



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

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# **Application Notes for British Telecom Wholesale/HIPCOM SIP Trunk Service and Avaya IP Office 7.0 – Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the British Telecom 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. British Telecom 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 Trunk 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 Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

### 2.1. Interoperability Compliance Testing

A simulated enterprise site with Avaya IP Office was connected to the HIPCOM SIP Trunk 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.
- 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:

- The Calling Line Identity (CLI) set at the enterprise and is hidden if the number is withheld at the enterprise in this case no number is presented to the called party.
- 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.

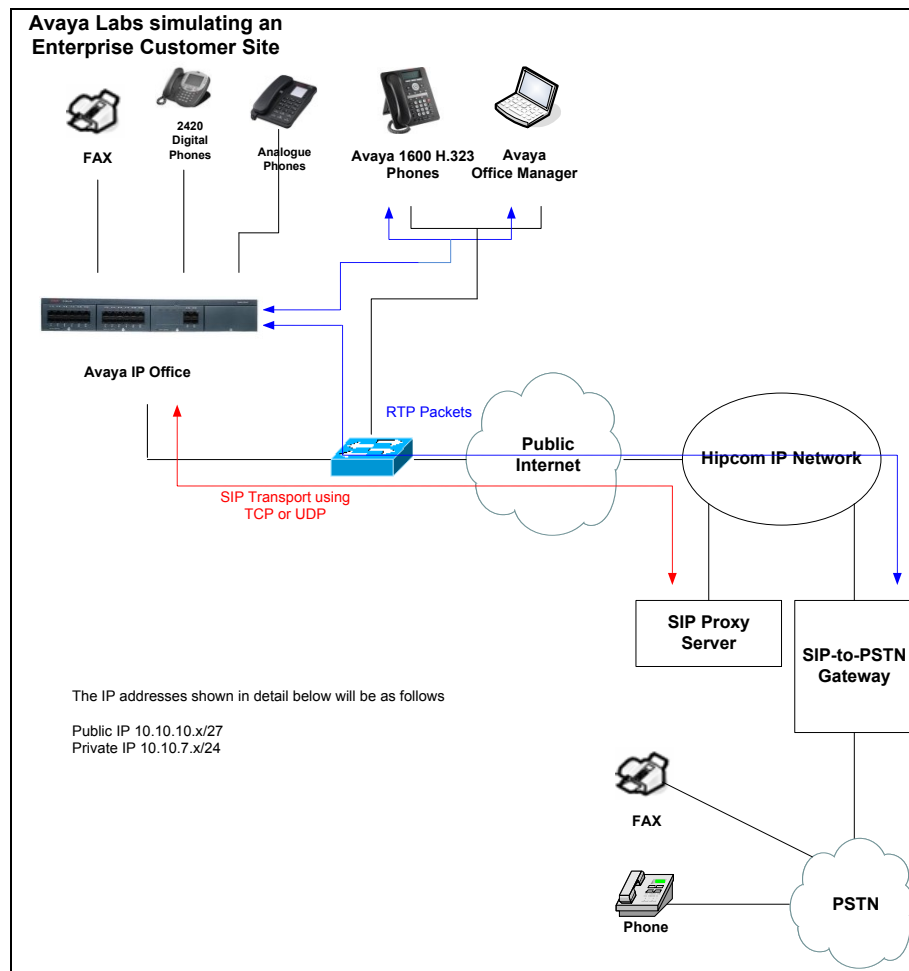
## 2.3. Support

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 Trunk 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 PC with Avaya IP Office Manager installed allowing configuration of 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: BTW/HIPCOM SIP Trunking Solution to Avaya 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 and the remaining N digits are 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	7.0(3)
Avaya 1620 Phone (H.323)	1.220
Avaya 2420 Digital Phone	NA
Avaya 98390 Analogue Phone	NA
HIPCOM SIP Trunk Service	A1B149G1

## 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 appropriate 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.

