Avaya Solution & Interoperability Test Lab

Application Notes for BT Wholesale/HIPCOM SIP Trunk Service and Avaya IP Office 8.0 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the BT Wholesale (BTW)/HIPCOM SIP Trunk Service and Avaya IP Office.

The BTW/HIPCOM SIP Trunk Service provides PSTN access via a SIP trunk connected to the BTW/HIPCOM Voice Over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. This approach generally results in lower cost for the enterprise. BT is a member of the Avaya DevConnect Service Provider program.

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

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the BTW/HIPCOM SIP Trunk Service and Avaya IP Office. BTW/HIPCOM SIP Trunking service provides PSTN access via a SIP trunk connected to the BTW/HIPCOM network as an alternative to legacy analogue or digital trunks. This approach generally results in lower cost for the enterprise.

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 the HIPCOM SIP Trunking 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

A simulated enterprise site with Avaya IP Office was connected to the HIPCOM SIP Trunking service. To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types
- Phone types included H.323, 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
- Phone types included H.323, 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 Phone Manager Lite clients
- Various call types including: local, long distance, international, toll free(outbound)and directory assistance (118118)
- Codecs G.729A, G.711A and G.711Mu
- 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 twinning
- T.38 fax

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the HIPCOM SIP Trunk Service with the following observations:

- For outbound Calling Line Identity (CLI) restriction to work correctly the P-Asserted Identity info on the SIP invite must be set to the registration user. The SIP Contact field of the extension in IP Office has to be set to the same value as the SIP User name credential field in the SIP Line configuration. See **Sections 5.5.1** and **5.7** for more details
- Contact info on the SIP invite must be set to the registration user for Transfer and Mobile Twinning to work
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested
- Routing to emergency numbers (such as 999) was not tested

2.3. Support

For technical support on HIPCOM products please contact the HIPCOM support team at:

http://www.hipcom.co.uk/support or http://ipvoicesupport.btwholesale.com

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the HIPCOM SIP Trunking service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), an Avaya 2420 Digital Telephone, an Avaya analogue Telephone and fax machine. The site also has a Windows 2003 Server running Avaya IP Office Manager to configure the Avaya IP Office.

For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with arbitrary numbers that bear no relevance to the test configuration.

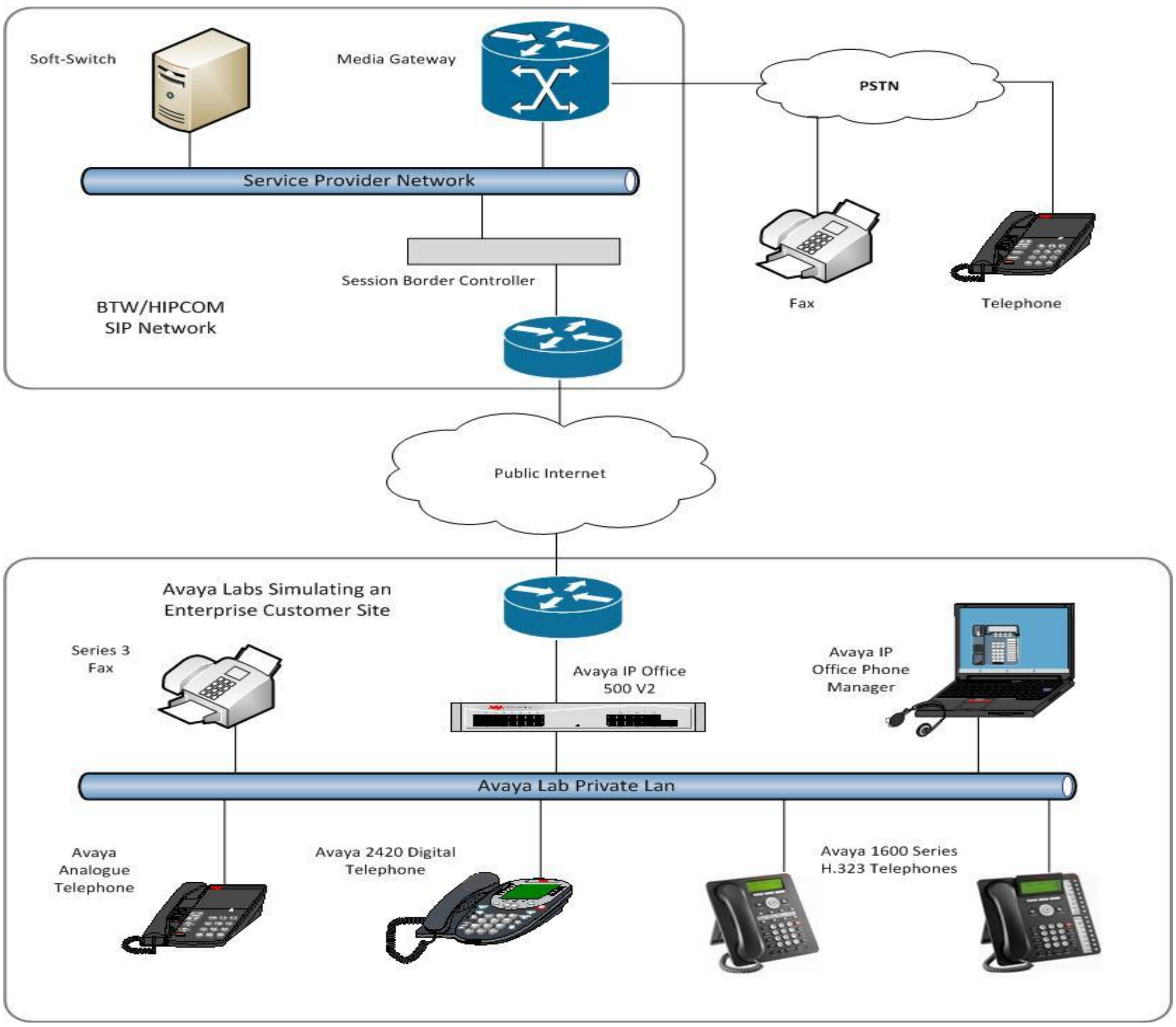


Figure 1: BT/HIPCOM SIP Trunking Solution to IP Office Topology

Avaya IP Office was configured to connect to a static IP address at the service provider and a registration username and password was required allowing the SIP line on the IP Office to Register with HIPCOM. For the purposes of the compliance test, users dialed a short code of 9 + N digits to send digits across the SIP trunk to HIPCOM. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent with the SIP domain provided by HIPCOM added.

In an actual customer configuration, the enterprise site may also include additional network components between the service provider and Avaya IP Office such as a session border controller or data firewall. A complete discussion of the configuration of these devices is beyond the scope of these Application Notes. However, it should be noted that SIP and RTP traffic between the service provider and Avaya IP Office must be allowed to pass through these devices. HIPCOM sends SIP signaling from one IP address. However, RTP traffic may originate from a different IP address and ports which may vary from customer to customer. Customers will need to work with HIPCOM to determine the proper IP addresses and ports that require access to their network.

