



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring Avaya IP Office Release 9.0 with Avaya Session Border Controller for Enterprise Release 6.2 to support BT Ireland SIP Trunk Service – Issue 1.0**

### **Abstract**

These Application Notes describe the steps for configuring Avaya IP Office R9.0 and the Avaya Session Border Controller for Enterprise 6.2 to support BT Ireland SIP Trunk Service.

The BT Ireland Trunk Service provides PSTN access via a SIP trunk connected to the BT Ireland Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or digital trunks. BT Ireland 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 BT Ireland SIP Trunk service and Avaya IP Office. In the sample configuration, the Avaya IP Office solution consists of an Avaya Session Border Controller for Enterprise Release 6.2, and Avaya IP Office 500 v2 Release 9.0 Essential Edition, Avaya Voicemail Pro, Avaya IP Office Softphone, and Avaya H.323, SIP, digital, and analog 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 BT Ireland SIP Trunk service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

BT Ireland SIP Trunk service provides PSTN access via a SIP trunk connected to the BT Ireland 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 BT Ireland 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.729A codecs
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38
- 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
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning

## 2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for BT Ireland's SIP Trunk service with the following observations:

- When there were no matching codecs in the SDP offer of an outbound call, "408 Service Unavailable" response was returned from the BT Ireland 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

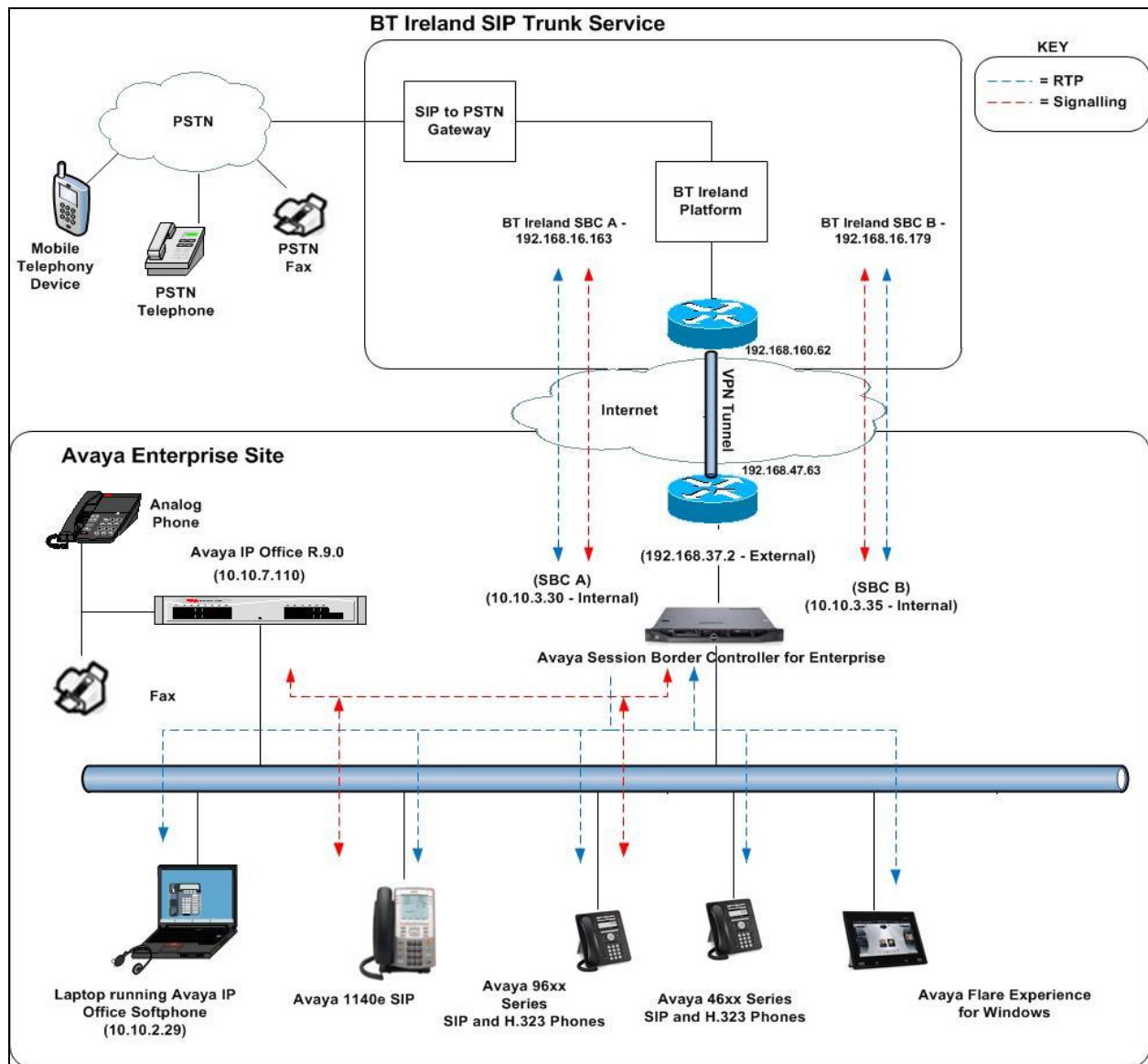
For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on BT Ireland products described in these Application Notes, please contact BT Ireland Customer Support at:

- Telephone: 1800 924 924
- Telephone: +353 1 4328846

### 3. Reference Configuration

**Figure 1** below illustrates the test configuration. The test configuration shows an enterprise site connected to the BT Ireland 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 SIP firmware), Avaya 1140e SIP Telephones, Avaya 2420 Digital Telephone, 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 an IP Office Softphone client and Flare Experience for Windows for mobility testing. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.



**Figure 1: Test Setup BT Ireland 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
<b>Avaya</b>	
Avaya Session Border Controller for Enterprise	Version 6.2.1.Q18
Avaya IP Office 500 V2	Version 9.0.4.0 build 965
Avaya 1603 Phone (H.323)	1.3.5
Avaya 9600 Series Phone (SIP)	6.3.0
Avaya SoftPhone (SIP)	3.056516
Avaya Flare Experience for Windows (SIP)	1.1.3.14
Avaya 1140e (SIP)	FW: 04.04.10.00.bin
Avaya 2420 Digital Phone	R6.0
Avaya 98390 Analogue Phone	N/A
<b>BT Ireland</b>	
BT Ireland SIP Trunk	Genband Q20 SBC R.8.3.2.0 Genband Experius Call Server R.17.0.2.15

## 5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the BT Ireland 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 BT Ireland.

System (1)	VMPro Recordings Administrators	Fq1@hobLd5aWzyc6eagw5Hb23woEx	255	Valid	Never	ADI Nodal
Line (9)	VMPro Outlook Interface	54m9l3k29v76DvWmu1_t8OpZz_PfXJ	255	Valid	Never	ADI Nodal
Control Unit (4)	VMPro TTS (ScanSoft)	SacVZavJDGI45PoVhZpLgHXNeaASNm	255	Valid	Never	ADI Nodal
Extension (31)	VMPro TTS (Generic)	2Uc@Z0qVPluZFu7QkErgko_e5c9Hxm	255	Valid	Never	ADI Nodal
User (31)	Conferencing Center	TGKkall3g5D2GwEQExog4kX8VfFe	255	Obsolete	Never	ADI Nodal
Group (1)	Small Office Edition VCM (channels)	cvHQ4uoe5l5V3gJPpyre1JGOCrospxu	255	Obsolete	Never	ADI Nodal
Short Code (79)	Small Office Edition WiFi	CttbR@gkEtQoakMjYr_wLCoz9TlmpCDJ	255	Obsolete	Never	ADI Nodal
Service (0)	IPSec Tunneling	MUMda@vDstN4V_yaz_0L2pmDij@w8e1	255	Valid	Never	ADI Nodal
RAS (1)	Proactive Reporting	@GTgg@b2xJ38Jg3n781Zbp1qLKdxFC	255	Valid	Never	ADI Nodal
Incoming Call Route (6)	Report Viewer	nXtL@dhfPNUaO7qhv9V_T7JA1ieYfzM	255	Valid	Never	ADI Nodal
Directory (0)	Mobility Features	9GMw6EgcDNRxt0LqFxbRdYGOIswABDC	255	Obsolete	Never	ADI Nodal
Time Profile (0)	Advanced Small Community Networking	Q4OzWky8AsqZyRQb1Wzgr6gXod@gJb1	255	Obsolete	Never	ADI Nodal
Firewall Profile (1)	IP500 Voice Networking Channels	x3IZTUsetWu@3TSwK_c4qr_3ghVfVfLm	255	Valid	Never	ADI Nodal
IP Route (7)	IP500 Upgrade Standard to Profession...	KT11qHieQlzpI854CA9kyZHNqGyZQZ	255	Obsolete	Never	ADI Nodal
Account Code (0)	IP500 Voice Networking Channels	KqTHPCvedI5rd1kwoXo01KEd9vOTp9B	4	Valid	Never	ADI Nodal
License (76)	VCM Channel Migration	@qww79ongGHFDU3SLHmECeOLALkDe	255	Valid	Never	ADI Nodal
Tunnel (0)	SIP Trunk Channels	yyWlB2V8Pmb1m13L1MFhNjv1ueSG8@J	255	Valid	Never	ADI Nodal
User Rights (8)	VPN IP Extensions	In2R0v99Dcbt48Pa7c16275V9PREm	255	Obsolete	Never	ADI Nodal
Auto Attendant (1)	IP500 Universal PRI (Additional chan...	@9@b1gbydvDqL97hc_EHOUxVVRAPs9	255	Valid	Never	ADI Nodal
ARS (1)	RAS LRQ Support (Rapid Response)	_1TOHD@H9Id9kRcIGCWx9PbDaFd7_x	255	Valid	Never	ADI Nodal
RAS Location Request (0)	IP Office Dealer Support - Standard E...	Ek28bu9YPIQyXhCh6KPeqMpWQe4dPEW	255	Valid	Never	ADI Nodal
Location (0)	IP Office Dealer Support - Profession...	sIn5JU@ZVUyglELFtsMng@Eze5OGAmB	255	Valid	Never	ADI Nodal
	IP Office Distributor Support - Stand...	qVbJCvY9dmf13WbQOpZLsgw13xNAej9	255	Valid	Never	ADI Nodal
	IP Office Distributor Support - Profes...	bAtvbZmhddda2o6_4Up6Bgf1Ejas7IM	255	Valid	Never	ADI Nodal
	UMS Web Services	3XnHkomVDSWbnEewfjgc_mTRg1LZ6wPc9	255	Valid	Never	ADI Nodal
	CCR SUP	YIzcnWmJdJRZUGQdfGE9m1Foj9e4EhbB	255	Valid	Never	ADI Nodal
	Customer Service Agent	g19PvPmDvVZDjOrfHhMkocddsk@xSPES	255	Valid	Never	ADI Nodal
	CCR Designer	Ay2PHfDX2ebJCh9cjo7YVwQFer20m	255	Valid	Never	ADI Nodal
	CCR CCC UPG	ZycnqhbQkhd38gtycwksyl9L2GgMc	255	Valid	Never	ADI Nodal
	1600 Series Phones	HcyTL5UL4dh9Kuz6x17MNSDkA1uu	255	Valid	Never	ADI Nodal
	Third Party API	8hxD_VWgQd10IWSGpPzwdRKYvDKPwse	255	Valid	Never	ADI Nodal
	one-X Portal for IP Office	bnHC3dVWQdJ_yLRFF_x8ckOMSH26GR_3	255	Valid	Never	ADI Nodal
	Avaya IP endpoints	8413qySRgY4GnpQ26WUpvLgDMV7EYem	255	Valid	Never	ADI Nodal

