Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 8.1 with Colt VoIP Access service – 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.

The Colt VoIP Access service 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 are 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 Colt VoIP Access and Avaya IP Office. Colt VoIP Access provides PSTN access via a SIP trunk connected to the Colt network as an alternative to legacy Analogue or Digital trunks. This approach generally results in lower cost for customers.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office to connect to 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, Digital and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider
- Outgoing PSTN calls from various phone types including H.323, Digital, and Analogue telephones at the enterprise.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider
- Inbound and outbound PSTN calls to/from an IP Office Softphone client
- Various call types including: local, long distance, international, toll free (outbound)and directory assistance
- Codecs G.711A and G.729A
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- T.38 fax

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Colt VoIP Access with the following observations:

- The No Answer timer for incoming calls at 3 minutes and 40 seconds was longer than that normally observed
- Although Colt support G.722 and IBLC, the commonly supported codecs were limited to G.711A and G.729A
- The Colt test system does not route to Operator services and these calls were not tested
- P-Asserted-ID and Privacy headers are not used for delivery of restricted CLI on incoming calls. The enterprise equipment displays the user portion of the Request URI, and in test this was “Unavailable”. This is potentially misleading where users may be more familiar with more commonly used indicators for restricted CLI such as “Private”.
- The CLI displayed on terminating equipment for outgoing calls was a single number provided by the network regardless of the extension used. This suggests that CLI validation was not used in the network during test.
- Delays in speech up to 1 second for PSTN to PSTN calls were experienced during test and echo was significant. This is a result of numerous IP to TDM network hops in the test network and is not thought to be an issue for the live network.

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 500 v2. Endpoints include two Avaya 1600 Series IP Telephones (with H.323 firmware), an 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 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.

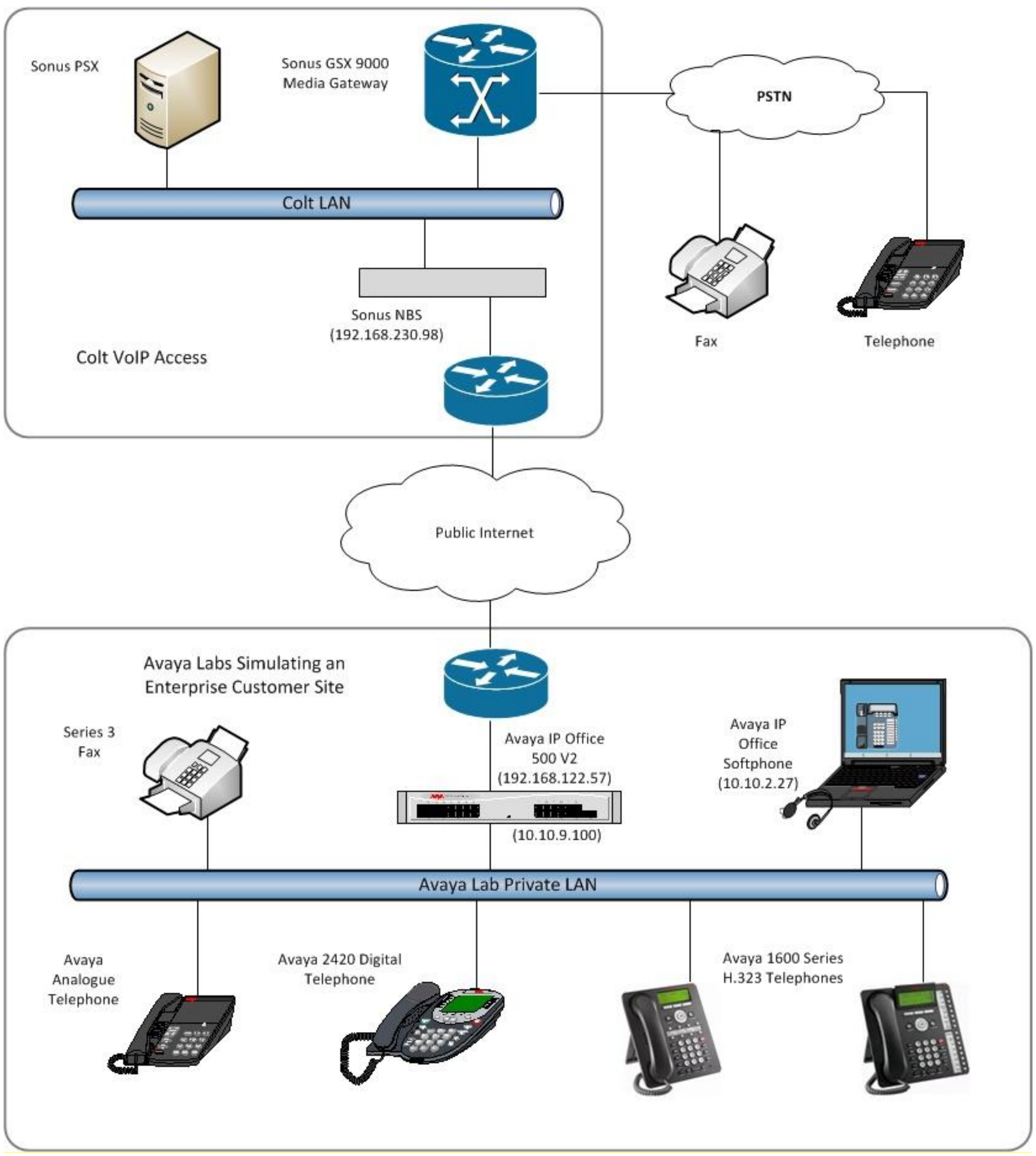


Figure 1: Colt VoIP Access Solution to Avaya IP Office Topology

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 9 + N digits to send digits across the SIP trunk to the Colt network. The short code of 9 is stripped off by Avaya IP Office and the remaining N digits sent with adaptation to E.164 format with leading "+".

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. Colt 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 Colt 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:

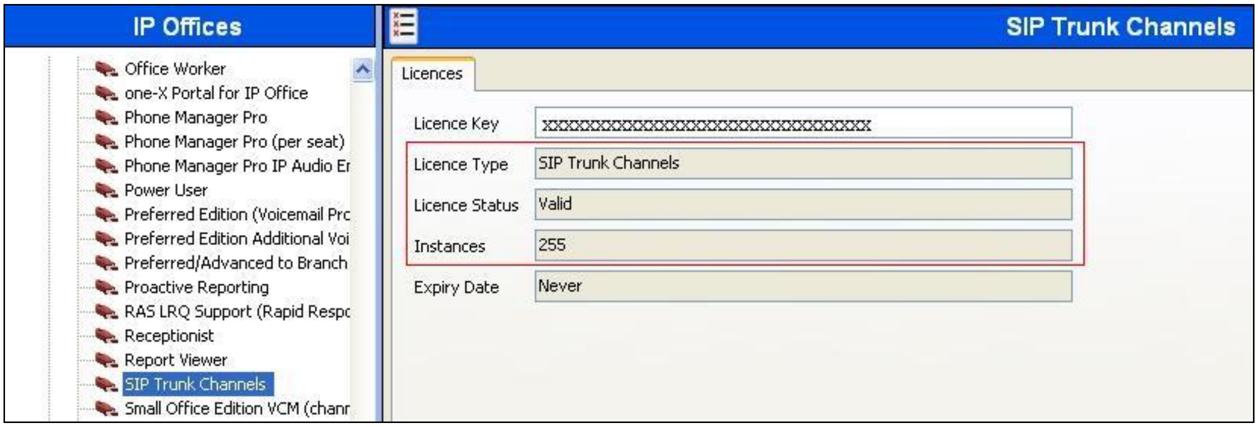
Equipment/Software	**Release/Version**
Avaya	
Avaya IP Office 500 V2	Avaya IP Office R8.1(43)
Avaya 1603 Phone (H.323)	1.3100
Avaya 1608 Phone (H.323)	1.3100
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Colt	
Sonus GSX/NBS 9000	V 8.4.2
Sonus PSX	V 8.4.2

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**, 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.