4. Equipment and Software Validated

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

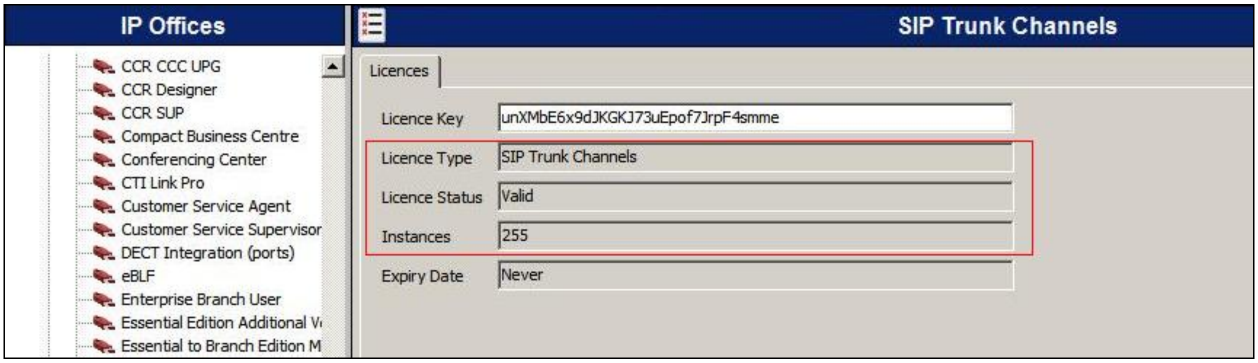
Equipment	**Software**
Avaya IP Office 500 V2	8.0(16)
Avaya 1620 Phone (H.323)	1.220
Avaya 2420 Digital Phone	NA
Avaya 98390 analogue Phone	NA
HIPCOM SIP Trunking	Acme Packet 4500 Net-Net SBC ver SCX6.1.0 Broadsoft – ver 14 Service Pack 9 Configuration version –HIPCOM v8.1.SIPConnect v1.0

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the HIPCOM SIP Trunking 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.

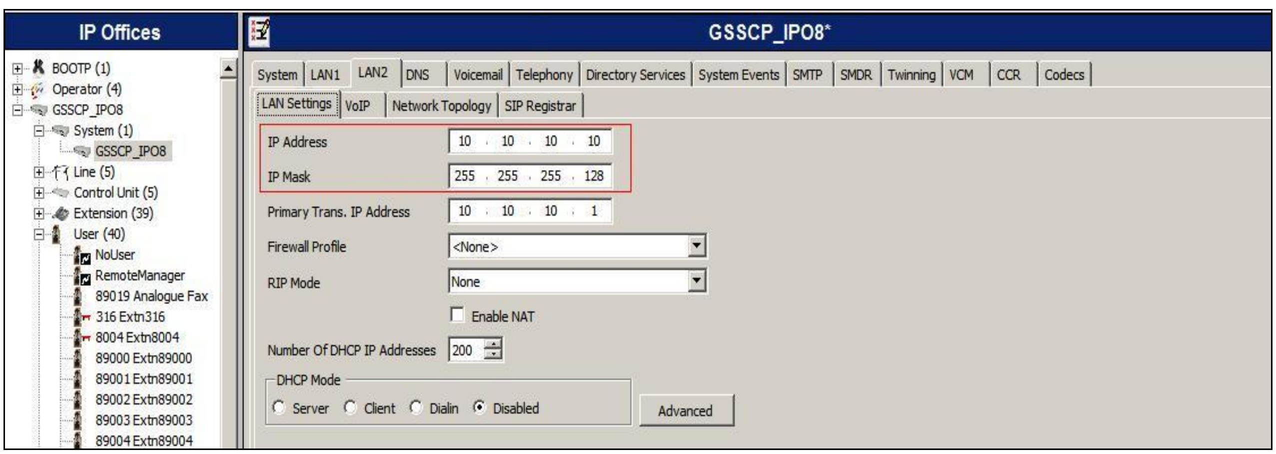
5.1. Verify System Capacity

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 HIPCOM.

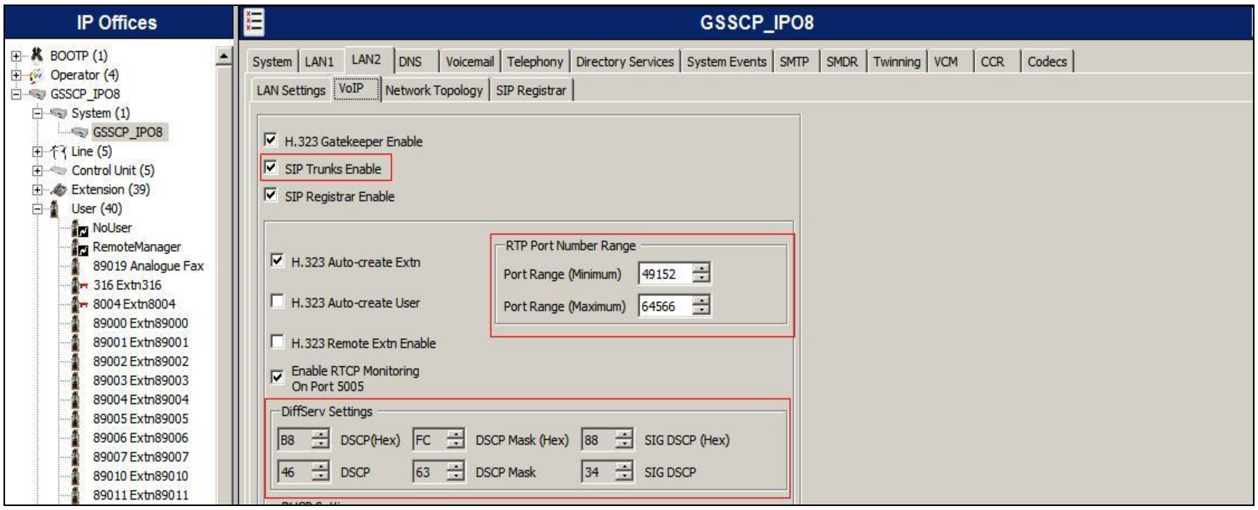


5.2. LAN2 Settings

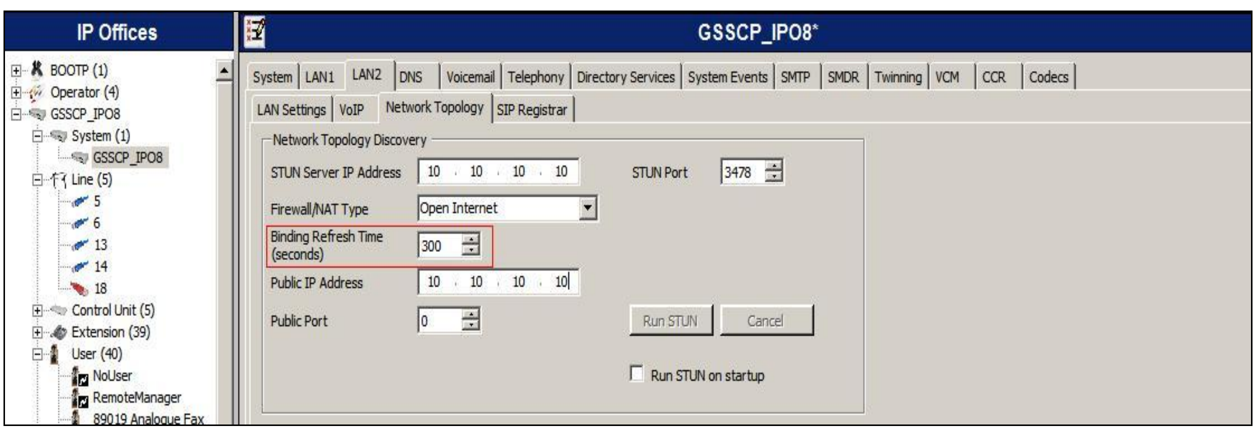
In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the external intranet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO8** in the Navigation Pane where GSSCP_IPO8 is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane



On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. 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 LAN2. Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signaling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signaling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements.

