Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.1 with Virgin Media Business SIP Trunking – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Virgin Media Business SIP Trunking and Avaya IP Office.

The Virgin Media Business SIP Trunking service provides PSTN access via a SIP trunk connected to the Virgin Media Business Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. Virgin Media Business 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.

©2016 Avaya Inc. All Rights Reserved.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Virgin Media Business SIP Trunking and Avaya IP Office. Virgin Media Business SIP Trunking provides PSTN access via a SIP trunk connected to the Virgin Media Business 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 Virgin Media Business SIP Trunking. 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 Virgin Media Business SIP Trunking. 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 Virgin Media Business.
- Outgoing PSTN calls from various phone types including H.323, SIP and analogue telephones at the enterprise. Calls were routed from the enterprise across the SIP trunk to Virgin Media Business.
- Inbound and outbound PSTN calls to/from an Avaya Communicator for Windows client.
- Various call types including: local, international, toll free (outbound) and directory assistance.
- Codecs G.711A and G.711MU.
- 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.

©2016 Avaya Inc. All Rights Reserved.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Virgin Media Business SIP Trunking with the following observations:

- The IP Office private IP address could be seen in the Proxy-Authorization header on outgoing calls. This was rectified by specifying a domain name in the ITSP Domain Name in the SIP Line settings on the IP Office (See Section 5.6.2).
- Initially there were no Min-SE or Session-Expires headers in the outbound INVITE messages from IP Office. This was resolved by setting the SIP Line Session Timer to 3600 in line with that used by Virgin Media (See Section 5.6.2).
- The calling number displayed on IP Office Extensions for inbound calls was not in diallable format. This was resolved by configuring the IP Office to use the number in the From header as opposed to P-Asserted-ID (See Section 5.6.2).
- At the time of testing, the Virgin Media Business SIP trunk was not configured to support the G.729 codec.
- When testing DTMF on outbound calls from a SIP extension, some digits were missed by the voicemail system at the destination. This was thought to be due to network quality issues encountered during testing.
- Toll-Free access to the IP Office was not available for testing.
- Network quality issues were encountered when testing Emergency Services, in particular when testing the text device (Minicom 5000). Quality was improved by configuring the IP Office 500 V2 as a standalone device as opposed to an extension of the IP Office Server.
- In the normal operation of blind call transfer, leg 2 information is passed to the network in a REFER message. During testing, the network responded with “405 Method Not Allowed”. To resolve this, Outgoing Blind REFER was unchecked on IP Office.
- During testing, occasional events were observed where the network responded to an INVITE with 491 Request Pending. The IP Office sent an ACK in response to these messages to which the network responded with 407 Proxy Authentication Required. The IP Office did not respond to these messages and although the call was successful, there was unnecessary signalling from the network.
- Network quality issues were encountered when testing outbound fax and the fax transmission was unreliable. Quality was improved by configuring the IP Office 500 V2 as a standalone device as opposed to an extension of the IP Office Server.
- The conferencing of outbound calls from Avaya Communicator for Windows was not tested. The softphone requires the use of a conferencing server that was not available at the time of testing.
- During testing, Virgin Media Business observed that re-INVITEs were seen after call set-up. These were found to be due to the redirection of media to establish a path directly between IP endpoints or the IP Office 500 V2 extension and the Avaya SBCE. This was seen as undesirable and was turned off by unchecking “Allow Direct Media” (See Section 5.6.2).

©2016 Avaya Inc. All Rights Reserved.

2.3. Support

For technical support on Virgin Media Business products please contact the following website: http://www.virginmediabusiness.co.uk/help/s/

©2016 Avaya Inc. All Rights Reserved.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Virgin Media Business SIP Trunking. 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 Avaya Communicator for Windows for mobility testing. For security purposes, public IP addresses have been changed and 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.

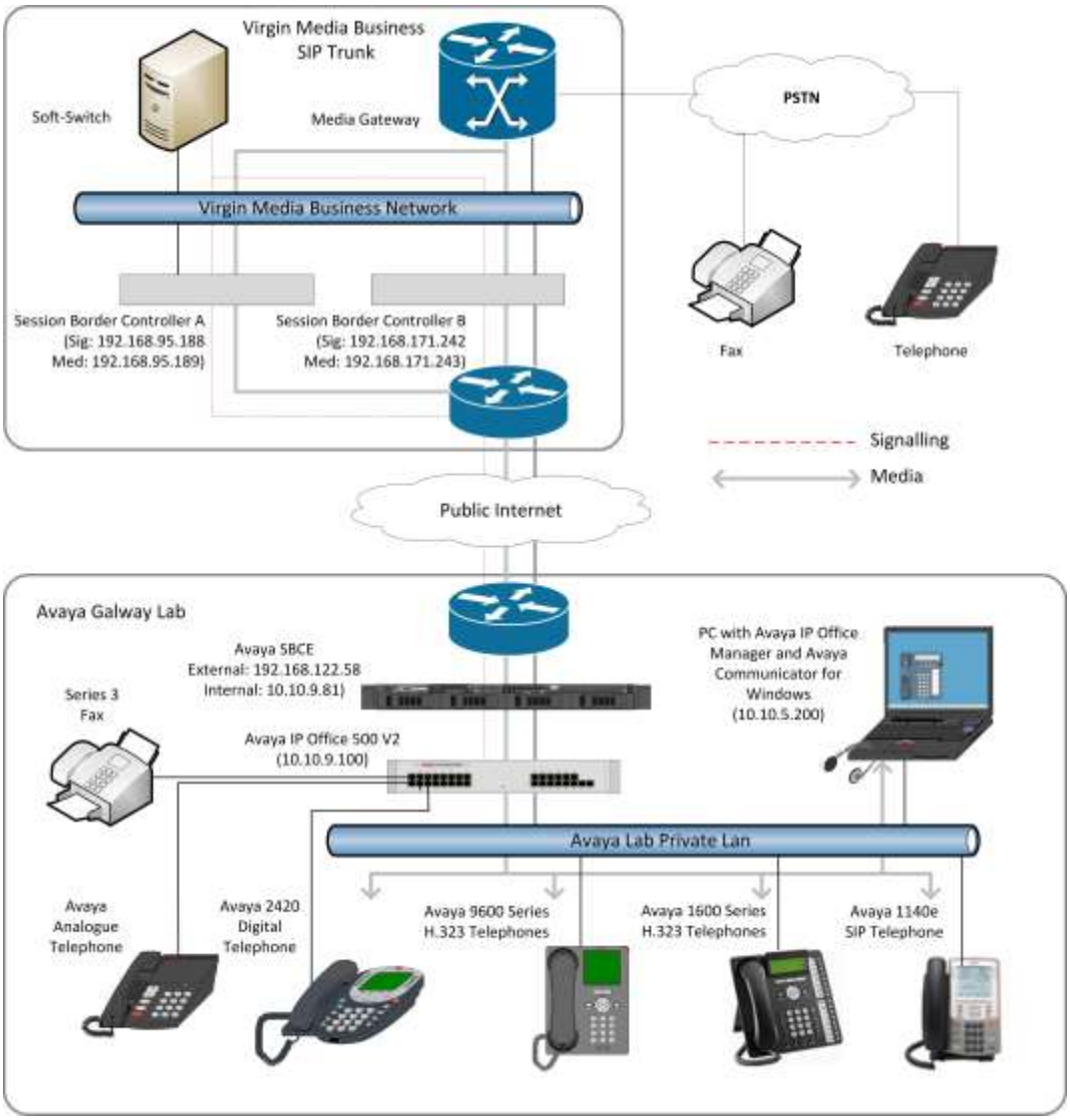


Figure 1: Virgin Media Business SIP Trunking to Avaya IP Office Topology

©2016 Avaya Inc. All Rights Reserved.

4. Equipment and Software Validated

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

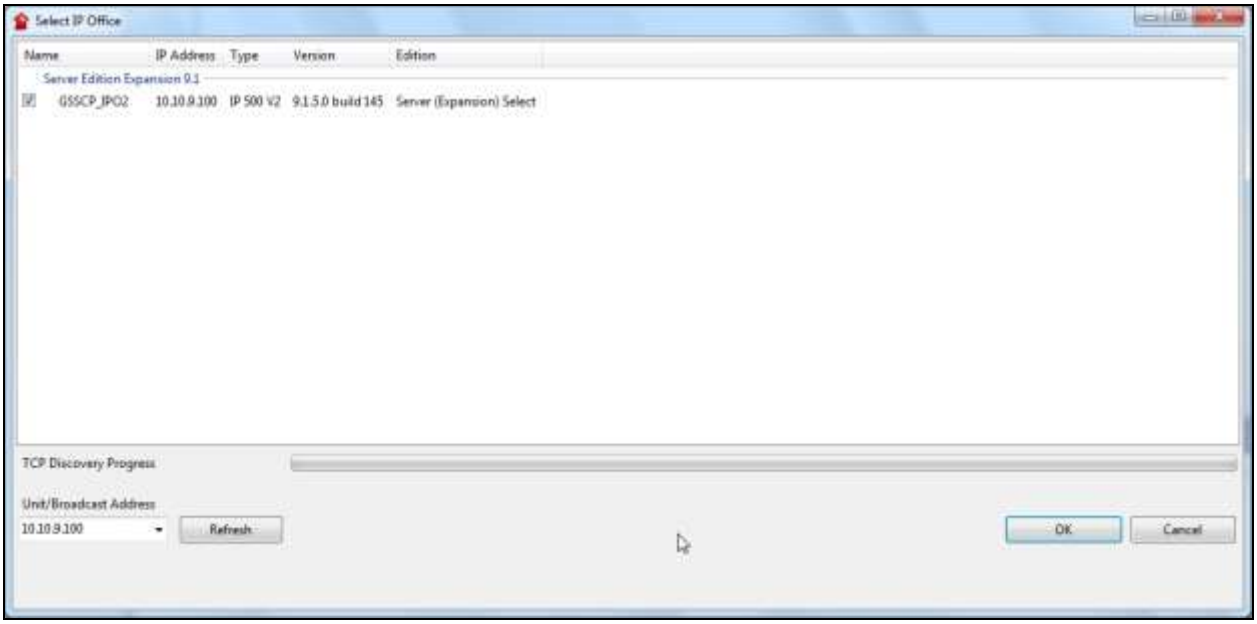
Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Avaya IP Office R9.1.500.145
Avaya Session Border Controller for Enterprise	7.0.1-03-8739
Avaya 1140e IP SIP Telephone	04.04.18.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.6.0.29
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Avaya Communicator	2.1.3.80
Avaya IP Office Manager	Version 9.1.5.0 build 145
Virgin Media Business	
Genband C20 Call Session Controller	CVM17
Genband Q20 Network SBC	8.3.8.7

Testing was performed with IP Office 500 V2 R9.1.500.145. 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. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

©2016 Avaya Inc. All Rights Reserved.

5. Configure Avaya IP Office

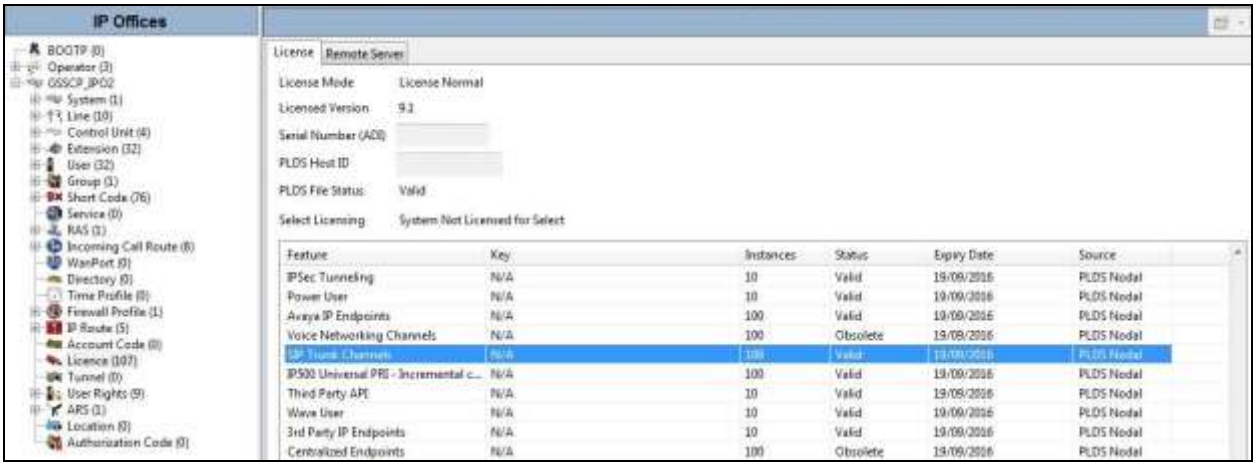
This section describes the Avaya IP Office configuration to support connectivity to the Virgin Media Business SIP Trunking. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → 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.



©2016 Avaya Inc. All Rights Reserved.

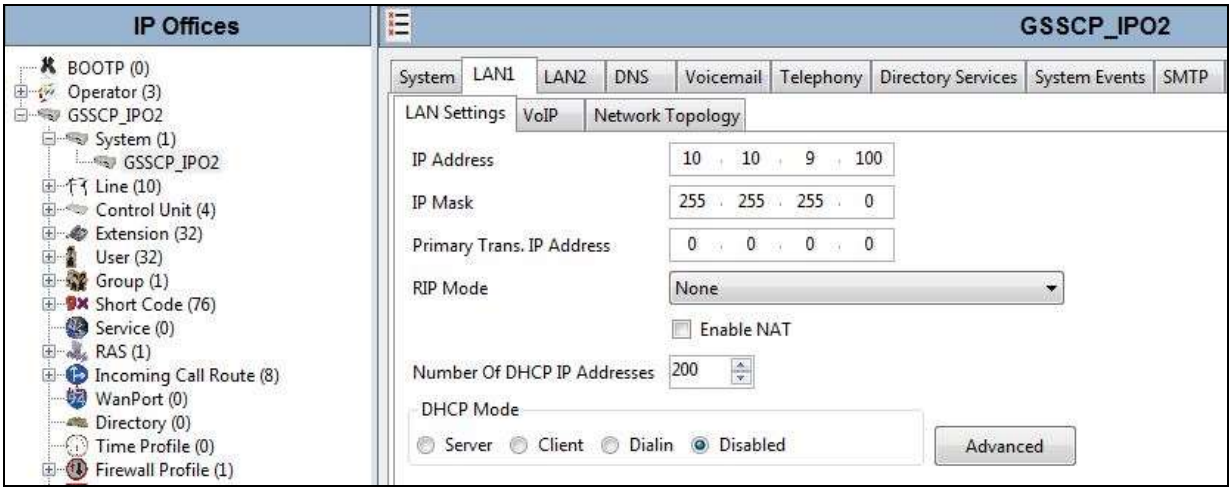
5.1. Verify System Capacity

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 Virgin Media Business.



5.2. LAN1 Settings

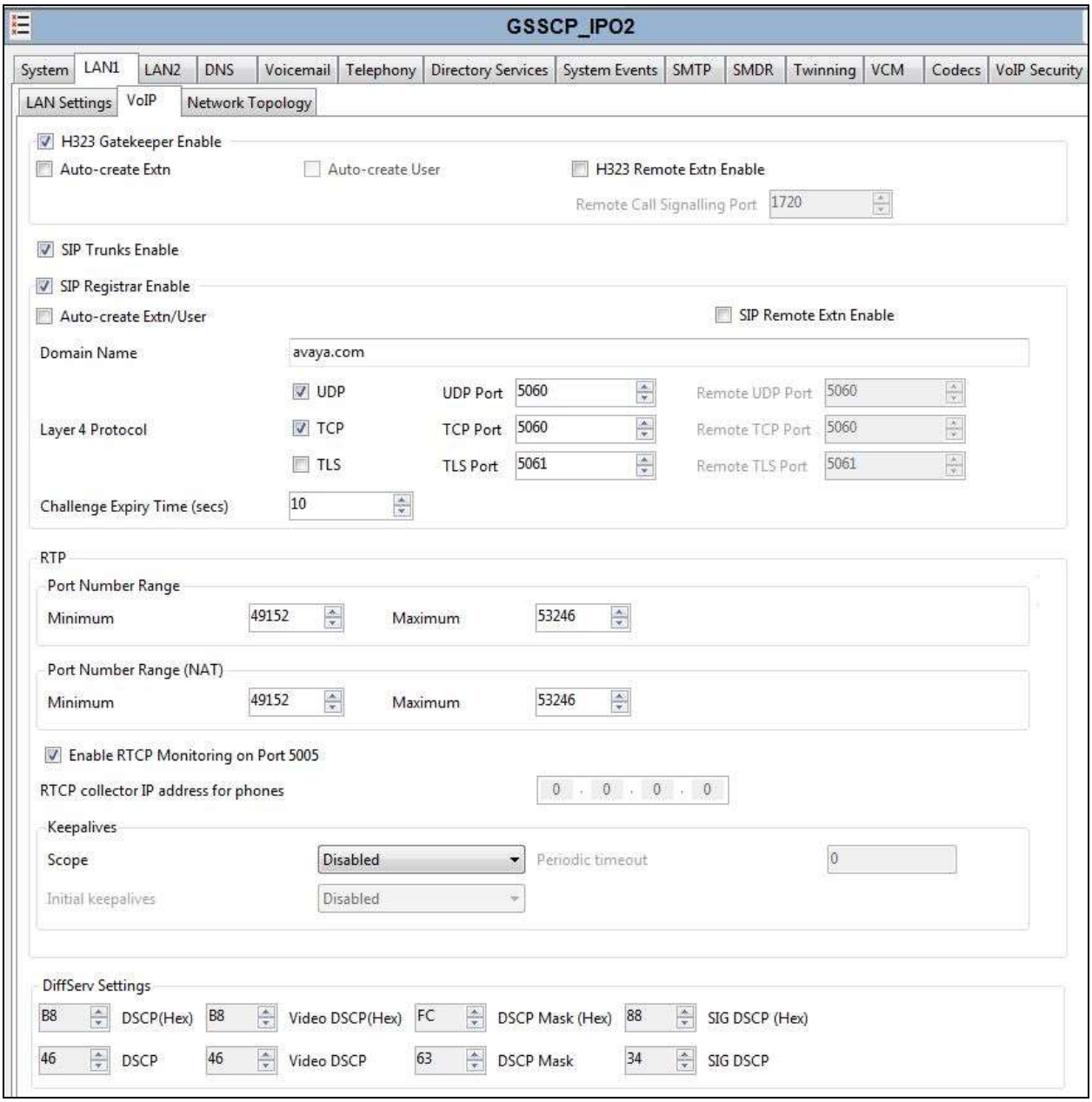
The configuration tested with the Virgin Media Business SIP trunk used an Avaya Session Border Controller for Enterprise (Avaya SBCE) so the only connection from IP Office was to the local network using the LAN1 connection. To access the LAN1 settings, first navigate to **System → <IP Office Name>** in the Navigation Pane where <IP Office Name> is the name of the Avaya IP Office. This is **GSSCP_IPO2** in the GSSCP test environment. Navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the interface of the Avaya IP Office; **Primary Trans. IP Address** is the next hop, usually the default gateway address. In this configuration, **Primary Trans IP Address** is not set as the default route is configured in the **IP Route** settings (not shown). All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



©2016 Avaya Inc. All Rights Reserved.

