DevConnect Program

Application Notes for Configuring Avaya IP Office Release 11.1 with Avaya Session Border Controller Release 10.1 to support M-net Premium SIP Trunk Service using TLS/SRTP – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between the M-net Premium SIP Trunk Service and Avaya IP Office R11.1 with Avaya Session Border Controller R10.1 using Transport Layer Security (TLS) for signalling and Secured Real-Time Protocol (SRTP) for media encryption.

The M-net Premium SIP Trunk provides PSTN access via a SIP trunk connected to the M-net Premium Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analog 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.

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1. Introduction

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between M-net Premium SIP Trunk service and Avaya IP Office with Avaya Session Border Controller (Avaya SBC) using TLS for signalling and SRTP for media encryption.

Avaya IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service.

The Avaya SBC is the point of connection between Avaya IP Office and M-net Premium SIP Trunk service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signalling for interoperability.

M-net Premium SIP Trunk service provides PSTN access via a SIP trunk connected to the M-net network as an alternative to legacy Analog 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 and Avaya SBCE 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

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 Analog 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 Analog telephones at the enterprise.
- All outbound PSTN calls were routed from the enterprise across the SIP trunk to the Service Provider.
- Incoming and Outgoing PSTN calls to/from Avaya Workplace Client for Windows soft phone.
- Calls using G.722 and G.711A codecs.
- Fax calls to/from Client a group 3 fax machine to a PSTN-connected fax machine using T.38 Fallback transmission.
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls.
- 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, call mute, transfer, and conference.
- Off-net call forwarding and mobile twinning.
- Transmission and response of SIP OPTIONS messages sent by M-net requiring Avaya response and sent by Avaya requiring M-net response.

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:

- It was observed during compliance testing that both inbound and outbound T38 Fallback fax calls were failing when "Media Security" was set to "Preferred" on the SIPLine connections between IP Office Server Edition and IP Office 500 V2 Expansion. In order for inbound and outbound T38 Fallback fax calls to terminate successfully, "Media Security" was set to "Disabled" on the SIPLine connections between IP Office Server Edition and IP Office500 V2 Expansion as per **Section 5.11.2**. This issue is currently under investigation with the Avaya IP Office Support team.
- G.729 codec is not supported by M-net and therefore was not tested.
- No Inbound Toll-Free access available for test.
- No Emergency Services test call booked with Operator.

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/sip-trunk.

3. Reference Configuration

Figure 1 below 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 Server Edition, an Avaya IP Office 500 V2 as an Expansion and an Avaya SBC. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 9600 Series IP Telephones (with H.323 firmware), Avaya 1140e SIP Telephones, Avaya 1400 Series Digital Deskphones, Analog Telephone and a fax machine. The site also has a Windows 7 PC running Avaya IP Office Manager to configure the Avaya IP Office as well as Avaya Workplace Client for Windows for softphone testing.

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.

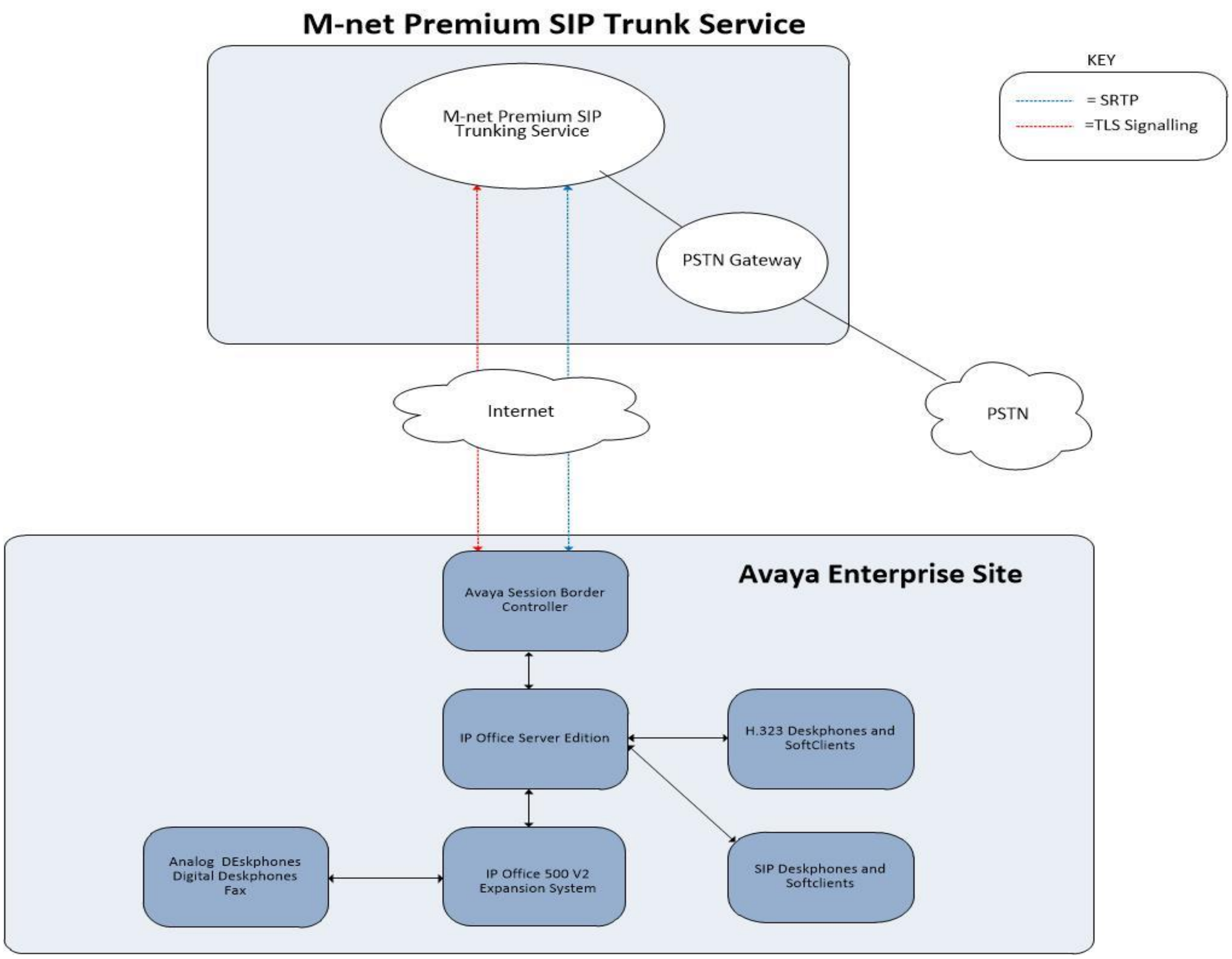


Figure 1: Test setup M-net Premium SIP Trunk service to simulated Avaya Enterprise

4. Equipment and Software Validated

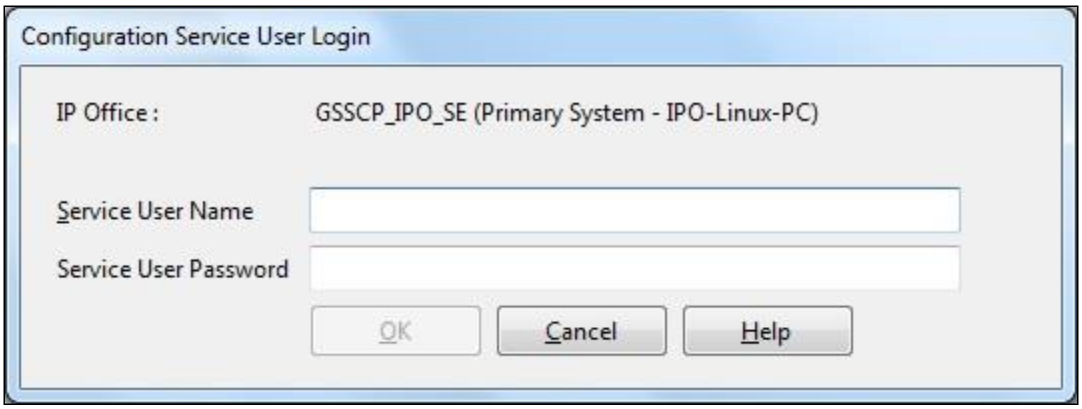
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office Server Edition	Version 11.1.3.1.0 build 34
Avaya IP Office 500 V2	Version 11.1.3.1.0 build 34
Avaya Voicemail Pro Client	Version 11.1.3.0.0
Avaya IP Office Manager	Version 11.1.3.1.0 build 34
Avaya Session Border Controller	10.1.2.0-64-23285
Avaya 1608 Phone (H.323)	1.3.12
Avaya 9611G Series Phone (H.323)	6.8.3
Avaya J179 Series Phone (SIP)	4.0.10
Avaya Workplace Client for Windows (SIP)	3.34.1
Avaya 1140e (SIP)	FW: 04.04.23.00.bin
Avaya Analogue Phone	N/A
M-net	
Metaswitch Perimeta SBC and IPX (Class 4 Switch/Routing and SBC)	GA
Metaswitch CFS (Class 5 Switch)	GA

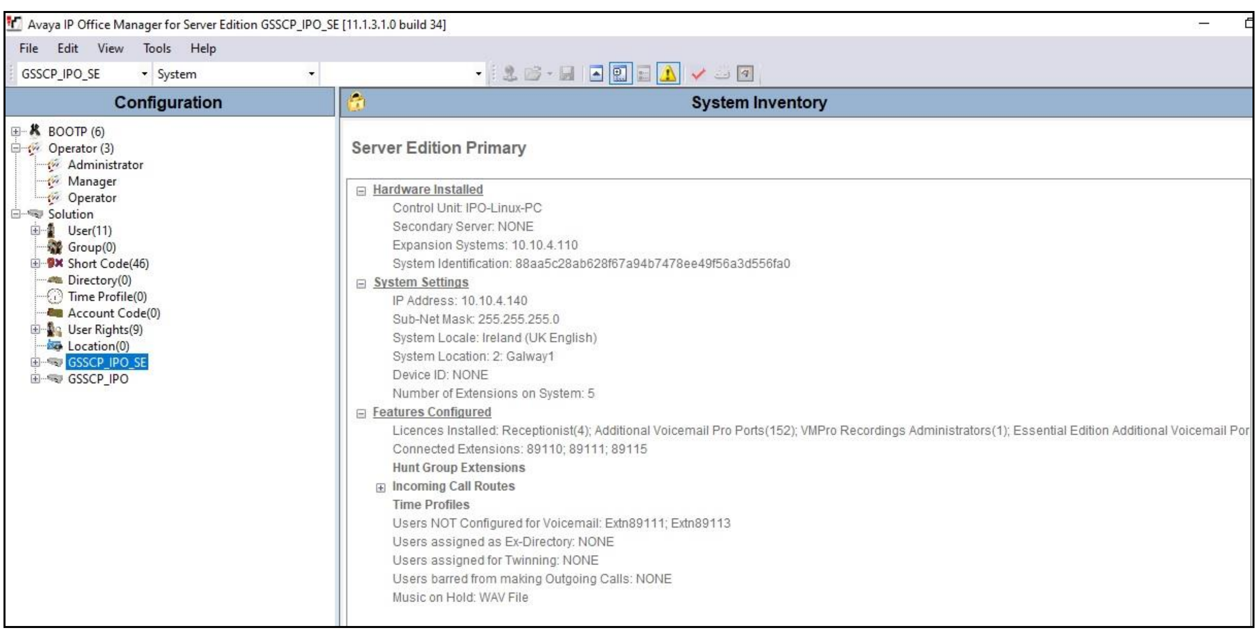
Note – Testing was performed with IP Office Server Edition with Expansion IP Office500 V2. Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks, this includes T.38 fax.

5. Configure Avaya IP Office

This section describes the Avaya IP Office configuration to support connectivity to the M-net Premium SIP Trunk 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.