## 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 sample 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).

GSSCP\_IPO9

System
LAN1
LAN2
DNS
Voicemail
Telephony
Directory Services
System Events
SMTP
SMDR
Twinning
VCM
CCR
Codecs

LAN Settings
VoIP
Network Topology

IP Address
10 . 10 . 7 . 110
IP Mask
255 . 255 . 255 . 0
Primary Trans. IP Address
0 . 0 . 0 . 0
RIP Mode
None
☐ Enable NAT
Number Of DHCP IP Addresses
200
DHCP Mode
☐ Server
☐ Client
☐ Dialin
☒ Disabled
Advanced

On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. 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).

The screenshot displays the GSSCP\_IP09 configuration window. The 'VoIP' tab is selected, showing various SIP-related settings. The 'RTP' section is expanded, showing port ranges for RTP and NAT. The 'DiffServ Settings' section is also expanded, showing DSCP and SIG DSCP values. Red boxes highlight the 'SIP Trunks Enable' and 'SIP Registrar Enable' checkboxes, the 'RTP' section, and the 'DiffServ Settings' section.

**GSSCP\_IP09**

System | LAN1 | LAN2 | DNS | Voicemail | Telephony | Directory Services | System Events | SMTP | SMDR | Twinning | VCM | CCR | Codecs

LAN Settings | **VoIP** | Network Topology

☒ H323 Gatekeeper Enable  
☐ Auto-create Extn ☐ Auto-create User ☐ H323 Remote Extn Enable

☒ **SIP Trunks Enable**  
☒ **SIP Registrar Enable**  
☐ Auto-create Extn/User ☐ SIP Remote Extn Enable

Domain Name: avaya.com

Layer 4 Protocol: ☒ UDP ☒ TCP ☐ TLS

UDP Port: 5060 TCP Port: 5060 TLS Port: 5061

Remote UDP Port: 5060 Remote TCP Port: 5060 Remote TLS Port: 5061

Challenge Expiry Time (secs): 10

**RTP**

Port Number Range  
 Minimum: 49152 Maximum: 53246

Port Number Range (NAT)  
 Minimum: 49152 Maximum: 53246

☐ Enable RTCP Monitoring on Port 5005

**Keepalives**  
 Scope: Disabled Periodic timeout: 5  
 Initial keepalives: Enabled

**DiffServ Settings**

B8 DSCP(Hex) B8 Video DSCP(Hex) FC DSCP Mask (Hex) 88 SIG DSCP (Hex)  
 46 DSCP 46 Video DSCP 63 DSCP Mask 34 SIG DSCP



Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to “None” in **Section 5.6**. It is important that the **Binding Refresh Time** is set to the correct value. Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. Below is a sample configuration. On completion, click the **OK** button (not shown).

The screenshot shows the 'GSSCP\_IP09\*' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following settings:

- STUN Server Address: 0.0.0.0
- STUN Port: 3478
- Firewall/NAT Type: Unknown
- Binding Refresh Time (seconds): 200
- Public IP Address: 0 . 0 . 0 . 0
- Run STUN button and Cancel button

The 'Public Port' section contains the following settings:

- UDP: 0
- TCP: 0
- TLS: 0

At the bottom, there is a checkbox labeled 'Run STUN on startup' which is currently unchecked.



### 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).

The screenshot shows the GSSCP\_IP09\* configuration window with the 'Telephony' tab selected. The 'Companding Law' section is highlighted with a red box, showing 'A-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is also highlighted with a red box and is unchecked. Other settings visible include 'Default Outside Call Sequence' set to 'Normal', 'Default Inside Call Sequence' set to 'Ring Type 1', 'Default Ring Back Sequence' set to 'Ring Type 2', and 'Restrict Analogue Extension Ringer Voltage' set to 'No'.

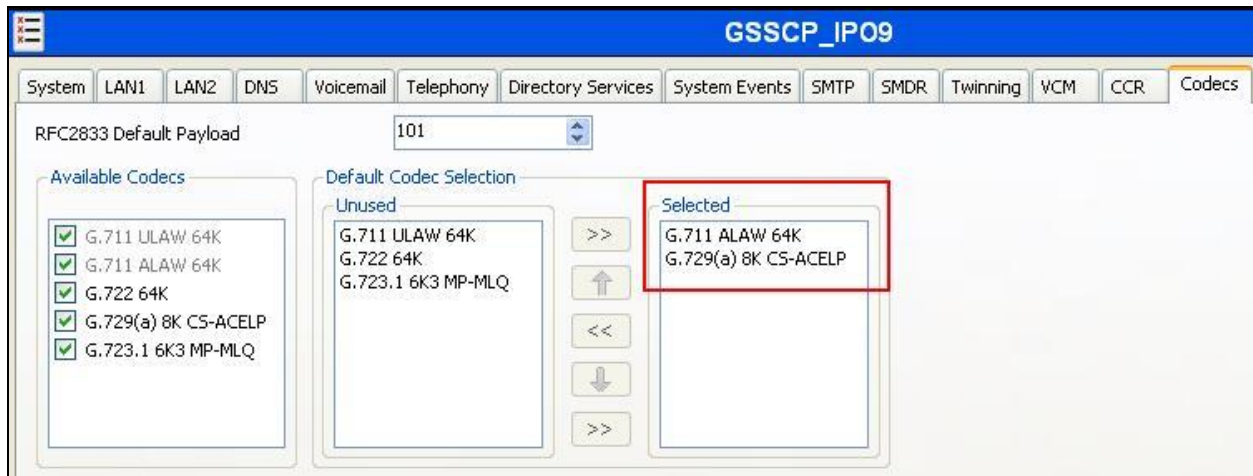
### 5.4. System Twinning Settings

To view or change Twinning settings, select the **Twining** 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 BT Ireland SIP Trunk.

The screenshot shows the GSSCP\_IP09 configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked, and the 'Calling party information for Mobile Twinning' field is empty.

## 5.5. Codec Settings

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

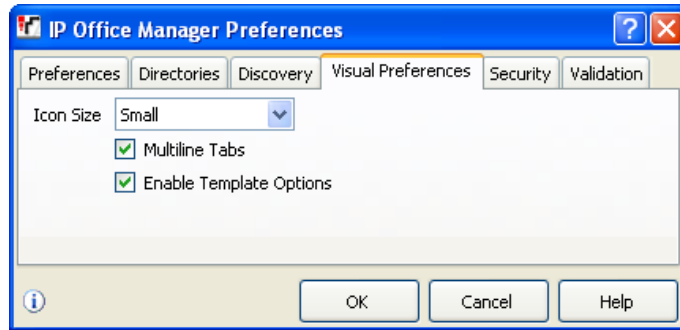
Also, the following SIP Line settings are not supported on Basic Edition:

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

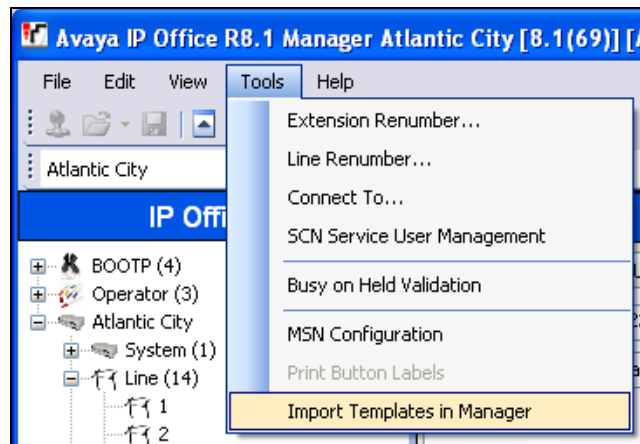
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

### 5.6.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **IE\_ BT Ireland \_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**.