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 Avaya Communicator 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 **TCP** 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 LAN1. The range used for testing was the default values of **49152** to **53246**.



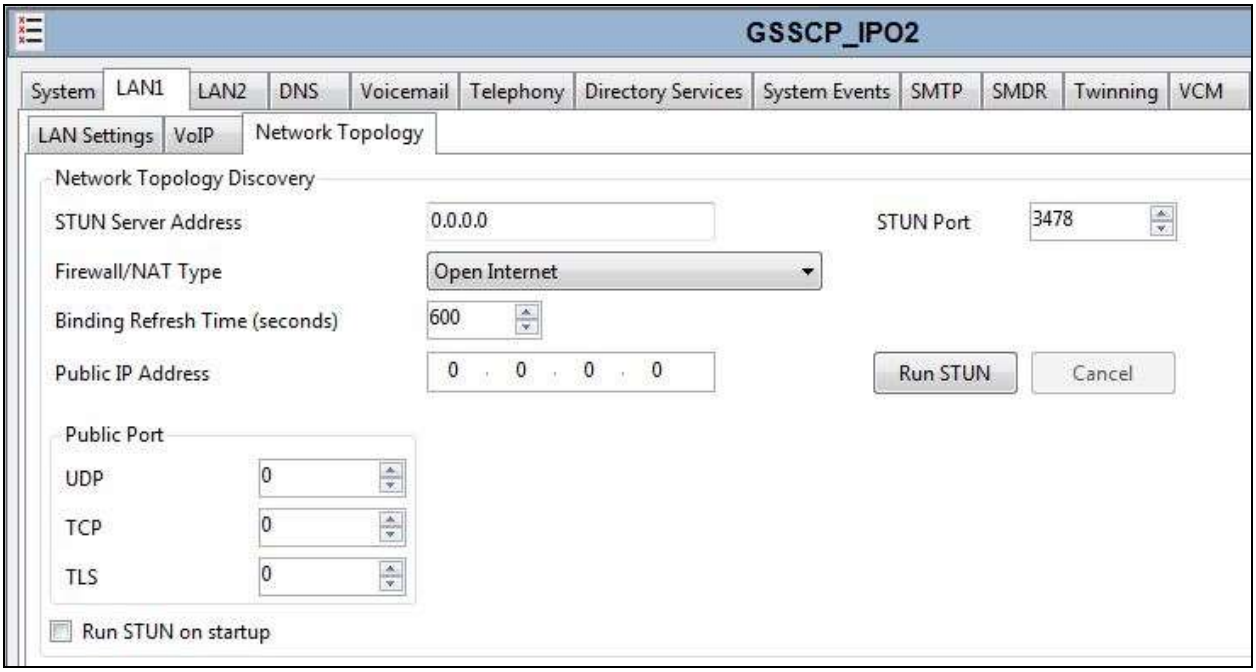
©2016 Avaya Inc. All Rights Reserved.

Note: 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 (not shown). 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.

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, leave the **STUN Server IP Address** at the default setting of **0.0.0.0** and the Firewall/NAT Type at **Open Internet** as NAT is not required in this configuration.

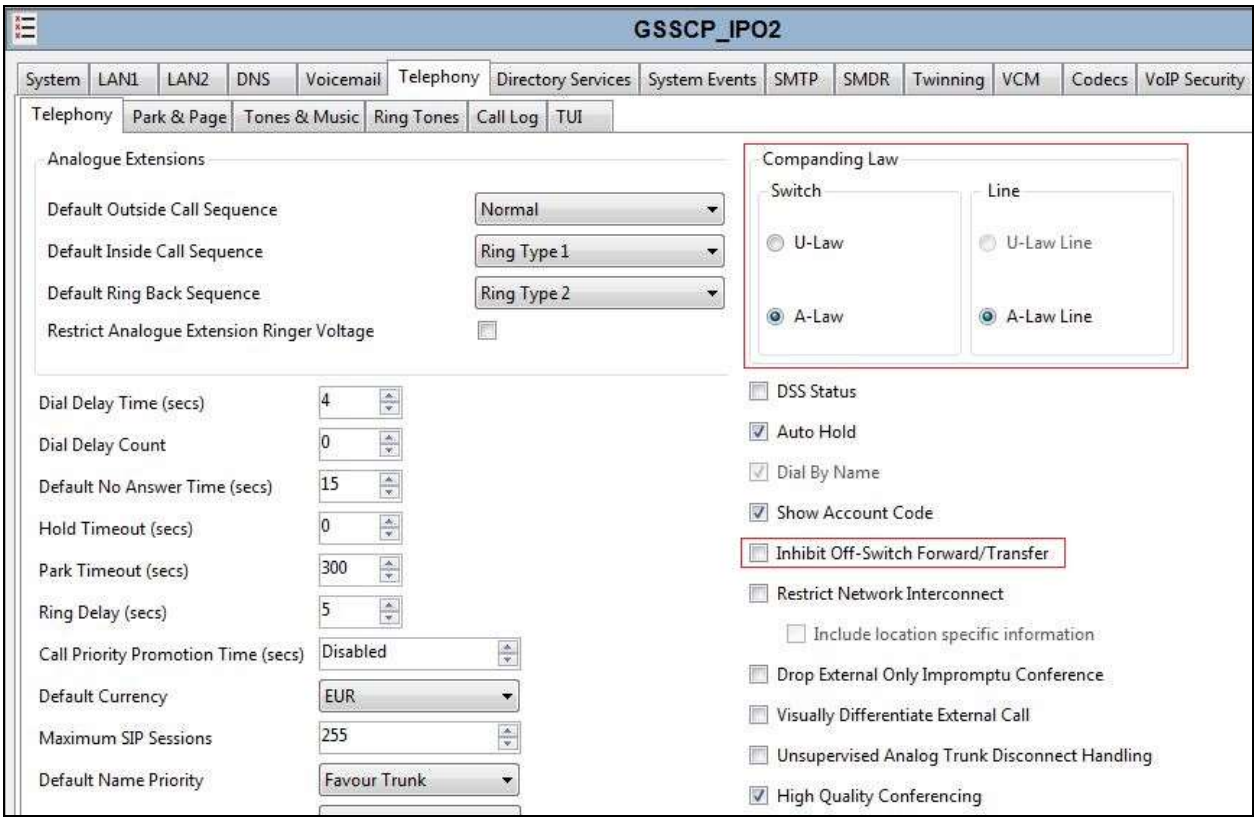
The Network Topology tab can be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. During testing, IP Office sent OPTIONS messages at an interval of 10 minutes. This was achieved by setting the **Binding Refresh Time** to **600**.



©2016 Avaya Inc. All Rights Reserved.

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).



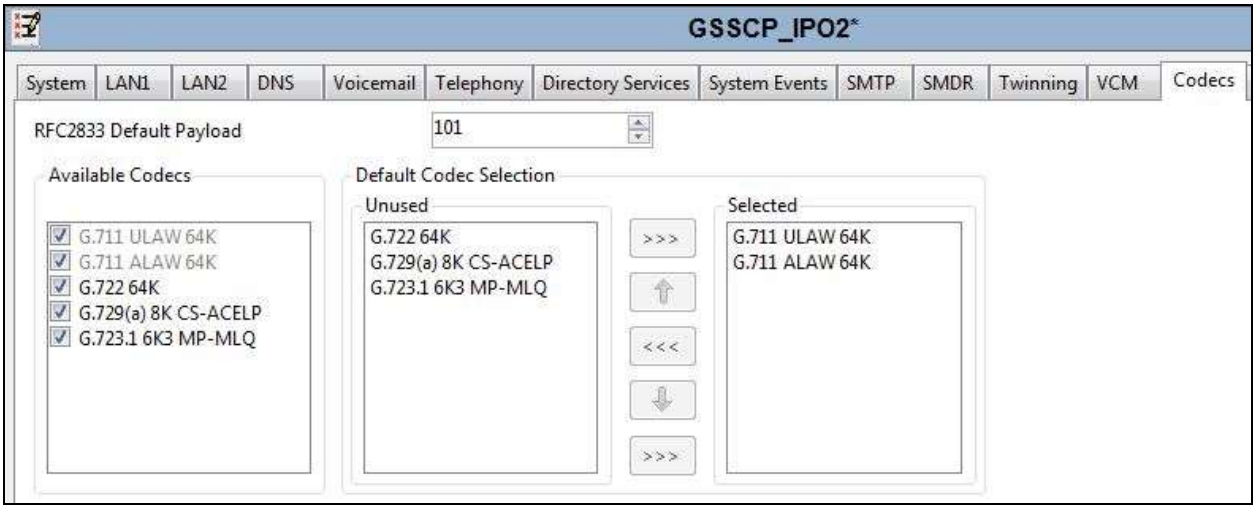
©2016 Avaya Inc. All Rights Reserved.

5.4. System Twinning Settings

Navigate to the **Twinning** tab and ensure that the box labeled **Send original calling party information for Mobile Twinning** is unchecked and that the **Calling party information for Mobile Twinning** field is blank. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header and will use SIP Line settings for sending the calling party number of the host phone. The SIP Line settings for sending caller ID are described in **Section 5.6.2**. On completion, click the **OK** button (not shown).

5.5. Codec Settings

Navigate to the **Codecs** tab on the Details Pane. Check the Available Codecs boxes as required. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test, **G.711 ALAW 64K** and **G.711 ULAW 64K** were used as default codecs. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



©2016 Avaya Inc. All Rights Reserved.

5.6. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Virgin Media Business SIP Trunking. 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

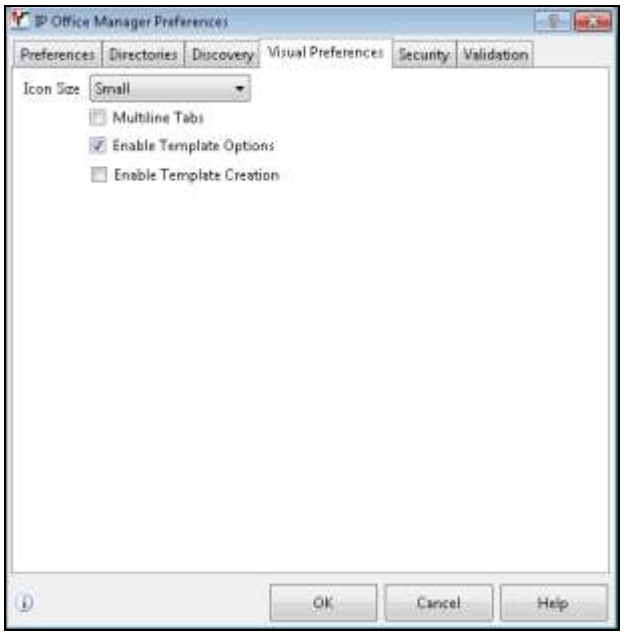
Alternatively, a SIP Line can be created manually. To do so, right-click Line in the Navigation Pane and select **New→SIP Line** (not shown). Then, follow the steps outlined in **Section 5.6.2**.

©2016 Avaya Inc. All Rights Reserved.

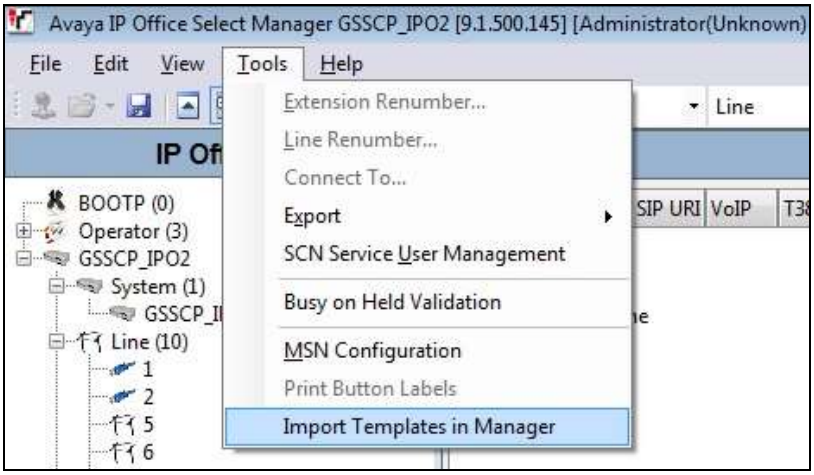
5.6.1. SIP Line From Template

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



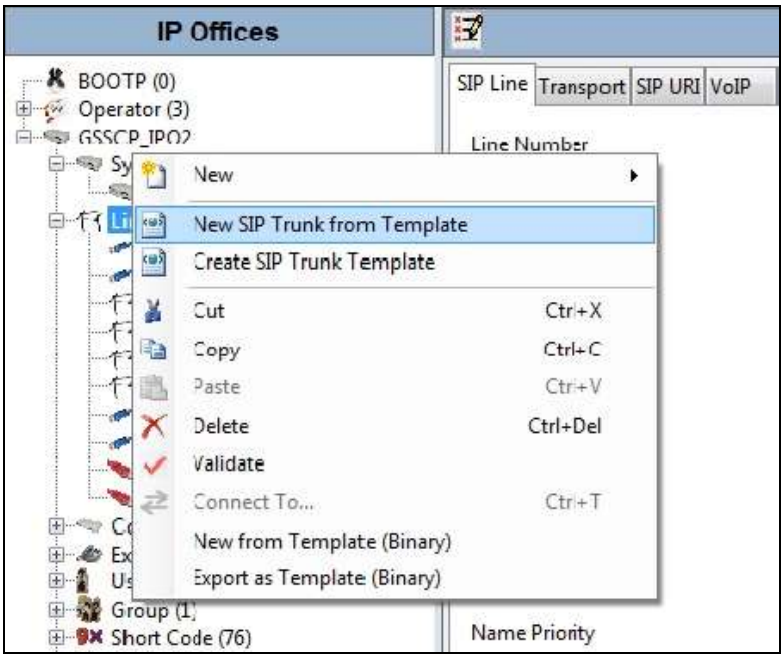
Import the template into IP Office Manager. Select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus. The default template location is **C:\Program Files\Avaya\IP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



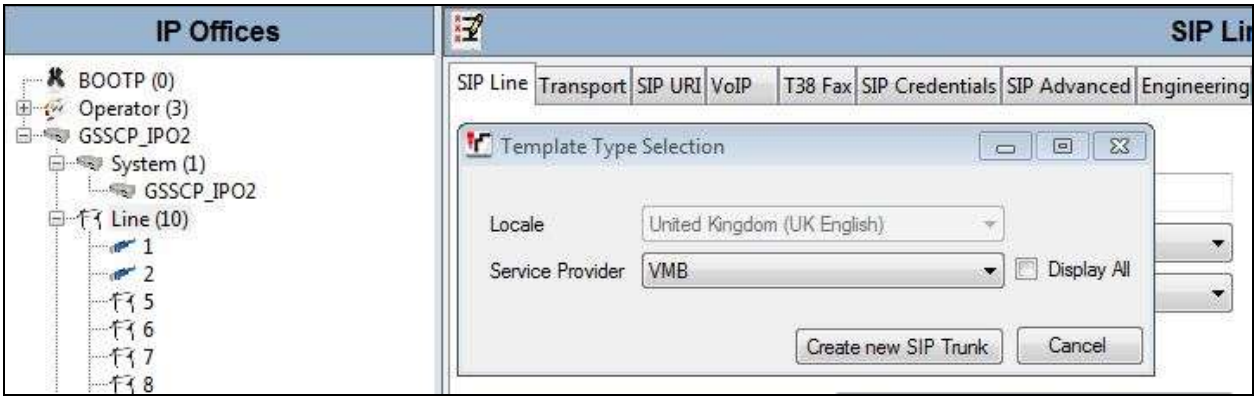
©2016 Avaya Inc. All Rights Reserved.

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 **VMB** from the **Service Provider** pull-down menu as shown below. This value corresponds to part of the file name (**AF_VMB_SIPTrunk.xml**) created earlier. Click **Create new SIP Trunk** to finish creating the trunk.