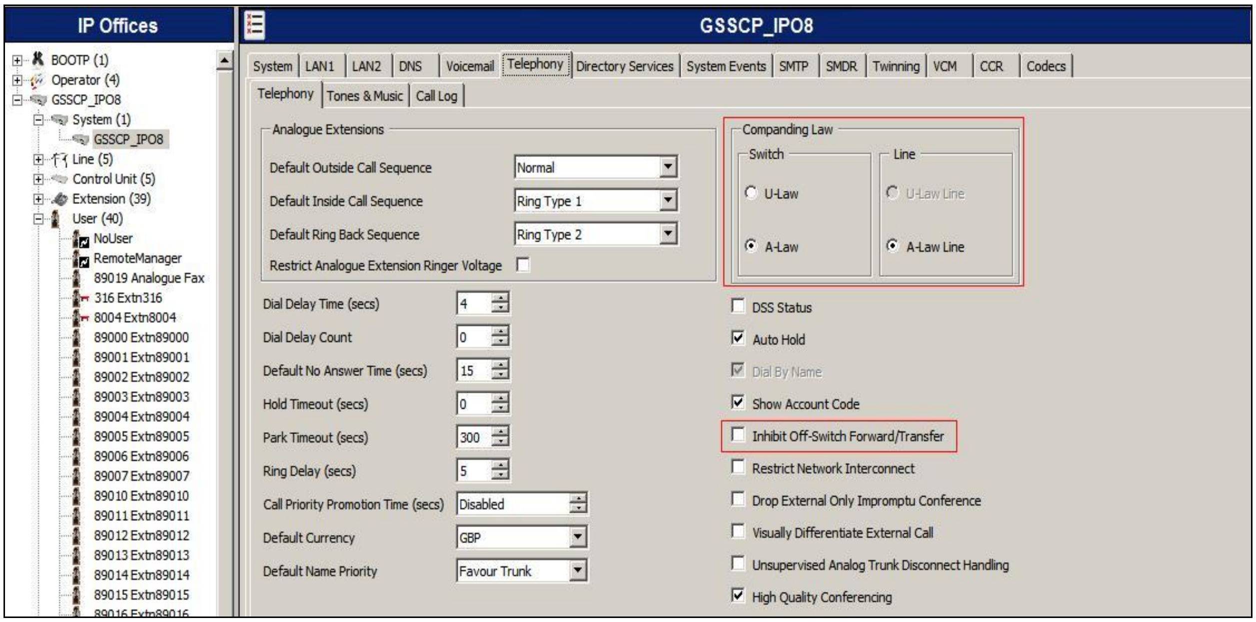


On the **Network Topology** tab **Binding Refresh Time** is set to **300.** This value is used to determine the frequency at which Avaya IP Office will send SIP OPTION messages to HIPCOM.



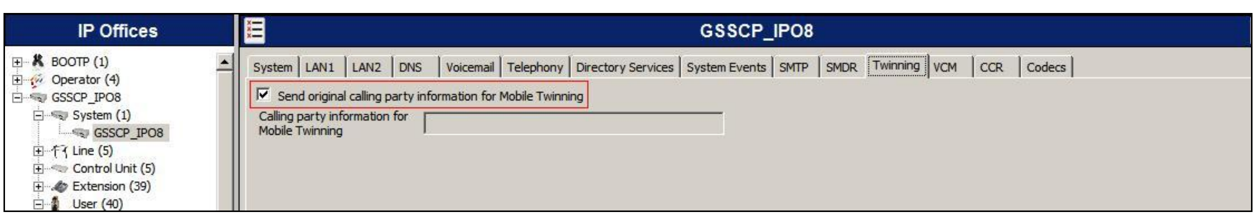
5.3. System Telephony Settings

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.



5.4. System Twinning Settings

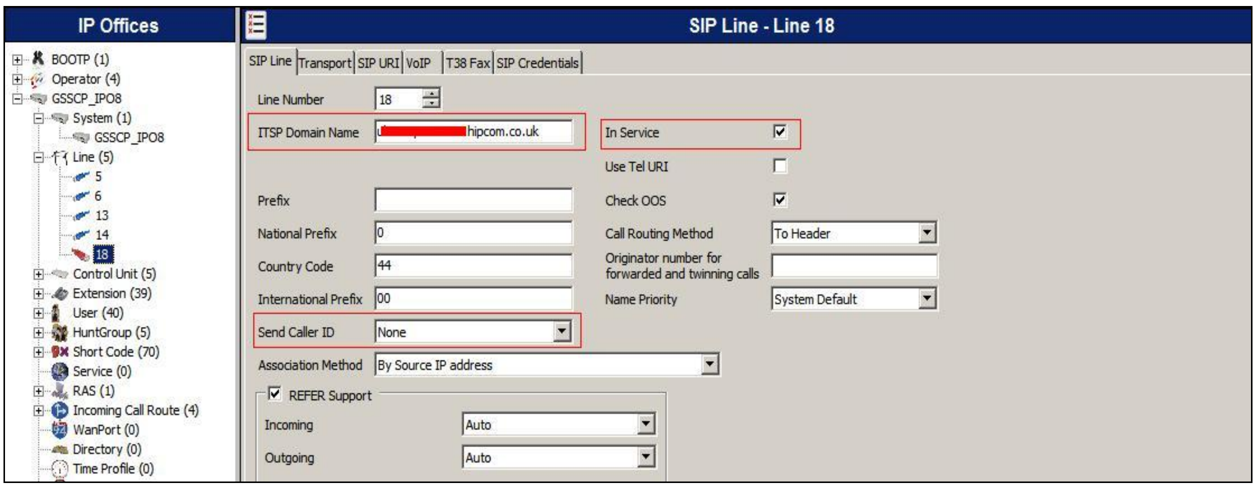
Navigate to the **Twinning** tab; check the box labeled **Send original calling party information for Mobile Twinning.** With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a twinned enabled phone, Avaya IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of the **Send Caller ID** parameter on the SIP line (**Section 5.5**).



5.5. Administer SIP Line

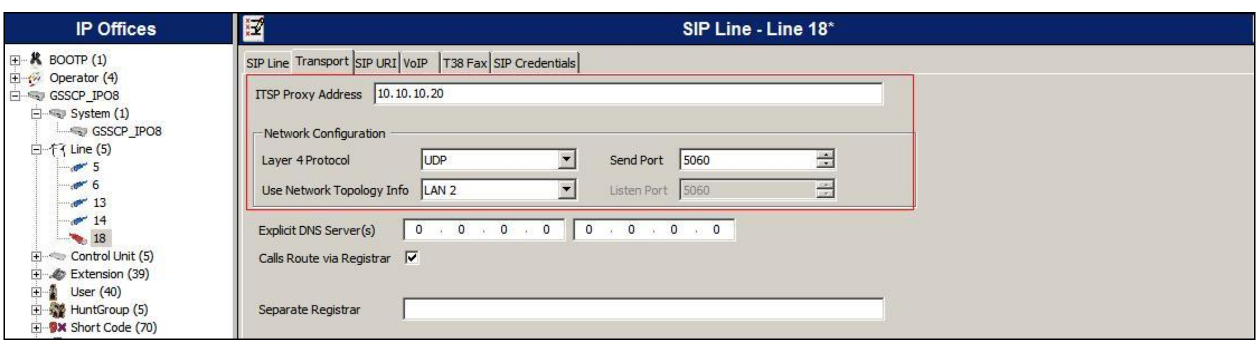
A SIP line is needed to establish the SIP connection between Avaya IP Office and the HIPCOM SIP Trunking service. To create a SIP line, begin by navigating to **Line** in the Navigation Pane. Right-click and select **New→SIP Line** (not shown). On the **SIP Line** tab in the Details Pane, configure the parameters below to connect to the SIP Trunking service.

