Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 9.1 to support M-net Premium SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between M-net Premium SIP Trunk Service and Avaya IP Office 9.1.

The M-net Premium SIP Trunk Service provides PSTN access via a SIP trunk connected to the M-net Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or Digital trunks. M-net is a member of the Avaya DevConnect Service Provider program.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the M-net Premium SIP Trunk and Avaya IP Office 9.1. Customers using this Avaya SIP-enabled enterprise solution with M-net's SIP Trunk are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using Avaya IP Office 9.1 to connect to the M-net Premium SIP 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

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.
- 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, SIP, Digital, and Analogue telephones at the enterprise.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Calls using the G.711A and G.729 codecs.
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using G.711 pass-through transmission.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- Inbound and outbound PSTN calls to/from Avaya Communicator Softphone client.
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance.
- Caller ID presentation and Caller ID restriction.
- User features such as hold and resume, transfer, and conference.
- Off-net call forwarding and mobile twinning.

2.2. Test Results

Interoperability testing of the test configuration was completed with successful results for M-net's SIP Trunk service with the following observations:

- Inbound calls from the PSTN to the enterprise which are then call forwarded to another PSTN endpoint, show an incorrect caller ID display at the destination. The destination show the enterprise account DDI instead of the originating PSTN caller DDI
- If an inbound call from the PSTN to the enterprise contains only codecs that are not supported by the enterprise, then the enterprise will return a "488 Not Acceptable Here" response. M-net converts this SIP error message to a SS7 error message and sends it to the PSTN carrier. This should cause some error indication (e.g., fast busy) to be presented to the PSTN caller. However, during the testing no error indication was provided and the call was silently dropped. This issue is not critical since it should only occur if the enterprise and/or M-net have misconfigured codecs.
- When the SIP Trunk is disabled or taken out of service and an inbound call from the PSTN attempts to terminate, IP Office will return a "503 Service Unavailable" response to the M-net SIP platform. This should cause some error indication (e.g. fast busy) to be presented to the PSTN caller. However, during the testing no error indication was provided and the call was silently dropped after multiple reINVITE attempts.
- When the SIP REFER method was used for call transfer of an active PSTN call to another PSTN destination, then after the transfer was complete, unnecessary messaging (in the form of BYE message retransmissions) continued between the enterprise and M-net. The retransmissions from the enterprise continued until a timeout was reached. This behavior did not impact the call and the call was successful.
- T.38 fax transmission is not supported by M-net and therefore was not tested.
- No inbound toll free numbers were tested, however routing of inbound DDI numbers and the relevant number translation was successfully tested.
- Access to Emergency Services was not tested as no test call had been booked by the Service Provider with the Emergency Services Operator. However both three and four digit numbering format replicating Emergency Service's numbering formats was tested successfully.

2.3. Support

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

For technical support on the M-net Premium SIP Trunk Service, please contact M-net at www.M-net.de.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the M-net Premium SIP Trunk. Located at the enterprise site is an Avaya IP Office 500 V2. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 96x1 and 96x0 Series IP Telephones (with H.323 firmware), Avaya 1140e SIP Telephones, Avaya Analogue and Digital Telephones and fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure Avaya IP Office as well as Avaya Communicator for Windows SIP Softphone client.

For security purposes, all Service Provider IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, all IP addresses have been changed to a private format and all phone numbers have been obscured beyond the city code.

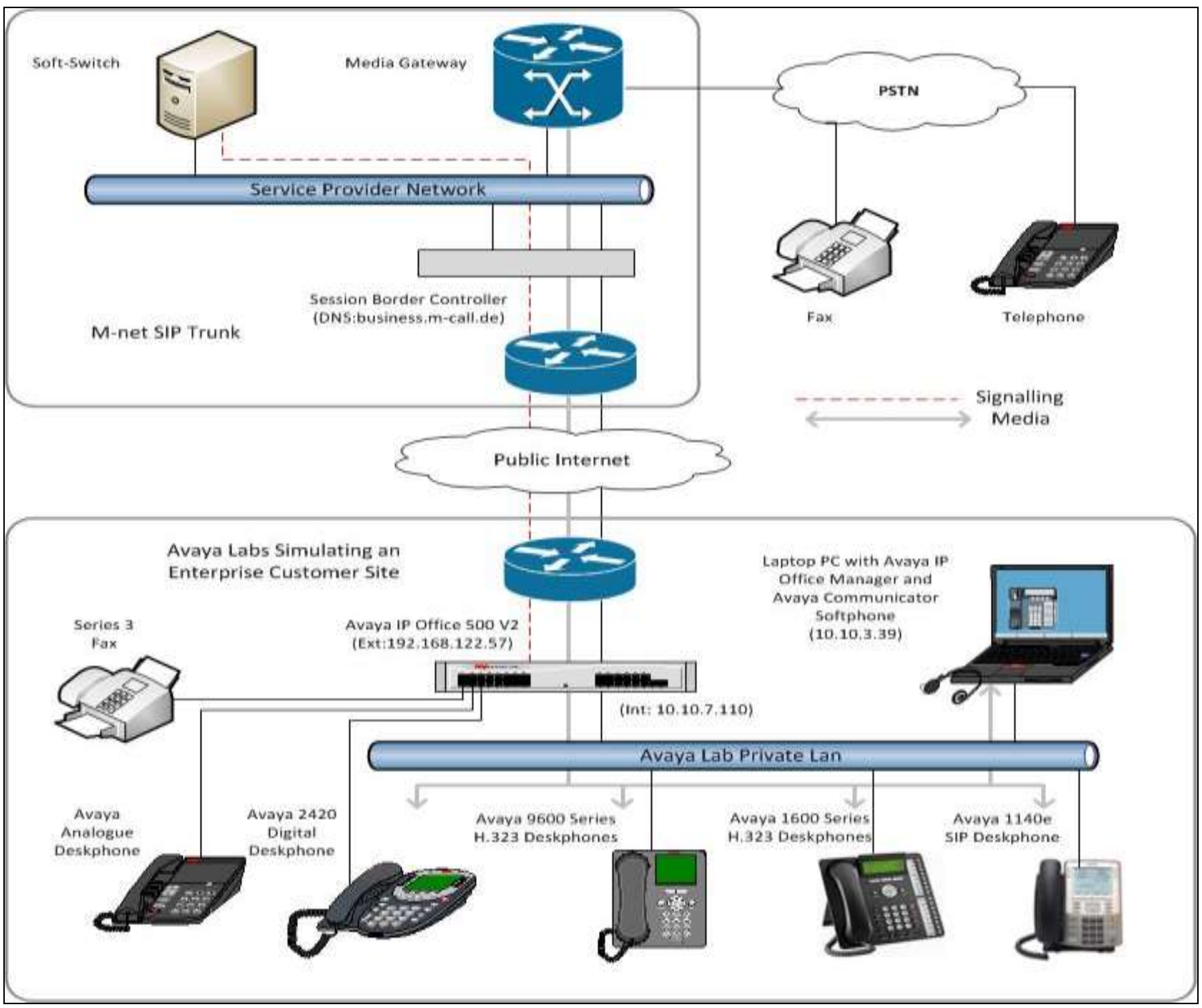


Figure 1: Test setup M-net Premium SIP Trunk to simulated Avaya Enterprise

4. Equipment and Software Validated

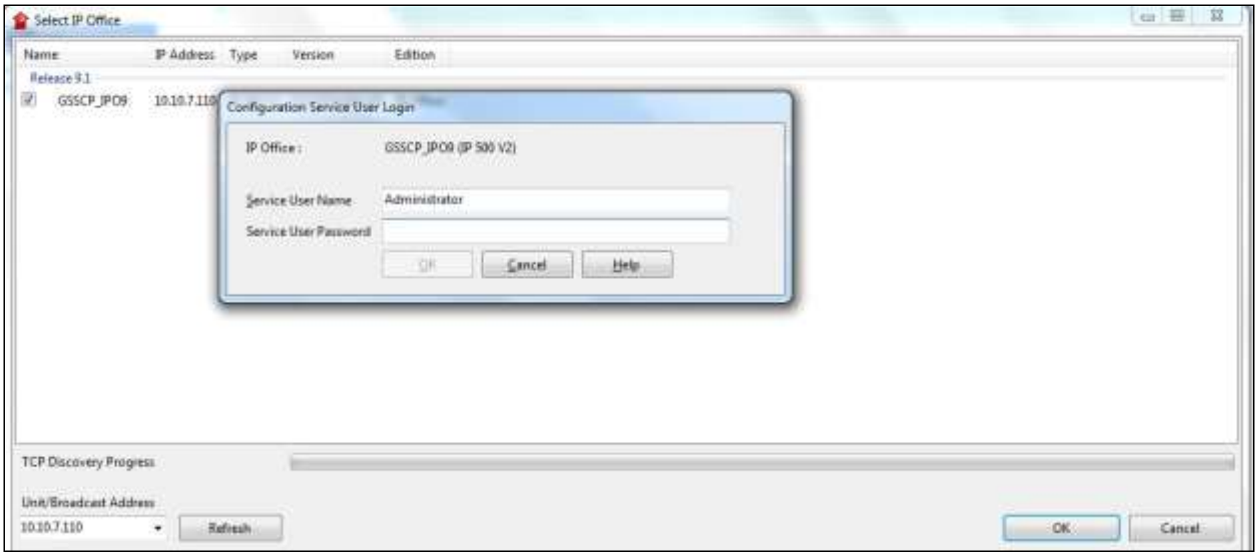
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Version 9.1.5.0 build 145
Avaya Voicemail Pro Client	Version 9.1.5.0
Avaya IP Office Manager	Version 9.1.5.0 build 145
Avaya 1603 Phone (H.323)	1.3.7
Avaya 9611G Series Phone (H.323)	6.4.0
Avaya 9608 Series Phone (H.323)	6.4.0
Avaya Communicator for Windows (SIP)	2.1.1.74
Avaya 1140e (SIP)	FW: 04.04.18.00.bin
Avaya 98390 Analogue Phone	N/A
M-net	
Oracle ACME Packet Net-Net SD 4500 Session Border Controller (SBC)	SCX6.4
Nokia Siemens Networks HiQ4200 Telephone Application Server (TAS)	R14
Nokia Siemens Networks CFX5000 IP Multimedia Subsystem (IMS)	IMS 7.2

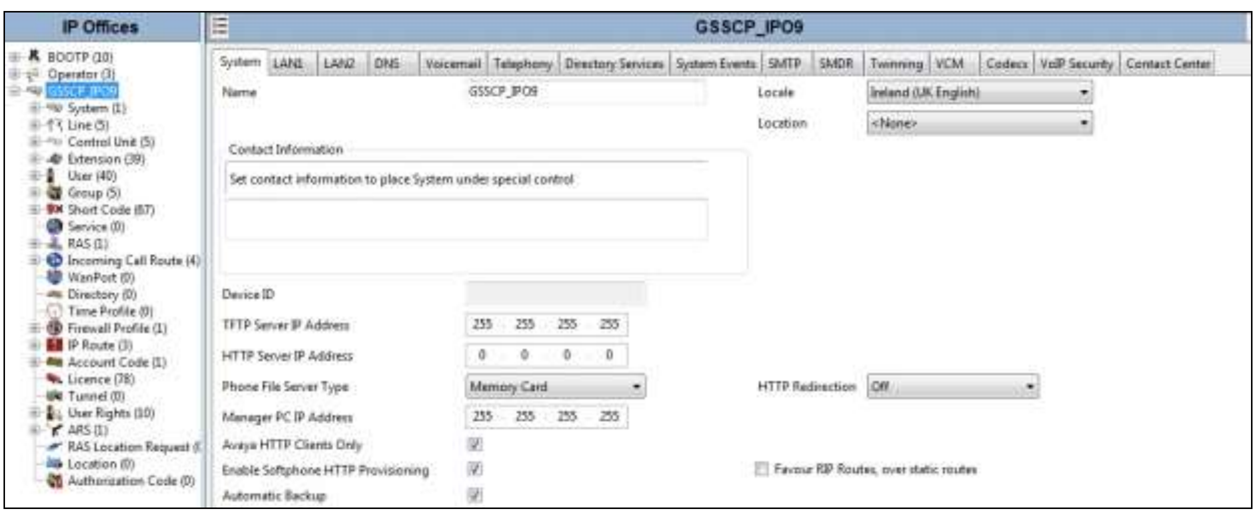
Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition **without T.38 Fax Service**.

