



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 10.0 with the BT Business SIP Trunk platform – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the BT Business SIP Trunk platform and Avaya IP Office. Note that the BT Business SIP Trunk platform uses Broadsoft BroadWorks, formally HIPCOM in the UK.

The BT Business SIP Trunk platform provides PSTN access via BroadWorks SIP Trunking as an alternative to legacy analogue or digital trunks. BT 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 the BT Business SIP Trunk platform and Avaya IP Office. The BT Business SIP Trunk platform provides PSTN access via BroadWorks SIP Trunking (formally HIPCOM in the UK) as an alternative to legacy analogue or digital trunks. This approach generally results in lower cost for customers.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to the BT Business SIP Trunk platform. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

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

2.1. Interoperability Compliance Testing

Avaya IP Office was connected to the BT Business SIP Trunk platform. To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types including H.323, SIP, digital and analogue telephones at the enterprise. Calls were routed to the enterprise across the SIP trunk from BT Business.
- Outgoing PSTN calls from various phone types including H.323, SIP and analogue telephones at the enterprise. Calls were routed from the enterprise across the SIP trunk to BT Business.
- Inbound and outbound PSTN calls to/from an Avaya Communicator for Windows client.
- Various call types including: local, international, toll free (outbound) and directory assistance.
- Codecs G.729A, G.711A and G.711MU.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38.
- Caller ID presentation and Caller ID restriction.
- DTMF transmission using RFC 2833.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and mobile twinning.

2.2. Test Results

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

- The BT Business SIP Trunk platform doesn't send OPTIONS messages to monitor the SIP Line
- Outbound call failures, in particular unanswered calls and calls to unallocated numbers, received appropriate announcements from the network but signalling indicates normal call clearing.
- When no matching codec is found on outbound calls, the network sends "480 Temporarily Unavailable", a more commonly used response for this failure is "488 Not Acceptable Here"
- Inbound Toll-Free access was not available for test
- Calls to Emergency Services were not checked as a test call was not booked with the Emergency Services Operator
- The network call hold test was not a true test of SIP trunk functionality as it was handled in the network and there was nothing in signalling to indicate the call was on hold. Similarly, IP Office provides no indication to the network that a call is on hold.

2.3. Support

For technical support on HIPCOM products please contact the HIPCOM support team at: <http://support.hipcom.co.uk/support> or <http://ipvoicesupport.btwholesale.com> or <http://info.broadcloudpbx.com>.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to BT Business SIP Trunk. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), an Avaya 1140e SIP Telephone, an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as Avaya Communicator for Windows for mobility testing. For security purposes, public IP addresses have been changed and any PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead the phone numbers have been obscured beyond the city code.

