Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office 7.0 with Tele2 VoIPConnect Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the Tele2 VoIPConnect Service and Avaya IP Office. The Tele2 VoIPConnect Service provides PSTN access via a SIP trunk connected to the Tele2 Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or digital trunks. Tele2 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 the Tele2 VoIPConnect Service and Avaya IP Office. Tele2 VoIPConnect Service provides PSTN access via a SIP trunk connected to the Tele2 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 the Tele2 VoIPConnect Service. This configuration (shown in **Figure 1**) was used to exercise the features and functionality listed in **Section 2.1**.

2.1. Interoperability Compliance Testing

Avaya IP Office was connected to the Tele2 VoIPConnect Service. To verify SIP trunking interoperability the following features and functionality were exercised during the interoperability compliance test:

- Incoming PSTN calls to various phone types
- Phone types included H.323, Digital, and Analogue telephones at the enterprise.
- All inbound PSTN calls were routed to the enterprise across the SIP trunk from the Service Provider
- Outgoing PSTN calls from various phone types
- Phone types included H.323, Digital, and Analogue telephones at the enterprise
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider
- Inbound and outbound PSTN calls to/from Phone Manager Lite clients
- Various call types including: local, long distance, international, toll free (outbound)and directory assistance (1802)
- Codecs G.711A and G.711Mu
- Caller ID presentation and Caller ID restriction
- DTMF transmission using RFC 2833
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning
- T.38 fax

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Tele2 VoIPConnect Service with the following observations:

- Transport Protocol used was UDP
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested
- Only the number translation for routing to emergency number 112 was tested, no call was made to the Operator

2.3. Support

For technical support on Tele2 products please contact the Tele2 support team at:

Telephone number:	+31 (0) 900 – 240 1602
Web url:	www.tele2.nl/zakelijk/customer-service.html

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Tele2 VoIPConnect Service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include two Avaya 1600 Series IP Telephones (with H.323 firmware), 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. 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 obscured and all phone numbers have been obscured beyond the city code.

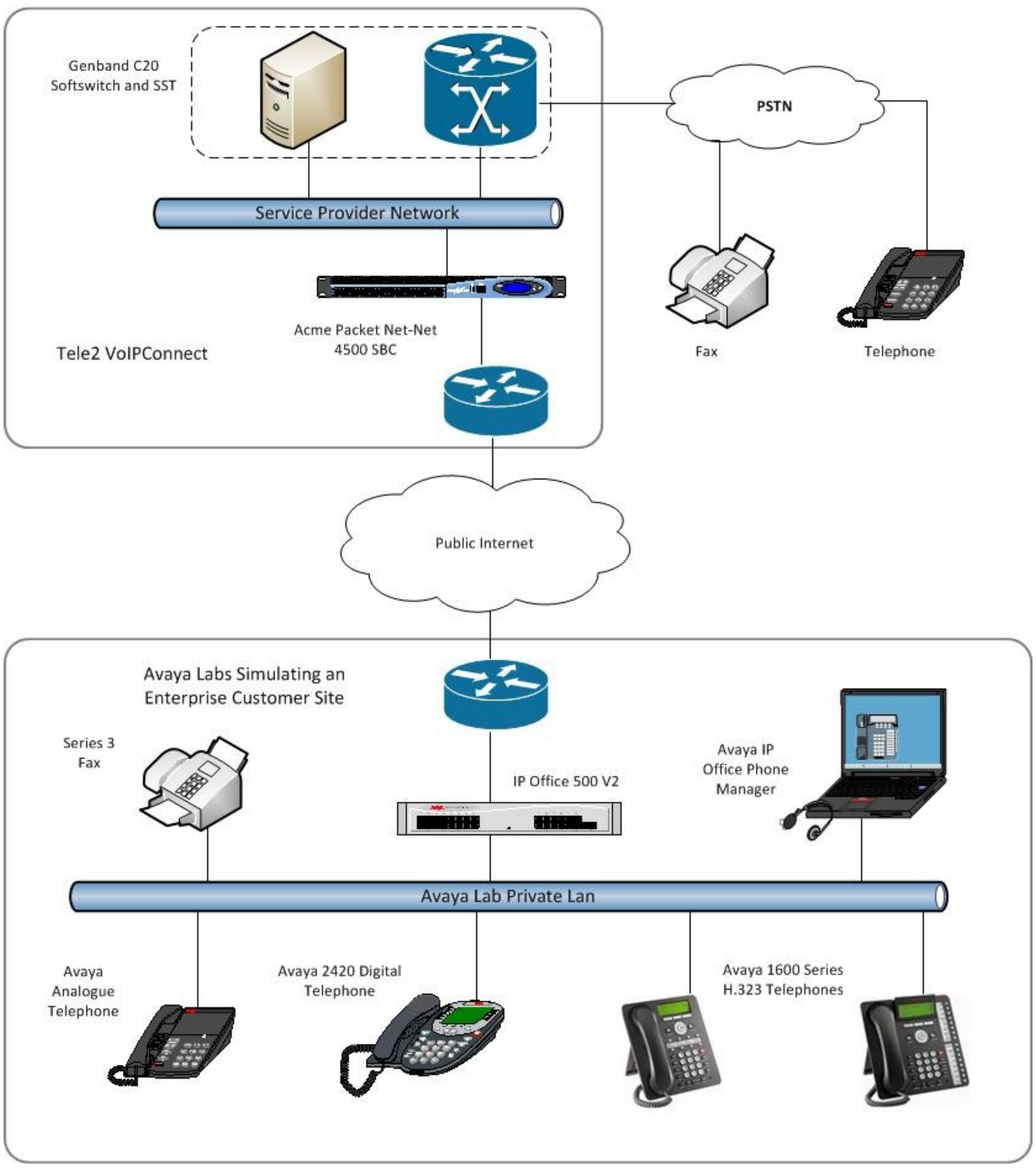


Figure 1: Tele2 VoIPConnect Service Solution to Avaya IP Office Topology

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 Tele2. The short code of 9 is stripped off by Avaya IP Office but the remaining N digits were sent with the SIP domain provided by Tele2 added. The only exception was emergency services number 112 which was configured to work without the 9 prefix.