SIP Trunk Channels	
Licences	
Licence Key	unXMbE6x9dJKGKJ73uEpof7JrpF4smme
Licence Type	SIP Trunk Channels
Licence Status	Valid
Instances	255
Expiry Date	Never

## 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\_IPO7** in the Navigation Pane where GSSCP\_IPO7 is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are set from values shown in **Figure 1**. All other parameters should be set according to customer requirements.

GSSCP_IPO7*	
System   LAN1   LAN2   DNS   Voicemail   Telephony   Directory Services   System Events   SMTP   SMDR   Twinning   VCM   CCR	
LAN Settings   VoIP   Network Topology   SIP Registrar	
IP Address	10 . 10 . 10 . 10
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	10 . 10 . 10 . 1
Firewall Profile	<None>
RIP Mode	None
<input type="checkbox"/> Enable NAT	
Number Of DHCP IP Addresses	200
DHCP Mode: <input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled	
Advanced	

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.

The screenshot shows the Avaya IP Office configuration interface. On the left is a tree view of the system configuration. The main pane is titled 'GSSCP\_IPO7\*' and has tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and CCR. The 'VoIP' tab is selected. Within the VoIP tab, there are sub-tabs for LAN Settings, Network Topology, and SIP Registrar. The 'SIP Registrar' sub-tab is active. In this sub-tab, the following settings are visible:

- ☒ H323 Gatekeeper Enable
- ☒ SIP Trunks Enable (highlighted with a red box)
- ☒ SIP Registrar Enable
- ☒ H323 Auto-create Extn
- ☐ H323 Auto-create User
- ☒ Enable RTCP Monitoring On Port 5005
- RTP Port Number Range** (highlighted with a red box):
  - Port Range (Minimum): 49152
  - Port Range (Maximum): 53246
- DiffServ Settings** (highlighted with a red box):
 

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

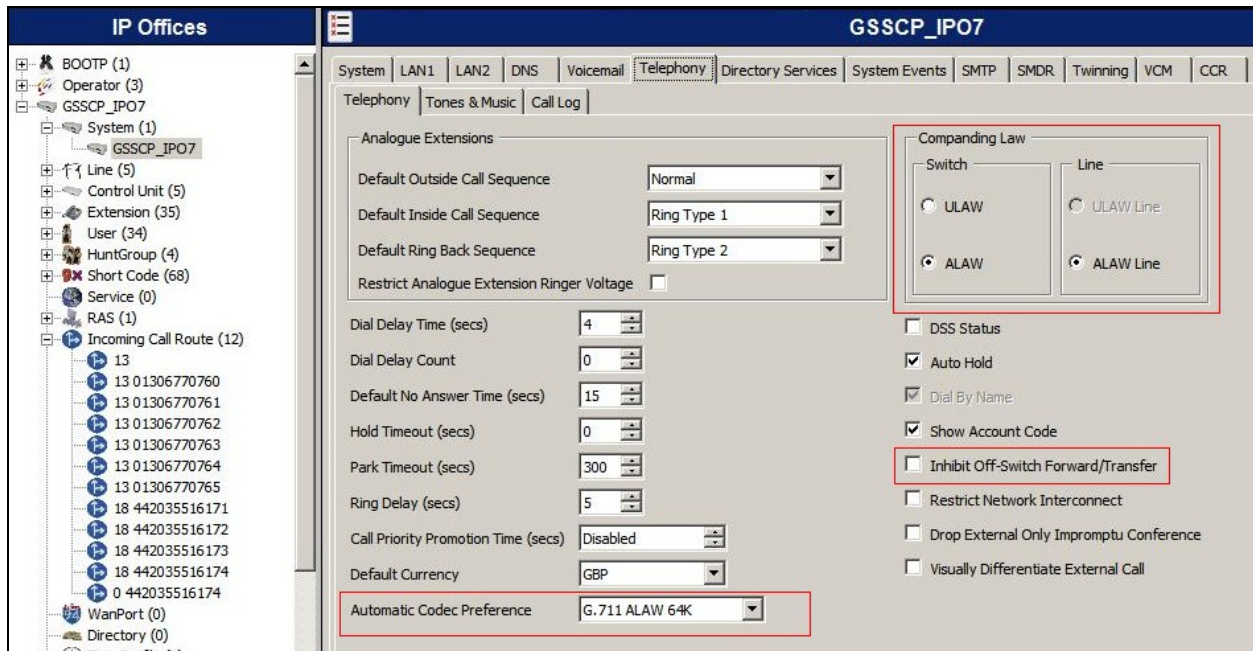
On the **Network Topology** tab set the STUN Server IP Address as this is required even if STUN is not used. Set the **Binding Refresh Time (seconds)** to a figure agreed.