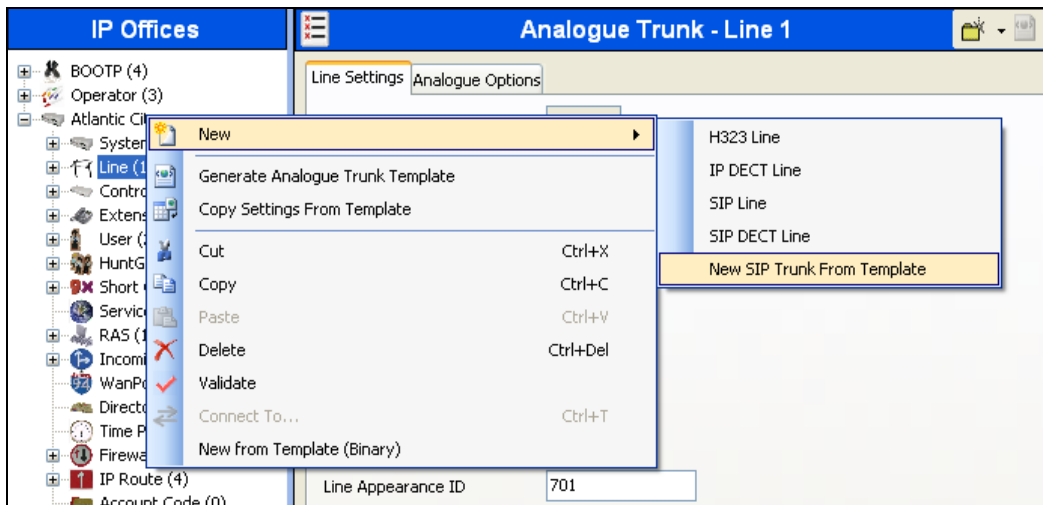


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



In the pop-up window (not shown) that appears, select the directory where the template file was copied 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 **Ireland** from the **Country** pull-down menu and select **BT\_Ireland** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**IE\_BT\_Ireland\_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



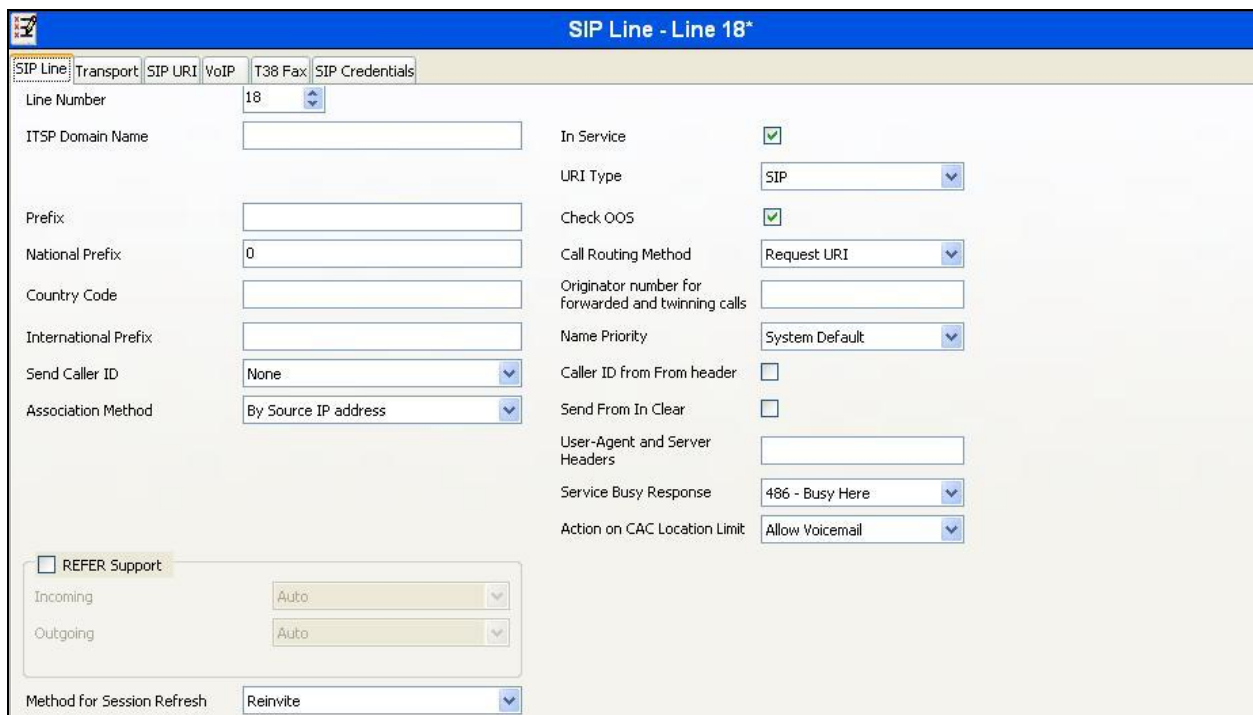
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.6.2**.

## 5.6.2. SIP Line – SIP Line Tab

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 **REFER Support** is unchecked as Zen Internet does not support **REFER**.
- Set **Method for Session Refresh** to **Reinvite**.
- Ensure the **In Service** box is checked.
- Default values may be used for all other parameters.

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



**SIP Line - Line 18\***

**SIP Line** | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials

Line Number: 18

ITSP Domain Name:

Prefix:

National Prefix: 0

Country Code:

International Prefix:

Send Caller ID: None

Association Method: By Source IP address

In Service: ☒

URI Type: SIP

Check OOS: ☒

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

Name Priority: System Default

Caller ID from From header: ☐

Send From In Clear: ☐

User-Agent and Server Headers:

Service Busy Response: 486 - Busy Here

Action on CAC Location Limit: Allow Voicemail

☐ **REFER Support**

Incoming: Auto

Outgoing: Auto

Method for Session Refresh: Reinvite

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).

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field is set to '10.10.3.30,10.10.3.35'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'TCP', 'Send Port' is '5060', 'Use Network Topology Info' is 'None', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '0.0.0.0' and '0.0.0.0'. 'Calls Route via Registrar' is checked. The 'Separate Registrar' field is empty.

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

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'SIP URI' tab selected. A table with columns 'Channel', 'Groups', 'Via', 'Local URI', 'Contact', 'Display Name', 'PAI', 'Credential', and 'Max Calls' is visible. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'. The 'Add...' button is highlighted with a red rectangle.



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 (Registration /Authentication to the BT SIP trunk is handled and configured on the Avaya SBCE. The configuration details are outlined in **Section 6.2.5**.
- 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.

**SIP Line - Line 18**

Channels: SIP Line | Transport | **SIP URI** | VoIP | T38 Fax | SIP Credentials

Channel	Groups	Via	Local URI	Contact	Display Name	PAI
1	18 18	<...>				

Buttons: Add..., Remove, Edit...

**Edit Channel**

Via: <None>

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 10

Buttons: OK, Cancel

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

- Select **Custom** 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 **T.38**.
- 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 BT Ireland network.
- Default values may be used for all other parameters.

The screenshot displays the 'SIP Line - Line 18\*' configuration window. The 'VoIP' tab is selected, showing a 'Codec Selection' section with 'Custom' chosen from a dropdown. Below this, two lists of codecs are shown: 'Unused' (G.711 ULAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ) and 'Selected' (G.711 ALAW 64K, G.729(a) 8K CS-ACELP). Arrows indicate the movement of codecs between these lists. To the right, several checkboxes are visible: 'VoIP Silence Suppression' (unchecked), 'Allow Direct Media Path' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'PRACK/100rel Supported' (checked), 'Force direct media with phones' (unchecked), and 'G.711 Fax ECAN' (unchecked). At the bottom, four fields are highlighted with red boxes: 'Fax Transport Support' set to 'T38', 'Location' set to 'Cloud', 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'.

Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the Use Default Values box (not shown) and select **0** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate (bps)** to **14400**. All other field may retain their default values. On completion, click the **OK** button (not shown).

The screenshot shows the 'SIP Line - Line 18\*' configuration window with the 'T38 Fax' tab selected. The window has a blue title bar and a tabbed interface with tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'T38 Fax' tab is active, displaying various configuration options. On the left, there are settings for 'T38 Fax Version' (set to 0), 'Transport' (set to UDPTL), a 'Redundancy' section with 'Low Speed' and 'High Speed' (both set to 0), 'TCF Method' (set to Trans TCF), 'Max Bit Rate (bps)' (set to 14400), 'EFlag Start Timer (msecs)' (set to 2600), 'EFlag Stop Timer (msecs)' (set to 2300), and 'Tx Network Timeout (secs)' (set to 150). On the right, there are checkboxes for 'Scan Line Fix-up' (checked), 'TFOP Enhancement' (checked), 'Disable T30 ECM' (unchecked), 'Disable EFlags For First DIS' (unchecked), 'Disable T30 MR Compression' (unchecked), and 'NSF Override' (unchecked). Below these are 'Country Code' and 'Vendor Code' fields, both set to 0.

**Note:** It is advisable at this stage to save the configuration as described in **Section 5.11**.

## 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).

9N;; Dial	
Short Code	
Code	9N;;
Feature	Dial
Telephone Number	N
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

For incoming calls from mobility extension to FNE features hosted by IP Office to provide dial tone or mobile callback functionalities, Short Code **FNE00** was created.

- In the **Code** field, enter the FNE feature code as **FNE00**.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00** for **FNE00**.
- 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**.