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. Tele2 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 Tele2 to determine the proper IP addresses and ports that require access to their network.

4. Equipment and Software Validated

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

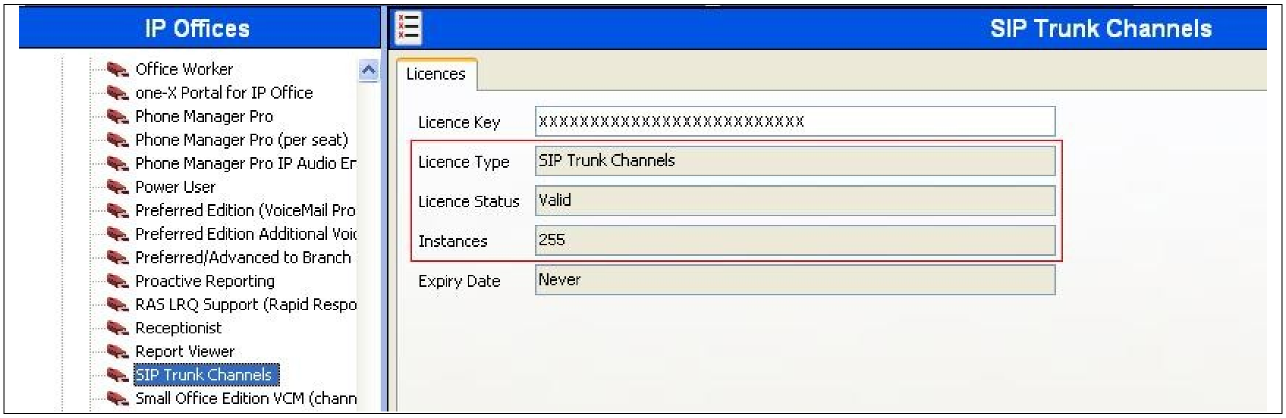
Equipment	Software
Avaya IP Office 500 V2	Avaya IP Office 7.0(3)
Avaya 1603 Phone (H.323)	1.3
Avaya 1608 Phone (H.323)	1.3
Avaya 2420 Digital Phone	NA
Avaya 98390 Analogue Phone	NA
Tele2 Voipconnect platform	Version 2.0
Genband C20 (Nortel CS2K)	CVM14
Genband SST (Session Server Trunks)	CVM14
Acme Packet Net-net 4500	6.2.0 MR8

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the Tele2 VoIPConnect Service. Avaya IP Office is configured through the Avaya IP Office Manager PC application. From a PC running the Avaya IP Office Manager application, select **Start → Programs → IP Office → Manager** to launch the application. Navigate to **File → Open Configuration**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials. A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All licensing and feature configuration that is not directly related to the interface with the Service Provider (such as twinning) is assumed to already be in place.

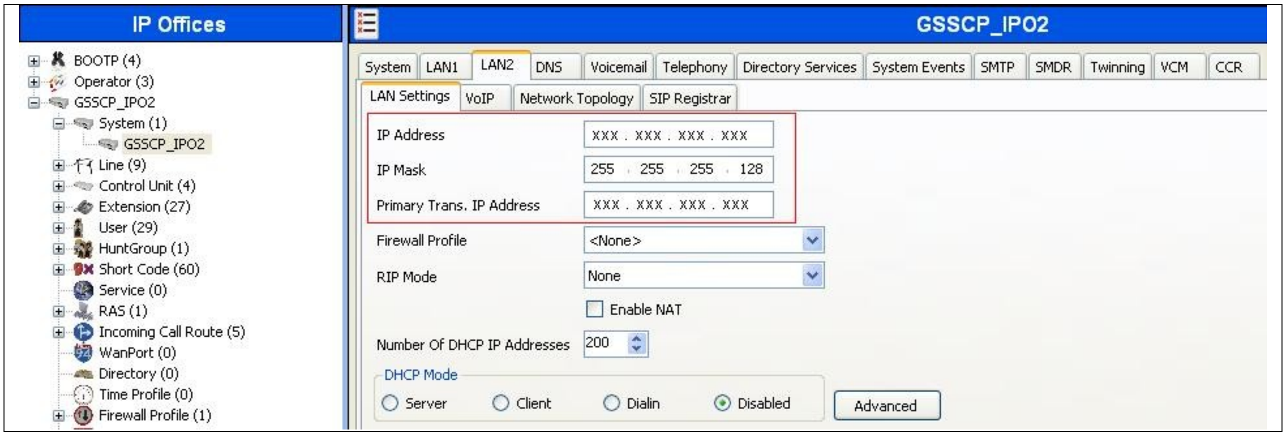
5.1. Verify System Capacity

Navigate to **License → SIP Trunk Channels** in the Navigation Pane. In the Details Pane verify that the **License Status** is **Valid** and that the number of **Instances** is sufficient to support the number of SIP trunk channels provisioned by Tele2.

