Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 9.0 with TDC Business Trunk – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between TDC Business Trunk and Avaya IP Office.

The TDC Business Trunk provides PSTN access via a SIP trunk connected to the TDC Voice Over Internet Protocol (VoIP) network as an alternative to legacy analogue or digital trunks. TDC 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.

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between TDC Business Trunk and Avaya IP Office. TDC Business Trunk provides PSTN access via a SIP trunk connected to the TDC 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 TDC Business Trunk. 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 TDC Business Trunk. 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 TDC.
- Outgoing PSTN calls from various phone types including H.323, SIP, digital, and analogue telephones at the enterprise. Calls were routed from the enterprise across the SIP trunk to TDC.
- Inbound and outbound PSTN calls to/from an IP Office Softphone client and Avaya Flare® Experience for Windows.
- Various call types including: local, international, toll free (outbound) and directory assistance.
- Codecs G.711A, G.711MU and G.729A.
- 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.
- Mobile Extension (MEX) incoming and outgoing calls.

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2.2. Test Results

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

- When IP Office sends OPTIONS to TDC Business Trunk, the network responds with 404 Not Found. This is accepted by IP Office as a valid response showing the SIP trunk is still in service.
- No Inbound Toll Free access was available for test.
- When the CLI is blocked on IP Office, the number must still be sent in the "From" header. This is achieved by selecting "Send From in Clear" on the SIP Line. The network will not display the number as the "Privacy" header is set to "id".
- When a call is forwarded off-net, the DDI number of the extension is required in the Diversion Header. This is achieved by selecting "Diversion Header" in the "Send Caller ID" field on the SIP Line.
- Initial tests of incoming T.38 fax calls failed. When the version was changed on IP Office from 3 to 2, calls were successful.
- Calls to an extension with Mobile Extension (MEX) with the CLI withheld failed to be routed to the MEX mobile. This was because the From header contained "Anonymous" instead of a valid DDI number. A change was made in the TDC network to resolve this issue.
- Calls from one extension to another with MEX active are routed to the MEX mobile with the full DDI number in the From header where the extension number would be preferred. This is not considered to be a major issue.
- When an outbound call from a MEX mobile is completed, the caller hears dial tone for 30 seconds. Normally when the Mobile Twinning function of IP Office is used, multiple calls can be made from the twinned mobile and the caller hears dial tone after a call is complete so he or she can make an additional call. In this case however, no additional call could be made. This issue is under investigation.

2.3. Support

For technical support on TDC products please contact the following website:
http://www.tdc.se

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3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to TDC Business Trunk. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include an Avaya 1600 Series IP Telephone (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), an Avaya 1140e SIP Telephone, an Avaya 2420 Digital Telephone, an Avaya Analogue Telephone and a fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as an IP Office Softphone client and Avaya Flare® Experience for Windows for mobility testing. For security purposes, 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.

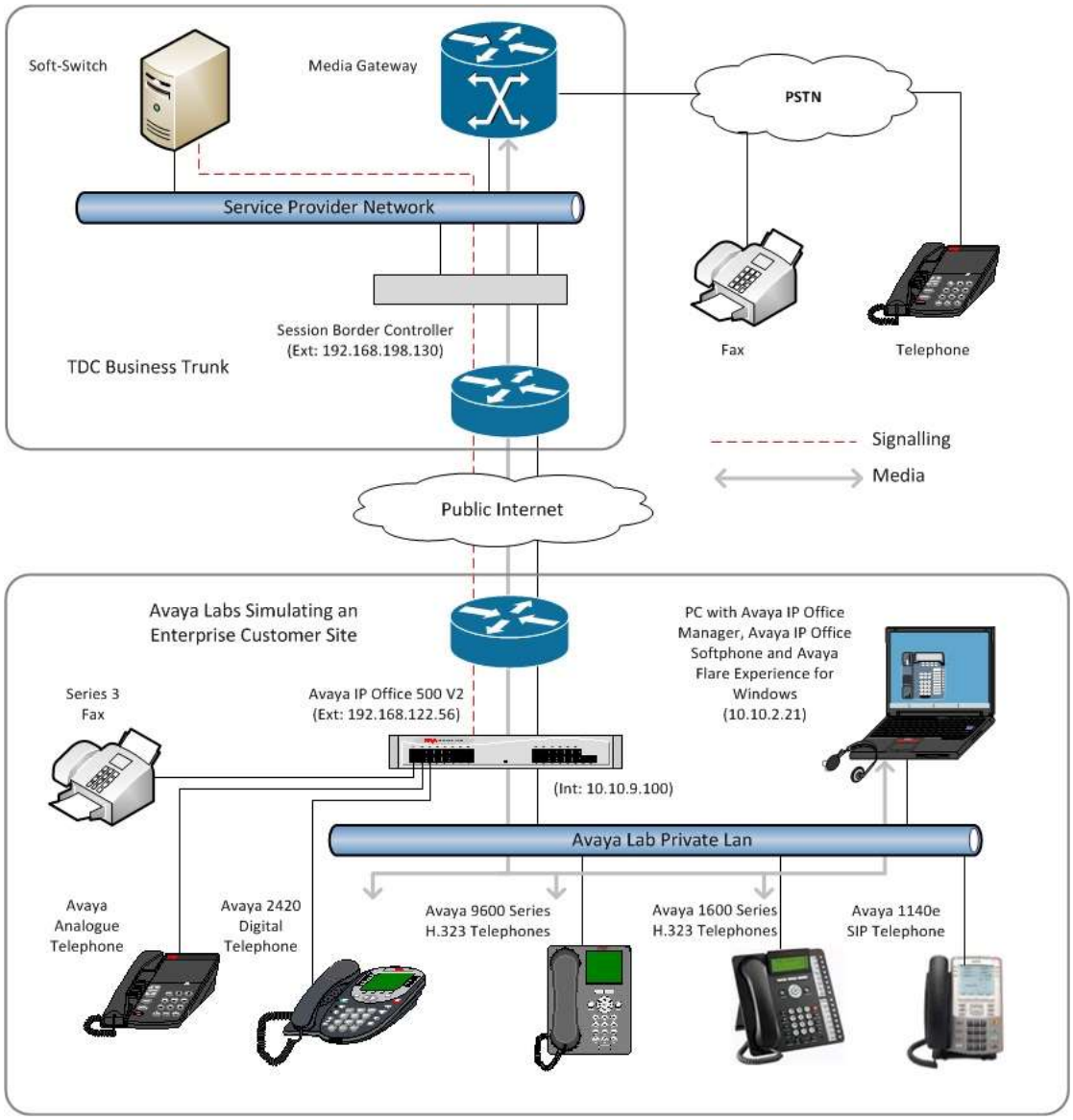

Figure 1: TDC Business Trunk to Avaya IP Office Topology

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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 R9.0.500.972
Avaya 1140e IP SIP Telephone	04.04.10.00
Avaya 1608 IP Phone (H.323)	1.350B
Avaya 9608 IP Phone (H.323)	6.3.1.16
Avaya 2420 Digital Phone	N/A
Avaya 98390 Analogue Phone	N/A
Avaya Softphone	3.2.3.49
Avaya Flare® Experience for Windows	1.1.3.14
Avaya IP Office Manager	Version 9.0.5.0 build 972
TDC	
Acme Packet SD3820	6.3
Ericsson IMS	N/A
Broadsoft Broadworks	r17sp4
Cisco PGW2200	N/A

Testing was performed with IP Office 500 V2 R9.0.1, but this testing also applies to IP Office Server Edition 9.0.1. Note that IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

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5. Configure Avaya IP Office

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

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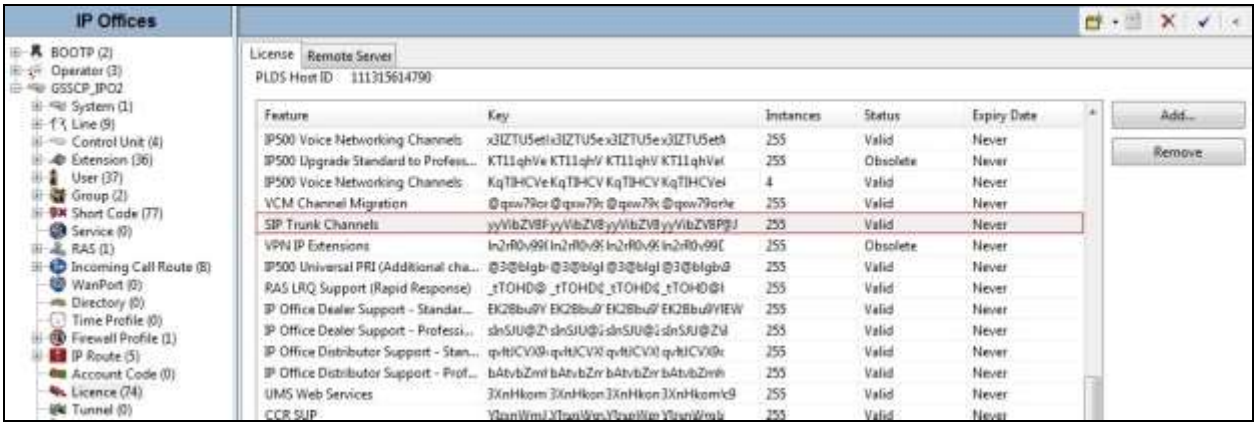
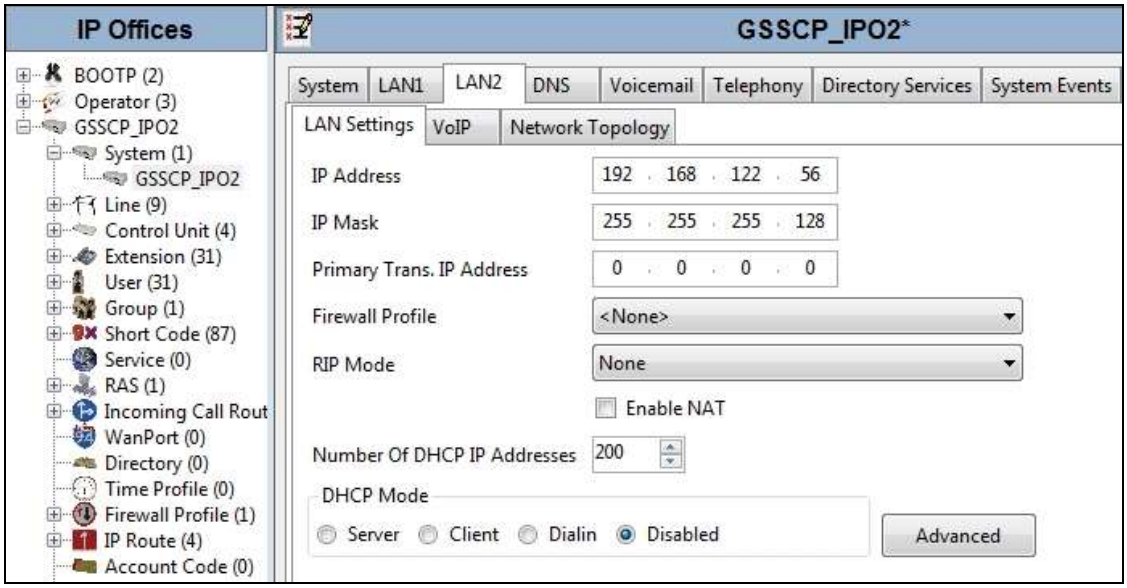
5.1. Verify System Capacity

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 TDC.