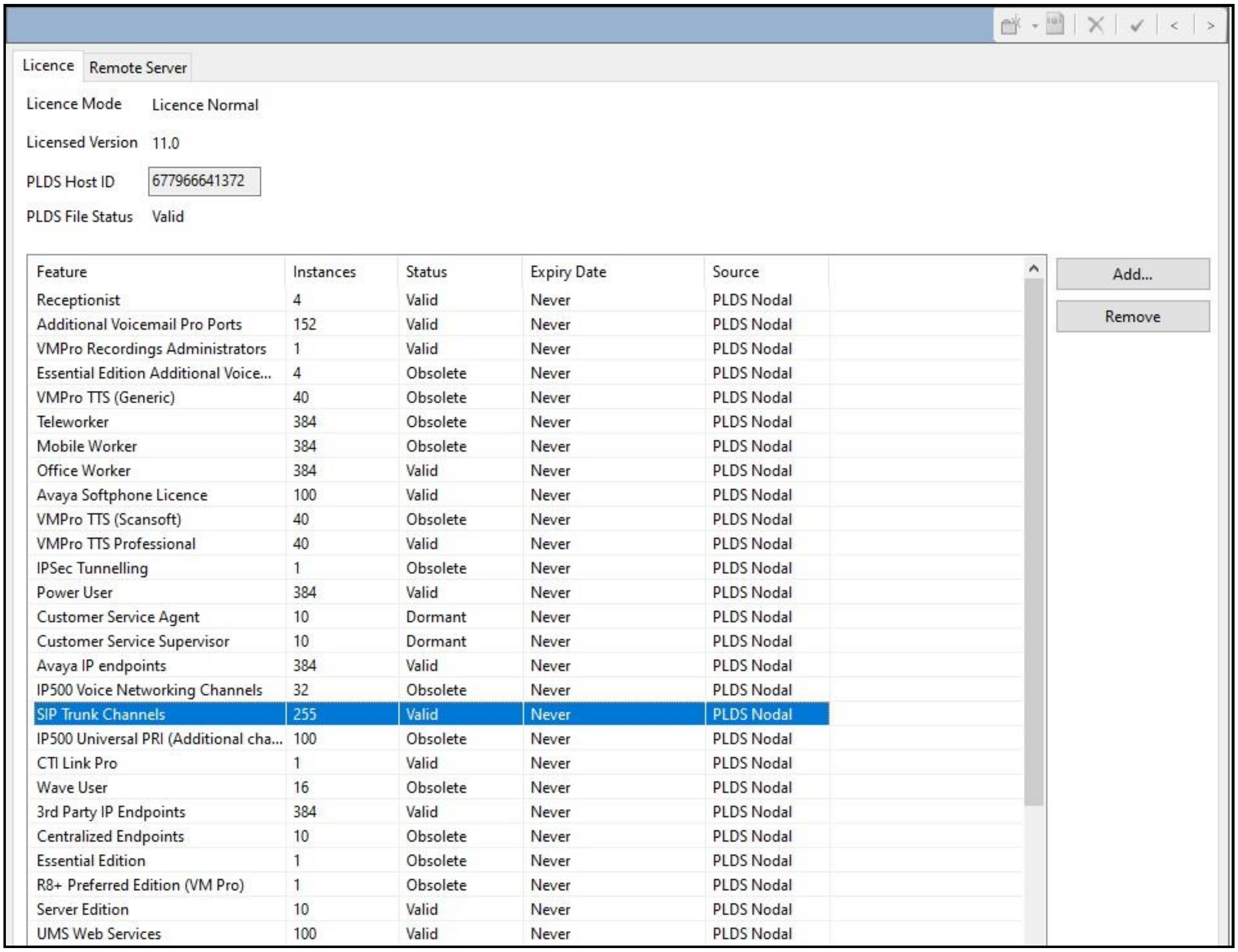


A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All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

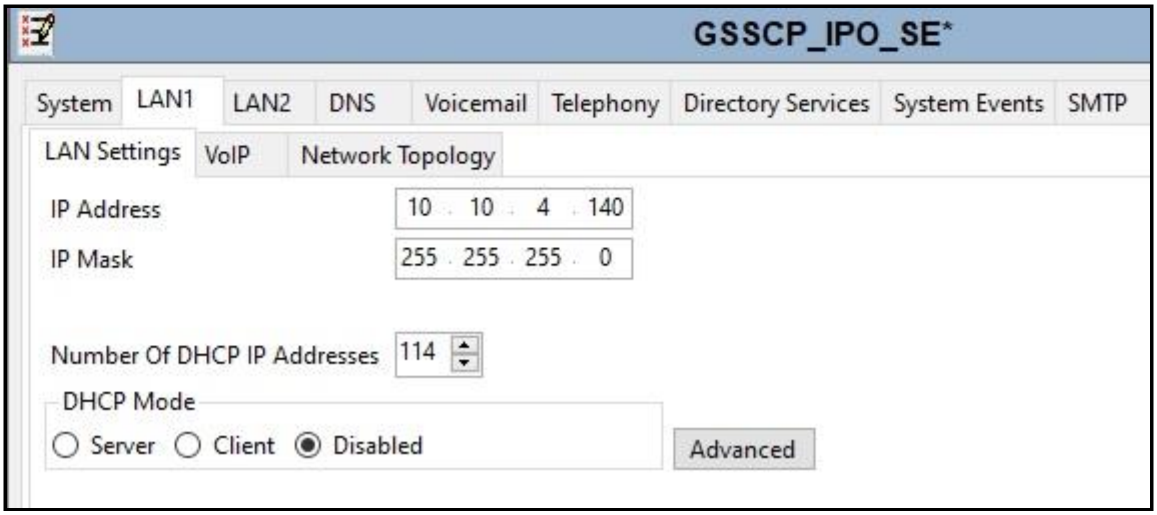


Feature	Instances	Status	Expiry Date	Source
Receptionist	4	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	152	Valid	Never	PLDS Nodal
VMPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	4	Obsolete	Never	PLDS Nodal
VMPro TTS (Generic)	40	Obsolete	Never	PLDS Nodal
Teleworker	384	Obsolete	Never	PLDS Nodal
Mobile Worker	384	Obsolete	Never	PLDS Nodal
Office Worker	384	Valid	Never	PLDS Nodal
Avaya Softphone Licence	100	Valid	Never	PLDS Nodal
VMPro TTS (Scansoft)	40	Obsolete	Never	PLDS Nodal
VMPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
Customer Service Agent	10	Dormant	Never	PLDS Nodal
Customer Service Supervisor	10	Dormant	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Obsolete	Never	PLDS Nodal
SIP Trunk Channels	255	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	384	Valid	Never	PLDS Nodal
Centralized Endpoints	10	Obsolete	Never	PLDS Nodal
Essential Edition	1	Obsolete	Never	PLDS Nodal
R8+ Preferred Edition (VM Pro)	1	Obsolete	Never	PLDS Nodal
Server Edition	10	Valid	Never	PLDS Nodal
UMS Web Services	100	Valid	Never	PLDS Nodal

5.2. LAN1 Settings

In an Avaya IP Office, the LAN2 tab settings correspond to the Avaya IP Office WAN port (public network side) and the LAN1 tab settings correspond to the LAN port (private network side). For the compliance test, the **LAN1** interface was used to the Avaya IP Office to the internal side of the Avaya SBCE as these are on the same LAN, **LAN2** was not used.

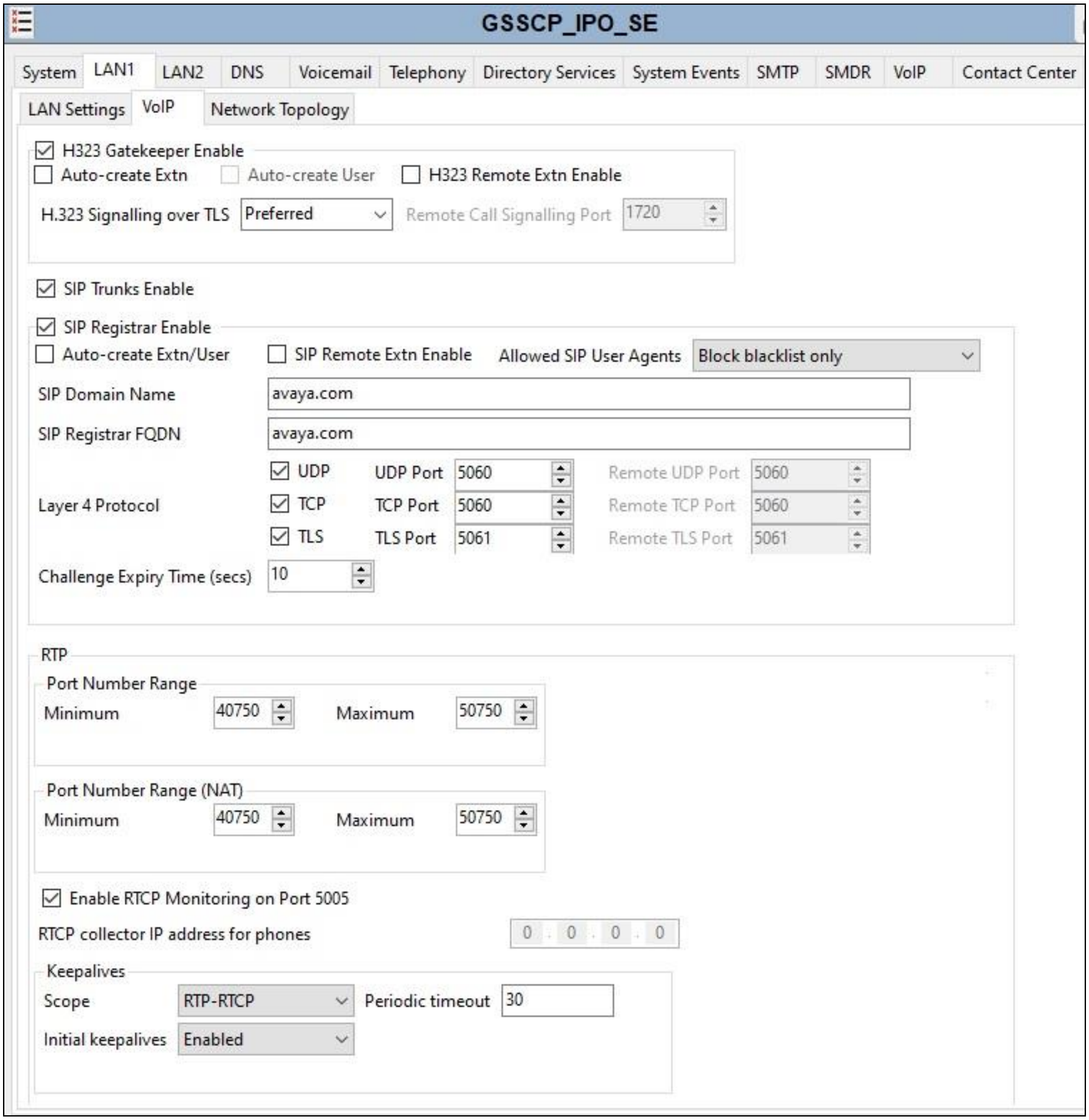
To access the LAN1 settings, first navigate to **System → GSSCP_IPO_SE** in the Navigation Pane where GSSCP_IPO_SE is the name of the IP Office. Navigate to the **LAN1 → LAN Settings** tab in the Details Pane. The **IP Address** and **IP Mask** fields are the private interface of the IP Office. All other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



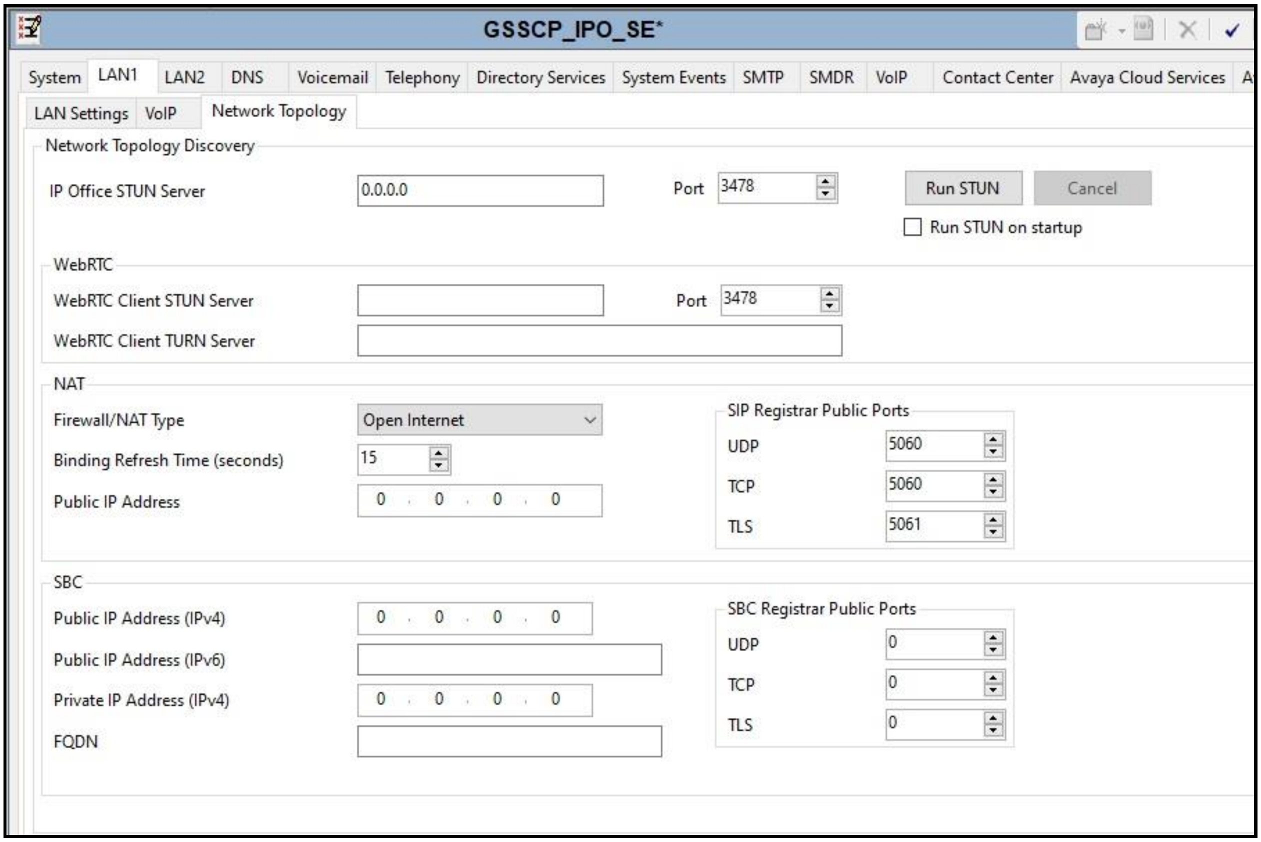
On the **VoIP** tab in the Details Pane, the **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol. Set **H.323 Signalling over TLS** to **Preferred** to allow IP Office endpoints to use TLS for signalling. Check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. If SIP Endpoints are to be used such as the Avaya Communicator for Windows and the Avaya 1140e, the **SIP Registrar Enable** box must also be checked. 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.

