



Application Notes for Configuring Avaya IP Office 10.0 with Colt VoIP Access – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Colt VoIP Access and Avaya IP Office.

Colt VoIP Access provides PSTN access via a SIP Trunk connected to the Colt Voice over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. Colt 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 Colt VoIP Access and Avaya IP Office. Customers using this Avaya SIP-enabled enterprise solution with Colt's 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 Colt VoIP Access. 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 Colt VoIP Access. 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 Colt VoIP Access.
- 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 Colt VoIP Access.
- 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.
- Calls using G.729A and G.711A codec's.
- 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 Colt VoIP Access with the following observations:

- No inbound Toll-Free access was available to test.
- Routing was not in place to test Operator or Directory enquiries calls.
- Emergency calls were not tested as there was no test call booked with the Emergency Services Operator
- IP Office has a facility to "Indicate HOLD" that, if enabled, uses re-INVITEs to indicate to the network that call is put on and taken off hold. When enabled, the network cleared the call after 5 minutes. The feature was disabled for testing.
- The Colt network supports REFER and inbound blind call transfer failed when REFER was used. This is not supported in IP Office and support of REFER was disabled for this test.
- T.38 Fax tests only worked when NAT was used on the Colt IP address. This is a known bug on IP Office for IP addresses starting with hex "D". It is resolved in Service Pack 3 due for release on the 24th of February 2017. All fax calls worked when using G.711.

2.3. Support

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

For technical support on Colt products please contact the Colt authorized representative at: www.colt.net or Colt Local Support numbers.

Austria	0800 880 990	Belgium	0800 507 01
Germany	0800 111 1230	France	0800 948 888
Italy	192090	Netherlands	0800 265 8023
Portugal	808 780 222	Spain	901 888400
Switzerland	0800 560 560	UK	0800 136 166

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to Colt VoIP Access. Located at the enterprise site is an Avaya IP Office Server and 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 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.