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 → <IP Office Name>** in the Navigation Pane where <IP Office Name> is the name of the IP Office. This is **GSSCP_IPO2** in the GSSCP test environment. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the IP Office; **Primary Trans. IP Address** is the next hop, usually the default gateway address. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



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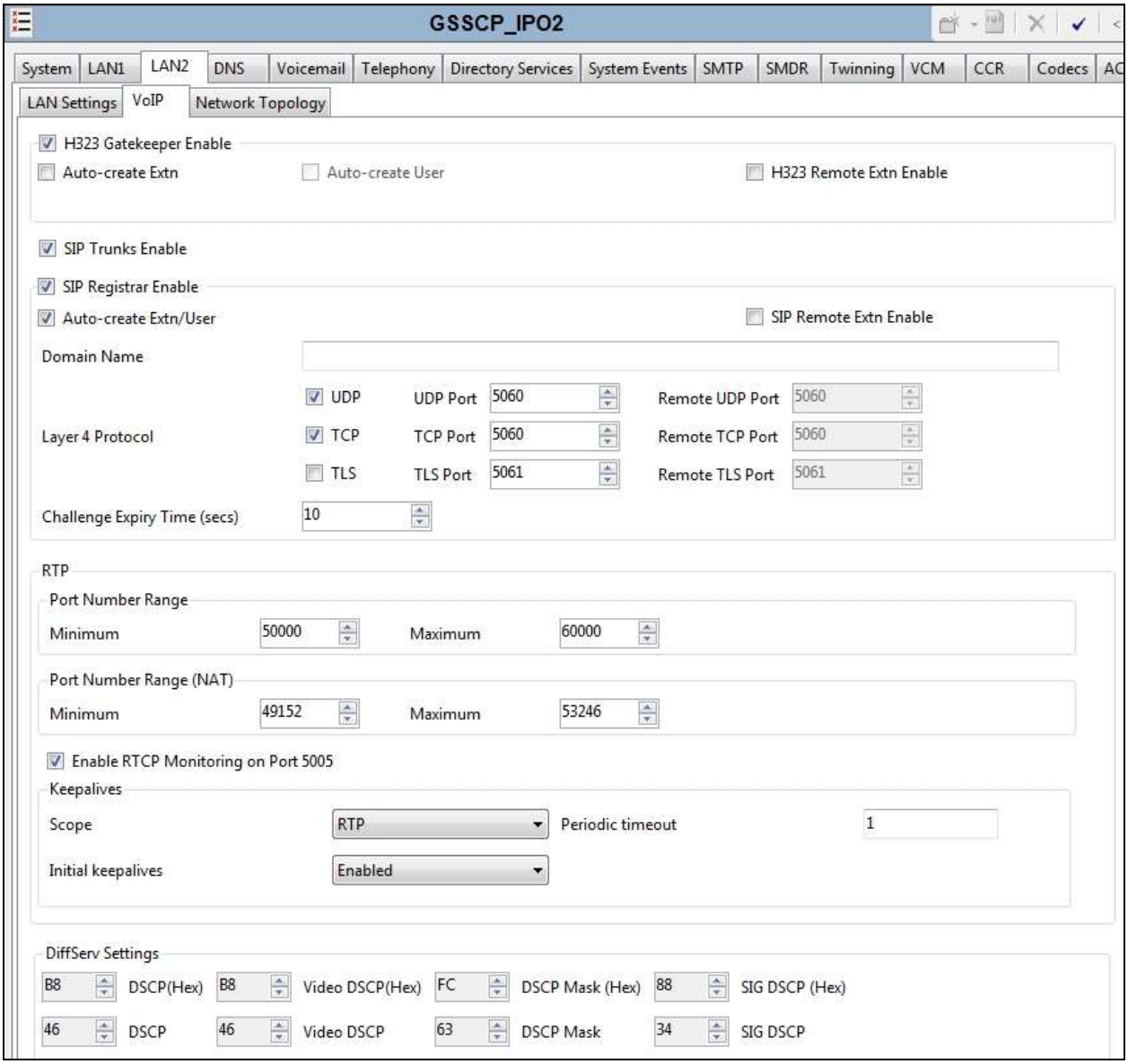
On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If SIP Endpoints are to be used such as the IP Office Softphone, Avaya Flare® Experience for Windows and the Avaya 1140e, the **SIP Registrar Enable** box must also be checked. Define 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 **5000** to **6000** as specified in the Business Trunk Technical Specification.



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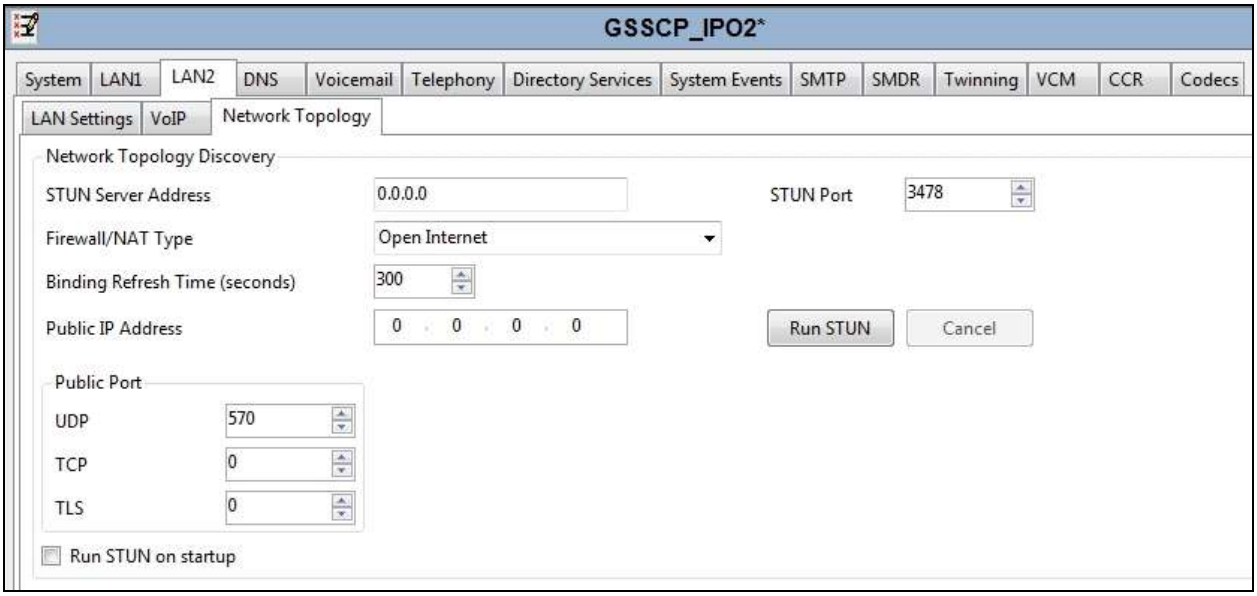
Note: Avaya IP Office can also be configured to mark the Differentiated Services Code Point (DSCP) in the IP Header with specific values to support Quality of Services policies for both signalling and media. DSCP for media can be set for both voice and video. The **DSCP** field is the value used for 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 enter the information required if Network Address Translation (NAT) is to be used. During test, NAT was not required and there was no requirement for a STUN server. To disable this facility, 0.0.0.0 is entered in the **STUN Server IP Address** and **Public IP Address** fields.

The Network Topology tab can also 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**.



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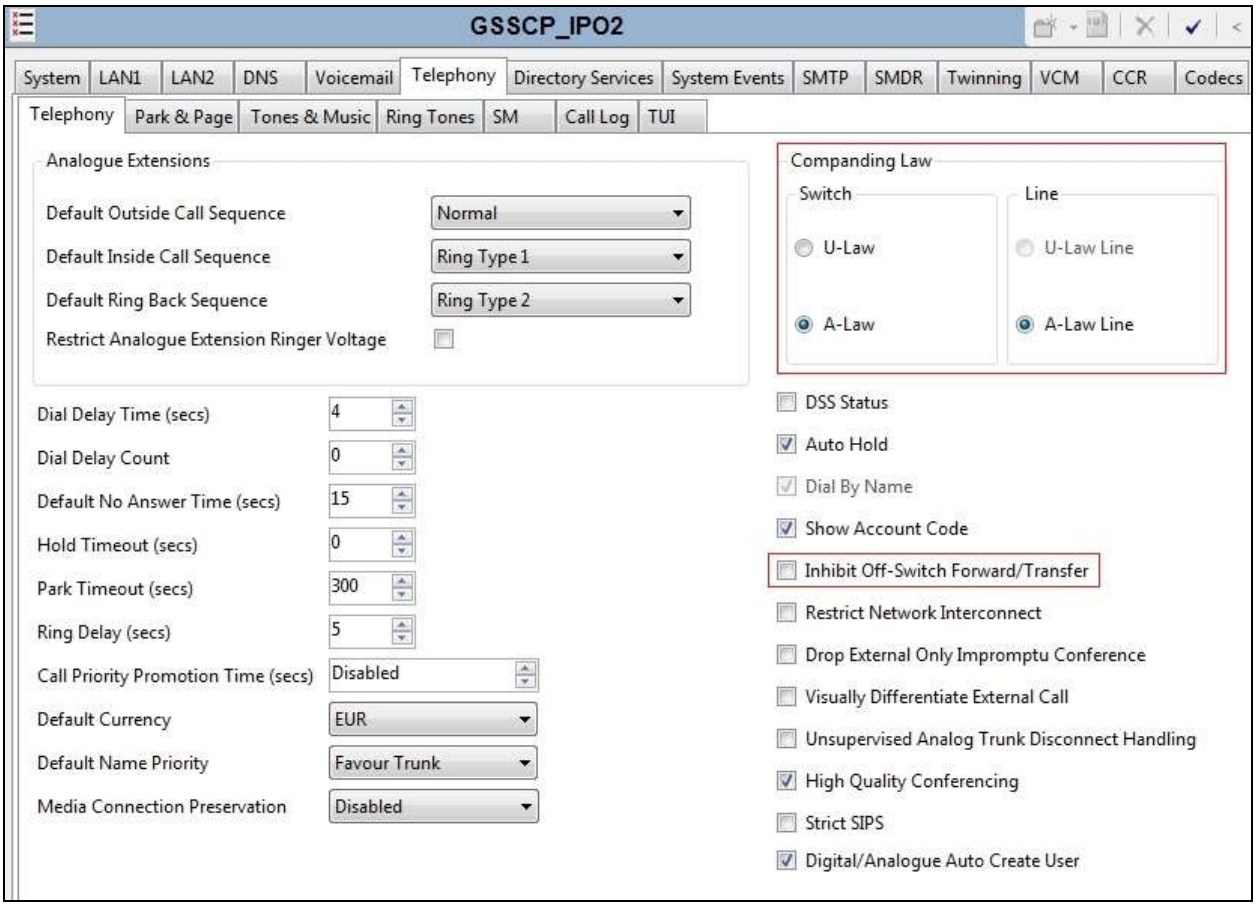
5.3. System Telephony Settings

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).