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the M-net Premium SIP 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**, select the proper Avaya IP Office system from the pop-up window, and log in with the appropriate credentials.

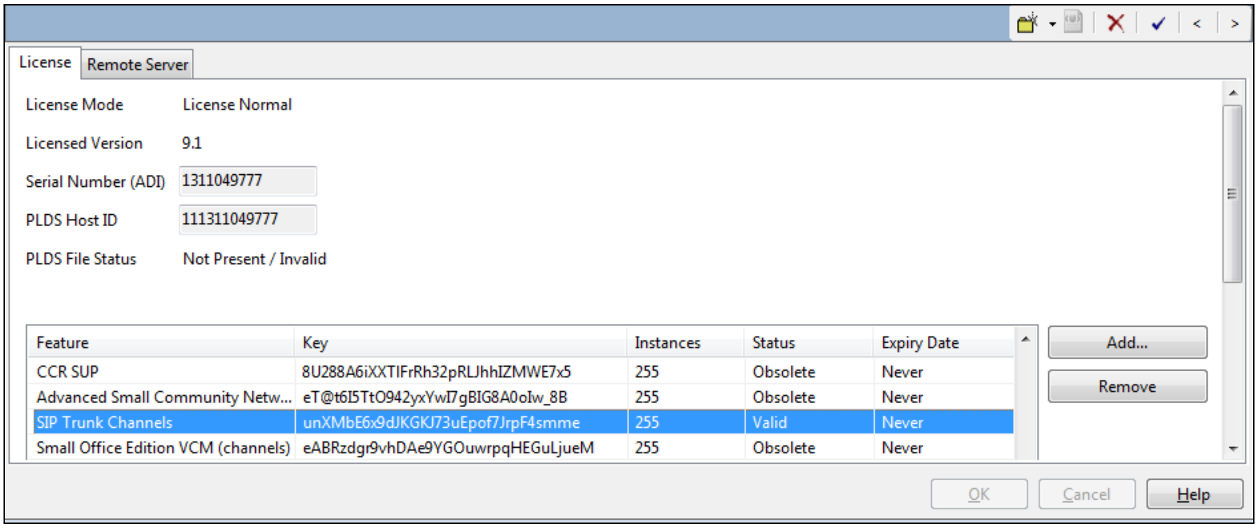


A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All 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.



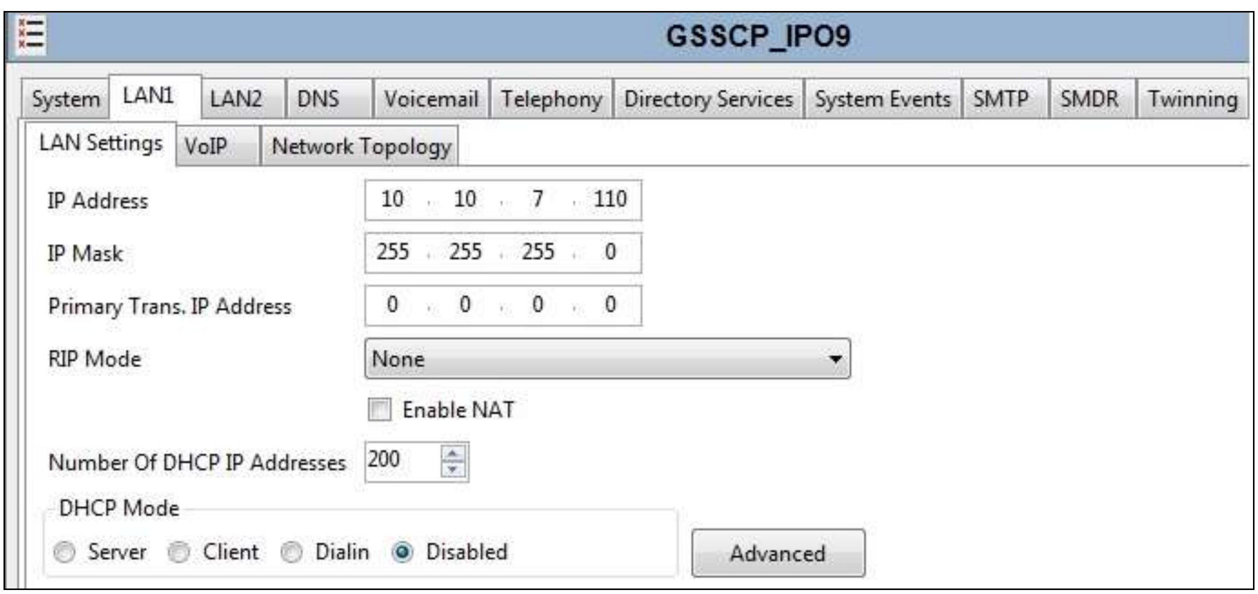
5.1. Verify System Capacity

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 M-net.

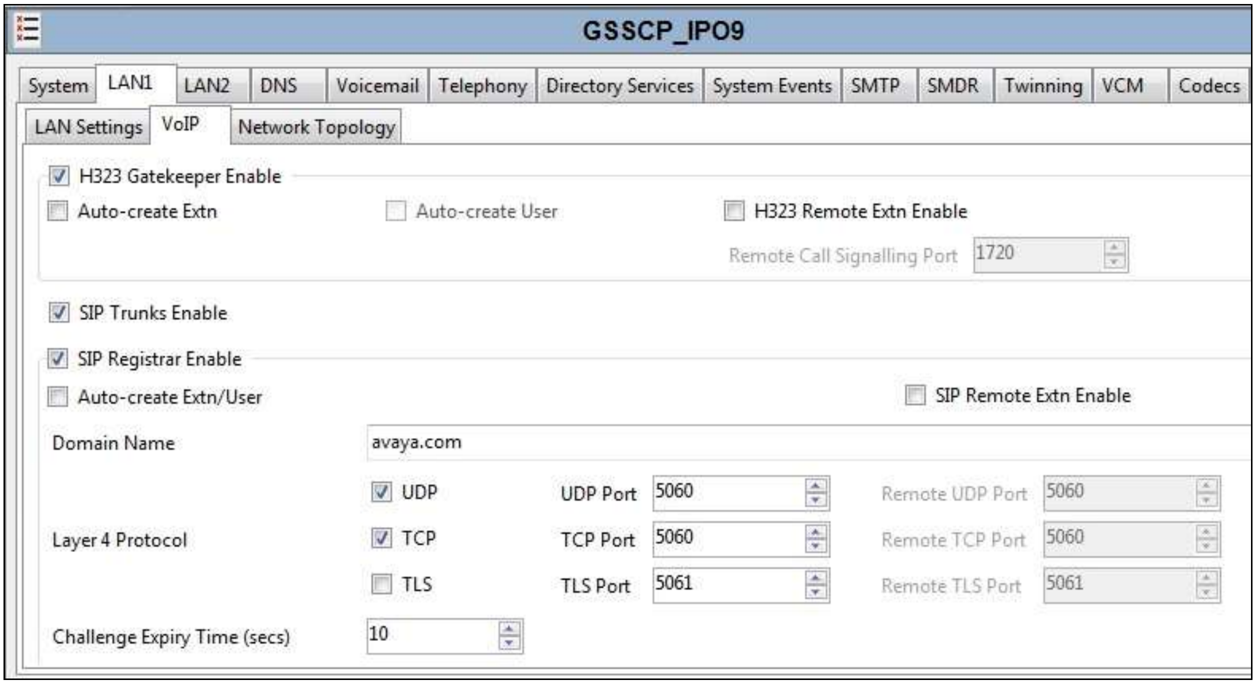


5.2. LAN1 Settings

In the test configuration, the LAN1 port is used to configure the behavior of the services provided by the systems first LAN interface. To access the LAN1 settings, first navigate to **System → GSSCP_IPO9** in the Navigation Pane where GSSCP_IPO9 is the name of the IP Office. Navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the private interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).

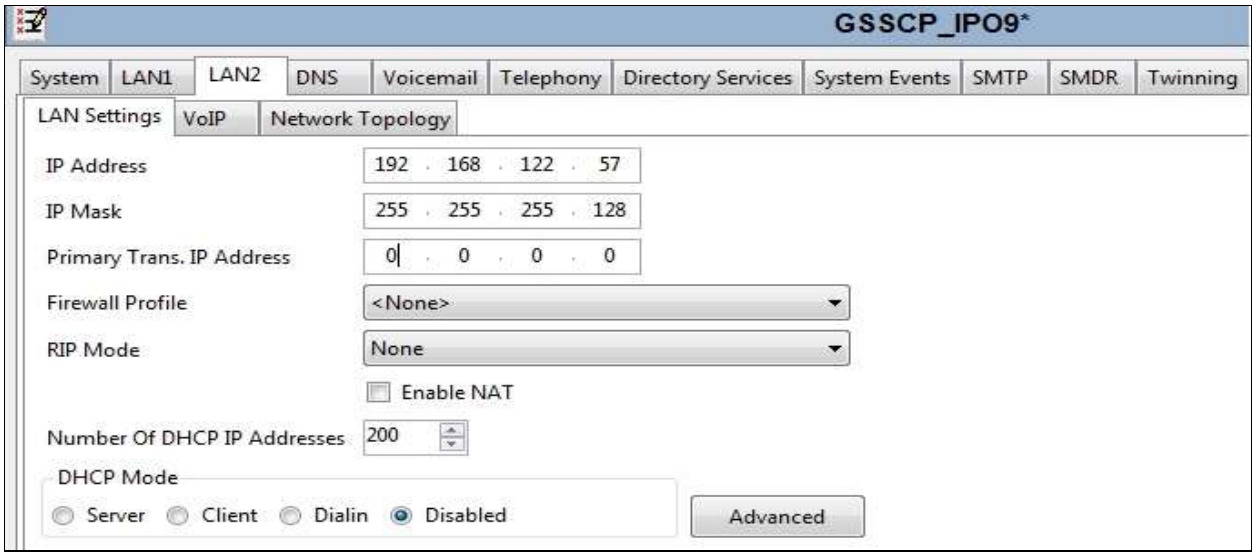


On the **VoIP** tab in the Details Pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If Avaya Communicator along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **Domain Name** has been set to the customer premises equipment domain "**avaya.com**". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN1 IP Address. All other parameters shown are default values.