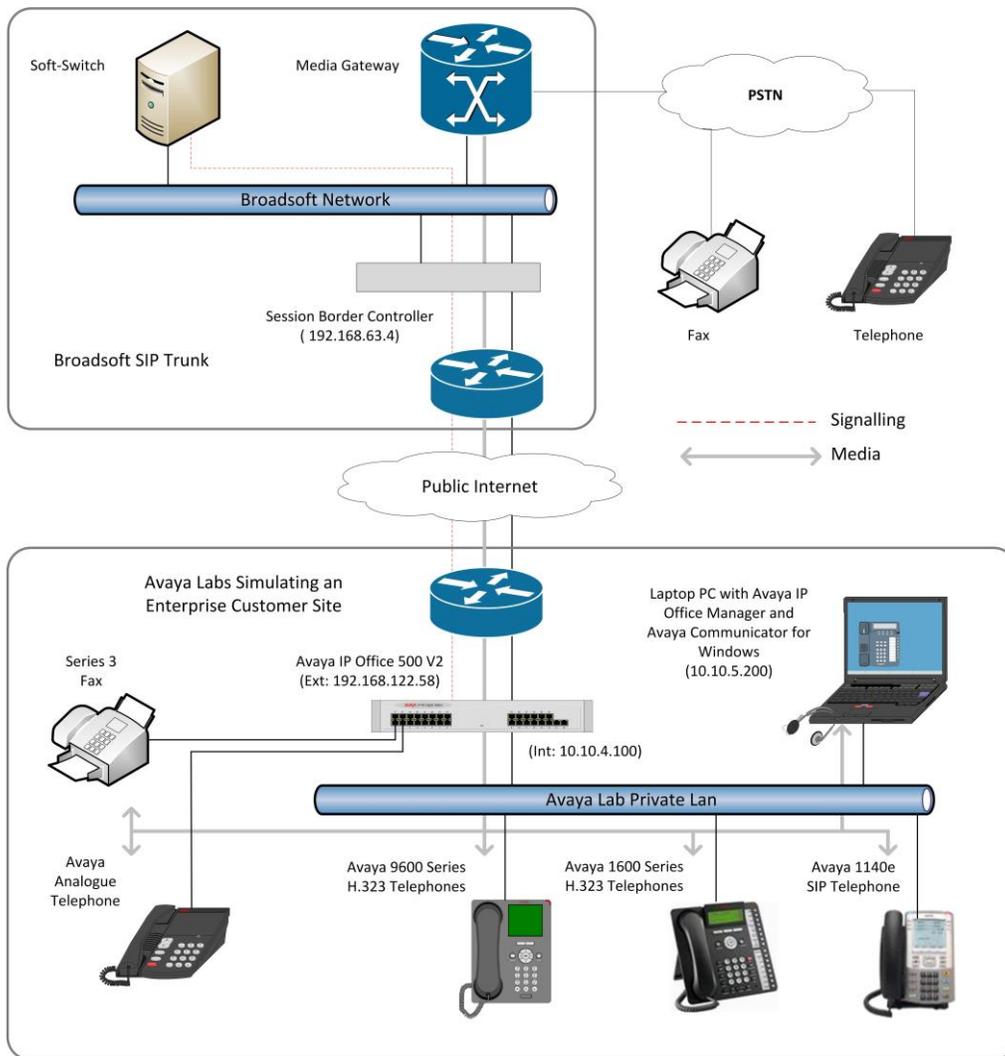


Figure 1: BT Business SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

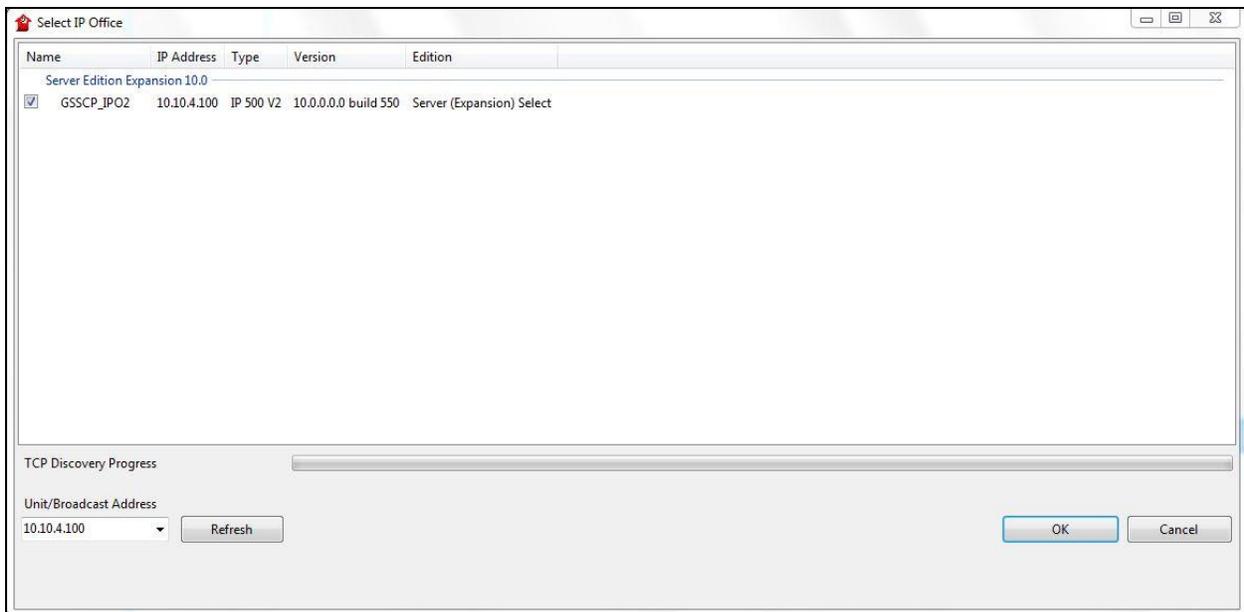
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	10.0.0.0.0 Build 550
Avaya 1140e IP SIP Telephone	04.04.23.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.6.2.29 V474
Avaya 98390 Analogue Phone	N/A
Avaya Communicator	2.1.3.802.1.3.0-NGUE-FLAREWINIPOREGRESSION10-JOB1.237
Avaya IP Office Manager	Version 10.0.0.0.0 build 550
HIPCOM SIP Trunking	
Acme Packet 6300 Net-Net SBC	SCX7.2.0
BroadWorks	17 Service Pack 4
Configuration version	HIPCOM v8.2.SIPConnect v1.0

Testing was performed with IP Office 500 V2 R10.0. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