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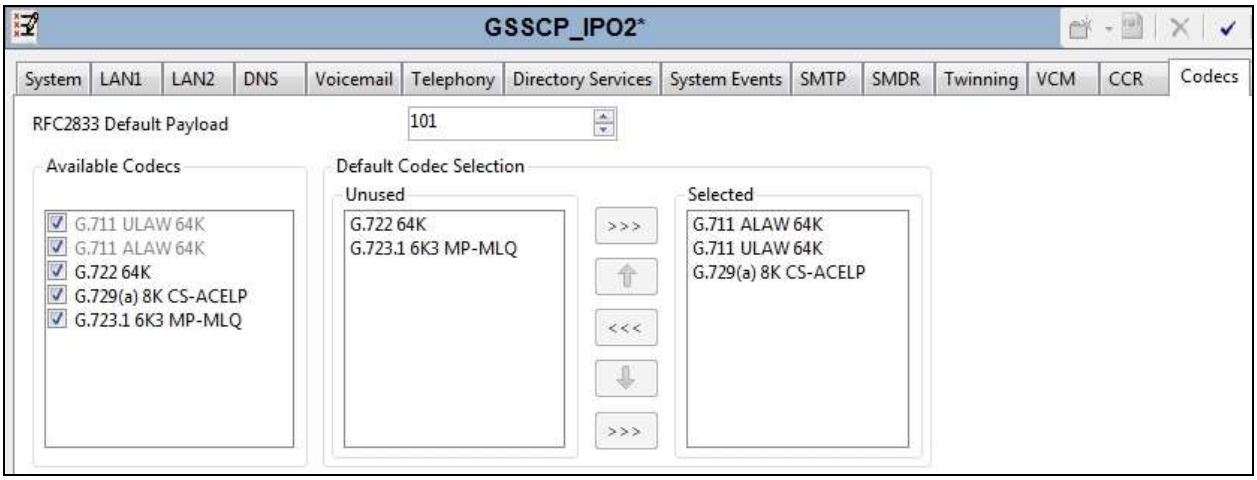
5.4. System Twinning Settings

Navigate to the **Twinning** tab (not shown) and ensure that the box labeled **Send original calling party information for Mobile Twinning** is unchecked and that the **Calling party information for Mobile Twinning** field is blank. With this setting, Avaya IP Office will send the original calling party number to the twinned phone in the SIP From header and will use SIP Line settings for sending the calling party number of the host phone. The SIP Line settings for sending caller ID are described in **Section 5.6.2**. 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 used as default codecs. The order of priority can be changed using the vertical arrows. On completion, click the **OK** button (not shown).



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5.6. Administer SIP Line

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

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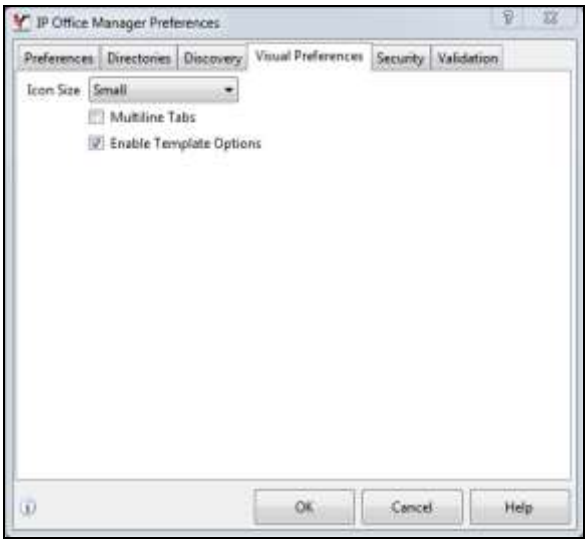
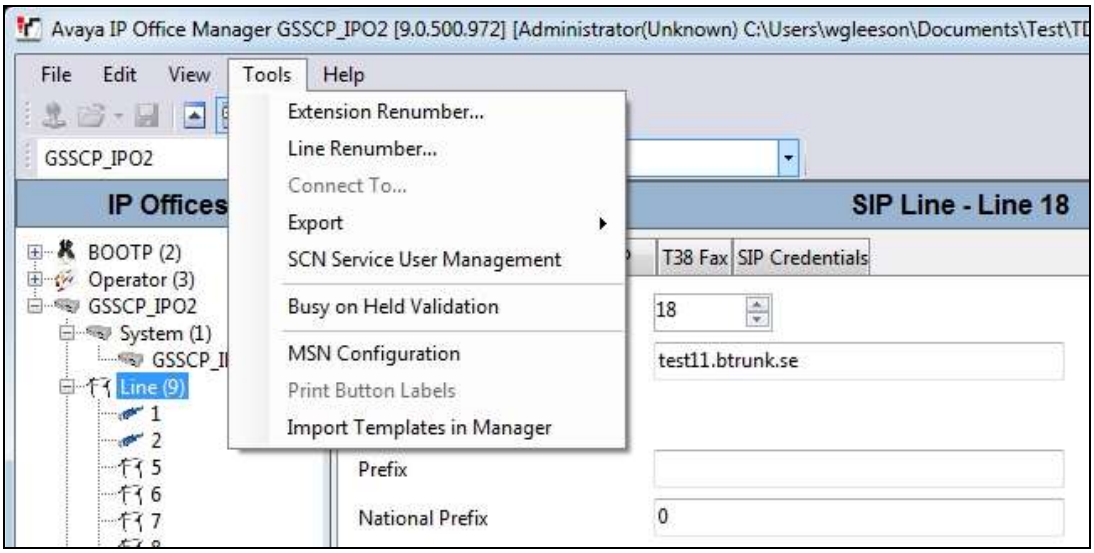
5.6.1. SIP Line From Template

Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **SE_TDC_SIPTrunk.xml**. The file name is important in locating the proper template file.

Verify that template options are enabled in IP Office Manager. Navigate to **File → Preferences**. 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**.



Import the template into 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. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



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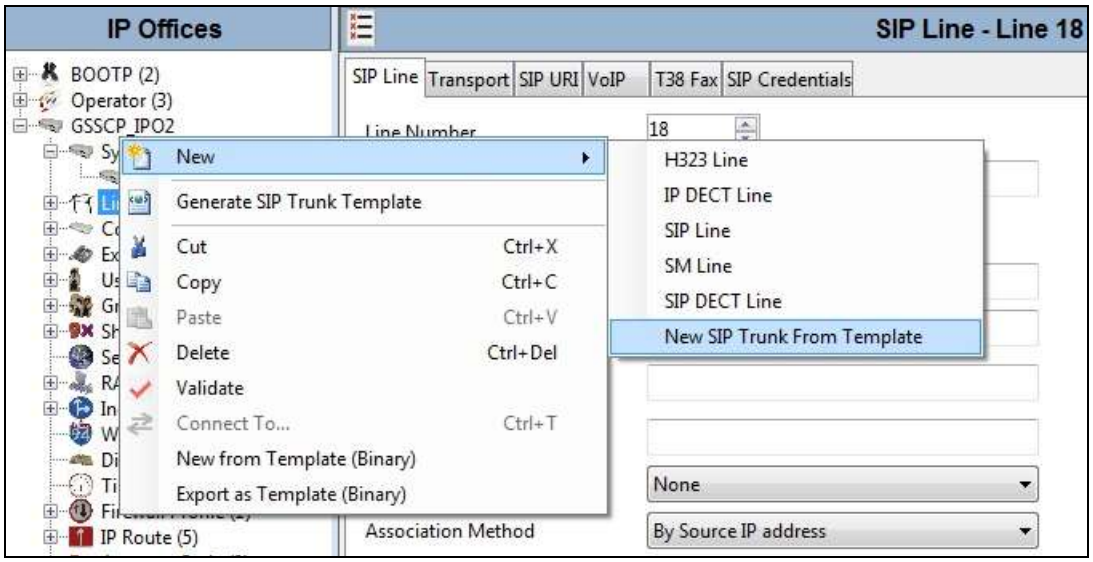
In the pop-up window (not shown) that appears, select the directory where the template file was copied. After the import is complete, a final import status pop-up window (not shown) will appear stating success or failure. Click **OK** (not shown) to continue. If preferred, this step may be skipped if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

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**.



In the subsequent Template Type Selection pop-up window, select **Sweden** from the **Country** pull-down menu and select **TDC** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**SE_TDC_SIPTrunk.xml**) created earlier. Click **Create new SIP Trunk** to finish creating the trunk.

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

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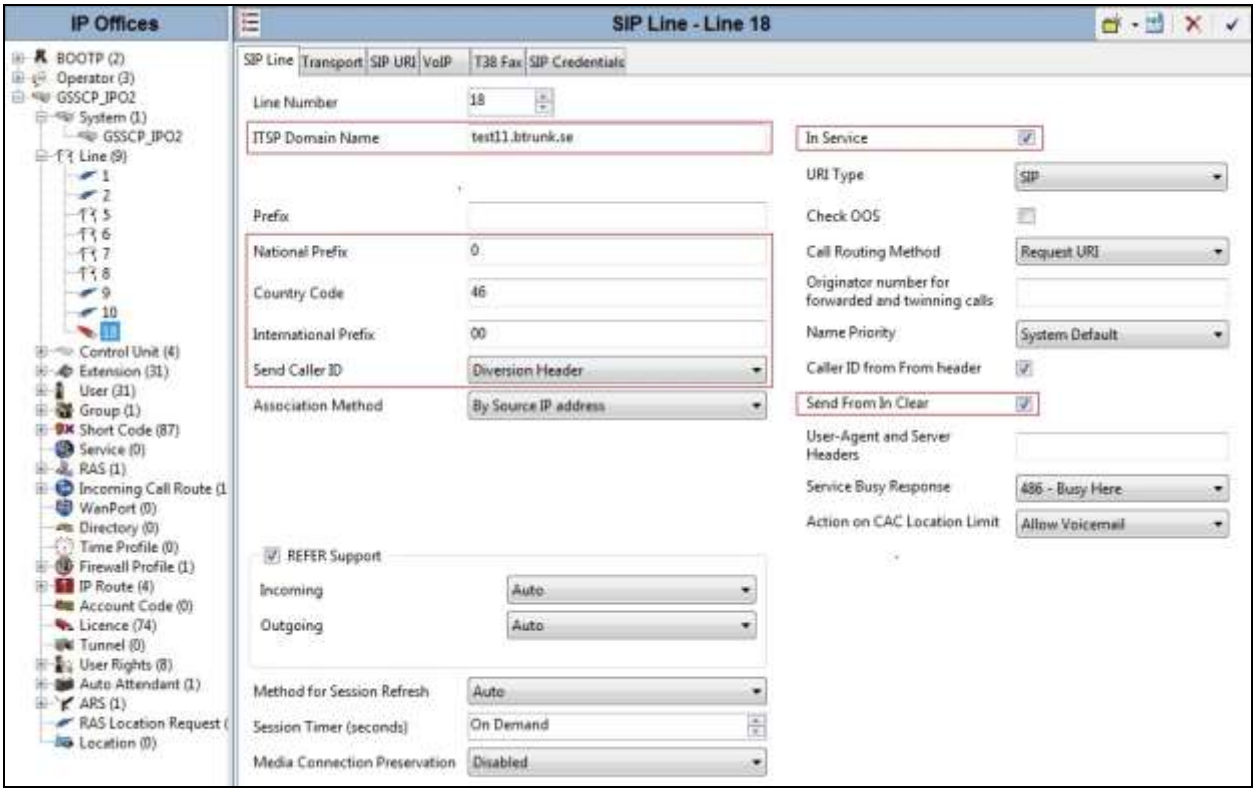
5.6.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** field to the domain name used by TDC. In test this was **test11.btrunk.se**.
- Define **National Prefix**, **Country Code** and **International Prefix** appropriately for Sweden. This displays the calling party number in diallable format on the IP Office phones. It also allows conversion back to E.164 format with leading + for inbound MEX calls.
- Set **Send Caller ID** to **Diversion** to send the calling party number of the IP Office phone in the Diversion header for forwarded and Mobile Twinning calls.
- Ensure the **In Service** box is checked.
- Check the **Send From In Clear** box so that the calling party number is inserted in the From header for outgoing calls with CLI restricted.
- Default values may be used for all other parameters.