The screenshot shows the Avaya IP Office configuration interface. On the left is a tree view of the system configuration. The main pane is titled 'GSSCP\_IPO7' and has tabs for LAN Settings, VoIP, Network Topology, and SIP Registrar. The 'Network Topology' tab is selected. Within the Network Topology tab, the following settings are visible:

- Network Topology Discovery** (highlighted with a red box):
  - STUN Server IP Address: 10 . 10 . 10 . 52 (highlighted with a red box)
  - STUN Port: 3478
  - Firewall/NAT Type: Open Internet
  - Binding Refresh Time (seconds): 300 (highlighted with a red box)
  - Public IP Address: 10 . 10 . 10 . 120
  - Public Port: 0
  - Run STUN: ☐ Run STUN on startup

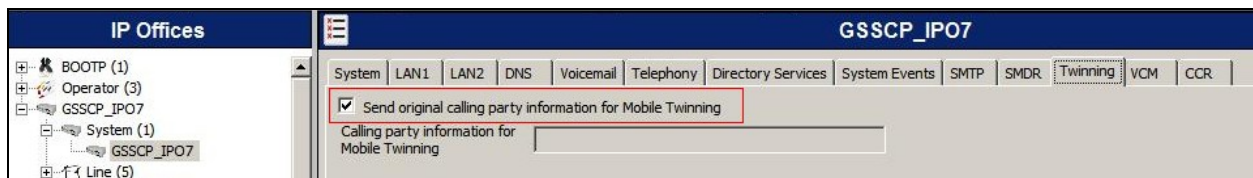
## System Telephony Settings

Navigate to the **Telephony**→**Telephony** Tab on the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. 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.



## 5.3. System Twinning Settings

Navigate to the **Twining** 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**).



## 5.4. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the HIPCOM SIP Trunk 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.
- 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.

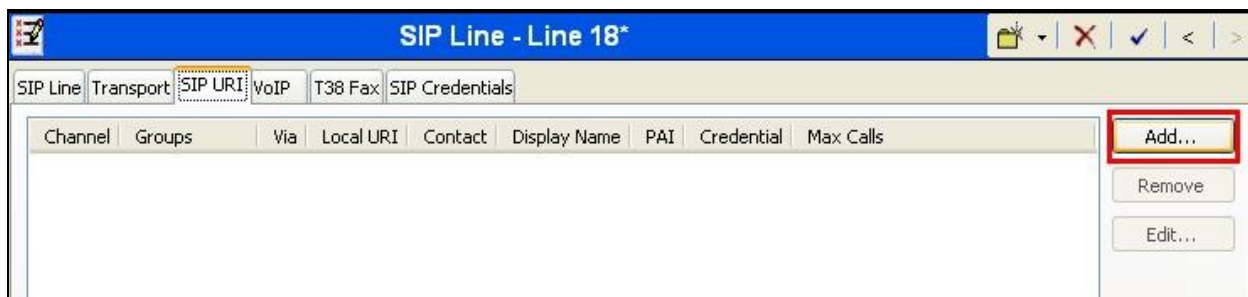
The screenshot shows the 'SIP Line - Line 18' configuration window. The 'SIP Line' tab is active. The 'ITSP Domain Name' field is set to 'sipconnect.hipcom.co.uk'. The 'In Service' checkbox is checked. The 'Send Caller ID' dropdown is set to 'None'. The 'Association Method' is 'By Source IP address'. The 'REFER Support' checkbox is checked. The 'Incoming' and 'Outgoing' dropdowns are set to 'Auto'.