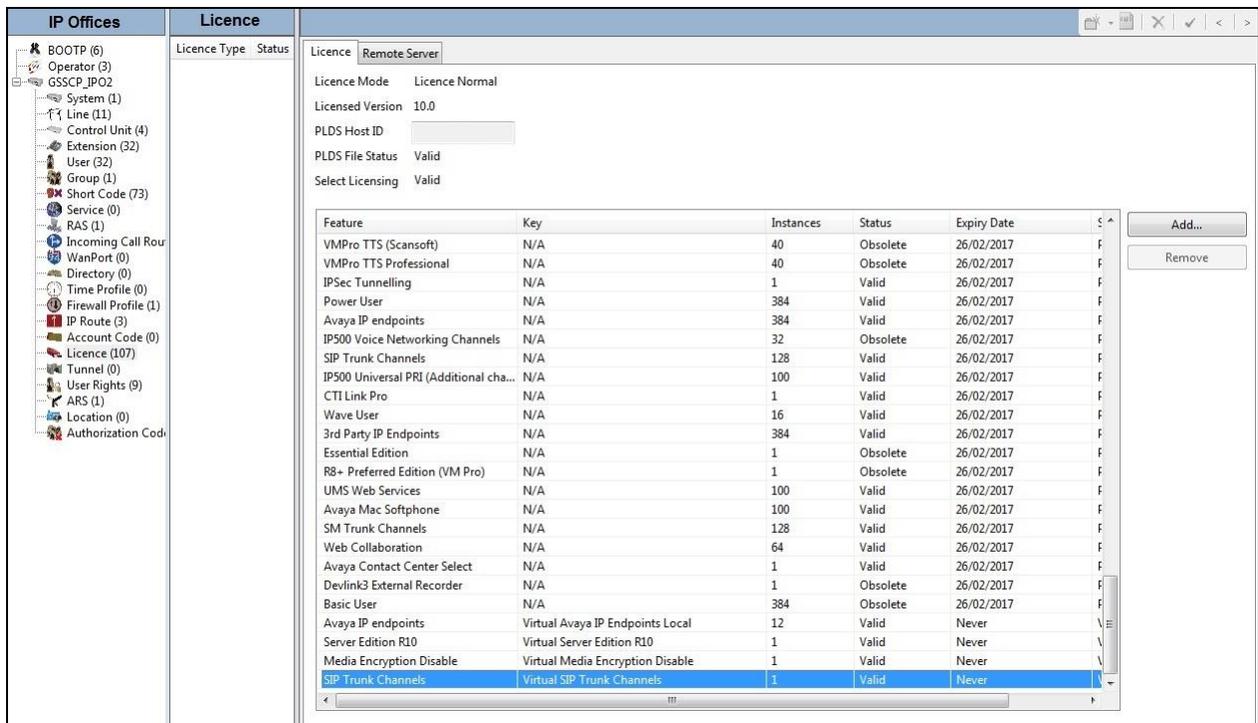
5. Configure Avaya IP Office

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



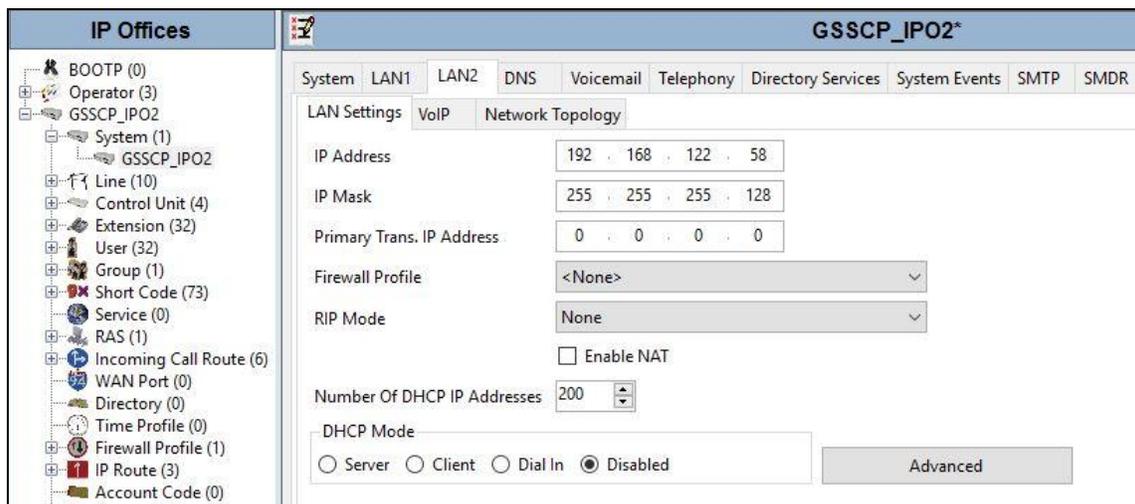
5.1. Verify System Capacity

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



5.2. LAN2

In the sample configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System** → **<IP Office Name>** in the Navigation Pane where IP Office Name is the name of the IP Office. This is **GSSCP_IPO2** in the GSSCP test environment. Navigate to the **LAN2** → **LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the IP Office, **Primary Trans. IP Address** is the next hop, usually the default gateway, though this is not set in the test environment as a default route is specified in **IP Route** (not shown). All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).



On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If SIP Endpoints are to be used such as the Avaya Communicator for Windows and the Avaya 1140e, the **SIP Registrar Enable** box must also be checked. Define the port to be used for the signalling transport, in the test environment **UDP** was used and the port number was left at the default value of **5060**.

Scroll down for further configuration. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office requests RTP media to be sent to a UDP port in the configurable range for calls using LAN2. The range used for testing was 50000 to 60000, though in this case the default values of **49152** to **53246** would have been equally effective.

The screenshot displays the configuration interface for GSSCP_IPO2*. The interface is divided into several sections:

- System Settings:** Includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, VoIP, VoIP Security, and Contact Center. The current view is under LAN Settings > VoIP > Network Topology.
- H.323 Settings:**
 - H.323 Gatekeeper Enable
 - Auto-create Extension
 - Auto-create User
 - H.323 Remote Extension Enable
 - H.323 Signaling over TLS: Disabled
 - Remote Call Signaling Port: 1720
- SIP Settings:**
 - SIP Trunks Enable
 - SIP Registrar Enable
 - Auto-create Extension/User
 - SIP Remote Extension Enable
 - SIP Domain Name: avaya.com
 - SIP Registrar FQDN: (empty)
 - Layer 4 Protocol:
 - UDP, UDP Port: 5060, Remote UDP Port: 5060
 - TCP, TCP Port: 5060, Remote TCP Port: 5060
 - TLS, TLS Port: 5061, Remote TLS Port: 5061
 - Challenge Expiration Time (sec): 10
- RTP Settings:**
 - Port Number Range:
 - Minimum: 50000, Maximum: 60000
 - Port Number Range (NAT):
 - Minimum: 49152, Maximum: 53246
 - Enable RTCP Monitoring on Port 5005
 - RTCP collector IP address for phones: 0 . 0 . 0 . 0
 - Keepalives:
 - Scope: RTP
 - Periodic timeout: 1
 - 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

Note: Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signalling and media (not shown). DSCP for media can be set for both voice and video. The **DSCP** field is the value used for voice and the **SIG DSCP** is the value used for signalling. For the compliance test, the DSCP values were left at their default values.