- Set the **ITSP Domain Name** to the domain name provided by HIPCOM, partially removed for security purposes
- Set **Send Caller ID** to ***None***. This parameter determines how the calling party number is sent in the SIP messaging for twinning if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4.** This parameter was set to ***None*** and the box in **Section 5.4** was checked
- Ensure the **In Service** box is checked
- Default values may be used for all other parameters

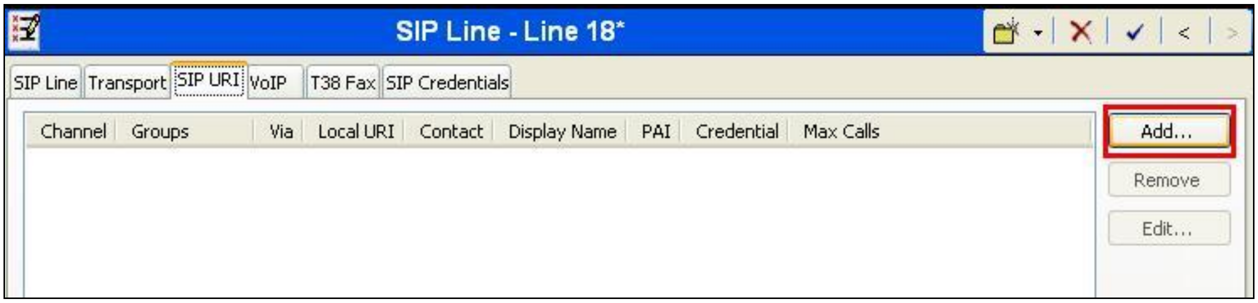


Select the **Transport** Tab and set the following:

- Set **ITSP IP Address** to the IP address of the HIPCOM SIP proxy
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** and **Listen Port** to **5060**



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

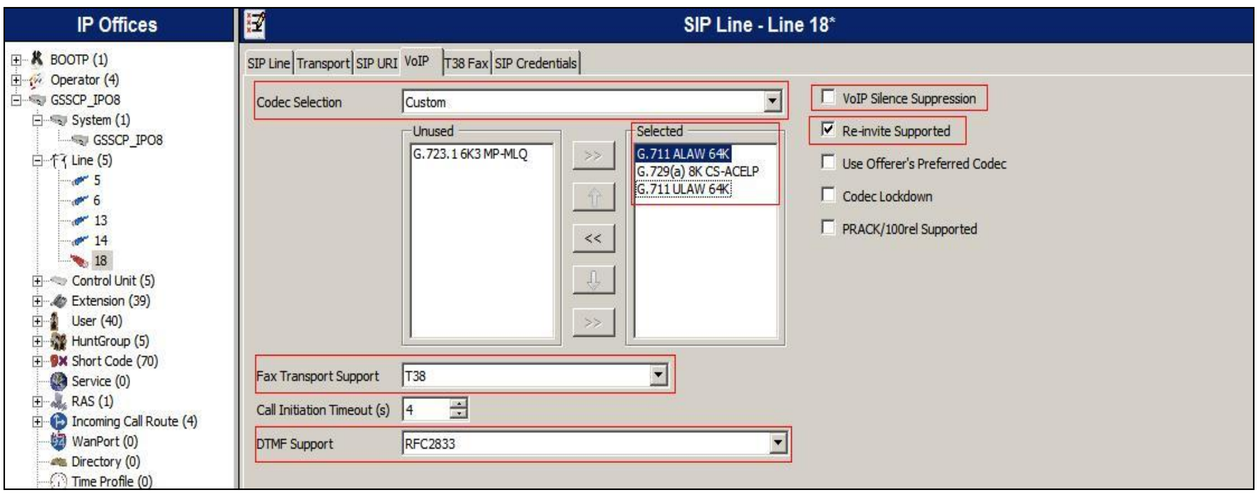


For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

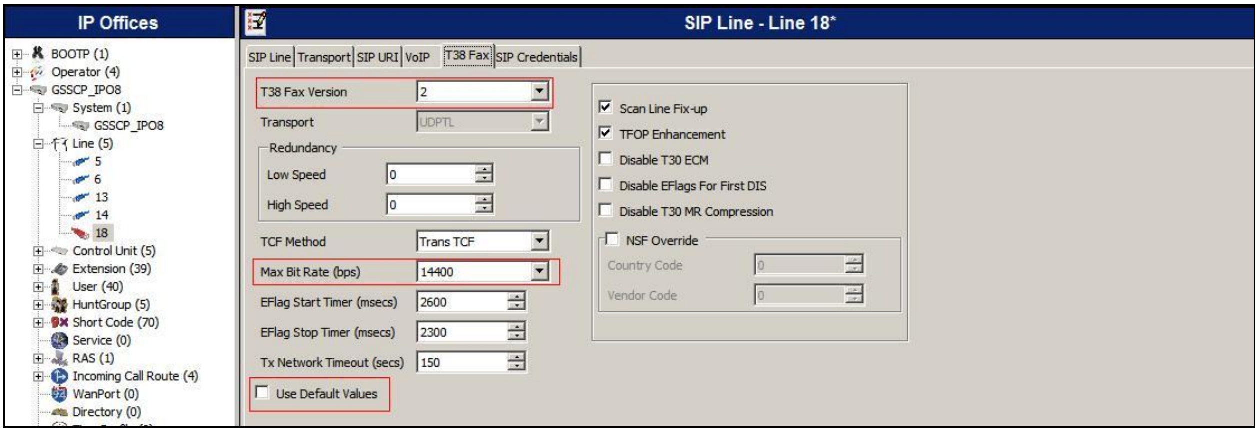
- Set **Local URI** to **Use Internal Data.** This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**
- Set **Contact, Display Name** and **PAI** to **Use Credentials User Name.** This value is set on the credentials tab in **Section 5.5.1**
- For **Registration**, select ***1: 442031111111*** from the pull-down menu.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **18** was defined that was associated to a single line (line 18)
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern

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

- Configure the **Codec Selection** as **Custom**. In the **Selected** box select the codecs and the order based on the needs of the customer. For the compliance test, **G.711 ALAW 64K** was selected first followed by **G.729 (a) 8K CS-ACELP** then by **G.711 ULAW 64K**
- 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 **Fax Transport Support** box to allow T.38 fax operation
- 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
- Default values may be used for all other parameters

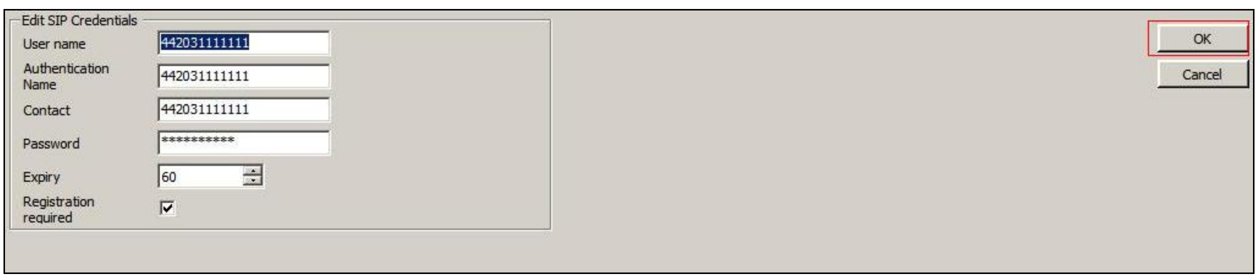


Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the **Use Default Values** box and select **2** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate(bps)** to **14400**. All other field may retain their default values.