The screenshot shows the configuration window for a Short Code named 'FNE00: FNE Service'. The window has a blue title bar and a toolbar with icons for help, save, delete, and confirm. The main area contains a form with the following fields:

Short Code	
Code	FNE00
Feature	FNE Service
Telephone Number	00
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

A red rectangular box highlights the 'Code', 'Feature', 'Telephone Number', and 'Line Group ID' fields.

## 5.8. User s 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 screenshot shows the 'SIP Extension: 8003 89107' configuration window with the 'Extn' tab selected. The fields are as follows:

Field	Value
Extension Id	8003
Base Extension	89107
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Avaya 1140E SIP
Location	Automatic
Module	0
Port	0
Force Authorization	<input checked="" type="checkbox"/>

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. Check the **Reserve Avaya IP endpoint license** box. 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.

The screenshot shows the 'SIP Extension: 8003 89107' configuration window with the 'VoIP' tab selected. The fields are as follows:

Field	Value
IP Address	0 . 0 . 0 . 0
Codec Selection	System Default
Reserve License	None
Fax Transport Support	None
TDM->IP Gain	Default
IP->TDM Gain	Default
DTMF Support	RFC2833

On the right side, the following checkboxes are present:

- ☐ VoIP Silence Suppression
- ☐ Local Hold Music
- ☒ Allow Direct Media Path
- ☒ Re-invite Supported
- ☐ Codec Lockdown

The 'Codec Selection' section shows a list of codecs: 'Unused' (G.711 ULAW 64K, G.722 64K) and 'Selected' (G.711 ALAW 64K, G.729(a) 8K C5-ACELP).

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.

The screenshot shows the 'User' configuration page for 'Ext89107: 89107\*'. The 'User' tab is selected, showing various configuration fields and checkboxes.

Field	Value
Name	Ext89107
Password	*****
Confirm Password	*****
Account Status	Enabled
Full Name	Ext 89107
Extension	89107
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Power User

Below the profile selection, there are several checkboxes for additional features:

- ☐ Receptionist
- ☐ Enable Softphone
- ☐ Enable one-X Portal Services
- ☒ Enable one-X TeleCommuter
- ☒ Enable Remote Worker
- ☐ Enable Flare
- ☐ Enable Mobile VoIP Client



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.

Ext89107: 89107\*

User Voicemail DND Short Codes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

Call Settings **Supervisor Settings** Multi-line Options Call Log TUI

Login Code \*\*\*\*\*

Login Idle Period (secs)

Monitor Group <None>

Coverage Group <None>

Status on No-Answer Logged On (No change)

Reset Longest Idle Time

☒ All Calls

☐ External Incoming

After Call Work Time (secs) System Default (10)

☐ Force Login

☐ Force Account Code

☐ Incoming Call Bar

☐ Outgoing Call Bar

☐ Inhibit Off-Switch Forward/Transfer

☐ Can Intrude

☒ Cannot be Intruded

☐ Can Trace Calls

☐ CCR Agent

☐ Automatic After Call Work

☐ Deny Auto Intercom Calls

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.

Ext89107: 89107\*

User Voicemail DND Short Codes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

**Call Settings** Supervisor Settings Multi-line Options Call Log TUI

Outside Call Sequence Default Ring

Inside Call Sequence Default Ring

Ringback Sequence Default Ring

No Answer Time (secs) System Default (15)

Wrap-up Time (secs) 2

Transfer Return Time (secs) Off

Call Cost Mark-Up 100

☒ **Call Waiting On**

☐ Answer Call Waiting On Hold

☐ Busy On Held

☐ Offhook Station

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 BT Ireland.

Extn89107: 89107\*

Personal Directory

User Voicemail DND Short Codes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming Menu Programming Mobility Group Membership Announcements SIP

SIP Name 014xxxx01

SIP Display Name (Alias) 014xxxx01

Contact 014xxxx01

☐ Anonymous

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.

Extn89107: 89107\*

Personal Directory

User Voicemail DND Short Codes Source Numbers Telephony Forwarding Dial In Voice Recording Button Programming Menu Programming Mobility

☐ Internal Twinning

Twinned Handset: <None>

Maximum Number of Calls: 1

☐ Twin Bridge Appearances

☐ Twin Coverage Appearances

☐ Twin Line Appearances

☒ Mobility Features

☒ Mobile Twinning

Twinned Mobile Number (including dial access code) 90894xxxxx1

Twining Time Profile: <None>

Mobile Dial Delay (secs) 3

Mobile Answer Guard (secs) 0

☐ Hunt group calls eligible for mobile twinning

☐ Forwarded calls eligible for mobile twinning

☐ Twin When Logged Out

☐ one-X Mobile Client

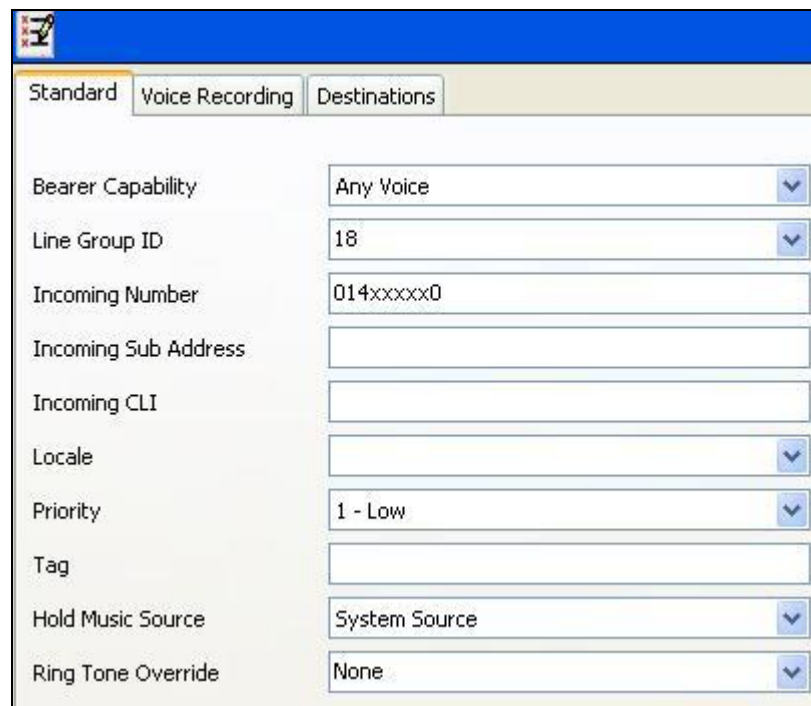
☐ Mobile Call Control

☐ Mobile Callback

## 5.9. Incoming Call Routing

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

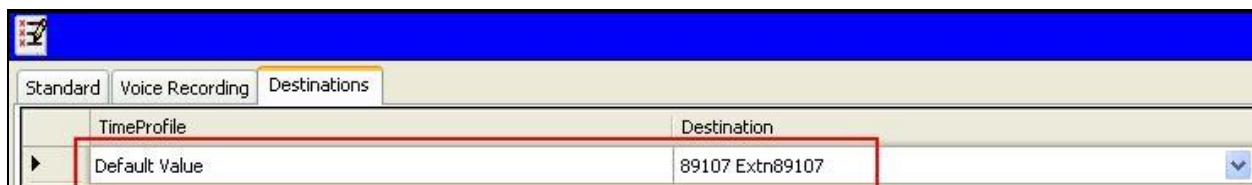
- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.2**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other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



The screenshot shows the 'Standard' tab of the Incoming Call Routes configuration window. The fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	014xxxxx0
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

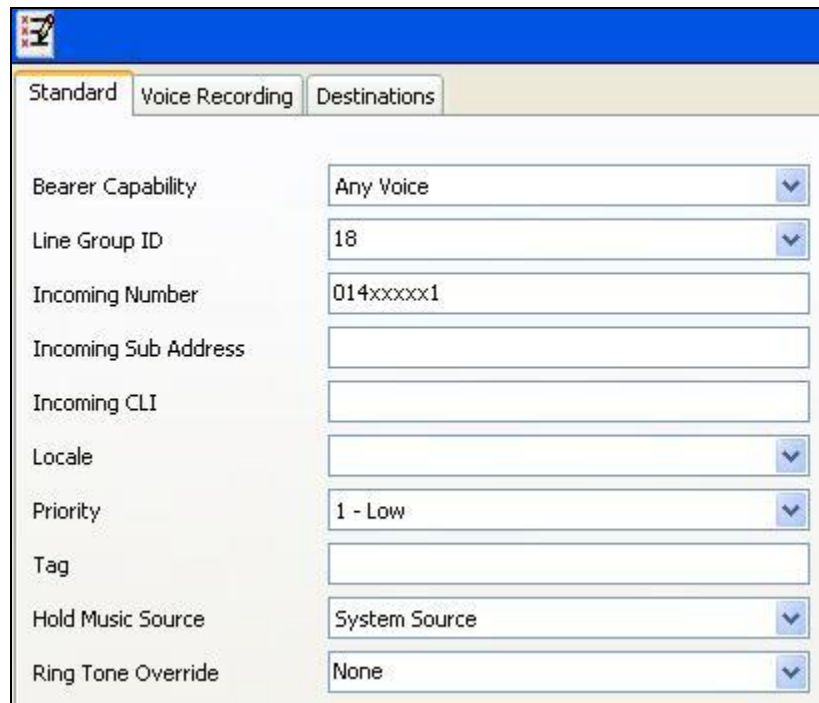
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89107.