5.3. LAN2 Settings

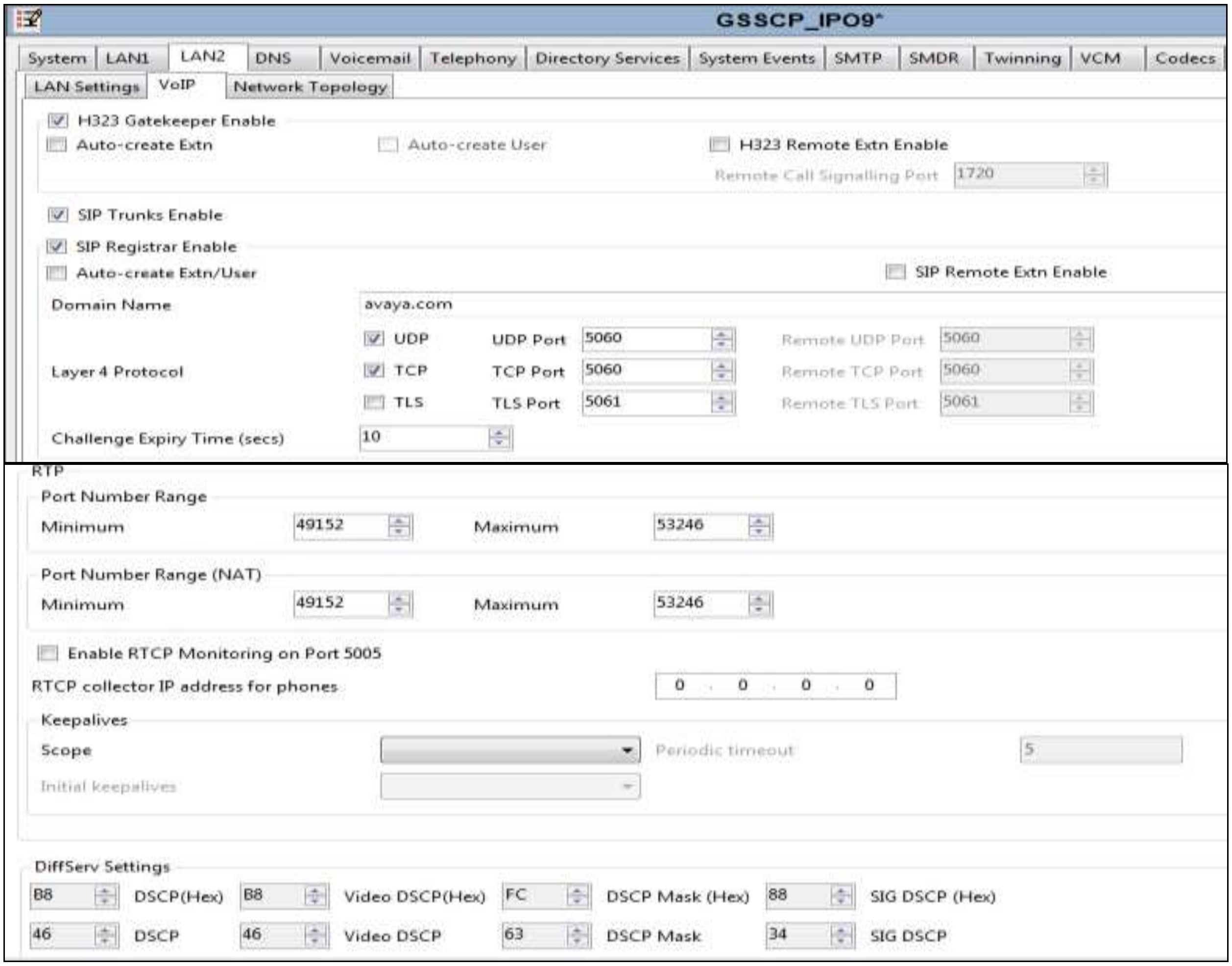
In the test configuration, the LAN2 port was used to connect the Avaya IP Office to the external internet. To access the LAN2 settings, first navigate to **System → GSSCP_IPO9** in the Navigation Pane where GSSCP_IPO9 is the name of the IP Office. Navigate to the **LAN2 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the public interface of the IP Office. All other parameters should be set according to customer requirements. On completion, click the **OK** button (not shown).



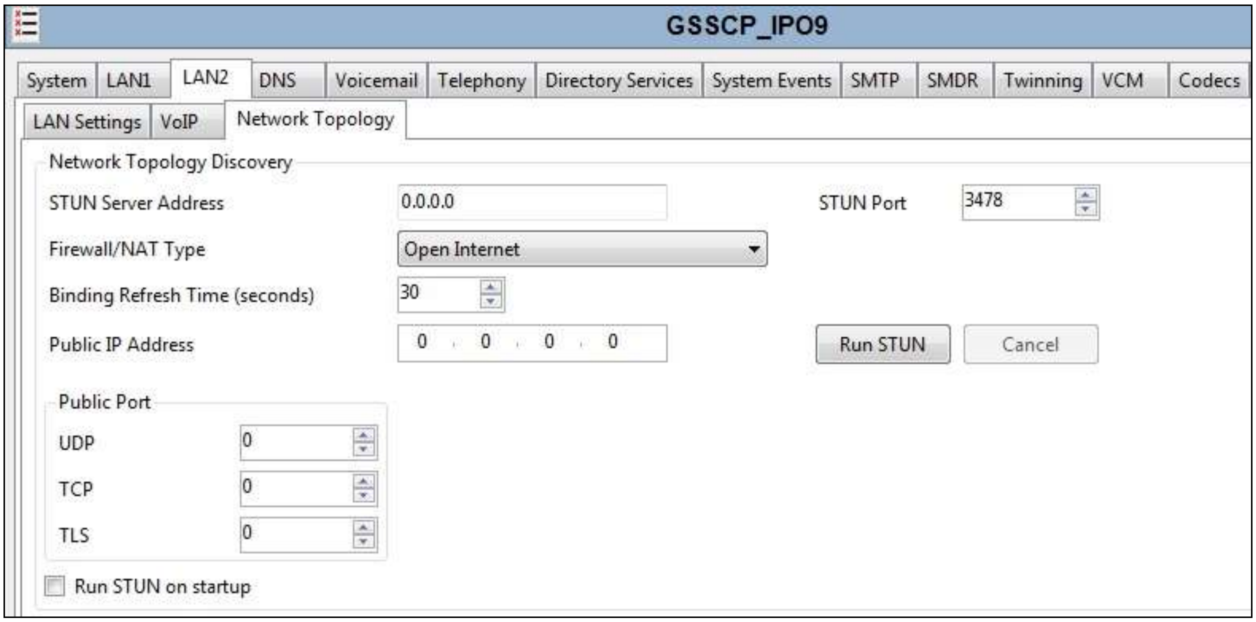
On the **VoIP** tab in the Details Pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If Avaya Communicator along with any other SIP endpoint is to be used, the **SIP Registrar Enable** box must also be checked. The **Domain Name** has been set to the customer premises equipment domain "**avaya.com**". If the **Domain Name** is left at the default blank setting, SIP registrations may use the IP Office LAN2 IP Address. All other parameters shown are default values.

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).

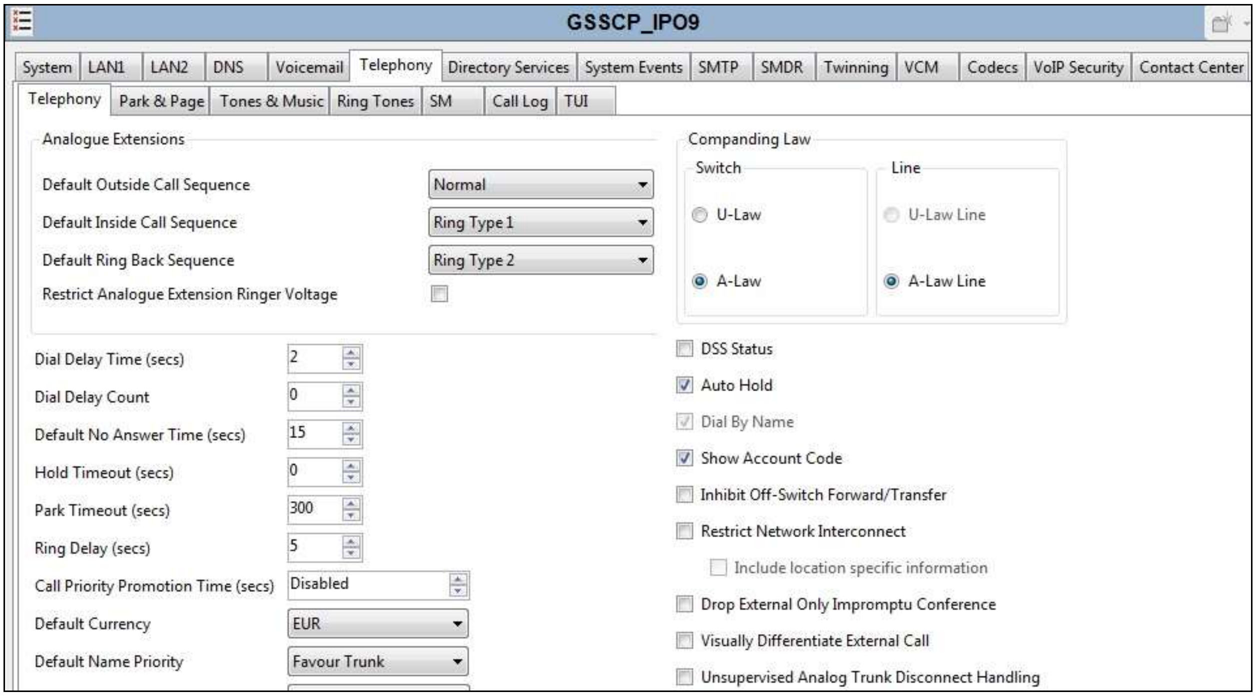

GSSCP_IPO9*
System
LAN1
LAN2
DNS
Voicemail
Telephony
Directory Services
System Events
SMTP
SMDR
Twinning
VCM
Codecs
LAN Settings
VoIP
Network Topology
H323 Gatekeeper Enable
Auto-create Extn
Auto-create User
H323 Remote Extn Enable
Remote Call Signalling Port
1720
SIP Trunks Enable
SIP Registrar Enable
Auto-create Extn/User
SIP Remote Extn Enable
Domain Name
avaya.com
Layer 4 Protocol
UDP
TCP
TLS
UDP Port
5060
TCP Port
5060
TLS Port
5061
Remote UDP Port
5060
Remote TCP Port
5060
Remote TLS Port
5061
Challenge Expiry Time (secs)
10
RTP
Port Number Range
Minimum
49152
Maximum
53246
Port Number Range (NAT)
Minimum
49152
Maximum
53246
Enable RTCP Monitoring on Port 5005
RTCP collector IP address for phones
0 . 0 . 0 . 0
Keepalives
Scope
Periodic timeout
5
Initial keepalives
DiffServ Settings
B8
DSCP(Hex)
B8
Video DSCP(Hex)
FC
DSCP Mask (Hex)
88
SIG DSCP (Hex)
46
DSCP
46
Video DSCP
63
DSCP Mask
34
SIG DSCP

On the **Network Topology** tab, select the **Firewall/NAT Type** from the pulldown menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used as NAT was not required for this configuration, therefore resulting in no requirement for a STUN server. The **Use Network Topology Info** in the **SIP Line** was set to **None** in **Section 5.7.2**. Set **Binding Refresh Time (seconds)** to **30** as requested by M-net. This value is used to determine the frequency at which Avaya IP Office will send SIP OPTIONS messages to the service provider. Default values were used for all other parameters. On completion, click the **OK** button (not shown).



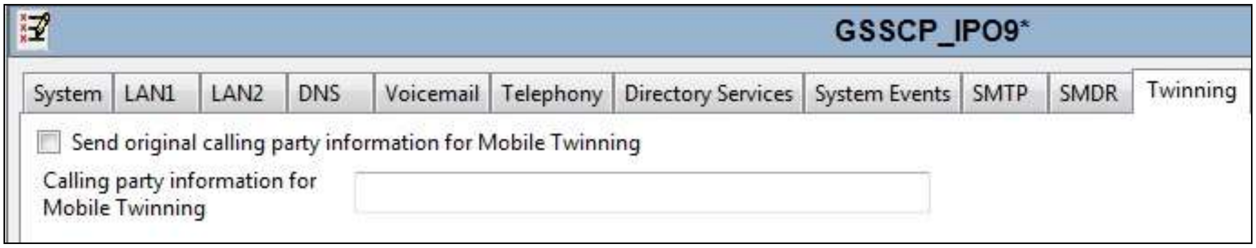
5.4. System Telephony Settings

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).