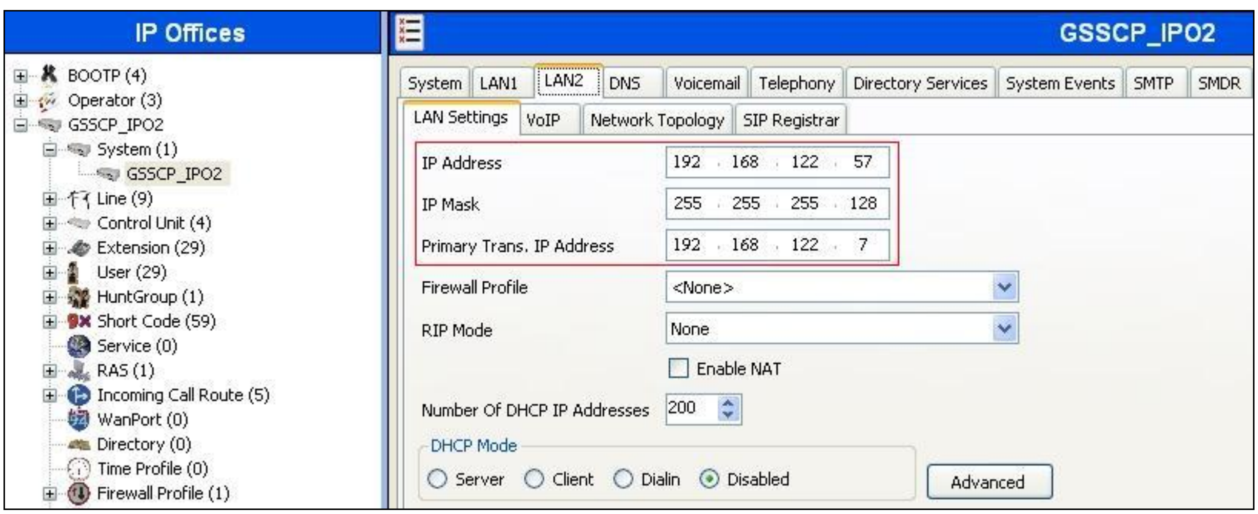
5.1. Verify System Capacity

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 Colt.



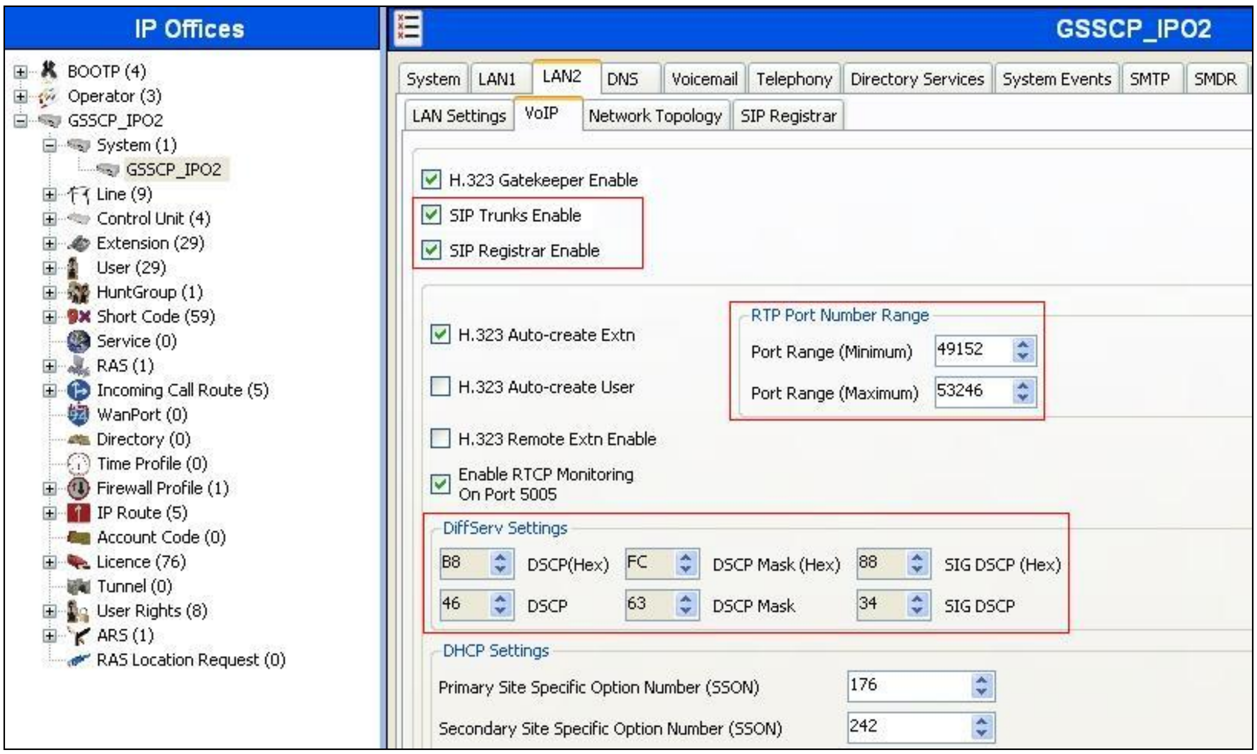
5.2. LAN2 Settings

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, **Primary Trans. IP Address** is the next hop, usually the default gateway address. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

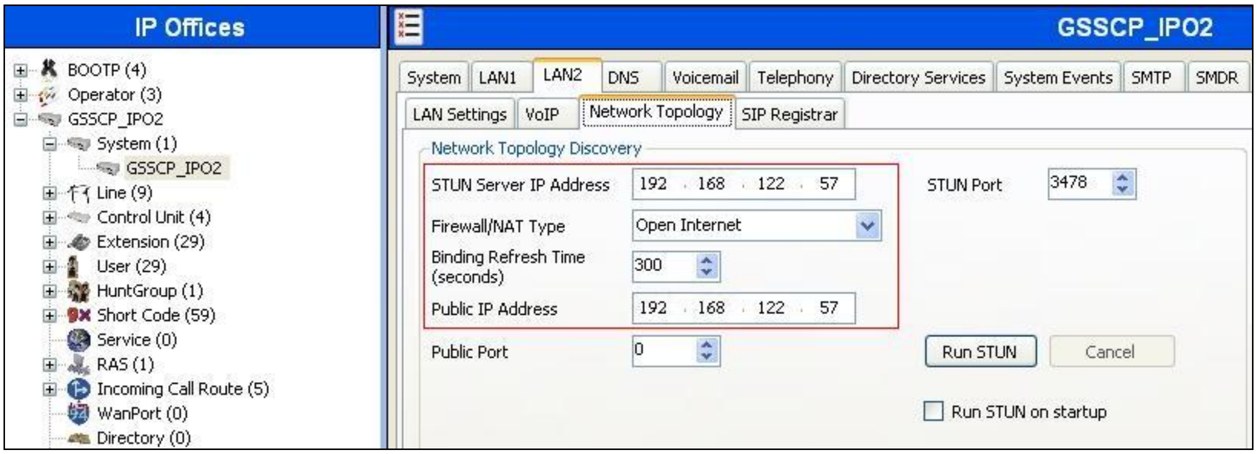


On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Based on this setting, Avaya IP Office would request RTP media be sent to a UDP port in the configurable range for calls using LAN2.

Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both 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).