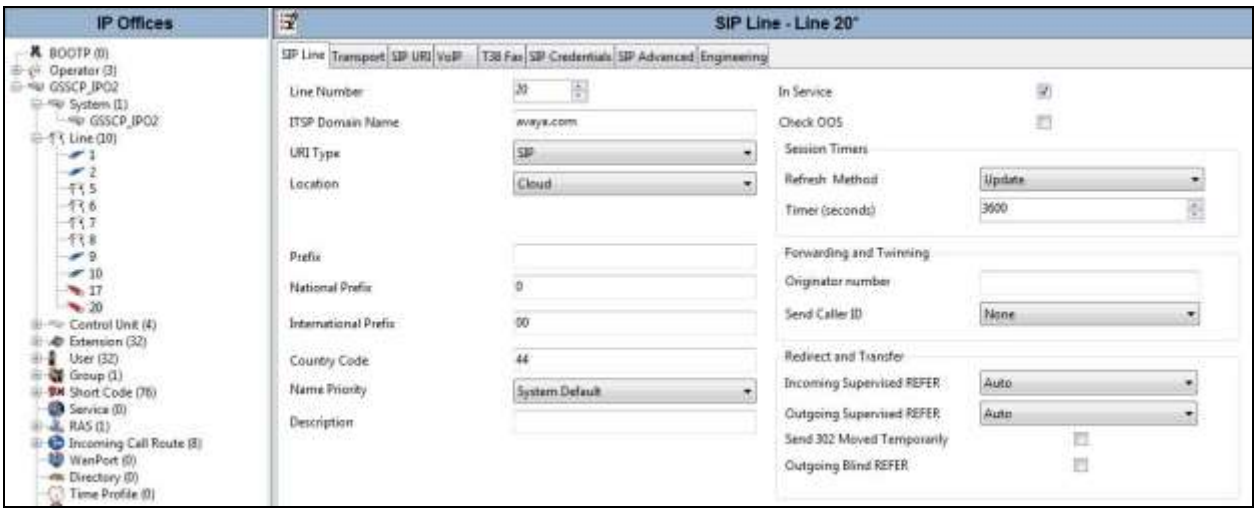
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

©2016 Avaya Inc. All Rights Reserved.

5.6.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to a domain name agreed with Virgin Media Business. In test this was set to **avaya.com**. This setting was required so that the internal IP address of the Avaya SBCE was not sent in the Proxy Authentication header on outgoing calls.
- Leave **Prefix blank and set the**, **National Prefix** and **International Prefix** to those used in the UK. This ensures that Calling Party Numbers are presented on the IP Office extensions in diallable format. It also removes the prefixes on outgoing dialled numbers for conversion to E.164 format.
- Set **Country Code** to 44 for the UK, this prefixes the country code on outgoing dialled numbers for conversion to E.164 format with leading "+". This conforms to the Virgin Media Business Global dial plan used during testing.
- Uncheck the **Check OOS** box so that the SIP Trunk is not taken out of service when there is no response to OPTIONS. Virgin Media Business use two SBCs for resilience and the SIP Trunk must not be taken out of service when one of them is unavailable.
- Set **Refresh Method** to **Update** and specify the Timer (seconds) value as **3600**. This ensures that Session Refresh messages are sent according to the timer indicated in the Session-Expires header.
- Ensure the **In Service** box is checked.



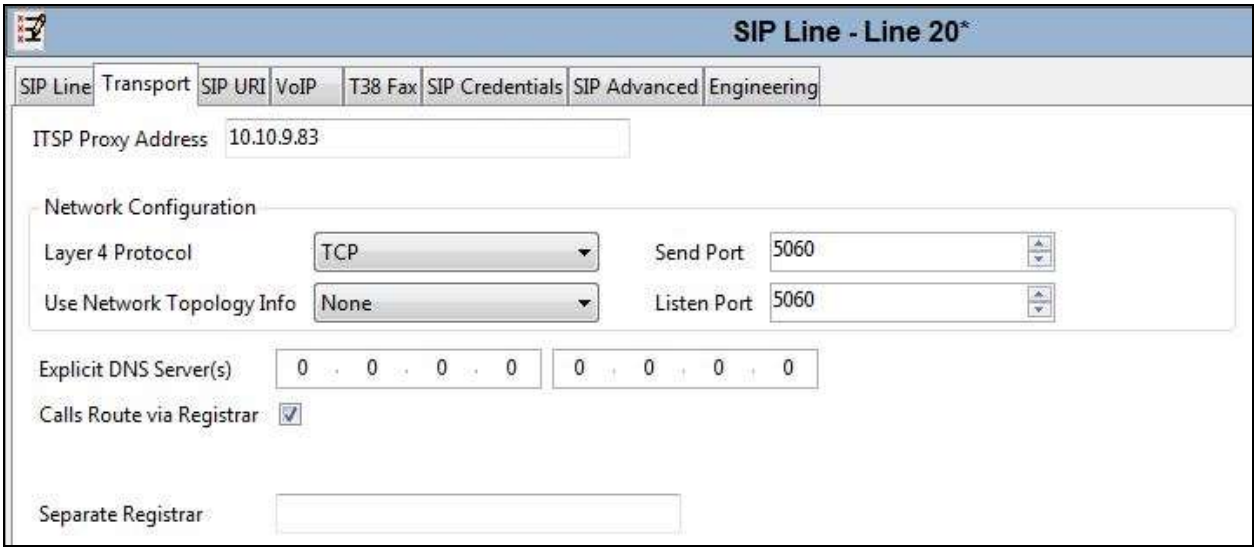
On completion, click the **OK** button (not shown).

©2016 Avaya Inc. All Rights Reserved.

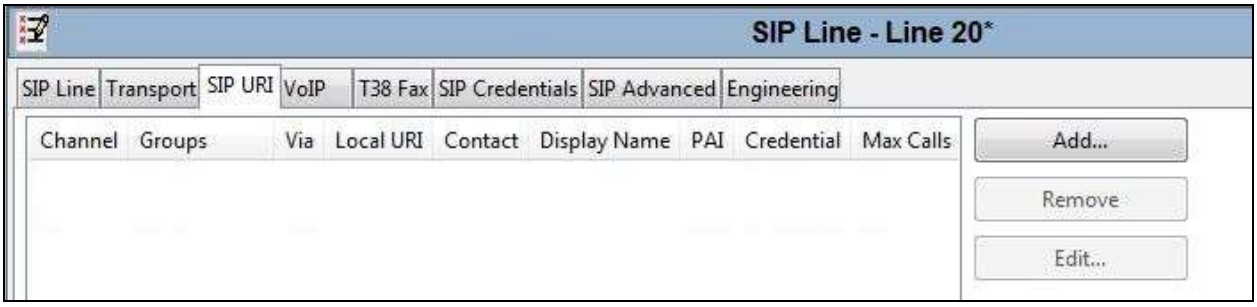
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Addres**s to the internal IP address of the Avaya SBCE.
- Set **Use Network Topology Info** to **None** as NAT is not used in this configuration and the Network Topology settings defined in **Section 5.2** are not required.
- Set **Layer 4 Protocol** to **TCP**.
- Set **Send Port** and **Listen Port** to **5060**.

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



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



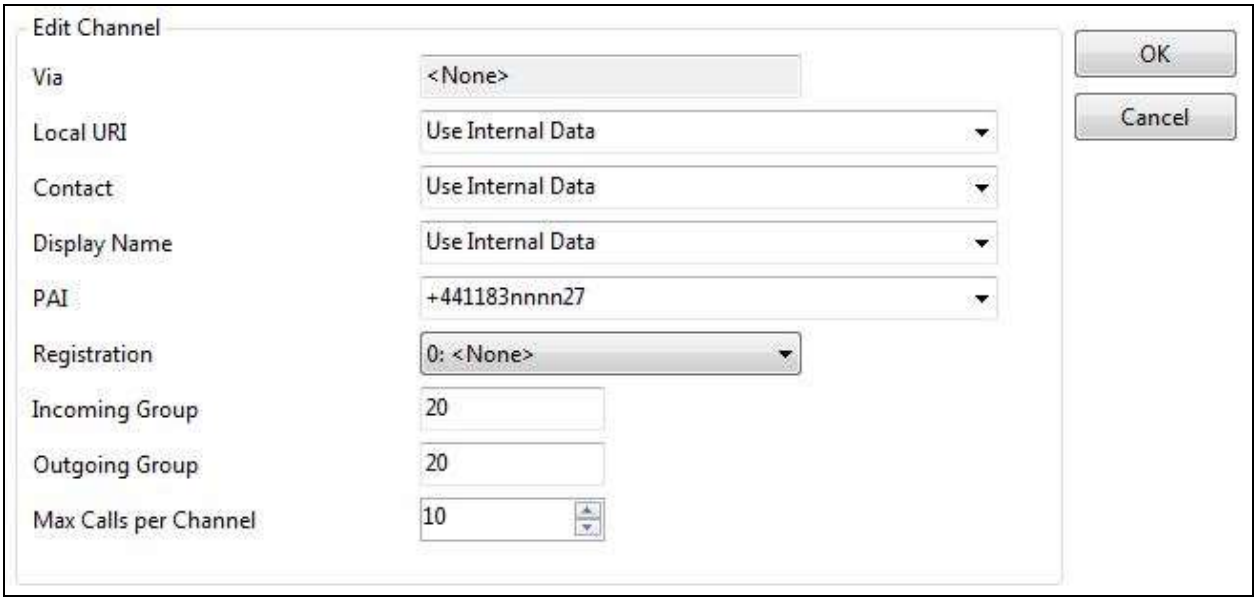
For the compliance test, two SIP URI entries were created. One specified internal data for the local URI and the other specified a wild card (*). Internal data was used for outgoing calls so that calling party number could be sent in the correct format. The wild card was specified for incoming calls to numbers where there was no corresponding IP Office extension. For example Voice Mail and Feature Name Extensions (FNE)

©2016 Avaya Inc. All Rights Reserved.

The entry for outgoing calls was created with the parameters shown below.

- Set **Local URI**, **Contact** and **Display Name** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.8**. It is the default setting when no SIP Credentials are specified.
- Set **PAI** to a group number for the IP Office if required. Note that the number used during testing has been obscured.
- Leave the **Registration** field at the default value of **None** as registration is not used for Virgin Media Business SIP Trunking.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **20** was defined that was associated to a single line (line 20).
- Associate this line with an outgoing line group by entering a line group number in the **Outgoing Group** field. For the compliance test, a new outgoing group **20** was defined that was also associated to line 20.
- 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.

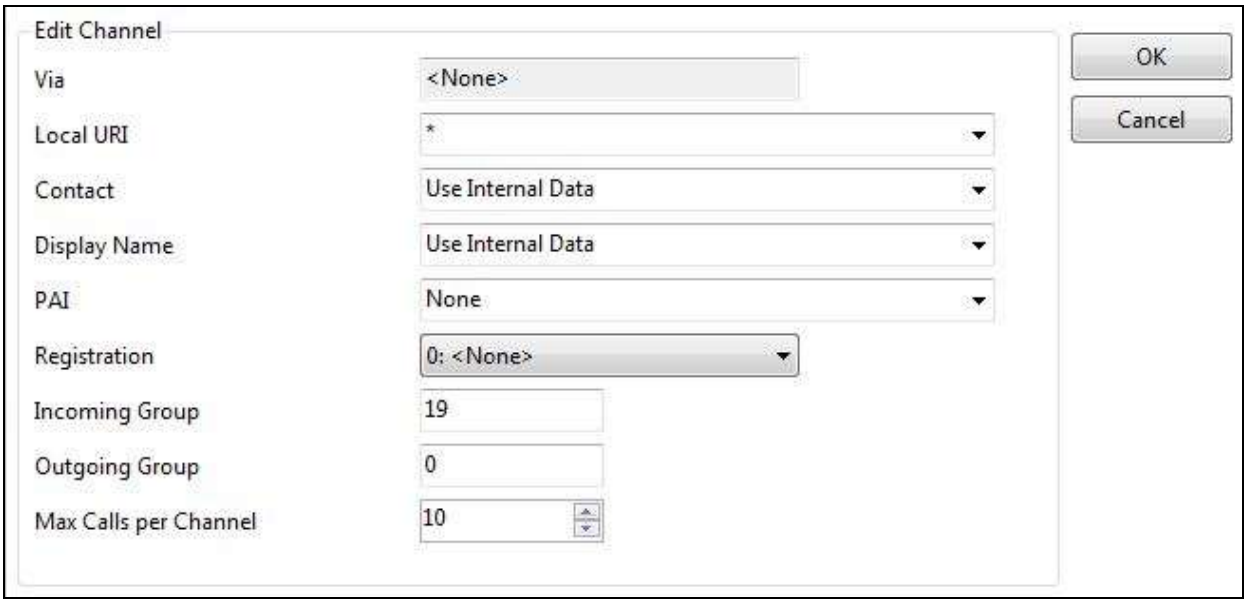


©2016 Avaya Inc. All Rights Reserved.

The entry for incoming calls was created with the parameters shown below.

- Set **Local URI** to *****. This will match anything other than the DDI numbers applied to specific extensions, for example Voice mail and FNE.
- Leave **Contact**, **Display Name** and **PAI** at default settings. This URI is only used only for incoming calls and these settings are not required. During testing, Contact and Display Name were left at **Use Credentials User Name** and **PAI** was left at **None**.
- Leave the **Registration** field at the default setting of **None** as registration is not required.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **19** was defined that was associated to line 20. As this SIP URI entry is not to be used for outgoing calls, the **Outgoing Group** was set to **0**
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

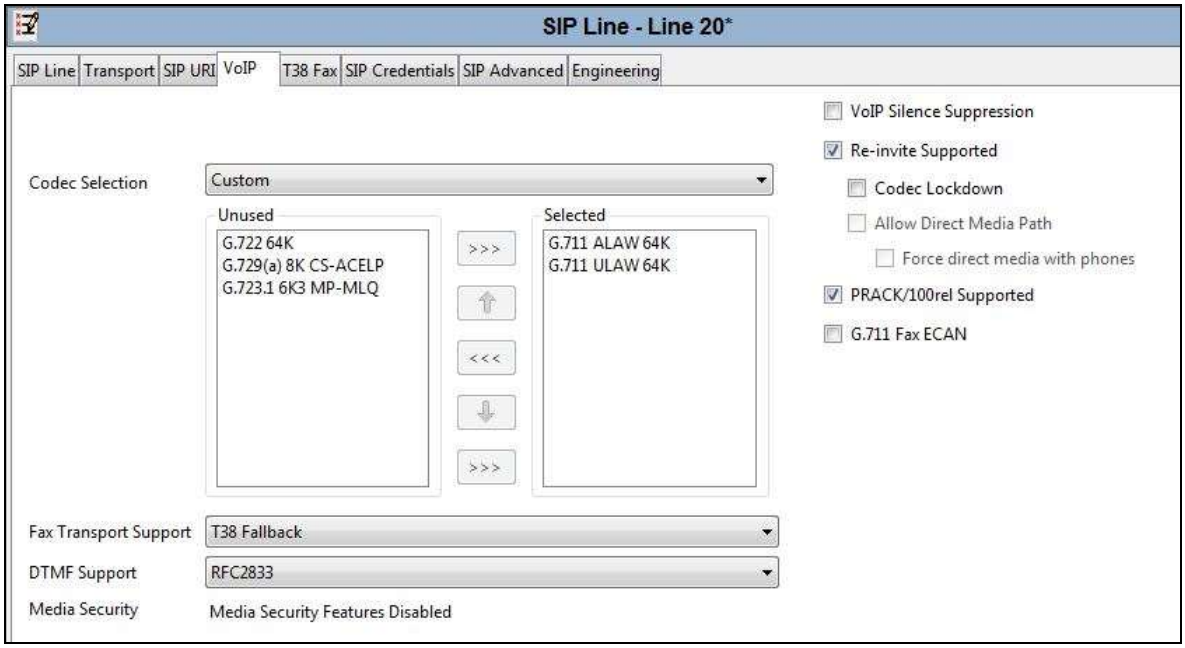
On completion, click the **OK** button.



©2016 Avaya Inc. All Rights Reserved.