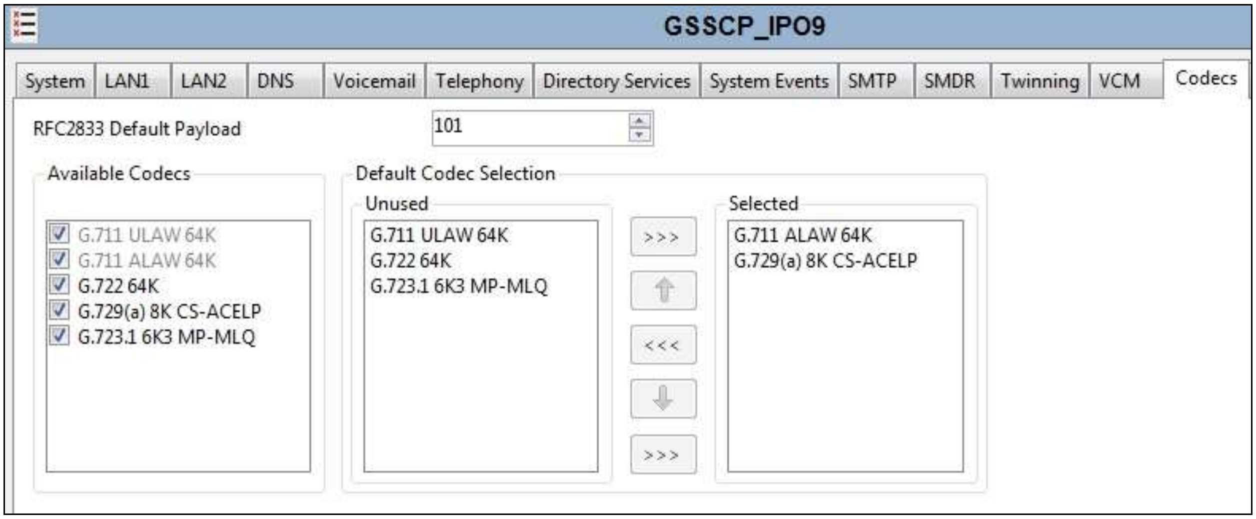
5.5. System Twinning Settings

To view or change Twinning settings, select the **Twinning** tab as shown in the following screen. The **Send original calling party information for Mobile Twinning** box is not checked, and the **Calling party information for Mobile Twinning** is left blank in the reference configuration. With this configuration, the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the M-net SIP Trunk.



5.6. 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**, and **G.729(a) 8K CS-ACELP** were the supported codecs used for testing.



5.7. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the M-net Premium service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.7.1** 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 5.7.2**.

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

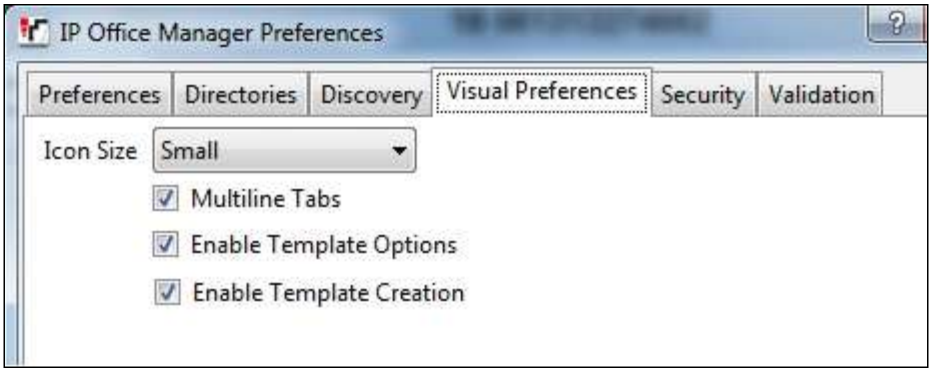
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.7.2**.

5.7.1. SIP Line From Template

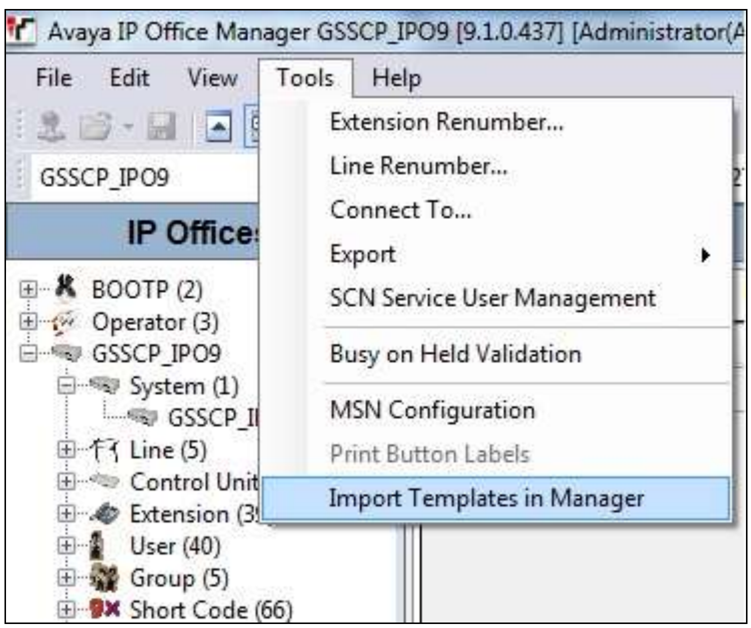
DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer's environment.

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed. Rename the template file to **AF_M-net_SIPTrunk.xml**. The file name is important in locating the proper template file in **Step 5**.

2. Verify that template options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Verify that the box is checked next to **Enable Template Options**. Click **OK**.

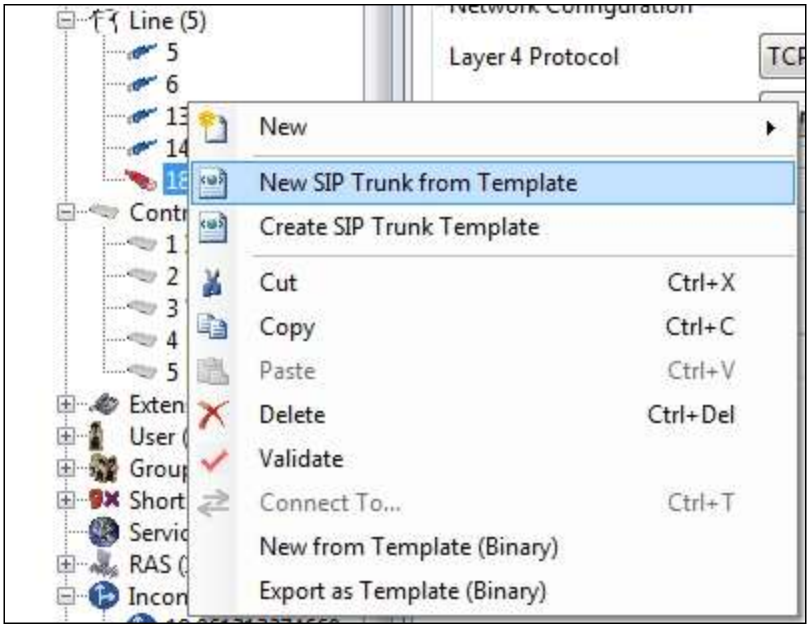


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**.



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

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



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

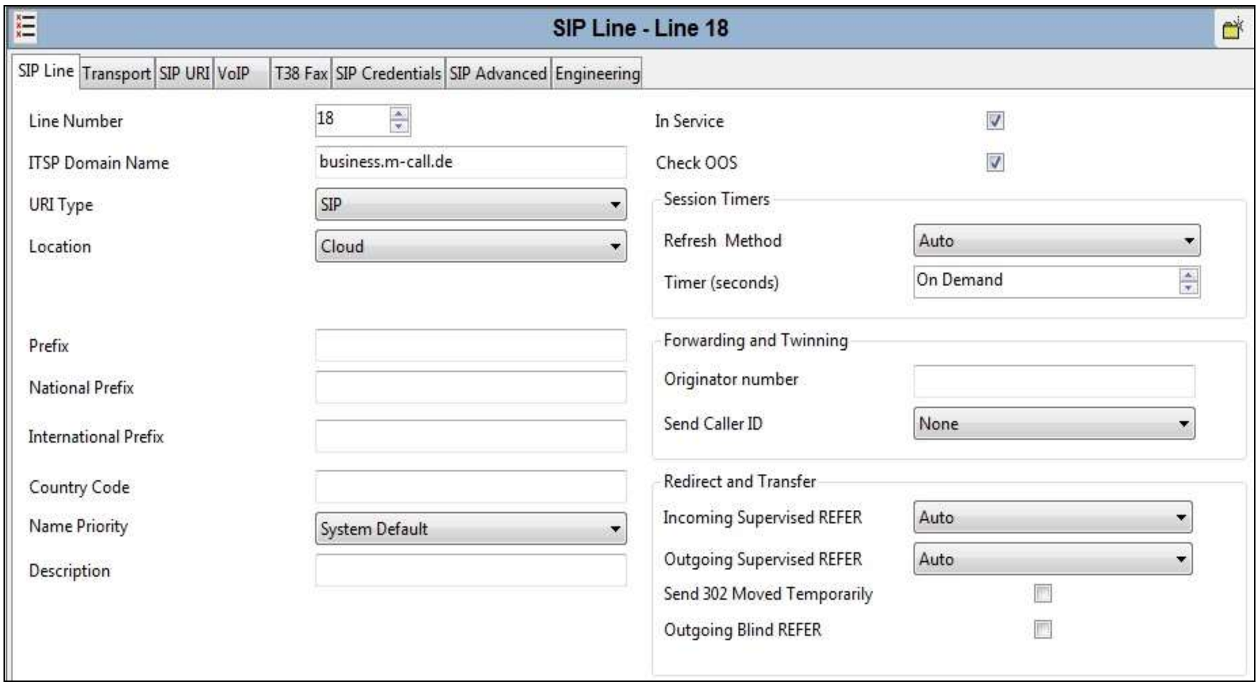
6. Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Section 5.7.2**.

5.7.2. Manual SIP Line Configuration

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

- Set **ITSP Domain Name** to the M-net DNS address.
- Ensure the **In Service** box is checked.
- Ensure the **Check OOS** box is checked.
- Set **Refresh Method** to **Auto**.
- Set **Send Caller ID** to **None**.
- Set **Incoming Supervised REFER** and **Outgoing Supervised REFER** to **Auto**.
- Default values may be used for all other parameters.

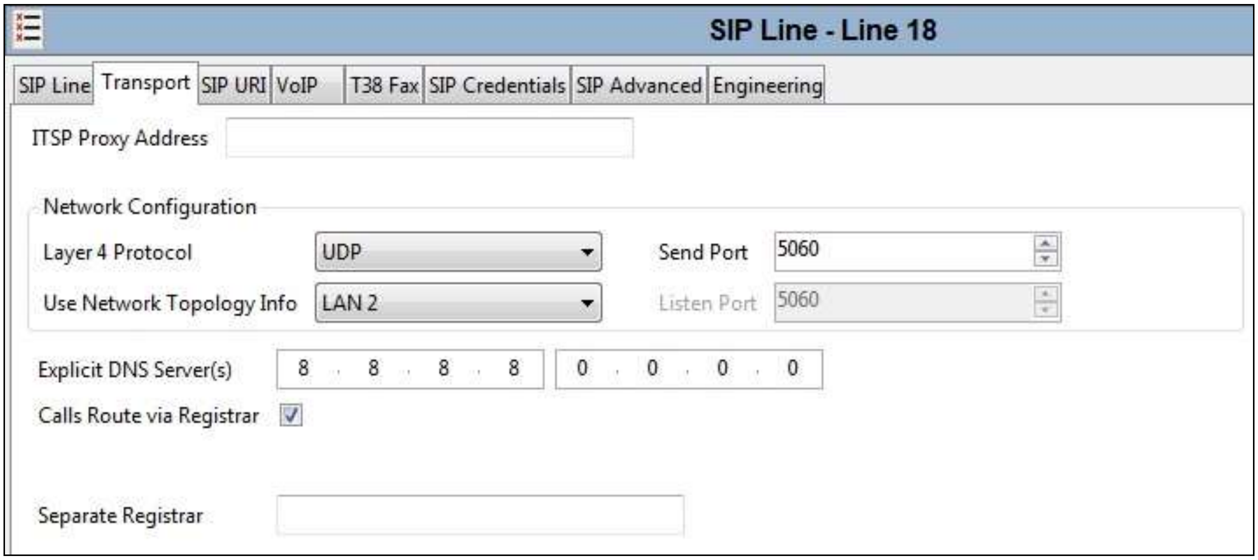
On completion, click the **OK** button (not shown).