5.5.1. Setting the Registration Credentials

Select the **SIP Credentials** tab to administer registration details provided by HIPCOM. This allows the SIP Trunk to authenticate to the HIPCOM SIP Trunking solution. Choose **Add** (not shown) and enter the registration credentials provided by HIPCOM as shown below. Click the **OK** button to complete the SIP line administration



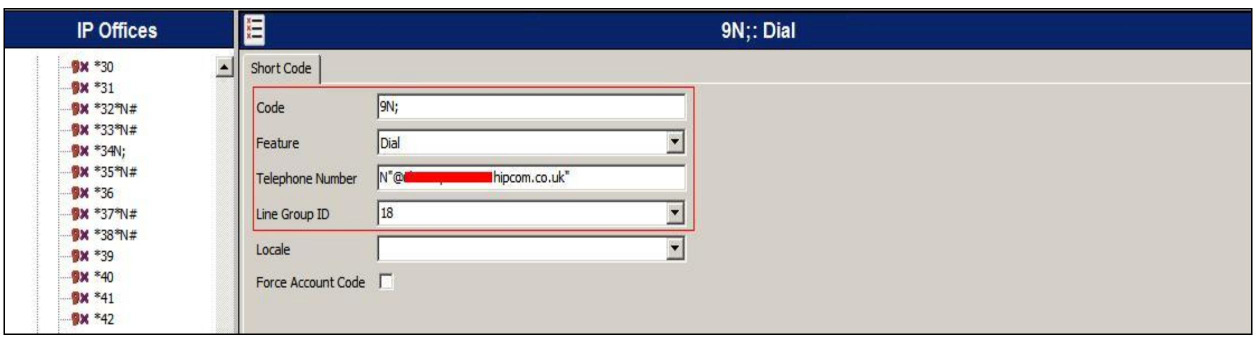
Note: It is advisable at this stage to save the configuration as described in **Section 5.9** to make the Line Group ID available in **Section 5.6**.

5.6. 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** (not shown). 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. In this case, **9N;**. This short code will be invoked when the user dials 9 followed by any number
- Set **Feature** to **Dial**. This is the action that the short code will perform
- Set **Telephone Number** to **N"@xx.xx.xxxxx.hipcom.co.uk"**. This field is used to construct the Request URI and To headers in the outgoing SIP INVITE message. The value **N** represents the number dialed by the user. **xx.xx.xxxxx.hipcom.co.uk** is the SIP domain.
- Set the **Line Group Id** to the outgoing line group number defined on the **SIP URI** tab on the **SIP Line** in **Section 5.5**. This short code will use this line group when placing the outbound call

Click the **OK** button (not shown).



5.7. User

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **SIP** tab in the Details Pane. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls and 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 5.5**). As such, these fields should be set to one of the DID numbers assigned to the enterprise from HIPCOM.

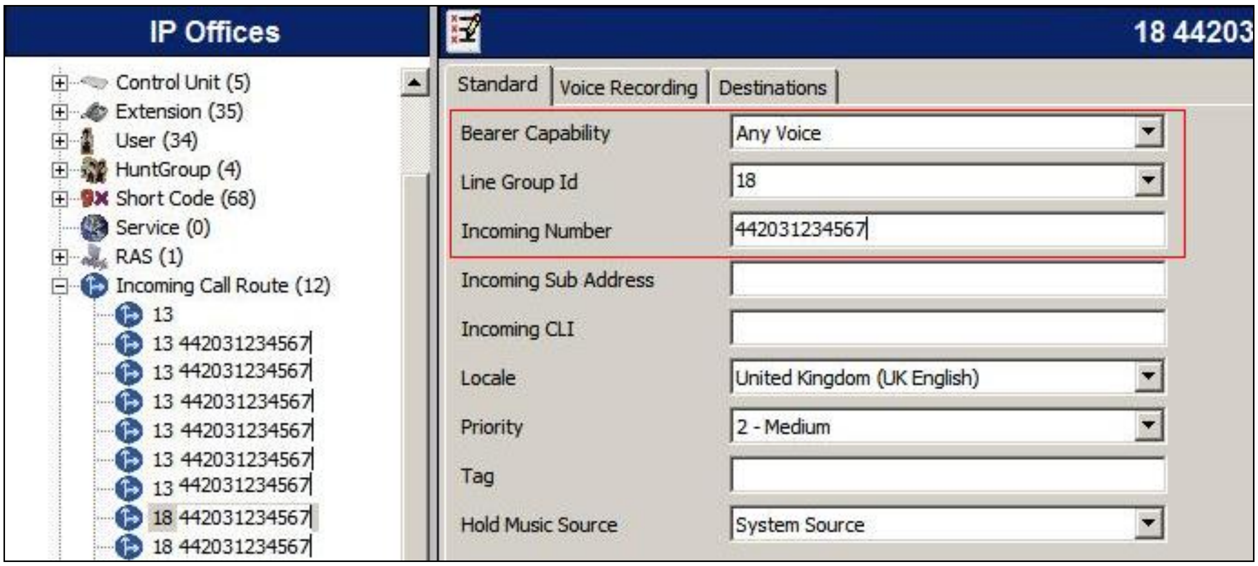
In the example below, the DID number **442031111111** is used. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. Click the **OK** button (not shown).

Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly.

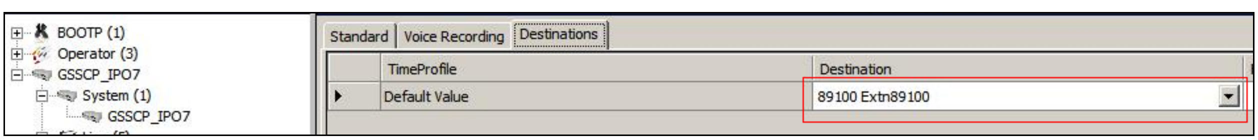
5.8. Incoming Call Routing

An incoming call route maps an inbound DID number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New** (not shown). On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.



On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. Click the **OK** button (not shown). In this example, incoming calls to 442031234567 on line 18 are routed to extension 89100.