All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

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

The Network Topology tab can be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. During testing, IP Office sent OPTIONS messages at an interval of 5 minutes. This was achieved by setting the **Binding Refresh Time** to **300**.

The screenshot shows the configuration window for GSSCP_IP02*. The 'Network Topology' tab is selected. The 'Network Topology Discovery' section contains the following settings:

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

Buttons for 'Run STUN' and 'Cancel' are visible at the bottom right of the configuration area.

5.3. System Telephony Settings

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

The screenshot displays the configuration interface for GSSCP_IPO2*. The 'Telephony' tab is active, and the 'Companding Law' section is highlighted with a red box. The 'Switch' and 'Line' sub-sections are also highlighted. The 'Switch' section has 'A-Law' selected, and the 'Line' section has 'A-Law Line' selected. The 'Inhibit Off-Switch Forward/Transfer' checkbox is highlighted with a red box and is unchecked. Other settings include 'Default Outside Call Sequence' set to 'Normal', 'Default Inside Call Sequence' set to 'Ring Type 1', and 'Default Ring Back Sequence' set to 'Ring Type 2'. The 'Dial Delay Time (sec)' is set to 4, 'Dial Delay Count' to 0, 'Default No Answer Time (sec)' to 15, 'Hold Timeout (sec)' to 0, 'Park Timeout (sec)' to 300, and 'Ring Delay (sec)' to 5. The 'Call Priority Promotion Time (sec)' is set to 'Disabled', 'Default Currency' to 'EUR', 'Default Name Priority' to 'Favor Trunk', 'Media Connection Preservation' to 'Disabled', and 'Phone Failback' to 'Manual'. Other settings include 'DSS Status' (unchecked), 'Auto Hold' (checked), 'Dial By Name' (checked), 'Show Account Code' (checked), 'Restrict Network Interconnect' (unchecked), 'Include location specific information' (unchecked), 'Drop External Only Impromptu Conference' (unchecked), 'Visually Differentiate External Call' (unchecked), 'Unsupervised Analog Trunk Disconnect Handling' (unchecked), 'High Quality Conferencing' (checked), and 'Digital/Analogue Auto Create User' (unchecked).

5.4. 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.729(a) 8K CS-ACELP**, **G.711 ALAW 64K** and **G.711 ULAW 64K** were used as default codecs. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).

The screenshot displays the configuration interface for GSSCP_IPO2*. The interface includes a navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, and VoIP. The DNS tab is currently selected. Below the navigation bar, there are several configuration options: 'Ignore DTMF Mismatch For Phones' (checkbox), 'Allow Direct Media Within NAT Location' (checkbox), and 'RFC2833 Default Payload' (dropdown menu set to 101). The main section is titled 'Available Codecs' and is divided into three columns: 'Available Codecs', 'Default Codec Selection', and 'Selected'. The 'Available Codecs' column contains a list of codecs with checkboxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' column is further divided into 'Unused' and 'Selected' sub-columns. The 'Unused' sub-column contains G.722 64K and G.723.1 6K3 MP-MLQ. The 'Selected' sub-column contains G.729(a) 8K CS-ACELP, G.711 ALAW 64K, and G.711 ULAW 64K. Between the 'Unused' and 'Selected' sub-columns are four buttons: '>>>', an up arrow, '<<<', and a down arrow. The 'G.711 ULAW 64K' and 'G.711 ALAW 64K' options in the 'Available Codecs' list are greyed out.

5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the BT Business SIP Trunk. 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 5.5.1** 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 5.5.2**.

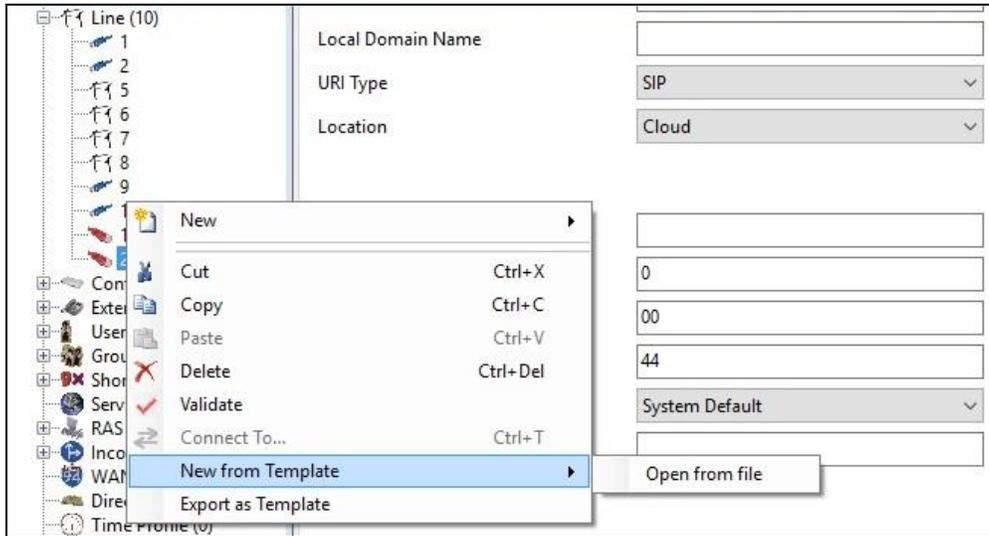
Also, the following SIP Line settings are not supported on Basic Edition:

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

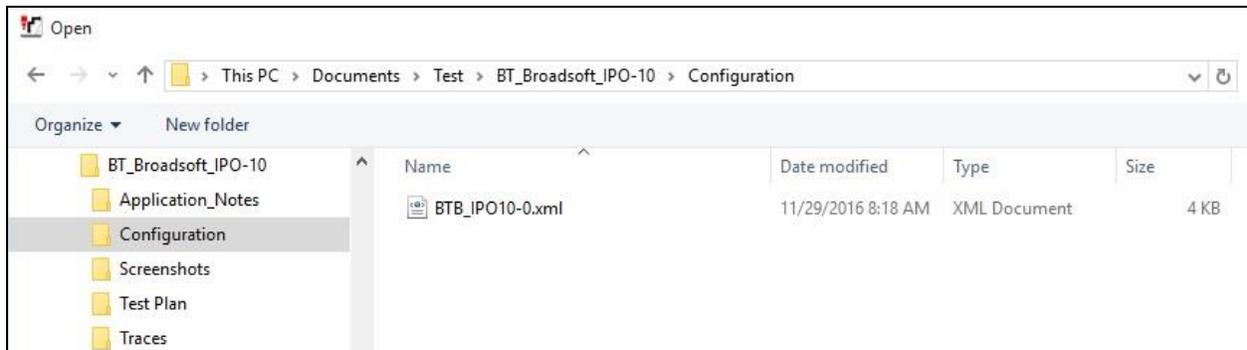
Alternatively, a SIP Line can be created manually. To do so, right-click Line in the Navigation Pane and select **New→SIP Line** (not shown). Then, follow the steps outlined in **Section 5.5.2**.

5.5.1. SIP Line From Template

Copy the template file to the computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New** → **New from Template**.



Navigate to the directory on the local machine where the template was copied and select the template as required.



The SIP Line is automatically created and can be verified and edited as required using the configuration described in **Section 5.6.2**.

5.5.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to a domain name agreed with BT Business. In test this was set to **uk.ic.sipconnect.hipcom.co.uk**. This ensures that private IP address isn't used in any SIP messages from IP Office to the network.
- Leave **Prefix blank** and set the **National Prefix** and **International Prefix** to those used in the UK. This ensures that Calling Party Numbers are presented on the IP Office extensions in diallable format. It also removes the prefixes on outgoing dialled numbers for conversion to E.164 format.
- Set **Country Code** to 44 for the UK, this prefixes the country code on outgoing dialled numbers for conversion to E.164 format.
- Check the **Check OOS** box so that the SIP Trunk is taken out of service when there is no response to OPTIONS.
- Ensure the **In Service** box is checked.

The screenshot shows the configuration window for 'SIP Line - Line 20'. The left pane shows a tree view of 'IP Offices' with 'Line (10)' selected. The main pane has tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', 'SIP Credentials', 'SIP Advanced', and 'Engineering'. The 'SIP Line' tab is active, showing the following configuration:

Line Number	20	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	uk.ic.sipconnect.hipcom.co.uk	Check OOS	<input checked="" type="checkbox"/>
Local Domain Name			
URI Type	SIP	Session Timers	
Location	Cloud	Refresh Method	Auto
		Timer (sec)	On Demand
Prefix		Redirect and Transfer	
National Prefix	0	Incoming Supervised REFER	Auto
International Prefix	00	Outgoing Supervised REFER	Auto
Country Code	44	Send 302 Moved Temporarily	<input type="checkbox"/>
Name Priority	System Default	Outgoing Blind REFER	<input type="checkbox"/>
Description			

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

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

- Set **ITSP Proxy Address** to the domain name for the BT Business SIP Trunk platform.
- Set **Use Network Topology Info** to **None** as NAT is not used in this configuration and the Network Topology settings defined in **Section 5.2** are not required.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** and **Listen Port** to **5060**.

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

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' is set to 'uk.ic.sipconnect.hipcom.co.uk'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is set to 'None', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are both set to '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

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

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'SIP URI' tab selected. The table below shows one entry with columns: URI, Groups, Local URI, Contact, Display Name, Identity, Header, Originator Number, Send Caller ID, Diversion Header, Credential, Max Calls. There are 'Add...' and 'Remove' buttons on the right.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls

Only one SIP URI is shown in this example using internal data. That means that for incoming SIP INVITE messages, the user part of the Request URI is matched to the SIP settings for the Users as described in **Section 5.7**. Where the user can't be matched, which is the case in calls to voicemail and the Mobile Twinning FNE for example, the SIP INVITE is rejected with a "404 Not Found". To avoid this, an additional incoming SIP URI can be defined with a wildcard (*) as the **Local URI**, or a SIP URI can be defined for each number that is not associated with a User. During testing an additional SIP URI was used with a wildcard and also a different Incoming Group (19).