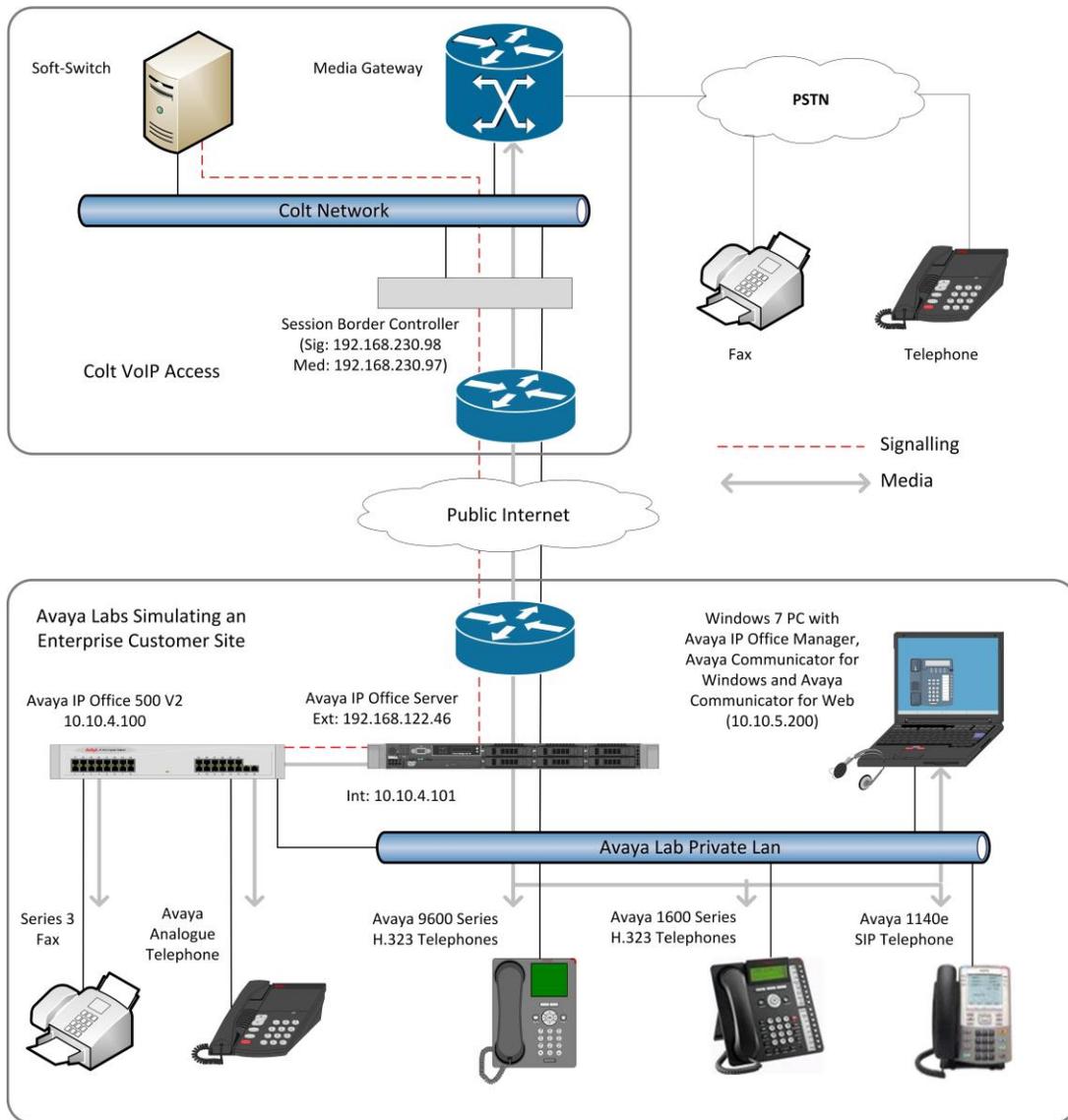


Figure 1: Colt SIP Trunk to Avaya IP Office Topology

4. Equipment and Software Validated

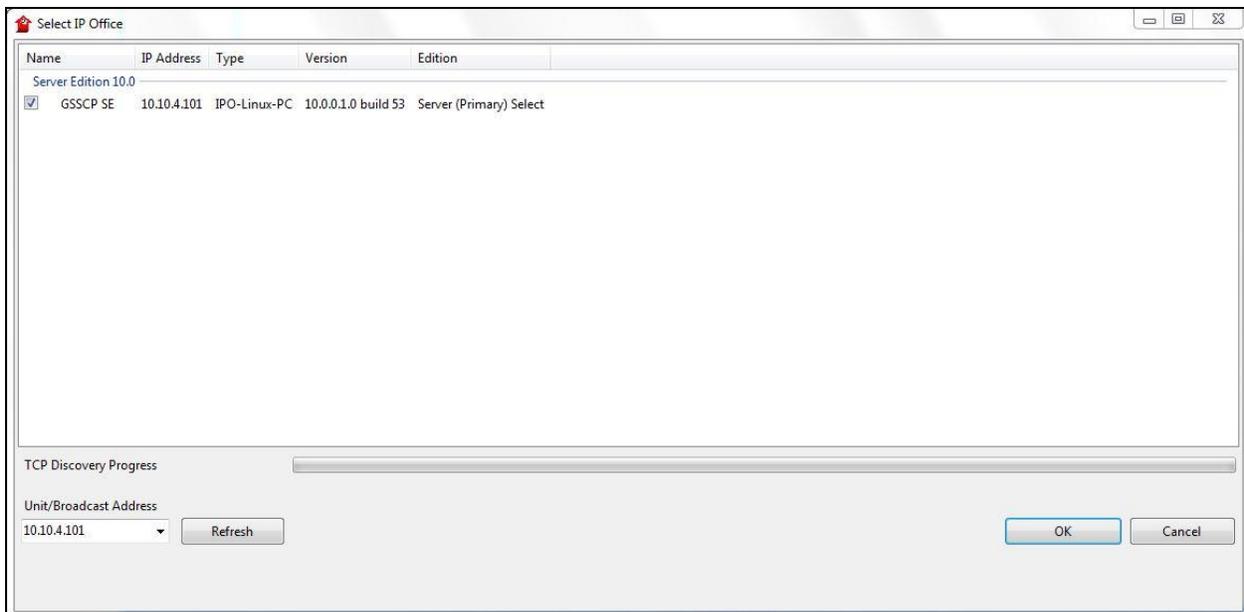
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	10.0.0.1.0 Build 53
Avaya 1140e IP SIP Telephone	04.04.23.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.6.3.02 V474
Avaya 98390 Analogue Phone	N/A
Avaya Communicator for Windows	2.1.3.0-NGUE- FLAREWINIPOREGRESSION10- JOB1.237
Avaya IP Office Manager	Version 10.0.0.1.0 build 53
Colt	
Sonus GSX	GSX9000: V09.02.05 F004
Sonus PSX	PSX: V09.03.02 R000

Testing was performed with IP Office Server Edition R10.0 with IP Office 500 V2 R10.0 as an Expansion. 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. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks, this includes T.38 fax.

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to Colt VoIP Access. 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 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 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 Colt.

Configuration window showing License details for a Remote Server. The License Mode is License Normal, Licensed Version is 10.0, and PLDS File Status is Valid. The Select Licensing is Valid.

Feature	Instances	Status	Expiry Date	Source
Receptionist	10	Valid	08/03/2017	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	08/03/2017	PLDS Nodal
VMPro Recordings Administrators	1	Valid	08/03/2017	PLDS Nodal
Office Worker	1000	Valid	08/03/2017	PLDS Nodal
VMPro TTS Professional	40	Valid	08/03/2017	PLDS Nodal
IPSec Tunnelling	1	Obsolete	08/03/2017	PLDS Nodal
Power User	1000	Valid	08/03/2017	PLDS Nodal
Avaya IP endpoints	1000	Valid	08/03/2017	PLDS Nodal
SIP Trunk Channels	256	Valid	08/03/2017	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	08/03/2017	PLDS Nodal
CTI Link Pro	1	Valid	08/03/2017	PLDS Nodal
Wave User	16	Obsolete	08/03/2017	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	08/03/2017	PLDS Nodal
Server Edition R10	150	Valid	08/03/2017	PLDS Nodal
UMS Web Services	1000	Valid	08/03/2017	PLDS Nodal
Avaya Mac Softphone	1000	Valid	08/03/2017	PLDS Nodal
Avaya Softphone Licence	1000	Valid	08/03/2017	PLDS Nodal
SM Trunk Channels	128	Valid	08/03/2017	PLDS Nodal
Web Collaboration	64	Valid	08/03/2017	PLDS Nodal