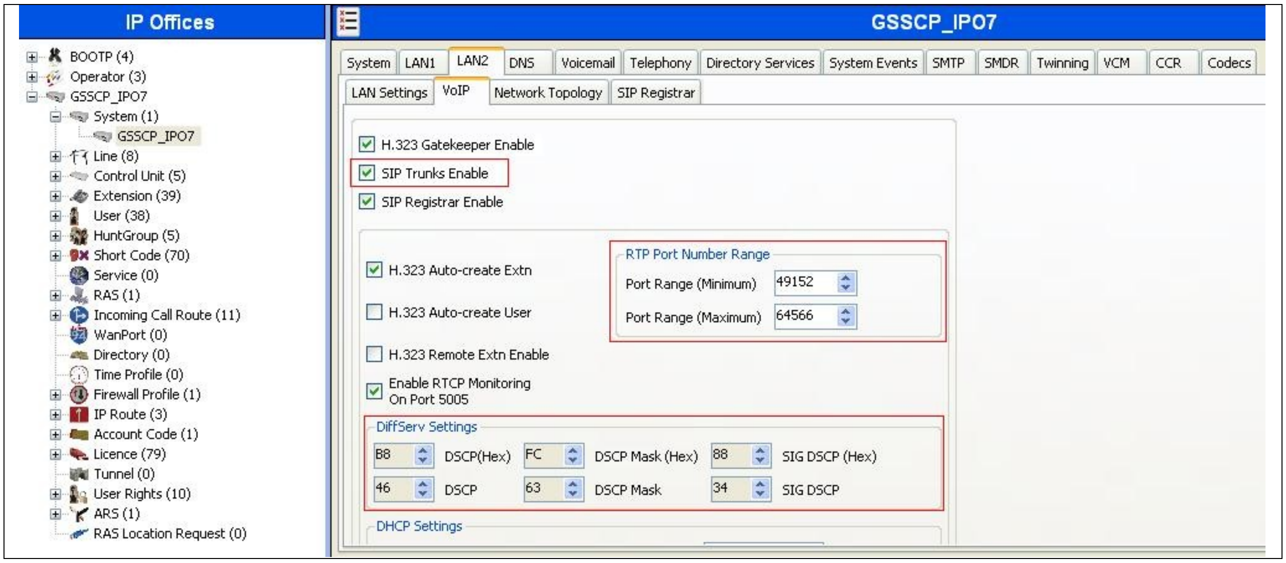


5.2. LAN2 Settings

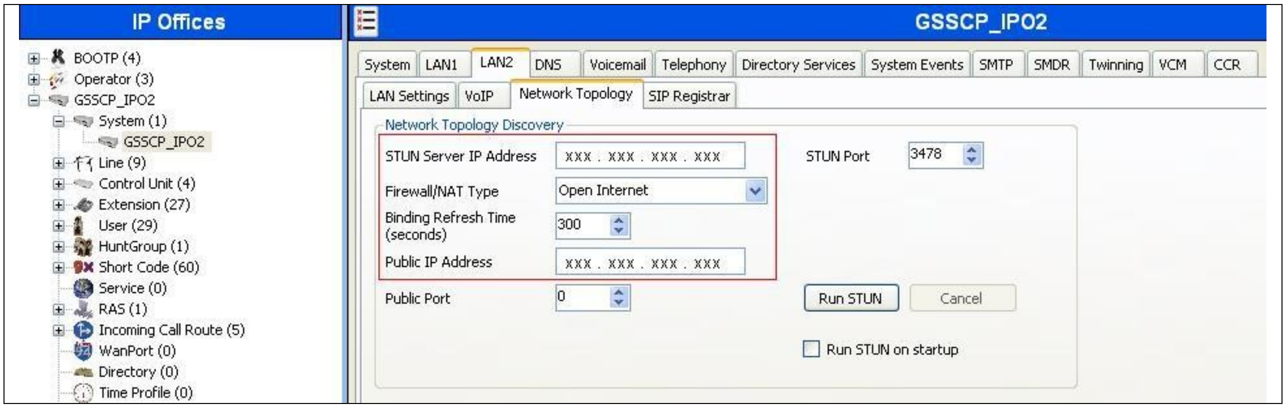
In the sample configuration, the WAN port defined by LAN2 settings was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO7** in the Navigation Pane where GSSCP_IPO7 is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the Avaya 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 **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).

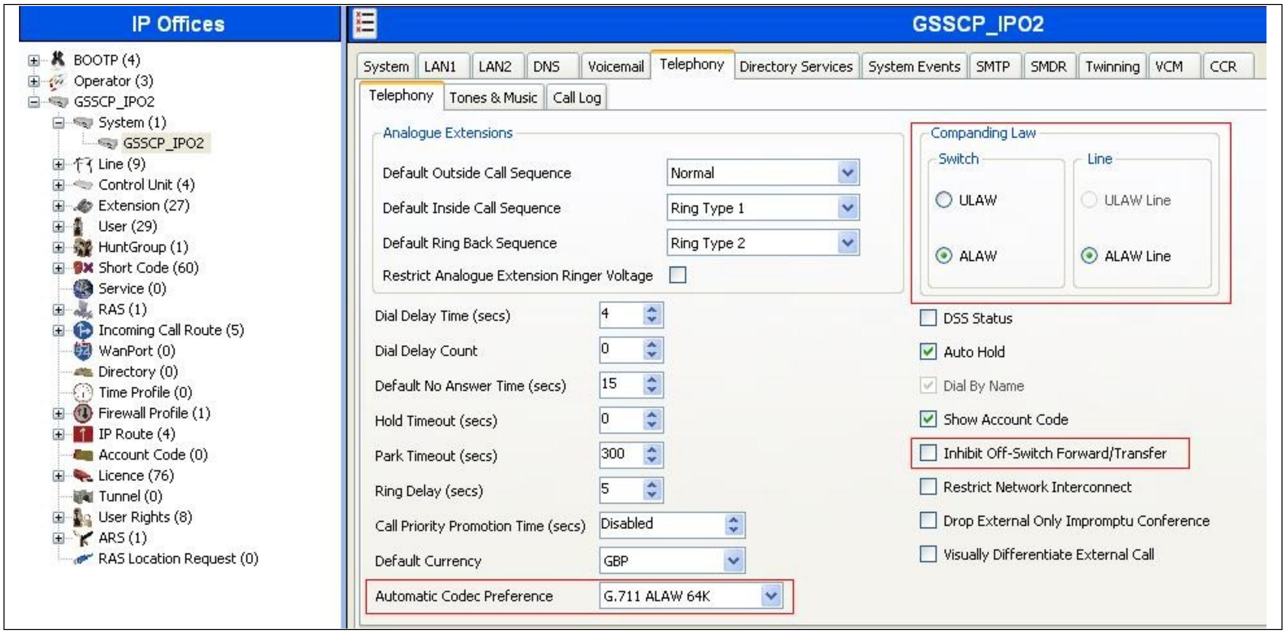


On the **Network Topology** tab in the Details Pane enter the **Public IP Address** for the IP Office. The same **Public IP Address** number is used in the **STUN Server IP Address** field, even if not running STUN. It is important that the **Binding Refresh Time** is set to the correct value. Avaya 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, see **Section 5.9** for more details. Below is a sample configuration. On completion, click the **OK** button (not shown).



