



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 9.0 to support BT Global Services NOAS SIP Trunk – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between BT Global Services NOAS SIP Trunk service and Avaya IP Office.

The BT NOAS SIP Trunk Service provides PSTN access via a SIP trunk connected to the BT NOAS Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or Digital trunks. 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 BT SIP Trunk service and Avaya IP Office. Customers using this Avaya SIP-enabled enterprise solution with BT NOAS SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

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 NOAS SIP Trunk Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

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

2.1. Interoperability Compliance Testing

To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming calls to the enterprise site from PSTN phones using the SIP Trunk provided by BT, calls made to SIP and H.323 telephones at the enterprise
- Outgoing calls from the enterprise site completed via BT NOAS SIP trunk to PSTN destinations, calls made from SIP and H.323 telephones
- Calls using the G.711A, G.711MU and G.729A codecs
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using T.38
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- Inbound and outbound PSTN calls to/from an IP Office Softphone clients
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance
- Caller ID presentation and Caller ID restriction
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- Transmission and response of SIP OPTIONS messages sent by BT requiring Avaya response and sent by Avaya requiring BT response

2.2. Test Results

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

- During compliance testing it was observed that when BT NOAS initiates a call-hold, BT NOAS will send a reINVITE with the attribute “SendOnly” in the SDP. IP Office correctly responds with 200OK with the attribute “RecvOnly” in the SDP. However, once call-hold is initiated, RTCP packets are just sent in one direction from BT NOAS to Avaya IP Office. IP Office does not send RTCP packets when BT NOAS initiated call-hold using the Media attribute “SendOnly” in the SDP. BT NOAS expect to receive RTCP from IP Office and have 25 second RTCP timers configured on their SIP Media Gateway. As BT NOAS don’t receive any RTCP packets from IP Office, after 25 seconds, BT NOAS issue a BYE and the call is torn down. An Avaya IP Office MR was opened on this issue and a fix has been provided in a custom build **IP Office 9.0.110.1**. With this build, it requires the configuration of a **NoUserSourceNumber** on IP Office. The details and configuration of the **NoUserSourceNumber** are outlined in **Section 5.10**. During testing, IP Office 9.0FP was used for testing with custom build **IP Office 9.0.110.1** applied as a release-specific patch. **IP Office 9.0.110.1** will be incorporated in feature pack **IP Office 9.0.4** later this year. Versions of IP Office prior to the release of **IP Office 9.0.4** will require the release specific patch **IP Office 9.0.110.1** to address this issue.
- Inbound call hold and resume from PSTN was not tested as PSTN was unable to initiate the call-hold due to their environment set-up.
- PSTN called party hang-up during an active call did not cause the call to drop. Avaya IP Office caller must hang-up first, or wait for the PSTN T2ISUP timer to expire.
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked with the Emergency Services Operator.

2.3. Support

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

For technical support on BT products please use the following web link. <http://btbusiness.custhelp.com/app/contact>

3. Reference Configuration

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

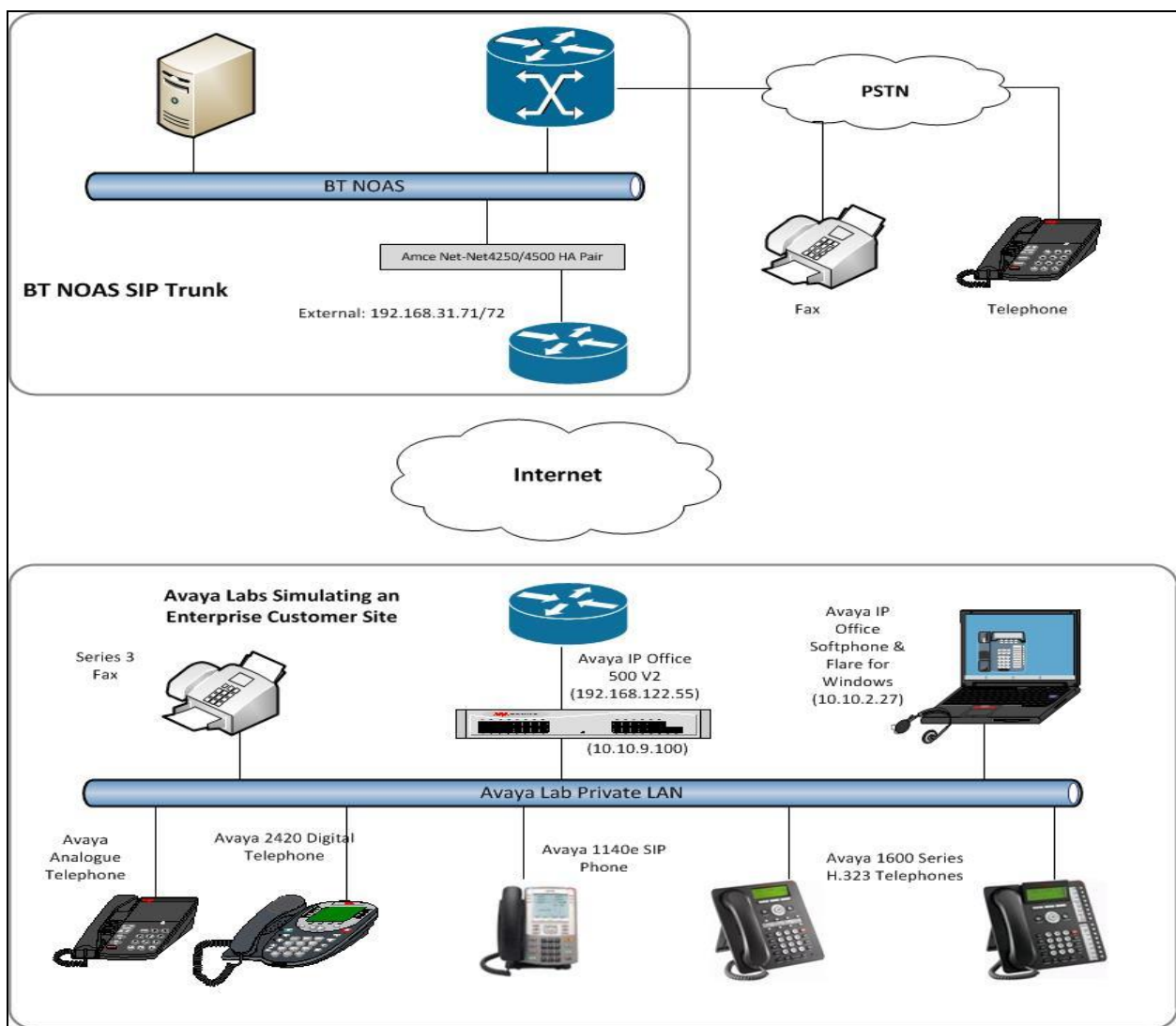


Figure 1: Test Setup BT NOAS SIP Trunk Service to Simulated Enterprise

Avaya IP Office was configured to connect to a static IP address at the Service Provider. For the purposes of the compliance test, users dialed a short code of 9N digits to send digits across the SIP trunk to the BT network. The short code of 9 is stripped off by Avaya IP Office and the remaining N digits sent.