The entry for IP Office extensions was created with the parameters shown below.

- Set **Local URI**, **Contact** and **Display Name** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.7**. It is the default setting when no SIP Credentials are specified.
- Set **Identity** to **Use Internal Data** and leave the Header at default **P Asserted ID**.
- Set the **Originator Number** for **Forwarding and Twinning** to a number in the DDI range assigned to the IP Office. During testing, the number used was the same as that used for SIP registration. Select **None** as the **Send Caller ID** Value to ensure that the **Originator Number** is used.
- Select **None** in the **Diversion Header** drop down menu as Diversion Header is not used.
- Leave the **Registration** field at the default value of **None** as registration credentials are not used in the SIP URI. During testing, this was set but is not required.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **20** was defined that was associated to a single line (line 20).
- Associate this line with an outgoing line group by entering a line group number in the **Outgoing Group** field. For the compliance test, a new outgoing group **20** was defined that was also associated to line 20.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

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

The screenshot shows the 'Edit URI' configuration form with the following settings:

- Local URI:** Use Internal Data
- Contact:** Use Internal Data
- Display Name:** Use Internal Data
- Identity:**
 - Identity:** Use Internal Data
 - Header:** P Asserted ID
- Forwarding And Twinning:**
 - Originator Number:** 4420355nnnn0
 - Send Caller ID:** None
- Diversion Header:** None
- Registration:** 0: <None>
- Incoming Group:** 20
- Outgoing Group:** 20
- Max Sessions:** 10

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

- In **Section 5.5**, system default codecs were defined. If any other codec combination is required for this SIP Line, select **Custom** in the **Codec Selection** drop down menu.
- Highlight codecs in the **Unused** box that are to be used on this line and click on the right arrows to move them to the **Selected** box.
- Highlight codecs in the **Selected** box that are not to be used and click on the left arrows to move them to the **Unused** box.
- Highlight codecs in the **Selected** box and use the up and down arrows to change the priority order of the offered codecs, for testing with BT Business this was **G.729(a) 8K CS-ACELP**, **G.711 ALAW 64K** and **G.711 ULAW 64K**. This reflected the codec list received from the network.
- Select **T38 Fallback** in the **Fax Transport Support** drop down menu to allow both T.38 and G.711 fax operation.
- Select **RFC2833** in the **DTMF Support** drop down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- 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.
- Leave **Allow Direct Media Path** unchecked as direct media can't be used in this configuration.
- Check the **PRACK/100rel Supported** box if early media is required. This was checked during compliance testing.
- On completion, click the **OK** button (not shown).

The screenshot shows the configuration interface for a SIP Line (Line 20) in the VoIP tab. The interface includes several sections:

- Codec Selection:** A dropdown menu is set to "Custom". Below it are two boxes: "Unused" and "Selected". The "Unused" box contains "G.722 64K" and "G.723.1 6K3 MP-MLQ". The "Selected" box contains "G.729(a) 8K CS-ACELP", "G.711 ALAW 64K", and "G.711 ULAW 64K". Arrows between the boxes allow for moving and reordering codecs.
- Fax Transport Support:** A dropdown menu is set to "T38 Fallback".
- DTMF Support:** A dropdown menu is set to "RFC2833".
- Media Security:** A dropdown menu is set to "Media Security Features Disabled".
- Checkboxes:** On the right side, there are several checkboxes: "VoIP Silence Suppression" (unchecked), "Local Hold Music" (unchecked), "Re-invite Supported" (checked), "Codec Lockdown" (unchecked), "Allow Direct Media Path" (unchecked), "Force direct media with phones" (unchecked), "PRACK/100rel Supported" (checked), and "G.711 Fax ECAN" (unchecked).

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

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'T38 Fax' tab selected. The settings are as follows:

- T38 Fax Version: 3
- Transport: UDPTL
- Redundancy:
 - Low Speed: 0
 - High Speed: 0
- TCF Method: Trans TCF
- Max Bit Rate (bps): 14400
- EFlag Start Timer (ms): 2600
- EFlag Stop Timer (ms): 2300
- Tx Network Timeout (sec): 150
- Use Default Values:
- Advanced Settings (unchecked):
 - Scan Line Fix-up:
 - TFOP Enhancement:
 - Disable T30 ECM:
 - Disable EFlags For First DIS:
 - Disable T30 MR Compression:
 - NSF Override:
 - Country Code: 0
 - Vendor Code: 0

Select the **SIP Credentials** tab to configure the authentication parameters for the BT Business SIP Trunk platform. Click on Add (not shown).

The screenshot shows the 'SIP Line - Line 20' configuration window with the 'SIP Credentials' tab selected. The table below is used for managing SIP credentials:

Index	User Name	Authentication Name	Contact	Expiration (mins)	Register
<input type="button" value="Add..."/> <input type="button" value="Remove"/>					

The registration details are entered at the bottom of the screen.

The authentication takes place on the SIP registration. Define the login details in the **User name** and **Authentication Name** fields. Define the domain in the **Contact** field and the password in the **Password** field. Check the **Registration required** box.

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

- Check the **Send From In Clear** box to ensure that the calling number is inserted into the SIP From header even when the calling number is restricted. These numbers won't be displayed as the Privacy header is used to indicate that a number is restricted.
- Default values may be used for all other parameters.