The **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media. Set **Scope** to **RTP-RTCP** and **Initial keepalives** to **Enabled** and **Periodic timeout** to **30**.

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).

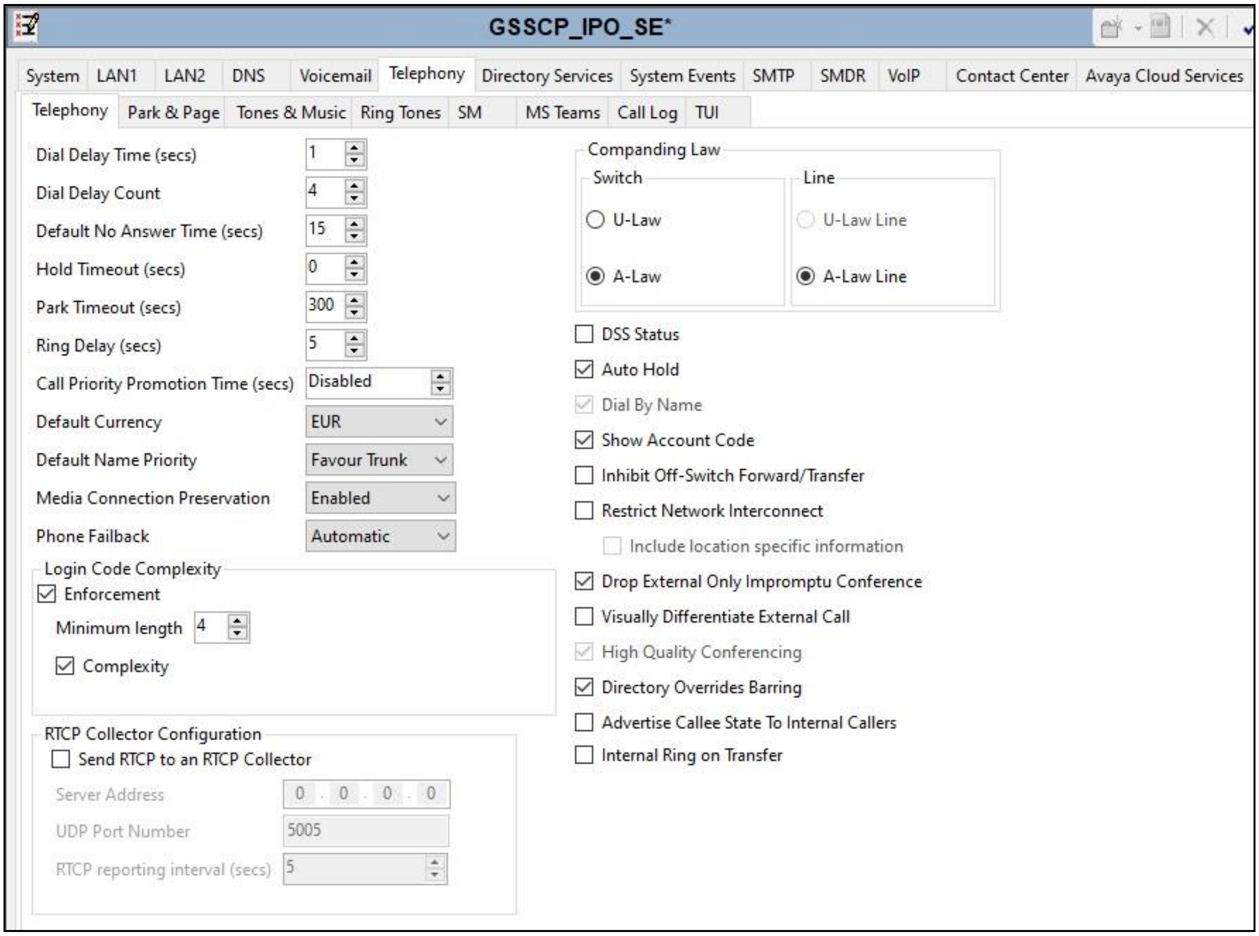


On the **Network Topology** tab, set the **Firewall/NAT Type** from the pulldown menu to **Open Internet**. With this configuration, the **STUN Server IP Address** and **STUN Port** are not used as NAT was not required for this configuration, therefore resulting in no requirement for a STUN server. The **Use Network Topology Info** in the **SIP Line** was set to **None** in **Section 5.6.2**. Set **Binding Refresh Time (seconds)** to **15**. 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).



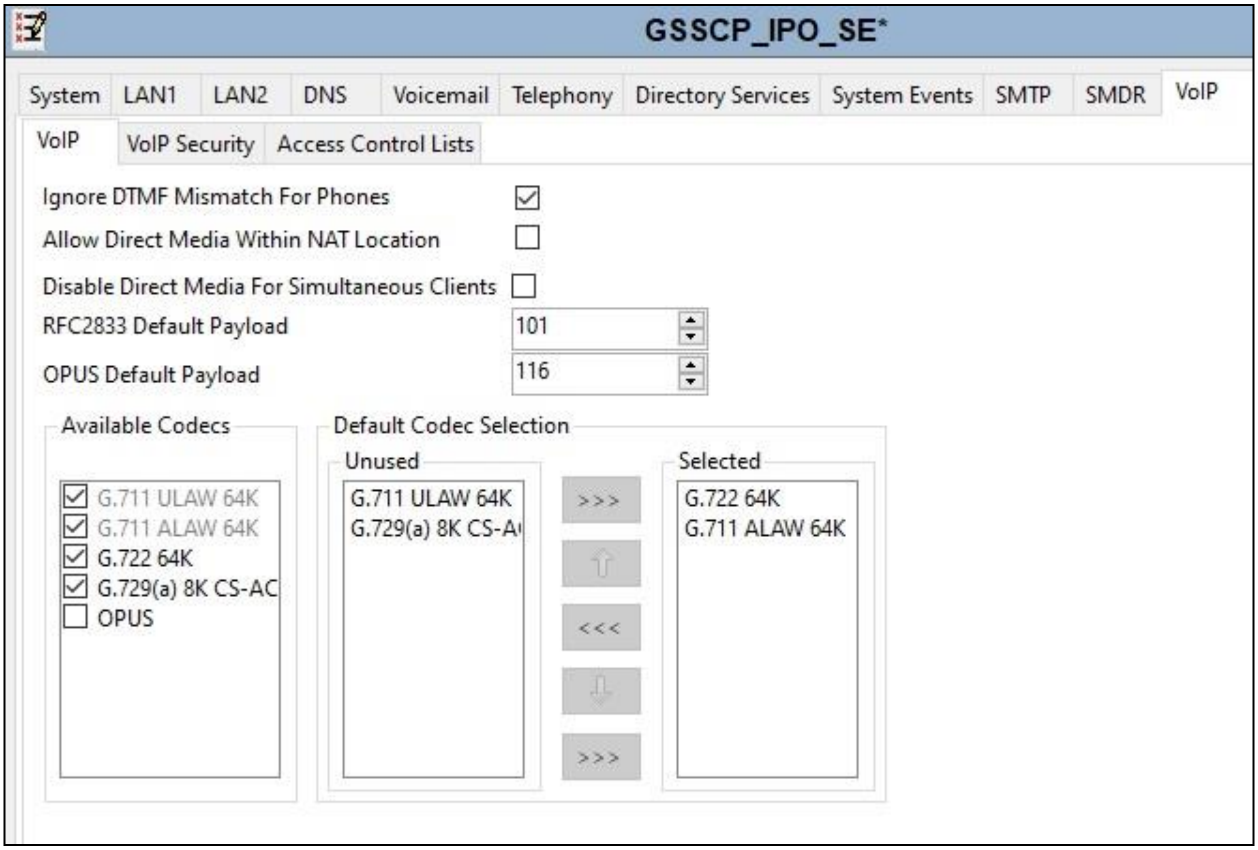
5.3. System Telephony Settings

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



5.4. VoIP Settings

Navigate to the **VoIP** 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.722.64K** and **G.711 ALAW 64K** is set as the priority codecs as per screenshot below.



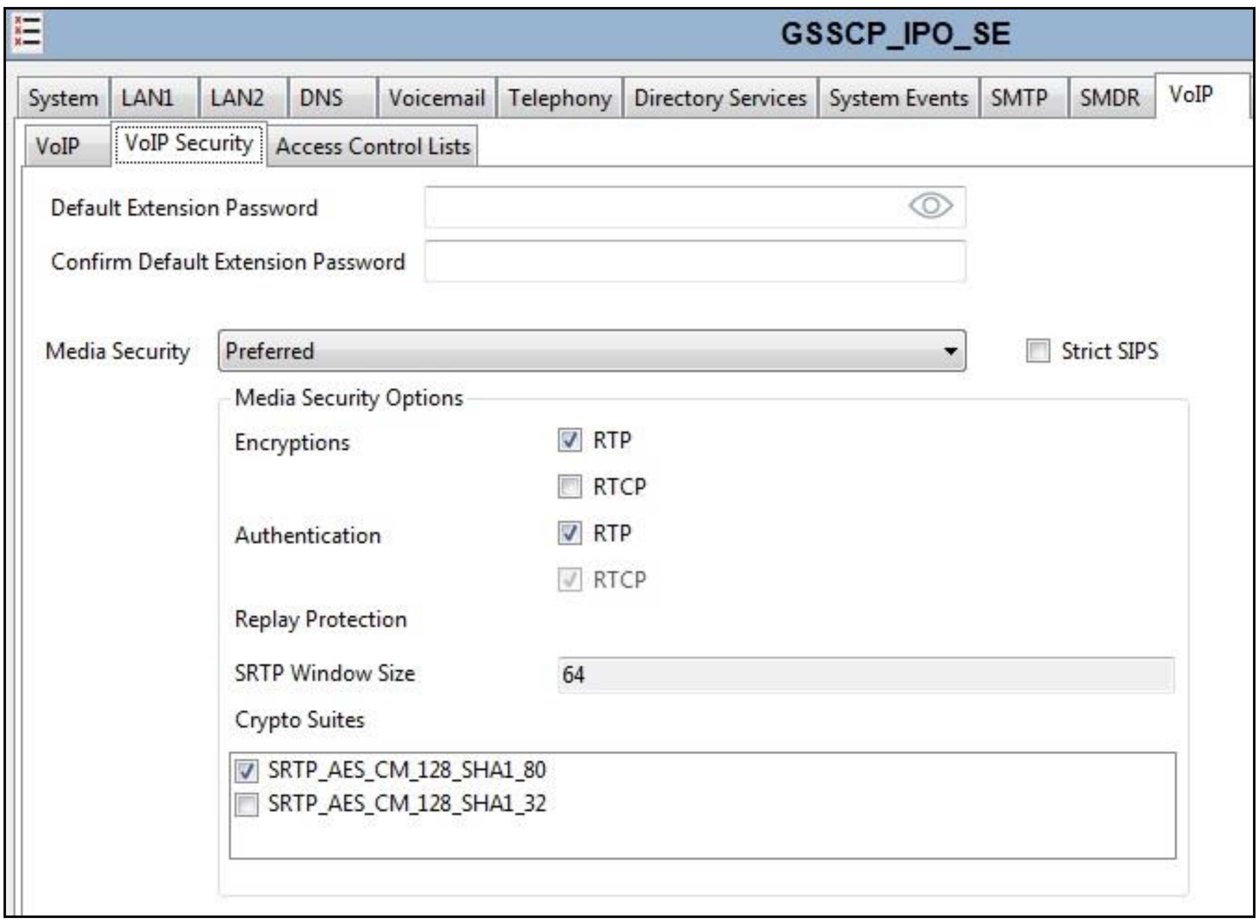
5.5. VoIP Security

When enabling SRTP on the system, the recommended setting for **Media** is **Preferred**. In this scenario, IP Office uses SRTP if supported by the other end, and otherwise uses RTP. If the **Enforced** setting is used, and SRTP is not supported by the other end, the call is not established.