The screenshot shows the 'Destinations' tab of the Incoming Call Routes configuration window. The fields are as follows:

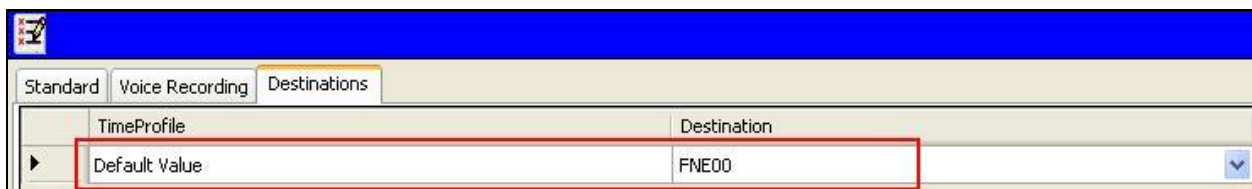
Field	Value
TimeProfile	
Destination	89107 Extn89107

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 **014xxxxx1** mapped to a shortcode **FNE** is illustrated.



Field	Value
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	014xxxxx1
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

The **Destination** tab contains the Destination field **FNE00** which was entered manually. An incoming call to **014xxxxx1** 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.



Field	Value
TimeProfile	Default Value
Destination	FNE00

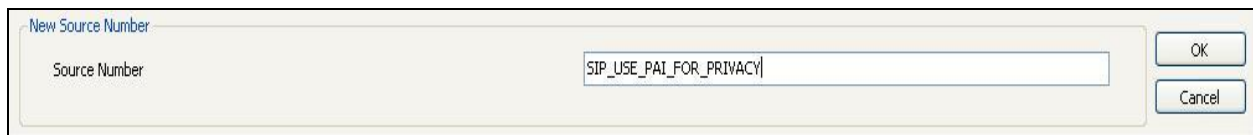
## 5.10. Privacy / Anonymous Calls

There are multiple methods for a user to withhold outgoing identification:

- Dialing the short code \*67 to access the SIP Line.
- Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user.
- Avaya Telephones equipped with a “Features” button can also request privacy for a specific call, without dialing a unique short code, using **Features** → **Call Settings** → **Withhold Number**, on the phone itself.

To configure IP Office to include the caller’s DID number in the P-Asserted-Identity SIP header to admit an otherwise anonymous caller to the network, the following procedure may be used.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field, type **SIP\_USE\_PAI\_FOR\_PRIVACY**. Click **OK**.



The source number **SIP\_USE\_PAI\_FOR\_PRIVACY** should now appear in the list of Source Numbers as shown below.



## 5.11. Save Configuration

Navigate to **File** → **Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

## 6. Configure Avaya Session Border Controller for Enterprise

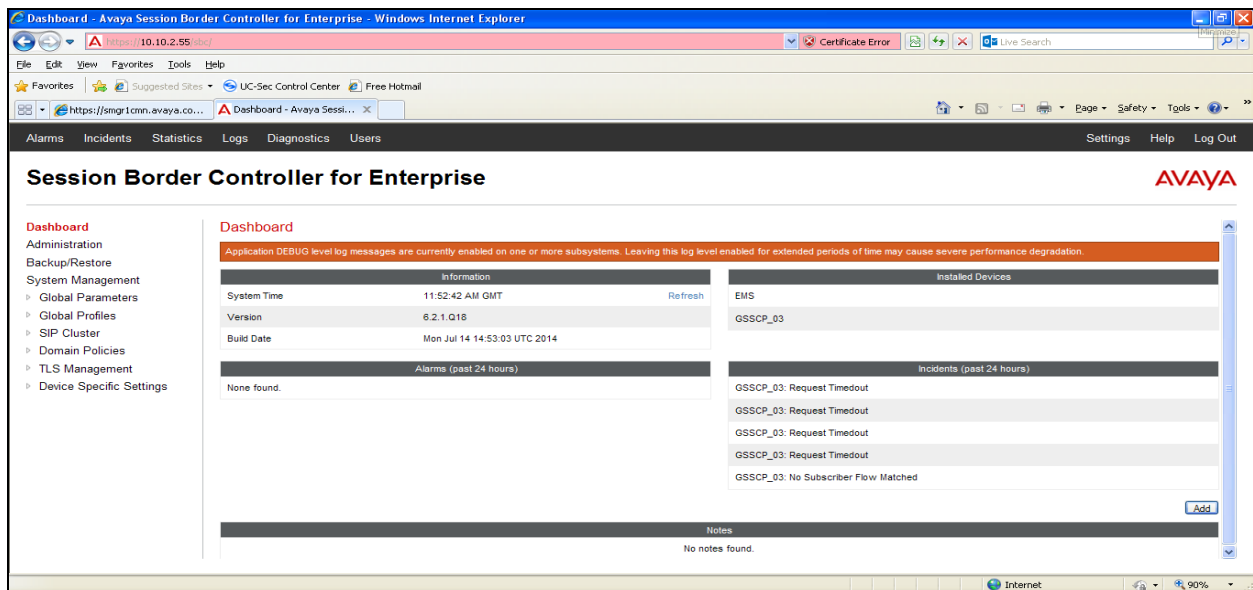
This section describes the configuration of the Session Border Controller for Enterprise (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP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

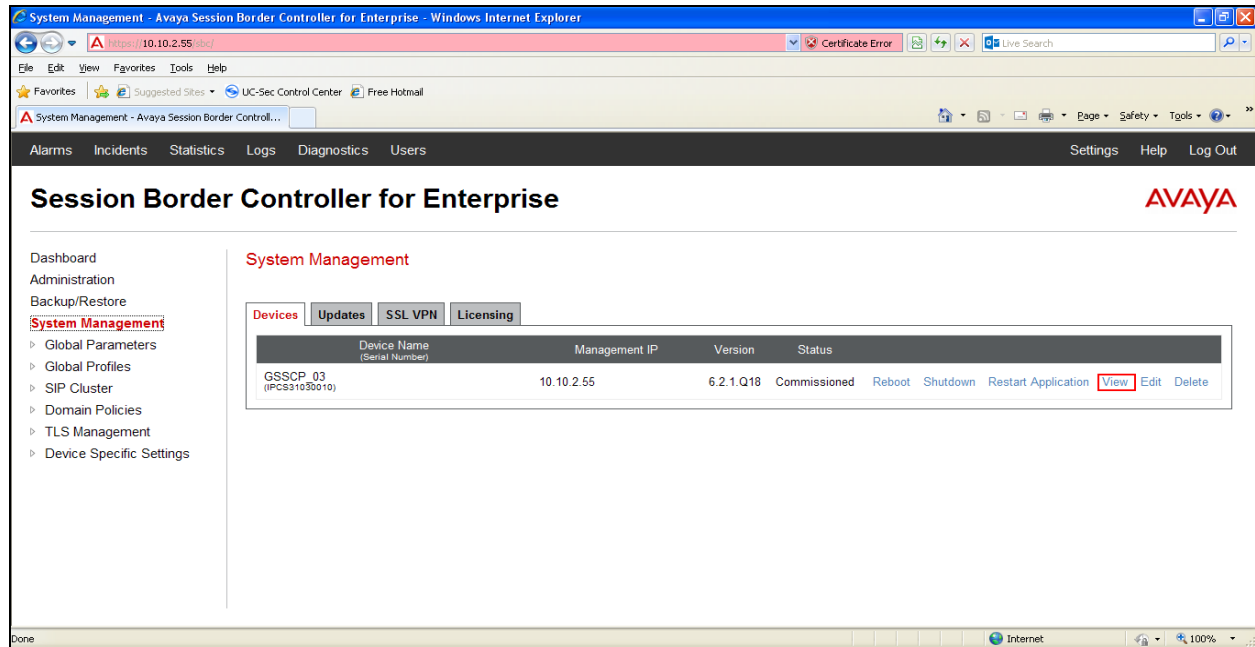


Once logged in, a dashboard is presented with a menu on the left-hand side. The menu is used as a starting point for all configuration of the Avaya SBCE.





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 sample configuration, a single device named **GSSCP\_03** 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**, **Network Configuration**, **DNS Configuration** and **Management IP** information.

### System Information: GSSCP\_03

#### General Configuration

Appliance Name	GSSCP_03
Box Type	SIP
Deployment Mode	Proxy

#### Device Configuration

HA Mode	No
Two Bypass Mode	No

#### Network Configuration

IP	Public IP	Netmask	Gateway	Interface
10.10.3.30	10.10.3.30	255.255.255.0	10.10.3.1	A1
192.168.37.2	192.168.37.2	255.255.255.128	192.168.37.1	B1
10.10.3.35	10.10.3.35	255.255.255.0	10.10.3.1	A1

#### DNS Configuration

Primary DNS	8.8.8.8
Secondary DNS	10.10.7.100
DNS Location	DMZ
DNS Client IP	192.168.37.2

#### Management IP(s)

IP	10.10.2.55
----	------------



## 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

The screenshot shows a configuration window titled "Profile: Avaya" with a close button (X) in the top right corner. The window contains a "General" tab with various configuration options. The options and their current states are as follows:

Option	Current State
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None (dropdown menu)
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

At the bottom right of the window is a "Next" button.

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

**Profile: Avaya** X

Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

Finish

### 6.2.2. Server Interworking – BT Ireland

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 **BT** 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**.