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

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

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**. On the **Short Code** tab in the Details Pane, configure the parameters as shown in the example below for national numbers.

- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **9N;** which will be invoked when the user dials 9 followed by a public number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N** which removes the access code and inserts the public number into the Request URI and To headers in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.5**.
- On completion, click the **OK** button (not shown).

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' navigation pane lists various offices, with '9N;' selected. The main area displays the configuration for the selected short code, '9N;: Dial'. The configuration fields are as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	20
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

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

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

The screenshot shows the Avaya IP Office configuration interface for a short code. The configuration fields are as follows:

Field	Value
Code	9*67N;
Feature	Dial
Telephone Number	NW
Line Group ID	20
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.7. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required.

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

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

The screenshot displays the Avaya IP Office configuration interface for user 89022. On the left, a navigation tree shows the hierarchy: IP Offices > User (32) > 89022 Extn89022. The main panel is titled 'Extn89022: 89022' and contains several tabs: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The 'User' tab is active, showing the following configuration fields:

Name	Extn89022
Password	••••••••
Confirm Password	••••••••
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	
Extension	89022
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User

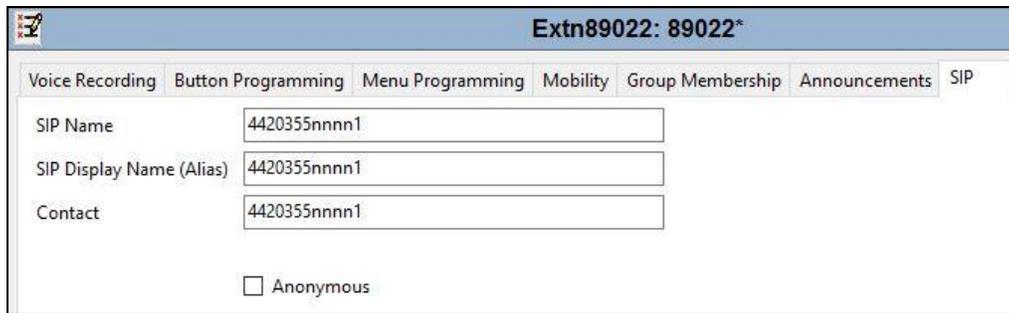
Below the Profile dropdown, there are several checkboxes for additional features:

- Receptionist
- Enable Softphone
- Enable one-X Portal Services
- Enable one-X TeleCommuter
- Enable Remote Worker
- Enable Communicator
- Enable Mobile VoIP Client
- Send Mobility Email
- Web Collaboration

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

Next select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details Pane until it becomes visible. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These fields should be set to the DDI numbers assigned to the enterprise from BT Business in E.164 format.

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



The screenshot shows a configuration window titled "Extn89022: 89022*" with a tabbed interface. The "SIP" tab is selected. The fields are as follows:

Field	Value
SIP Name	4420355nnnn1
SIP Display Name (Alias)	4420355nnnn1
Contact	4420355nnnn1

There is also an unchecked checkbox labeled "Anonymous".

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

5.8. Incoming Call Routing

An incoming call route maps an inbound DDI number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Route** in the Navigation Pane and select **New**, (not shown).

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

- Set the **Bearer Capability** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**.
- 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.

The screenshot shows the configuration interface for an Incoming Call Route. The left pane shows a tree view of IP Offices, with 'Incoming Call Route (6)' expanded and '20 4420355nnnn1' selected. The right pane shows the configuration details for this route, with the 'Standard' tab selected. The configuration fields are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	20
Incoming Number	4420355nnnn1
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

Note: A number of digits of the DDI have been obscured. Number format is E.164.

On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 20 are routed to extension 89022.

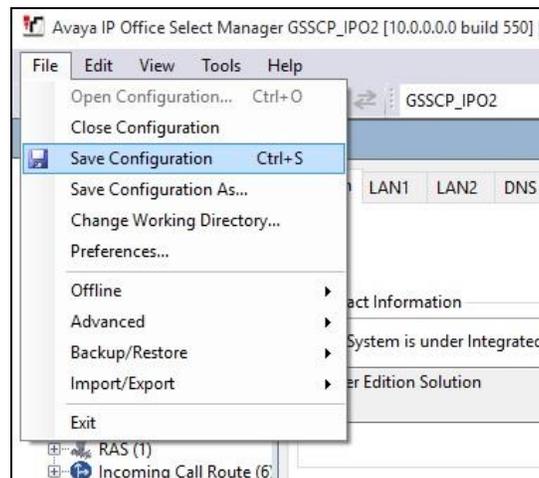
The screenshot shows the 'Destinations' tab configuration interface. The configuration is as follows:

TimeProfile	Destination	Fallback Extension
Default Value	89022 Extn89022	

Note: Calls coming in to destinations not associated with an extension such as Voice Mail and FNE appear on line 19 in this configuration. This is because the incoming number can't be matched to a DDI number specified in the **User** settings shown in **Section 5.7**.

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.