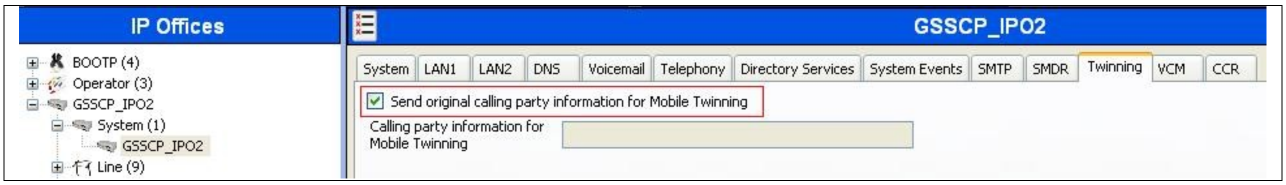
5.3. System Telephony Settings

Navigate to the **Telephony → Telephony** tab on the Details Pane. Set the **Automatic Codec Preference** for the default codec to be used for intra-enterprise traffic. Choose the **Companding Law** typical for the enterprise location. For Europe, **ALAW** is used. Uncheck the **Inhibit Off-Switch Forward/Transfer** box to allow call forwarding and call transfer to the PSTN via the Service Provider across the SIP trunk. 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 5.5**). On completion, click the **OK** button (not shown).

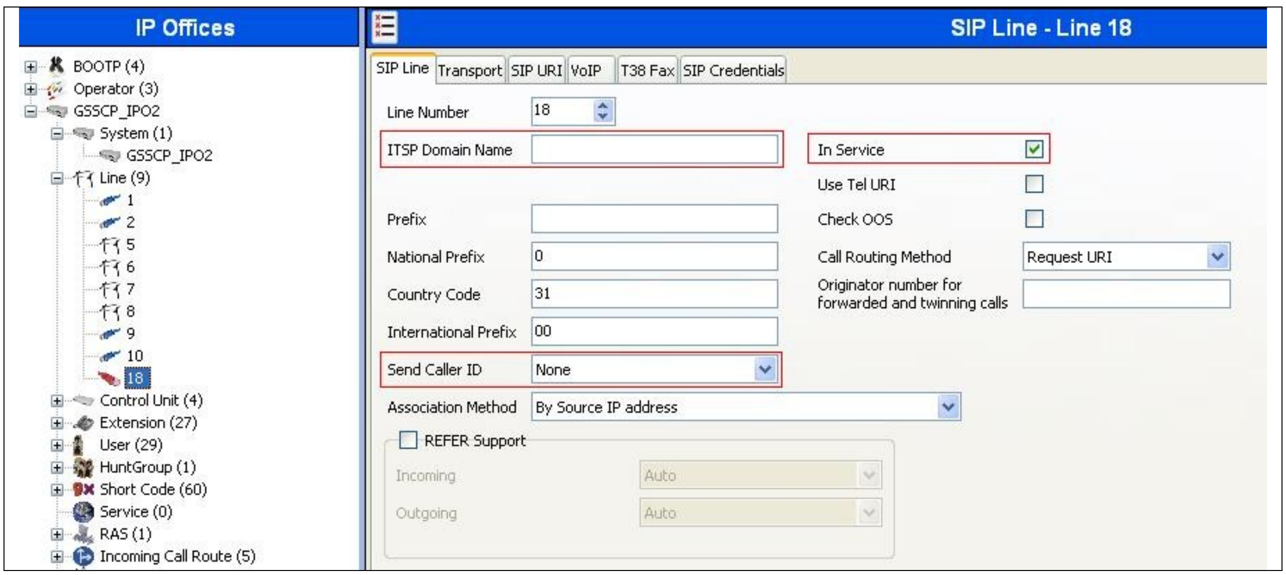


5.5. Administer SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Tele2 VoIPConnect 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 Tele2 VoIPConnect 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

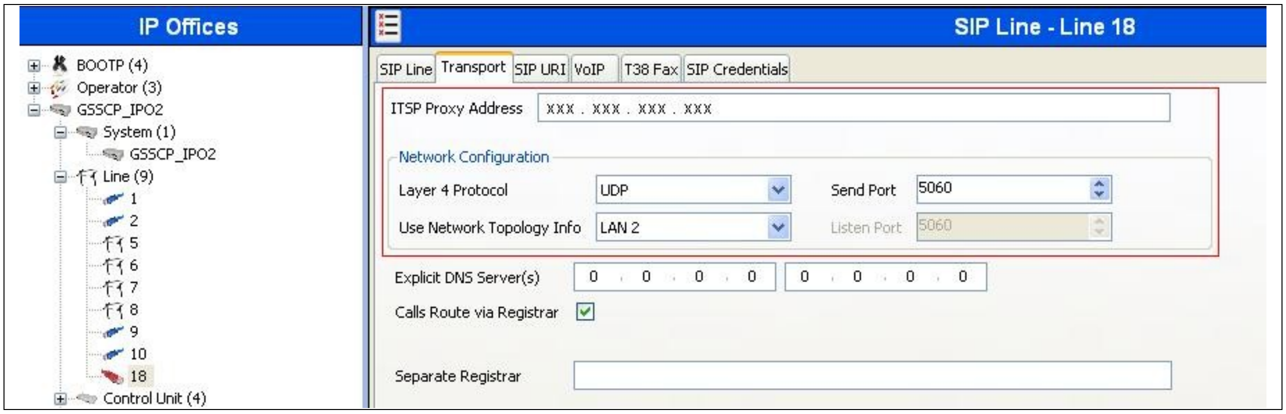
On completion, click the **OK** button (not shown).