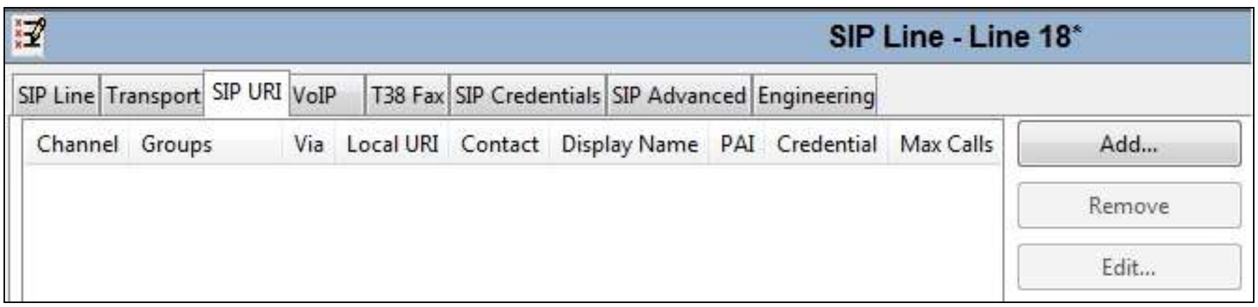
Select the **Transport** tab and set the following:

- Leave **ITSP Proxy Address** blank as none was used in this configuration.
- Set **Layer 4 Protocol** to **UDP**.
- Set **Send Port** to **5060** and **Listen Port** to **5060**.
- Set **Network Topology Info** to **LAN 2**.

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

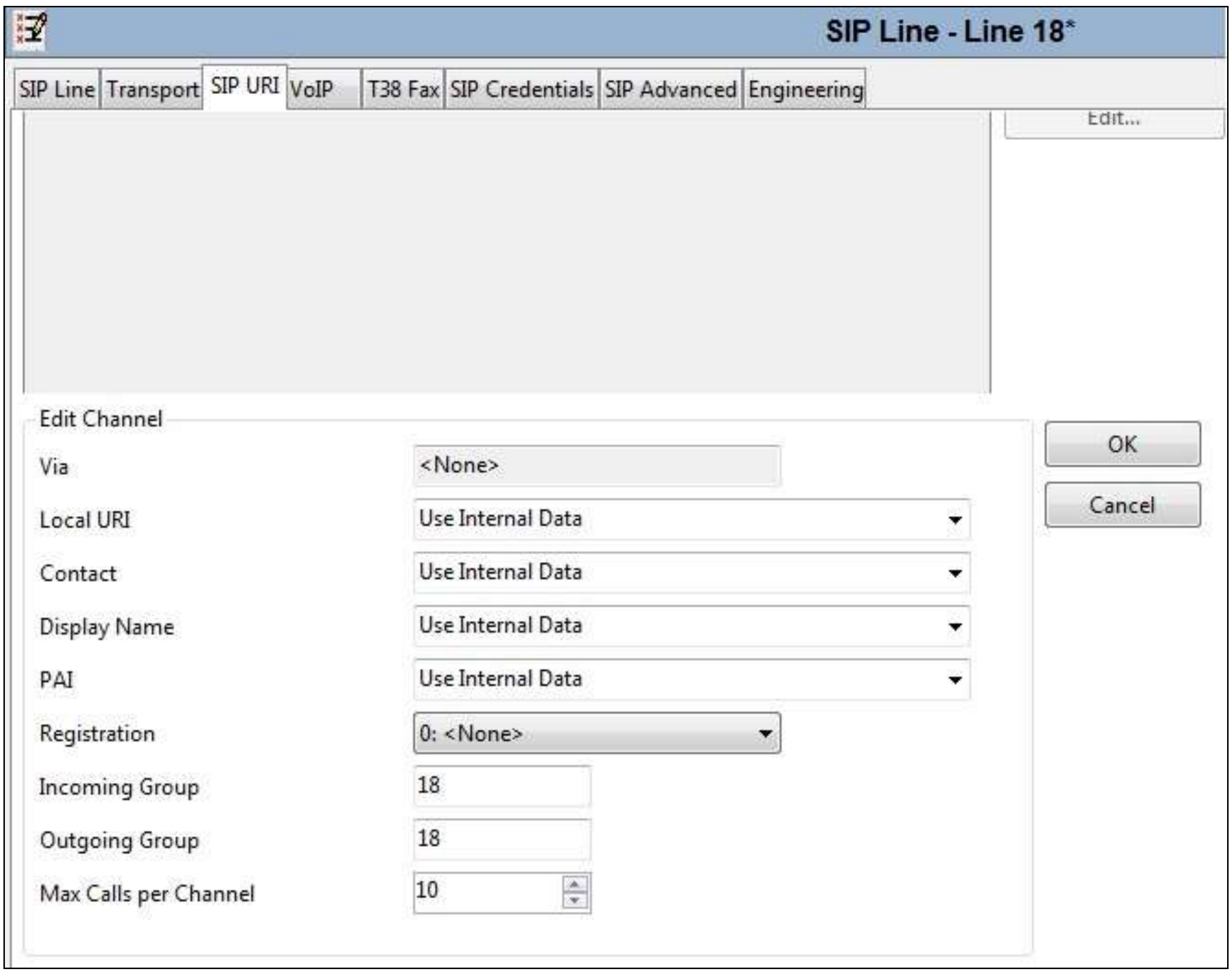


After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, first select the **SIP URI** tab. Click the **Add** button and the **New Channel** area will appear at the bottom of the pane.



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.

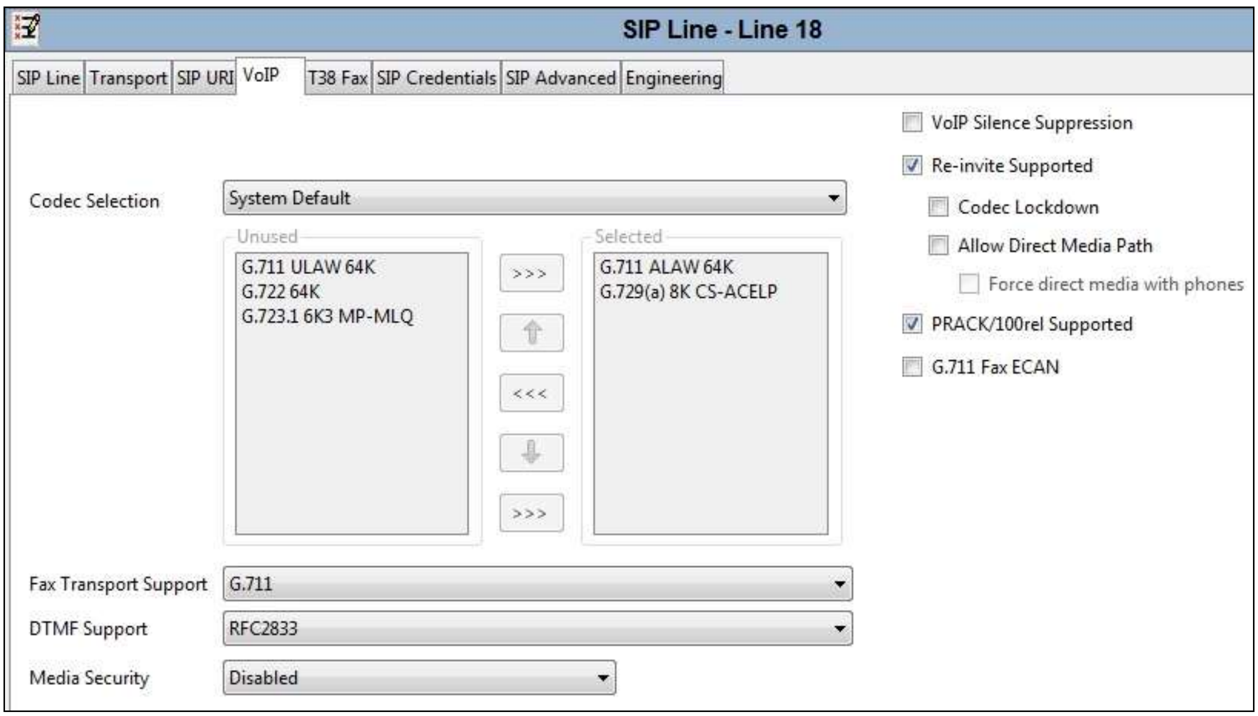
- Set **Local URI, Contact, Display Name and PAI** to **Use Internal Data**. This setting allows calls on this line whose SIP URI matches the number set in the **SIP** tab of any **User** as shown in **Section 5.9**.
- For **Registration**, select **0: <None>** from the pull-down menu.
- Associate this 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.



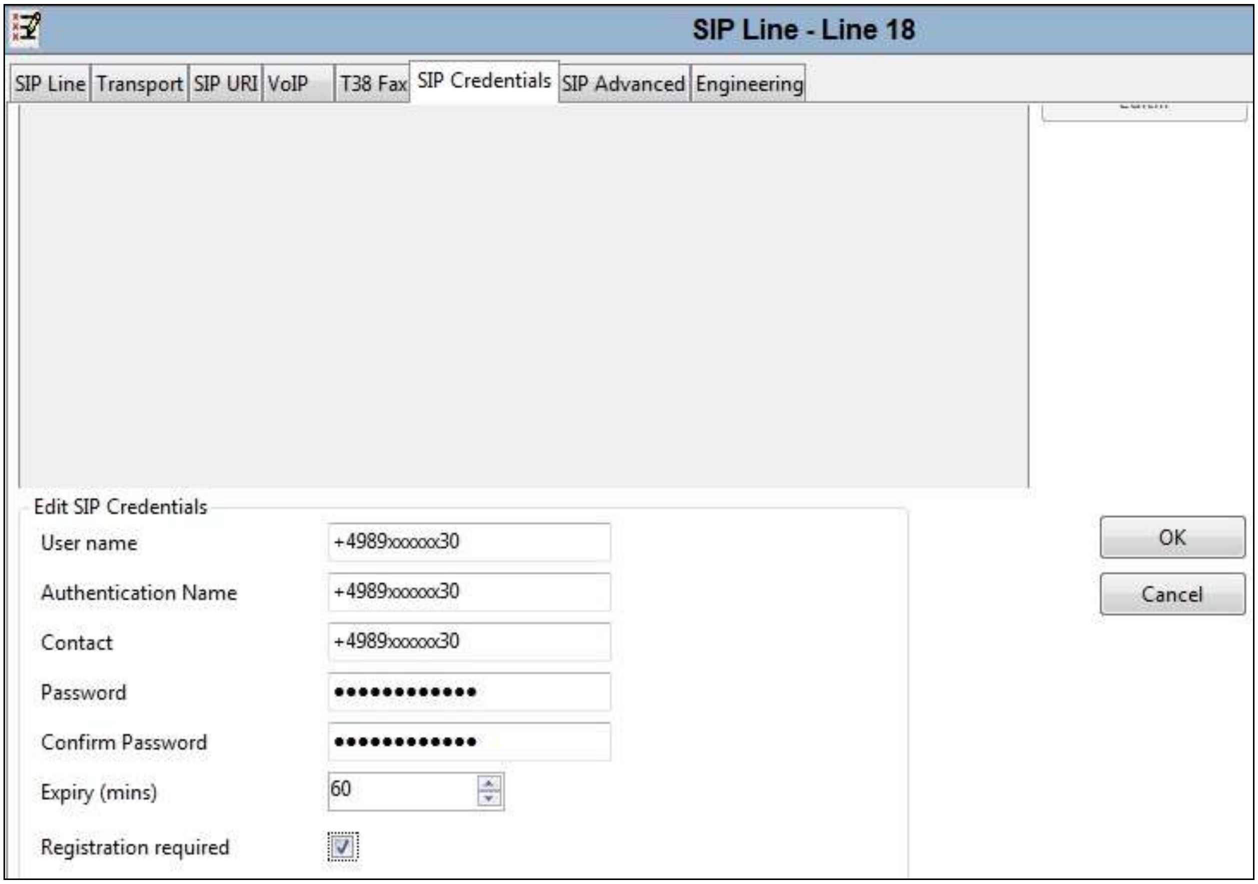
Select the **VoIP** tab to set the Voice over Internet Protocol parameters of the SIP line. Set the parameters as shown below:

- Select **System Default** from the drop-down menu.
- Select **G.711 ALAW 64K**, and **G.729(a) 8K CS-ACELP** codecs.
- Set the **Fax Transport Support** box to **G.711** as this is the preferred method of fax transmission for M-net.
- 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.
- Check **PRACK/100rel Supported** to advertise the support for provisional responses and Early Media to the M-net network.