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. BT sends SIP signalling 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 BT 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 | Release/Version |
|--|--|
| Avaya | |
| Avaya IP Office 500 V2 | Avaya IP Office R9.0 with release-specific patch. IP Office 9.0.110.1 |
| Avaya 1603 Phone (H.323) | 1.3100 |
| Avaya 1608 Phone (H.323) | 1.3100 |
| Avaya SoftPhone (SIP) | 3.056516 |
| Avaya Flare Experience for Windows (SIP) | 1.1.3.14 |
| Avaya 1140e (SIP) | FW: 04.01.13.00.bin |
| Avaya 2420 Digital Phone | R6.0 |
| Avaya 98390 Analogue Phone | N/A |
| BT NOAS | |
| Acme Packet Net-Net 4250 Session border Controller | SC6.1.0 MR-11 patch 1 (build 1036) |
| | |

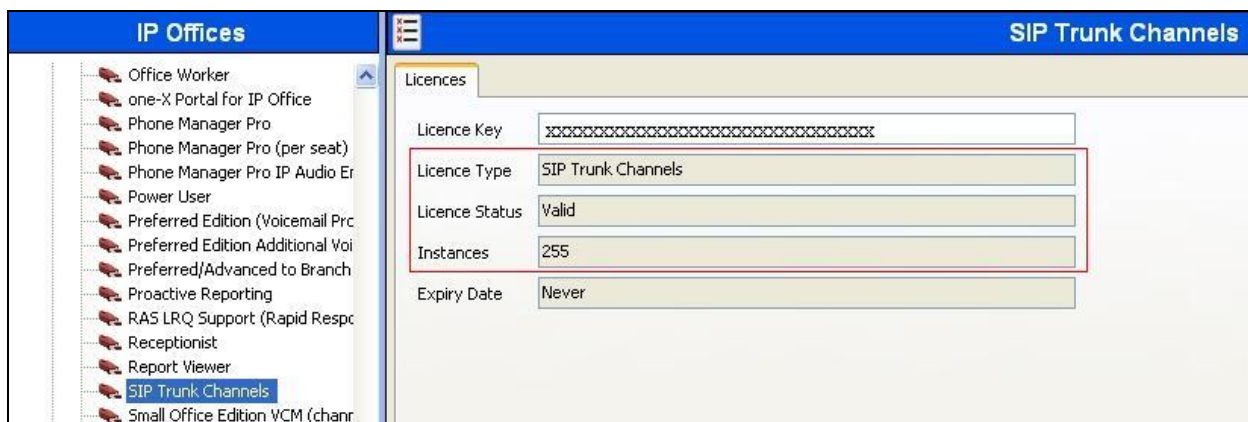
5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to BT NOAS SIP Trunk service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced

throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider (such as twinning) is assumed to already be in place.

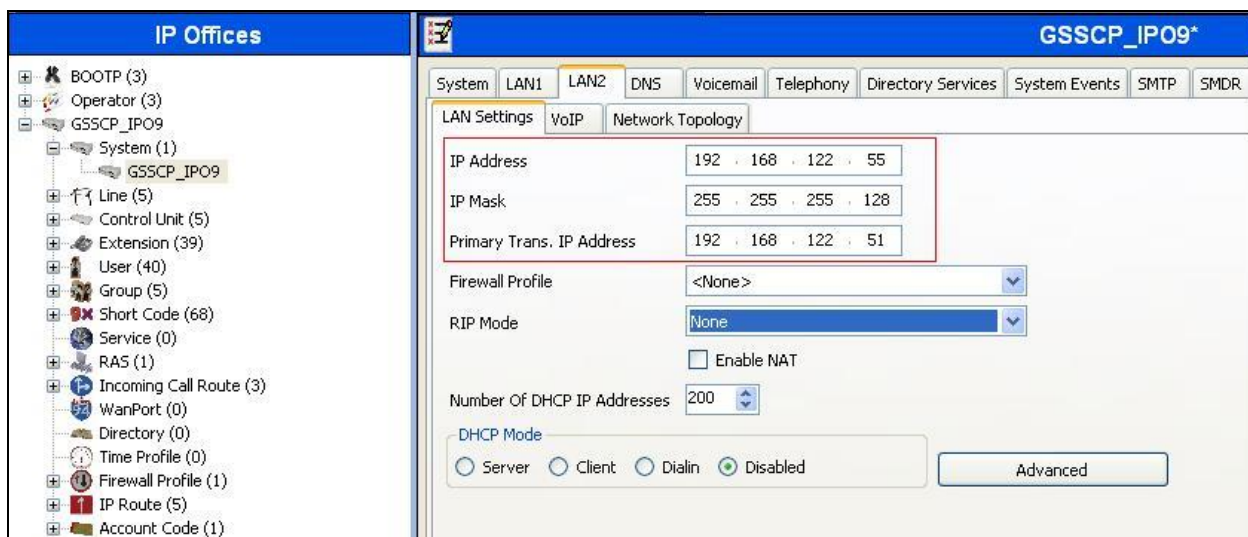
5.1. Verify System Capacity

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



5.2. LAN2 Settings

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 → GSSCP_IPO2** in the Navigation Pane where GSSCP_IPO2 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 the public interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