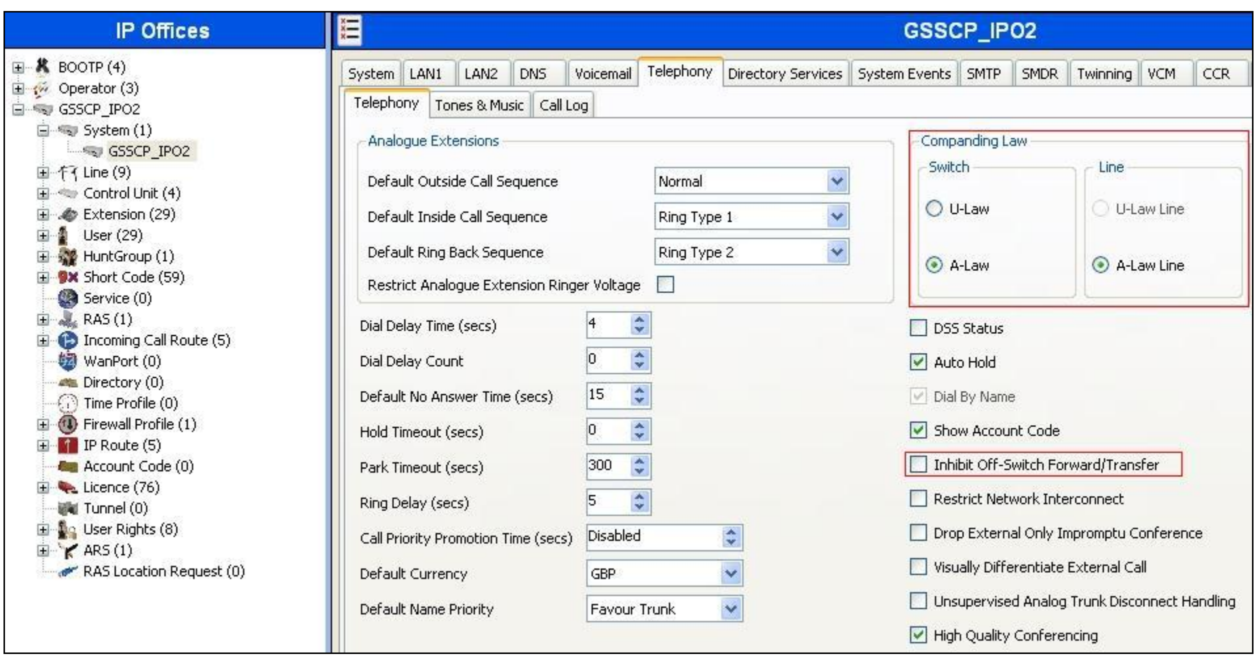


On the **Network Topology** tab in the Details Pane enter the **Public IP Address** for the IP Office. The same Public IP Address is used in the **STUN Server IP Address** field, even if not running STUN. It is important that the **Binding Refresh Time** is set to the correct value. Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. The rate at which the messages are sent is determined by the combination of the **Binding Refresh Time** (in seconds) set on the **Network Topology** tab and User settings, see **Section 5.10** for more details. Below is a sample configuration. On completion, click the **OK** button (not shown).



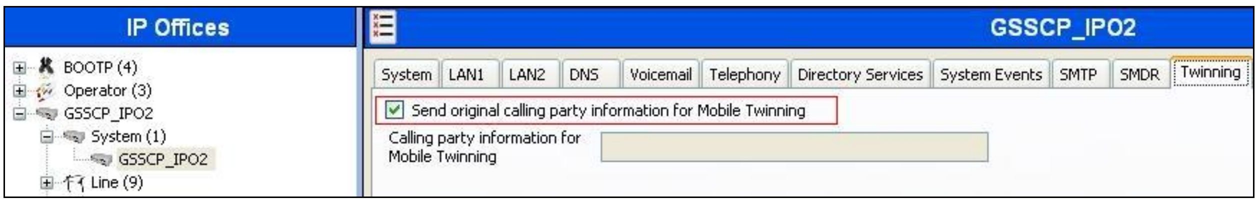
5.3. System Telephony Settings

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).



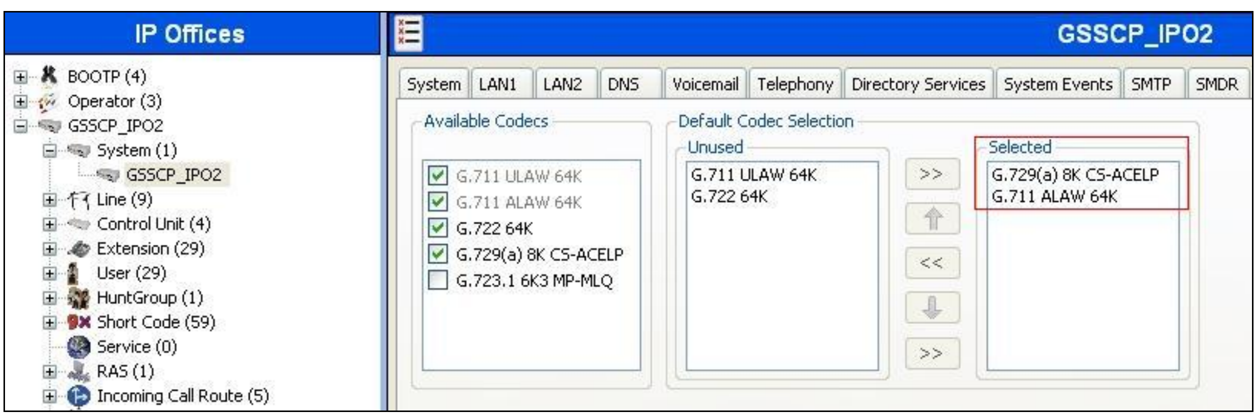
5.4. System Twinning Settings

Navigate to the **Twinning** tab, check the box labeled **Send original calling party information for Mobile Twinning**. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header (not the associated desk phone number) for calls that originate from an internal extension. For inbound PSTN calls to a twinned enabled phone, Avaya IP Office will continue to send the associated host phone number in the SIP From header (used for the caller display). This setting only affects twinning and does not impact the messaging of other redirected calls such as forwarded calls. If this box is checked, it will also override any setting of the **Send Caller ID** parameter on the SIP line (**Section 5.6**). On completion, click the **OK** button (not shown).



5.5. Codec Settings

Navigate to the **Codecs** tab (not shown) on the Details Pane. Check the Available Codecs boxes as required. Note that **G.711 ULAW 64K** and **G.711 ALAW 64K** are greyed out and always available. Once available codecs are selected, they can be used or unused by using the horizontal arrows as required. Note that in test, **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** were used. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).

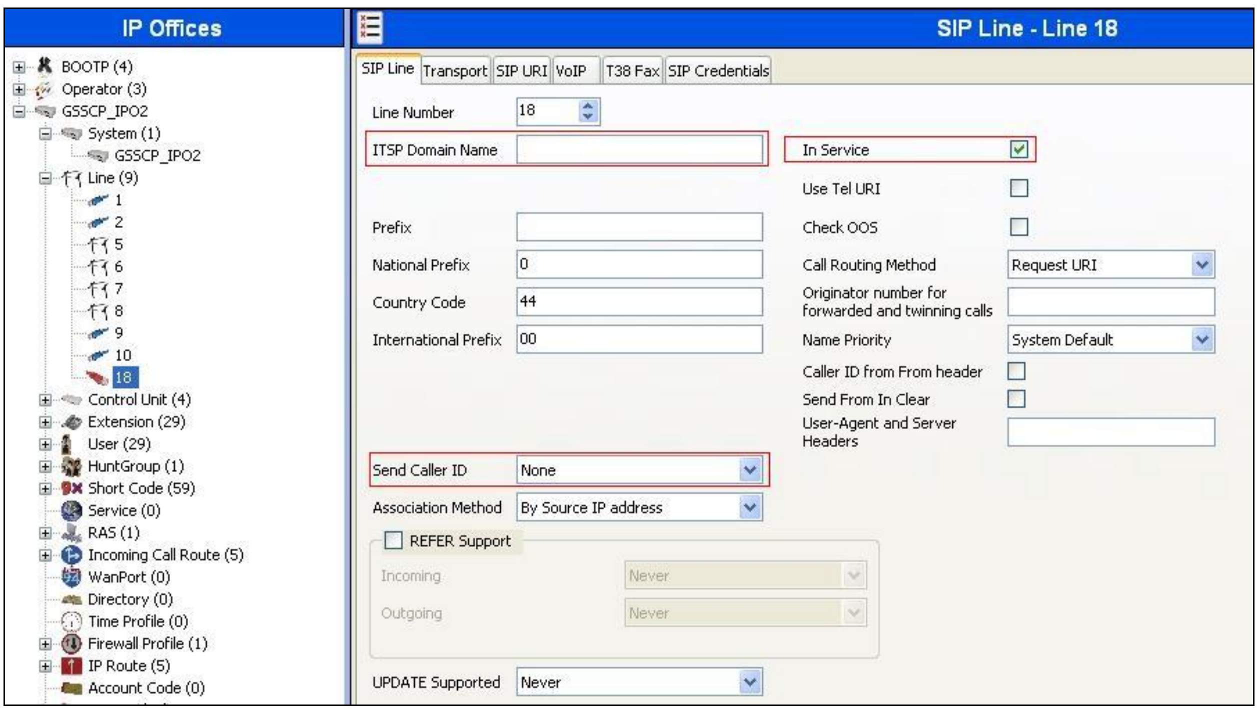


5.6. Administer SIP Line

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

- **ITSP Domain Name** field should remain blank as Colt VoIP Access have not provided a Domain Name
- **Set Send Caller ID** to **None** as it is only required if the box labeled **Send original calling party information for Mobile Twinning** is unchecked in **Section 5.4**
- Ensure the **In Service** box is checked
- Default values may be used for all other parameters