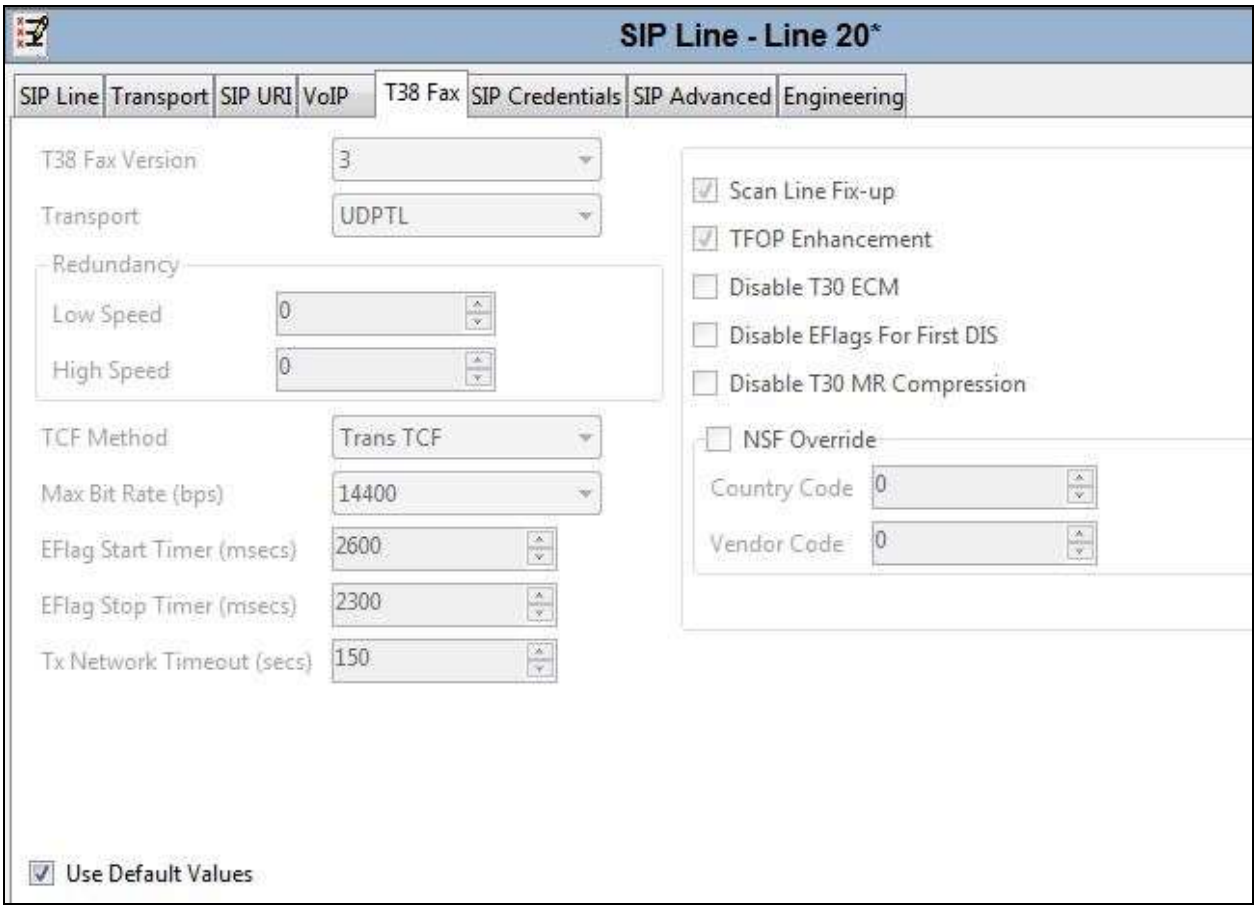
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- In **Section 5.5**, system default codecs were defined. If any other codec combination is required for this SIP Line, select **Custom** in the **Codec Selection** drop down menu.
- 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 Virgin Media Business this was **G.711 ALAW 64K** and **G.711 ULAW 64K**. 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. Both T.38 and G.711 fax transmission were successfully tested with Virgin Media Business, T38 is the Avaya preferred method.
- 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. Both RFC2833 and inband DTMF transmission were successfully tested with Virgin Media Business, RFC2833 is the Avaya preferred method.
- 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.
- Ensure that **Allow Direct Media Path** is unchecked.
- Check the **PRACK/100rel Supported** box if early media is required. This was checked during compliance testing.
- On completion, click the **OK** button (not shown).



©2016 Avaya Inc. All Rights Reserved.

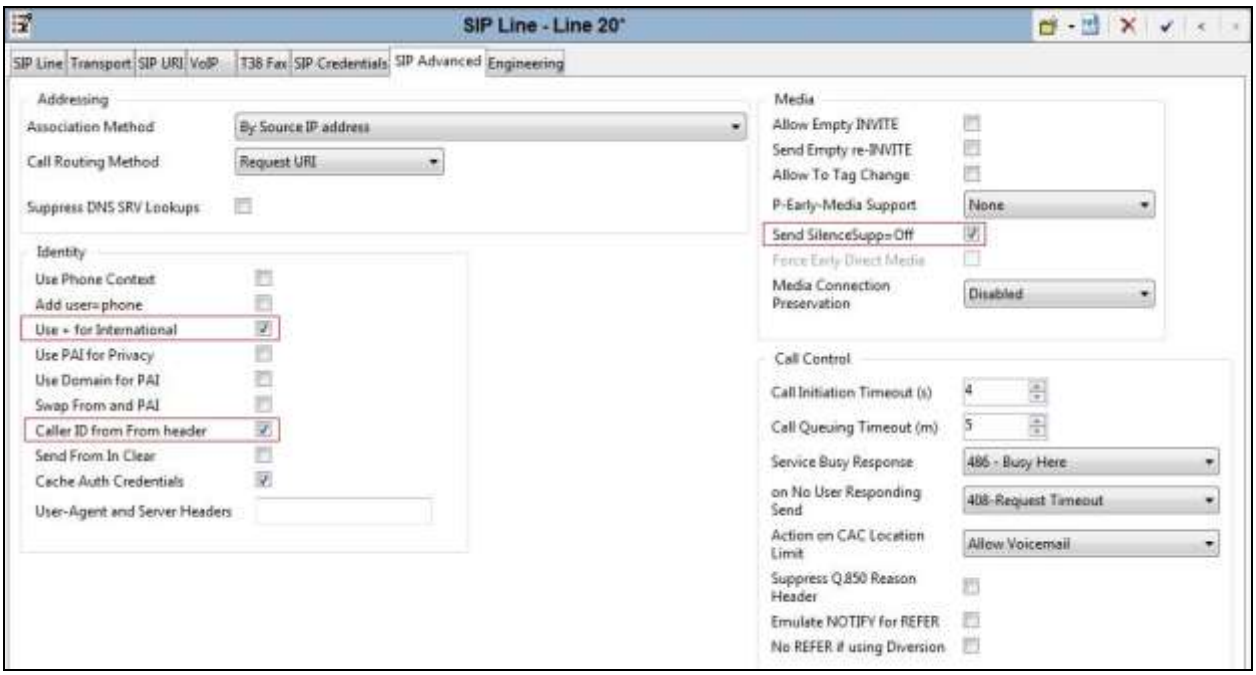
Select the **T.38 Fax** tab to set the T.38 parameters for the line. During compliance testing, default values were used by checking the **Use Default Values** box. If other settings are required, uncheck this box so that parameters can be individually set. On completion, click the **OK** button (not shown).



©2016 Avaya Inc. All Rights Reserved.

Select the **SIP Advanced** tab and set the following:

- If the Virgin Media Business global dialplan is used, check the **Use + for International** box to prefix the outbound E.164 called party numbers with "+". This setting also ensures that the leading "+" is removed from calling party numbers on inbound calls for display on IP Office extensions in diallable format.
- Check the **Caller ID from From header** box as P-Asserted-ID is not provided by the network.
- During testing, the **Send SilenceSupp=Off** box was checked to ensure that silence suppression was not a factor in the poor quality of text Emergency Services calls, though there was no indication in the call traces that silence suppression was being used.
- Default values may be used for all other parameters.



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

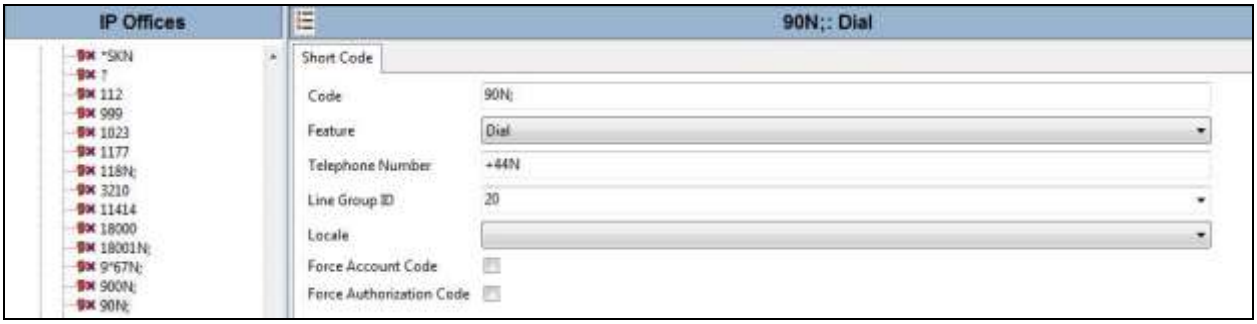
Note: It is advisable at this stage to save the configuration as described in **Section 5.10**to make the Line Group ID defined in **Section 5.6** available.

©2016 Avaya Inc. All Rights Reserved.

5.7. Short Codes

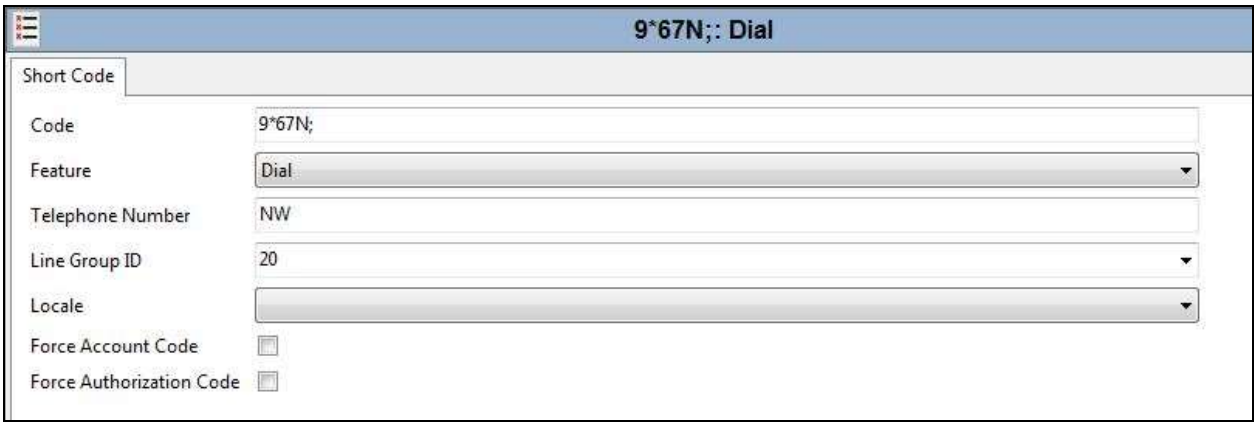
Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as shown in the example below for national numbers.

- 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 national number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **+44N** which removes the access code and national prefix and inserts the country code and number with leading "+" 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).



A further example is shown of a short code to route numbers where CLI is to be withheld:

- The **Code** is **9*67N** which is an outbound call prefixed with *67 which indicates that CLI is to be withheld.
- Set **Telephone Number** to **NW** which removes the access code and the *67 and inserts the dialled number with a "W" suffix that causes Avaya IP Office to withhold the CLI.



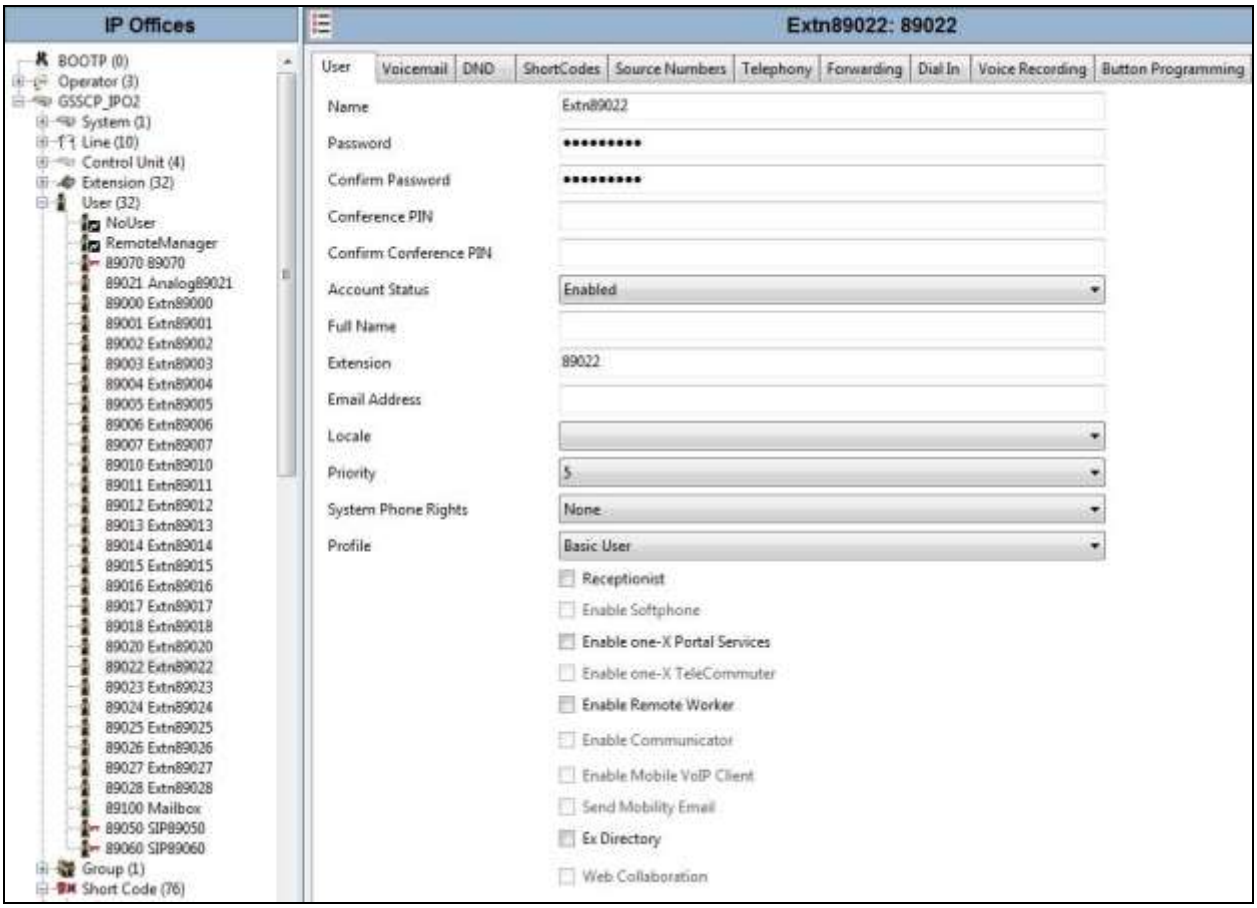
©2016 Avaya Inc. All Rights Reserved.

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 following example shows the configuration required for an analogue endpoint, the same fields are configured for H.323 and SIP.

- Change the **Name** of the User if required, this will be used for login to the Avaya IP Office Softphone.
- The **Password** and **Confirm Password** fields are set, though these are not used for analogue phones.
- Select the required profile from the **Profile** drop down menu. **Basic User** is used commonly used, **Power User** can be selected for SIP softphone and Remote Worker endpoints.

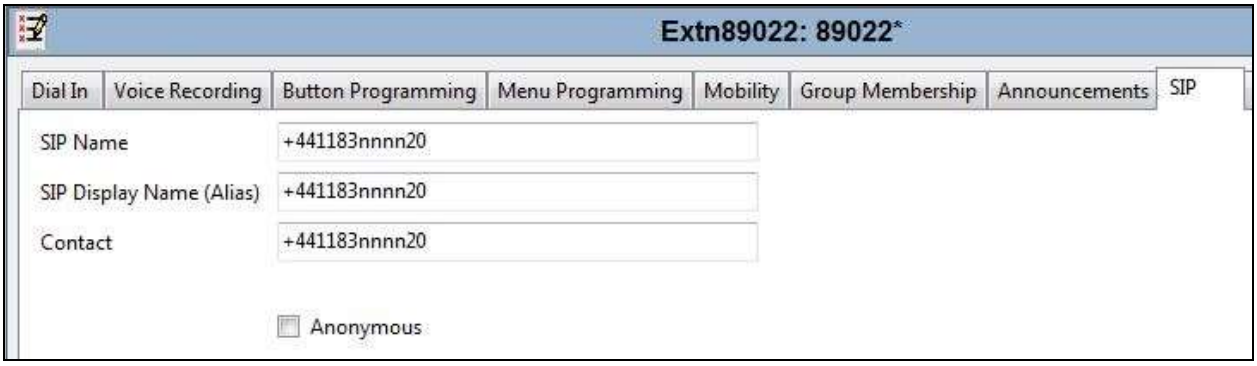


SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**.

©2016 Avaya Inc. All Rights Reserved.

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 fields should be set to the DDI numbers assigned to the enterprise from Virgin Media Business in E.164 format with leading +.

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



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

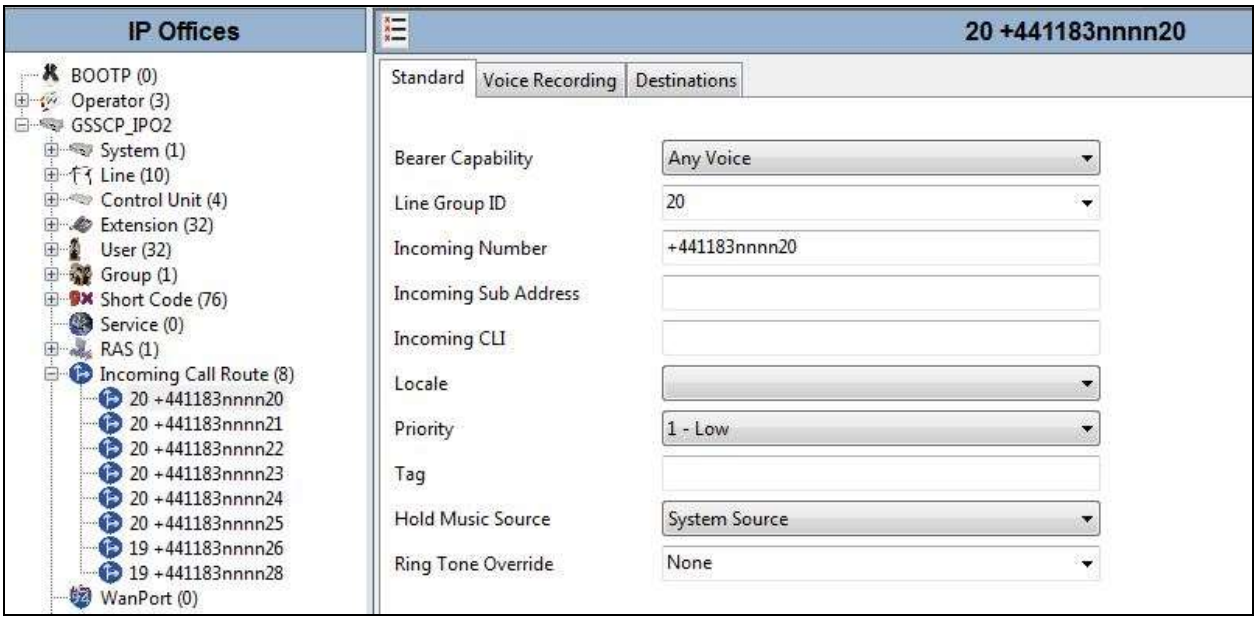
©2016 Avaya Inc. All Rights Reserved.