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

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

The screenshot shows the 'SIP Line - Line 18' configuration window, Transport tab. The 'ITSP Proxy Address' field is set to '10.10.10.20'. The 'Layer 4 Protocol' dropdown is set to 'UDP'. The 'Send Port' and 'Listen Port' fields are both set to '5060'. The 'Use Network Topology Info' dropdown is set to 'LAN 2'. The 'Calls Route via Registrar' checkbox is checked.

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**. These settings allows call on this line to send the User Name set in credentials in **Section 5.4.1**
- For **Registration**, select **1: 442031111111** from the pull-down menu since this configuration does not use SIP registration. This value is set on the credentials tab in **Section 5.5.1**
- 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 **Compression Mode** with the **Advanced** button to specify the preferred order of the offered codecs. Select the codecs and their order based on the needs of the customer. Click the box next to the codec with the highest preference first, followed by the second preference. For the compliance test, **G.729(a) 8K CS-ACELP** was selected first followed by **G.711 ALAW 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.

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'VoIP' tab selected. The 'IP Offices' tree on the left shows the hierarchy: BOOTP (1), Operator (3), GSSCP\_IPO7, System (1), GSSCP\_IPO7, Line (5), and its sub-lines (5, 6, 13, 14, 18). The main configuration area includes tabs for SIP Line, Transport, SIP URI, VoIP, T38 Fax, and SIP Credentials. The 'VoIP' tab is active, showing the 'Compression Mode' set to 'Advanced'. A list of codecs is displayed, with 'G.729(a) 8K CS-ACELP' selected. Other settings include 'Fax Transport Support' set to 'T38', 'Call Initiation Timeout (s)' set to '4', and 'DTMF Support' set to 'RFC2833'. Checkboxes for 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), and 'Codec Lockdown' (unchecked) are visible.

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 fields may retain their default values.

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'T38 Fax' tab selected. The 'IP Offices' tree on the left is the same as in the previous screenshot. The main configuration area shows the 'T38 Fax' tab active. The 'T38 Fax Version' is set to '2'. The 'Transport' is set to 'UDPTL'. Under 'Redundancy', 'Low Speed' and 'High Speed' are both set to '0'. The 'TCF Method' is set to 'Trans TCF'. The 'Max Bit Rate (bps)' is set to '14400'. The 'EFlag Start Timer (msecs)' is '2600', 'EFlag Stop Timer (msecs)' is '2300', and 'Tx Network Timeout (secs)' is '150'. The 'Use Default Values' checkbox is unchecked. On the right, a group of checkboxes includes '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 fields for 'Country Code' and 'Vendor Code', both set to '0'.

### 5.4.1. 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 Trunk Service. 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



The screenshot shows a dialog box titled "Edit SIP Credentials". It contains the following fields and controls:

- User name:** Text box containing "442031111111".
- Authentication Name:** Text box containing "442031111111".
- Contact:** Text box containing "442031111111".
- Password:** Text box containing "\*\*\*\*\*".
- Expiry:** Dropdown menu with "60" selected.
- Registration required:** Check box that is checked.
- Buttons:** "OK" and "Cancel" buttons on the right. The "OK" button is highlighted with a red rectangle.

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

## 5.5. 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 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"@sipconnect.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.
- 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).