5.9. 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. BT/HIPCOM SIP Trunking Service Configuration

BT/HIPCOM is responsible for the configuration of the SIP Trunking service. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. BT/HIPCOM will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

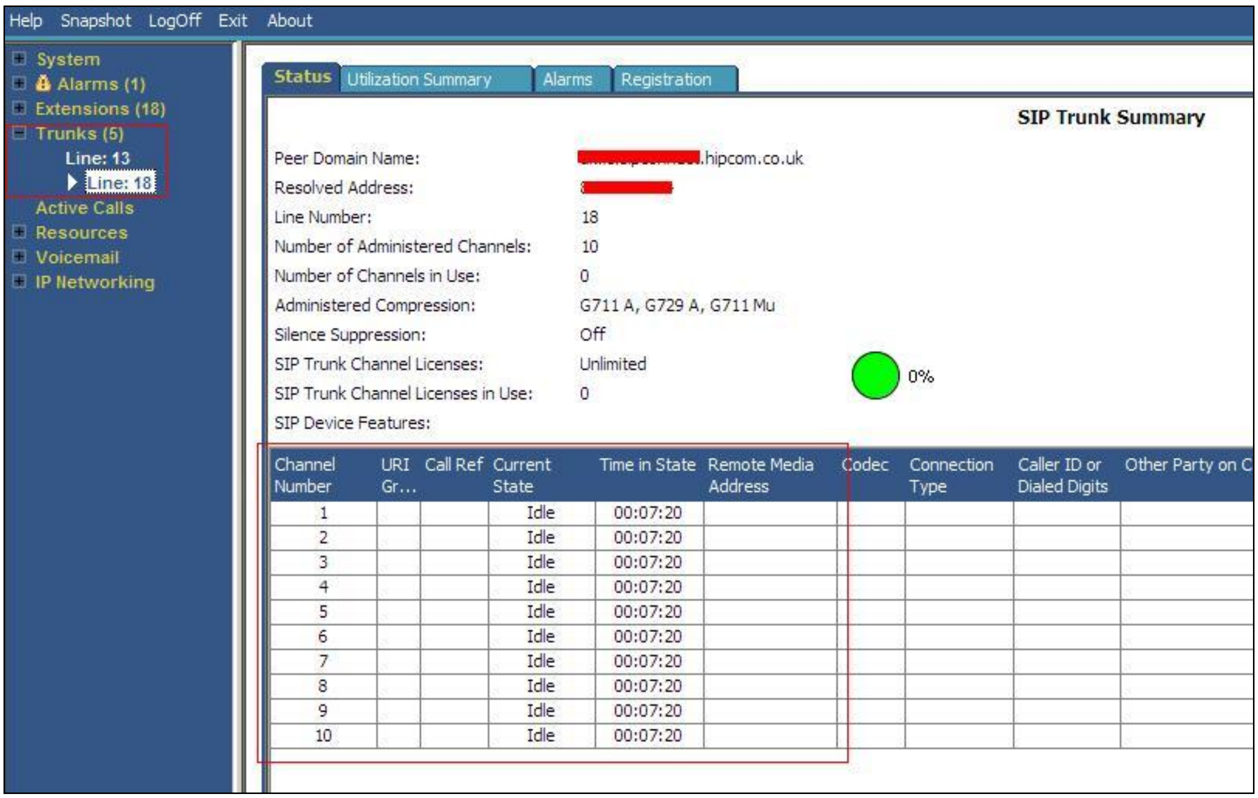
- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DID numbers
- All IP addresses and port numbers used for signaling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

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

7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found under **Start →All Programs →IP Office →System Status.** From the left hand menu expand **Trunks** and choose the SIP trunk (18 in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.



8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the BT/HIPCOM SIP Trunking service 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 can be configured to interoperate successfully with the BT/HIPCOM SIP Trunking Service. This solution provides IP Office users the ability to access the Public Switched Telephone Network (PSTN) via a SIP trunk using the BT/HIPCOM SIP Trunking service.

9. Additional References

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

[1] IP Office 8 Documentation CD, April 2012.
[2] IP Office Installation, Document number15-601042, April 2012.
[3] IP Office Manager, Document number15-601011, April 2012.
[4] System Status Application, Document number15-601758, April 2012.

Product documentation for the BT/HIPCOM SIP Trunking service is available from http://www.hipcom.co.uk/support

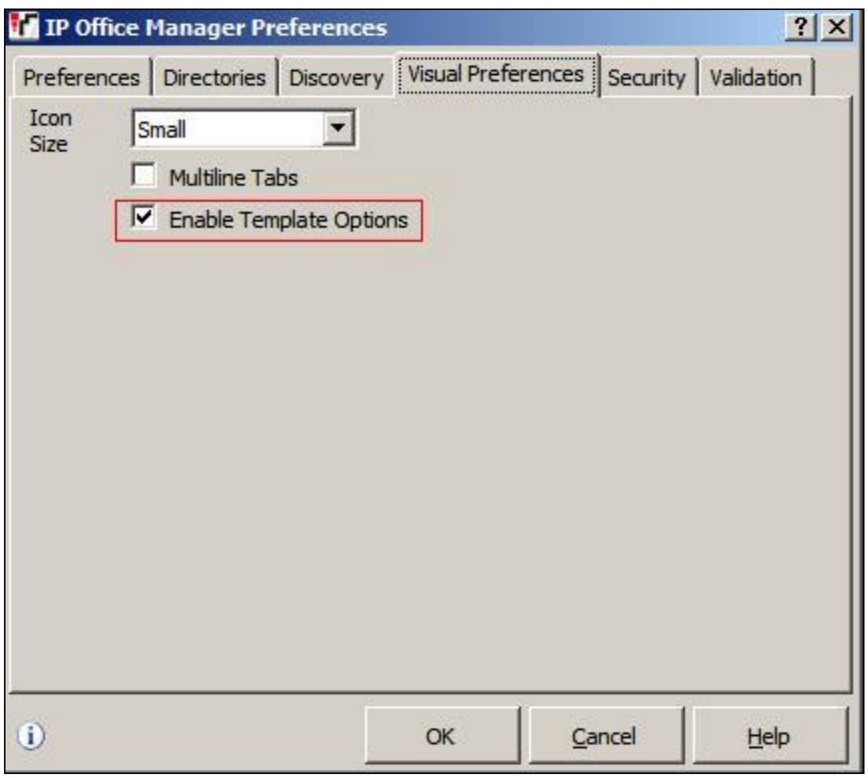
Appendix: SIP Line Template

Avaya IP Office Release 8.0 supports a SIP Line Template (in xml format) that can be created from an existing configuration and imported into a new installation to simplify configuration procedures as well as to reduce potential configuration errors.