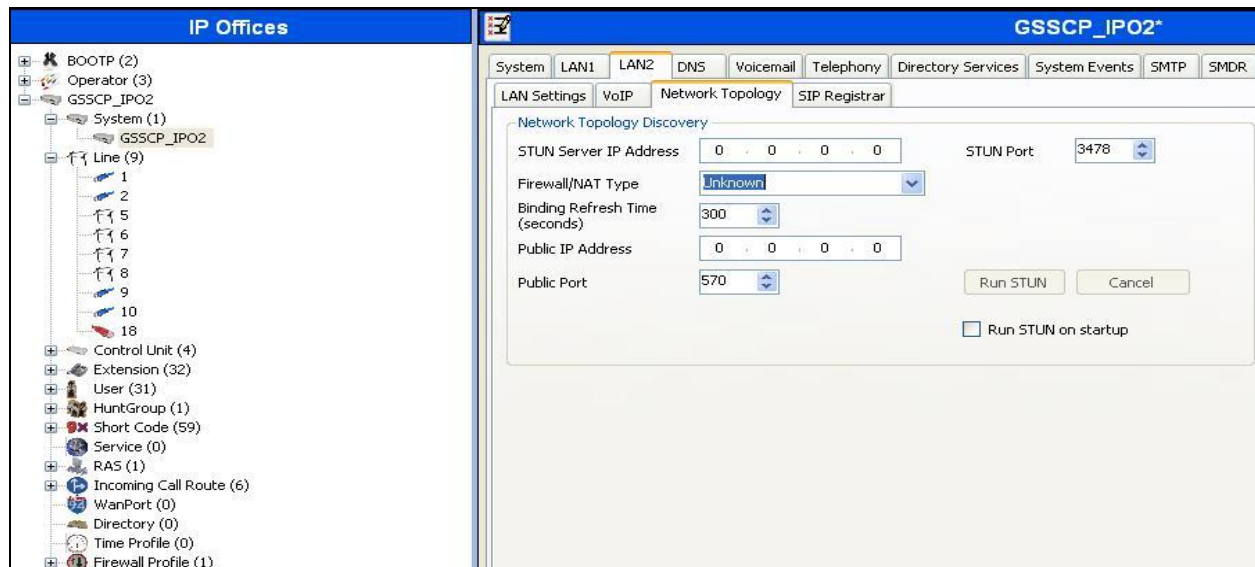
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **RTP Port Number Range** can be customized to a specific range of receive ports for 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 signalling and media. The **DSCP** field is the value used for media and the **SIG DSCP** is the value used for signalling. The specific values used for the compliance test are shown in the example below. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

The screenshot shows the 'GSSCP_IP09*' configuration window with the 'VoIP' tab selected. The 'LAN Settings' sub-tab is active. The 'SIP Trunks Enable' and 'SIP Registrar Enable' checkboxes are checked and highlighted with a red box. The 'Domain Name' is set to 'avaya.com'. The 'Layer 4 Protocol' section shows 'UDP' and 'TCP' checked, with 'UDP Port' and 'TCP Port' both set to '5060'. The 'Challenge Expiry Time (secs)' is set to '10'. The 'RTP' section shows the 'Port Number Range' with 'Minimum' set to '49152' and 'Maximum' set to '64566', also highlighted with a red box.

| Setting | Value |
|------------------------------|-------------------------------------|
| H323 Gatekeeper Enable | <input checked="" type="checkbox"/> |
| Auto-create Extn | <input checked="" type="checkbox"/> |
| Auto-create User | <input type="checkbox"/> |
| H323 Remote Extn Enable | <input type="checkbox"/> |
| SIP Trunks Enable | <input checked="" type="checkbox"/> |
| SIP Registrar Enable | <input checked="" type="checkbox"/> |
| Auto-create Extn/User | <input type="checkbox"/> |
| SIP Remote Extn Enable | <input type="checkbox"/> |
| Domain Name | avaya.com |
| UDP | <input checked="" type="checkbox"/> |
| UDP Port | 5060 |
| Remote UDP Port | 5060 |
| TCP | <input checked="" type="checkbox"/> |
| TCP Port | 5060 |
| Remote TCP Port | 5060 |
| TLS | <input type="checkbox"/> |
| TLS Port | 5061 |
| Remote TLS Port | 5061 |
| Challenge Expiry Time (secs) | 10 |
| RTP Port Number Range | Minimum: 49152, Maximum: 64566 |

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



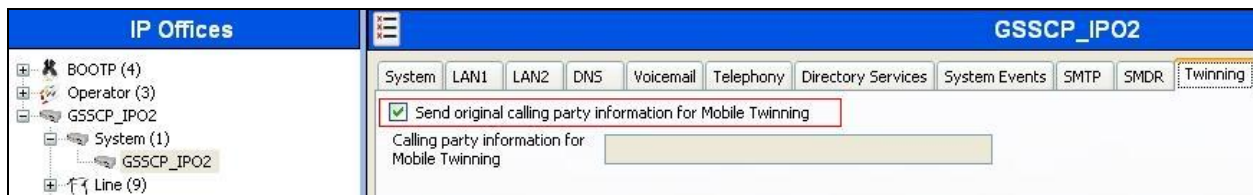
5.3. System Telephony Settings

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



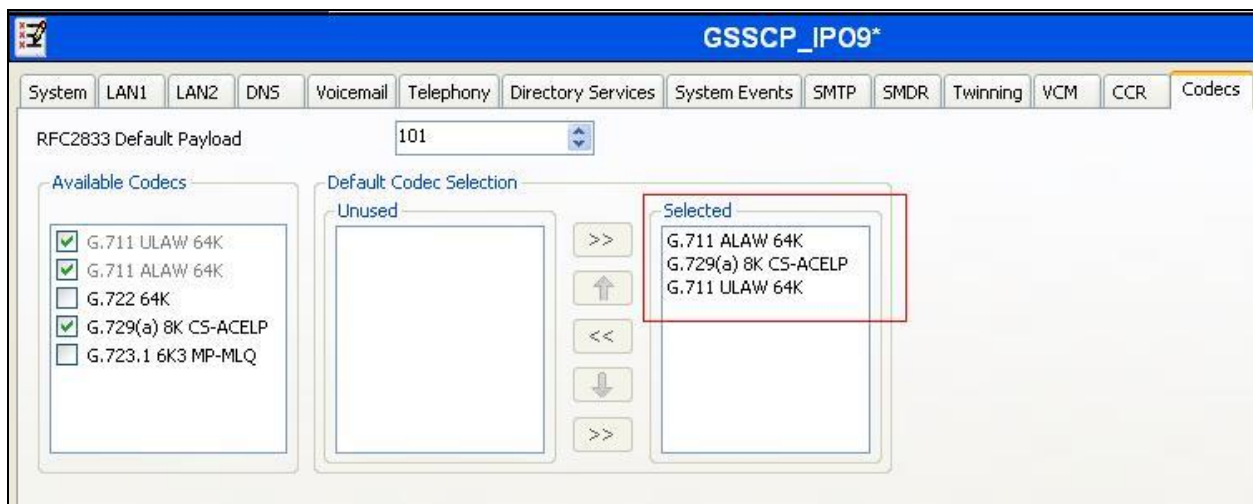
5.4. System Twinning Settings

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. On calls from the PSTN to a twinned phone, Avaya IP Office will send the calling party number of the host phone associated with the twinned destination (instead of the number of originating caller). 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 5.6). On completion, click the **OK** button (not shown).