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** in the Navigation Pane and select **New**, (not shown**)**.

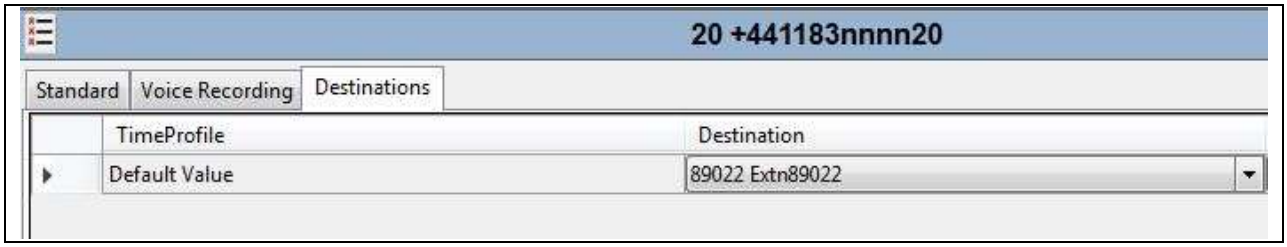
On the **Standard** tab of the Details Pane, enter the parameters as shown below:

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



Note: A number of digits of the DDI have been obscured. Number format is prefixed with the country code for the UK with leading "+" in line with the Virgin Media Business global dialplan.

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 20 are routed to extension 89022.

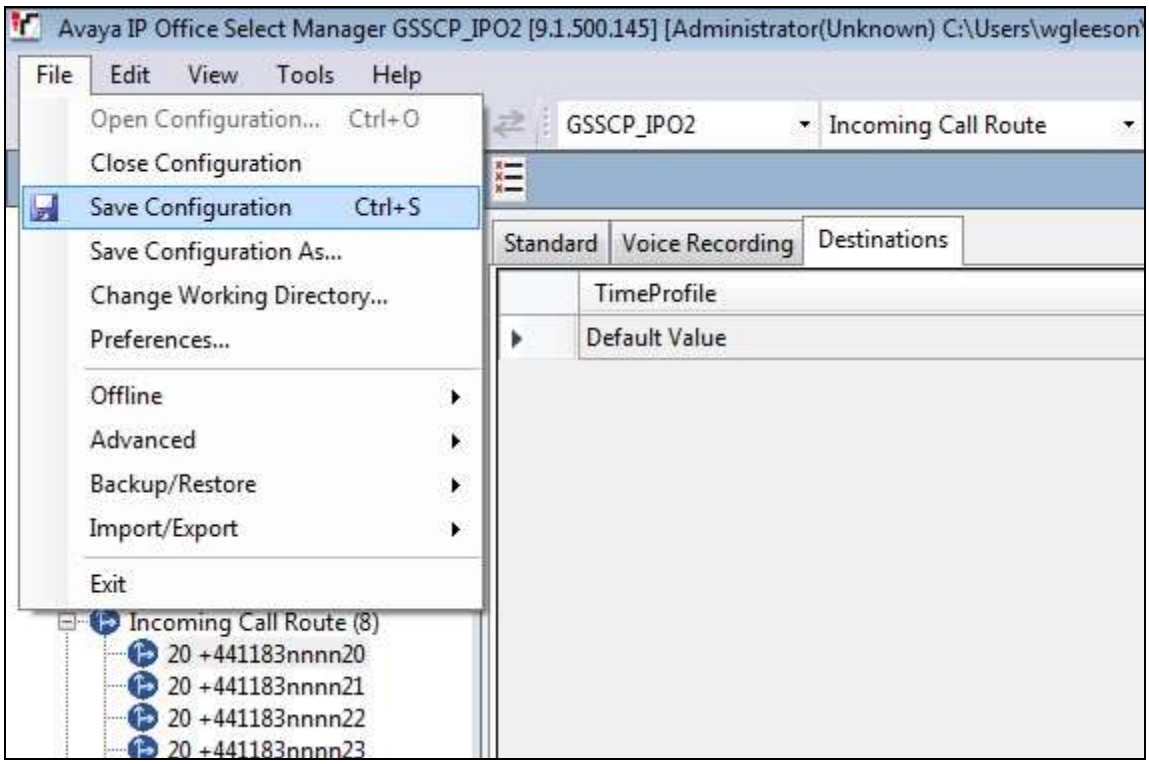


©2016 Avaya Inc. All Rights Reserved.

Note: Calls coming in to destinations not associated with an extension such as Voice Mail and FNE appear on line 19 in this configuration. This is because the incoming number can't be matched to a DDI number specified in the **User** settings shown in **Section 5.8**.

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.



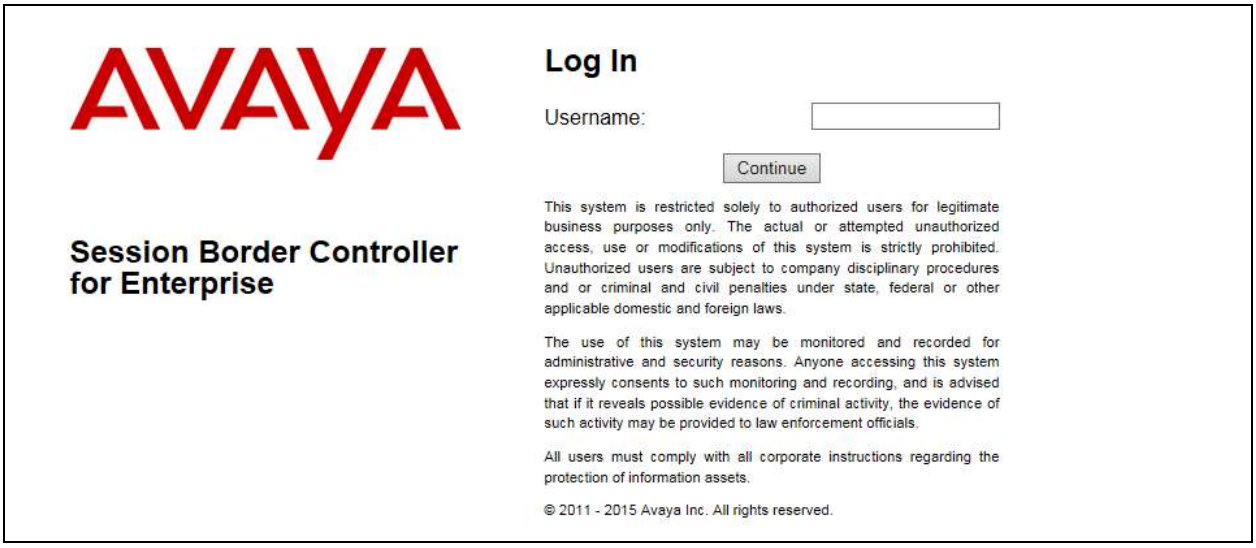
©2016 Avaya Inc. All Rights Reserved.

6. Configure Avaya Session Border Controller for Enterprise

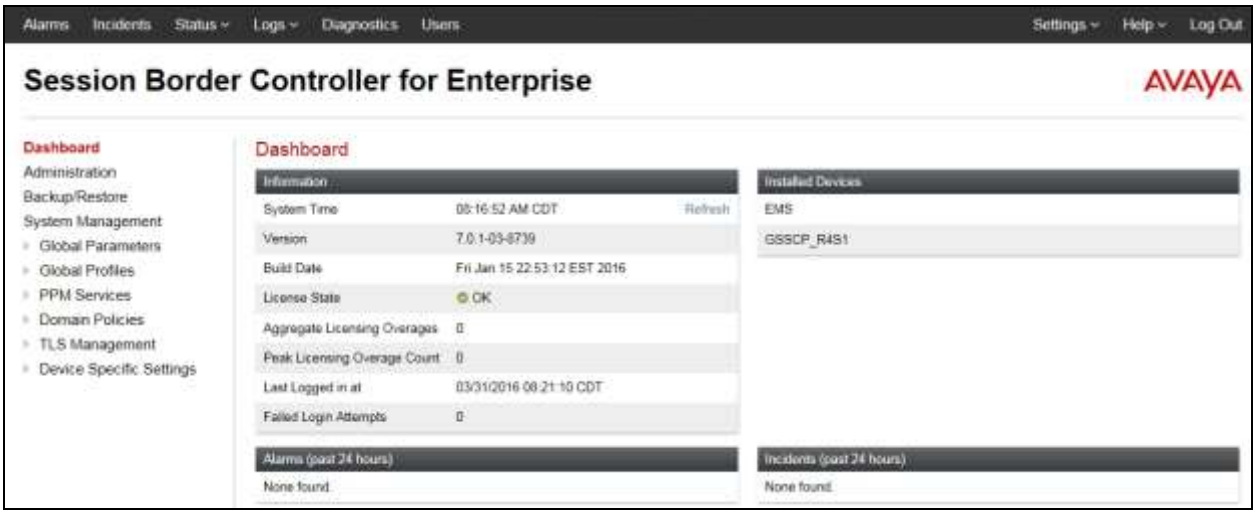
This section describes the configuration of the Avaya Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's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

6.1. Access Avaya Session Border Controller for Enterprise

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. A log in screen is presented. Log in using the appropriate 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.

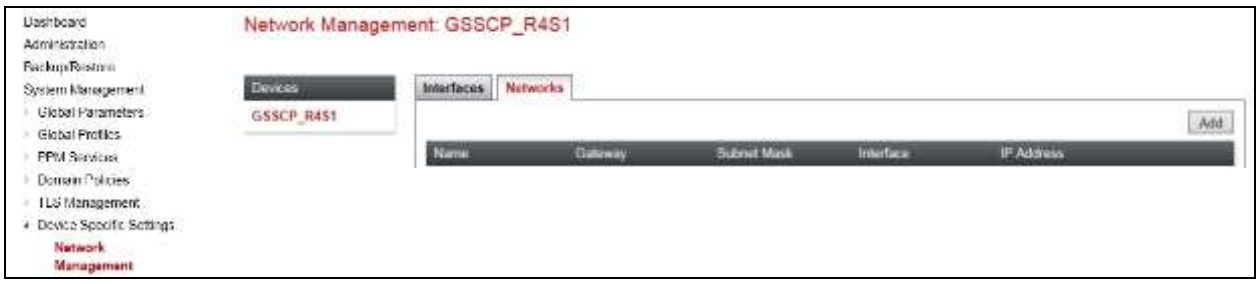


©2016 Avaya Inc. All Rights Reserved.

6.2. Define Network Management

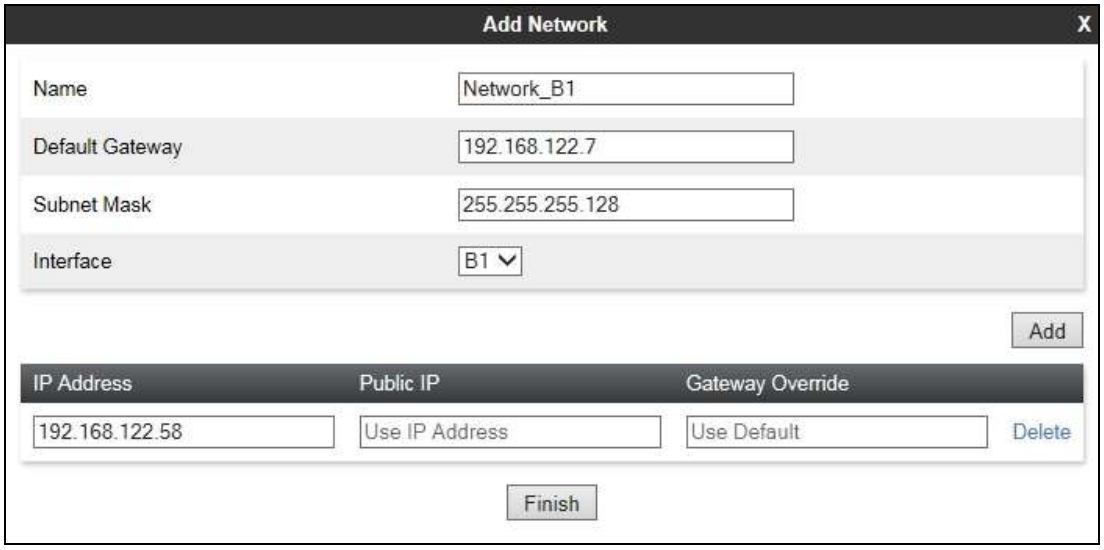
Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one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

To define the network information, navigate to **Device Specific Settings → Network Management** in the main menu on the left hand side and click on **Add**.



Enter details for the external interface in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the test environment, this was **B1.**
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the external interface IP address for the SIP trunk in the IP Address field and leave the Public IP and Gateway Override fields blank.
- Click on **Finish** to complete the interface definition.

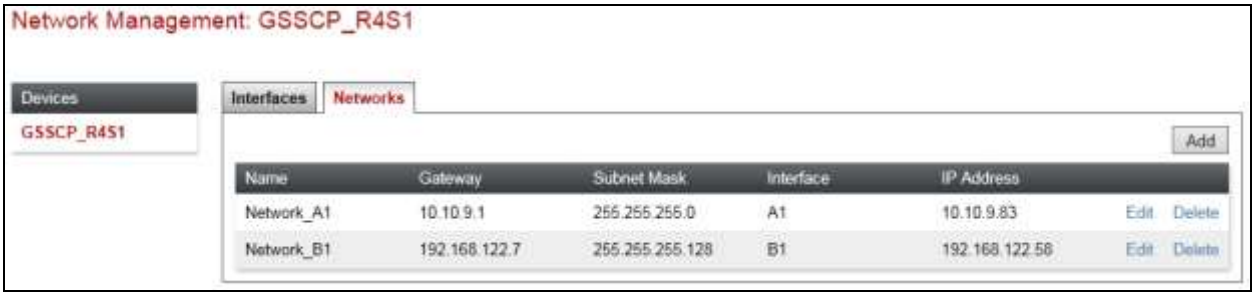


©2016 Avaya Inc. All Rights Reserved.

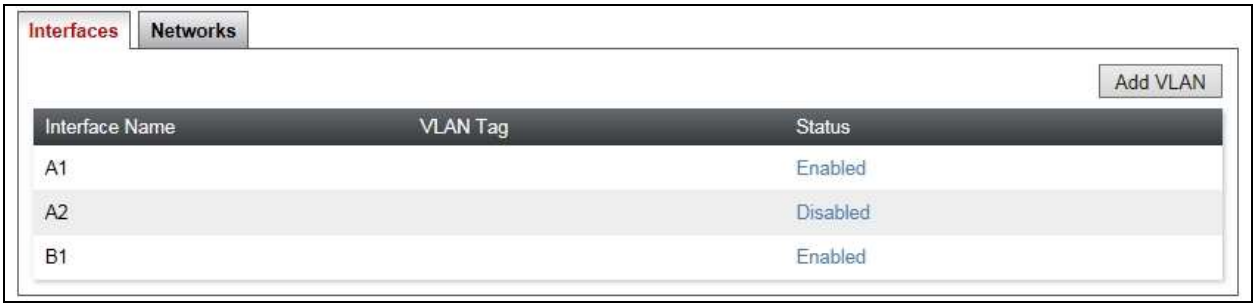
Click on **Add** again to define the internal interface. Enter details in the dialogue box (not shown):

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1.**
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal interface IP address for the SIP Trunk in the IP Address field and leave the Public IP and Gateway Override fields blank.
- Click on **Finish** to complete the interface definition.

The following screenshot shows the completed Network Management configuration:



Select the **Interface Configuration** tab and click on the **Status** of the physical interface to toggle the state. Change the state to **Enabled** where required.



Note: to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

- Click on **System Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear that will indicate when the restart is complete.

©2016 Avaya Inc. All Rights Reserved.

6.3. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces. Testing was carried out with TCP used for transport of signalling between IP Office and the Avaya SBCE, and UDP for transport of signalling between the Avaya SBCE and the Virgin Media Business SIP Trunking Service. Signalling and media interfaces were required on both the internal and external sides of the Avaya SBCE to handle traffic between the Avaya SBCE and IP Office, and Avaya SBCE and Virgin Media Business SIP Trunk. This document shows the configuration for TCP and UDP, if additional security is required, it's recommended to use TLS and port 5061.

6.3.1. Signalling Interfaces

To define the signalling interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Signaling Interface** (not shown) in the main menu on the left hand side. Details of transport protocol and ports for the external and internal SIP signalling are entered here.