The screenshot shows a configuration window titled "Profile: BT" with a close button (X) in the top right corner. The window contains a "General" tab with various configuration options. The options and their current values are as follows:

Option	Value
Hold Support	<input checked="" type="radio"/> None <input type="radio"/> RFC2543 - c=0.0.0.0 <input type="radio"/> RFC3264 - a=sendonly
180 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
181 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
182 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
183 Handling	<input checked="" type="radio"/> None <input type="radio"/> SDP <input type="radio"/> No SDP
Refer Handling	<input type="checkbox"/>
URI Group	None (dropdown menu)
3xx Handling	<input type="checkbox"/>
Diversion Header Support	<input type="checkbox"/>
Delayed SDP Handling	<input type="checkbox"/>
Re-Invite Handling	<input type="checkbox"/>
T.38 Support	<input checked="" type="checkbox"/>
URI Scheme	<input checked="" type="radio"/> SIP <input type="radio"/> TEL <input type="radio"/> ANY
Via Header Format	<input checked="" type="radio"/> RFC3261 <input type="radio"/> RFC2543

At the bottom of the window, there is a "Next" button.

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

**Profile: BT** X

Record Routes	<input type="radio"/> None <input type="radio"/> Single Side <input checked="" type="radio"/> Both Sides
Topology Hiding: Change Call-ID	<input type="checkbox"/>
Call-Info NAT	<input type="checkbox"/>
Change Max Forwards	<input checked="" type="checkbox"/>
Include End Point IP for Context Lookup	<input type="checkbox"/>
OCS Extensions	<input type="checkbox"/>
AVAYA Extensions	<input type="checkbox"/>
NORTEL Extensions	<input type="checkbox"/>
Diversion Manipulation	<input type="checkbox"/>
Diversion Header URI	<input type="text"/>
Metaswitch Extensions	<input type="checkbox"/>
Reset on Talk Spurt	<input type="checkbox"/>
Reset SRTP Context on Session Refresh	<input type="checkbox"/>
Has Remote SBC	<input checked="" type="checkbox"/>
Route Response on Via Port	<input type="checkbox"/>
Cisco Extensions	<input type="checkbox"/>

Finish

### 6.2.3. 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.

Create a Routing Profile for IP Office and a Routing Profile for BT Ireland. To add a routing profile, navigate to **Global Profiles → Routing** and select **Add**. Enter a **Profile Name** and click **Next** to continue.

In the new window that appears, enter the following values. Use default values for all remaining fields:

- **URI Group:** Select “\*” from the drop down box
- **Next Hop Server 1:** Enter the Domain Name or IP address of the Primary Next Hop server
- **Next Hop Server 2:** (Optional) Enter the Domain Name or IP address of the secondary Next Hop server
- **Routing Priority Based on Next Hop Server:** Checked
- **Use Next Hop for In-Dialog Messages:** Select only if there is no secondary Next Hopserver
- **Outgoing Transport:** Choose the protocol used for transporting outgoing signaling packets

Click **Finish**.

Each URI group may only be used once per Routing Profile.

**Next Hop Routing**

URI Group: \*

Next Hop Server 1:  (IP, IP:Port, Domain, or Domain:Port)

Next Hop Server 2:  (IP, IP:Port, Domain, or Domain:Port)

Routing Priority based on Next Hop Server: ☒

Use Next Hop for In Dialog Messages: ☐

Ignore Route Header for Messages Outside Dialog: ☐

NAPTR: ☐

SRV: ☐

Outgoing Transport: ☐ TLS ☐ TCP ☒ UDP

Finish

The following screen shows the Routing Profile to IP Office.

Routing Profiles: Avaya

Click here to add a description.

Routing Profile

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	
1	*	10.10.3.19	---	<a href="#">View</a> <a href="#">Edit</a>

The following screen shows the Routing Profile to BT Ireland SBC A.

Routing Profiles: BT\_SBC\_A

Click here to add a description.

Routing Profile

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	
1	*	192.168.16.163	---	<a href="#">View</a> <a href="#">Edit</a>

The following screen shows the Routing Profile to BT Ireland SBC B.

Routing Profiles: BT\_SBC\_B

Click here to add a description.

Routing Profile

Priority	URI Group	Next Hop Server 1	Next Hop Server 2	
1	*	62.172.16.179	---	<a href="#">View</a> <a href="#">Edit</a>

#### 6.2.4. Server Configuration– Avaya IP Office

Servers are defined for each server connected to the Avaya SBCE. In this case, BT Ireland 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** tab, set the following:

- Select **Server Type** to be **Call Server**
- Enter **IP Addresses / Supported FQDNs** to **10.10.7.110** (IP Office IP Address)
- For **Supported Transports**, check **TCP**
- **TCP Port:5060**
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs

The screenshot shows the 'Server Configuration Profile - General' window. The 'Server Type' dropdown is set to 'Call Server'. The 'IP Addresses / Supported FQDNs' text box contains '10.10.7.110'. The 'Supported Transports' section has 'TCP' checked, 'UDP' unchecked, and 'TLS' unchecked. The 'TCP Port' text box contains '5060'. The 'UDP Port' and 'TLS Port' text boxes are empty. A 'Finish' button is located at the bottom of the window.



On the **Advanced** tab:

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

The screenshot shows a window titled "Server Configuration Profile - Advanced" with a close button (X) in the top right corner. The window contains several configuration options:

- Enable DoS Protection**: A checkbox that is currently unchecked.
- Enable Grooming**: A checkbox that is currently unchecked.
- Interworking Profile**: A dropdown menu with "Avaya" selected. This dropdown is highlighted with a red rectangular box.
- Signaling Manipulation Script**: A dropdown menu with "None" selected.
- TCP Connection Type**: Three radio buttons labeled "SUBID", "PORTID", and "MAPPING". The "SUBID" radio button is selected.

At the bottom center of the window is a button labeled "Finish".

### 6.2.5. Server Configuration – BT Ireland

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

- Select **Server Type** as **Trunk Server**
- Set **IP Address** to **192.168.16.163** (BT Ireland SBC A)
- **Supported Transports**: Check **UDP**
- **UDP Port**: **5060**
- Click **Next**(not shown)

Server Configuration Profile - General

Server Type: Trunk Server

IP Addresses / Supported FQDNs  
Separate entries with commas: 192.168.16.163

Supported Transports:  
☐ TCP  
☒ UDP  
☐ TLS

TCP Port:

UDP Port: 5060

TLS Port:

Finish

In the new window that appears, enter the following values as BT Ireland require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

Click **Next** to continue.

Server Configuration Profile - Authentication

Enable Authentication ☒

User Name

Realm   
(Leave blank to detect from server challenge)

Password   
(Leave blank to keep existing password)

Confirm Password

Finish

On the Advanced tab:

- Select **BT** for Interworking Profile
- Click Finish

Server Configuration Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile

Signaling Manipulation Script

UDP Connection Type ☒ SUBID ☐ PORTID ☐ MAPPING

Finish

To define the second BT Ireland SBC B as a Trunk Servers, navigate to select **Global Profiles** → **Server Configuration** and click on **Add Profile** and enter a descriptive name. On the **Add Server Configuration Profile** tab, click on **Edit** and set the following:

- Select **Server Type** as **Trunk Server**
- Set **IP Address** to **192.168.16.179** (BT Ireland SBC B)
- **Supported Transports**: Check **UDP**
- **UDP Port**: **5060**
- Click **Next**

Server Configuration Profile - General

Server Type: Trunk Server

IP Addresses / Supported FQDNs  
Separate entries with commas  
192.168.16.179

Supported Transports:  
☐ TCP  
☒ UDP  
☐ TLS

TCP Port:

UDP Port: 5060

TLS Port:

Finish

In the new window that appears, enter the following values as BT Ireland require authentication to connect to their network:

- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider
- **Realm:** Enter realm details provided by the Service Provider
- **Password** Enter password provided by the Service Provider
- **Confirm Password** Re-enter password provided by the Service Provider

Click **Next** to continue.

Server Configuration Profile - Authentication

Enable Authentication ☒

User Name

Realm   
(Leave blank to detect from server challenge)

Password   
(Leave blank to keep existing password)

Confirm Password

Finish

On the Advanced tab:

- Select **BT** for Interworking Profile
- Click Finish

Server Configuration Profile - Advanced

Enable DoS Protection ☐

Enable Grooming ☐

Interworking Profile

Signaling Manipulation Script

UDP Connection Type ☒ SUBID ☐ PORTID ☐ MAPPING

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**.
- Click **Finish** (not shown)

The screenshot shows the 'Topology Hiding Profiles: Avaya' configuration page. On the left, a sidebar lists 'Topology Hiding Profiles' with options: 'default', 'cisco\_th\_profile', 'Avaya' (selected), and 'BT'. The main area has a blue header bar with 'Click here to add a description.' and buttons for 'Rename', 'Clone', and 'Delete'. Below this is a 'Topology Hiding' tab and a table with the following data:

Header	Criteria	Replace Action	Overwrite Value
From	IP/Domain	Overwrite	avaya.com
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	avaya.com
To	IP/Domain	Overwrite	avaya.com
Record-Route	IP/Domain	Auto	---

An 'Edit' button is located at the bottom right of the table.

To define Topology Hiding for BT Ireland, 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).

- Enter a descriptive **Profile Name** such as **BT** and click **Next**
- If the required Header is not shown, click on **Add Header**
- Select **Request-Line**, **To** and **From** as the required headers from the **Header** drop down menu
- Select the required action from the **Required Action** drop down menu, **Auto** was used for test
- Click **Finish** (not shown)

### Topology Hiding Profiles: BT

Add

Topology Hiding Profiles

default

cisco\_th\_profile

Avaya

BT

Click here to add a description.

Topology Hiding

Header	Criteria	Replace Action	Overwrite Value
From	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
To	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---

Edit

CMN; Reviewed:  
SPOC 1/27/2015

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### 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.

In the test configuration, two IP addresses were used on the internal interface so that different server flows could be assigned depending on which interface address the SIP messages were received on. These server flows were used to direct traffic to the two BT Ireland SBCs separately.