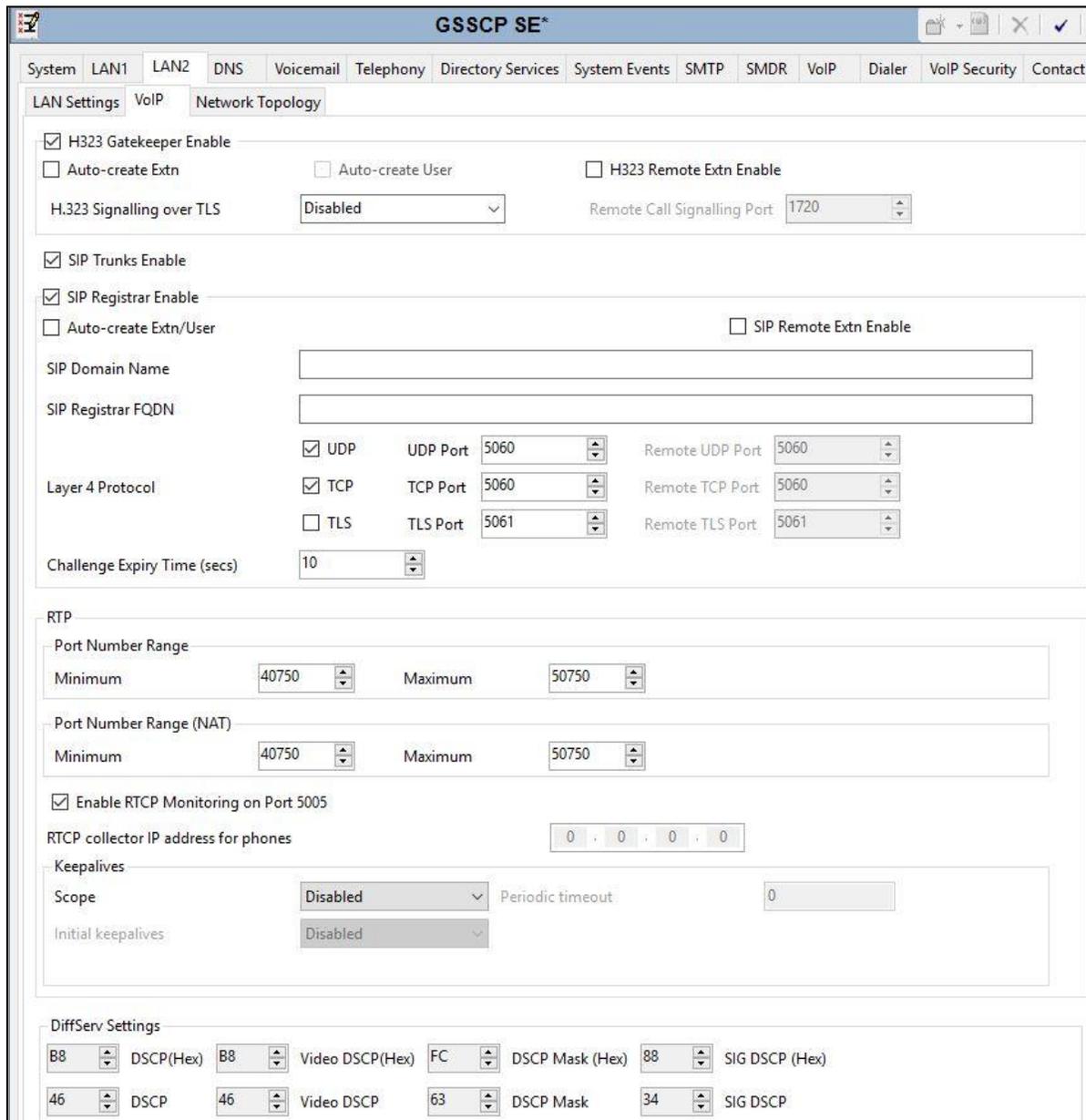
5.2. LAN2

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_SE** 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 used for incoming IP packets not associated with a service or session and is not used in the test environment. All other parameters should be set according to customer requirements. On completion, click the OK button (not shown).

Configuration window showing LAN2 settings for GSSCP SE. The LAN Settings tab is selected, and the VoIP sub-tab is active. The IP Address is 192.168.122.46 and the IP Mask is 255.255.255.128. The Number Of DHCP IP Addresses is 200. The DHCP Mode is Disabled.

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 **40750 to 50750**, though in this case the default values of **49152 to 53246** would have been equally effective.



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, clear the **STUN Server IP Address** field and select **Open Internet** in the **Firewall/NAT Type** drop down menu as NAT is not required in this configuration.

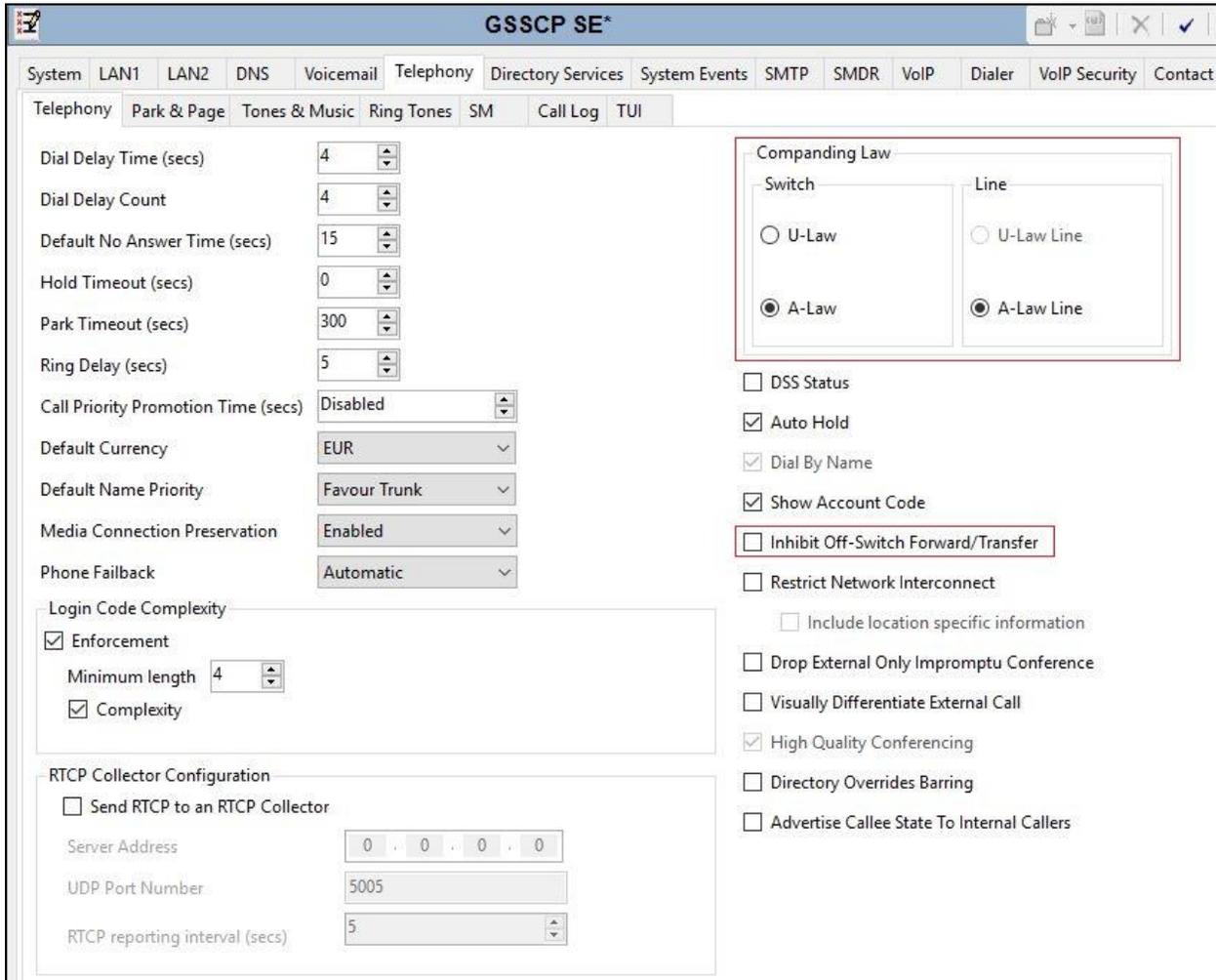
The Network Topology tab can be used to set the **Binding Refresh Time** for the periodic sending of OPTIONS. During testing, IP Office sent OPTIONS messages at an interval of 5 minutes. This was achieved by setting the **Binding Refresh Time** to **300**.