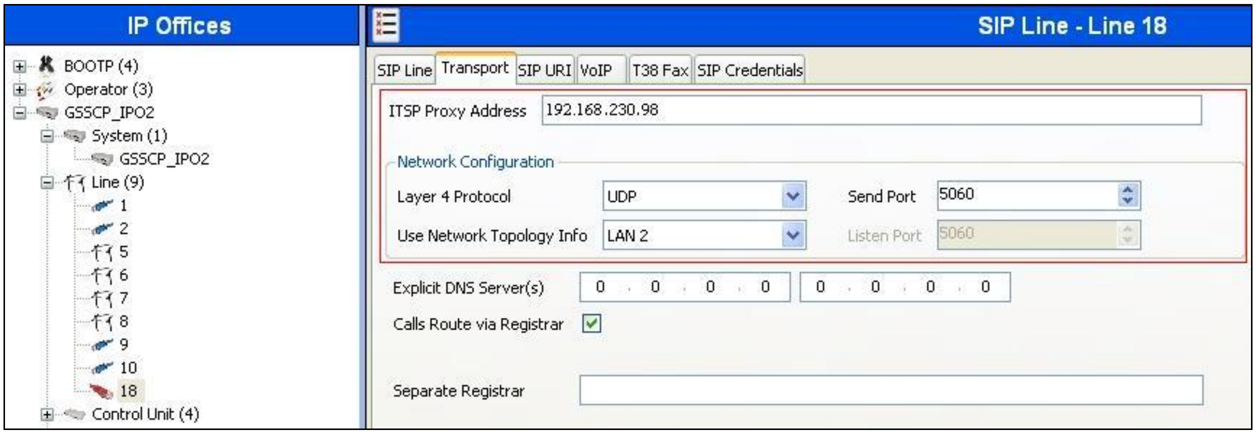
On completion, click the **OK** button (not shown).



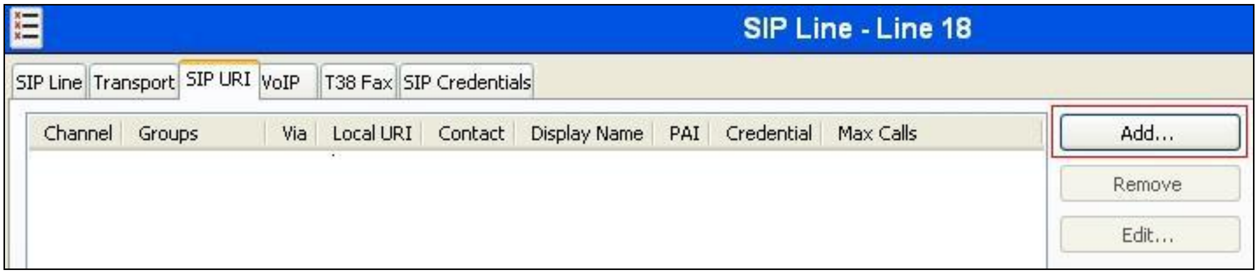
Select the **Transport** tab and set the following:

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

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



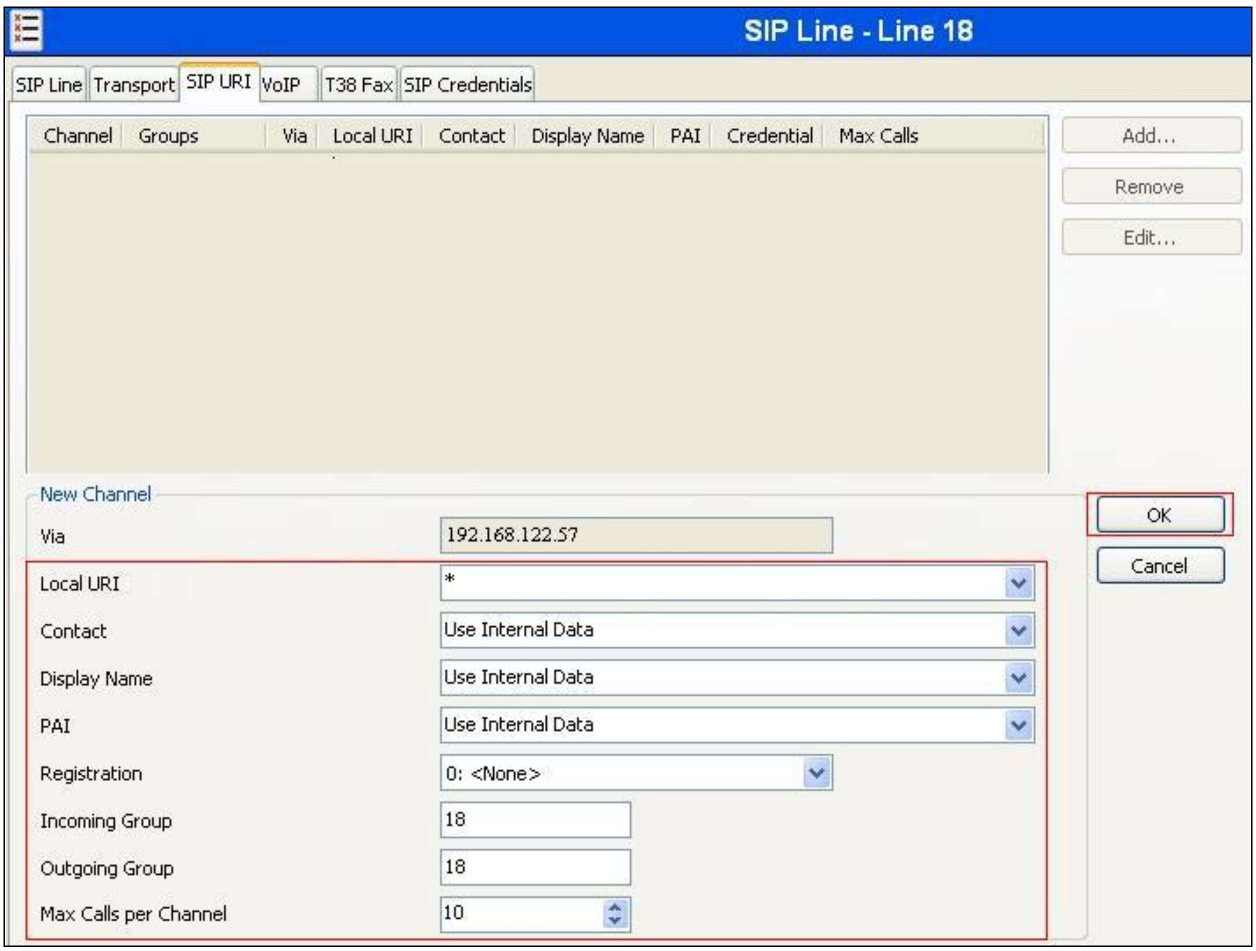
After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



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.