In the compliance testing, **Preferred** is selected as this allows IP Office to fall back to non-secure media if the attempt to use secure media is unsuccessful.

Navigate to **System → VoIP Security** tab and configure as follows:

- Select **Preferred** for **Media**.
- Check **RTP** for **Encryptions**.
- Check **RTP** for **Authentication**.
- Check **SRTP_AES_CM_128_SHA1_80** for **Crypto Suites**.
- Other parameters are left as default.
- Click **OK**.



5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the M-net Premium SIP Trunk 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 5.6.1** 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:

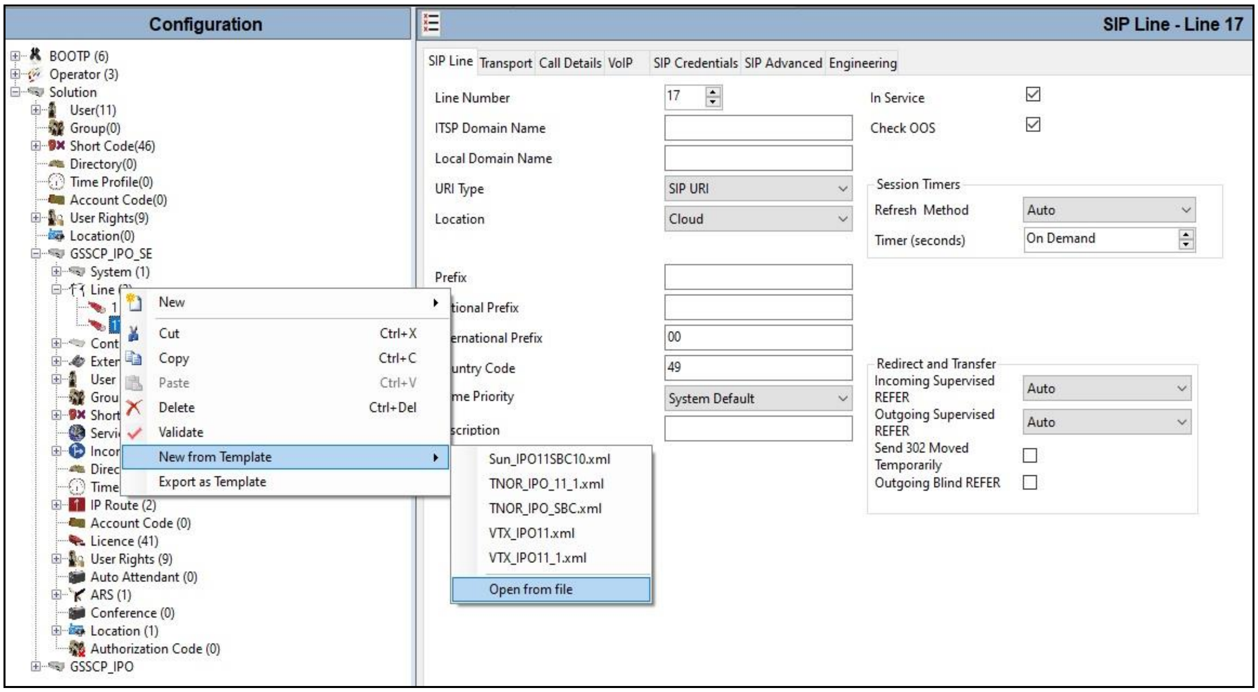
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

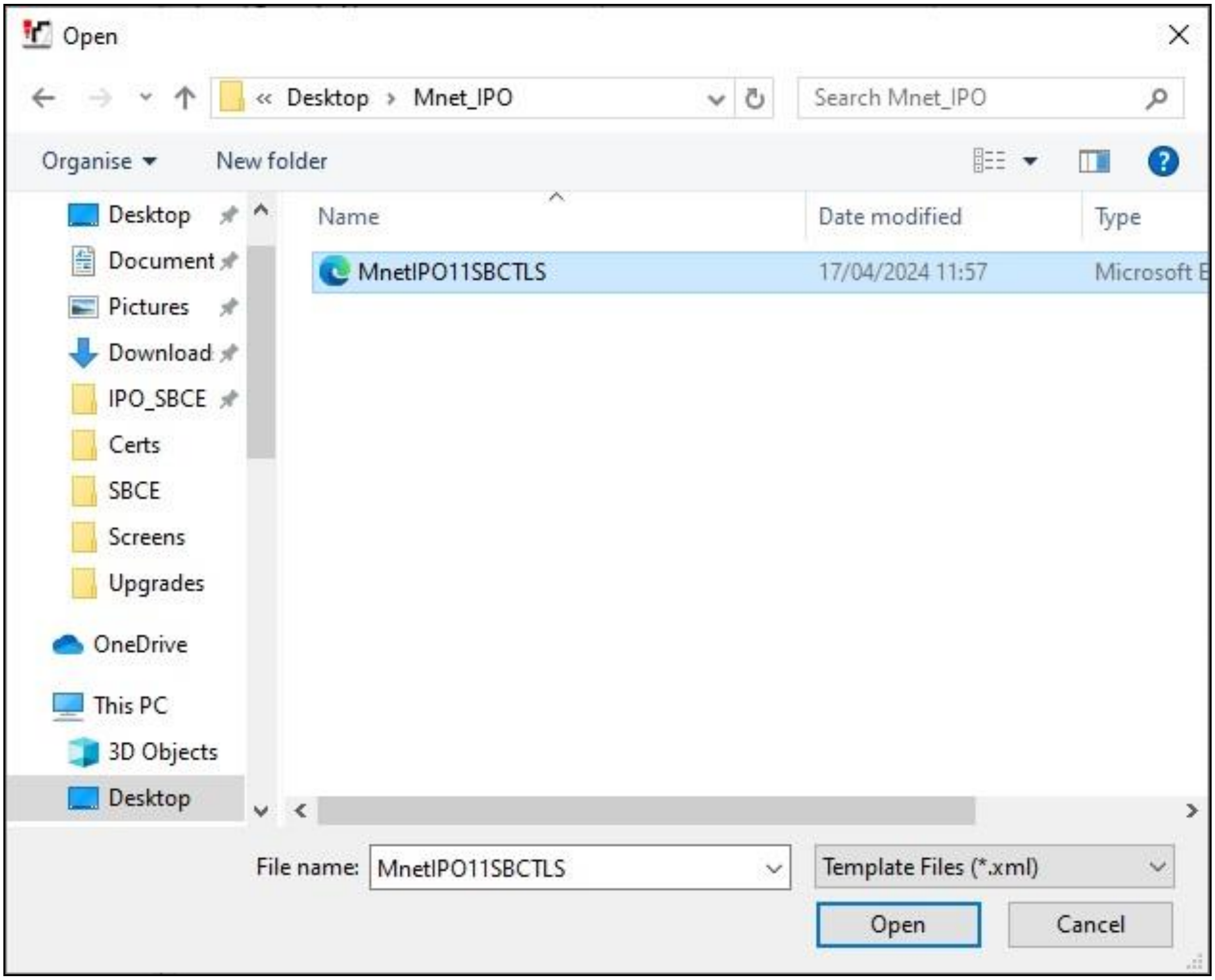
5.6.1. SIP Line From Template

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.

Copy a previously created template file to a location (e.g., *temp*) on the same computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New → New from Template →Open from file**.



Navigate to the directory on the local machine where the template was copied and select the template as required.



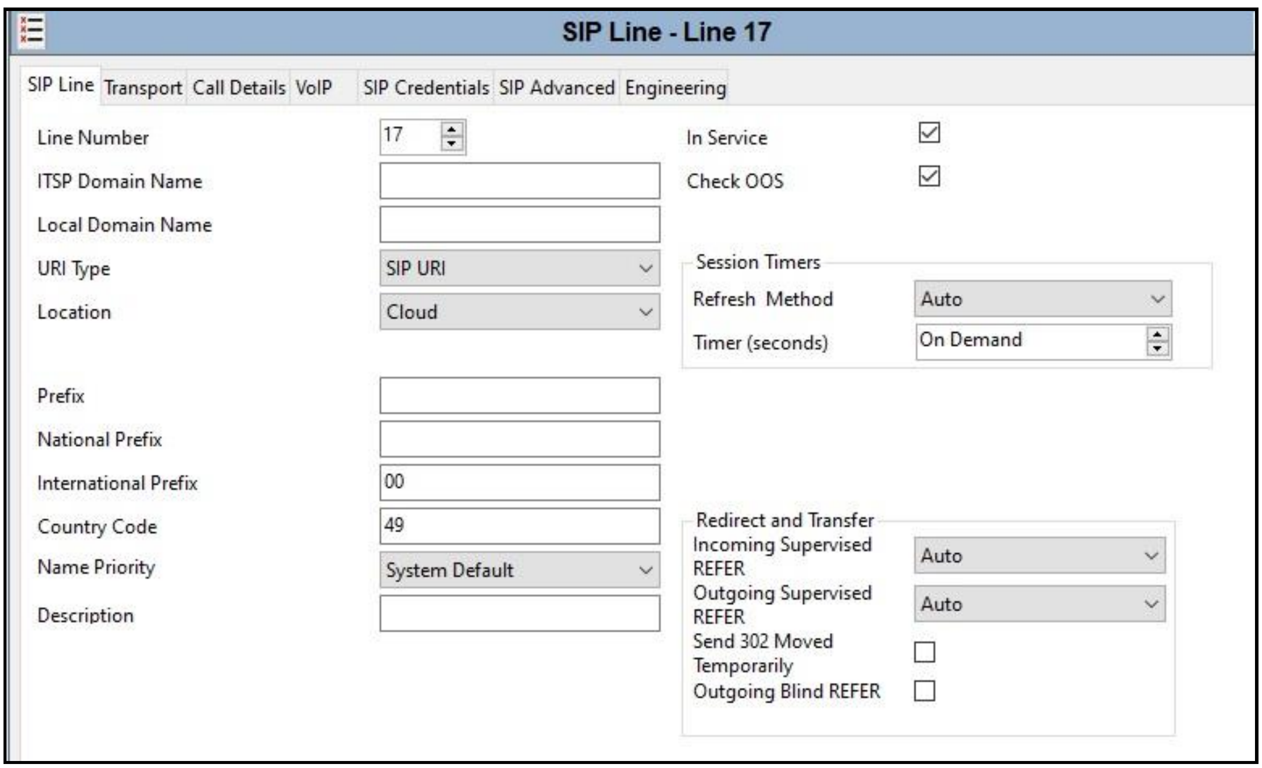
The SIP Line is automatically created and can be verified and edited as required using the configuration described in **Section 5.6.2**.

5.6.2. 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 a domain name provider by the Service Provider if required, however no ITSP Domain Name was used in this configuration.
- Set **Country Code** to 49 and **International Prefix** to 00 so international numbers can be correctly identified.
- Ensure the **In Service** box is checked.
- Ensure the **Check OSS** box is checked.
- Leave the **Refresh Method** at the default value of **Auto** which results in re-INVITE being used for Session Refresh.
- Leave **Timer (seconds)** at the default value of **On Demand**. This value allows the Session Refresh interval to be set by the network.
- Set **Incoming Supervised REFER** and **Outgoing Supervise REFER** to **Auto**.
- Default values may be used for all other parameters.

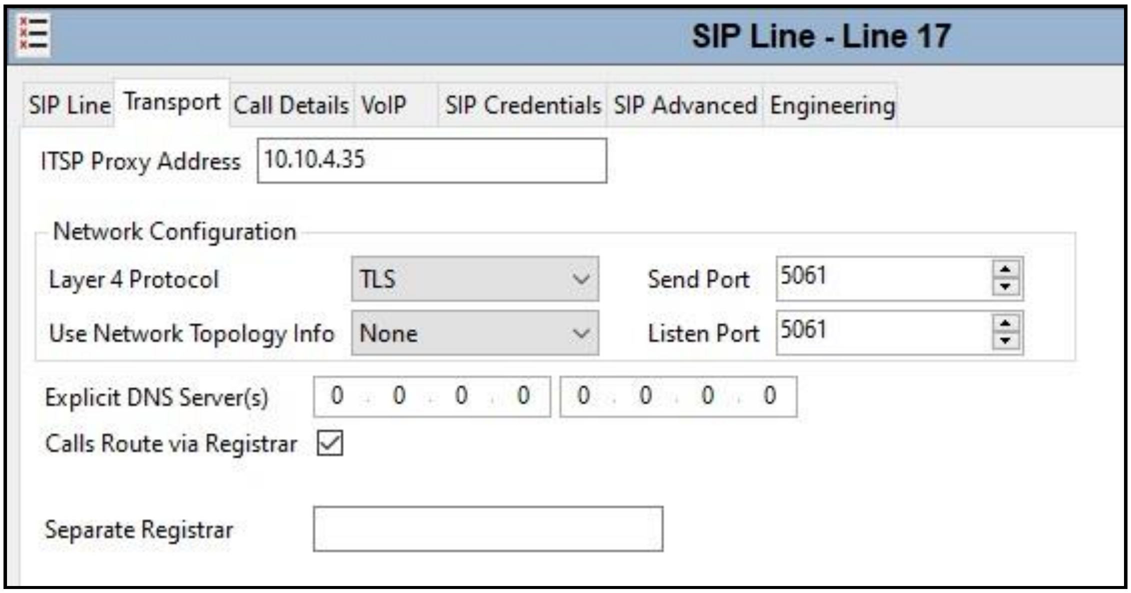
On completion, click the **OK** button (not shown).