Default values may be used for all other parameters.

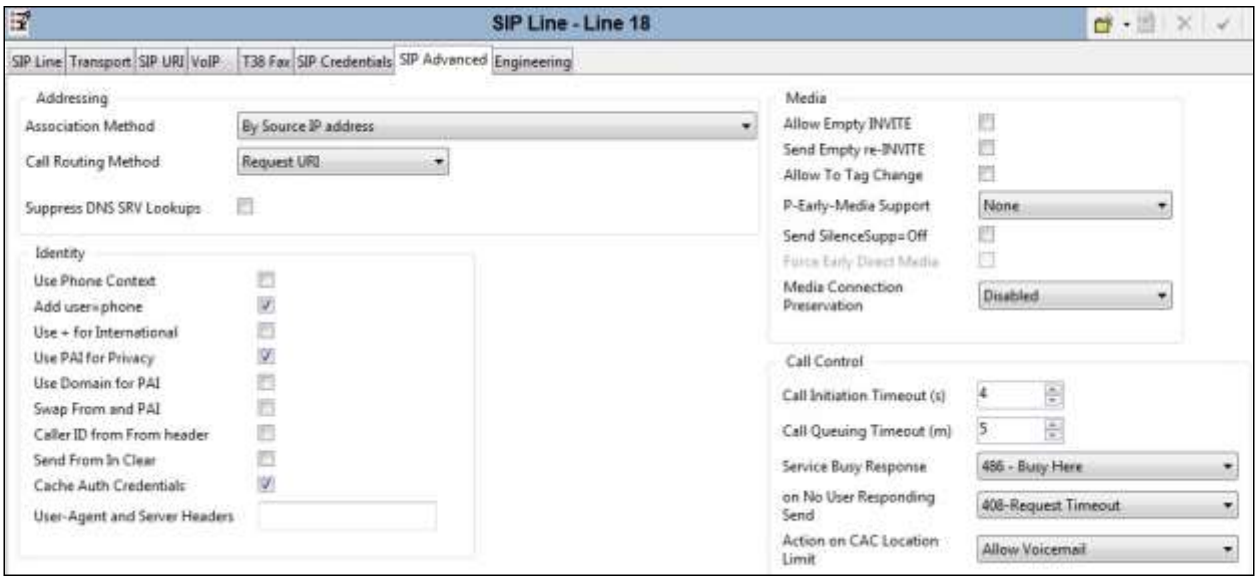


Select the **SIP Credentials** tab to administer registration details provided by M-net. This allows the SIP Trunk to authenticate to the M-net SIP Trunk service. Choose **Add** (not shown) and enter the registration credentials provided by M-net as shown below. Click the **OK** button to complete the SIP line administration.



Select the **SIP Advanced** tab. For outbound calls with privacy enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with “anonymous”. Avaya IP Office can be configured to use the P-Preferred-Identity (PPI) or P-Asserted-Identity (PAI) header to pass the actual calling party information for authentication and billing purposes. By default, Avaya IP Office will use the PPI header for privacy. For the compliance test, PAI was used for the purposes of privacy.

To configure Avaya IP Office to use the PAI header for privacy calls, on the **SIP Advanced** tab, check **Use PAI for Privacy**. Check **Add user=phone** and **Use + for International** as M-net require outgoing international calls to be presented in E.164/International format. All other fields retained their default values.



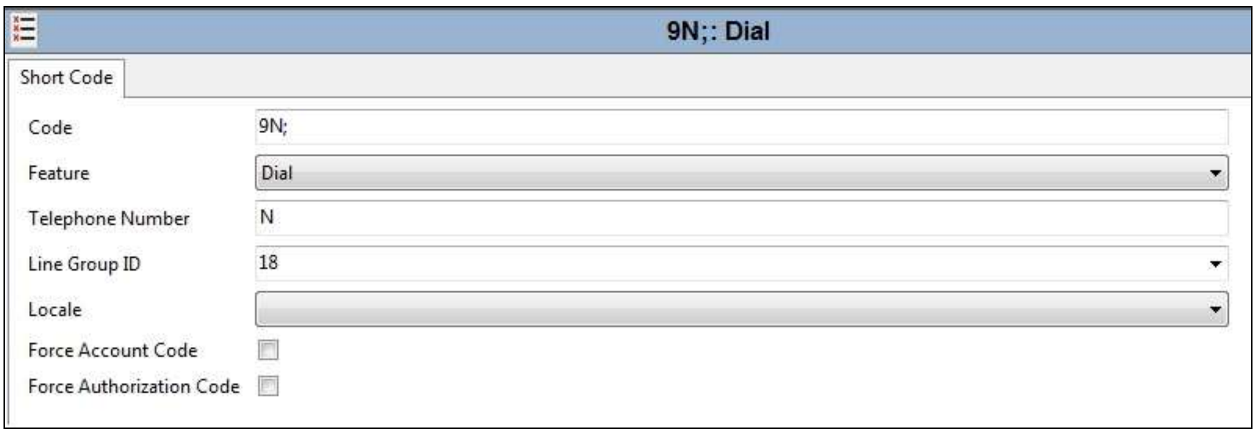
Note: It is advisable at this stage to save the configuration as described in **Section 5.11**.

5.8. ShortCodes

Define a short code to route outbound traffic to the SIP line and route incoming calls from mobility extensions to access Feature Name Extensions (FNE) hosted on IP Office. 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 **9N**; which will be invoked when the user dials 9 followed by the dialed number.
- Set **Feature** to **Dial**. This is the action that the short code will perform.
- Set **Telephone Number** to **N**. The **Telephone Number** field is used to construct the Request URI and To Header in the outgoing SIP INVITE message.
- Set the **Line Group Id** to the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.7.2**.

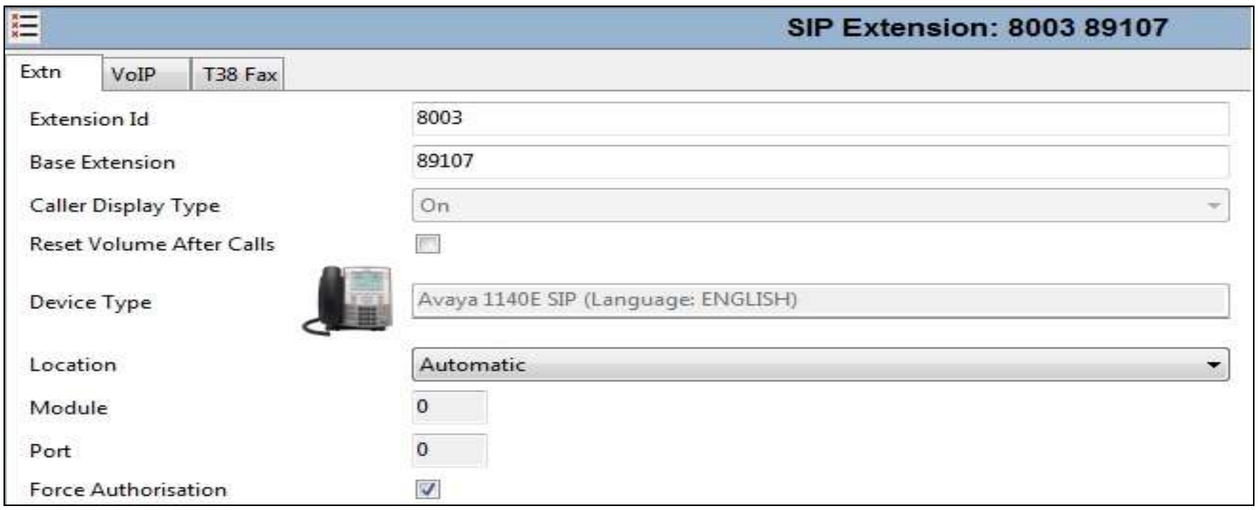
On completion, click the **OK** button (not shown).