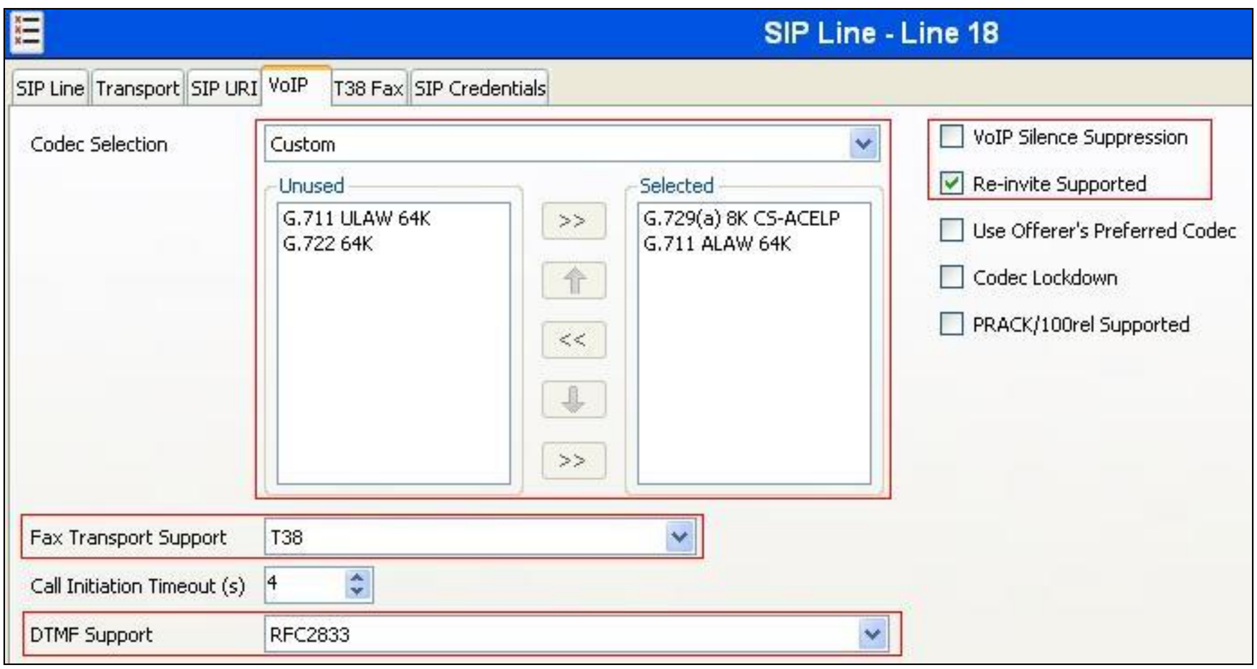
- Set **Local URI** to *****, This setting allows all calls with numbers defined in Incoming Call Routing as shown in **section 5.9.**
- Set **Contact**, **Display Name** and **PAI** to **Use Internal Data**
- For **Registration**, select **0: <None>** from the pull-down menu since this configuration 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.

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

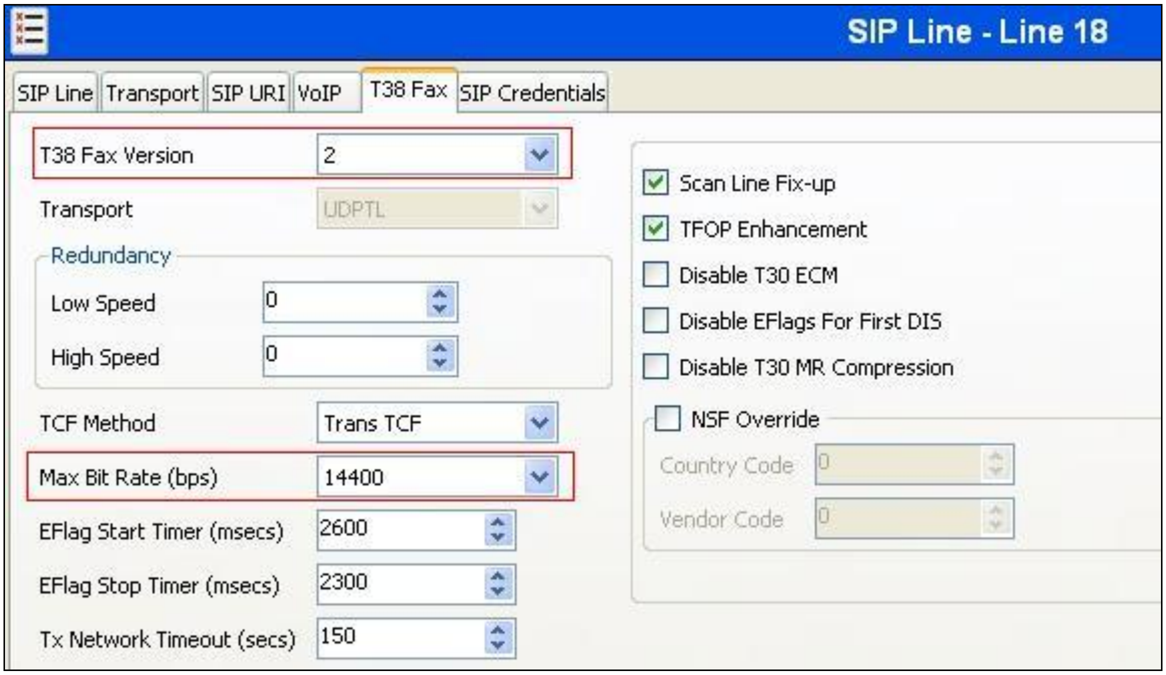


Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- Select **Custom** in the Codec Selection drop down menu to specify the preferred codecs
- 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** followed by **G.711 ALAW 64K**
- Select **T38** in the **Fax Transport Support** drop down menu to allow T.38 fax operation
- Select **RFC2833** in the **DTMF Support** drop down menu. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833
- Uncheck the **VoIP Silence Suppression** box
- Check the **Re-invite Supported** box, to allow for codec re-negotiation in cases where the target of the incoming call or transfer does not support the codec originally negotiated on the trunk
- Default values may be used for all other parameters
- On completion, click the **OK** button (not shown).



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).



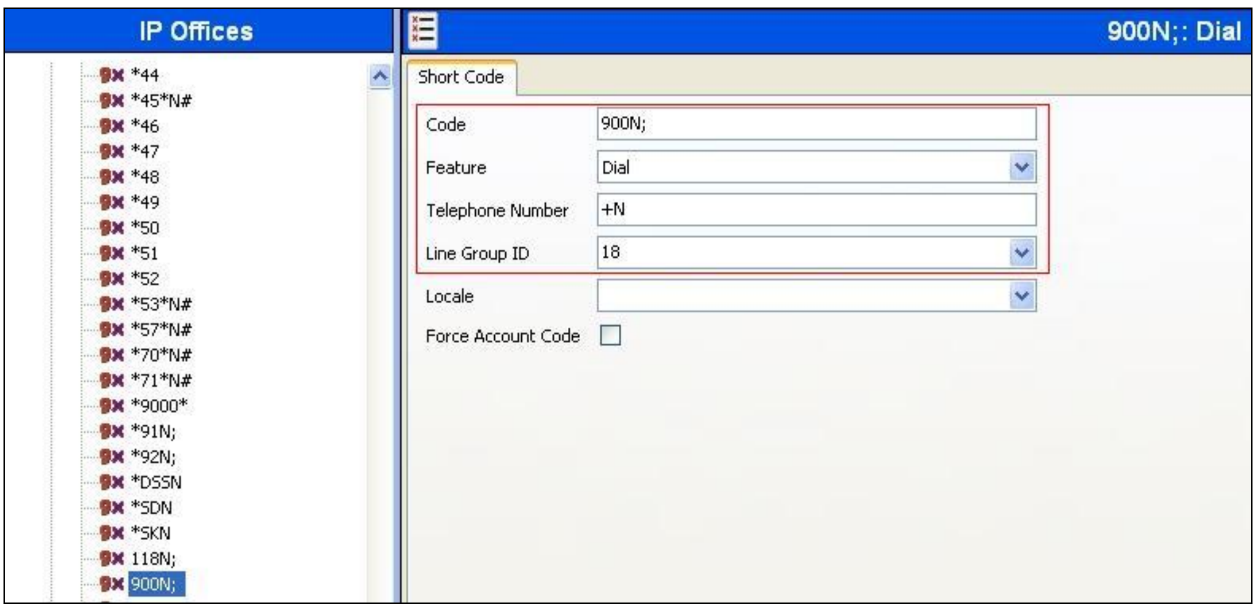
Note: It is advisable at this stage to save the configuration as described in **Section 5.11** to make the Line Group ID defined in **Section 5.6** available.

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.