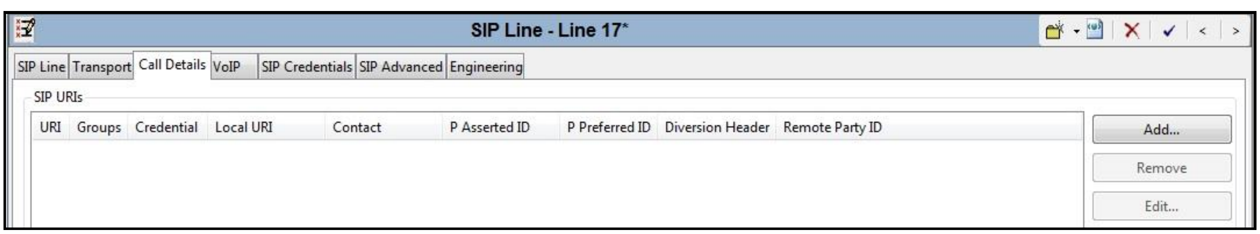
Select the **Transport** tab and set the following:

- Set **ITSP Proxy Address** to the inside interface IP address (**10.10.4.35**) of the Avaya SBCE as shown in **Figure 1**.
- Set **Layer 4 Protocol** to **TLS**.
- Set **Send Port** to **5061** and **Listen Port** to **5061**.
- Set **Use Network Topology Info** to **None**.

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



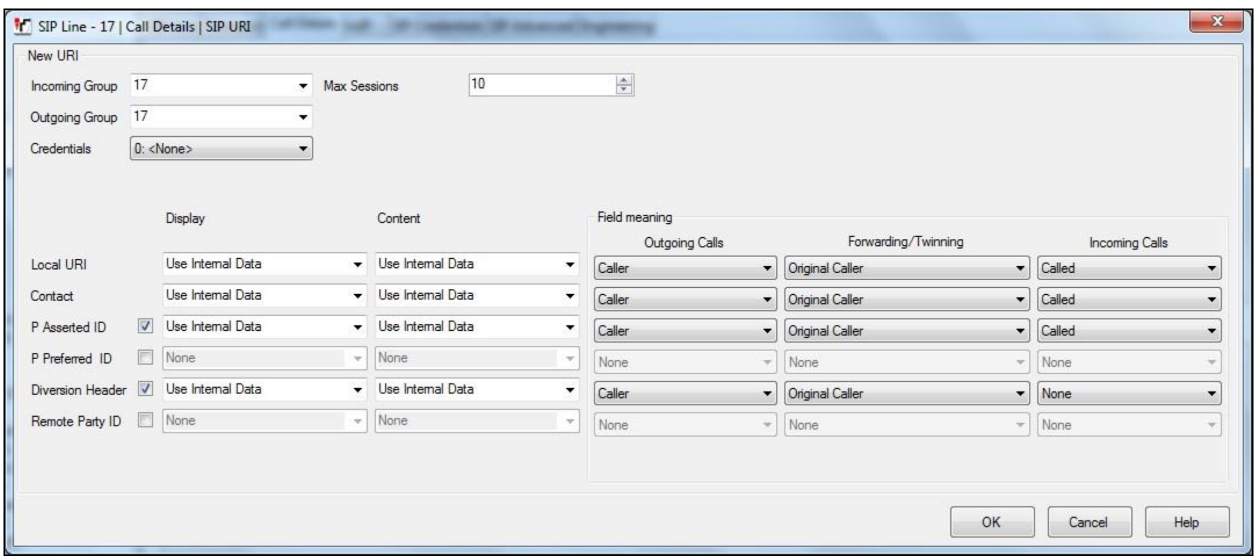
After the SIP line parameters are defined, the SIP URIs that Avaya IP Office will accept on this line must be created. To create a SIP URI entry, select the **Call Details** tab and click on **Add**.



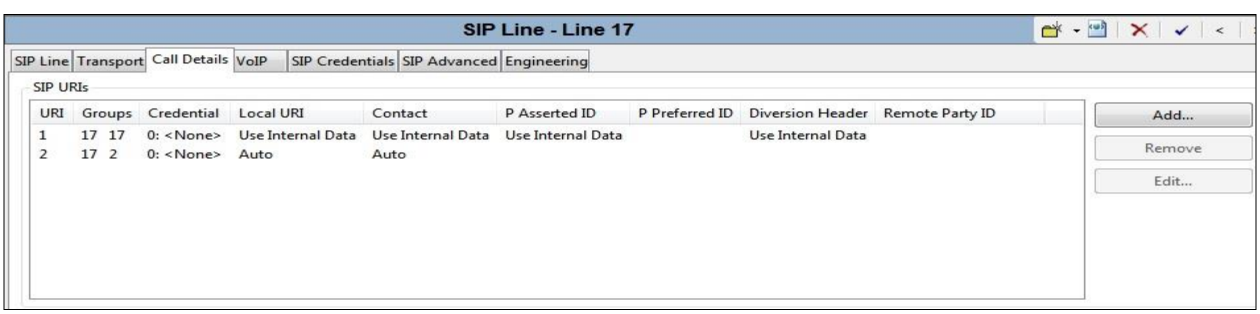
A SIP URI is shown in this example that is used for calls to and from extensions that have a DDI number assigned to them. Additional SIP URI's may be required for calls to services such as Voicemail Collect and the Mobile Twinning FNE, these would be for incoming calls only.

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

- Set **Incoming Group**. This is the value assigned for incoming calls that's analysed in the Incoming Call Route settings described in **Section 5.9**. In the test environment a value of **17** was used for the M-net SIP platform.
- Set **Outgoing Group**. This is the value assigned for outgoing calls that can be selected directly in the short code settings described in **Section 5.7**. In the test environment a value of **17** was used.
- Set **Max Sessions** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern
- Set **Local URI**, **Contact** and **P Asserted ID** to **Use Internal Data** for both the **Display** name and **Content**. On incoming calls, this will analyse the Request-Line sent by M-net and match to the SIP settings in the User profile as described in **Section 5.8**. On outgoing calls this will insert the SIP settings in the User profile into the relevant headers in the SIP messages.
- Leave the **Outgoing Calls**, **Forwarding/Twinning** and **Incoming Calls** at their respective default values of **Caller**, **Original Caller** and **Called** for the **Local URI**, **Contact** and **P Asserted ID** call details.



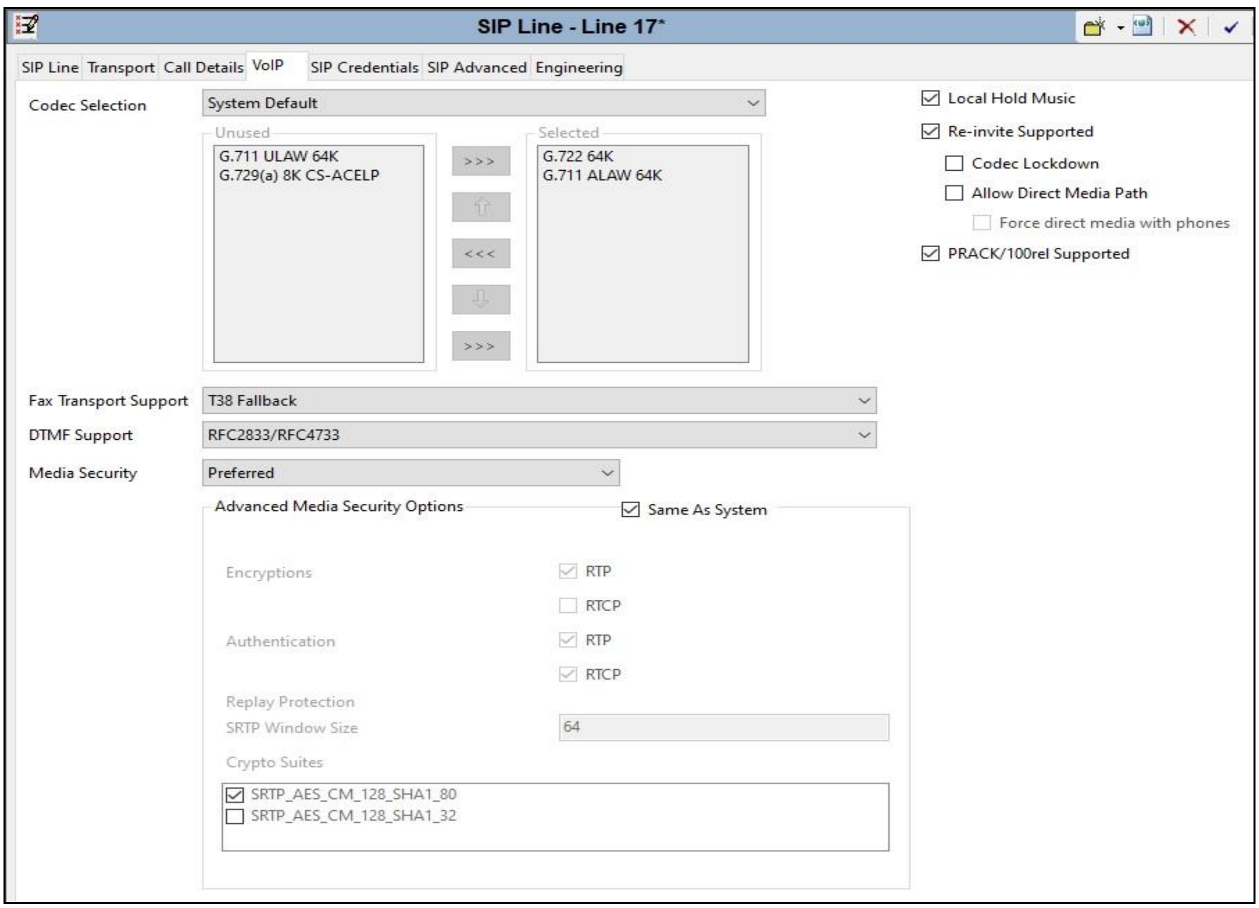
The following screenshot shows the completed configuration:



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

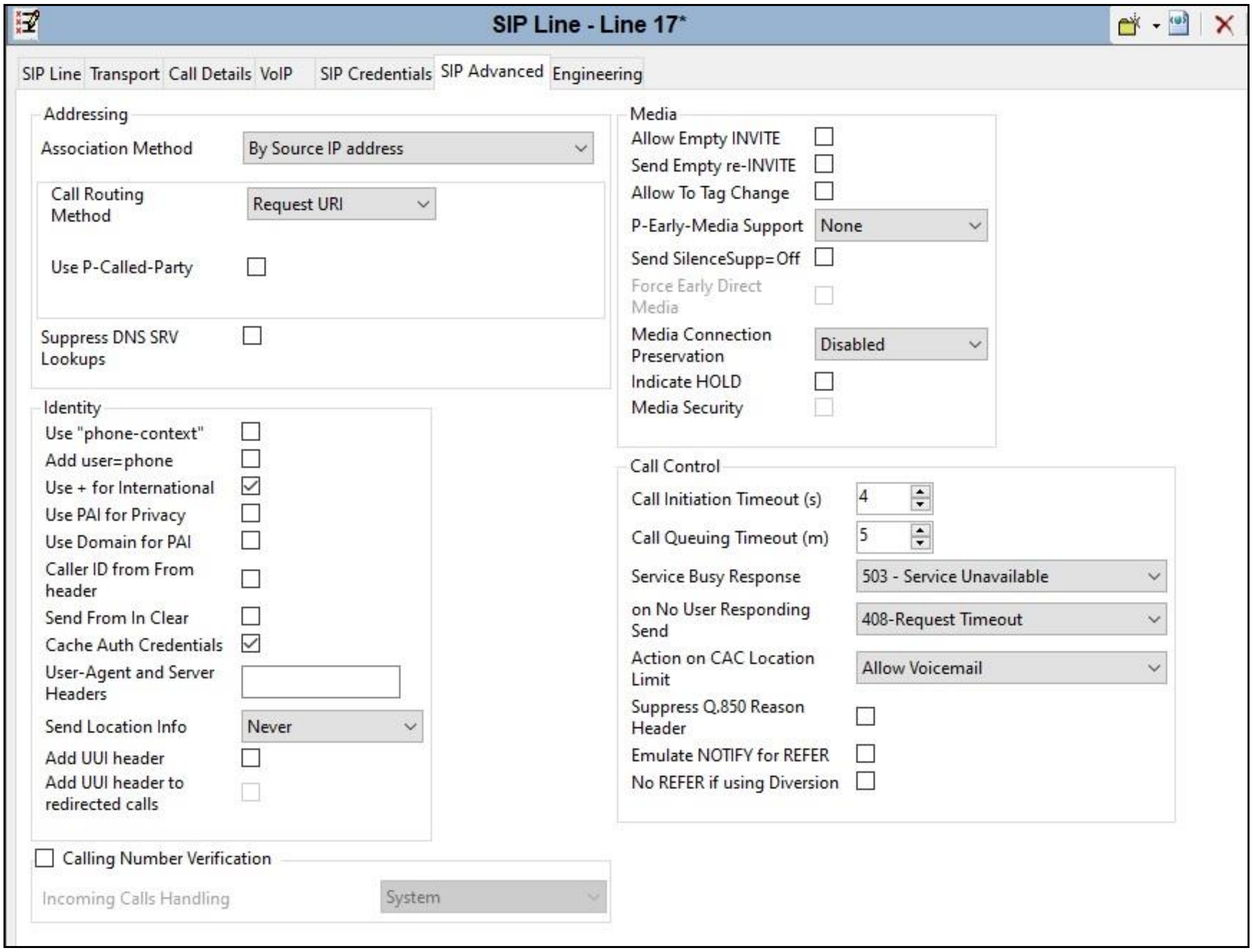
- Select **System Default** from the drop-down menu as system default codecs were already defined in **Section 5.4**.
- Set the **Fax Transport Support** box to **T.38 Fallback** as this is the preferred method of fax transmission for M-net.
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs Avaya IP Office to send DTMF tones using RTP events messages as defined in RFC2833.
- Check **Media Security** to **Same as System (Preferred)** and ensure that the **Same as System** box is checked. This ensures that system level media security is set to **Preferred** specifying that SRTP is preferred over RTP as configured in **Section 5.5**.
- Check the **Local Hold Music** 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.
- Check the **PRACK/100rel Supported** box if early media is required. This was checked during compliance testing.
- On completion, click the **OK** button (not shown).

Default values may be used for all other parameters.



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

- Check the **Use + for International** as E.164 numbering is used on the SIP Trunk.
- Default values may be used for all other parameters.



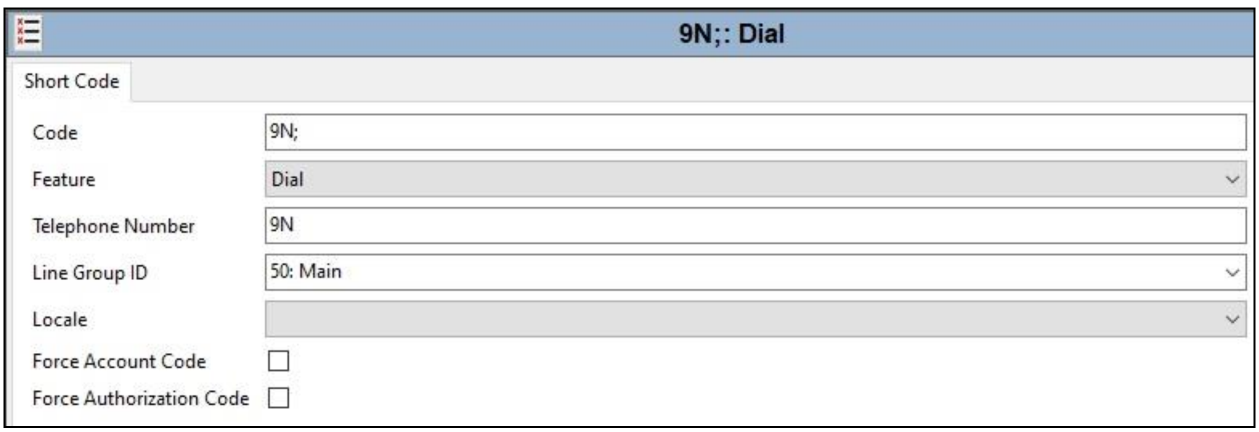
Note: It is advisable at this stage to save the configuration as described in **Section 5.12** to add the Line Group ID defined in **Section 5.6.2** available.

5.7. Short Codes

Define a short code to route outbound traffic to the SIP line. To create a short code, right-click **Short Code** in the Navigation Pane and select **New**. On the **Short Code** tab in the Details Pane, configure the parameters as required. The example below shows the configuration used during testing 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 **9N**; 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 **9N** so that the call is passed to the ARS function with the dialled number unchanged.
- Set the **Line Group Id** to the ARS route number described in **Section 5.9**.
- On completion, click the **OK** button (not shown).

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

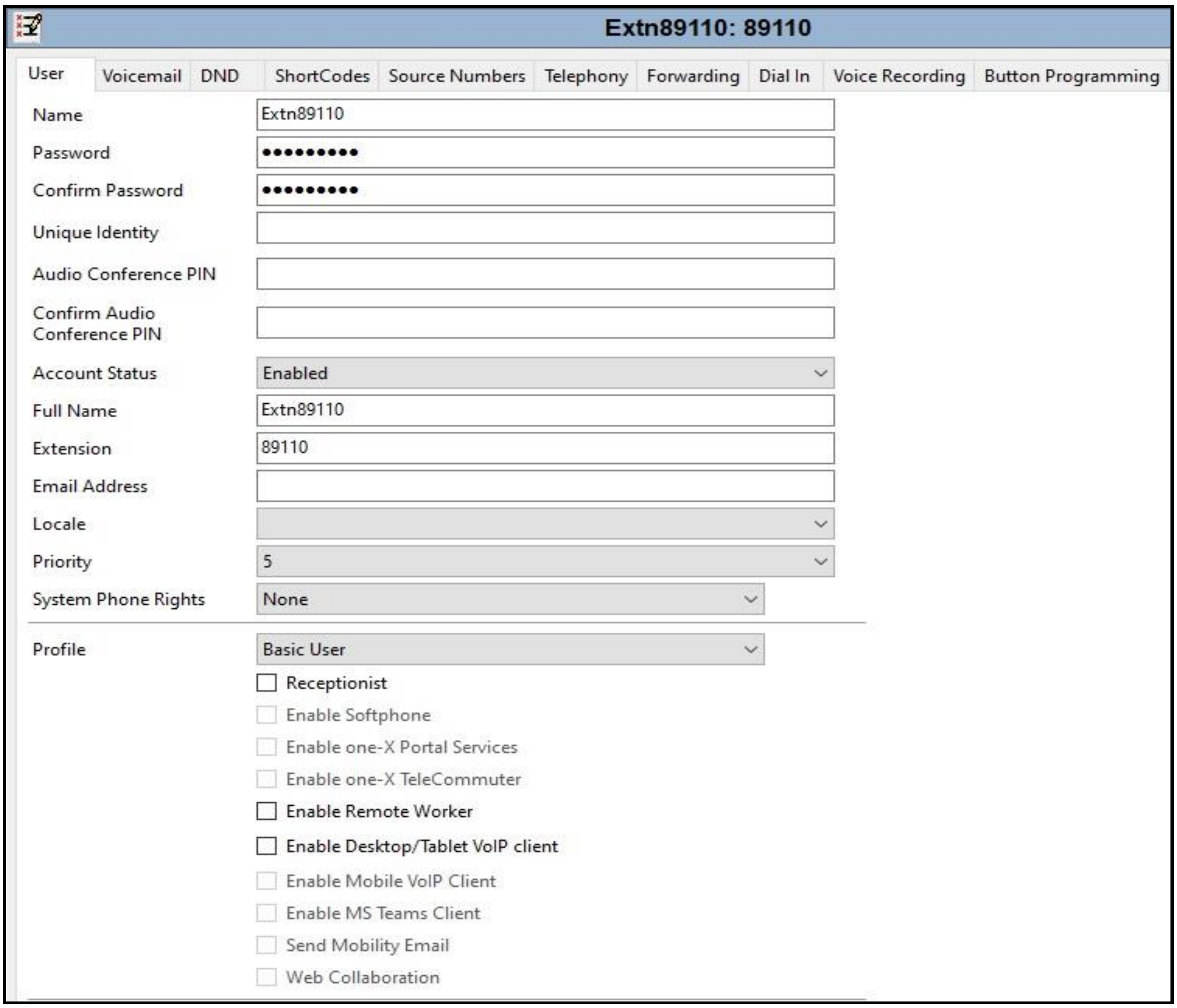


5.8. User

Configure the SIP parameters for each user that will be placing and receiving calls via the SIP line defined in **Section 5.6.2**. To configure these settings, first navigate to **User** in the Navigation Pane. Select the **User** tab if any changes are required.

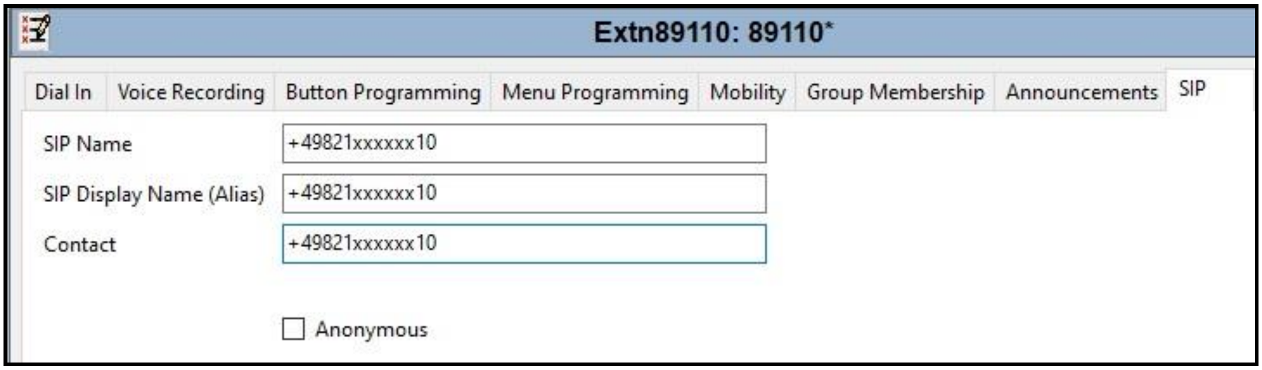
The following example shows the configuration required for a SIP Endpoint.

- Change the **Name** of the User if required.
- Set the **Password** and **Confirm Password**.
- Select the required profile from the **Profile** drop down menu. **Basic User** is commonly used; **Power User** can be selected for SIP softphone, WebRTC and Remote Worker endpoints.



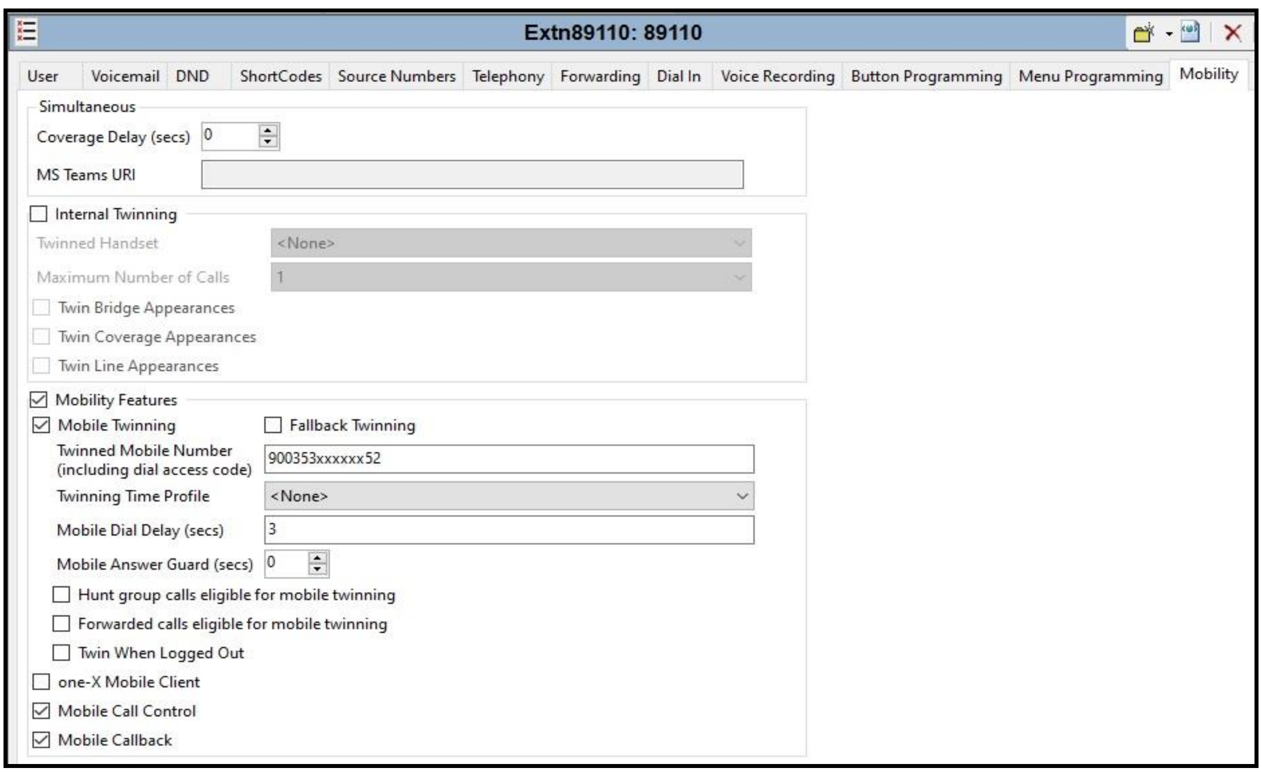
SIP endpoints require setting of the **SIP Registrar Enable** as described in **Section 5.2**.