5.5. Codec Settings

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



5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the BT NOAS SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.6.2**.

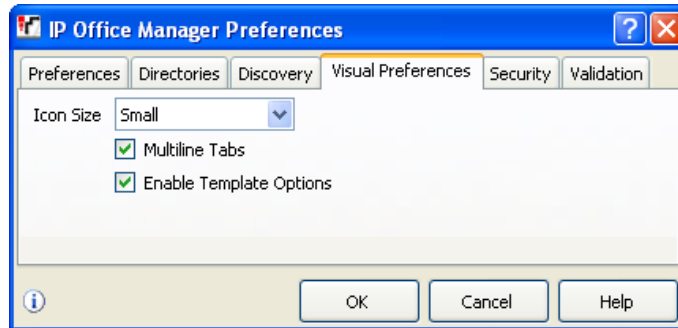
Also, the following SIP Line settings are not supported on Basic Edition:

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

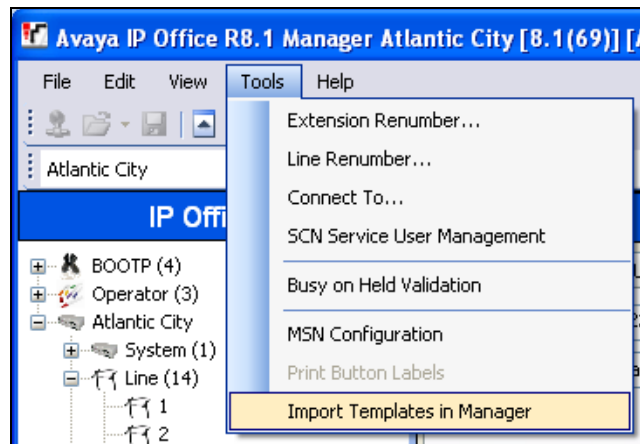
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line** (not shown). Then, follow the steps outlined in **Section 5.6.2**.

5.6.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **IE_BTNOAS_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.
2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences** (not shown). In the IP Office Manager Preferences window that appears, select the Visual Preferences tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

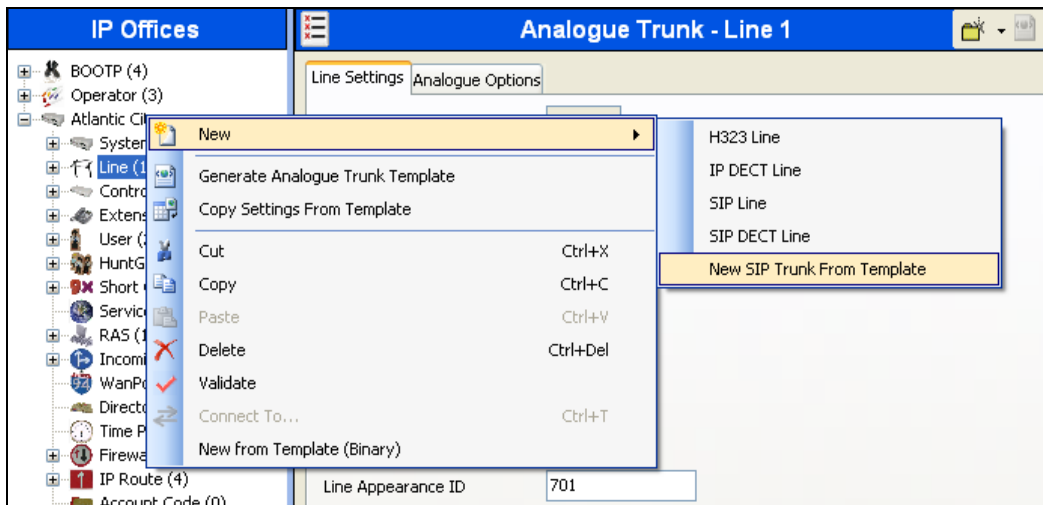


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default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



In the pop-up window (not shown) that appears, select the directory where the template file was copied in **Step 1**. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if the template file is copied directly to the IP Office template directory.

4. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New SIP Trunk From Template**.



5. In the subsequent Template Type Selection pop-up window, select **Ireland** from the **Country** pull-down menu and select **BT NOAS** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**IE_BTNOAS_SIPTrunk.xml**) created in **Step 1**. Click **Create new SIP Trunk** to finish creating the trunk.



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

5.6.2. SIP Line – SIP Line Tab

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

- **ITSP Domain Name** field should remain blank as BT have not provided a Domain Name
- Set **Send Caller ID** to *None*
- Ensure the **In Service** box is checked
- Set **REFER Supported** to Auto
- Set **Method for Session Refresh** to **Reinvite**
- Default values may be used for all other parameters

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

SIP Line - Line 18

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

Line Number: 18

ITSP Domain Name:

Prefix:

National Prefix: 0

Country Code: 44

International Prefix: 00

Send Caller ID: None

Association Method: By Source IP address

In Service: ☒

URI Type: SIP

Check OOS: ☐

Call Routing Method: Request URI

Originator number for forwarded and twinning calls:

Name Priority: System Default

Caller ID from From header: ☐

Send From In Clear: ☒

User-Agent and Server Headers:

Service Busy Response: 486 - Busy Here

Action on CAC Location Limit: Allow Voicemail

☒ REFER Support

Incoming: Auto

Outgoing: Auto

Method for Session Refresh: Reinvite

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

- Set **ITSP Proxy Address** to the IP address of the BT NOAS SIP proxy
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** to **5060** and **Listen Port** to **5060**
- Set **Network Topology Info** to **None**

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

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field contains '192.168.31.71,192.168.31.72'. The 'Network Configuration' section includes 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'None', and 'Listen Port' set to '5060'. Below this, 'Explicit DNS Server(s)' are set to '0.0.0.0' and '0.0.0.0'. The 'Calls Route via Registrar' checkbox is checked. The 'Separate Registrar' field is empty.