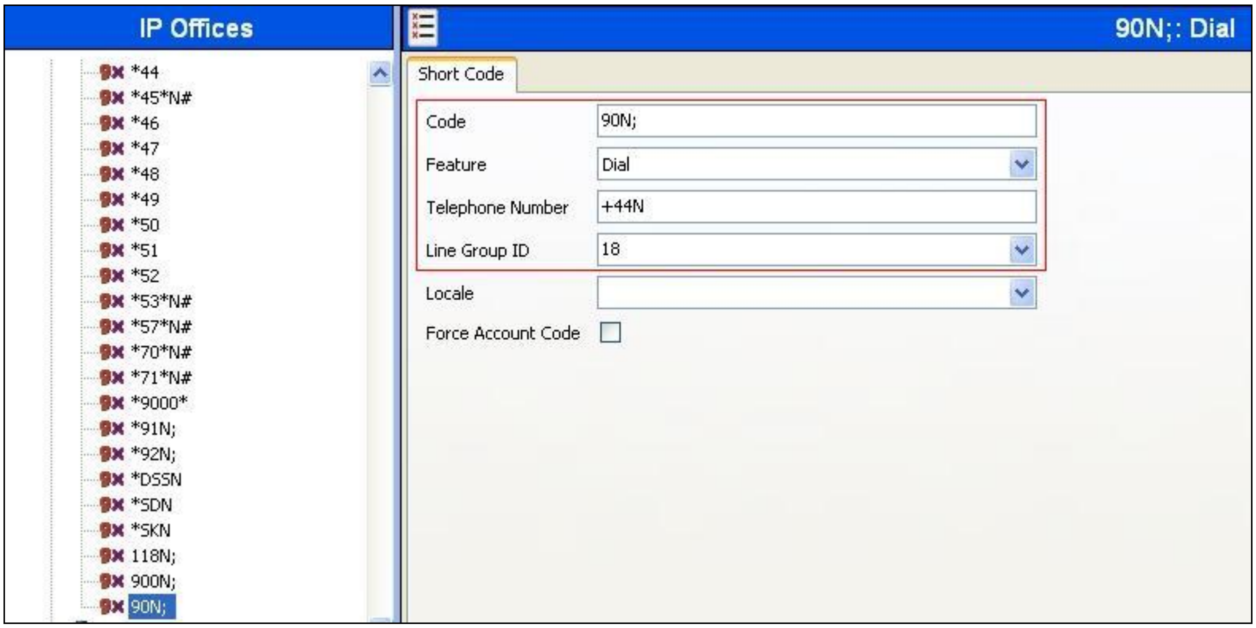
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon
- The example shows **900N**; which will be invoked when the user dials 9 followed by an international number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **+N** which will insert the E.164 number prefixed with a + in 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 5.6**

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



Short codes are also used for routing of national calls and Operator calls. An example for national calls is shown below.

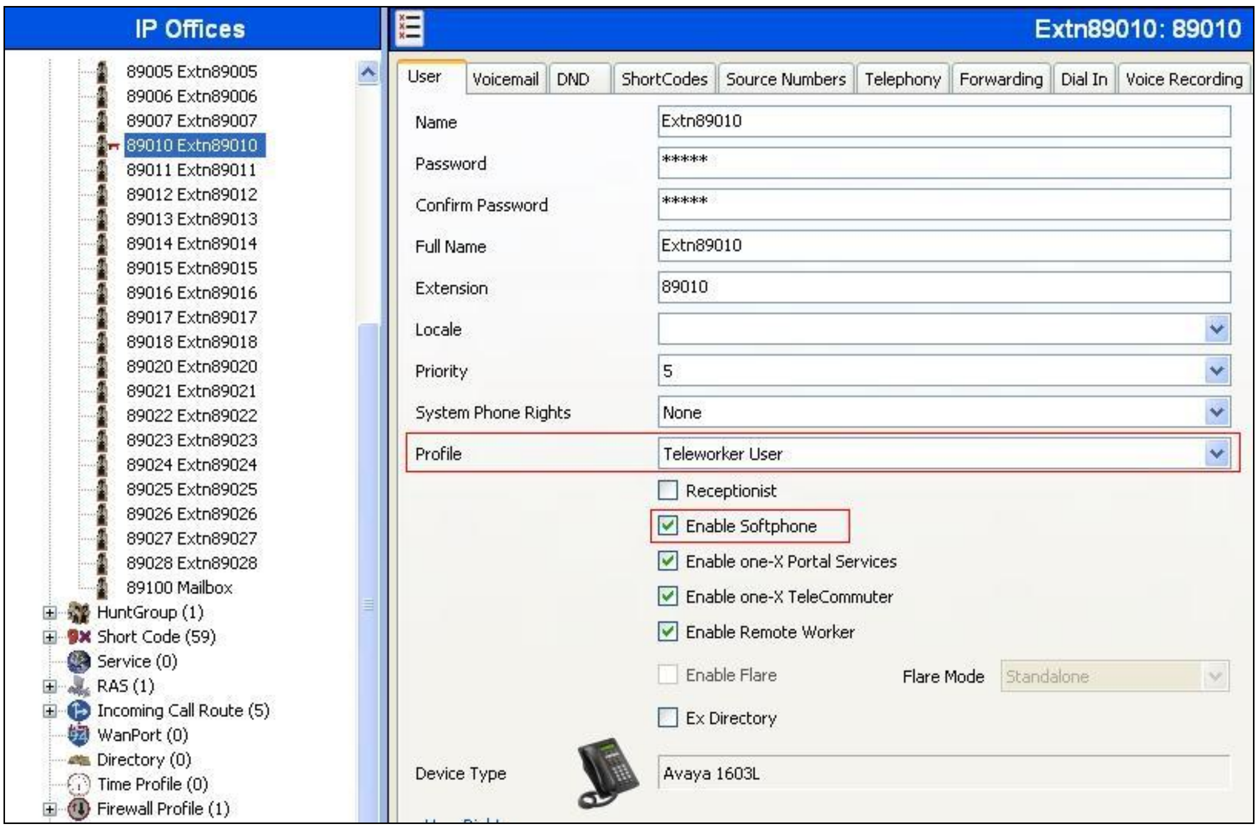
- The example of a national call shows **90N**; which will be invoked when the user dials 9 followed by a national number.
- Set **Telephone Number** to **+44N** which will insert the E.164 number prefixed with a + in the Request URI and To headers in the outgoing SIP INVITE message
- Set other parameters as shown in the previous example.



5.8. User

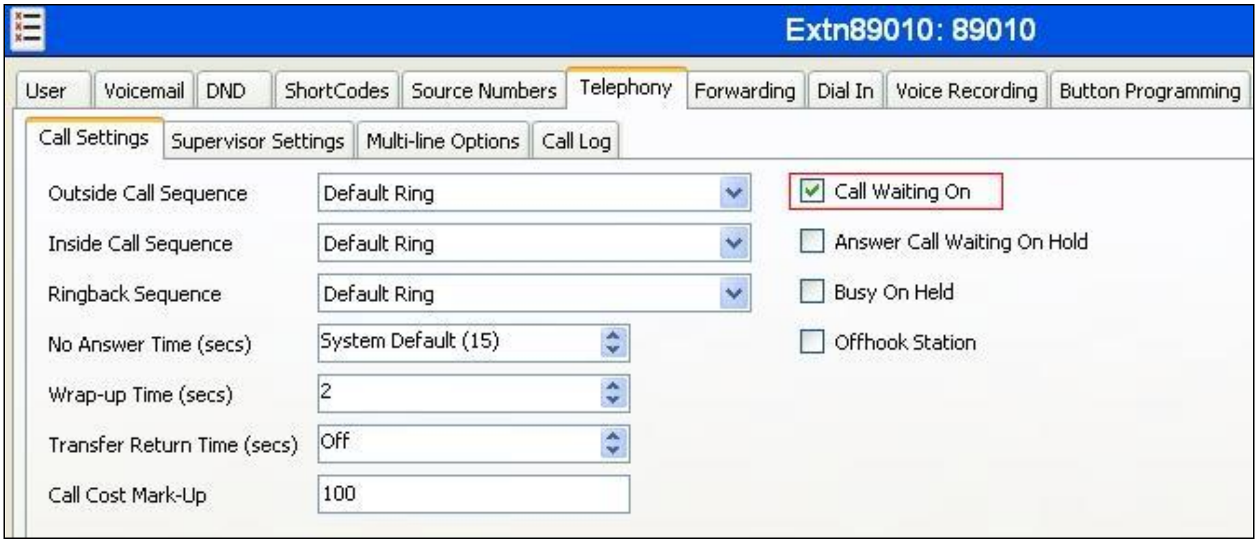
Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.6**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required. The example below shows the changes required to use IP Office Softphone which was used in test. Softphone replaced Phone Manager at IP Office 8.0.