The screenshot shows the 'GSSCP SE*' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** An empty text input field.
- STUN Port:** A dropdown menu set to '3478'.
- Firewall/NAT Type:** A dropdown menu set to 'Open Internet'.
- Binding Refresh Time (seconds):** A spinner control set to '300'.
- Public IP Address:** A field containing '0 . 0 . 0 . 0'.
- Public Port:** A section with three sub-fields: 'UDP' (0), 'TCP' (0), and 'TLS' (0), each with a spinner control.
- Run STUN on startup:** An unchecked checkbox.
- Buttons:** 'Run STUN' and 'Cancel' buttons are located to the right of the Public IP Address field.

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).



5.4. Codec Settings

Navigate to the **Codecs** tab on the Details Pane. Check the Available Codecs boxes as required for the IP endpoints. 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** and **G.711 ALAW 64K** were used as the default codec's. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).

The screenshot displays the 'GSSCP SE' configuration interface for VoIP. The 'VoIP' tab is selected, showing various settings. Under 'RFC2833 Default Payload', the value is set to '101'. The 'Available Codecs' section contains a list of codecs with checkboxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, and G.729(a) 8K CS-ACELP. The 'Default Codec Selection' section is divided into 'Unused' and 'Selected' lists. The 'Unused' list contains G.711 ULAW 64K and G.722 64K. The 'Selected' list contains G.729(a) 8K CS-ACELP and G.711 ALAW 64K. Navigation buttons (right arrow, up arrow, left arrow, down arrow, right arrow) are positioned between the 'Unused' and 'Selected' lists.

Note: The codec settings for IP endpoints can also be used for the SIP Trunk by selecting **System Default** in the **Codec Selection** as shown in **Section 5.5.2**.

5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and Colt VoIP Access. 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**.

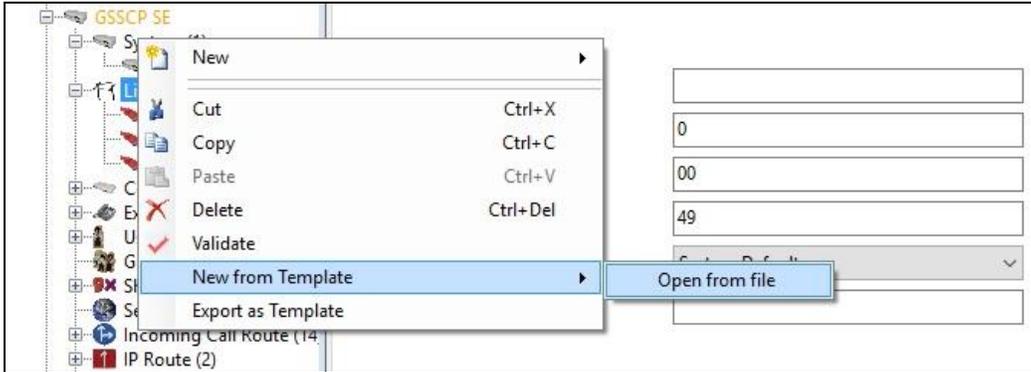
Also, the following SIP Line settings are not supported on Basic Edition:

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

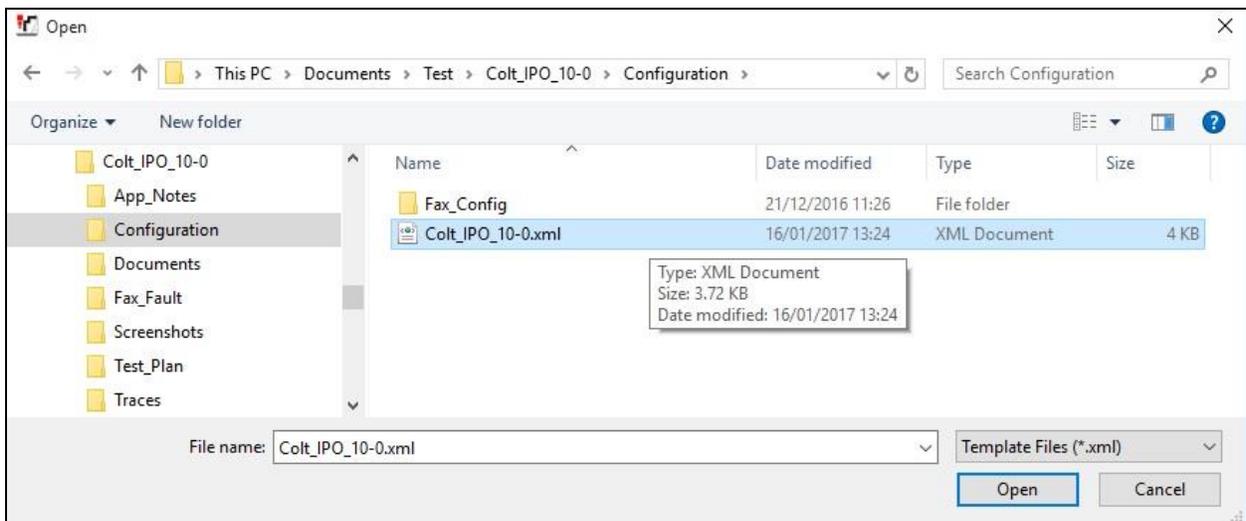
Alternatively, a SIP Line can be created manually. To do so, right-click Line in the Navigation Pane and select **New→SIP Line** (not shown). Then, follow the steps outlined in **Section 5.5.2**.