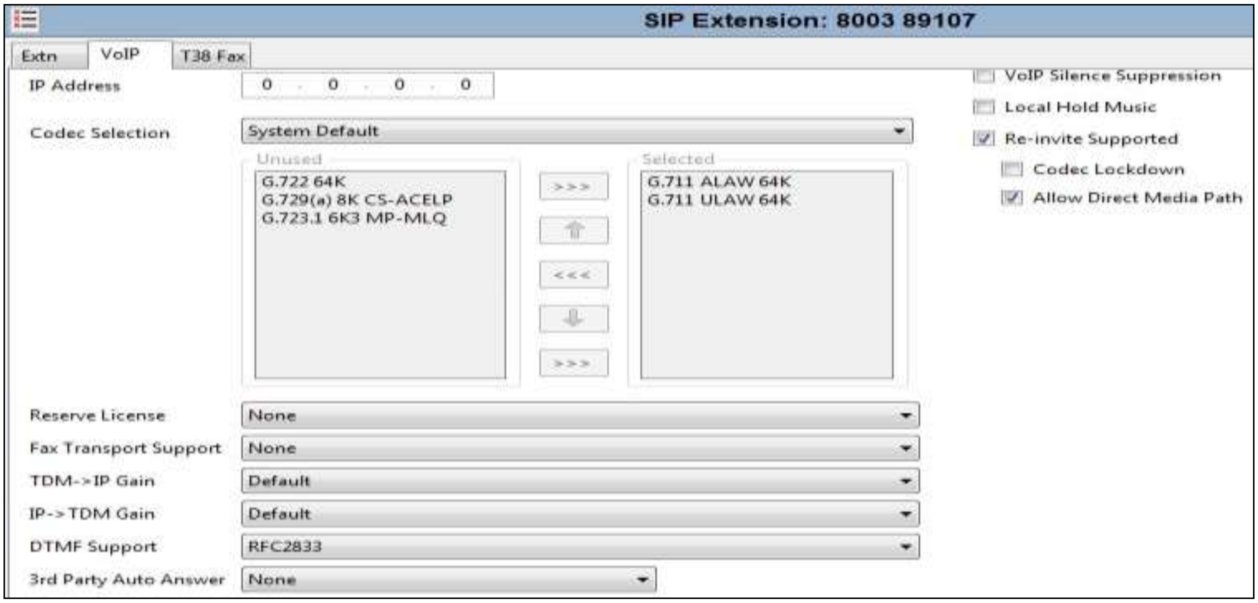
5.9. User and Extensions

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

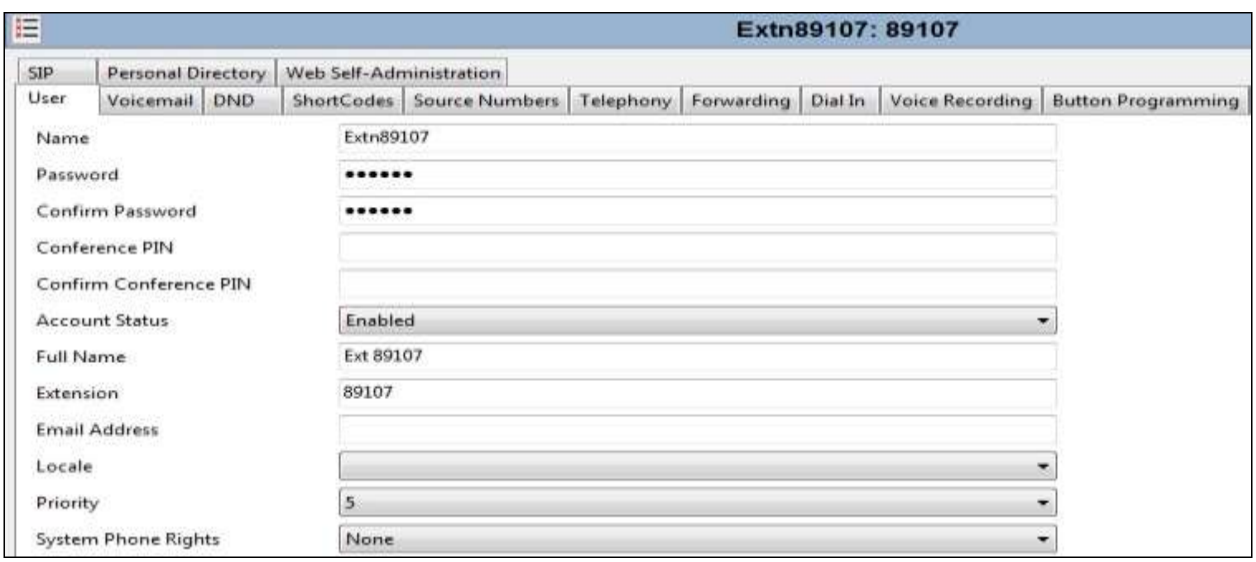
A new SIP extension may be added by right-clicking on **Extension** (not shown) in the Navigation pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the group pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya 1140E. The **Base Extension** field is populated with **89107**, the extension assigned to the Avaya 1140E. Ensure the **Force Authorisation** box is checked.



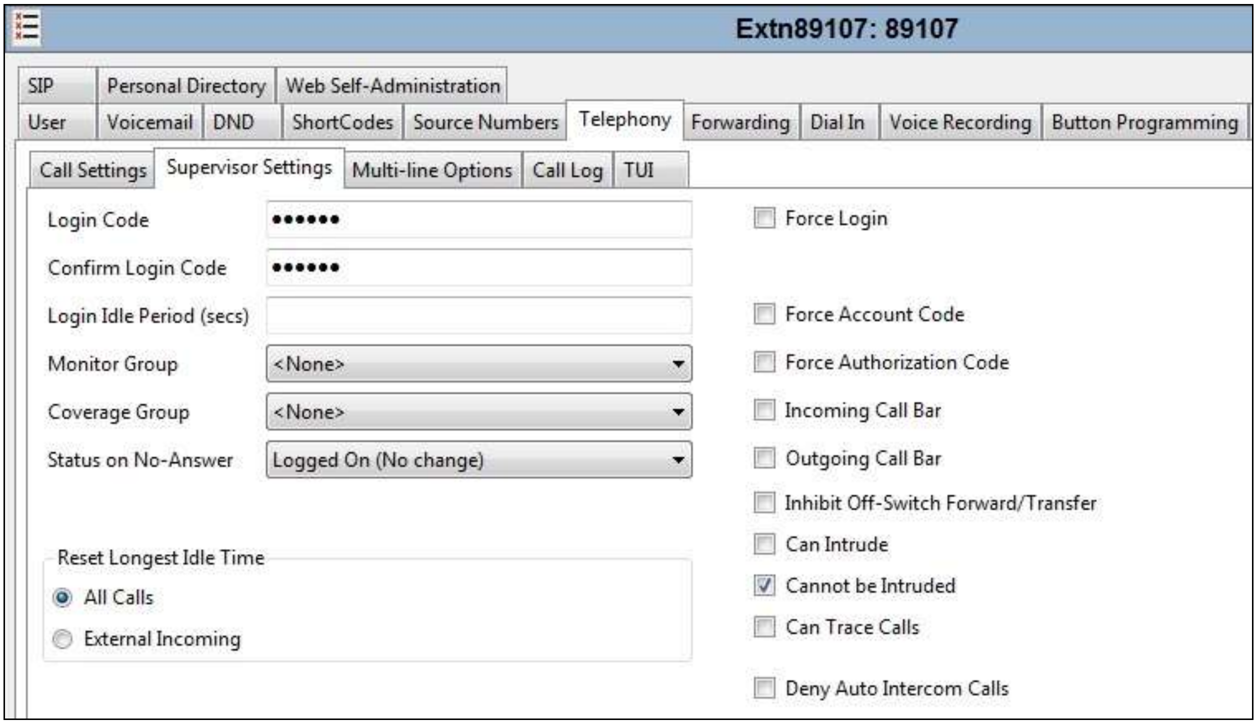
The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank or populated with a static IP address. The new **Codec Selection** parameter may retain the default setting **System Default** to follow the system configuration shown in **Section 5.6**. Alternatively, **Custom** may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.



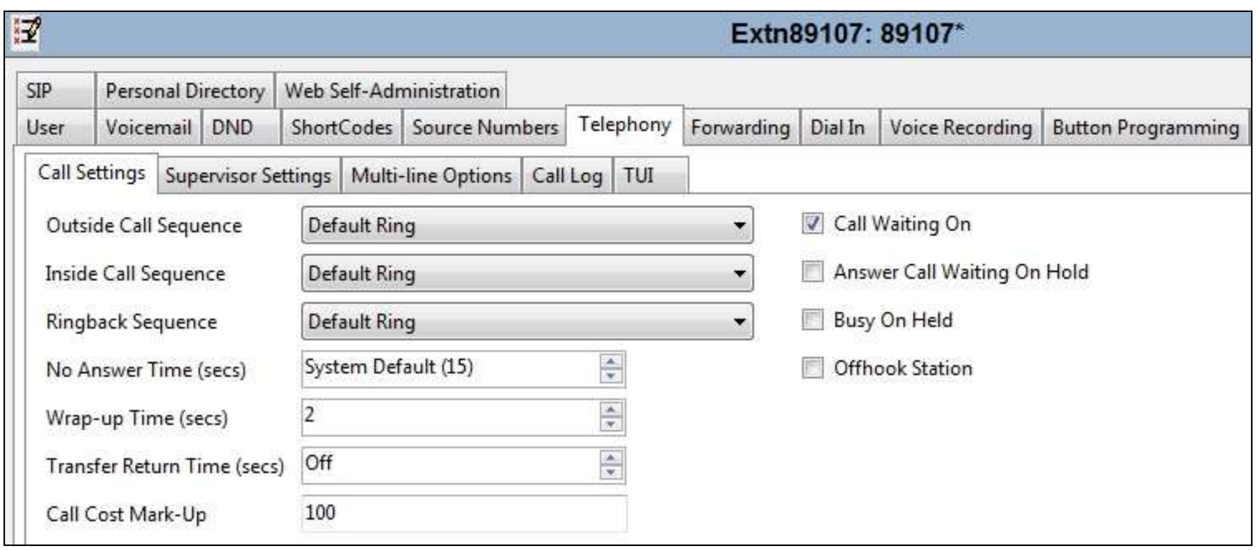
To add a User, right-click on **User** in the Navigation pane, and select **New**. To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane. Configure the SIP parameters for each User that will be placing and receiving calls via the SIP line defined in **Section 5.7.2**. To configure these settings, select the **User** tab if any changes are required. The example below shows the changes required to use Avaya 1140E which was used in test.



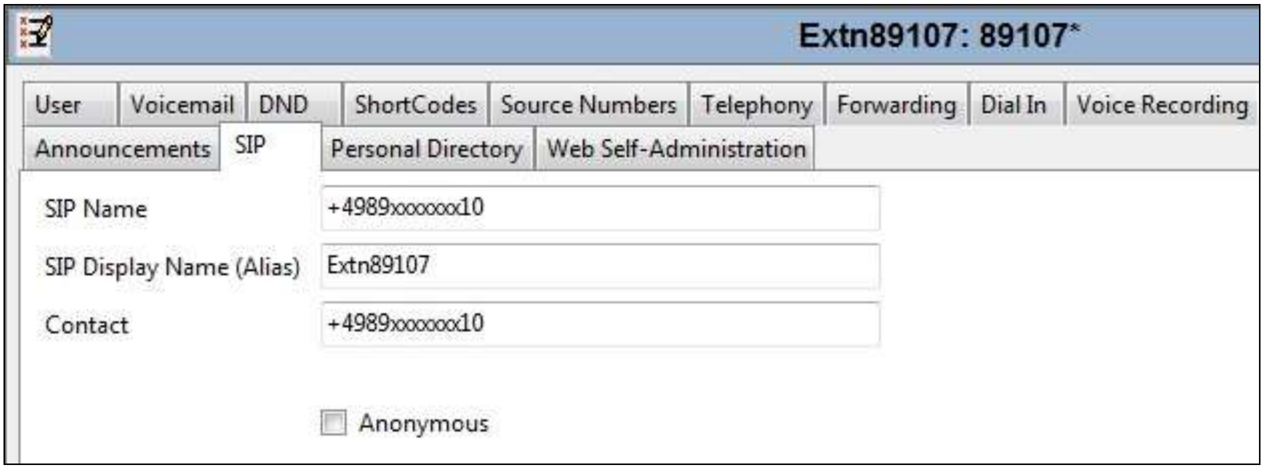
Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya 1140E telephone user as the login password.



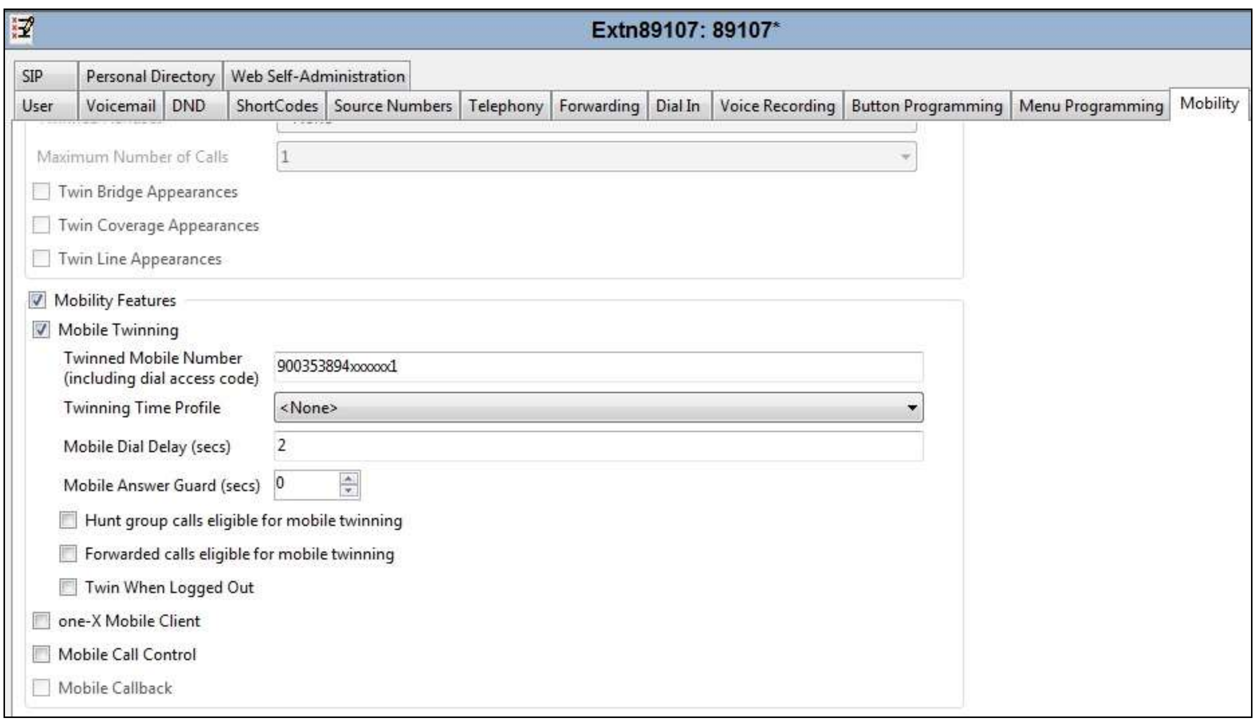
Remaining in the **Telephony** tab for the user, select the **Call Settings** tab as shown below. Check the **Call Waiting On** box to allow multiple call appearances and transfer operations.