- Change the **Name** of the User if required, this will be used for login to the IP Office Softphone
- Select **Teleworker** User from the Profile drop down menu
- Check the **Enable Softphone** box



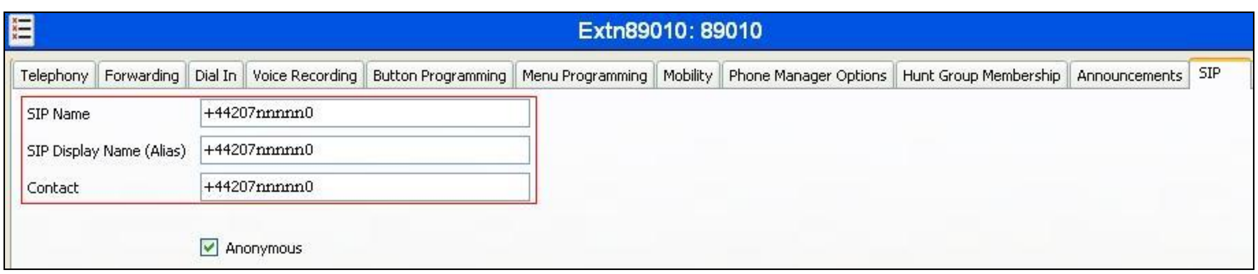
IP Office Softphone uses SIP for signalling and hence required setting of the **SIP Registrar Enable** as described in **Section 5.2**. Call forwarding and transfer make use of the SIP REFER message. To handle SIP REFER on IP Office, the Call waiting function is used.

To turn on Call Waiting, navigate to **Telephony→Call Settings**. Check the **Call Waiting On** box.



Next Select the **SIP** (not shown) 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 Colt.

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).

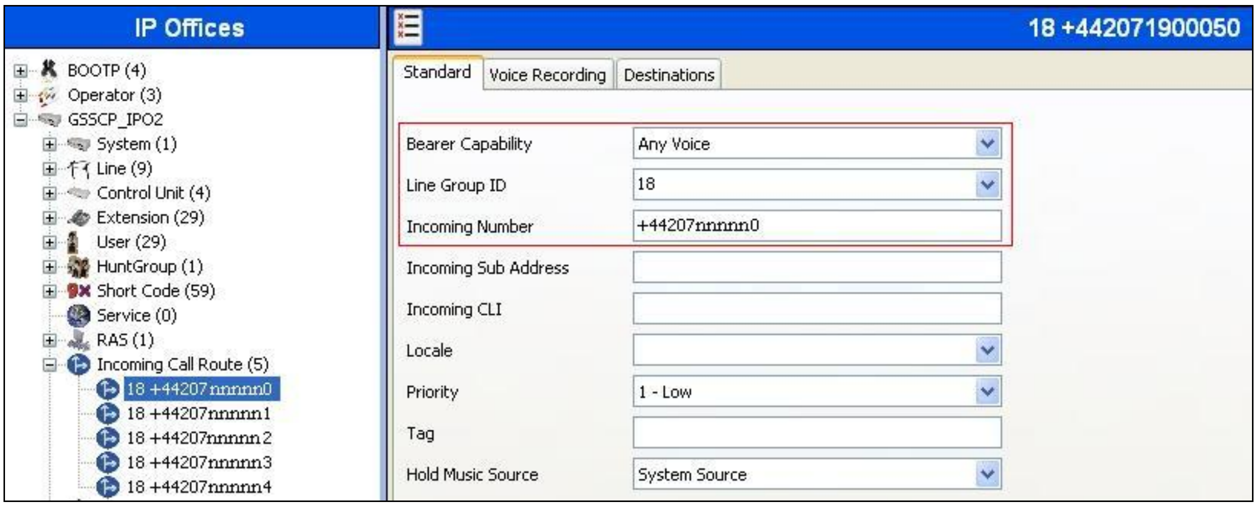


Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly. Also note that the **Anonymous** box is checked. This was done as part of the Calling Party Number presentation test and is not normally checked.

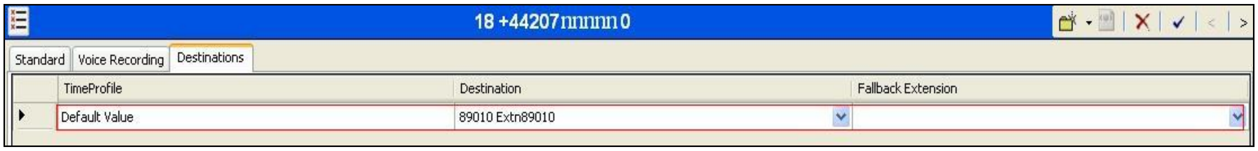
5.9. Incoming Call Routing

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 Route**s 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



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 89010.



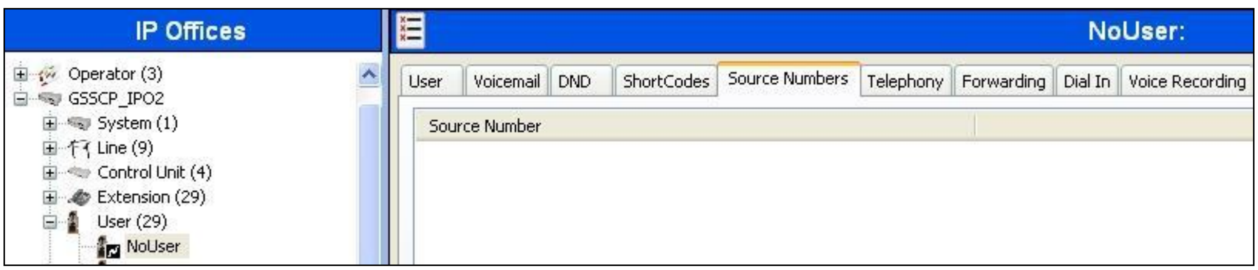
5.10. SIP Options

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

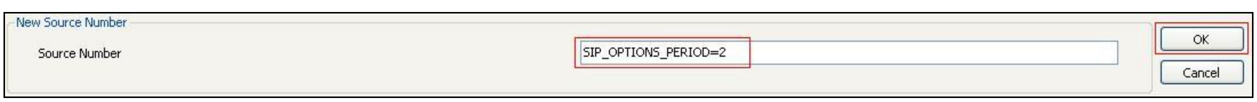
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to the value required
- To establish a period greater than 42 seconds, a **SIP_OPTIONS_PERIOD** parameter must be set to the value required

Note: The OPTIONS message period will be the smaller of the **Binding Refresh Time** and the **SIP_OPTIONS_PERIOD**.

To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button (Not shown)



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



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

5.11. 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. Colt VoIP Access Configuration