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

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

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 all calls on this line whose SIP URI matches the number set in the **SIP** tab of any User as shown in **Section 5.8**.
- Set **Contact**, **Display Name** and **PAI** to *.
- For **Registration**, select **0: <None>** from the pull-down menu since this configuration does not use SIP registration.
- 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.

The screenshot shows the 'Edit Channel' dialog box with the following configuration:

| Field | Value |
|-----------------------|-------------------|
| Via | <None> |
| Local URI | Use Internal Data |
| Contact | * |
| Display Name | * |
| PAI | Use Internal Data |
| Registration | 0: <None> |
| Incoming Group | 18 |
| Outgoing Group | 18 |
| Max Calls per Channel | 10 |

Buttons: OK, Cancel

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

- Select **Custom** from the drop-down Codec Selection menu.
- Select **G.711 ALAW 64K, G.711 ULAW 64K and G.729(a) 8K CS-ACELP** codecs.
- 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.
- Select the **Fax Transport Support** box to **T.38**.
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk.
- Check **PRACK/100rel Supported** to advertise the support for provisional responses and Early Media to the BT NOAS network.
- 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 'Codec Selection' section shows 'Custom' selected in the dropdown. Below this, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.722 64K'. The 'Selected' list contains 'G.711 ALAW 64K', 'G.729(a) 8K CS-ACELP', and 'G.711 ULAW 64K'. To the right of these lists are buttons for moving items between them. Below the codec lists, there are several checkboxes and dropdowns: 'Fax Transport Support' is set to 'T38', 'Location' is set to 'Cloud', 'Call Initiation Timeout (s)' is set to '4', and 'DTMF Support' is set to 'RFC2833'. On the right side of the window, there are several checkboxes: 'VoIP Silence Suppression' (unchecked), 'Allow Direct Media Path' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'PRACK/100rel Supported' (checked), 'Force direct media with phones' (unchecked), and 'G.711 Fax ECAN' (unchecked).

Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the Use Default Values box (not shown) and select **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. On completion, click the **OK** button (not shown).

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'T38 Fax' tab selected. The window has a blue header bar with the title 'SIP Line - Line 18*'. Below the header is a tab bar with 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'T38 Fax' tab is active. The configuration area is divided into two main sections. The left section contains: 'T38 Fax Version' (dropdown menu set to '2'), 'Transport' (dropdown menu set to 'UDP/TLS'), 'Redundancy' (grouped box containing 'Low Speed' and 'High Speed' spinners, both set to '0'), 'TCF Method' (dropdown menu set to 'Trans TCF'), 'Max Bit Rate (bps)' (dropdown menu set to '14400'), 'EFlag Start Timer (msecs)' (spinner set to '2600'), 'EFlag Stop Timer (msecs)' (spinner set to '2300'), and 'Tx Network Timeout (secs)' (spinner set to '150'). The right section contains a list of checkboxes: '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 checkboxes are two spinners: 'Country Code' (set to '0') and 'Vendor Code' (set to '0').

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

5.7. Short Codes

Define a short code to route outbound traffic to the SIP line. 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. The example shows **9N;** which will be invoked when the user dials 9 followed by the dialed number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N** which will allow an IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter N. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6.2**.

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

The screenshot shows the 'IP Offices' application window. On the left is a 'Navigation Pane' with a list of short codes, including *42, *43, *44, *45*N#, *46, *47, *48, *49, *50, *51, *52, *53*N#, *57*N#, *70*N#, *71*N#, *9000*, *91N;, *92N;, *DSSN, *SDN, *SKN, 118N;, 1802, and 9N;. The '9N;' entry is selected. On the right is the 'Details Pane' for the 'Short Code' tab, titled '9N;; Dial'. It contains the following fields: 'Code' (9N;), 'Feature' (Dial), 'Telephone Number' (N), 'Line Group ID' (18), 'Locale' (empty), and 'Force Account Code' (unchecked checkbox).

The screenshot below displays an example of a short code ***67N;** that can be used to withhold the sending of the calling ID number. **W** is a Telephone Number Field Character used to withhold outgoing CLI. The short code is similar to the shortcode **9N;** code used to route outbound traffic to the SIP line except that the Telephone Number field begins with **W** which will withhold the sending of the calling ID number. **Note:** This operation is service provider dependent.

The screenshot shows a configuration window titled '*67N;; Dial'. It contains a 'Short Code' tab and several input fields: 'Code' with the value '*67N;', 'Feature' with a dropdown menu showing 'Dial', 'Telephone Number' with the value 'WN', 'Line Group ID' with a dropdown menu showing '18', 'Locale' with a dropdown menu, and 'Force Account Code' with an unchecked checkbox.

| *67N;; Dial | |
|--------------------|--------------------------|
| Short Code | |
| Code | *67N; |
| Feature | Dial |
| Telephone Number | WN |
| Line Group ID | 18 |
| Locale | |
| Force Account Code | <input type="checkbox"/> |

5.8. User and Extensions

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users.

A new SIP extension may be added by right-clicking on **Extension** in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with 89107, the extension assigned to the Avaya 1140E. Ensure the **Force Authorization** box is checked.

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

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

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank or populated with a static IP address. Check the **Reserve Avaya IP endpoint license** box. The new **Codec Selection** parameter may retain the default setting "System Default" to follow the system configuration shown in **Section 5.5**. Alternatively, "Custom" may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.

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

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

On the right side, the following checkboxes are present:

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

The 'Codec Selection' section shows two lists: 'Unused' (G.711 ULAW 64K, G.722 64K) and 'Selected' (G.711 ALAW 64K, G.729(a) 8K CS-ACELP), with arrow buttons for moving items between them.

To add a User, right click on **User** in the Navigation pane, and select **New**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane. Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, select the **User** tab if any changes are required. The example below shows the changes required to use Avaya 1140E which was used in test.