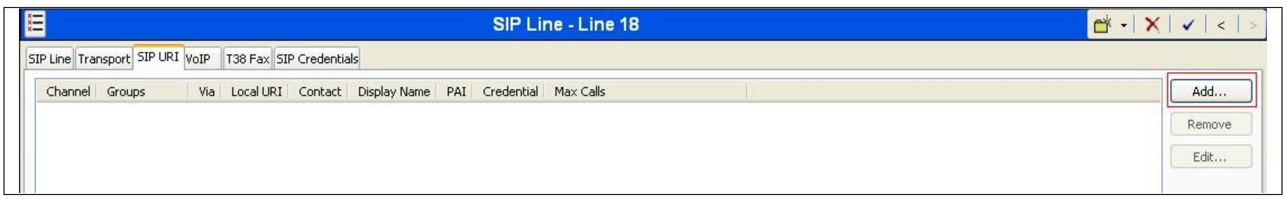
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the IP address of the Tele2 SIP proxy
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** and **Listen Port** to **5060**

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



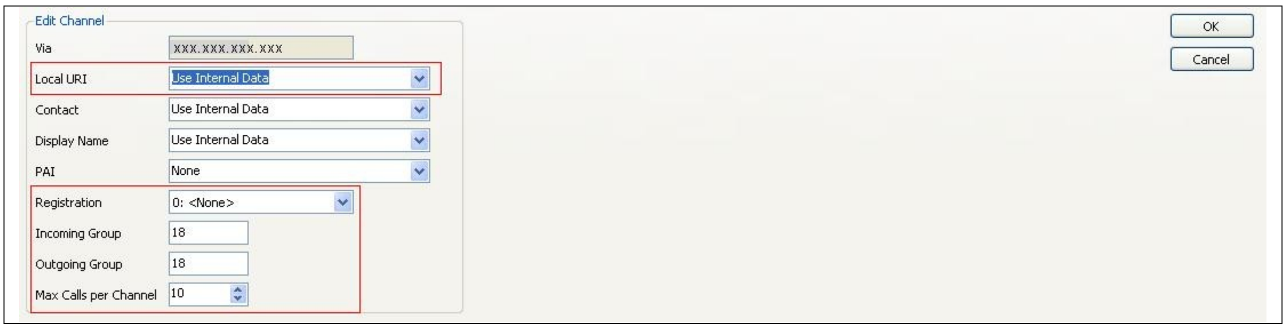
After the SIP line parameters are defined, each SIP URI that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



For the compliance test, a single SIP URI entry was created that matched any number assigned to an Avaya IP Office user. The entry was created with the parameters shown below.

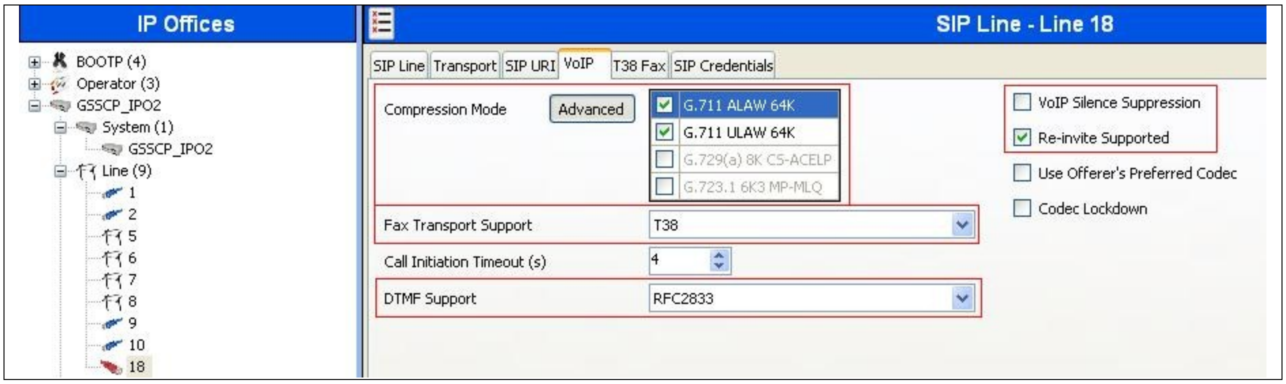
- Set **Local URI** to **Use Internal Data**, This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.7**.
- For **Registration**, select **0: <None>** from the pull-down menu since this configuration does not use SIP registration.
- 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 (not shown).



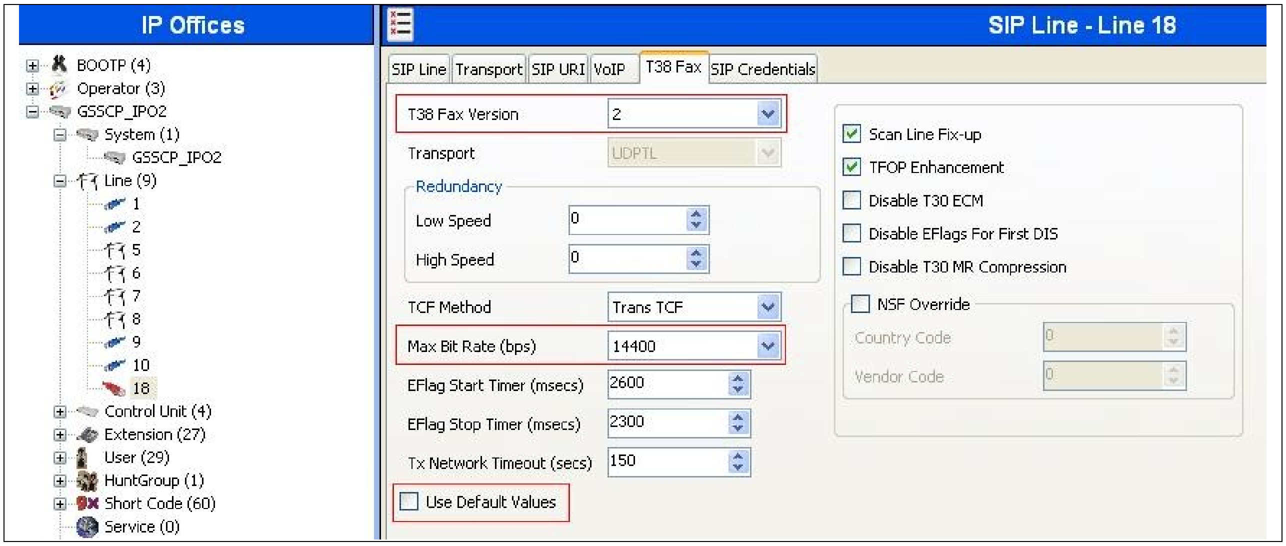
Select the **VoIP** tab, to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- Configure the **Compression Mode** with the **Advanced** button to specify the preferred order of the offered codecs, for testing with Tele2 this was **G.711 ALAW 64K** followed by **G.711 ULAW 64K**
- Check the **Fax Transport Support** box to allow T.38 fax operation
- Set the **DTMF Support** field to **RFC2833**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833
- Uncheck the **VoIP Silence Suppression** box
- Check the **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 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).