Next, select the **SIP** tab in the Details Pane. To reach the **SIP** tab click the right arrow on the right-hand side of the Details Pane until it becomes visible. The values entered for the SIP **Name** and **Contact** fields are used as the user part of the SIP URI in the From header for outgoing SIP trunk calls. These 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 5.6.2)**. As such, these fields should be set to one of the DDI numbers assigned to the enterprise from M-net.



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

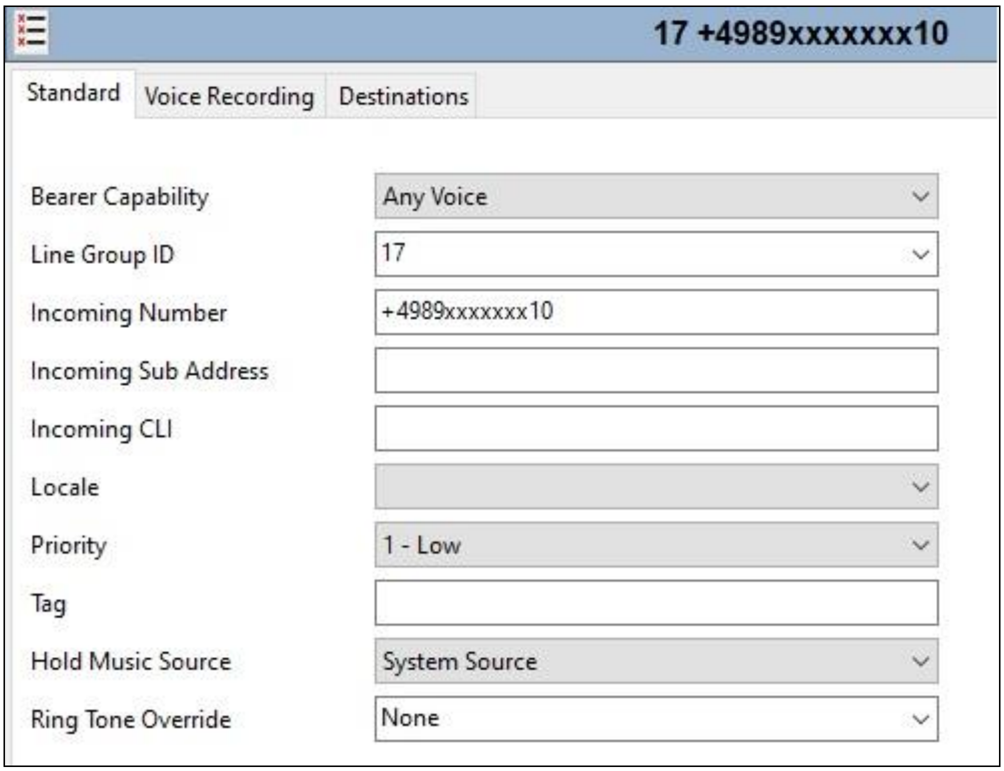
The following screen shows the Mobility tab for user 89110. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.



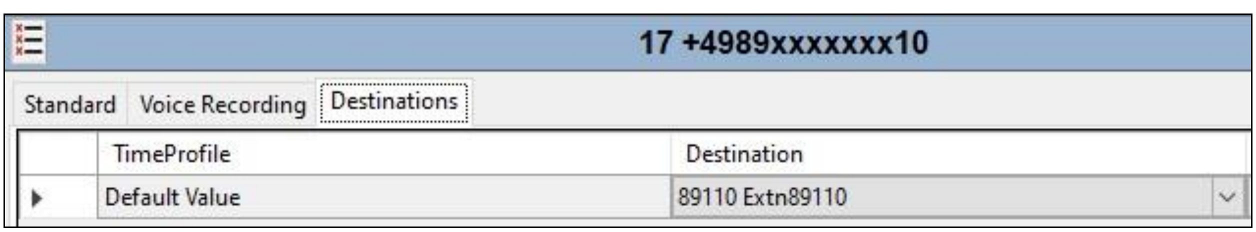
5.9. Incoming Call Routing

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 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 Capability** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6.2**.
- 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.

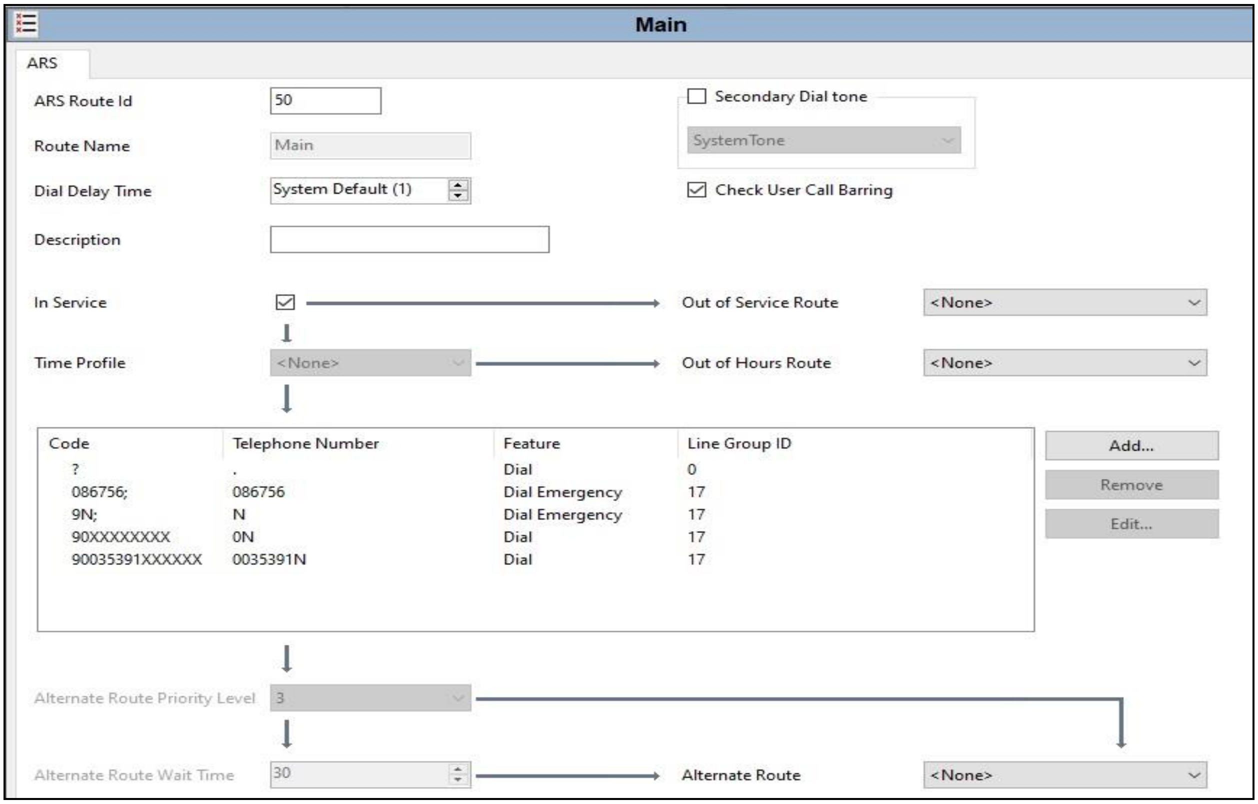


On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+4989xxxxxxx10** on line 17 are routed to extension 89110.



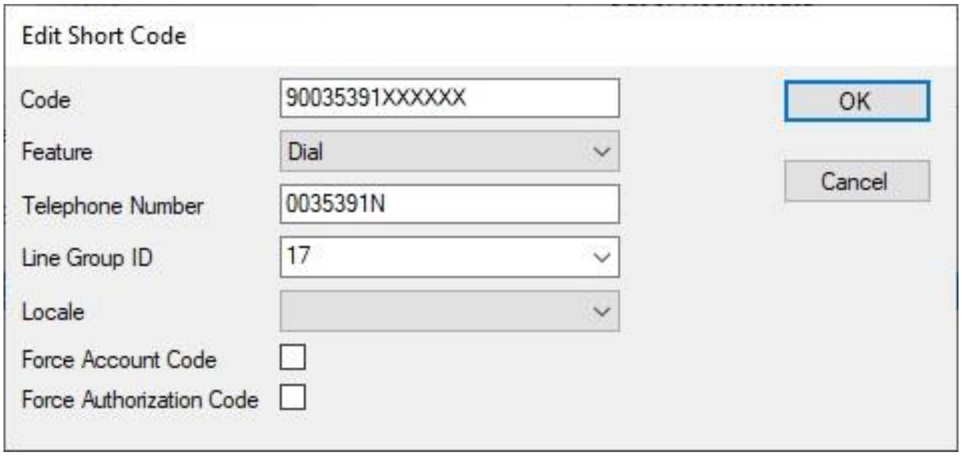
5.10. ARS

The Main ARS route exists by default and requires editing. Select the ARS **Main** route and click on **Add**.



Define numbers as required. An example for national numbers is as follows:

- Define the Short **Code**, the example shows a 15 international number with country code and city code prefixed with **9** for an outside line. Note that **X** indicates any digit and **;** causes the system to wait for the full number to be dialled.
- Select **Dial** in the **Feature** drop down menu.
- Define the **Telephone Number** without the **9** which removes it and sends the number as dialled. All **X** characters can be replaced with a single **N**.
- Select the **Line Group ID** defined in the SIP Line URI described in **Section 5.6.2**. During testing this was **17** for the SIP Trunk. Click on **OK**


Edit Short Code
Code
90035391XXXXXX
Feature
Dial
Telephone Number
0035391N
Line Group ID
17
Locale
Force Account Code
Force Authorization Code
OK
Cancel

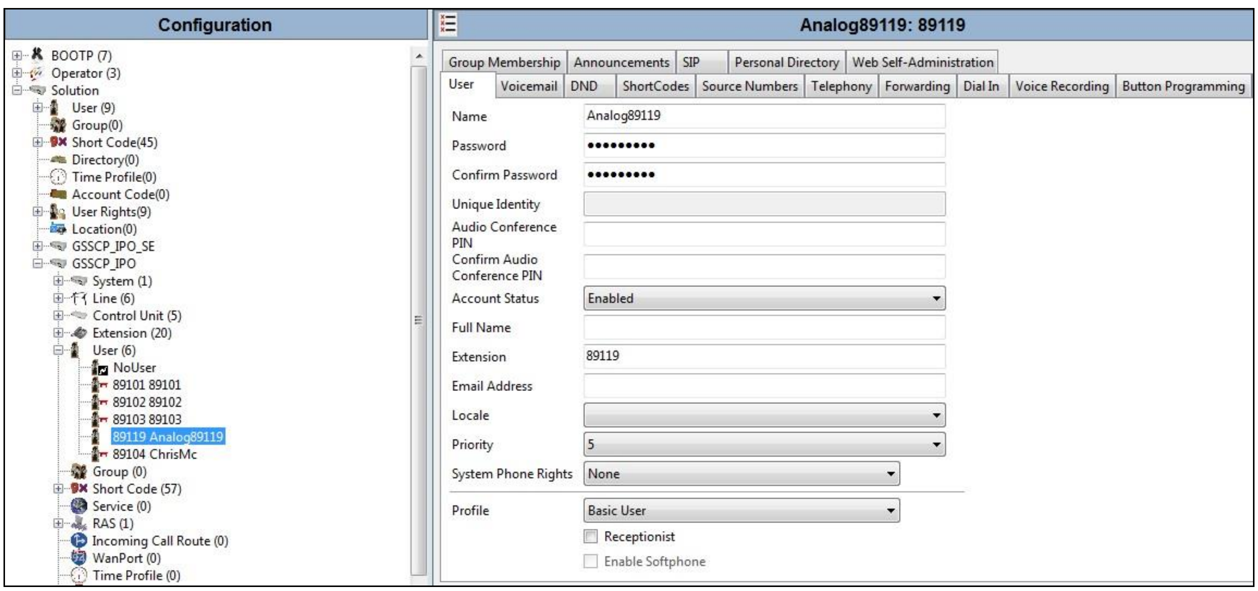
5.11. Fax

At Release 11, both G.711 and T.38 Fax is supported on IP Office Server Edition when using an IP Office Expansion (500 V2). The Mnet SIP Trunk testing was carried out using this configuration with only the analogue extension for the fax machine on the Expansion. In this configuration, the T.38 fax settings are configured on the SIP line between the Expansion and the Server.