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**. Enter details in the blank box that appears at the end of the list

- Define the two 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)

Network Management: GSSCP\_03

Devices: GSSCP\_03

Network Configuration | Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from [System Management](#)

A1 Netmask: 255.255.255.0 A2 Netmask: B1 Netmask: 255.255.255.128 B2 Netmask:

IP Address	Public IP	Gateway	Interface	
10.10.3.30		10.10.3.1	A1	Delete
192.168.37.2		192.168.37.1	B1	Delete
10.10.3.35		10.10.3.1	A1	Delete

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

Network Management: GSSCP\_03

Devices: GSSCP\_03

Network Configuration | Interface Configuration

Name	Administrative Status	
A1	Enabled	Toggle
A2	Disabled	Toggle
B1	Enabled	Toggle
B2	Disabled	Toggle

## 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 to be used in the server flow for BT Ireland SBC A:

- Select **Add** and enter details of the first 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 one of 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 internal interface to be used in the server flow for BT Ireland SBC B:

- 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 interface
- For **Signaling IP**, select the other **internal** signalling interface IP address 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 an **external** signalling interface IP address defined in **Section 6.3**
- Select **UDP** port number, **5060** is used for the SIP Trunk

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

Signaling Interface: GSSCP\_03

Devices	
GSSCP_03	

Signaling Interface							Add
Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
Int_Sig_A	10.10.3.30	5060	5060	---	None		Edit Delete
Ext_Sig	192.168.37.2	5060	5060	---	None		Edit Delete
Int_Sig_B	10.10.3.35	5060	5060	---	None		Edit Delete

## 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 enter details of the media IP and RTP port range on the internal interface to be used in the server flow for BT Ireland SBC A.

- 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 an **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 internal interface to be used in the server flow for BT Ireland SBC B.

- 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 an **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 an **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 sample configuration for the inside and outside IP interfaces.

Media Interface: GSSCP\_03

Devices

GSSCP\_03

Media Interface

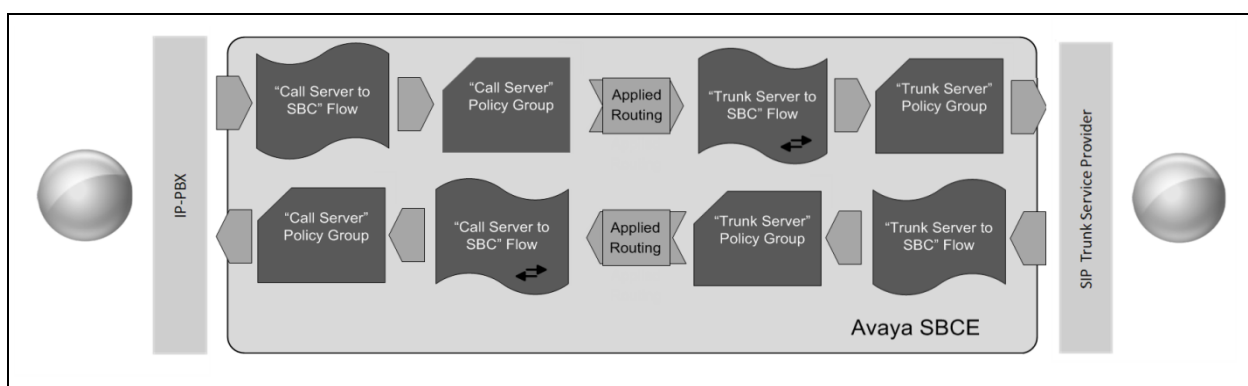
Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from [System Management](#).

Add

Name	Media IP	Port Range	
Int_Media_A	10.10.3.30	10000 - 51000	Edit Delete
Ext_Media	192.168.37.2	10000 - 51000	Edit Delete
Int_Media_B	10.10.3.35	10000 - 51000	Edit Delete

## 6.5. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from IP Office to BT Ireland's SIP Trunk and incoming flows from BT Ireland'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 BT Ireland network and vice versa. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.



Two server flows are required for outgoing traffic and two are required for incoming. This is so that traffic can be routed to both network SBCs and can also be received from both network SBCs. As mentioned previously, the network SBCs have been designated as BT Ireland SBC A and BT Ireland SBC B for the purposes of testing and documentation.

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

Subscriber Flows

Server Flows

Add

Hover over a row to see its description.

Server Configuration: Avaya

Update

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Call_Server_A	*	Ext_Sig	Int_Sig_A	default-low	BT_SBC_A	View	Clone	Edit	Delete
2	Call_Server_B	*	Ext_Sig	Int_Sig_B	default-low	BT_SBC_B	View	Clone	Edit	Delete

Server Configuration: BT\_SBC\_A

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server_A	*	Int_Sig_A	Ext_Sig	default-low	Avaya	View	Clone	Edit	Delete

Server Configuration: BT\_SBC\_B

Priority	Flow Name	URI Group	Received Interface	Signaling Interface	End Point Policy Group	Routing Profile				
1	Trunk_Server_B	*	Int_Sig_B	Ext_Sig	default-low	Avaya	View	Clone	Edit	Delete

To define a Server Flow for IP Office to each of the network SBCs, 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 **Name** field enter a descriptive name for the server flow for IP Office, in this case **Call\_Server\_A** was used.
- In the **Server Configuration** drop down menu, select the Server defined in **Section 6.2.4** for IP Office.
- In the **Received Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.4.1**.
- In the **Signaling Interface** drop-down menu, select the first internal SIP signalling interface defined in **Section 6.4.1**.
- In the **Media Interface** drop-down menu, select the internal media interface defined in **Section 6.4.2**.
- In the **Routing Profile** drop-down menu, select the routing profile of BT Ireland SBC A defined in **Section 6.2.3**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of IP Office defined in **Section 6.2.6** and click **Finish**.

The screenshot shows a configuration window titled "Flow: Call\_Server\_A". It contains a form with the following fields and values:

Field	Value
Flow Name	Call_Server_A
Server Configuration	Avaya
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Ext_Sig
Signaling Interface	Int_Sig_A
Media Interface	Int_Media_A
End Point Policy Group	default-low
Routing Profile	BT_SBC_A
Topology Hiding Profile	Avaya
File Transfer Profile	None

A "Finish" button is located at the bottom right of the form.

Repeat the above process for Call\_Server\_B, selecting the specific Call\_Server\_B entries for server flow configuration.

To define Server Flows for the BT Ireland network SBCs (BT Ireland SBC A and BT Ireland SBC B), 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 **Name** field enter a descriptive name for the server flow for BT Ireland SBC A, in this case **Trunk\_Server\_A** was used.
- In the **Server Configuration** drop down menu, select the Server defined in **Section 6.2.5** for BT Ireland SBC A
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 6.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 6.4.2**.
- In the **Routing Profile** drop-down menu, select the routing profile of IP Office defined in **Section 6.2.3**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of BT Ireland defined in **Section 6.2.6** and click **Finish**.

Flow: Trunk\_Server\_A

Flow Name	Trunk_Server_A
Server Configuration	BT_SBC_A
URI Group	*
Transport	*
Remote Subnet	*
Received Interface	Int_Sig_A
Signaling Interface	Ext_Sig
Media Interface	Ext_Media
End Point Policy Group	default-low
Routing Profile	Avaya
Topology Hiding Profile	BT
File Transfer Profile	None

Finish

Repeat the above process for BT Ireland SBC B, selecting the specific BT Ireland SBC B entries for server flow configuration.



## 7. BT Ireland SIP Trunk Service Configuration

BT Ireland 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. BT Ireland 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.

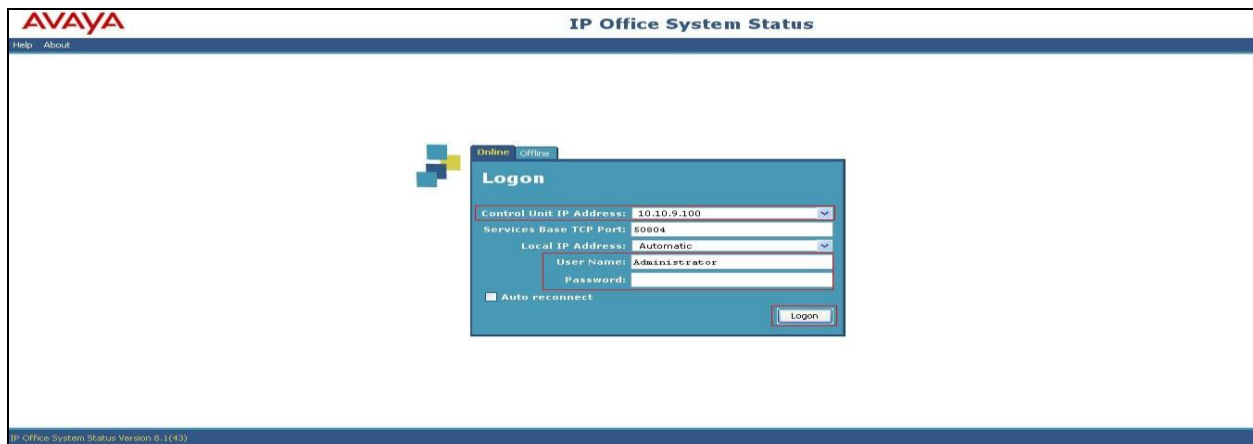
## 8. Verification Steps

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.