The screenshot shows the 'User' configuration page for 'Extn89107: 89107*'. The 'User' tab is selected, showing various configuration fields and options.

| Field | Value |
|---------------------|------------|
| Name | Extn89107 |
| Password | ***** |
| Confirm Password | ***** |
| Account Status | Enabled |
| Full Name | Ext 89107 |
| Extension | 89107 |
| Email Address | |
| Locale | |
| Priority | 5 |
| System Phone Rights | None |
| Profile | Power User |

Additional options under the Profile section:

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

Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.

Ext89107: 89107*

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

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

Login Code *****

Login Idle Period (secs)

Monitor Group <None>

Coverage Group <None>

Status on No-Answer Logged On (No change)

Reset Longest Idle Time

☒ All Calls

☐ External Incoming

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

☐ Force Login

☐ Force Account Code

☐ Incoming Call Bar

☐ Outgoing Call Bar

☐ Inhibit Off-Switch Forward/Transfer

☐ Can Intrude

☒ Cannot be Intruded

☐ Can Trace Calls

☐ CCR Agent

☐ Automatic After Call Work

☐ Deny Auto Intercom Calls

Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.

Ext89107: 89107*

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

Call Settings Supervisor Settings Multi-line Options Call Log TUI

Outside Call Sequence Default Ring

Inside Call Sequence Default Ring

Ringback Sequence Default Ring

No Answer Time (secs) System Default (15)

Wrap-up Time (secs) 2

Transfer Return Time (secs) Off

Call Cost Mark-Up 100

☒ **Call Waiting On**

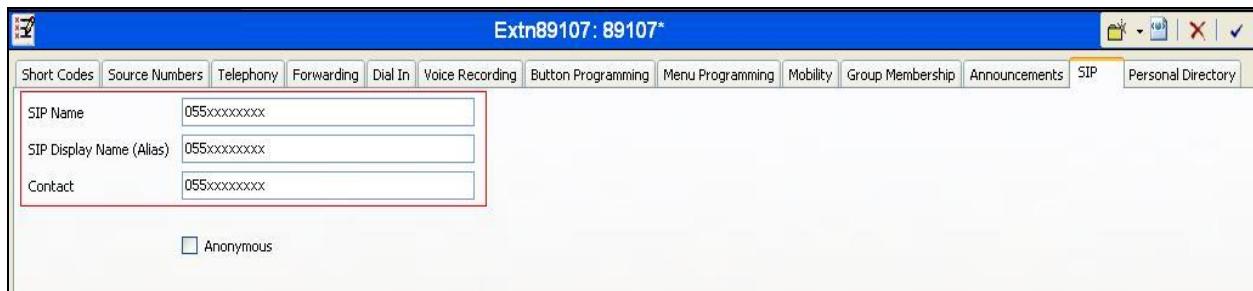
☐ Answer Call Waiting On Hold

☐ Busy On Held

☐ Offhook Station

Next select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right hand side of the Details Pane until it becomes visible. The values entered for the **SIP Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These allow matching of the SIP URI for incoming calls without having to enter this number as an explicit SIP URI for the SIP line (**Section 5.6**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from BT.

In the example below, one of the DDI numbers in the test range is used, though only country code, city code and least significant digit are shown. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. On completion, click the **OK** button (not shown).



The screenshot shows a software window titled "Extn89107: 89107*". The window has a tabbed interface with the following tabs: Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, Mobility, Group Membership, Announcements, SIP (selected), and Personal Directory. The SIP tab is active, displaying three text input fields: "SIP Name" with the value "055xxxxxxx", "SIP Display Name (Alias)" with the value "055xxxxxxx", and "Contact" with the value "055xxxxxxx". Below these fields is a checkbox labeled "Anonymous" which is currently unchecked. The window also features standard OS controls (minimize, maximize, close) and a confirmation button (checkmark) in the top right corner.

5.9. Incoming Call Routing

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

- Set the **Bearer Capacity** to **Any Voice**
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6**
- 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 window. The title bar at the top displays '18 055xxxxxxxxx'. The 'Standard' tab is selected, with 'Voice Recording' and 'Destinations' tabs also visible. The configuration fields are as follows:

| Field | Value |
|----------------------|---------------|
| Bearer Capacity | Any Voice |
| Line Group ID | 18 |
| Incoming Number | 055xxxxxxxxx |
| Incoming Sub Address | |
| Incoming CLI | |
| Locale | |
| Priority | 1 - Low |
| Tag | |
| Hold Music Source | System Source |
| Ring Tone Override | None |

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

The screenshot shows the 'Destinations' tab of the Incoming Call Route configuration window. The title bar at the top displays '18 055xxxxxxx'. The 'Destinations' tab is selected, with 'Standard' and 'Voice Recording' tabs also visible. The configuration table is as follows:

| TimeProfile | Destination |
|---------------|-----------------|
| Default Value | 89107 Extn89107 |

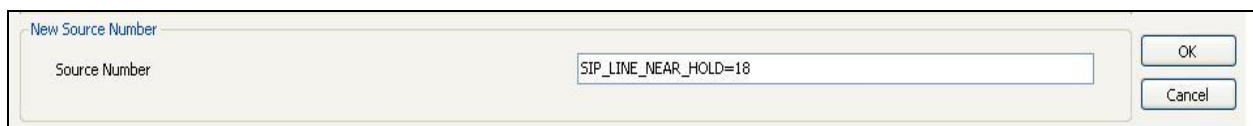
5.10. NoUserSourceNumber

NoUserSourceNumber is a string that is used to enter values in Avaya IP Office that have special uses.


During compliance testing it was observed that when BT NOAS initiates a call-hold, BT NOAS will send a reINVITE with the attribute “SendOnly” in the SDP. IP Office correctly responds with 200OK with the attribute “RecvOnly” in the SDP. However, once call-hold is initiated, RTCP packets are just sent in one direction from BT NOAS to Avaya IP Office. IP Office does not send RTCP packets when BT NOAS initiated call-hold using the Media attribute “SendOnly” in the SDP. BT NOAS expect to receive RTCP from IP Office and have 25 second RTCP timers configured on their SIP Media Gateway. As BT NOAS don’t receive any RTCP packets from IP Office, after 25 seconds, BT NOAS issue a BYE and the call is torn down. A fix has been provided in a custom build **IP Office 9.0.110.1**. With this build, it requires the configuration of a **NoUserSourceNumber (NUSN)** on IP Office.