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 5.5.2**.

5.5.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to a domain name agreed with Colt. In test this was set to **colt.net**. 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 page for 'SIP Line - Line 20'. The left sidebar contains a tree view of the system configuration, with 'Line 20' selected. The main area is divided into several sections:

- Line Information:** Line Number (20), In Service (checked), ITSP Domain Name (colt.net), Check OOS (checked), Local Domain Name (empty), URI Type (SIP), Location (Cloud).
- Session Timers:** Refresh Method (Auto), Timer (seconds) (3600).
- Prefixes:** Prefix (empty), National Prefix (0), International Prefix (00).
- Country Code:** 44.
- Name Priority:** System Default.
- Description:** (empty).
- Redirect and Transfer:** Incoming Supervised REFER (Never), Outgoing Supervised REFER (Never), Send 302 Moved Temporarily (unchecked), Outgoing Blind REFER (unchecked).

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

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

- Set **ITSP Proxy Address** to the IP address for Colt VoIP Access.
- 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**.
- Leave **Explicit DNS Server(s)** at default value unless domain name is to be used.

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 '192.168.230.98'. 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 set to '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. There is an empty 'Separate Registrar' field.

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 main area is empty, and there are 'Add...' and 'Remove' buttons on the right side.

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 specific DDI number for voicemail and the Mobile Twinning FNE.

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**.
- Leave the **Originator Number** for **Forwarding and Twinning** blank so that the DDI number for the User is sent as the calling party number. 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.
- 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 window 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	
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 5.4**, 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 Colt this was **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K**. This reflected the codec list received from the network. The G.726 codec is also supported on Colt VoIP Access, but this is not supported on IP Office.
- Select **T38** in the **Fax Transport Support** drop down menu to allow T.38 fax operation. Alternatively, if the T.38 fax fault described in **Section 2.2** is encountered, enter G.711. Refer to **Section 5.9** for T.38 Fax configuration.
- Select **RFC2833/RFC4733** 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).

SIP Line - Line 20*

SIP Line Transport SIP URI **VoIP** SIP Credentials SIP Advanced Engineering

Codec Selection: Custom

Unused: G.711 ULAW 64K, G.722 64K

Selected: G.729(a) 8K CS-ACELP, G.711 ALAW 64K

Fax Transport Support: T38

DTMF Support: RFC2833/RFC4733

Media Security: Media Security Features Disabled

Local Hold Music
 Re-invite Supported
 Codec Lockdown
 Allow Direct Media Path
 Force direct media with phones
 PRACK/100rel Supported

On an IP Office 500V2 system, the T.38 fax settings would be defined at this point. The system used for testing was an IP Office Server Edition with a 500V2 as an expansion. The configuration of the expansion is described in **Section 5.9**.

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

- Check the **Use + for International** box to convert outbound dialled numbers to E.164 with leading “+”.
- Check the **Use API for Privacy** box to send the calling party number for outbound calls with CLI Restricted in the P-Asserted-Identity header.
- Check the **Caller ID from From header** box so that the calling number displayed on IP Office extensions is from the From header and not an alternative such as P-Asserted-Identity.
- Default values may be used for all other parameters.

The screenshot shows the 'SIP Line - Line 20*' configuration window with the 'SIP Advanced' tab selected. The configuration is as follows:

Section	Parameter	Value
Addressing	Association Method	By Source IP address
	Call Routing Method	Request URI
	Suppress DNS SRV Lookups	<input type="checkbox"/>
Identity	Use "phone-context"	<input type="checkbox"/>
	Add user=phone	<input checked="" type="checkbox"/>
	Use + for International	<input checked="" type="checkbox"/>
	Use PAI for Privacy	<input checked="" type="checkbox"/>
	Use Domain for PAI	<input type="checkbox"/>
	Swap From and PAI/Diversion	<input type="checkbox"/>
	Caller ID from From header	<input checked="" type="checkbox"/>
	Send From In Clear	<input type="checkbox"/>
	Cache Auth Credentials	<input checked="" type="checkbox"/>
User-Agent and Server Headers	[Empty text box]	
Send Location Info	Never	
Media	Allow Empty INVITE	<input type="checkbox"/>
	Send Empty re-INVITE	<input type="checkbox"/>
	Allow To Tag Change	<input type="checkbox"/>
	P-Early-Media Support	None
	Send SilenceSupp=Off	<input type="checkbox"/>
	Force Early Direct Media	<input type="checkbox"/>
Media Connection Preservation	Media Connection Preservation	Disabled
	Indicate HOLD	<input type="checkbox"/>
Call Control	Call Initiation Timeout (s)	4
	Call Queuing Timeout (m)	5
	Service Busy Response	503 - Service Unavailable
	on No User Responding Send	408-Request Timeout
	Action on CAC Location Limit	Allow Voicemail
	Suppress Q.850 Reason Header	<input type="checkbox"/>
	Emulate NOTIFY for REFER	<input type="checkbox"/>
	No REFER if using Diversion	<input type="checkbox"/>

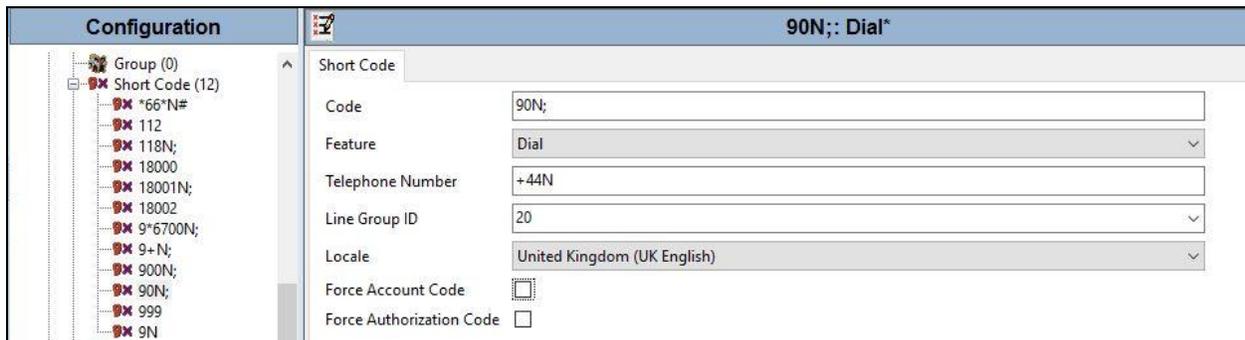
On completion, click the **OK** button (not shown).