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



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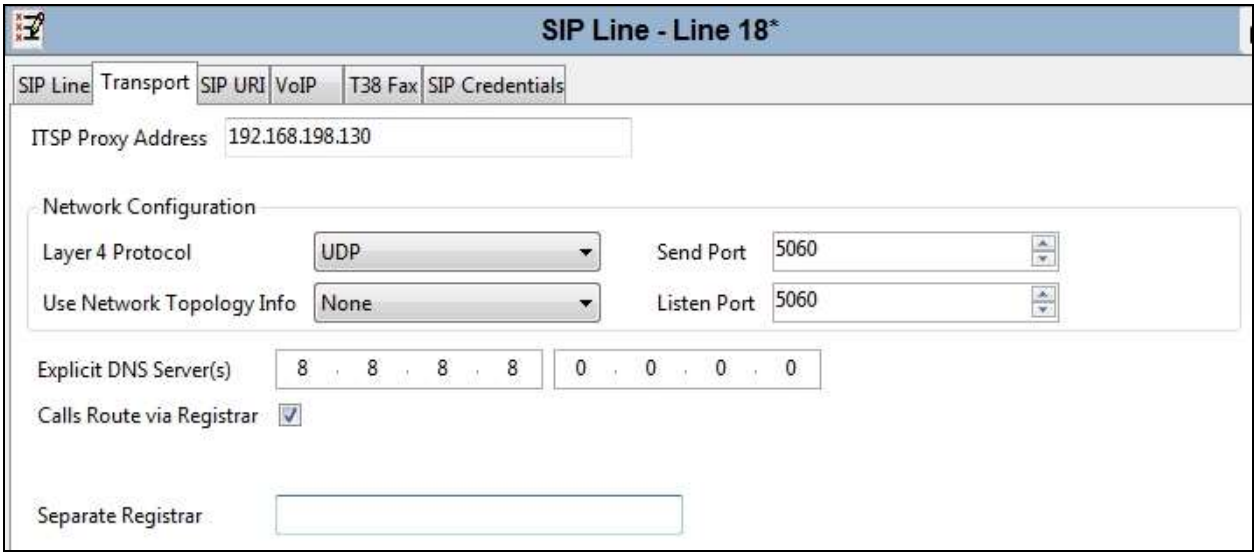
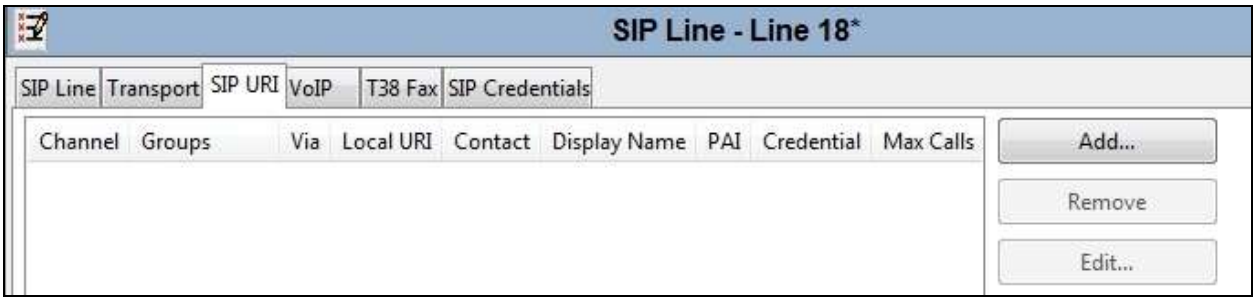
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Addres**s to the external IP address of the TDC SBC.
- Set **Use Network Topology Info** to **None** if NAT is not to be used and the Network Topology settings defined in **Section 5.2** are not required. This was the case during test.
- 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, two SIP URI entries were created. One specified internal data for the local URI and the other specified a wild card. Both could be used for incoming calls, but only the one with internal data specified could be used for outgoing. This is because when a wildcard is used in the local URI for outgoing calls, the From header is not populated correctly.

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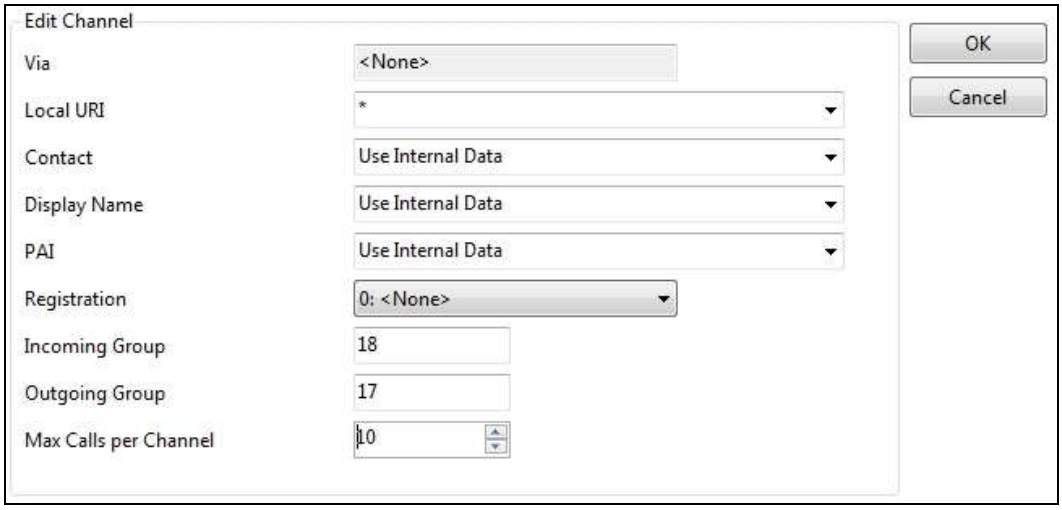
The entry for incoming calls only was created with the parameters shown below.

- Enter * in the **Local URI** field, this allows any call from the specified ITSP Proxy Address to be accepted by IP Office and is required so that incoming MEX calls are not rejected.
- Set **Contact**, **Display Name** and **PAI** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.8**. It is the default setting when no SIP Credentials are specified.
- The **Registration** field is not required as registration is not defined for TDC Business Trunk.
- 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. For the compliance test, a new incoming group **18** was defined that was associated to a single line (line 18). As this SIP URI entry is not to be used for outgoing calls, a different number must be specified in the **Outgoing Group** field. For the compliance test, outgoing group **17** was defined.
- 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.



Note: For outgoing calls, the **Local URI** field is used to populate the From header. The above SIP URI must not be used for outgoing calls as there is a wildcard character specified in this field. In this case, IP Office uses an internal buffer that differs depending on how the call was set up. In some cases, a string of invalid characters is used and the From header is rendered meaningless.

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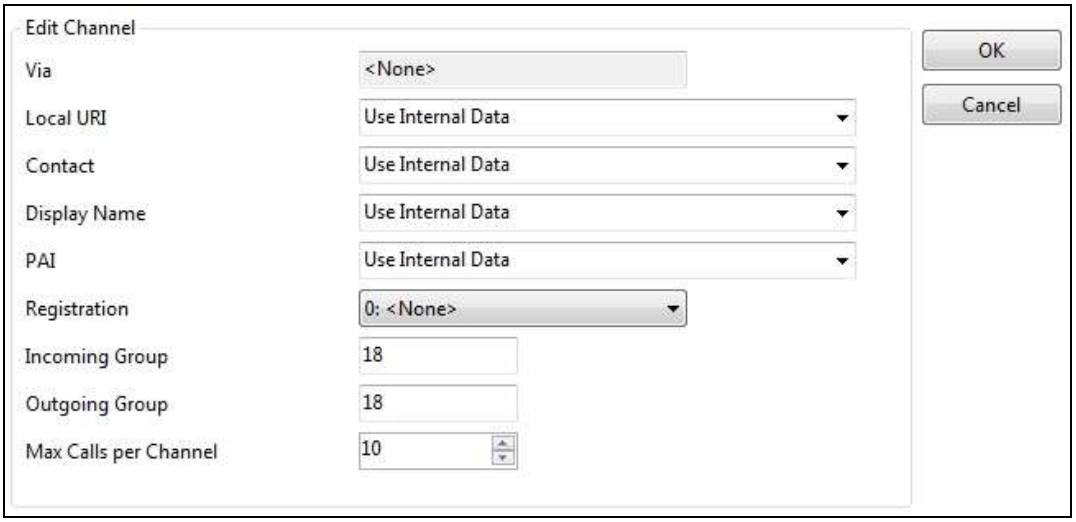
The entry for incoming and outgoing calls was created with the parameters shown below.

- Set **Local URI**, **Contact**, **Display Name** and **PAI** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.8**. It is the default setting when no SIP Credentials are specified.
- The **Registration** field is not required as registration is not defined for TDC Business Trunk.
- 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, incoming and outgoing group **18** was defined.
- 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.



Note: For outgoing calls, the **Local URI** field is used to populate the From header. In the above SIP URI, **Use Internal Data** is specified which ensures that the DDI number applied to the specific extension is used as described above. With this setting, the From header is populated correctly.