The NUSN string consists of **SIP_LINE_NEAR_HOLD=N** where **N** is the active SIP Line number. With this NUSN string configured in IP Office, both RTP and RTCP packets are sent from IP Office to BT NOAS SIP trunk when on-hold resolving the call-hold issue. To configure IP Office to send both RTP and RTCP when on-hold, required by BTs SIP Trunk service, the following procedure may be used.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field, type **SIP_LINE_NEAR_HOLD=18**. Click **OK**. **Note:** 18 is the active SIP Line number configured in **Section 5.6**.



The source number **SIP_LINE_NEAR_HOLD=18** should now appear in the list of Source Numbers as shown below.



5.11. Save Configuration

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

6. BT NOAS SIP Trunk Service Configuration

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

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

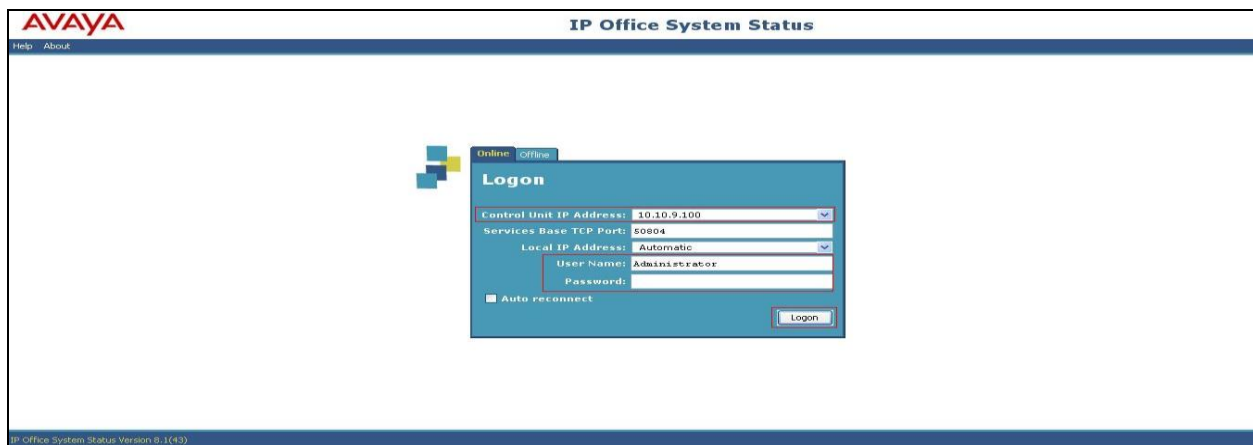
7. Verification Steps

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

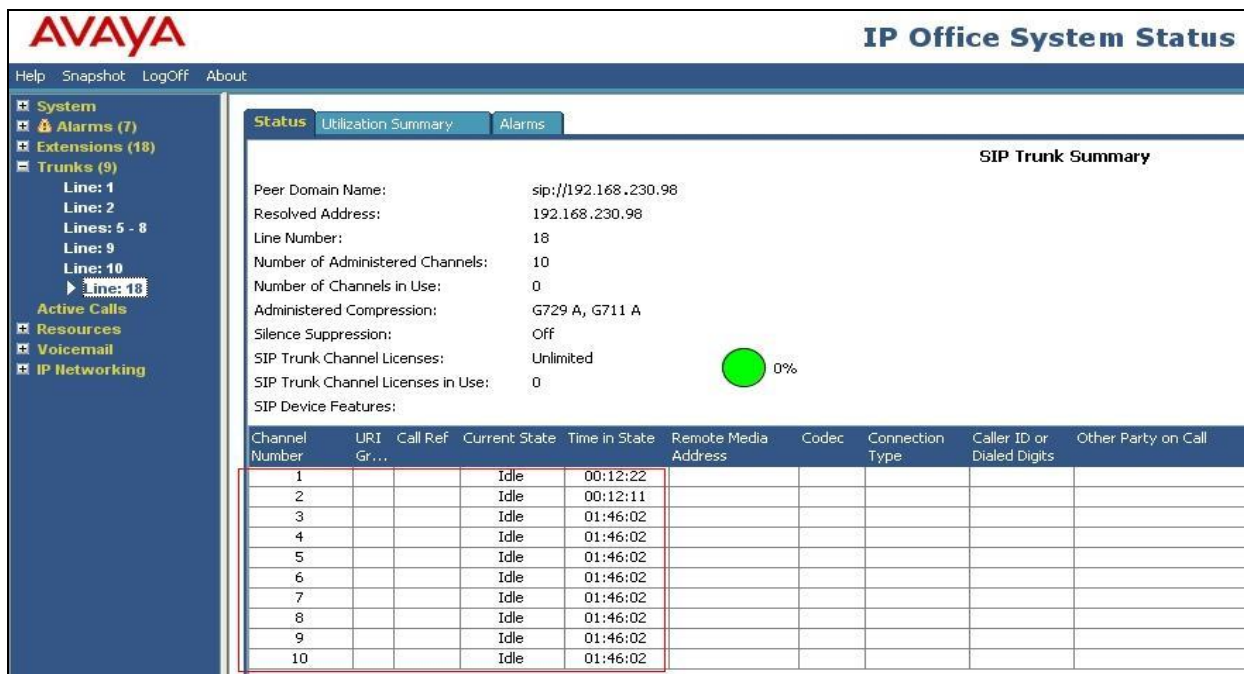
7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found on the PC where IP Office Manager is installed in PC programs under **Start → All Programs → IP Office → System Status** (not shown).

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



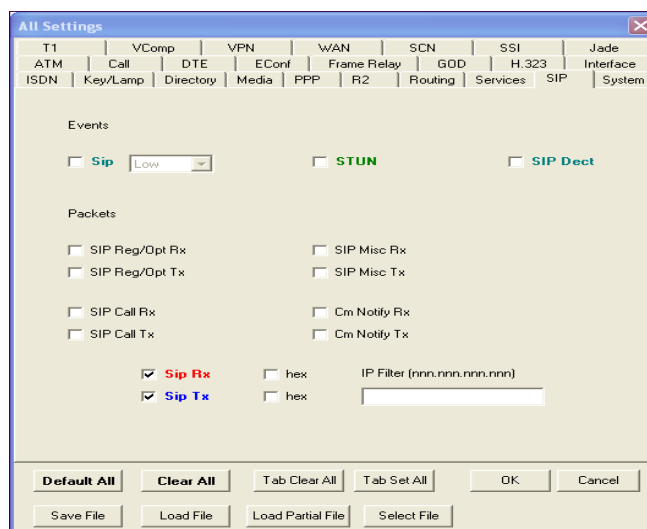
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational. IP address has been changed.