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 5.5** 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.

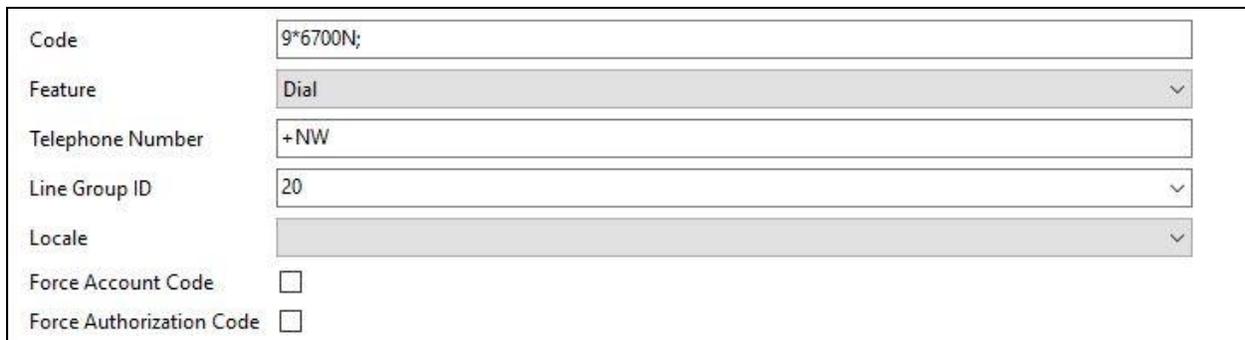
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **90N**; which will be invoked when the user dials 9 followed by a national number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **+44N** which removes the access code and inserts the public number with country code and leading “+” 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).



Configuration		90N;: Dial*	
Short Code			
Code		90N;	
Feature		Dial	▼
Telephone Number		+44N	
Line Group ID		20	▼
Locale		United Kingdom (UK English)	▼
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*6700N** which is an outbound international 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.



Code		9*6700N;	
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 5.5**. 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 a SIP Endpoint.

- Change the **Name** of the User if required, this will be used for login to the Avaya Communicator for Windows Softphone.
- The **Password** and **Confirm Password** fields are set for login from Softphones.
- Select the required profile from the **Profile** drop down menu. **Basic User** is commonly used, **Power User** can be selected for SIP softphone, WebRTC and Remote Worker endpoints.

The screenshot shows the Avaya User configuration interface for user 89108. The interface is titled "User" and "89108: 89108". The left sidebar shows a list of users, with 89108 selected. The main area displays various configuration fields and tabs. The tabs include User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The User tab is active, showing the following fields:

Name	89108
Password	••••••••
Confirm Password	••••••••
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	89108
Extension	89108
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Power User

Below the Profile dropdown, there are several checkboxes:

- 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 Colt in E.164 format with leading “+”.

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).

89108: 89108*						
Voice Recording	Button Programming	Menu Programming	Mobility	Group Membership	Announcements	SIP
SIP Name	<input type="text" value="+4420719nnnn2"/>					
SIP Display Name (Alias)	<input type="text" value="020719nnnn2"/>					
Contact	<input type="text" value="+4420719nnnn2"/>					
<input type="checkbox"/> 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.

Incoming Call Route		20 +4420719nnnn2	
Line Group ID	Incoming Number	Standard	Voice Recording
20	+4420719nnnn2	Destinations	
20	+4420719nnnn1	Bearer Capability: Any Voice	
20	+4420719nnnn3	Line Group ID: 20	
20	+4420719nnnn0	Incoming Number: +4420719nnnn2	
20	+4420719nnnn4	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 for incoming calls is E.164 with leading “+”.

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 89108.

Standard	Voice Recording	Destinations	
		TimeProfile	Destination
		Default Value	89108 89108
			Fallback Extension

Note: Calls coming in to destinations not associated with an extension such as Voice Mail and FNE also appear on line 20 in this configuration. The destinations are defined as the short codes for Voicemail Collect and FNE Service.

5.9. T.38 Fax

At Release 10, T.38 Fax is supported on IP Office Server Edition when using an IP Office Expansion (500 V2). Colt VoIP Access testing was carried out using this configuration with only the analogue extension for the fax machine on the Expansion. In this configuration, the T.38 fax settings are configured on the SIP line between the Expansion and the Server.

5.9.1. Analogue User

To configure the settings for the fax User, first navigate to **User** in the Navigation Pane for the Expansion. In the test environment, the 500V2 Expansion is called **GSSCP_IPO2**. Select the **User** tab.

The following example shows the configuration required for an analogue Endpoint.

- Change the **Name** of the User if required.
- The **Password** and **Confirm Password** fields are set but are not required for analogue endpoints.
- Select the required profile from the **Profile** drop down menu. **Basic User** is sufficient for fax.