- Select **Add** and enter details of the external signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external signalling interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 6.2**. In the test environment, this was IP address **192.168.122.58**.
- Enter the UDP port number in the **UDP Port** field, **5060** is used for the Virgin Media Business SIP Trunking Service.

The internal signalling interfaces are defined in the same way; the dialogue box is not shown:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal signalling interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.
- Select **TCP** port number, **5060** is used for IP Office.

©2016 Avaya Inc. All Rights Reserved.

The following screenshot shows details of the signalling interfaces:

Note: In the test environment, the internal IP address was **10.10.9.83**.

6.3.2. Media Interfaces

To define the media interfaces on the Avaya SBCE, navigate to **Device Specific Settings → Media Interface** in the main menu on the left hand side. Details of the RTP 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.

- Select **Add** and enter details of the external media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- In the **IP Address** drop down menus, select the external network interface and IP address. Note that when the external network interface is selected, the bottom drop down menu is populated with the available IP addresses as defined in **Section 6.2**. In the test environment, this was IP address **192.168.122.58**.
- Define the RTP **Port Range** for the media path with the Virgin Media Business SIP Trunking Service, during testing this was left at the default values.

©2016 Avaya Inc. All Rights Reserved.

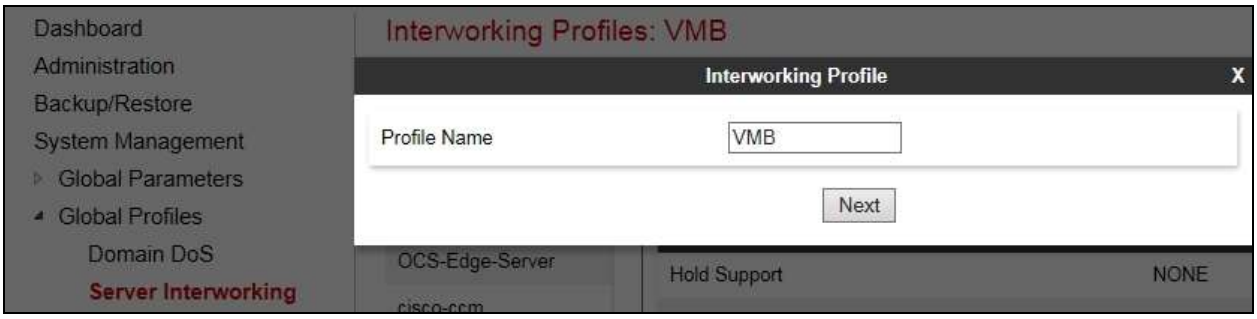
The internal media interfaces are defined in the same way; the dialogue box is not shown:

- Select **Add** and enter details of the internal media interface in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- In the **IP Address** drop down menus, select the internal network interface and IP address.

Note: In the test environment, the internal IP address was **10.10.9.83**.

6.4. Define Server Interworking

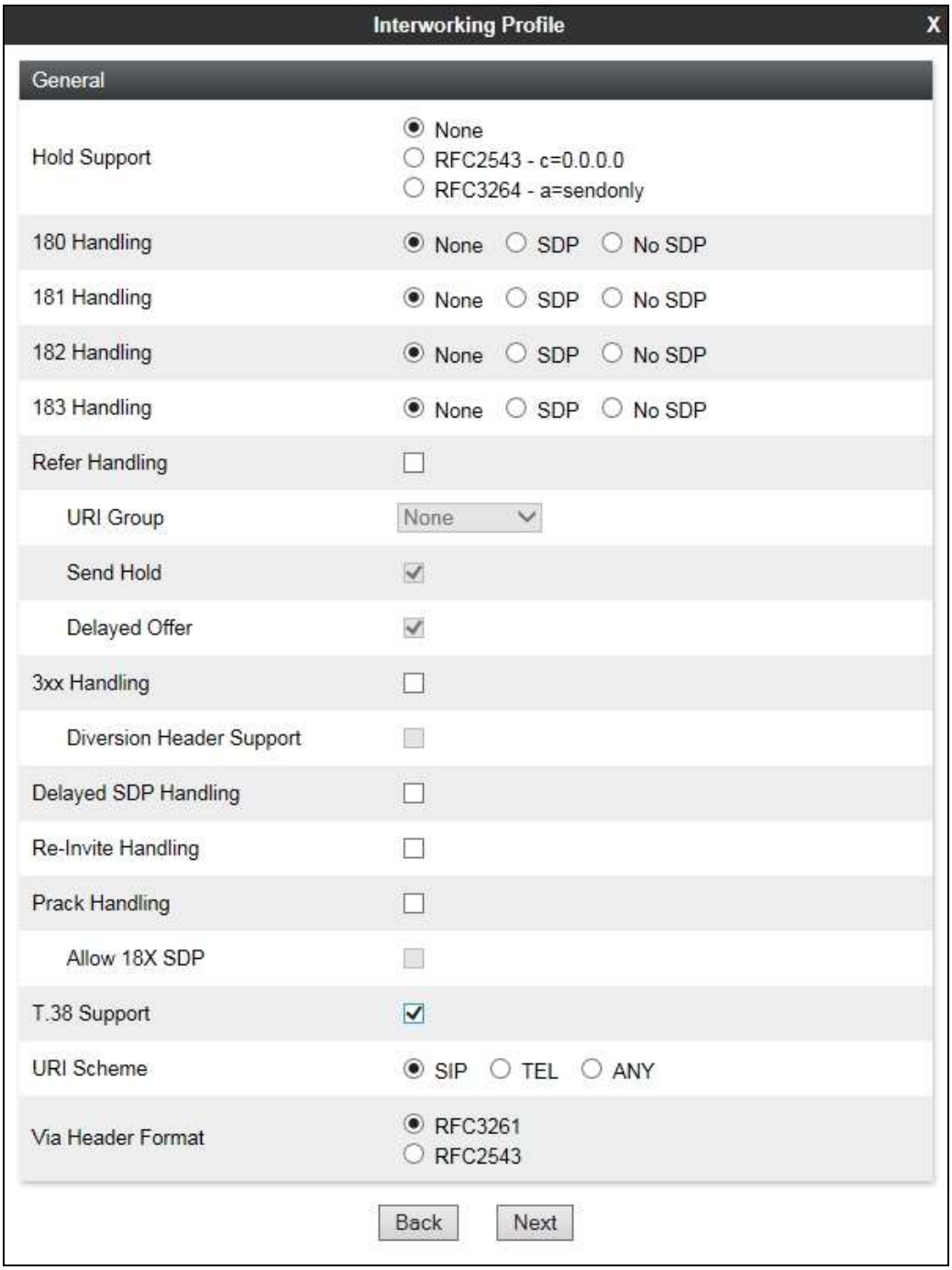
Server interworking is defined for servers connected to the Avaya SBCE. To define server interworking, navigate to **Global Profiles → Server Interworking** in the main menu on the left hand side. To define Server Interworking for the Virgin Media Business SIP Trunking Service, click on **Add** (not shown). A pop-up menu is generated. In the **Name** field enter a descriptive name for the Virgin Media Business SIP Trunking Service and click **Next**.



©2016 Avaya Inc. All Rights Reserved.

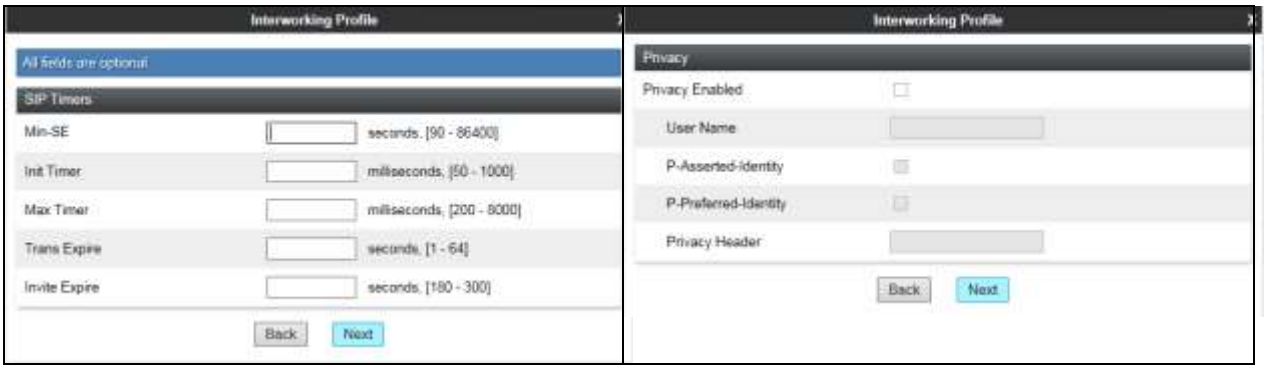
Configuration of interworking includes Hold support, T.38 fax support and SIP extensions. In the General dialogue box shown in the screenshot, define the interworking as follows:

- Check the **T.38 Support** box.
- During testing, the rest of the parameters were left at default values.

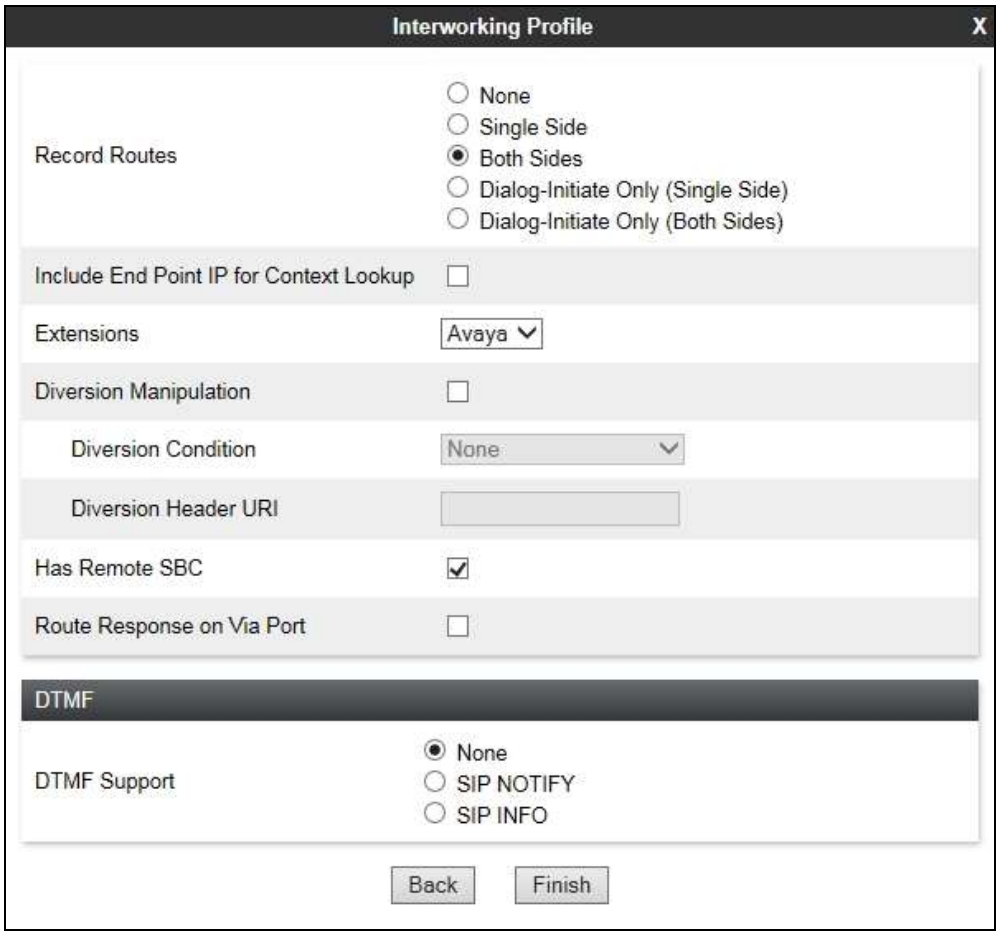


©2016 Avaya Inc. All Rights Reserved.

Click on **Next** and **Next** again to go through the next two dialogue boxes. During testing, these were left at default values.



In the final dialogue box, ensure that **Both Sides** is selected for **Record Routes** and that the **Has Remote SBC** box is checked. Note that although Avaya extensions are shown as selected in the screenshot, they are not supported on the Virgin Media Business SIP Trunk. Click on **Finish**

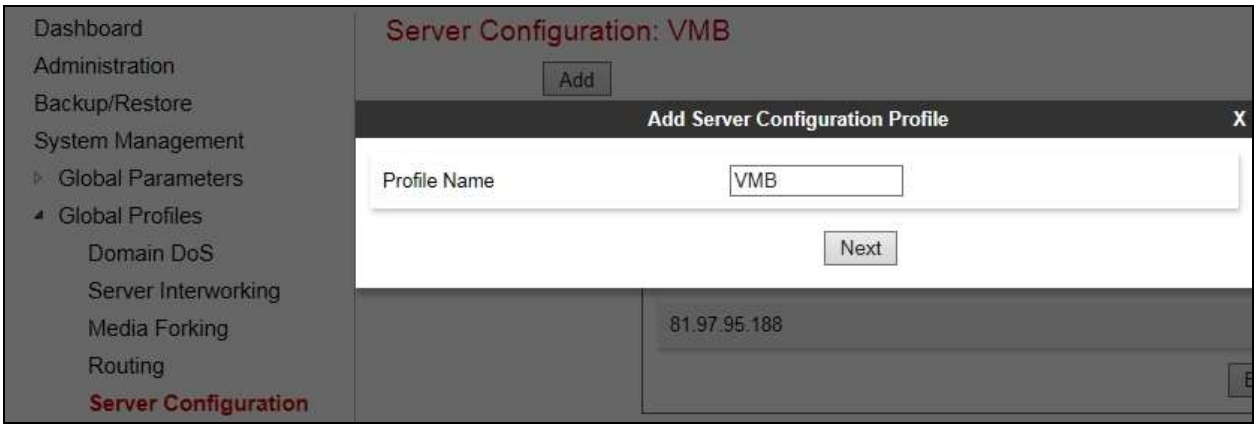


Repeat the process to define Server Interworking for IP Office with the same parameter settings.

©2016 Avaya Inc. All Rights Reserved.

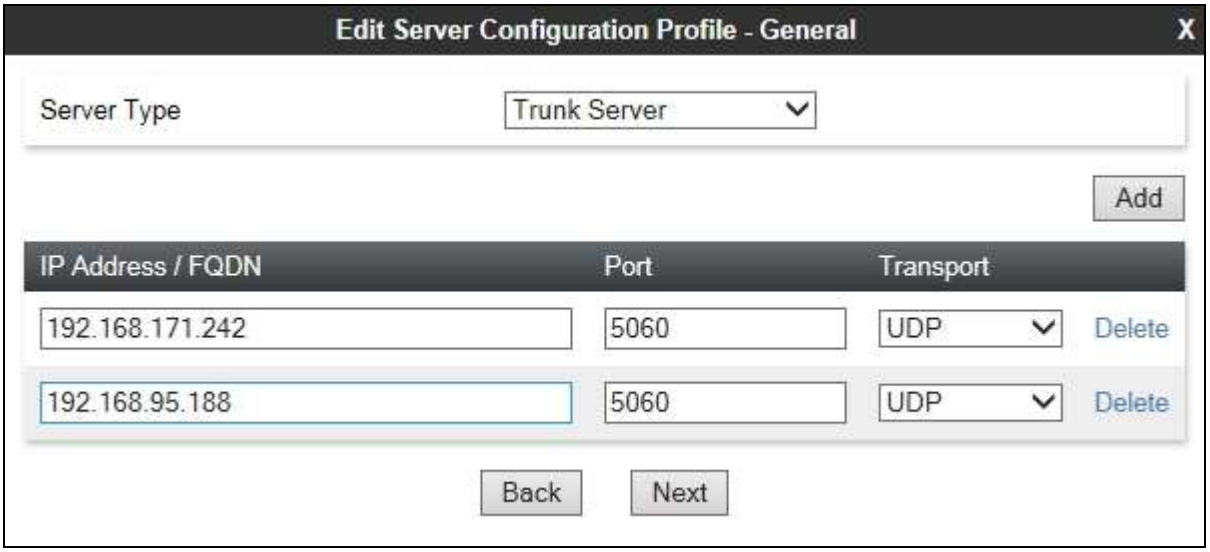
6.5. Define Servers

A server definition is required for each server connected to the Avaya SBCE. To define the Virgin Media Business SIP Trunk Server, navigate to **Global Profiles → Server Configuration** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the pop-up menu.



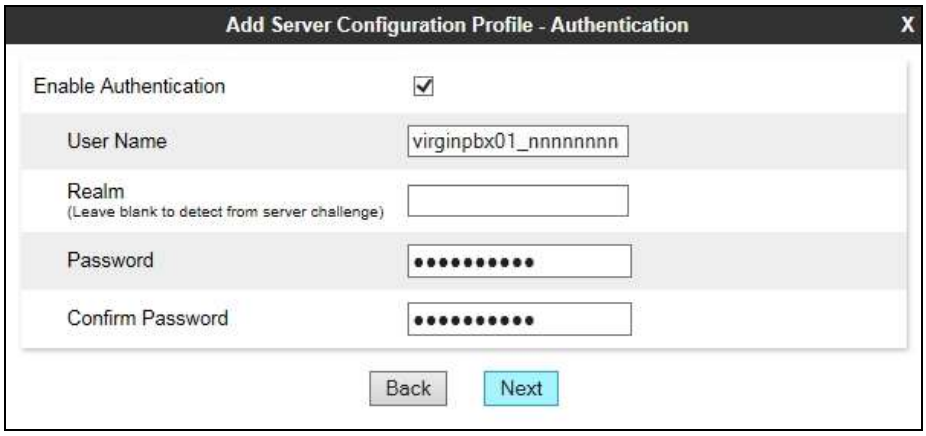
Click on **Next** and enter details in the dialogue box.