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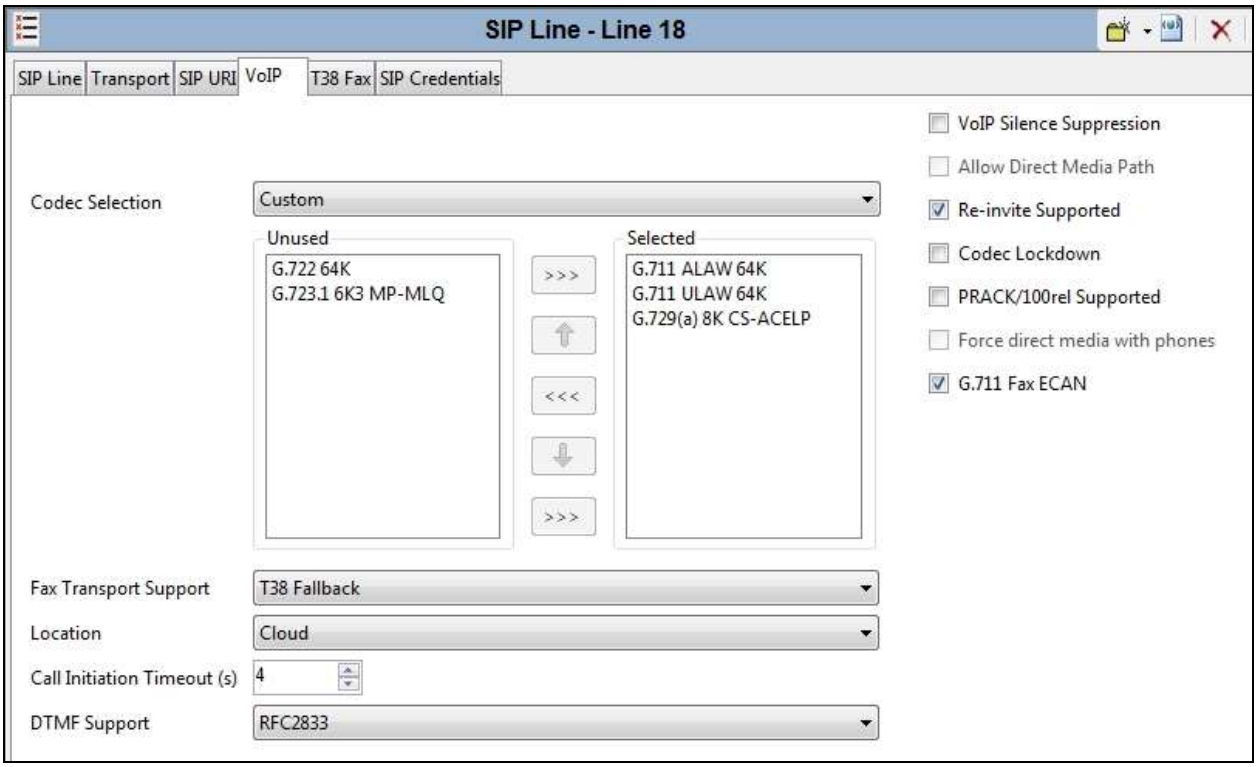
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- 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 TDC this was **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** in priority order from the highest to the lowest. This reflected the codec list received from the network.
- Select **T38 Fallback** in the **Fax Transport Support** drop down menu to allow both T.38 and G.711 fax operation, though only T.38 was supported by TDC at the time of testing.
- 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.
- Uncheck the **PRACK/100rel Supported** box as Reliable provisional responses as described in RFC3262 are not supported in Business Trunk.
- Default values may be used for all other parameters.
- On completion, click the **OK** button (not shown).



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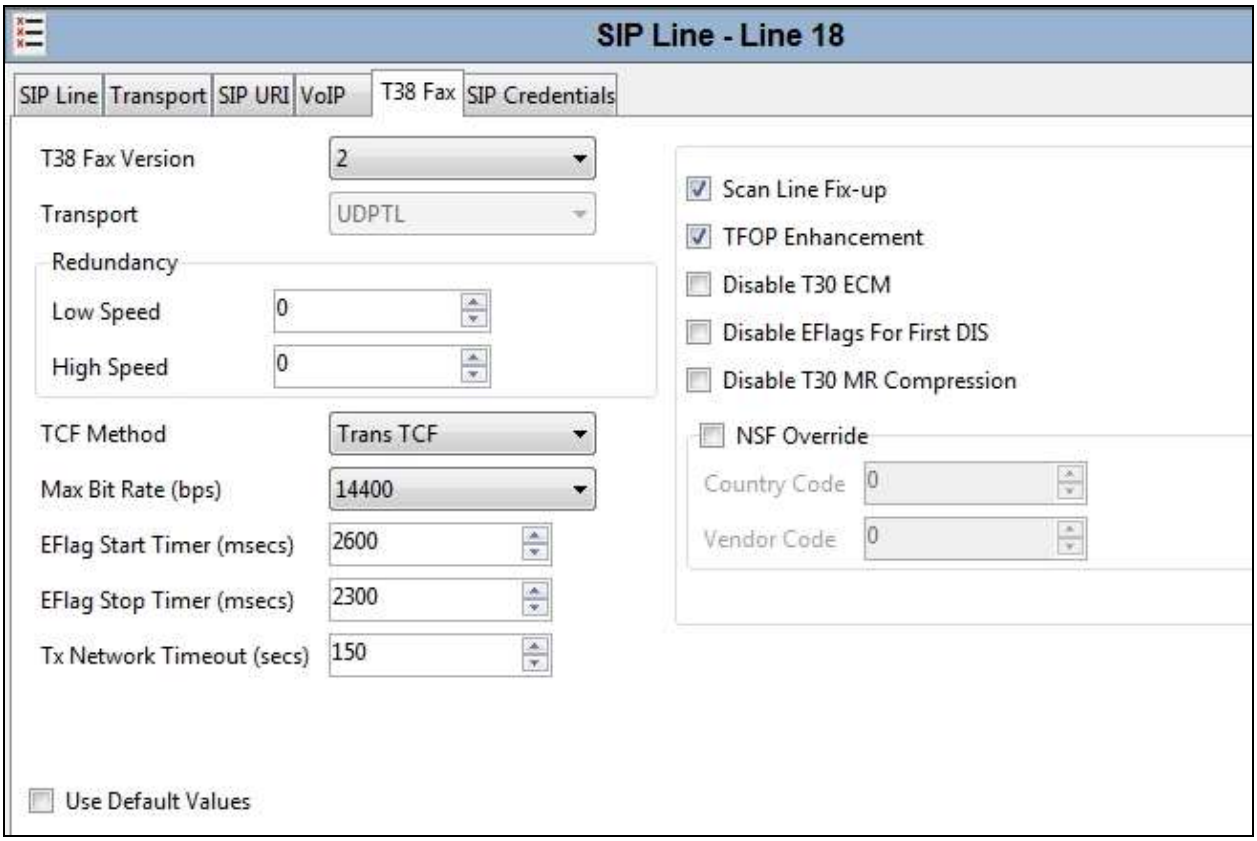
Select the **T.38 Fax** tab to set the T.38 parameters for the line. During compliance testing, Fax transmission was only reliable when **T38 Fax Version** was set to **2**. Un-check the **Use Default Values** box to change the fax settings and select **2** from the **T38 Fax Version** drop down menu. Leave all other settings at 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.

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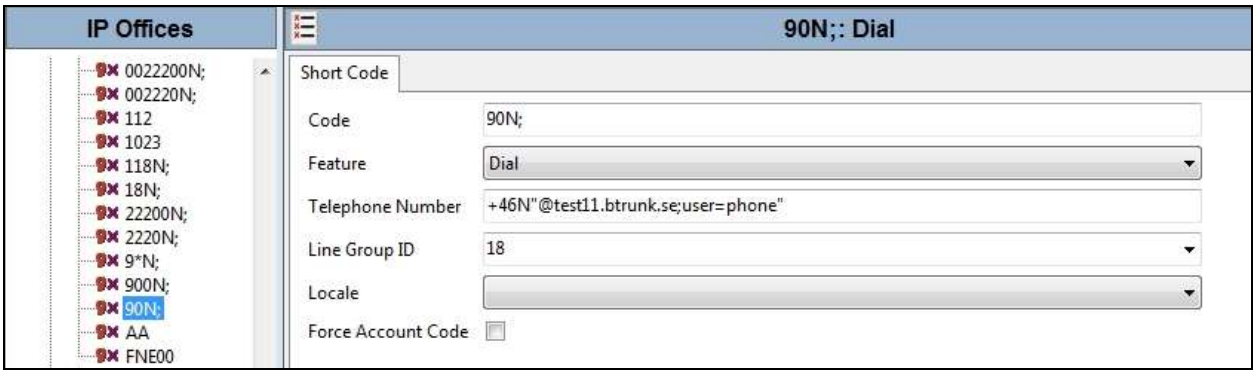
5.7. Short Codes

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**. 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 public number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **+46N** which removes the access code and inserts the number in E.164 format with leading "+" into the Request URI and To headers in the outgoing SIP INVITE message. Note that in the example the domain and "user=phone" were added though this is not necessary.
- 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).



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Define a short code to route outgoing MEX calls to the PSTN:

- The **Code** is **R1** which is a number containing 3 to 5 digits, used by the PBX to identify calls from a MEX mobile. A value of 222 was used for compliance testing and this example shows R1 followed by an international number. A single zero following R1 would indicate a national number. Note that this comes in from the PSTN with a leading "+".
- Set **Telephone Number** to **+N** which removes R1 and inserts the number in E.164 format with leading "+" into the Request URI and To headers in the outgoing SIP INVITE message. This example shows an international number. If the number was national, the **Telephone Number** would be **+46N** so that the country code would be added.