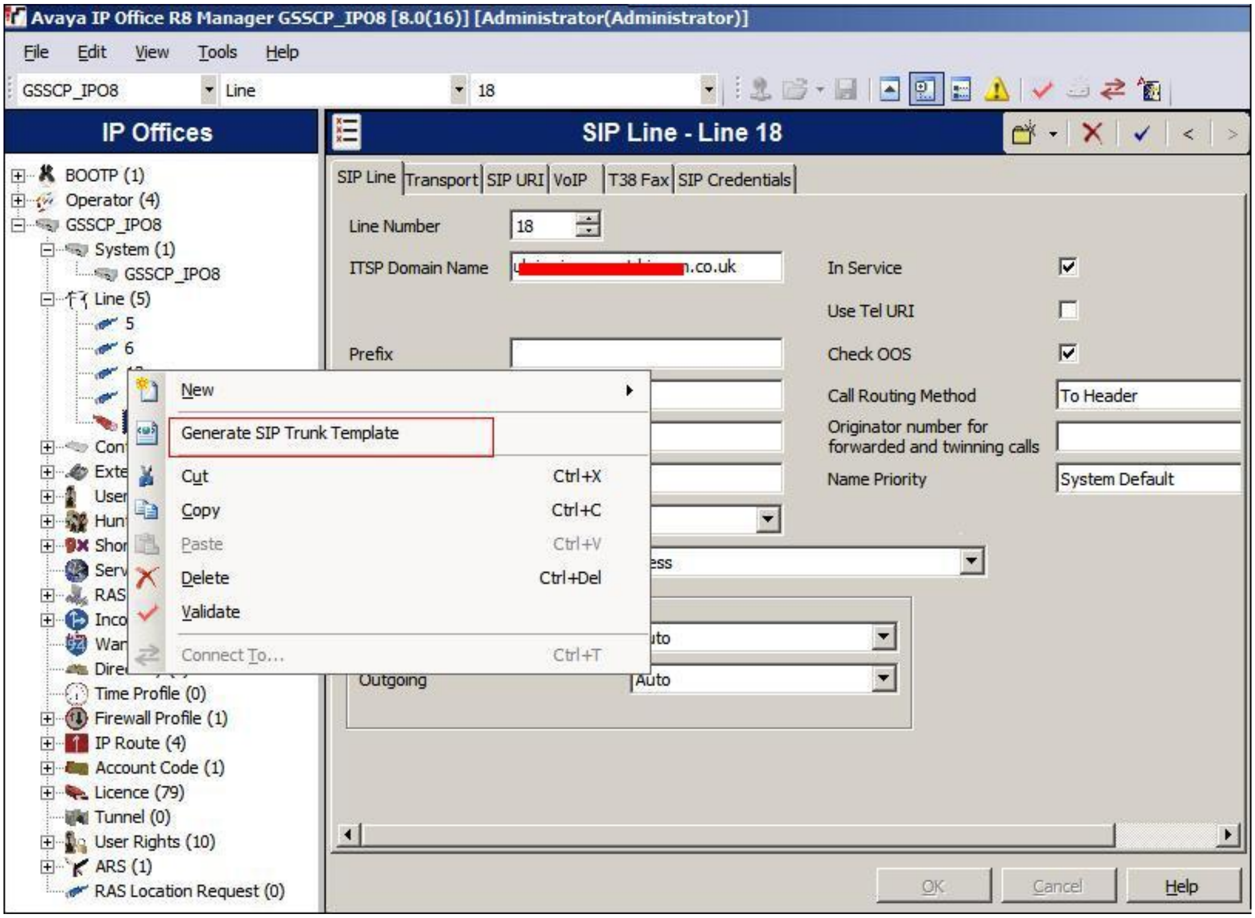
Not all of the configuration information is included in the SIP Line Template, therefore, it is critical that the SIP Line configuration be verified/updated after a template has been imported, and additional configuration be supplemented using **Section 5.5** in these Application Notes as a reference.

To create a SIP Line Template from the configuration described in these Application Notes, configure the parameters as described below.

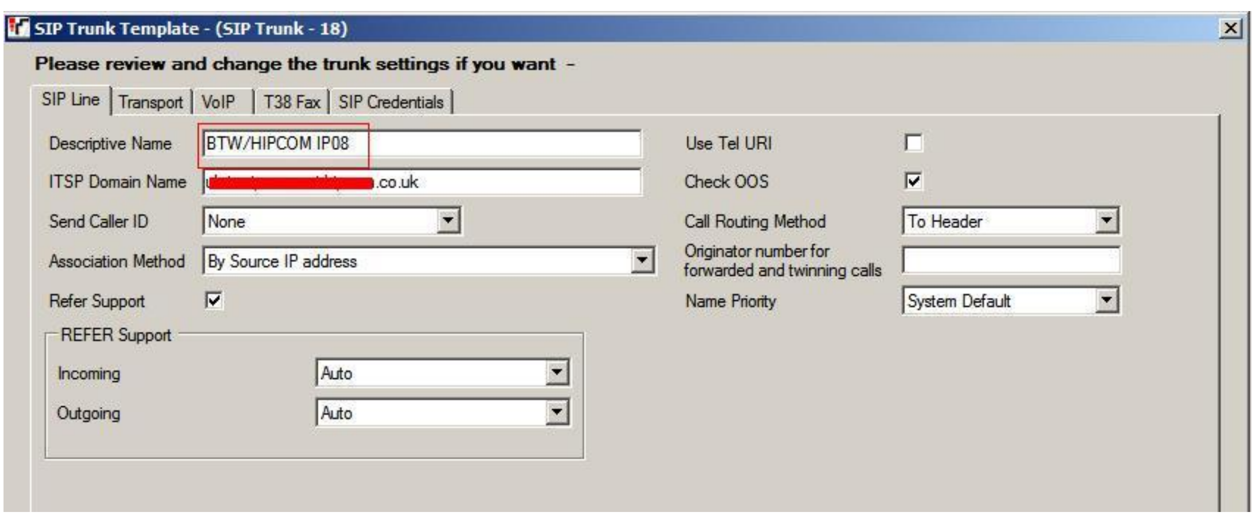
To enable template support, select **File**, then **Preferences**. On the **Visual Preferences** tab, check the **Enable Template Options** box.



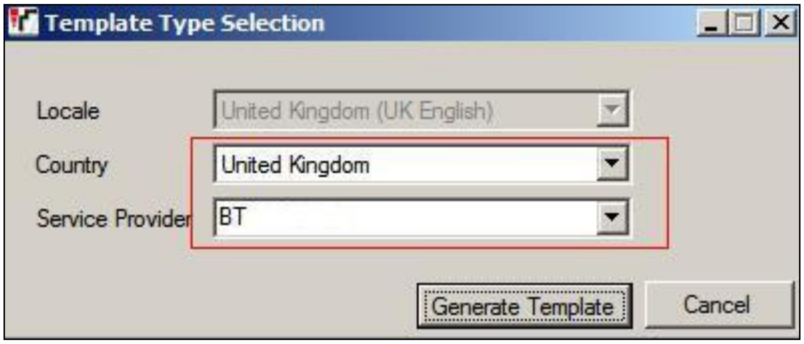
To create a SIP Line Template from the configuration, on the left Navigation Pane, right click the Sip Line (**18**), and select **Generate SIP Trunk Template**.



The trunk's settings are displayed as configured in **Section 5.5.** Enter a descriptive name for the template, adjust the settings if required, and then click on **Export**.



On the next screen, **Template Type Selection**, select the **Country,** enter the name for the **Service Provider**, and click **Generate Template**.