- In the **Server Type** drop down menu, select **Trunk Server**.
- Click on **Add** to enter an IP address
- In the **IP Addresses / FQDN** box, type the Virgin Media Business IP address SBC A.
- In the **Port** box, enter the port to be used for the SIP Trunk. This was left blank during testing which defaults to 5060 when UDP is used for transport.
- In the **Transport** drop down menu, select **UDP**.
- Click on **Add** and repeat the process for SBC B.
- Click on **Next**.



©2016 Avaya Inc. All Rights Reserved.

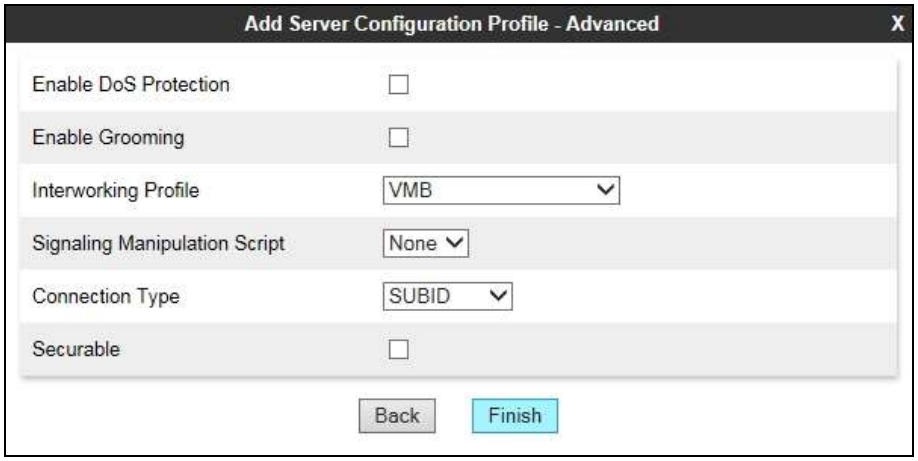
Click on **Next** to define the authentication parameters. In the **Username**, **Password** and **Confirm Password** fields, enter the required information as provided by Virgin Media Business.



Click on **Next** again to get the Heartbeat dialogue box. Leave the fields at default values.

Click on **Next** again to get to the final dialogue box. This contains the **Advanced** settings:

- In the **Interworking Profile** drop down menu, select the **Interworking Profile** for the Virgin Media Business SIP Trunking Service defined in **Section 6.4**.
- Leave the other fields at default settings.
- Click **Finish**.

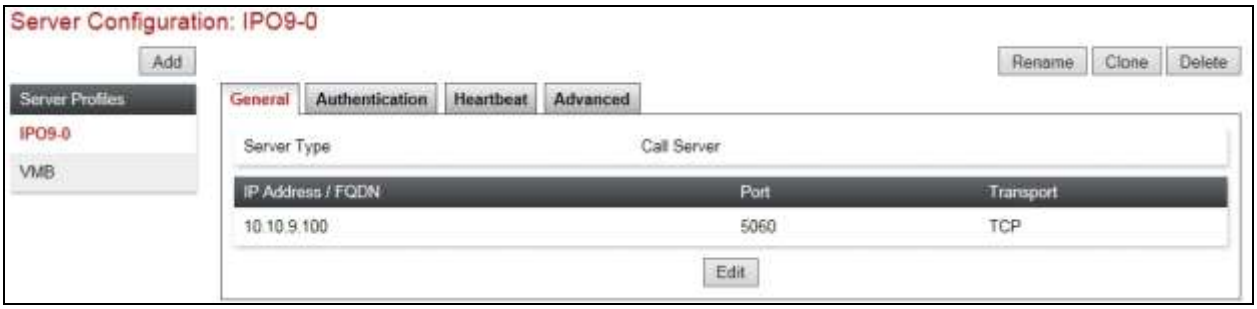


©2016 Avaya Inc. All Rights Reserved.

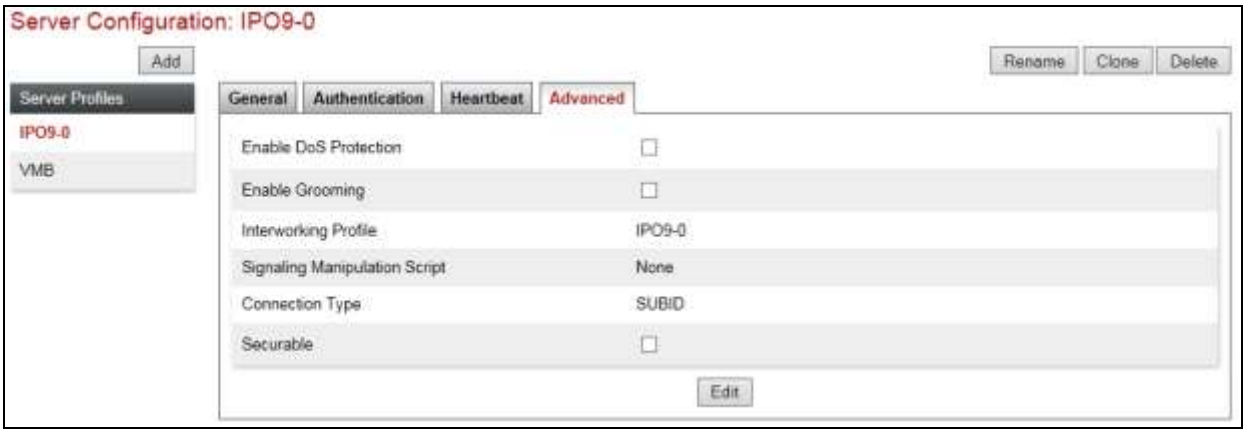
Use the previous process to define the Call Server configuration for IP Office if not already defined.

- Ensure that **Call Server** is selected in the **Server Type** drop down menu in the **General** dialogue box.
- Ensure that the Interworking Profile defined for IP Office in **Section 6.4** is selected in the **Interworking Profile** drop down menu in the Advanced dialogue box

The following screenshot shows the **General** tab of the completed Server Configuration for IP Office:



The next screenshot shows the **Advanced** tab.

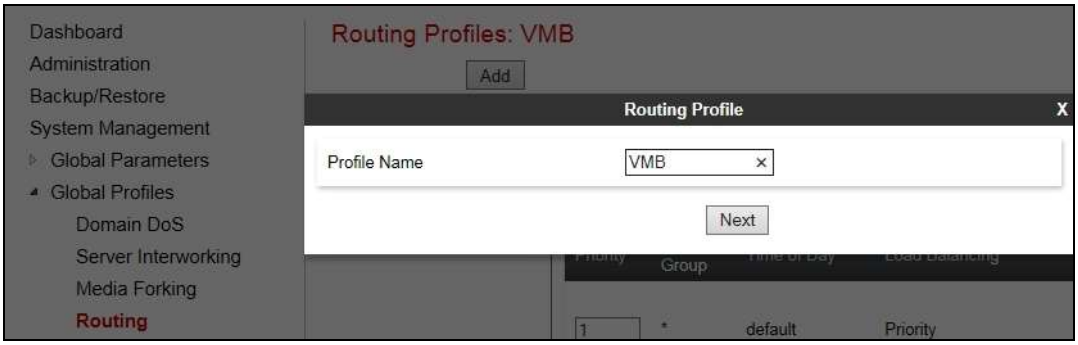


6.6. Define Routing

Routing information is required for routing to the Virgin Media Business SIP Trunking Service on the external side and IP Office on the internal side. The IP addresses and ports defined here will be used as the destination addresses for signalling.

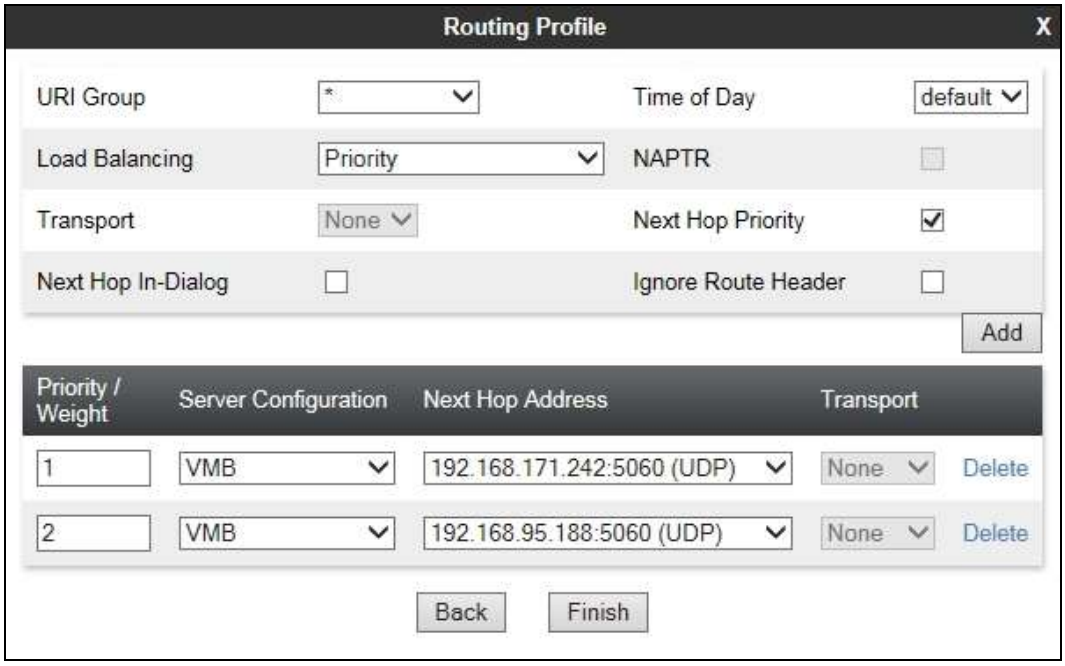
©2016 Avaya Inc. All Rights Reserved.

To define routing to the Virgin Media Business SIP Trunk, navigate to **Global Profiles → Routing** in the main menu on the left hand side. Click on **Add** and enter an appropriate name in the dialogue box.



Click on **Next** and enter details for the Routing Profile for the SIP Trunk:

- During testing, **Load Balancing** was not required and was left at the default value of **Priority**. This resulted in the network SBCs being selected in order of priority
- Click on **Add** to specify the IP address for SBC A.
- Assign a priority in the **Priority / Weight** field, during testing a value of **1** was used for SBC A so that it was selected as the first choice destination for signalling.
- Select the Server Configuration defined in **Section 6.5** in the **Server Configuration** drop down menu and select the **Next Hop Address** field for SBC A
- Click on **Add** again to repeat the process for SBC B. Note that a **Priority / Weight** value of **2** was used for SBC-B so that it was only selected when SBC A was unavailable.
- Click **Finish**.



©2016 Avaya Inc. All Rights Reserved.

Repeat the process for the Routing Profile for IP Office: The following screenshot shows the completed configuration:

6.7. 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 or external interfaces.

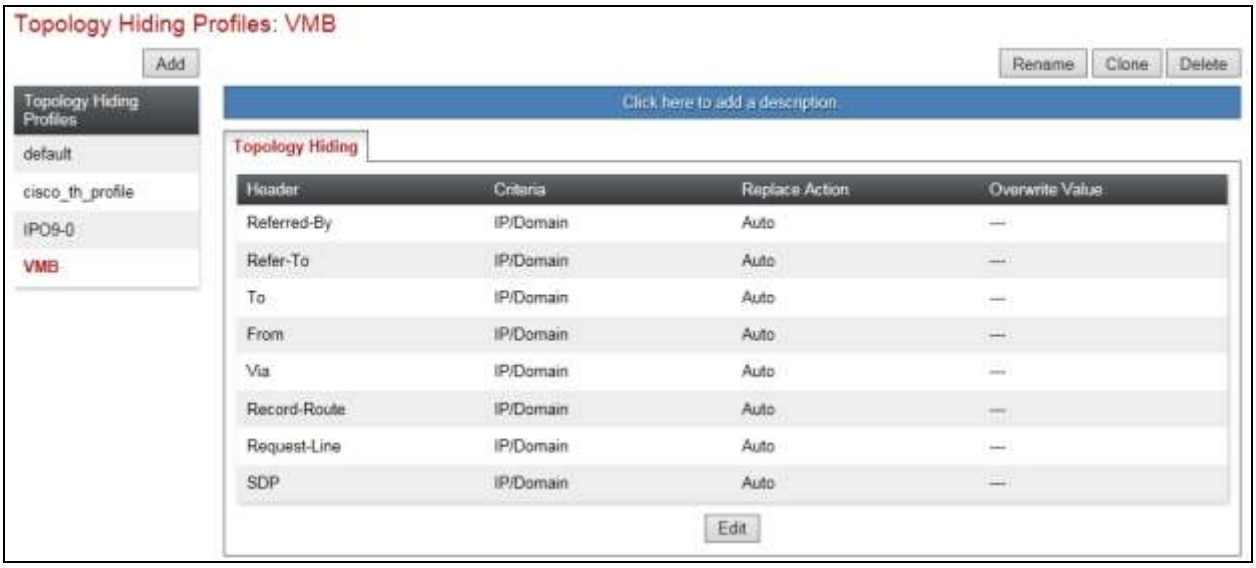
To define Topology Hiding for the Virgin Media Business SIP Trunking Service, navigate to **Global Profiles → Topology Hiding** in the main menu on the left hand side. Click on **Add** to bring up a dialogue box, assign an appropriate name and click on **Next** to configure Topology Hiding for each header as required:

©2016 Avaya Inc. All Rights Reserved.

Enter details in the **Topology Hiding Profile** pop-up menu.

- Click on **Add Header** and **s**elect from the **Header** drop down menu.
- Select **IP** or **IP/Domain** from the **Criteria** drop down menu depending on requirements. During testing default settings of **IP/Domain** were used so that both IP addresses and domain names were hidden.
- Leave the **Replace Action** at the default value of **Auto** unless a specific domain name is required. In this case, select **Overwrite** and define a domain name in the **Overwrite Value** field.
- Topology hiding was defined for all headers where the function is available.

The following screenshot shows the completed **Topology** Hiding configuration for the Virgin Media Business SIP Trunking Service.



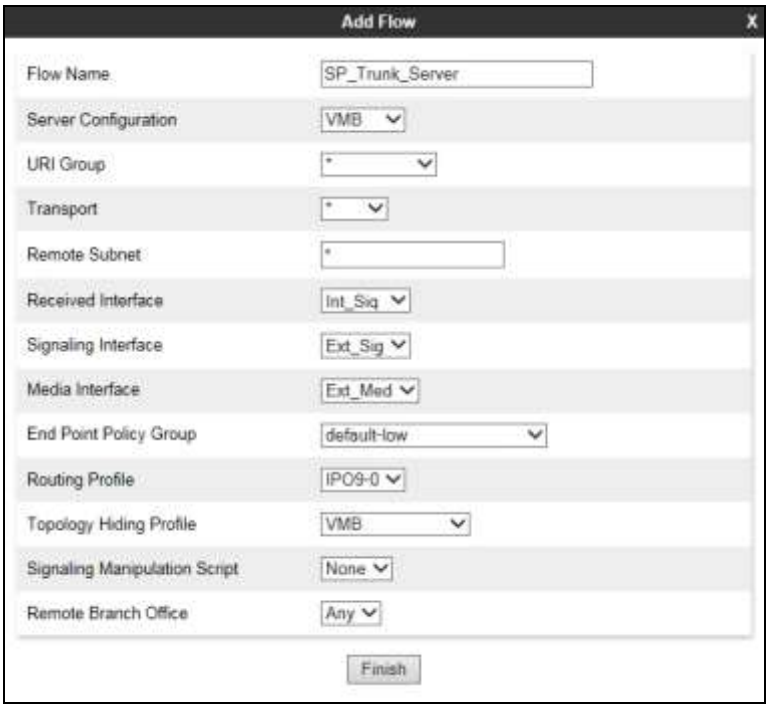
To define Topology hiding for IP Office, follow the same process. This can be simplified by cloning the profile defined for the Virgin Media Business SIP Trunking Service. Do this by highlighting the profile defined for the Virgin Media Business and clicking on **Clone**. Enter an appropriate name for IP Office and click on Next. Make any changes where required, in the test environment the settings were left at the same values.

©2016 Avaya Inc. All Rights Reserved.

6.8. Server Flows

Server Flows combine the previously defined profiles into two End Point Server Flows, one for the Virgin Media Business SIP Trunking Service and the other for IP Office. These End Point Server Flows allow calls to be routed from IP Office to the Virgin Media Business SIP Trunk and vice versa. To define a Server Flow for the Virgin Media Business SIP Trunk, navigate to **Device Specific Settings → End Point Flows**.