The screenshot shows a software interface with a left-hand navigation pane titled "IP Offices" and a main configuration area titled "9N;; Dial". The navigation pane lists several entries, each preceded by a red "X" icon: \*30, \*31, \*32\*N#, \*33\*N#, \*34N;, \*35\*N#, \*36, \*37\*N#, \*38\*N#, \*39, \*40, and \*41. The main configuration area contains a "Short Code" tab and several input fields. A red rectangular box highlights the "Feature", "Telephone Number", and "Line Group Id" fields. The "Code" field contains "9N;". The "Feature" dropdown menu is set to "Dial". The "Telephone Number" field contains "N"@sipconnect.hipcom.co.uk". The "Line Group Id" dropdown menu is set to "18". Below these fields are "Locale" and "Force Account Code" (with an unchecked checkbox).

## 5.6. 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 header 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 **44203111111** 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.

The screenshot displays the Avaya SIP configuration interface. On the left, the 'IP Offices' navigation pane shows a tree structure with 'User (34)' selected. The main pane on the right is titled 'Extn89101: 89101\*' and contains several tabs: 'Telephony', 'Forwarding', 'Dial In', 'Voice Recording', 'Button Programming', 'Menu Programming', and 'Mobility'. The 'SIP' tab is active, showing three input fields: 'SIP Name', 'SIP Display Name (Alias)', and 'Contact'. Each of these fields contains the value '44203111111' and is enclosed in a red rectangular box. Below these fields is an 'Anonymous' checkbox, which is currently unchecked.

## 5.7. 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**. 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.

The screenshot shows the 'Standard' tab of the 'Incoming Call Route' configuration. The 'Navigation Pane' on the left lists various system components, including 'Incoming Call Route (12)' with a list of entries. The 'Details Pane' on the right shows the configuration for a specific route. The 'Standard' tab is active, and the following fields are visible:

Field	Value
Bearer Capacity	Any Voice
Line Group Id	18
Incoming Number	442031234567
Incoming Sub Address	
Incoming CLI	
Locale	United Kingdom (UK English)
Priority	2 - Medium
Tag	
Hold Music Source	System Source

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.

The screenshot shows the 'Destinations' tab of the 'Incoming Call Route' configuration. The 'Navigation Pane' on the left lists various system components, including 'GSSCP\_IPO7'. The 'Details Pane' on the right shows the configuration for a specific route. The 'Destinations' tab is active, and the following fields are visible:

Field	Value
TimeProfile	
Default Value	89100 Extn89100

## 5.8. SIP Options

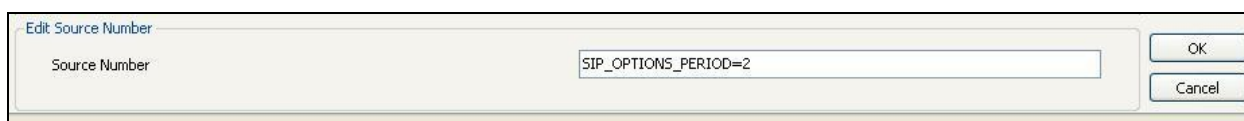
Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab in **Section 5.2** and the **SIP\_OPTIONS\_PERIOD** parameter (in minutes) that can be set on the **Source Number** tab of the **noUser** user. The OPTIONS period is determined in the following manner:

- If no **SIP\_OPTIONS\_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 42 seconds is used.
- To establish a period less than 42 seconds, do not define a **SIP\_OPTIONS\_PERIOD** parameter and set the **Binding Refresh Time** to a value less than 42 secs. The OPTIONS message period will be equal to the **Binding Refresh Time**.
- To establish a period greater than 42 seconds, a **SIP\_OPTIONS\_PERIOD** parameter must be defined. The **Binding Refresh Time** must be set to a value greater than 42 secs. The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP\_OPTIONS\_PERIOD**.