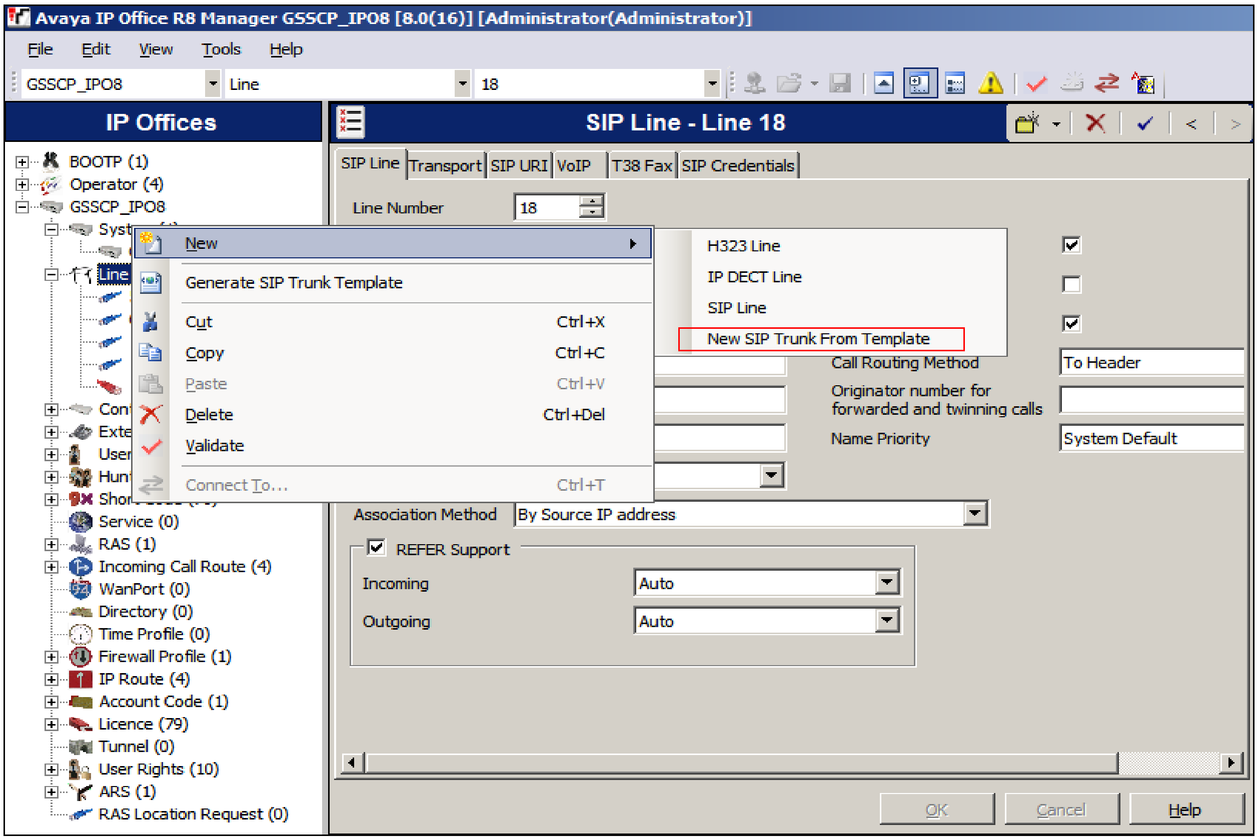
The following is the exported SIP Line Template file **GB_BT_SIPTrunk.xml,** domain name and ip address are masked for security purposes:


```
 <?xml version="1.0" encoding="utf-8" ?>
-<Template xmlns="urn:SIPTrunk-schema">
  <TemplateType>SIPTrunk</TemplateType>
  <Version>20120710</Version>
  <SystemLocale>eng</SystemLocale>
  <DescriptiveName>BTW/HIPCOM IP08</DescriptiveName>
  <ITSPDomainName>xx.xx.xxxxxxxxx.hipcom.co.uk</ITSPDomainName>
  <SendCallerID>CallerIDNone</SendCallerID>
  <ReferSupport>true</ReferSupport>
  <ReferSupportIncoming>2</ReferSupportIncoming>
  <ReferSupportOutgoing>2</ReferSupportOutgoing>
  <RegistrationRequired>false</RegistrationRequired>
  <UseTelURI>false</UseTelURI>
  <CheckOOS>true</CheckOOS>
  <CallRoutingMethod>0</CallRoutingMethod>
  <OriginatorNumber />
  <AssociationMethod>SourceIP</AssociationMethod>
  <LineNamePriority>SystemDefault</LineNamePriority>
  <ITSPProxy>10.10.10.20</ITSPProxy>
  <LayerFourProtocol>SipUDP</LayerFourProtocol>
  <SendPort>5060</SendPort>
  <ListenPort>5060</ListenPort>
  <DNSServerOne>0.0.0.0</DNSServerOne>
  <DNSServerTwo>0.0.0.0</DNSServerTwo>
  <CallsRouteViaRegistrar>true</CallsRouteViaRegistrar>
  <SeparateRegistrar />
  <CompressionMode>G72316K3</CompressionMode>
  <UseAdvVoiceCodecPrefs>true</UseAdvVoiceCodecPrefs>
  <AdvCodecPref>G.711 ALAW 64K,G.729(a) 8K CS-ACELP,G.711 ULAW
    64K</AdvCodecPref>
  <CallInitiationTimeout>4</CallInitiationTimeout>
```

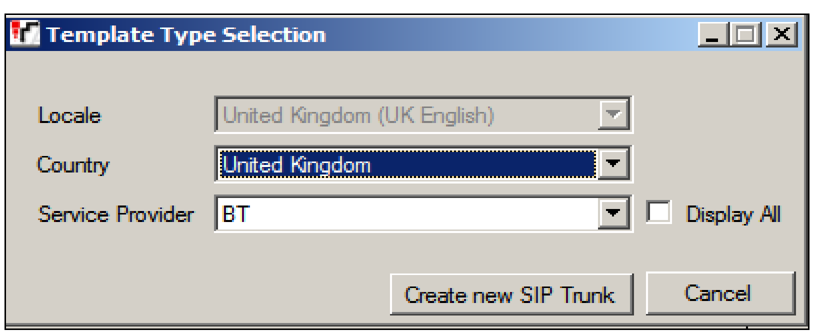
```
<DTMFSupport>DTMF_SUPPORT_RFC2833</DTMFSupport>
<VoipSilenceSupression>false</VoipSilenceSupression>
<ReinviteSupported>true</ReinviteSupported>
<FaxTransportSupport>FOIP_T38</FaxTransportSupport>
<UseOffererPrefferedCodec>false</UseOffererPrefferedCodec>
<CodecLockdown>false</CodecLockdown>
<Rel100Supported>false</Rel100Supported>
<T38FaxVersion>2</T38FaxVersion>
<Transport>UDPTL</Transport>
<LowSpeed>0</LowSpeed>
<HighSpeed>0</HighSpeed>
<TCFMethod>Trans_TCF</TCFMethod>
<MaxBitRate>FaxRate_14400</MaxBitRate>
<EflagStartTimer>2600</EflagStartTimer>
<EflagStopTimer>2300</EflagStopTimer>
<UseDefaultValues>false</UseDefaultValues>
<ScanLineFixup>true</ScanLineFixup>
<TFOPEnhancement>true</TFOPEnhancement>
<DisableT30ECM>false</DisableT30ECM>
<DisableEflagsForFirstDIS>false</DisableEflagsForFirstDIS>
<DisableT30MRCompression>false</DisableT30MRCompression>
<NSFOverride>false</NSFOverride>
- <SIPCredentials>
<Expiry>60</Expiry>
<RegistrationRequired>true</RegistrationRequired>
  </SIPCredentials>
  </Template>
```

To import the template into a new IP Office system, copy and paste the exported xml template file to the Templates directory (C:\Program Files\Avaya\IP Office\Manager\Templates) on the PC where IP Office Manager for the new system is running.

Next, import the template into the new IP Office system by creating a new SIP Line as shown in the screenshot below. In the Navigation Pane on the left, right-click on **Line** then navigate to **New, New SIP Trunk From Template**:



On the next screen, **Template Type Selection**, verify that the information in the **Country** and **Service Provider** fields is correct. If more than one template is present, use the drop-down menus to select the required template. Click **Create new SIP Trunk** to finish the process.



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.