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

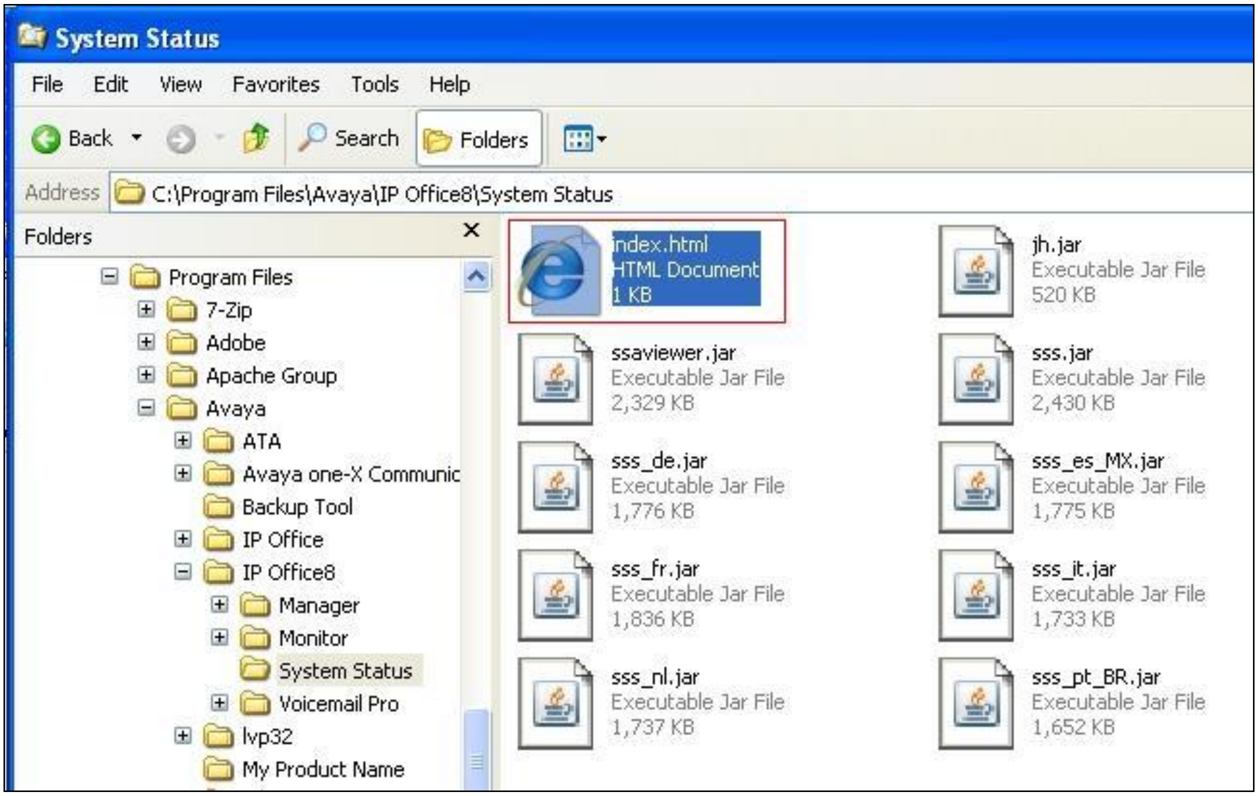
- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

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

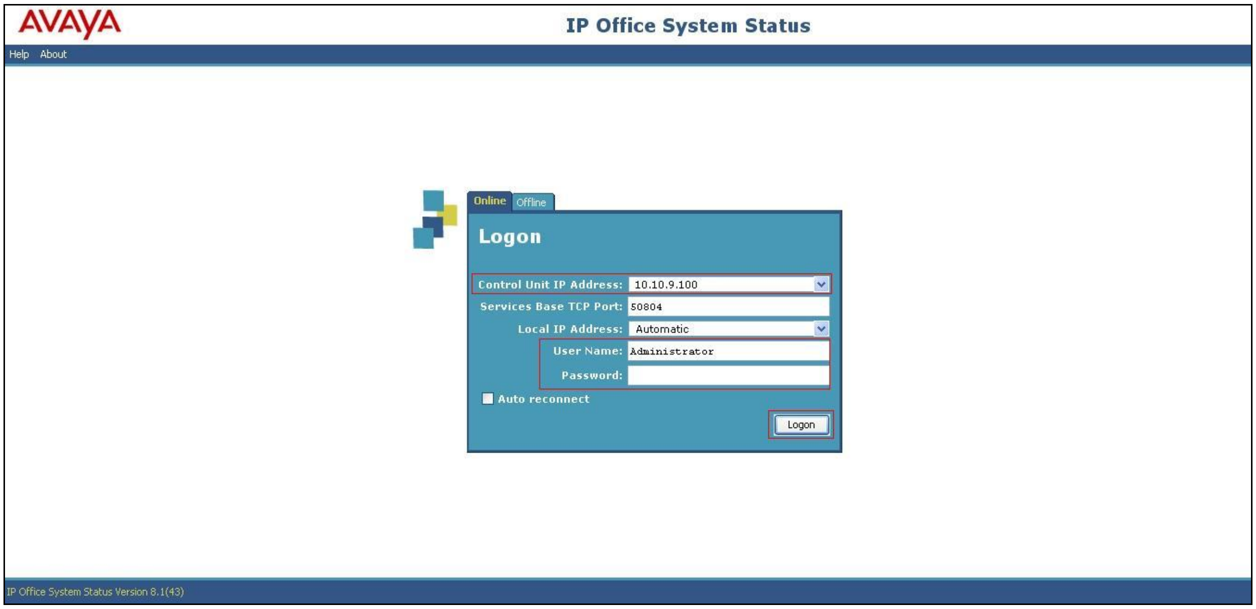
7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This opens in a browser and can be found in the IP Office folder at **Program Files → Avaya → IP Office → System Status**. Click on **Index.html**.

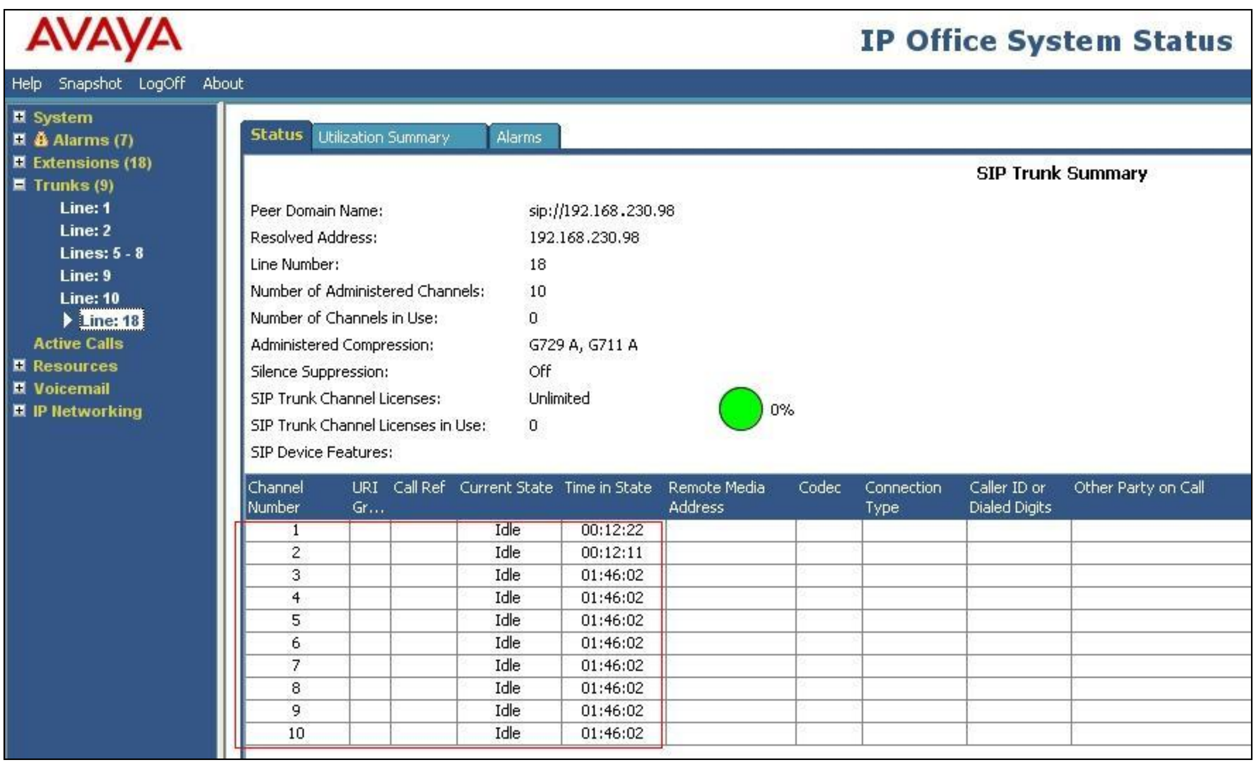


Note: in the example shown the **System Status** folder is in **IP Office8**. This is because of a previous installation; normally this would be in **IP Office.**

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.



8. Conclusion

The Colt VoIP Access service passed compliance testing. Interoperability testing of the sample configuration was completed with successful results for Colt VoIP Access. Refer to **Section 2.2** for test observations.

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] Avaya IP Office Installation, Document number15-601042, 14th August 2012.
[3] Avaya IP Office Manager, Document number15-601011, 3rd August 2012.
[4] System Status Application, Document number15-601758, 12th November 2011
[5] IP Office Softphone Installation, 28th September 2011
[6] IP Office SIP Extension Installation, 3rd October 2011

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.