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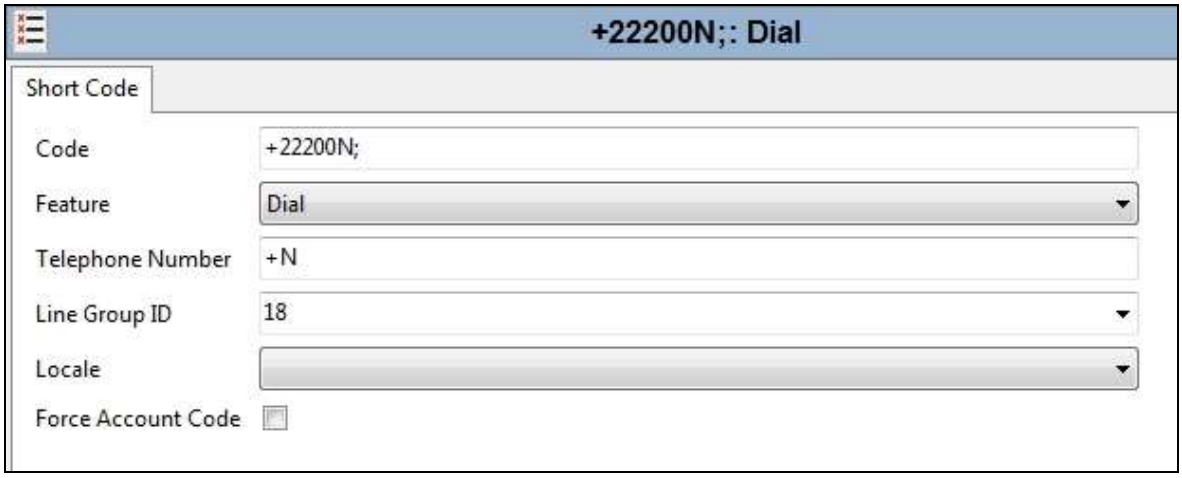
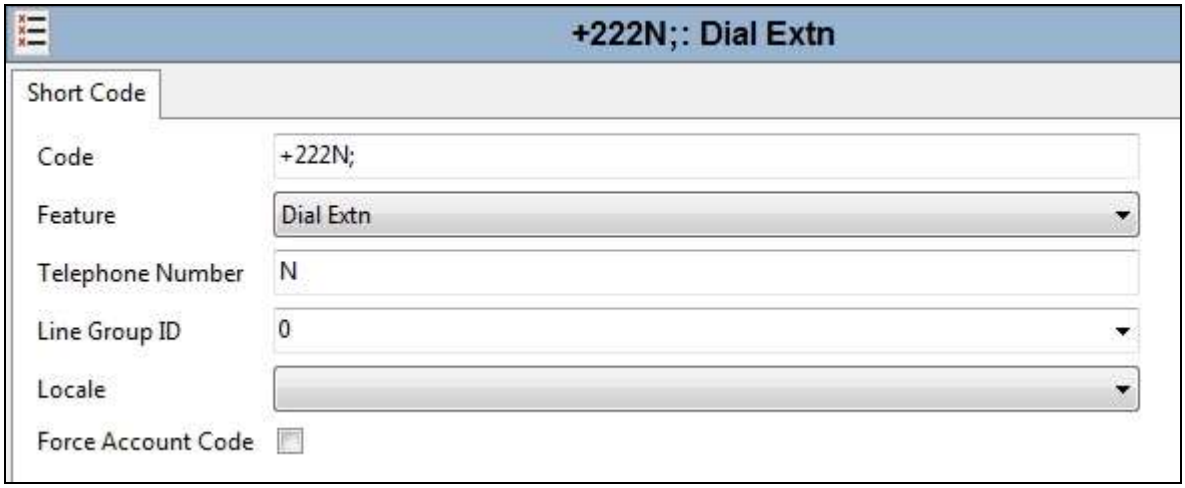


Define a short code to route outgoing MEX calls to an internal extension:

- The **Code** is **R1** followed by a number with no leading zeros. This is treated as an internal extension number.
- Set **Feature** to **Dial Extn**. This is the action that the short code will perform.
- Set **Telephone Number** to **N** which removes R1 and leading "+".



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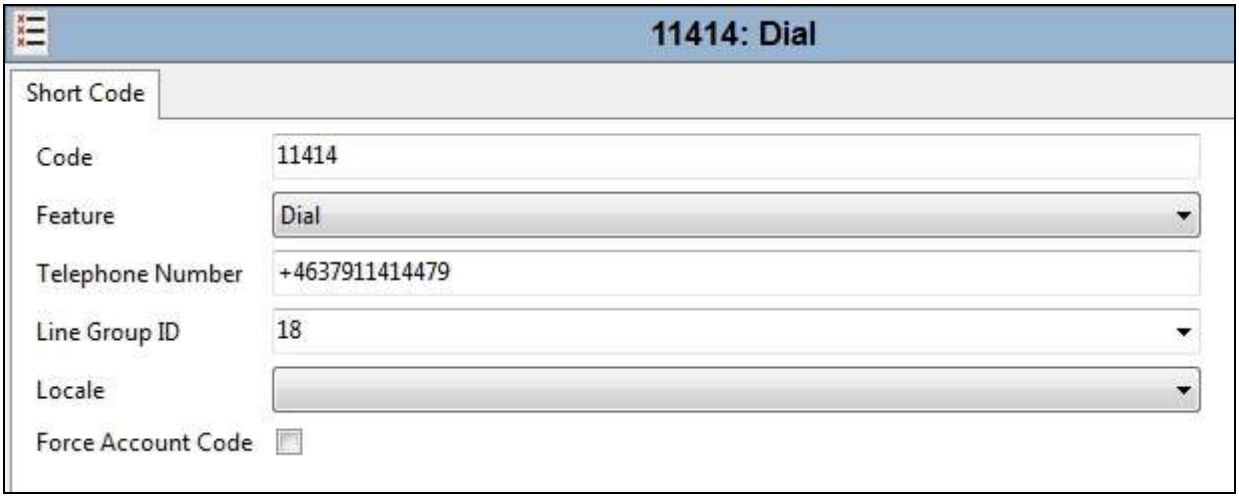
Note: In the previous screenshot, a **Line Group ID** of **0** is shown. This is not important as this feature dials an extension rather than selecting a line to dial out to the TDC network.

Define short codes for service numbers. The following example shows the short code for the Police. This requires a standard prefix of 379 and a suffix to indicate the area code.

- The **Code** is the service number as dialled by the caller.
- Set **Telephone Number** to the service number with prefix and suffix and converted to E.164 format with leading "+". In this case the prefix is **379** and the suffix is **479**.



Note: The above method can be used to reformat all service numbers. The suffix is not required for some service numbers, for example directory enquiries.

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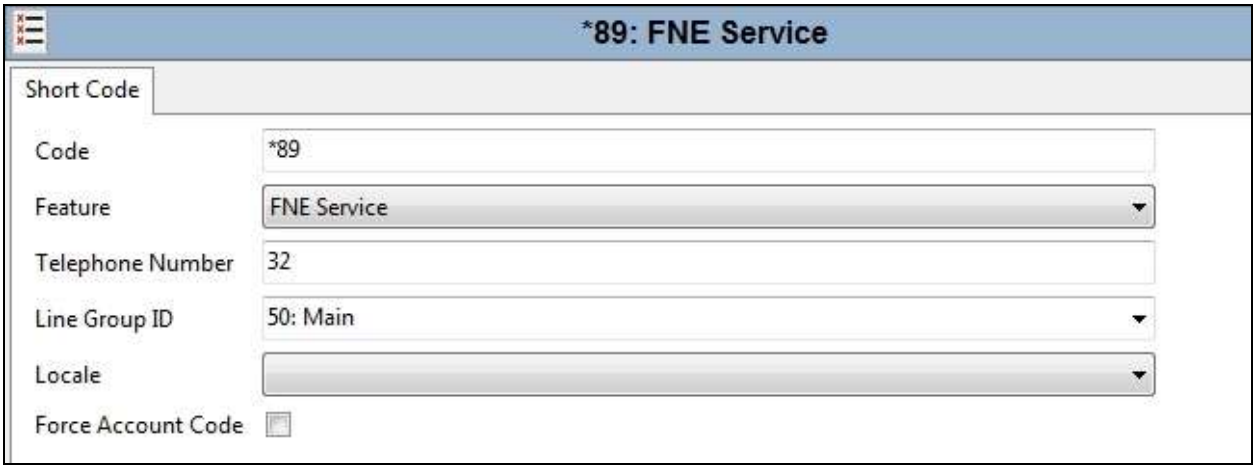
Define a short code for the Mobile Extension (MEX) outgoing calls. These calls are delivered to the IP Office from the MEX mobile for onward routing to the PSTN. A Feature Name Extension (FNE) is required to perform the onward routing and in this case, the feature required is Mobile Direct Access. This feature routes the call out directly to the received number. Mobile Direct Access is invoked by sending code 32 to the FNE function and is defined as follows:

- Set the **Code** to the short code used to invoke the FNE. This short code is defined as the destination in the Incoming Call Routing for outgoing MEX mobile calls as described in **Section 5.9**.
- Select **FNE Service** from the **Feature** drop down menu.
- Set the **Telephone Number** to **32** to invoke the Mobile Direct Access feature.
- Select the **Line Group ID** from the drop down menu. It was observed during testing that the calls go back into the short code analysis and pick up the Line Group ID for short code +222. That being the case, the setting here is not important. Testing was carried out with a setting of **50: Main**.



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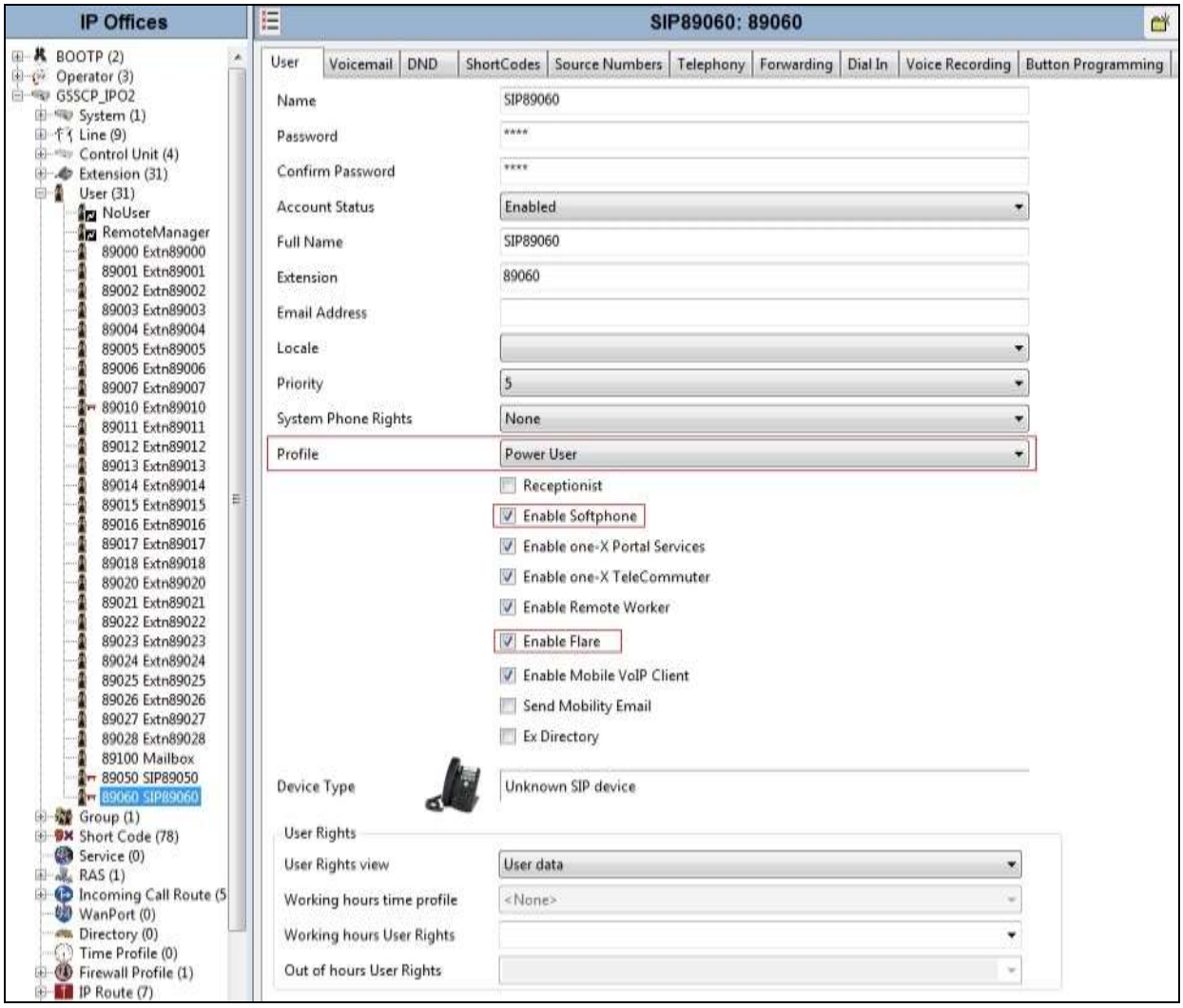
5.8. User

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 following example shows the configuration required for a SIP endpoint that can be used for a SIP phone or softphone. Two types of Softphone were successfully tested; these were IP Office Softphone and Avaya Flare® Experience for Windows. To configure a SIP User that can accommodate the different types of phone, it has to be given a profile that supports them. In test, **Power User** was selected.