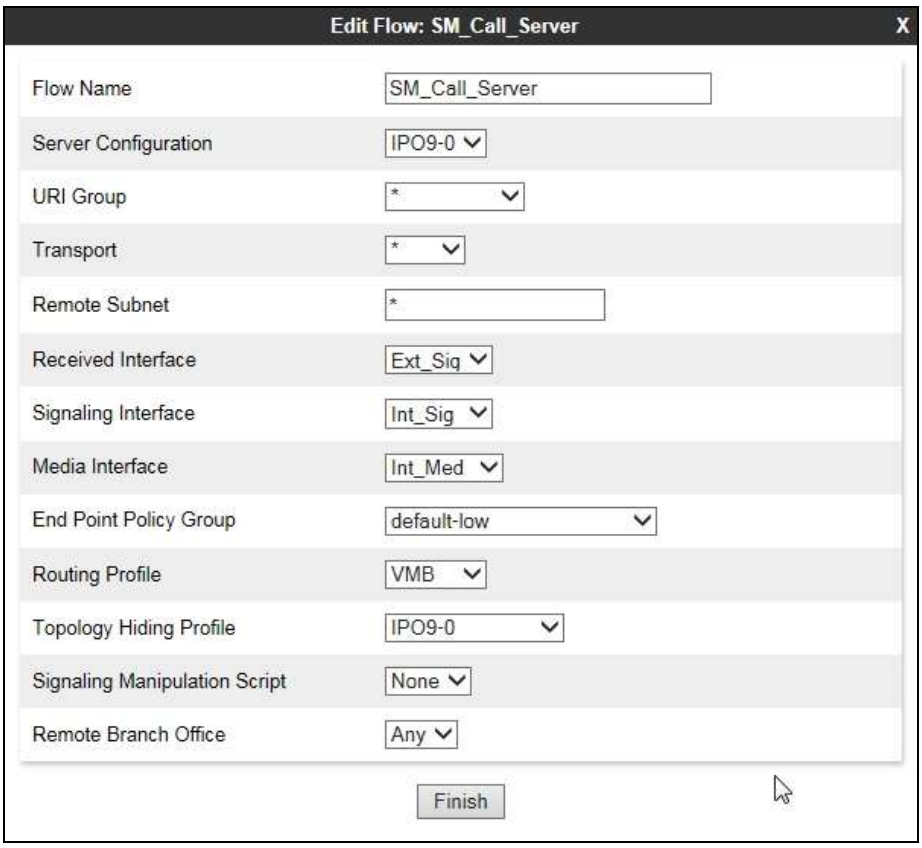
- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for the server flow for the Virgin Media Business SIP trunk, in the test environment **SP_Trunk_Server** was used.
- In the **Server Configuration** field drop-down menu, select the Virgin Media SIP Trunk server defined in **Section 6.5**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 6.3.1**. This is the interface that signalling bound for the SIP Trunk is received on.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.3.1**. This is the interface that signalling bound for the SIP Trunk is sent on.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 6.3.2**. This is the interface that media bound for the SIP Trunk is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of IP Office defined in **Section 6.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the Virgin Media Business SIP Trunking Service defined in **Section 6.7** and click **Finish**.



©2016 Avaya Inc. All Rights Reserved.

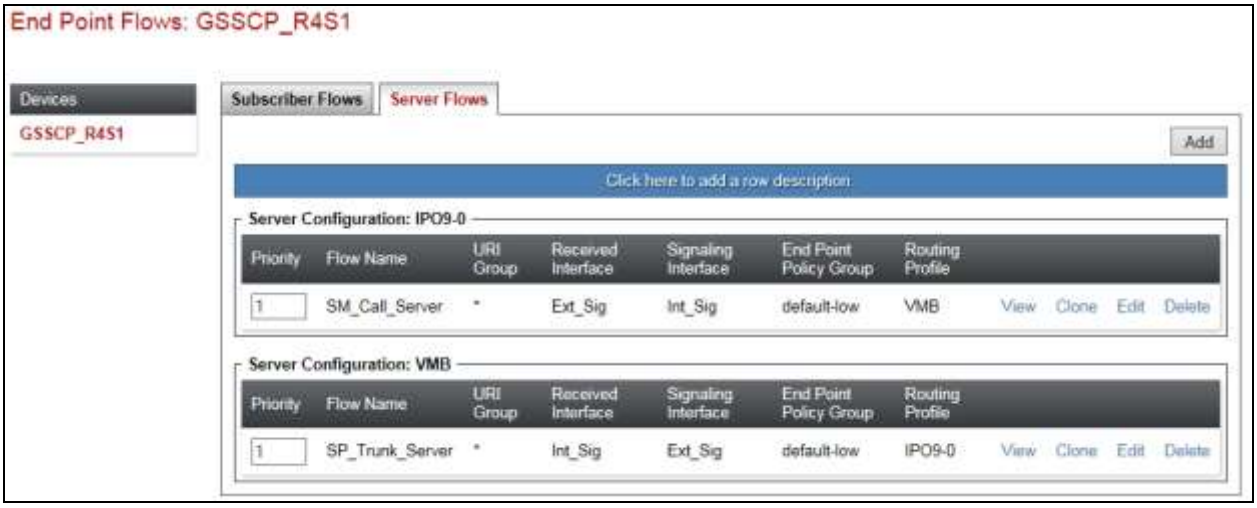
To define a Server Flow for IP Office, navigate to **Device Specific Settings → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Flow Name** field enter a descriptive name for server flow for IP Office, in the test environment **SM_Call_Server** was used.
- In the **Server Configuration** field drop-down menu, select the IP Office server defined in **Section 6.5**.
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.3.1**. This is the interface that signalling bound for IP Office is received on.
- In the **Signaling Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 6.3.1**. This is the interface that signalling bound for IP Office is sent on.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 6.3.2**. This is the interface that media bound for IP Office is sent on.
- In the **Routing Profile** drop-down menu, select the routing profile of the Virgin Media Business SIP Trunking Service defined in **Section 6.6**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of IP Office defined in **Section 6.7** and click **Finish**.



©2016 Avaya Inc. All Rights Reserved.

The information for all Server Flows is shown on a single screen on the Avaya SBCE.



7. Virgin Media Business SIP Trunking Configuration

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

- Domain Name or IP address of Virgin Media Business 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.

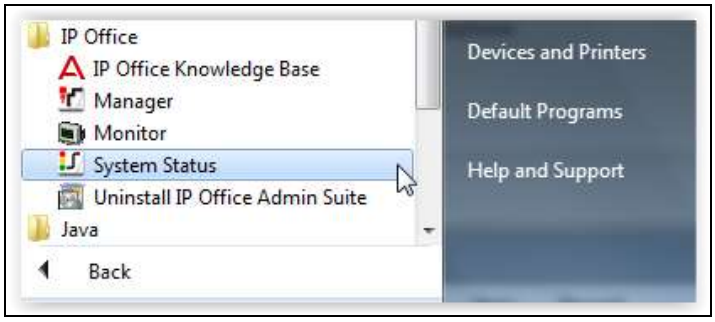
©2016 Avaya Inc. All Rights Reserved.

8. Verification Steps

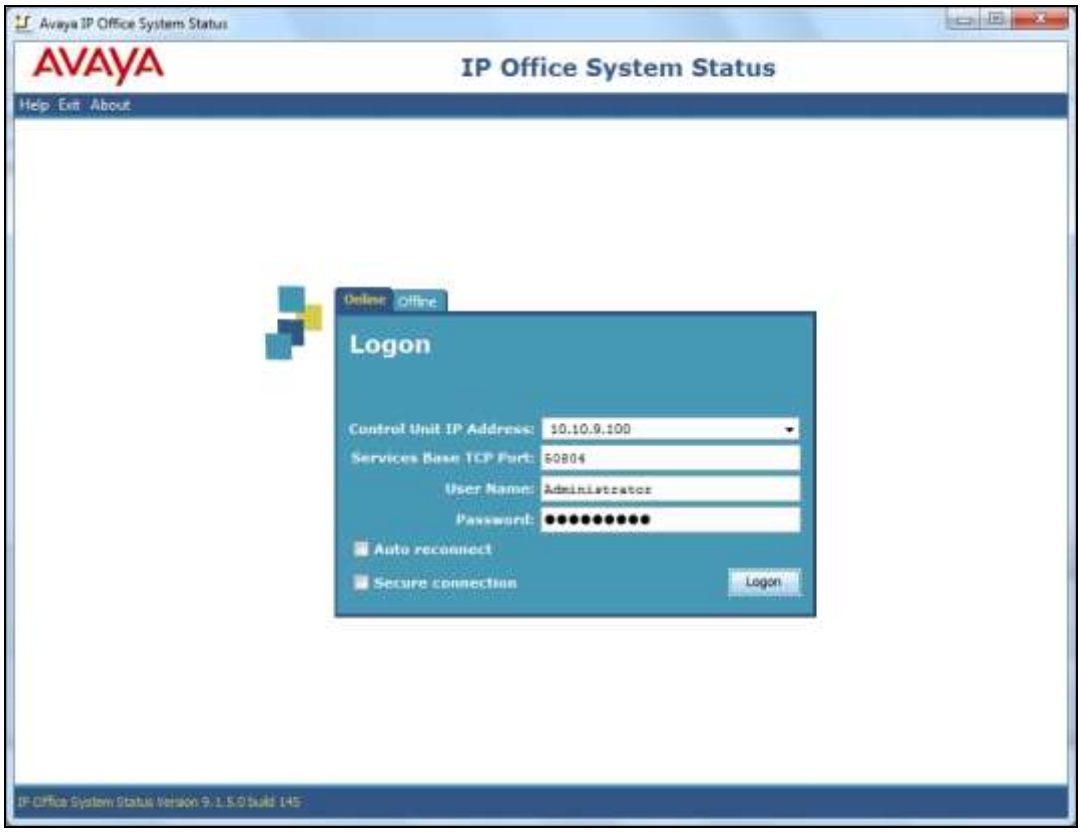
This section includes steps that can be used to verify that the configuration has been done correctly.

8.1. SIP Trunk status

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

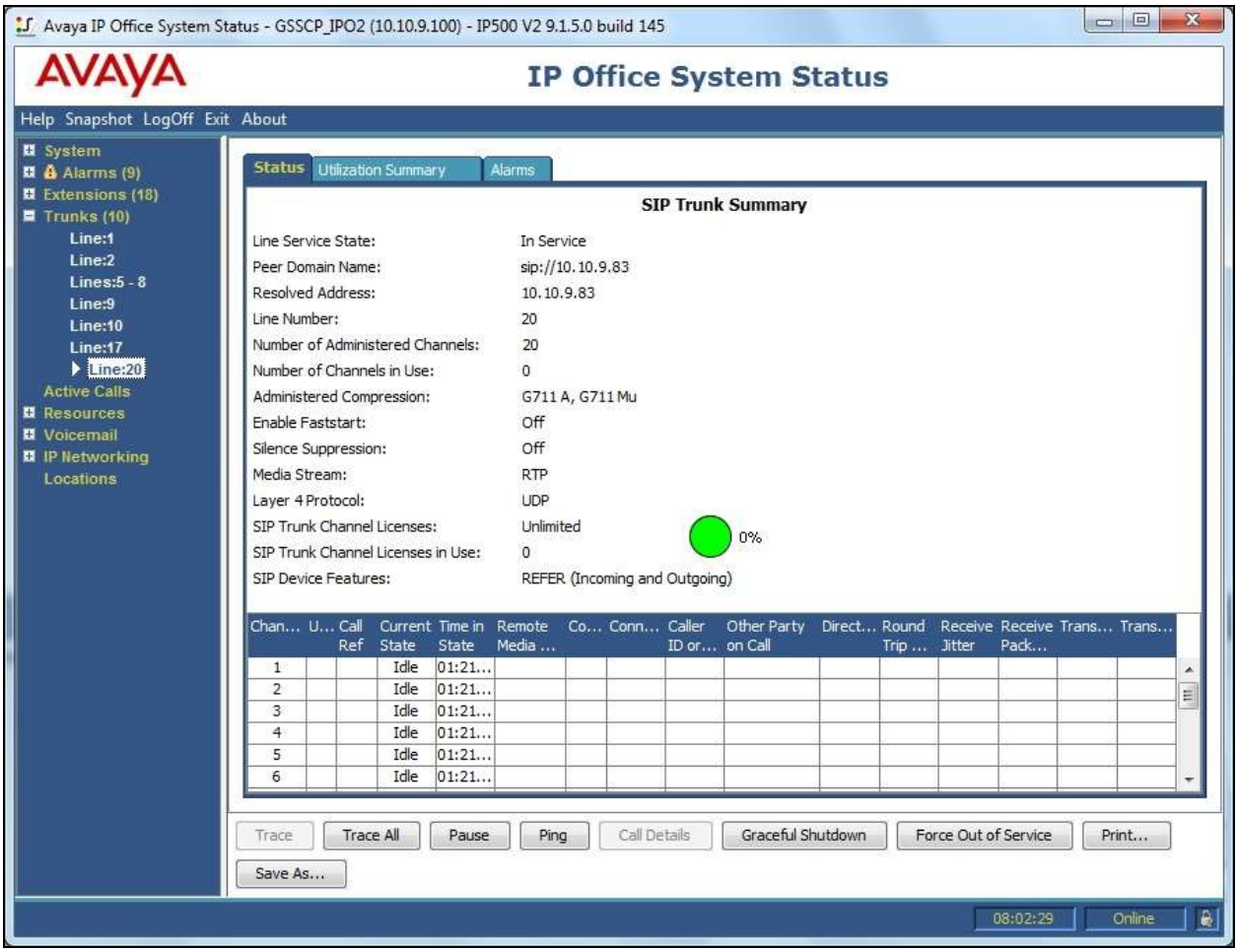


Log in to IP Office System Status at the prompt using the **Control Unit IP Address** for the IP Office. The **User Name** and **Password** are the same as those used for IP Office Manager.



©2016 Avaya Inc. All Rights Reserved.

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.

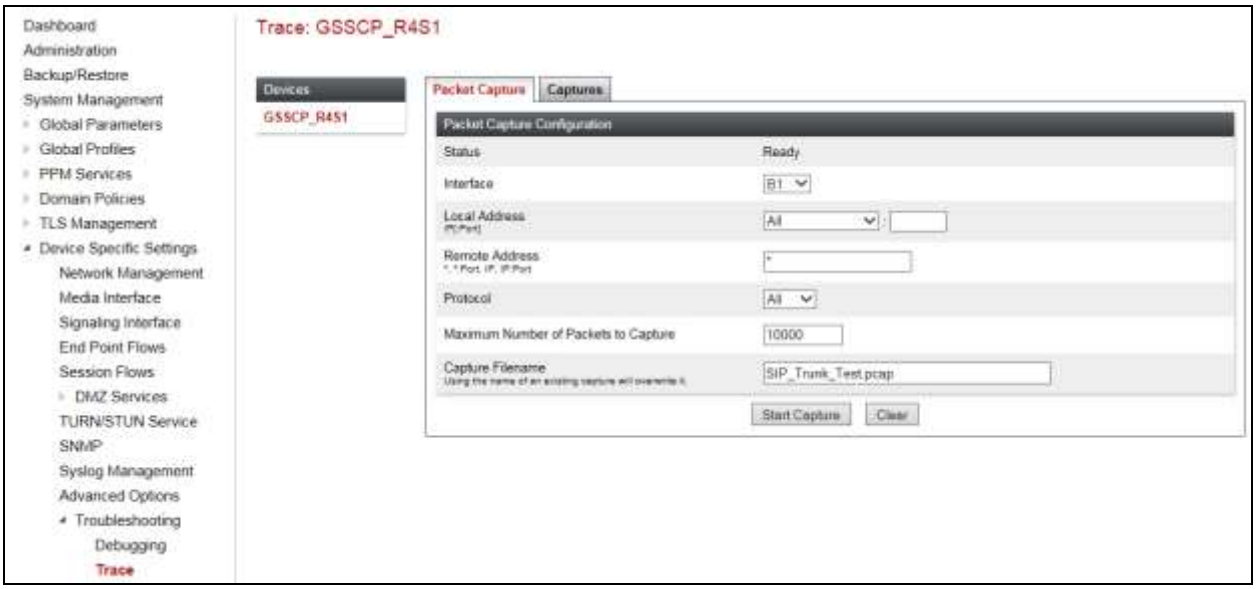


Should issues arise with the SIP trunk, use the Avaya SBCE trace facility to check that the OPTIONS requests sent from Session Manager via the Avaya SBCE to the network SBCs are receiving a response.

©2016 Avaya Inc. All Rights Reserved.

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

- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select the signalling interface IP address or **All** from the **Local Address** drop down menu.
- Enter the IP address of the network 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 to OPTIONS in the form of a 200 OK will be seen from the Virgin Media Business SIP Trunking service.

©2016 Avaya Inc. All Rights Reserved.

9. Conclusion

All tests for Virgin Media Business SIP Trunking were completed. Observations for the testing are listed in **Section 2.2**.

©2016 Avaya Inc. All Rights Reserved.

10. Additional References

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

[1] *IP Office 9.1 KnowledgeBase Technical Documentation*, Web based.

[2] *IP Office™ Platform 9.1 Deploying Avaya IP Office™ Platform IP500 V2*, Document number15-601042, 22nd October 2015.

[3] *IP Office™ Platform 9.1 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document number 15-601011, 30th September 2015.

[4] *Administering Avaya IP Office™ Platform with Manager*, Document number 101005673, October 2015

[5] *IP Office™ Platform 9.1 Using System Status*, Document number15-601758, 11th August 2015.

[6] *IP Office™ Platform 9.1 Using IP Office System Monitor*, Document number 15-601019, 19th May 2015.

[7] *Using Avaya Communicator for Windows on IP Office*, Document number 101005858, April 2015.

[8] *IP Office™ Platform 9.1 Third-Party SIP Extension Installation Notes*, Document number 101010620, 14th May 2015.

[9] *Deploying Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[10] *Upgrading Avaya Session Border Controller for Enterprise*, Release 7.0, August 2015

[11] *Administering Avaya Session Border Controller for Enterprise*, Release 7.0, Nov 2015

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

©2016 Avaya Inc. All Rights Reserved.

©2016 Avaya Inc. All Rights Reserved.
Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.