AVAYA

IP Office System Status

Help Snapshot LogOff About

System

Alarms (7)

Extensions (18)

Trunks (9)

Line: 1

Line: 2

Lines: 5 - 8

Line: 9

Line: 10

Line: 18

Active Calls

Resources

Voicemail

IP Networking

Status

Utilization Summary

Alarms

SIP Trunk Summary

Peer Domain Name:

192.168.230.98

Resolved Address:

192.168.230.98

Line Number:

18

Number of Administered Channels:

10

Number of Channels in Use:

0

Administered Compression:

G729 A, G711 A

Silence Suppression:

Off

SIP Trunk Channel Licenses:

Unlimited

SIP Trunk Channel Licenses in Use:

0

SIP Device Features:

0%

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call
1			Idle	00:12:22					
2			Idle	00:12:11					
3			Idle	01:46:02					
4			Idle	01:46:02					
5			Idle	01:46:02					
6			Idle	01:46:02					
7			Idle	01:46:02					
8			Idle	01:46:02					
9			Idle	01:46:02					
10			Idle	01:46:02					

## 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.

All Settings

T1

VComp

VPN

WAN

SCN

SSI

Jade

ATM

Call

DTE

EConf

Frame Relay

GOD

H.323

Interface

ISDN

Key/Lamp

Directory

Media

PPP

R2

Routing

Services

SIP

System

Events

☐ Sip
 

Low

☐ STUN

☐ SIP Dect

Packets

☐ SIP Reg/Opt Rx

☐ SIP Misc Rx

☐ SIP Reg/Opt Tx

☐ SIP Misc Tx

☐ SIP Call Rx

☐ Cm Notify Rx

☐ SIP Call Tx

☐ Cm Notify Tx

☒ Sip Rx
 

hex

IP Filter (nnn.nnn.nnn.nnn)

☒ Sip Tx
 

hex

Default All

Clear All

Tab Clear All

Tab Set All

OK

Cancel

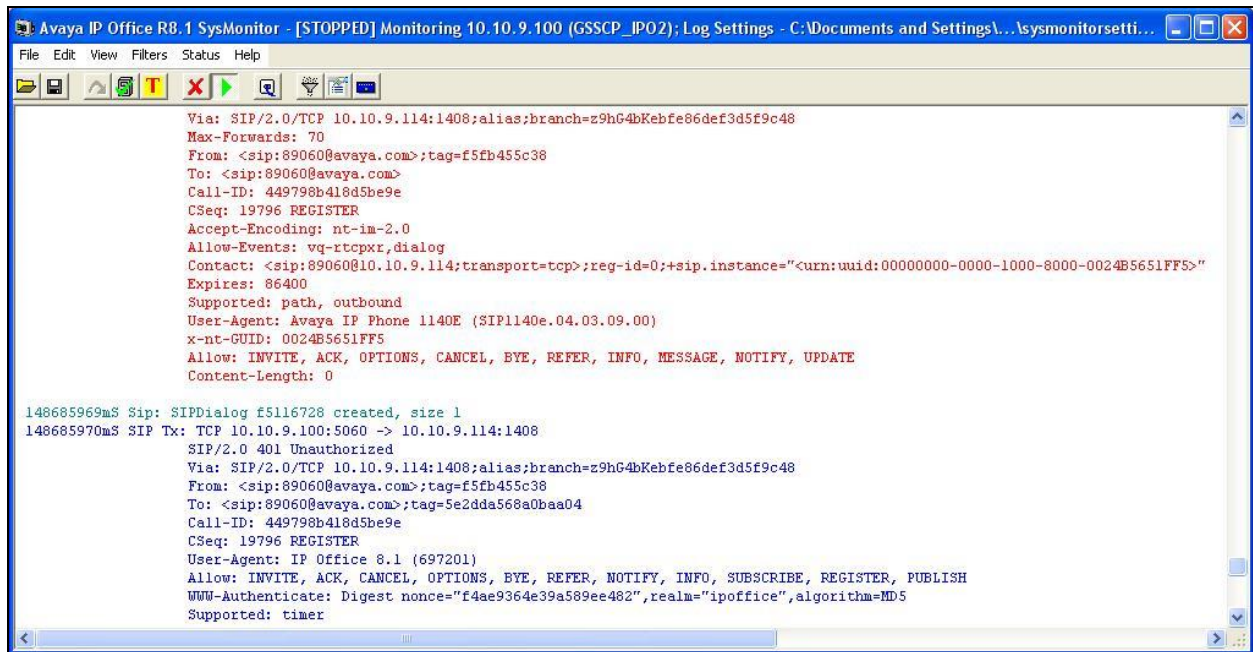
Save File

Load File

Load Partial File

Select File

As an example, the following shows a portion of the monitoring window for an outbound call.

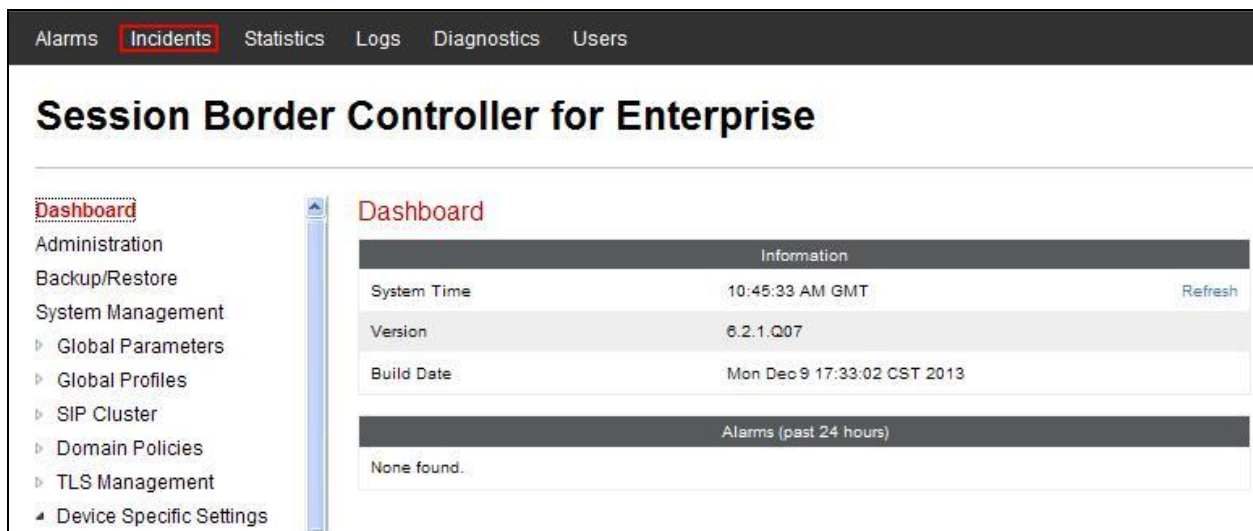


## 8.3. Avaya SBCE

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.

Incident Viewer

AVAYA

Device All ▼ Category All ▼ Clear Refresh Generate Report

Displaying results 1 to 15 out of 2000.

Type	ID	Date	Time	Category	Device	Cause
Routing Failure	686948871165253	7/15/13	2:15 PM	Policy	VLAN3_MicroSBC	Neither target nor source is Call Server, Sending 403 Forbidden
Routing Failure	686948811180314	7/15/13	2:13 PM	Policy	VLAN3_MicroSBC	Neither target nor source is Call Server, Sending 403 Forbidden
ACK Message Out of Dialog	686948761299324	7/15/13	2:12 PM	Protocol Discrepancy	VLAN3_MicroSBC	General Method not allowed Out-Of-Dialog
Message Dropped	686948761299222	7/15/13	2:12 PM	Policy	VLAN3_MicroSBC	No Subscriber Flow Matched
Call Denied	686948761263328	7/15/13	2:12 PM	Policy	VLAN3_MicroSBC	No Subscriber Flow Matched
Routing Failure	686948751195370	7/15/13	2:11 PM	Policy	VLAN3_MicroSBC	Neither target nor source is Call Server, Sending 403 Forbidden

### 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 the signalling interface IP address 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**

Trace: GSSCP\_03

Devices	Call Trace	Packet Capture	Captures
GSSCP_03	<div>Packet Capture Configuration</div> <div> <div>Status</div> <div>Ready</div> </div> <div> <div>Interface</div> <div>B1</div> </div> <div> <div>Local Address IP[:Port]</div> <div>192.168.122.57 : </div> </div> <div> <div>Remote Address *, *Port, IP, IP:Port</div> <div>*</div> </div> <div> <div>Protocol</div> <div>All</div> </div> <div> <div>Maximum Number of Packets to Capture</div> <div>10000</div> </div> <div> <div>Capture Filename Using the name of an existing capture will overwrite it.</div> <div>SIP_Trunk_Testpcap</div> </div> <div> <div>Start Capture</div> <div>Clear</div> </div>		

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

Trace: GSSCP\_03

Devices

GSSCP\_03

Call Trace

Packet Capture

Captures

Refresh

File Name	File Size (bytes)	Last Modified
<a href="#">SIP_Trunk_Test_20140916121852.pcap</a>	0	September 16, 2014 12:18:52 PM GMT <a href="#">Delete</a>

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 BT Ireland network

## 9. Conclusion

These Application Notes demonstrated how IP Office Release 9.0 and Avaya Session Border Controller for Enterprise can be successfully combined with BT Ireland 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 BT Ireland 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 BT Ireland 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 9.0 Documentation CD*, October 2013.
- [2] *IP Office 9.0 Installation Manual*, January 2014.
- [3] *IP Office Manager Manual 9.0*, February 2014
- [4] *IP Office Release 9.0 Implementing Voicemail Pro*, January 2014
- [5] *System Status Application*, November 2013
- [6] *IP Office Softphone Installation*, September 2013
- [7] *IP Office SIP Extension Installation*, October 2013
- [8] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>
- [9] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2
- [10] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2
- [11] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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