Change the **Name** of the User if required, this will be used for login to the IP Office Softphone

- Select **Power User** from the Profile drop down menu.
- Check the **Enable Softphone** box.
- Check the **Enable Flare** box.



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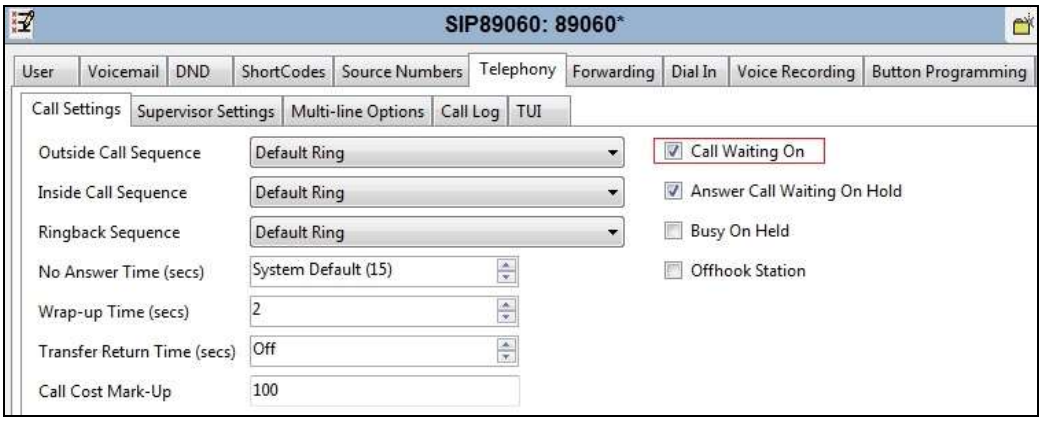
SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**. Call forwarding and transfer make use of the SIP REFER message. To handle SIP REFER from SIP endpoints on IP Office, the Call Waiting function is used. The screenshot over the page shows how this is set.

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



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 TDC 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).

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

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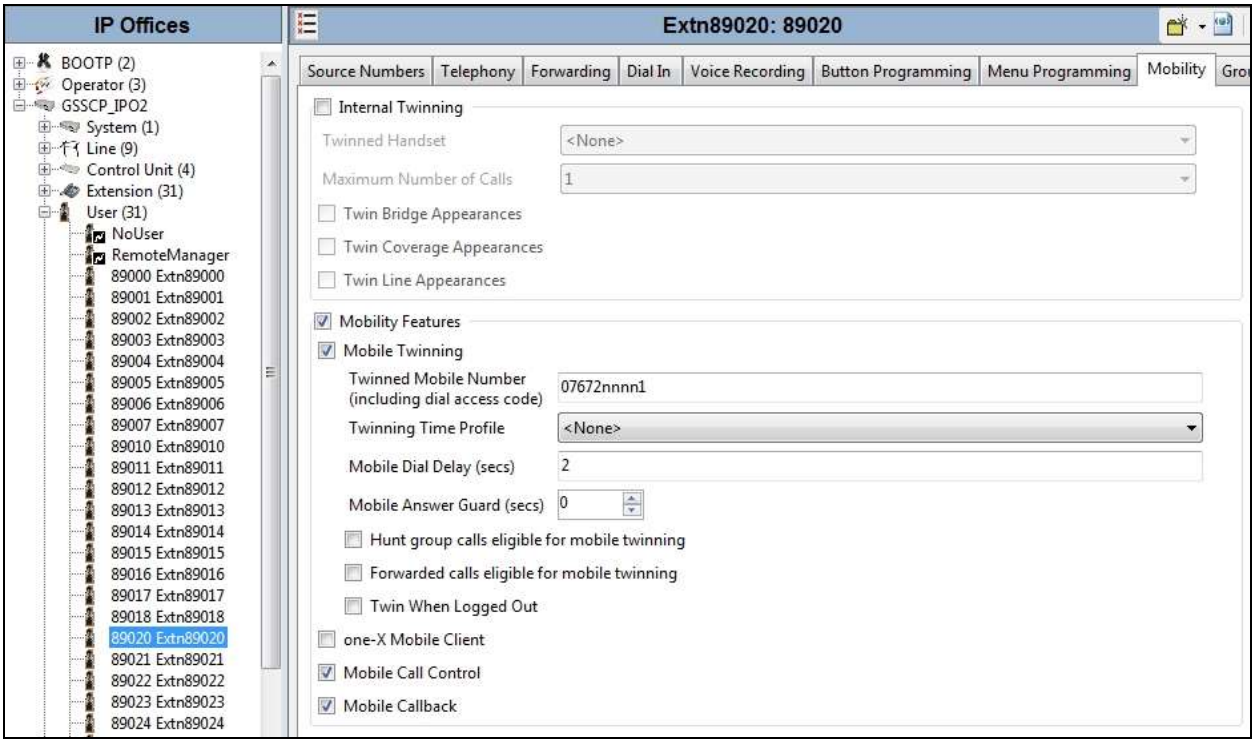
5.8.1. Mobile Extension (MEX)

A mobile extension (MEX) can be defined for each user. The following example shows a mobile defined for extension 89020. First select the **Mobility** tab and enter details as follows:

- Check the **Mobility Features** box.
- Check the **Mobile Twinning** box.
- Enter the mobile number in the **Twinned Mobile Number** field. This can be entered in national format provided the National Prefix, Country Code and International Prefix have been entered in the SIP Line as described in **Section 5.6.2**.
- Check the **Mobility Call Control** box.



Note: The CLI of outgoing MEX mobile calls must match the number specified in the **Twinned Mobile Number** field. This is required so that the FNE invoked when the call comes in to the IP Office recognizes the call as coming from a mobile extension.

The FNE is for Mobile Direct Access and is invoked by short code as described in **Section 5.7**. IP Office matches the number from right to left so that if the number specified here has a prefix, it will still match the received CLI. No prefix was used during compliance testing.

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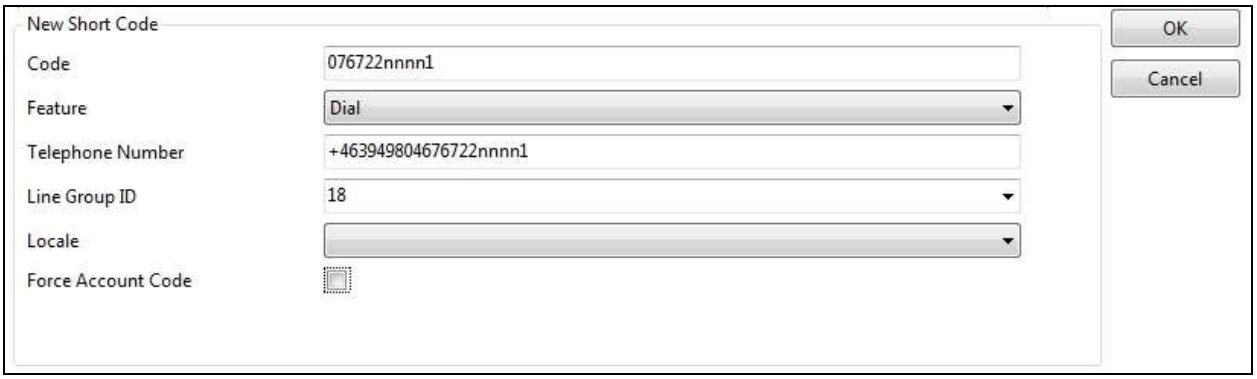
Next select the **ShortCodes** tab. Click the **Add** button and the **New Short Code** area will appear at the bottom of the pane.

The **ShortCodes** tab allows digit manipulation for a particular user. For the compliance test, the prefix for calls to the mobile extension for the user was added here. The prefix used for testing was +46394980. Enter details as follows:

- In the **Code** field, enter the number of the mobile extension as entered in the in the **Twinned Mobile Number** field in the **Mobility** tab.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to the mobile extension number and prefix with both the country code and the mobile extension prefix. The prefix used for compliance testing was +46394980
- 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.

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



Note: It may be possible to use the number with prefixes in the **Twinned Mobile Number** field in the **Mobility** tab. This would make it unnecessary to use the above short code. This is not what was done during compliance testing though, and care would have to be taken to ensure that the mobile number could still be matched.

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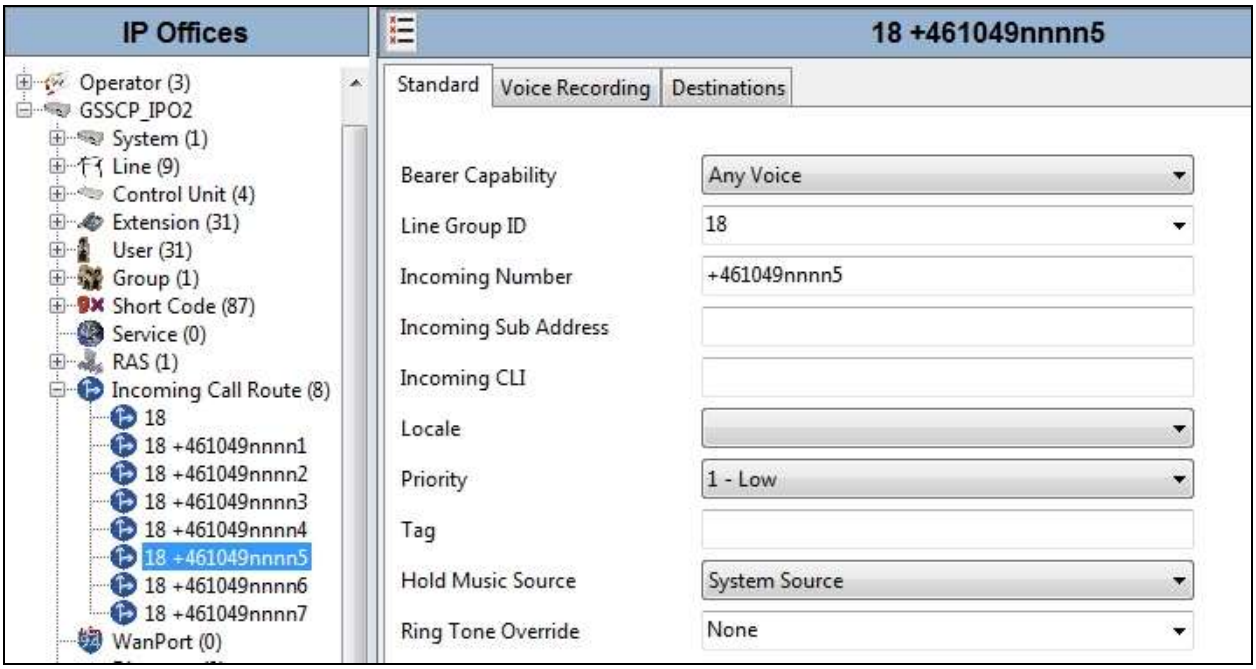
5.9. Incoming Call Routing

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**, (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 5.6**.
- 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.