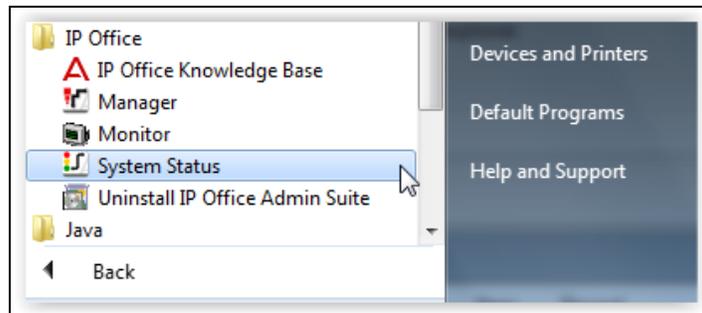


6. Verification Steps

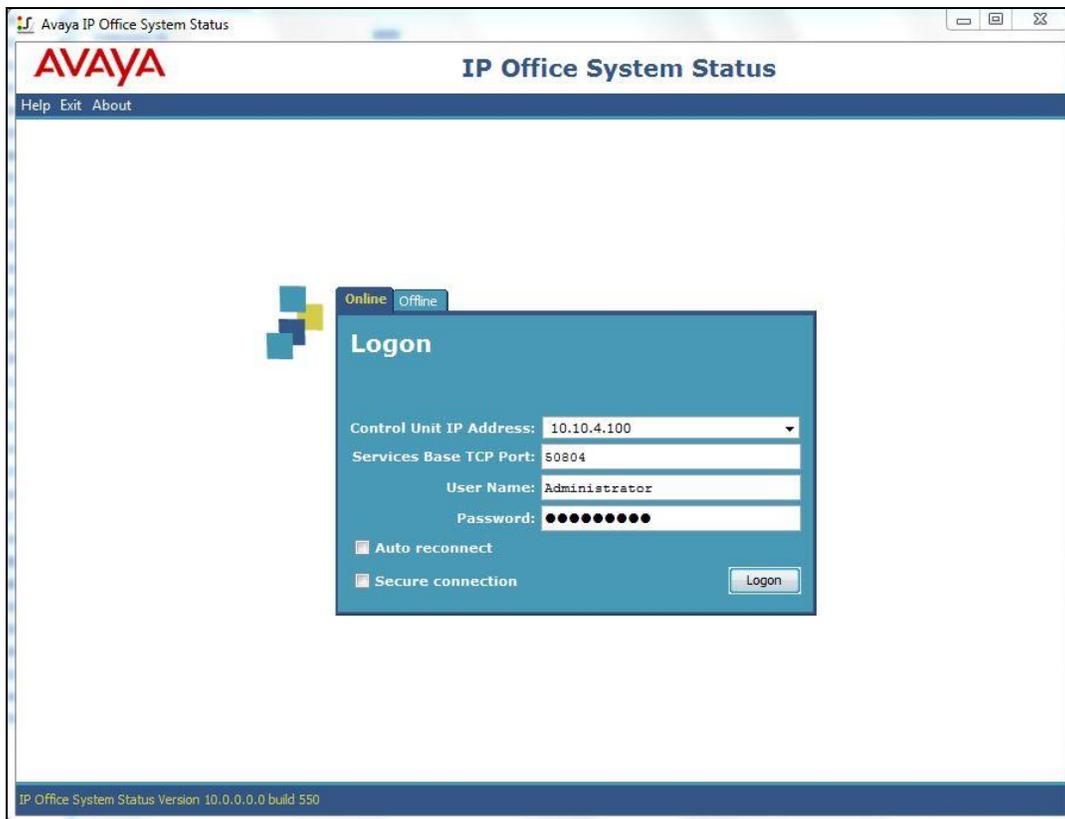
This section includes steps that can be used to verify that the configuration has been done correctly.

6.1. SIP Trunk status

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



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



From the left hand menu expand **Trunks** and choose the SIP trunk (**20** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.

The screenshot shows the Avaya IP Office System Status window. The left-hand menu is expanded to 'Trunks (11)' and 'Line: 20' is selected. The main window displays the 'SIP Trunk Summary' for Line 20. The status is 'In Service' and 'Idle'. A green progress indicator shows 0% utilization. Below the summary is a table of call logs.

Chan...	U...	Call Ref	Current State	Time in State	Remote Media	Co...	Conn...	Caller ID or...	Other Party on Call	Direct...	Round Trip ...	Receive Jitter	Receive Pack...	Trans...	Trans...
1			Idle	1 day...											
2			Idle	1 day...											
3			Idle	1 day...											
4			Idle	1 day...											
5			Idle	1 day...											
6			Idle	1 day...											
7			Idle	1 day...											

7. Conclusion

All tests for BT Business SIP Trunk were completed. Observations for the testing are listed in **Section 2.2**.

8. Additional References

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

- [1] *Avaya IP Office™ Platform Start Here First*, Release 10, August 2016.
- [2] *IP Office™ Platform 10.0 - Deploying Avaya IP Office Basic Edition*, Document number 15-601042, August 2016.
- [3] *IP Office™ Platform 9.1 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document number 15-601011, 30th September 2015.
- [4] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0, August 2016.
- [5] *IP Office™ Platform 9.1 Using System Status*, Document number 15-601758, 11th August 2015.
- [6] *IP Office™ Platform 9.1 Using IP Office System Monitor*, Document number 15-601019, 19th May 2015.
- [7] *Using Avaya Communicator for Windows on IP Office*, Release 10.0, August 2016.
- [8] *IP Office™ Platform 10.0 - Third-Party SIP Extension Installation Notes*, June 2016.
- [9] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

©2016 Avaya Inc. All Rights Reserved.

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

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