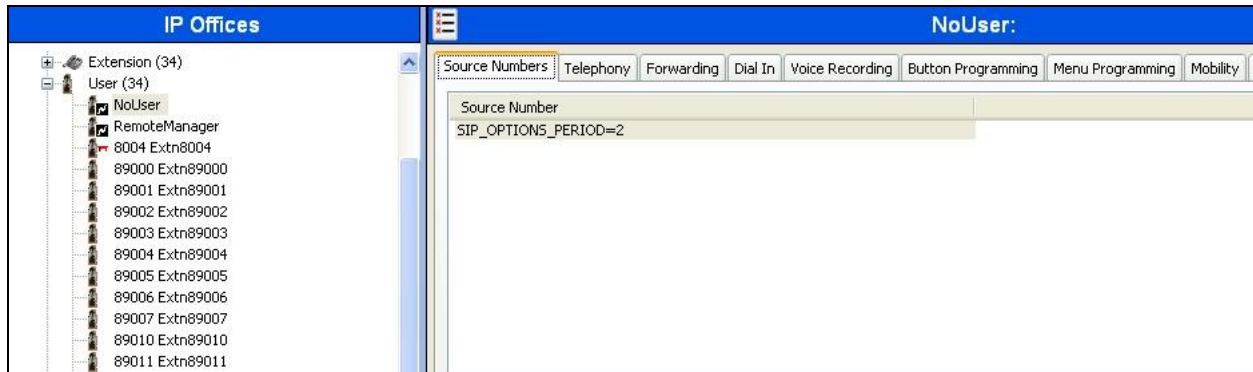
To configure the **SIP\_OPTIONS\_PERIOD** parameter, navigate to **User→NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add...** button.



At the bottom of the subsequent Details Pane, the **Source Number** field will appear. Enter **SIP\_OPTIONS\_PERIOD=X**, where **X** is the desired value in minutes. Click **OK**.



The **SIP\_OPTIONS\_PERIOD** parameter will appear in the list of Source Numbers as shown below. For the compliance test, an OPTIONS period of 2 minutes was desired. The **Binding Refresh Time** was set to **300** seconds (5 minutes) in **Section 5.2**. The **SIP\_OPTIONS\_PERIOD** was set to **2** minutes. Avaya IP Office chooses the OPTIONS period as the smaller of these two values (2 minutes). Click the **OK** button (not shown).



## 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. BTW/HIPCOM SIP Trunk Service Configuration

BTW/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 IP Office at the enterprise.

BTW/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 appropriate 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.

**SIP Trunk Summary**

Peer Domain Name: sipconnect.hipcom.co.uk  
Resolved Address: 85.119.63.4  
Line Number: 18  
Number of Administered Channels: 10  
Number of Channels in Use: 0  
Administered Compression: Auto  
Silence Suppression: Off  
SIP Trunk Channel Licences: Unlimited  
SIP Trunk Channel Licences in Use: 0  
SIP Device Features:

0%

Channel Number	URI Grou	Call Ref	Current State	Time in State	Remote RTP Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call
1			Idle	18:22:43					
2			Idle	18:22:43					
3			Idle	18:22:43					
4			Idle	18:22:43					
5			Idle	18:22:43					
6			Idle	18:22:43					
7			Idle	18:22:43					
8			Idle	18:22:43					
9			Idle	18:22:43					
10			Idle	18:22:43					

## 8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and the BTW/HIPCOM SIP Trunk 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 BTW/HIPCOM SIP Trunk Service. This solution provides IP Office users the ability to access the Public Switched Telephone Network (PSTN) via a SIP trunk using the BTW/HIPCOM SIP Trunk service.

## 9. Additional References

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

[1] IP Office 7 Documentation CD, 4<sup>th</sup> May 2011.

[2] IP Office Installation, Document number15-601042, 22<sup>nd</sup> May 2011.

[3] IP Office Manager, Document number15-601011, 22<sup>nd</sup> May 2011.

[4] System Status Application, Document number15-601758, 12<sup>th</sup> February 2010.

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

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