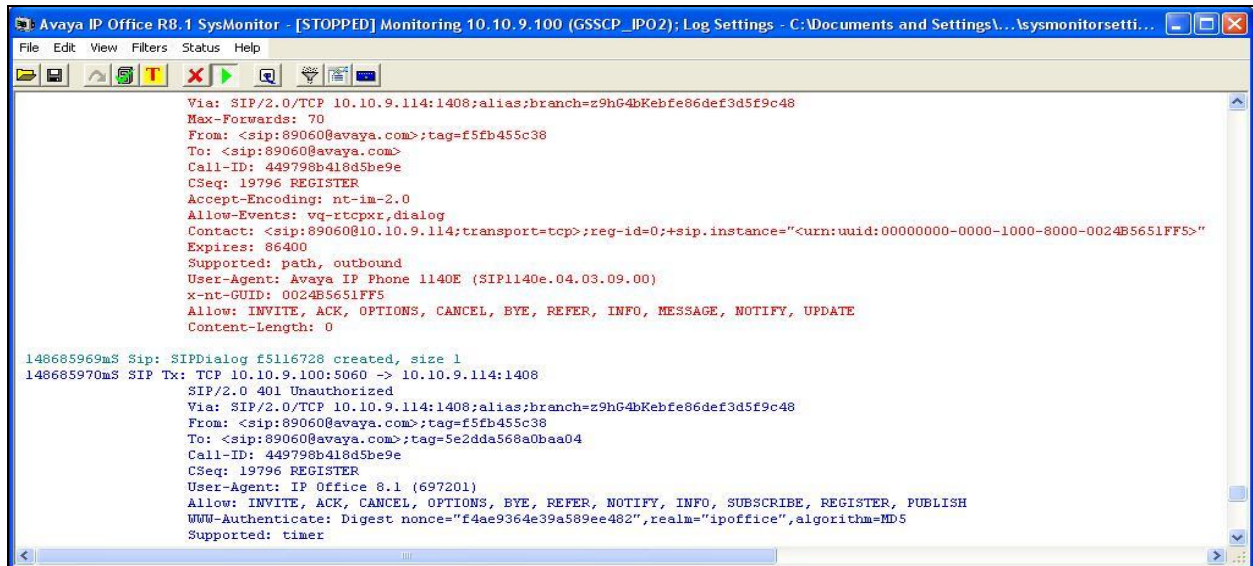
7.2. Monitor

The Monitor application can also be used to monitor and troubleshoot IP Office. Monitor can be accessed from **Start → Programs → IP Office → Monitor**. The application allows the monitored information to be customized. To customize, select the button that is third from the right in the screen below, or select **Filters → Trace Options**.

The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, the **SIP Rx** and **SIP Tx** boxes are checked. All SIP messages will appear in the trace with the color blue. To customize the color, right-click on **SIP Rx** or **SIP Tx** and select the desired color.



As an example, the following shows a portion of the monitoring window for a Registration attempt to the SIP trunk.



The screenshot shows the Avaya IP Office R8.1 SysMonitor application window. The title bar reads "Avaya IP Office R8.1 SysMonitor - [STOPPED] Monitoring 10.10.9.100 (GSSCP_IPO2); Log Settings - C:\Documents and Settings\...\sysmonitorsetti...". The window contains a log of SIP messages. The first message is a REGISTER request from 10.10.9.114 to 10.10.9.100. The second message is a 401 Unauthorized response from 10.10.9.100 to 10.10.9.114, indicating a failed registration attempt.

```
Via: SIP/2.0/TCP 10.10.9.114:1408;alias=branch=z9hG4bKebfe86def3d5f9c48
Max-Forwards: 70
From: <sip:890608avaya.com>;tag=f5fb455c38
To: <sip:890608avaya.com>
Call-ID: 449798b418d5be9e
CSeq: 19796 REGISTER
Accept-Encoding: nt-im-2.0
Allow-Events: vg-rtcpxr,dialog
Contact: <sip:89060810.10.9.114;transport=tcp>;reg-id=0;+sip.instance="urn:uuid:00000000-0000-1000-8000-0024b5651ff5"
Expires: 86400
Supported: path, outbound
User-Agent: Avaya IP Phone 1140E (SIP1140e.04.03.09.00)
x-nt-GUID: 0024b5651ff5
Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, REFER, INFO, MESSAGE, NOTIFY, UPDATE
Content-Length: 0

148685969mS Sip: SIPDialog f5116728 created, size 1
148685970mS SIP Tx: TCP 10.10.9.100:5060 -> 10.10.9.114:1408
SIP/2.0 401 Unauthorized
Via: SIP/2.0/TCP 10.10.9.114:1408;alias=branch=z9hG4bKebfe86def3d5f9c48
From: <sip:890608avaya.com>;tag=f5fb455c38
To: <sip:890608avaya.com>;tag=5e2dda568a0baa04
Call-ID: 449798b418d5be9e
CSeq: 19796 REGISTER
User-Agent: IP Office 8.1 (697201)
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, SUBSCRIBE, REGISTER, PUBLISH
WWW-Authenticate: Digest nonce="f4ae9364e39a589ee482",realm="ipoffice",algorithm=MD5
Supported: timer
```

8. Conclusion

These Application Notes describe the procedures required to configure the connectivity between Avaya IP Office and BT NOAS SIP Trunk solution as shown in **Figure 1**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and demonstrates Avaya IP Office can be configured to interoperate successfully with BT NOAS SIP Trunk Service. BT NOAS SIP Trunk Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

9. Additional References

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

- [1] *Avaya IP Office 8.1* Documentation CD, 16th July 2012.
- [2] *IP Office 8.1 Installation Manual*, Document Number 15-601042, August 2012.
- [3] *IP Office Manager Manual 10.0*, Document Number 15-601011, August 2012
- [4] *IP Office Release 8.1 Implementing Voicemail Pro*, Document Number 15-601064, June 2012
- [5] *System Status Application*, Document number 15-601758, 12th November 2011
- [6] *IP Office Softphone Installation*, 28th September 2011
- [7] *IP Office SIP Extension Installation*, 3rd October 2011
- [8] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>

©2014 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.