Note: A number of digits of the DDI have been obscured. Number format is E.164 and prefixed with a "+".

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 89060.

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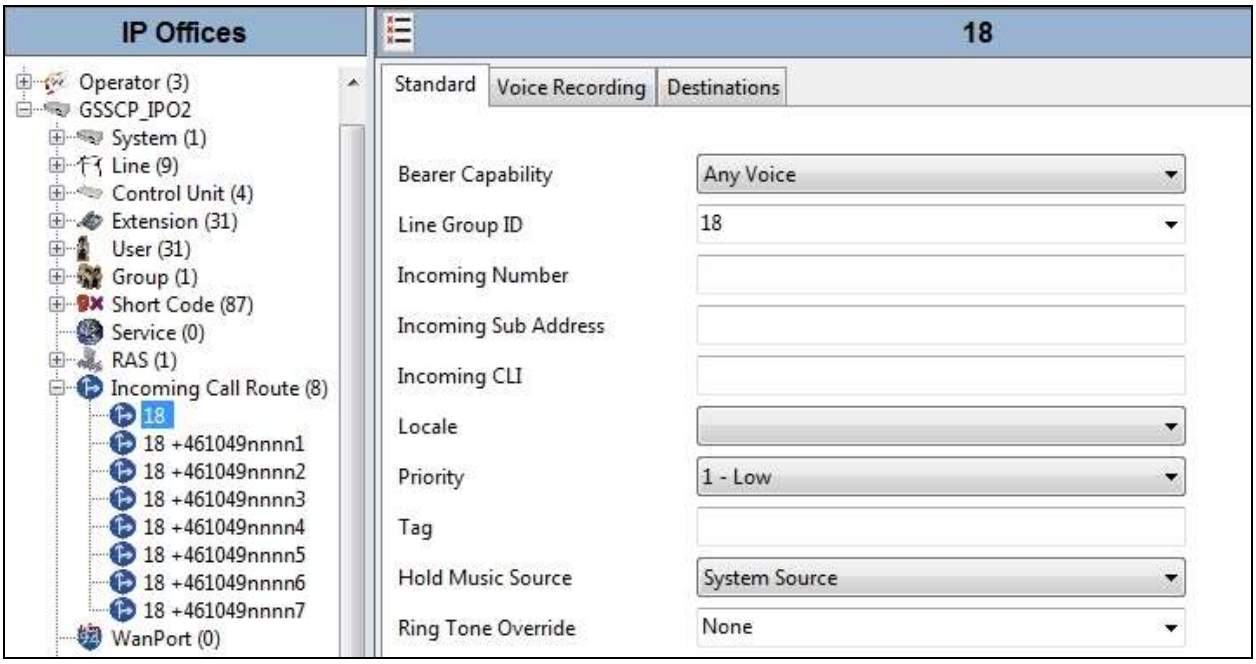
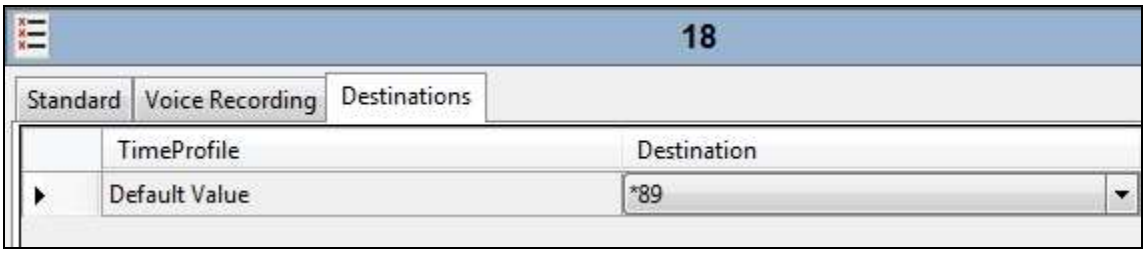
Outgoing calls from MEX mobiles must first be routed to the IP Office before onward routing to the PSTN. These calls come in to the IP Office with a prefix R1 which is a number containing 3 to 5 digits. During test, an R1 value of 222 was used so that the format of the called party number was +222<pstn number>.

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 5.6**.
- Leave the **Incoming Number** blank so that any number that does not match any of the DDI numbers allocated to the IP Office will use this incoming route.
- Set **Incoming CLI** to the number of the MEX mobile if required for screening. This was not done during compliance testing
- Default values can be used for all other fields.



On the **Destinations** tab, type the short code for the FNE described in **Section 5.7** in the **Destination** field. This FNE routes the call out to the received called party number. On completion, click the **OK** button (not shown)..



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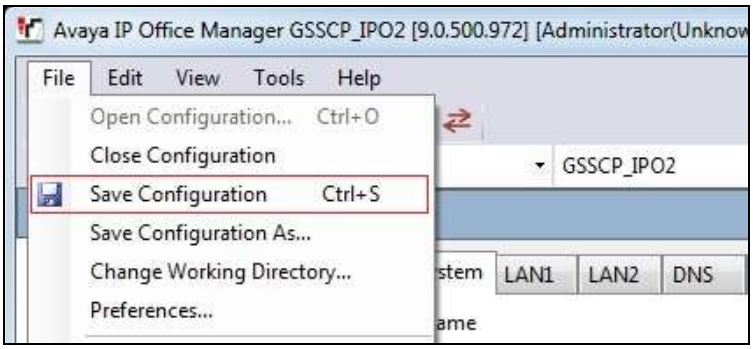
5.10. Save Configuration

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



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6. TDC Business Trunk Configuration

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

- Domain Name or IP address of TDC 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.

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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. 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.

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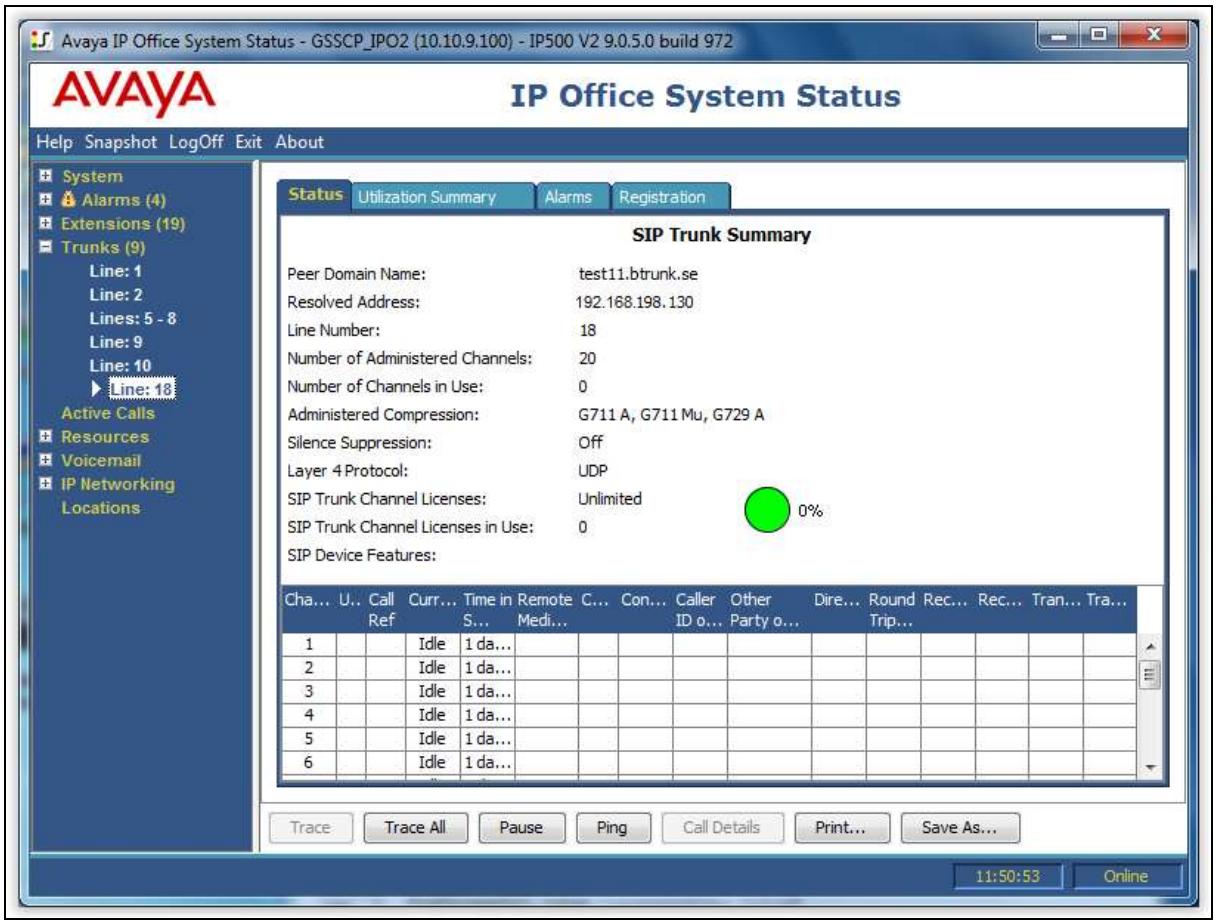
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.



8. Conclusion

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

9. Additional References

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

[1] *IP Office 9.0 KnowledgeBase Technical Documentation*, Web based.

[2] *IP Office 9.0 Installing IP500/IP500 V2*, Document number15-601042, 6th November 2014.

[3] *IP Office Application Server 9.0 Installation and Maintenance*, Document number15-601011, 13th November 2014.

[4] *IP Office Manager*, Document number 15-601011, 8th December 2014

[5] *IP Office 9.0 Using System Status*, Document number15-601758, 30th November 2014.

[6] *IP Office 9.0 Installing IP Office Video Softphone*, Document number 100173998, 21st August 2013.

[7] *IP Office 9.0 SIP Extension Installation*, 21st August 2013.

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