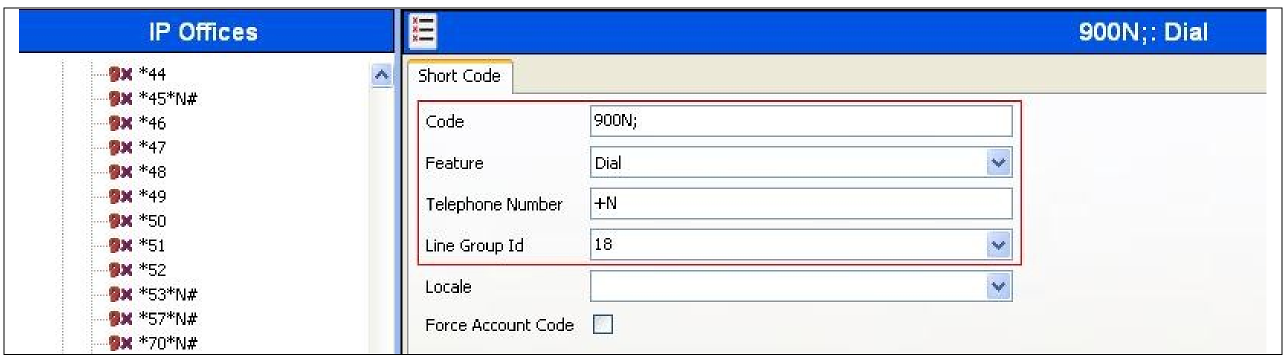
Note: It is advisable at this stage to save the configuration as described in **Section 5.10** to make the Line Group ID available in **Section 5.5**.

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 below.

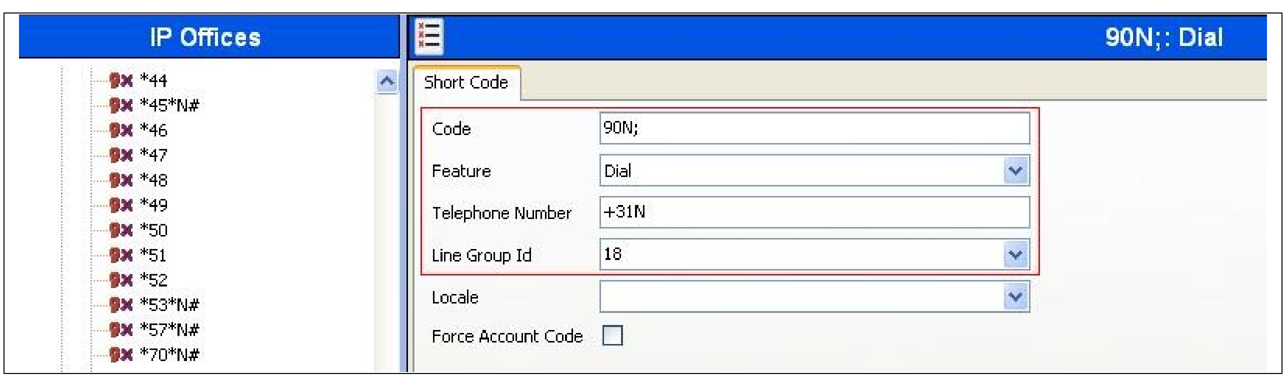
- In the **Code** field, enter the dial string which will trigger this short code, followed by a semi-colon.
- The example shows **900N;** which will be invoked when the user dials 9 followed by an international number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- 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 5.5.**

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



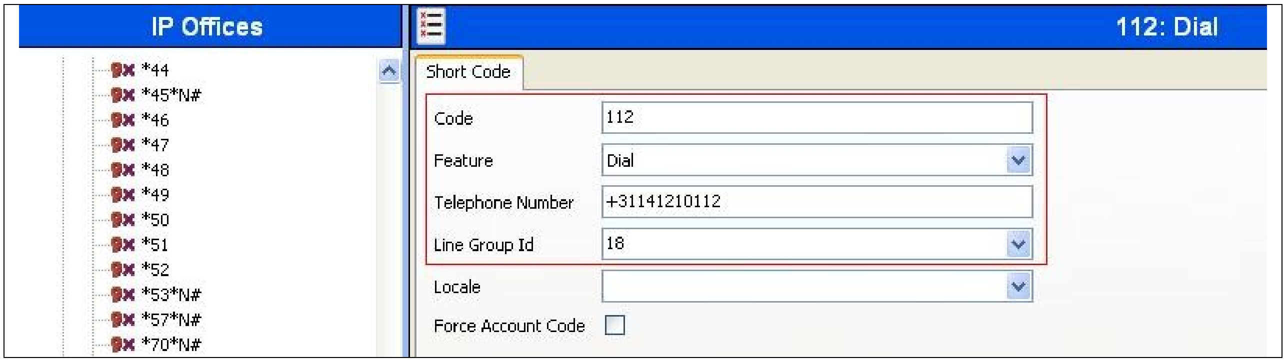
Short codes are also used for routing of national calls and Operator calls. An example for national calls is shown below.

- 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 **+31N** 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.



An example for Operator calls, in this case emergency services, is shown below.

- The example of an Operator call shows **112;** which will be invoked when the user dials emergency services
- Set **Telephone Number** to **+31141210112** which will translate the number to international format for routing and 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 first example



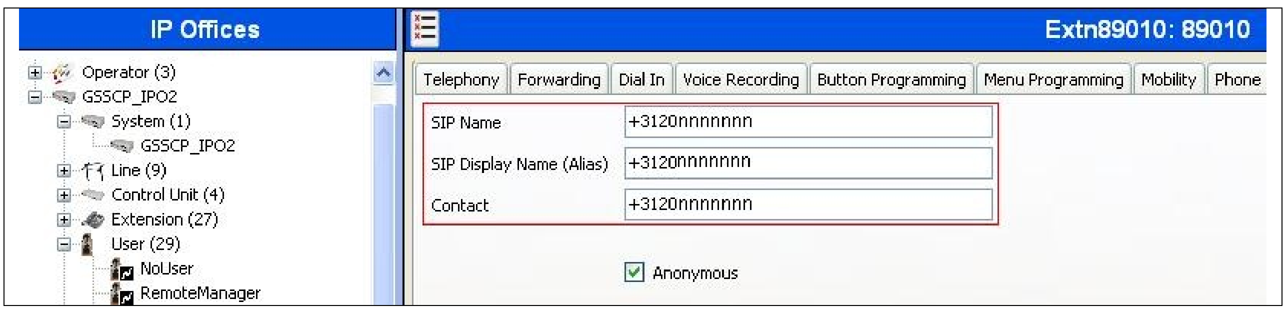
Note: The translated number shown is for test purposes only and is shown as an example. This should not be used in a live network installation.