Configuration	User	Extn89022: 89022									
BOOTP (0)	Name	User	Voicemail	DND	Short Codes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming
Operator (3)	89070	Name	Extn89022								
Solution	Analog89021	Password	••••••••								
User(34)	Extn89000	Confirm Password	••••••••								
Group(1)	Extn89001	Unique Identity									
Short Code(52)	Extn89002	Conference PIN									
Directory(0)	Extn89003	Confirm Audio Conference PIN									
Time Profile(0)	Extn89004	Account Status	Enabled								
Account Code(0)	Extn89005	Full Name									
User Rights(9)	Extn89006	Extension	89022								
Location(0)	Extn89007	Email Address									
GSSCP_SE	Extn89010	Locale									
GSSCP_IP02	Extn89011	Priority	5								
System (1)	Extn89012	System Phone Rights	None								
Line (10)	Extn89013	Profile	Basic User								
Control Unit (4)	Extn89014	<input type="checkbox"/> Receptionist									
Extension (32)	Extn89015	<input type="checkbox"/> Enable Softphone									
User (32)	Extn89016	<input type="checkbox"/> Enable one-X Portal Services									
Group (1)	Extn89017	<input type="checkbox"/> Enable one-X TeleCommuter									
Short Code (21)	Extn89018	<input type="checkbox"/> Enable Remote Worker									
Service (0)	Extn89020	<input type="checkbox"/> Enable Communicator									
RAS (1)	Extn89022	<input type="checkbox"/> Enable Mobile VoIP Client									
Incoming Call Route (8)	Extn89023	<input type="checkbox"/> Send Mobility Email									
WAN Port (0)	Extn89024	<input type="checkbox"/> Web Collaboration									
Firewall Profile (1)	Extn89025	<input type="checkbox"/> Exclude From Directory									
IP Route (3)	Extn89026										
License (107)	Extn89027										
Tunnel (0)	Extn89028										
ARS (1)	Mailbox										
Location (0)	NoUser										
Authorization Code (0)	RemoteManager										
	SIP89050										
	SIP89060										

Configure other settings as described in **Section 5.7**.

5.9.2. T.38 Fax Settings

The T.38 Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for T.38 Fax are required in three places in this configuration:

- The SIP Line for Colt VoIP Access as described in **Section 5.5**.
- The SIP Line between the Server and the Expansion at the Expansion end.
- The SIP Line between the Server and the Expansion at the Server end.

In all the above cases, the Fax Transport Support was set to T38.

The T.38 Fax settings are defined in the T38 Fax tab. This is available on the Expansion only. During testing, these settings were left at default values apart from **T38 Fax Version**.

- To change T.38 Fax settings, first uncheck **Use Default Values**.
- Select **0** from the **T38 Fax Version** drop down menu.

The screenshot shows the configuration window for 'IP Office Line - Line 17'. On the left is a 'Line' list with 21 entries, including Line 17 (IP Office Line) and Line 21 (SIP Line). The main area has tabs for 'Line', 'Short Codes', 'VoIP Settings', and 'T38 Fax'. The 'T38 Fax' tab is active, showing various settings:

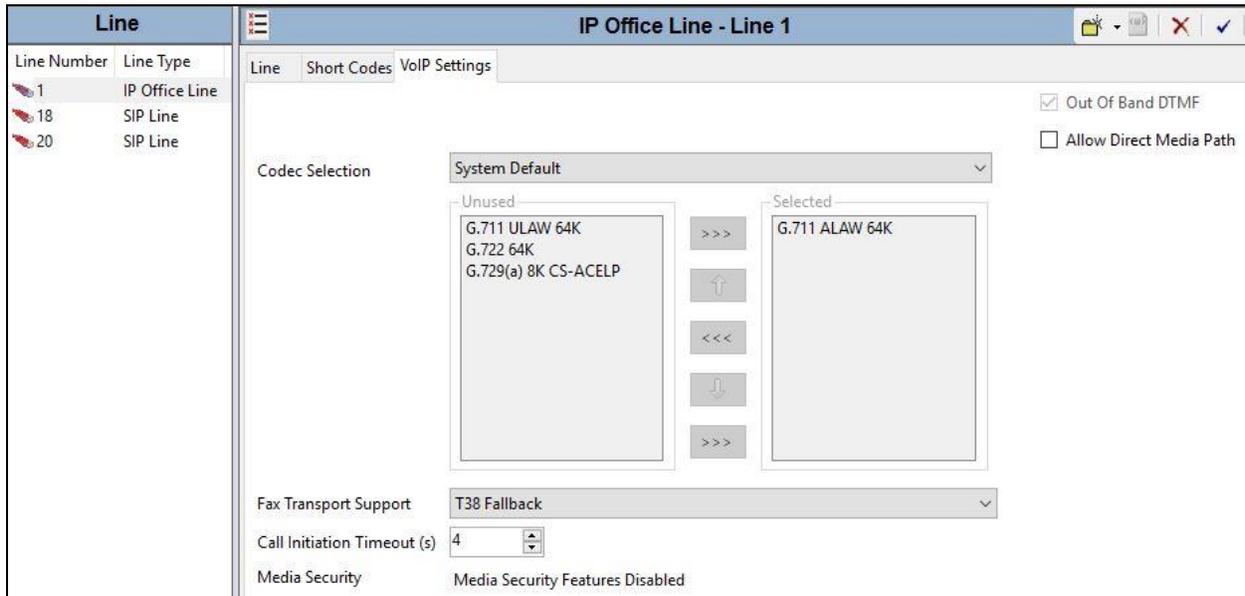
- T38 Fax Version: 0
- Transport: UDPTL
- Redundancy: Low Speed (0), High Speed (0)
- TCF Method: Trans TCF
- Max Bit Rate (bps): 14400
- EFlag Start Timer (msecs): 2600
- EFlag Stop Timer (msecs): 2300
- Tx Network Timeout (secs): 150
- Use Default Values:
- Scan Line Fix-up:
- TFOP Enhancement:
- Disable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

The following shows the **VoIP Settings** tab in the SIP Line for the Server in the Expansion configuration:

The screenshot shows the configuration window for 'IP Office Line - Line 17' with the 'VoIP Settings' tab active. The 'Line' list on the left highlights Line 17 (IP Office Line) and Line 21 (SIP Line). The 'VoIP Settings' tab shows the following configuration:

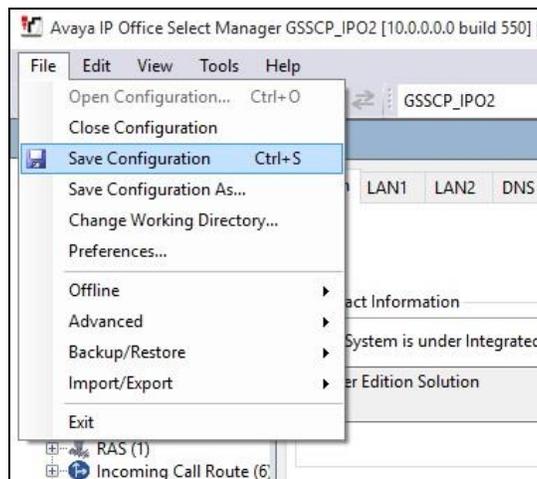
- Codec Selection: System Default
- Unused codecs: G.722 64K, G.723.1 6K3 MP-MLQ
- Selected codecs: G.711 ALAW 64K, G.729(a) 8K CS-ACELP, G.711 ULAW 64K
- Fax Transport Support: T38 Fallback
- Call Initiation Timeout (s): 4
- Media Security: Media Security Features Disabled
- VoIP Silence Suppression:
- Out Of Band DTMF:
- Allow Direct Media Path:

The following shows the **VoIP Settings** tab in the SIP Line for the Expansion in the Server configuration:



5.10. 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.

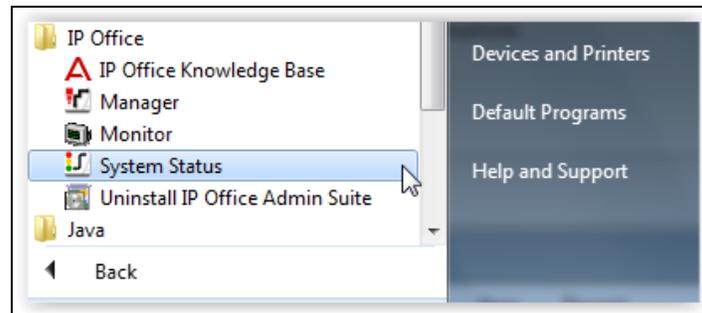


6. Verification Steps

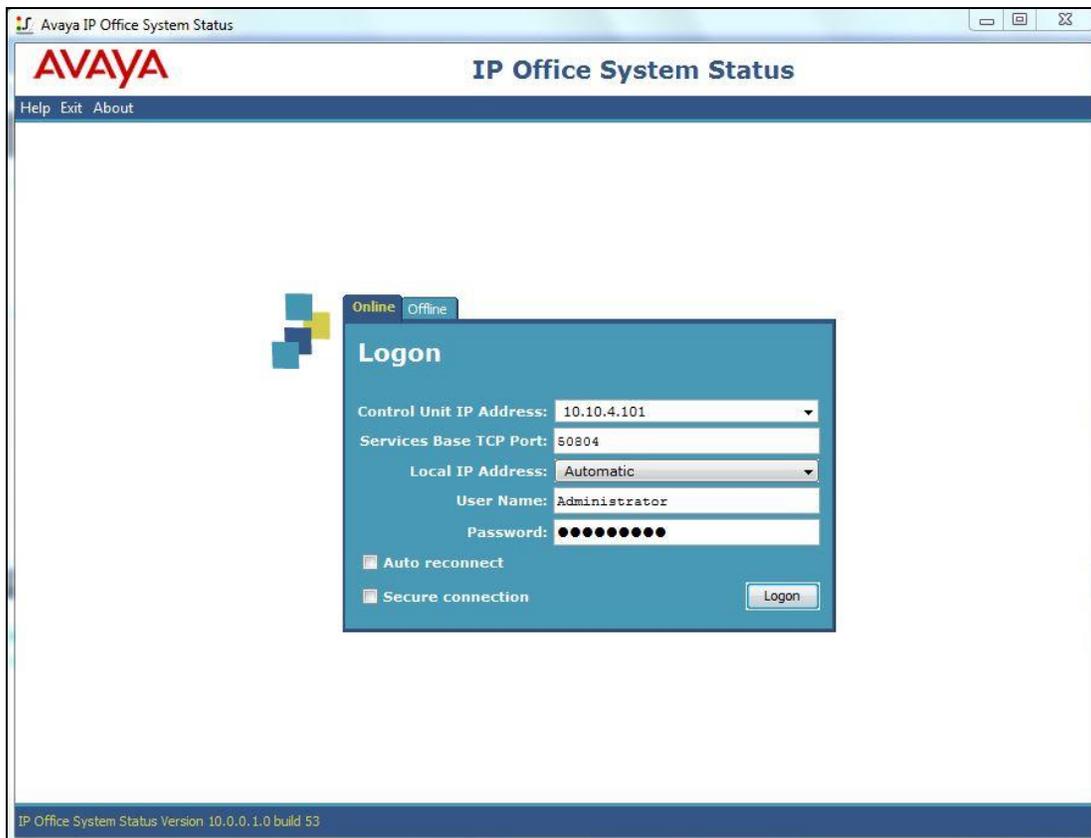
This section includes steps that can be used to verify that the configuration has been done correctly.

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 displays the Avaya IP Office System Status interface. The left-hand navigation menu is expanded to 'Trunks (2)', with 'Line: 20' selected. The main content area shows the 'SIP Trunk Summary' for Line 20. The status is 'In Service'. Key parameters include: Peer Domain Name: colt.net, Resolved Address: 192.168.230.98, Line Number: 20, Number of Administered Channels: 20, Number of Channels in Use: 0, Administered Compression: G729 A, G711 A, Enable Faststart: Off, Silence Suppression: Off, Layer 4 Protocol: TCP, SIP Trunk Channel Licenses: 256, and SIP Trunk Channel Licenses in Use: 0. A green circular progress indicator shows 0% utilization. Below the summary is a table with 7 channels, all in an 'Idle' state.

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	00:02...											
2			Idle	00:02...											
3			Idle	00:02...											
4			Idle	00:02...											
5			Idle	00:02...											
6			Idle	00:02...											
7			Idle	00:02...											

7. Conclusion

All tests for Colt 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/>

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