application. This is found on the PC where IP Office Manager is installed in PC programs under **Start →All Programs →IP Office →System Status** (not shown).
Log in to IP Office System Status at the prompt using the **Control Unit 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.

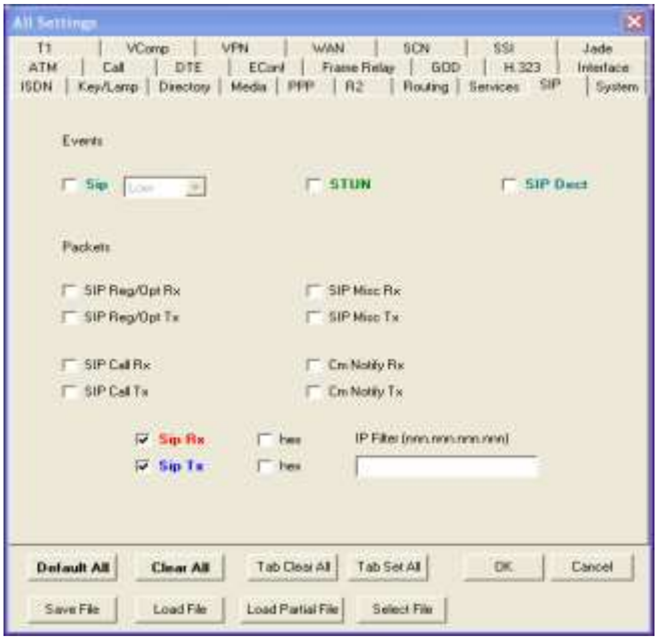


8.2. Monitor

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



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



As an example, the following shows a portion of the monitoring window of a SIP handset attempting registration to IP Office.

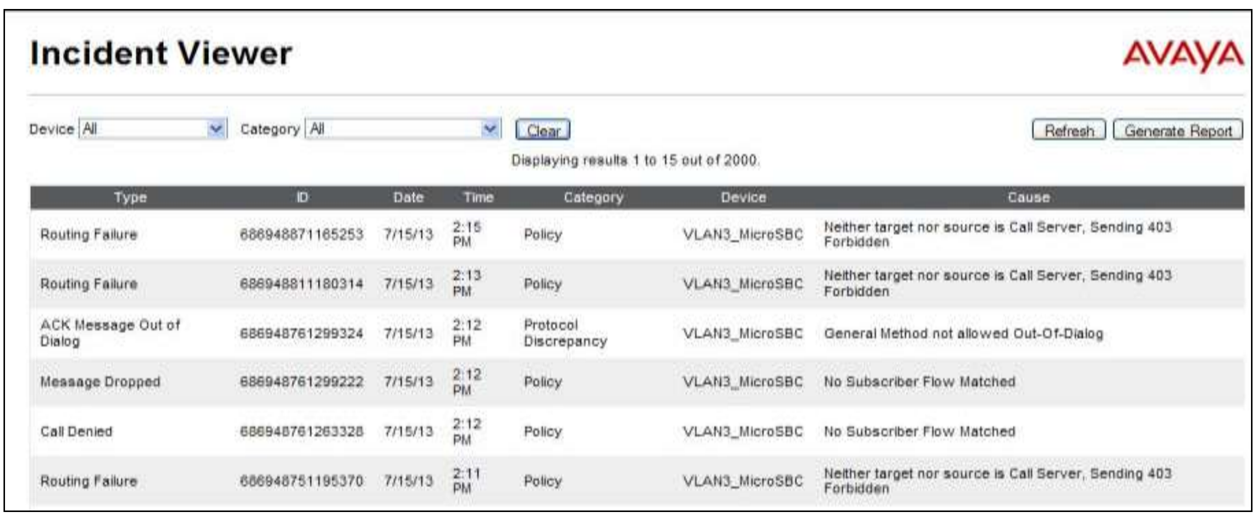
8.3. Avaya SBCE

This section provides verification steps that may be performed with the Avaya SBCE.

8.3.1. Incidents

The Incident Viewer can be accessed from the Avaya SBCE dashboard as highlighted in the screen shot below.