5.7. User Management

Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.5**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **SIP** (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 you see the **SIP** tab. 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 5.5**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Tele2.

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

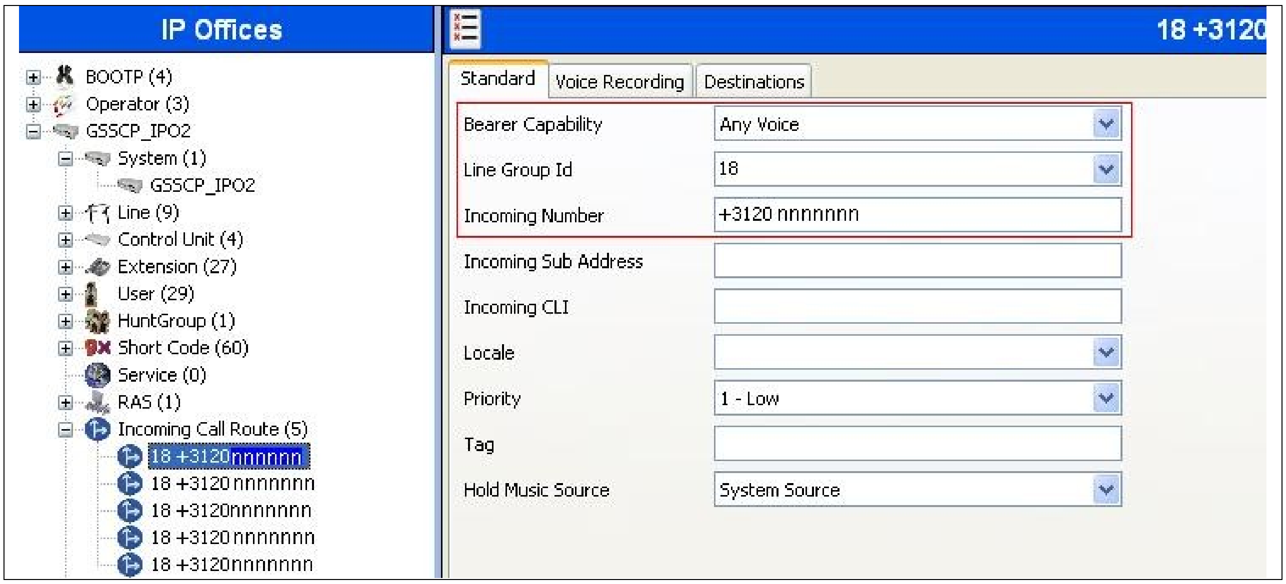
Note: The **Contact** field must be in E.164 format for the caller ID on the called phone to display properly.



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 Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to **Any Voice**
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5**
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left
- Default values can be used for all other fields



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.

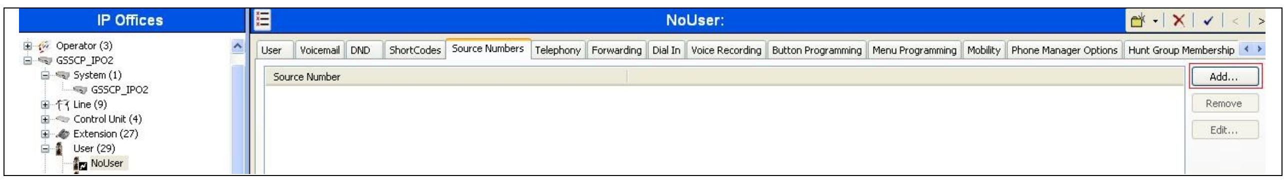
5.9. SIP Options

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:

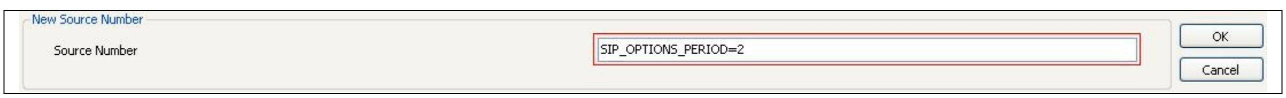
- If no **SIP_OPTIONS_PERIOD** parameter is defined and the **Binding Refresh Time** is 0, then the default value of 44 seconds is used
- To establish a period less than 42 seconds, do not define a **SIP_OPTIONS_PERIOD** parameter and set the **Binding Refresh Time** to the value required
- To establish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

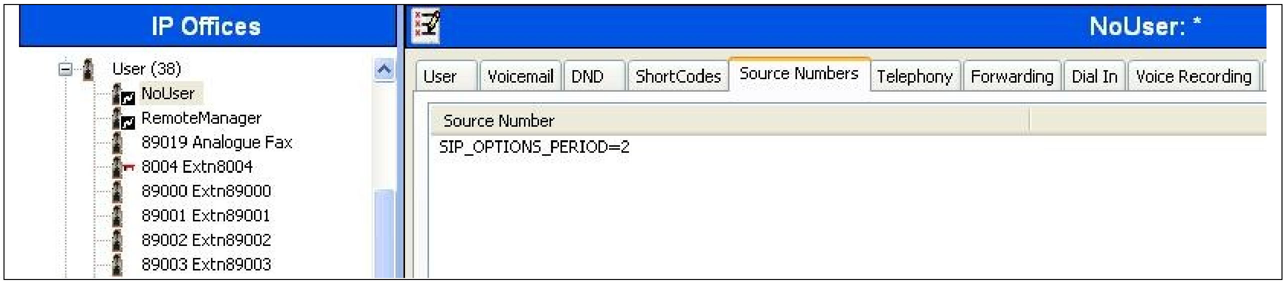
To configure the **SIP_OPTIONS_PERIOD** parameter, navigate to **User → NoUser** in the Navigation Pane. Select the **Source Numbers** tab in the Details Pane. Click the **Add** button.