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 5.7.2**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from M-net.



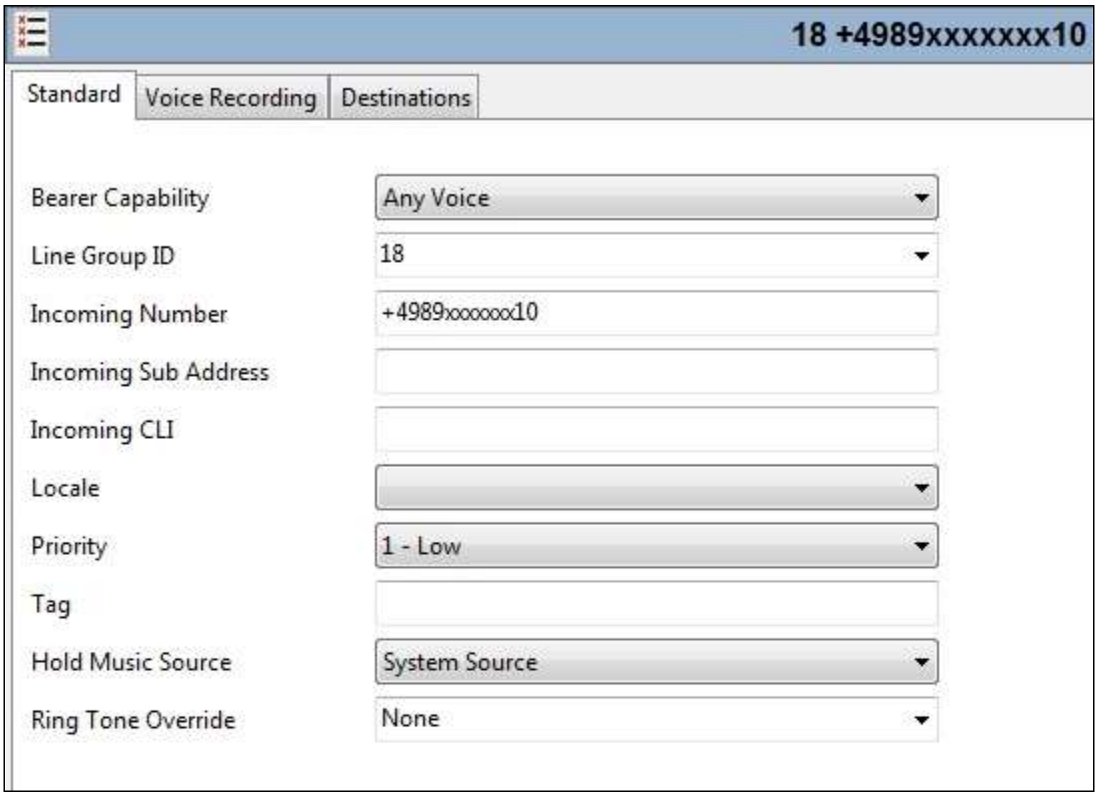
The following screen shows the Mobility tab for user 89107. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.



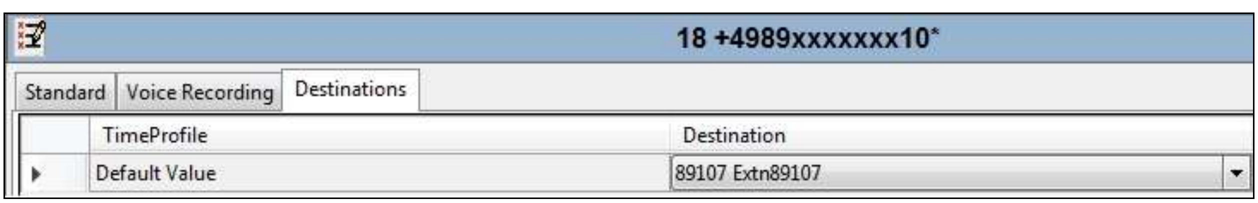
5.10. Incoming Call Routing

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 5.7.2**.
- 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.

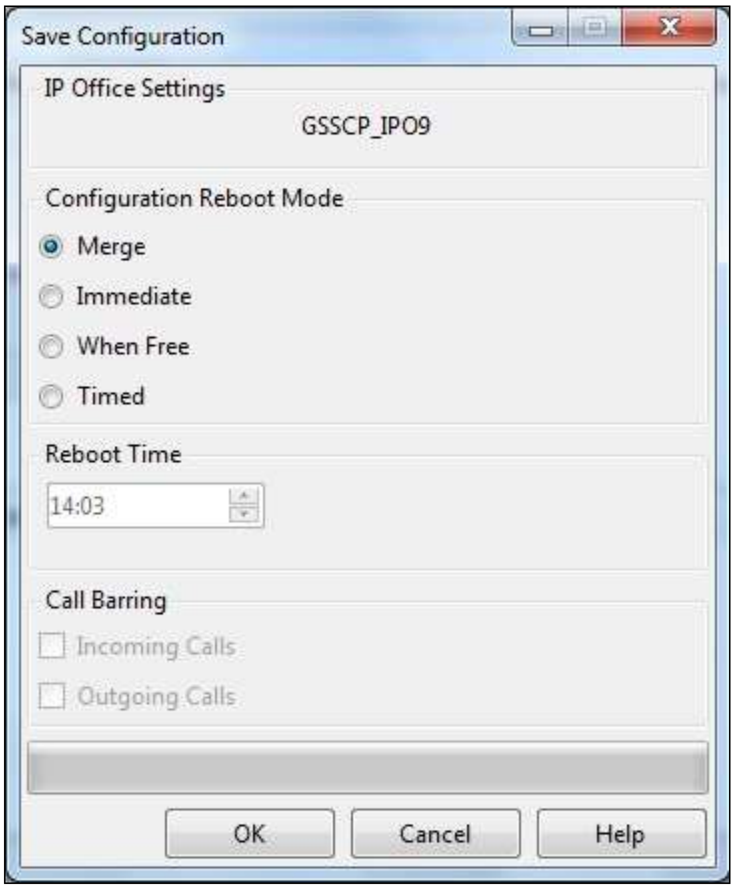


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+4989xxxxxxx10** on line 18 are routed to extension 89107.



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. A screen like the one shown below is displayed where the system configuration has been changed and needs to be saved on the system. **Merge, Immediate, When Free** or **Timed** is shown under the **Configuration Reboot Mode** column, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to save the configuration



6. M-net Premium SIP Trunk Configuration

The configuration of the M-net equipment used to support M-net's SIP trunk is outside of the scope of these Application Notes and will not be covered. To obtain further information on M-net equipment and system configuration please contact an authorized M-net representative.

7. Verification Steps

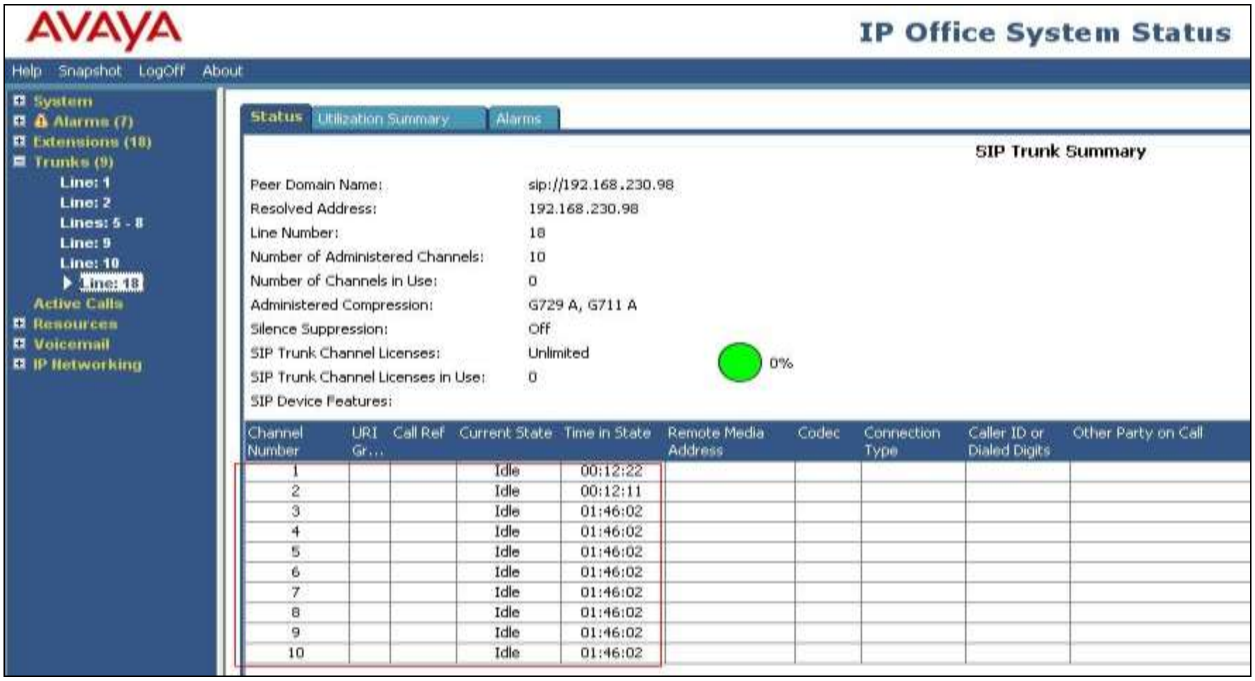
This section includes steps that can be used to verify that the configuration has been done correctly.

7.1. SIP Trunk status

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

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

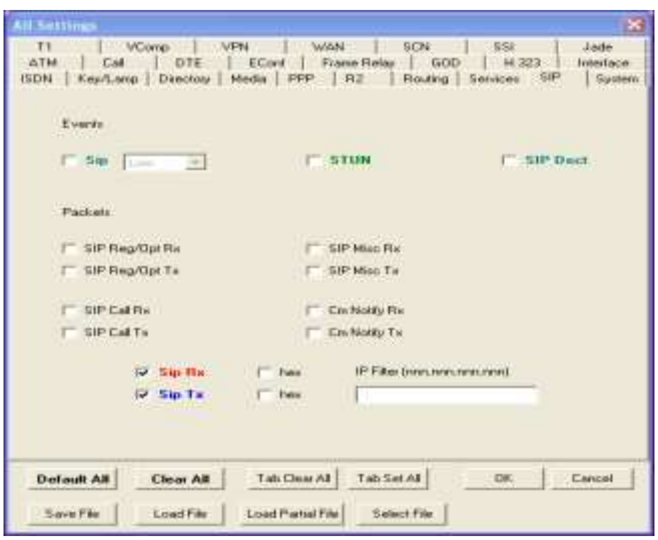
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the **Current State** as being **idle** and **Time in State** if the Trunk is operational. The IP address has been changed for security purposes.



7.2. Monitor

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

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



As an example, the following shows a portion of the monitoring window of a SIP handset attempting registration to IP Office.

8. Conclusion

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

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

9. Additional References

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

[1] *Deploying Avaya IP Office Platform IP500 V2*, December 2015.
[2] *Administrating Avaya IP Office with Manager*, December 2015.
[3] *Administrating Avaya IP Office Voicemail Pro*, January 2016.
[4] *Using IP Office System Status,* August 2015.
[5] *Administering Avaya Communicator for Windows*, October 2015
[6] *Avaya IP Office Knowledgebase,* http://marketingtools.avaya.com/knowledgebase

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.