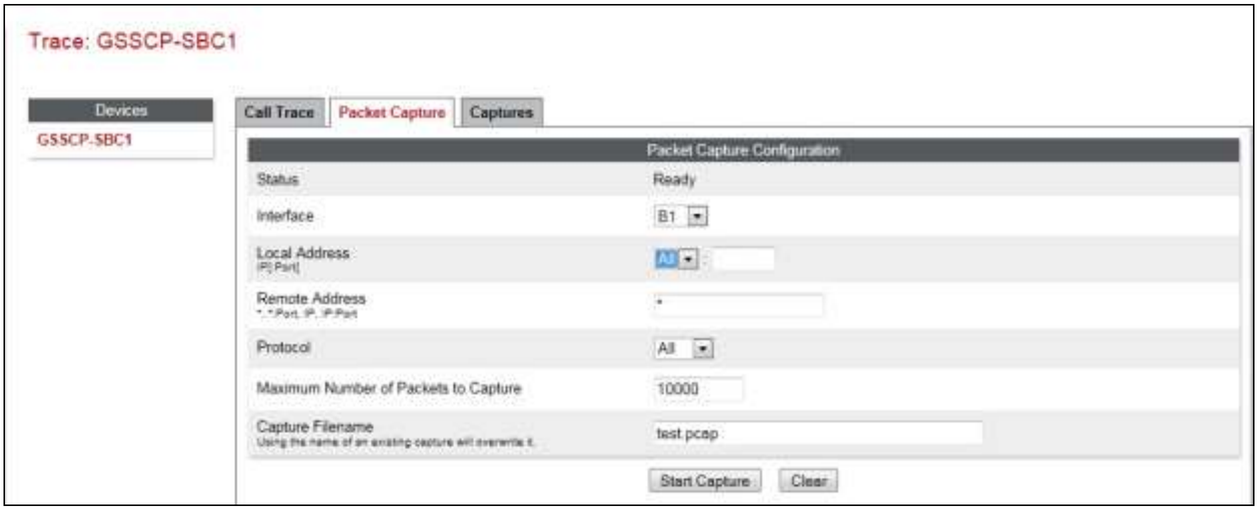
Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.



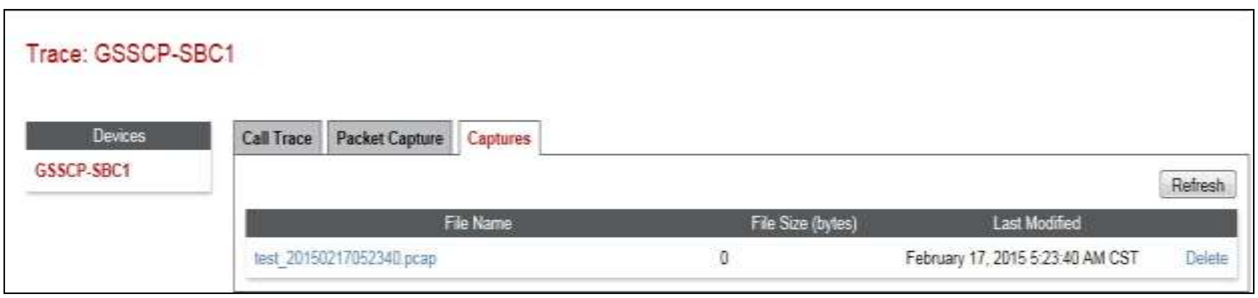
8.3.2. Trace Capture

To define the trace, navigate to **Device Specific Settings →Troubleshooting → Trace** in the menu on the left hand side and select the **Packet Capture** tab.

- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select **All** from the **Local Address** drop down menu.
- Enter the IP address of the Service Provider's SBC in the **Remote Address** field or enter a * to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, 10000 is shown as an example.
- Specify the filename of the resultant pcap file in the **Capture Filename** field.
- Click on **Start Capture**.



To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.



The trace is viewed as a standard pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response in the form of a 200 OK will be seen from the Vodafone Germany network.

9. Conclusion

These Application Notes demonstrated how IP Office Release 9.1 and Avaya Session Border Controller for Enterprise can be successfully combined with Vodafone Germany SIP Trunk service solution as shown in **Figure 1**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya IP Office with Avaya Session Border Controller for Enterprise can be configured to interoperate successfully with the Vodafone Germany SIP Trunk service. This solution provides IP Office and Avaya Session Border Controller for Enterprise users the ability to access the Public Switched Telephone Network (PSTN) via a SIP trunk using the Vodafone Germany SIP Trunk thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN.

10. Additional References

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

[1] *Avaya IP Office 9.1* Documentation CD, January 2015.

[2] *IP Office 9.1 Installation Manual*, January 2015.

[3] *IP Office Manager Manual 9.1, January* 2015.

[4] *IP Office Release 9.1 Implementing Voicemail Pro*, January 2015.

[5] *System Status Application*, November 2014.

[6] *Administering Avaya Communicator for Windows*, March 2015.

[7] *IP Office SIP Extension Installation*, October 2014.

[8] *Avaya IP Office Knowledgebase,* http://marketingtools.avaya.com/knowledgebase

[9] *Deploying Avaya Session Border Controller for Enterprise*, Release 6.3 October 2014.

[10] *Administering Avaya Session Border Controller for Enterprise*, Release 6.3 October 2014.

[11] RFC 3261 *SIP: Session Initiation Protocol*, http://www.ietf.org/

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