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).



5.10. 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. Tele2 VoIPConnect Configuration

Tele2 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. Tele2 will provide the customer the necessary information to configure the SIP connection to the VoIPConnect 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

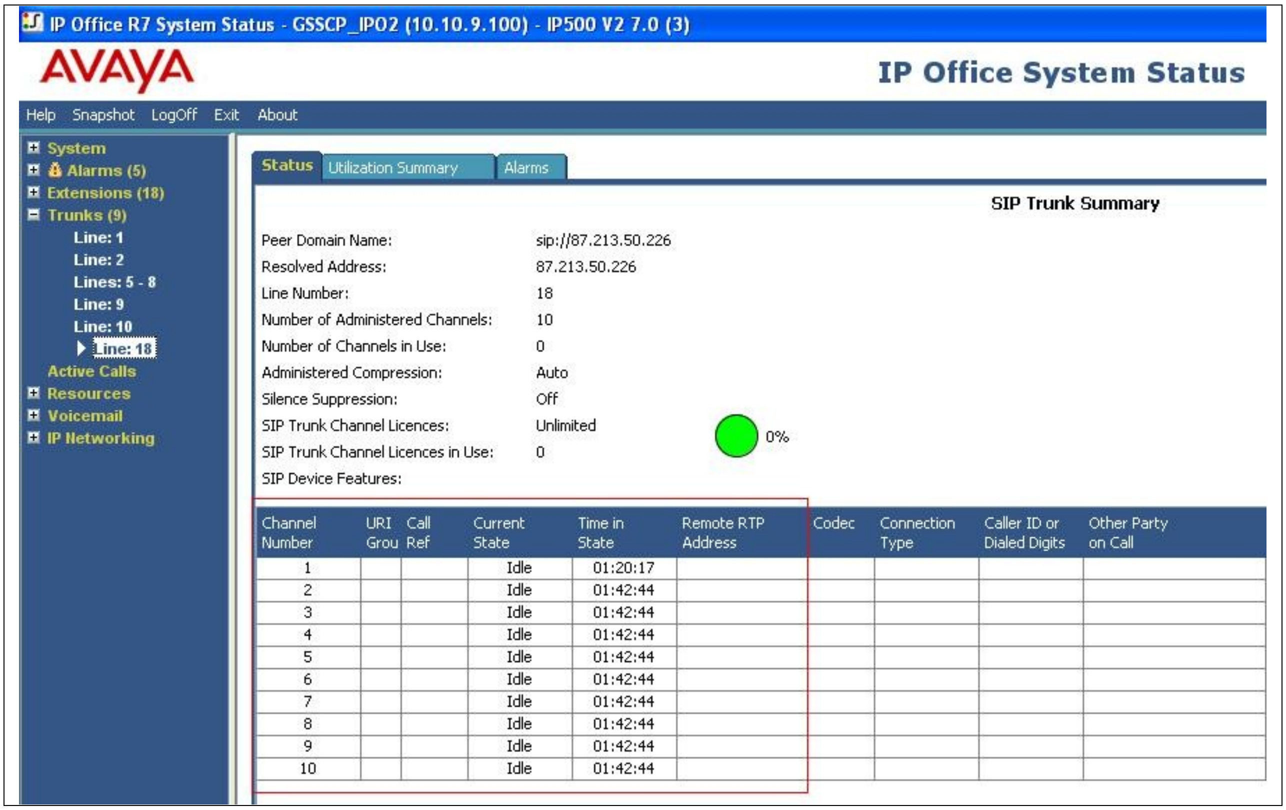
The first point of contact for any of the above information must be the Tele2 representative.

7. Verification Steps

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

7.1. SIP Trunk status

The status of the SIP trunk can be verified by opening the System Status application. This is found under **Start →All Programs →IP Office →System Status.** From the left hand menu expand **Trunks** and choose the 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 Tele2 VoIPConnect service passed compliance testing. Interoperability testing of the Tele2 VoIPConnect Service with the Avaya IP Office was completed with successful results. 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 7 Documentation CD, 4th May 2011.

[2] Avaya IP Office Installation, Document number15-601042, 22nd May 2011.

[3] Avaya IP Office Manager, Document number15-601011, 22nd May 2011.

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

10. Appendix A

10.1. Tele2 VoIPConnect Service Definition

Service definition for the Tele2 VoIPConnect service is as follows:

Basic specifications

- SIP protocol RFC 3261
- Offer/answer model RFC3264
- SDP RFC 4566

Mode of operation

- Dynamic registration modus: not supported
- Static mode (fixed IP address): supported

Security

- TLS: not supported
- SRTP: not supported

  N.B.: IPVPN Access intrisic secure

Formats

- E.164 style based SIP INVITES (SIPConnect compliant) e.g. sip:+31207501000@62.59.93.4:user=phone
- As dialed SIP INVITES (Tele2 ISDN30 compliant) e.g. sip:0207501500@62.59.93.4;user=phone Attention From header must still be +E.164 formaat.

Media

- RTP: G.711Alaw/UDP, @20ms,160 byte packets
- RTCP supported
- Early media supported
- Voice Activation Detection : not supported
- Comfort noise : not supported

SIP signaling

- Tele2 SIP interface IP address: 62.59.93.4 or 62.59.94.4 depending on location customer
- Destination port 5060
- Transport: TCP of UDP (TCP recommended above UDP)

Fax & modem support

- Fax / modem support : support passthrough mode for Tele2 offnet traffic
- Fax: T.38/UDPTL : Passtive supported (Tele2 network does not trigger T.38 but does support t.38 in SDP offer/answer and accepts Re-invites for t.38.

DTMF support

- DTMF inband: supported
- RFC2833: supported

DNS

- DNS name resolving: not supported
- Domain part URIs

Access:

- Via IP-VPN : ethernet, 10Mb/100Mb, full duplex, RJ45

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.