5.11.1. Analogue User

To configure the settings for the fax User, first navigate to **User** in the Navigation Pane for the Expansion. In the test environment, the 500V2 Expansion is called **GSSCP_IPO**. Select the **User** tab. The following example shows the configuration required for an analog Endpoint.

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



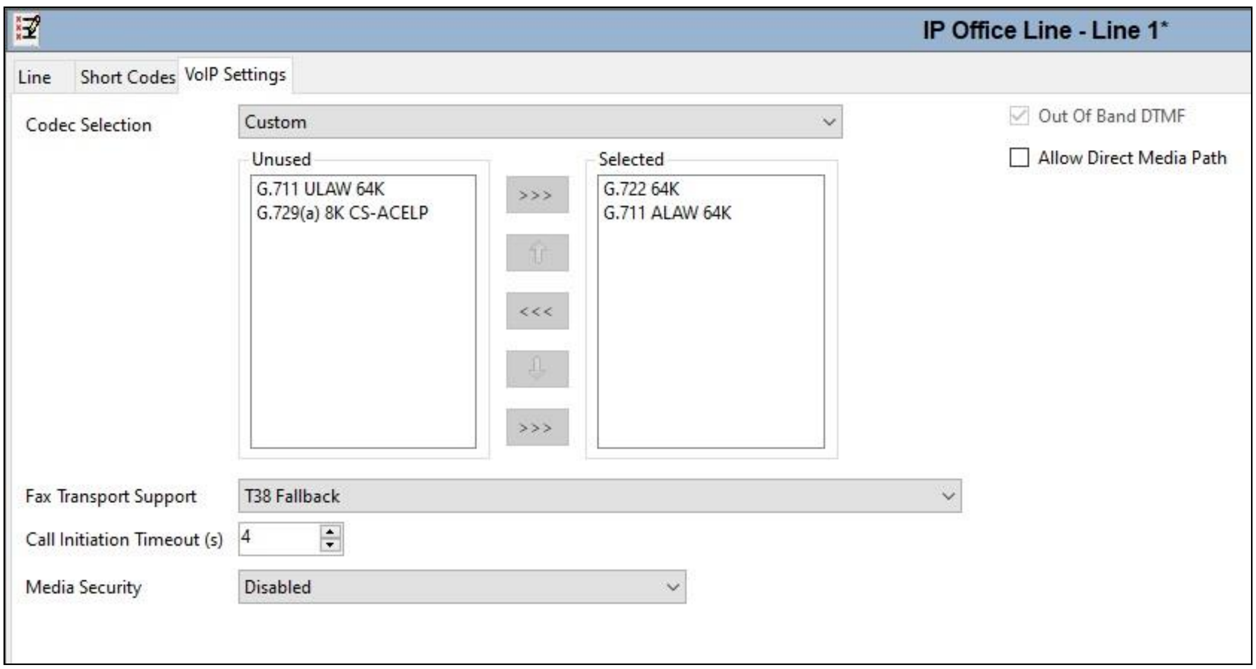
Configure other settings as described in **Section 5.8**.

5.11.2. T.38 Fallback Fax Settings

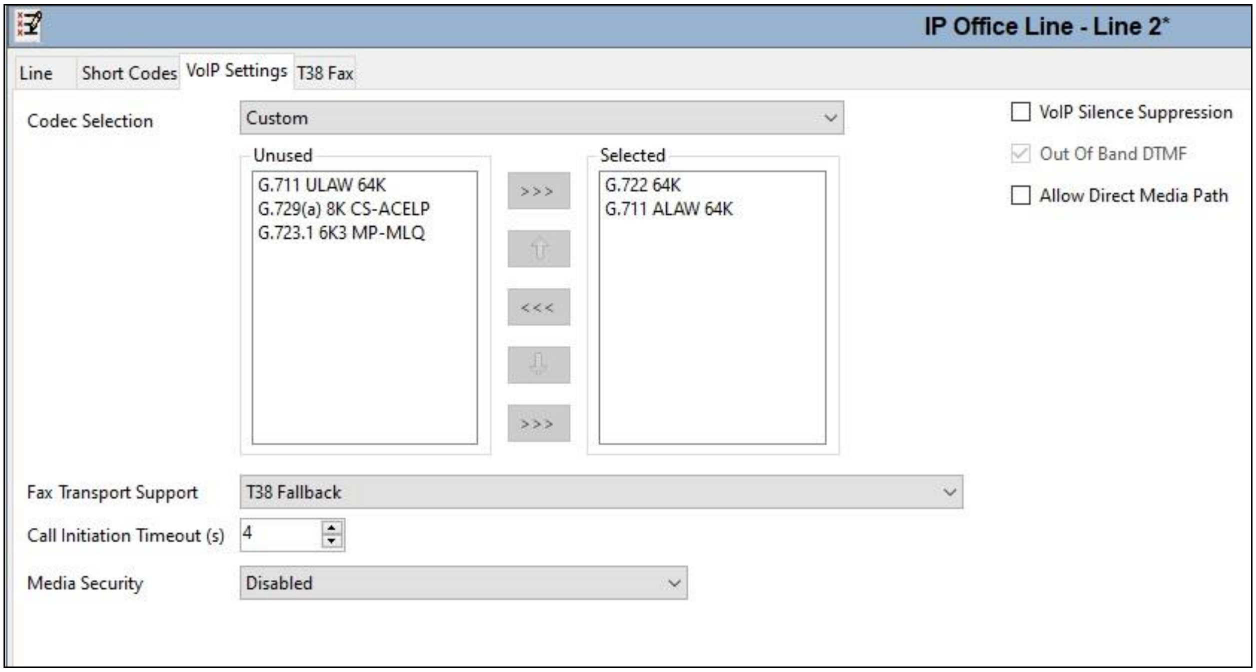
The T.38 Fallback Fax settings are defined on the SIP Line between the Expansion and the Server. Note that the VoIP settings for T.38 Fallback Fax are required in three places in this configuration:

- The SIP Line for the Mnet SIP Trunk as described in **Section 5.6.2**.
- The IP Office Line between the Server and the Expansion on the Expansion.
- The IP Office Line between the Server and the Expansion on the Server.

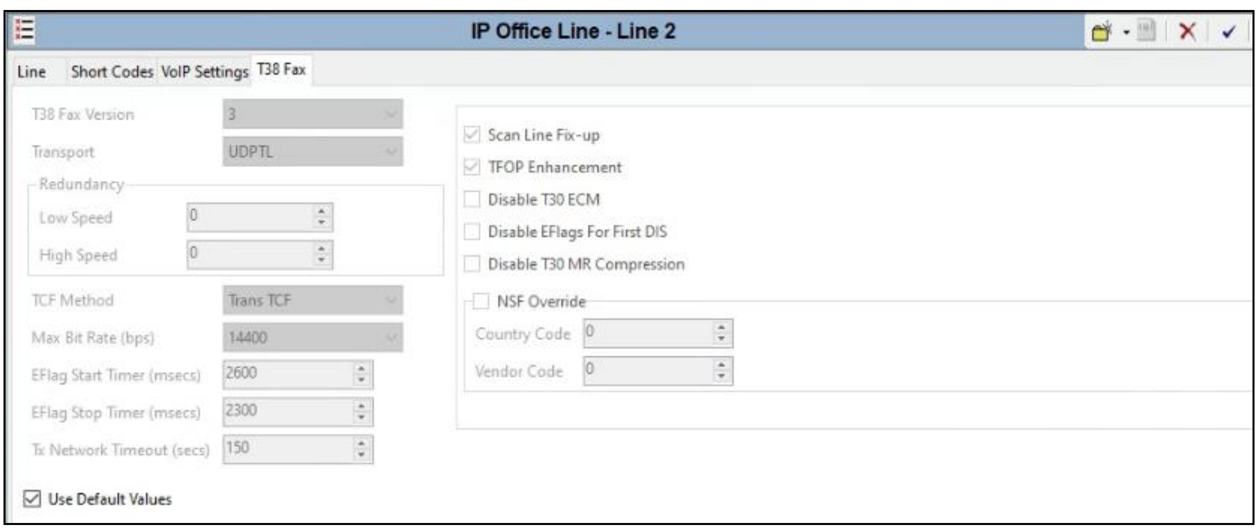
In all the above cases, the **Fax Transport Support** was set to **T38 Fallback**. Note: **Media Security** was set to **Disabled** as per **Section 2.2**. The following screenshot shows the VoIP Settings for the IP Office Line between the Server and the Expansion on the Server configuration:



The following shows the **VoIP Settings** tab in the IP Office Line for the Server in the Expansion configuration. Note: **Media Security** was set to **Disabled** as per **Section 2.2**.



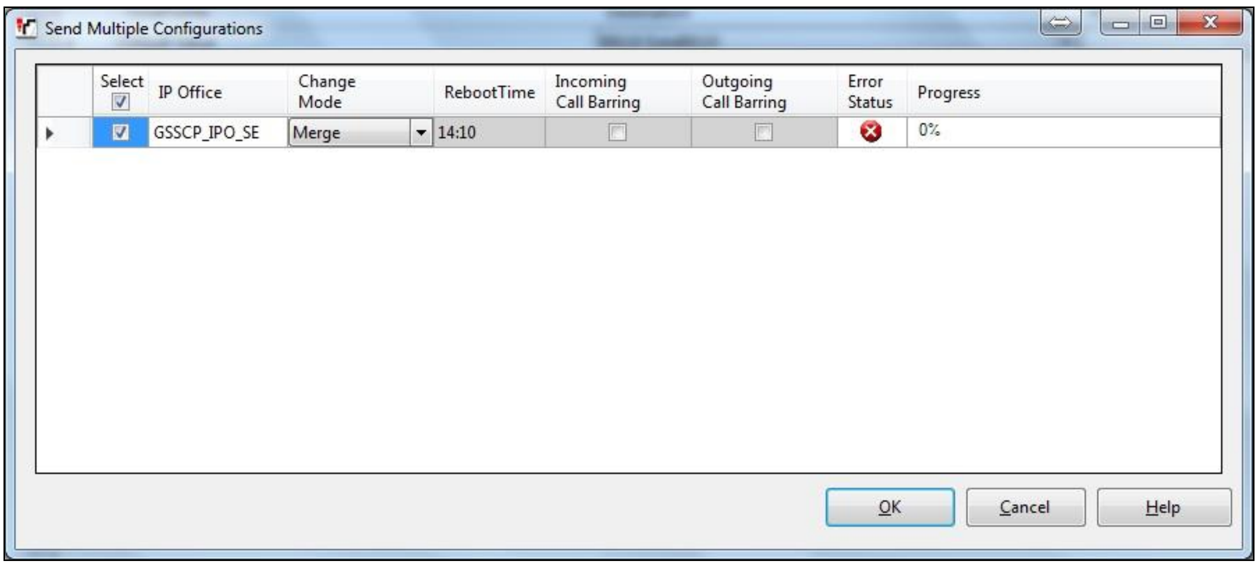
The following shows the T38 Fax tab in the IP Office Line for the Server in the Expansion configuration with **Use Default Values** enabled.



Refer to **Section 5.6.2** for the VoIP Settings on the SIP Line for the M-net Premium SIP Trunk.

5.12. 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
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, Reboot, Timed** or **RebootWhen Free** can be selected from the **Change Mode** drop-down menu 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.

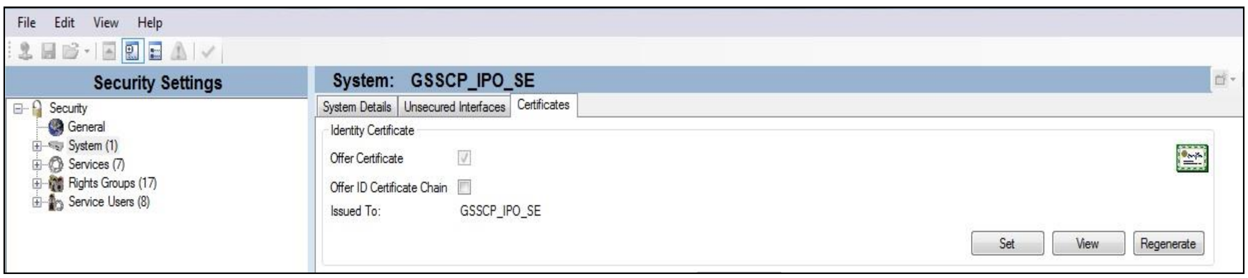


5.13. TLS Certificates

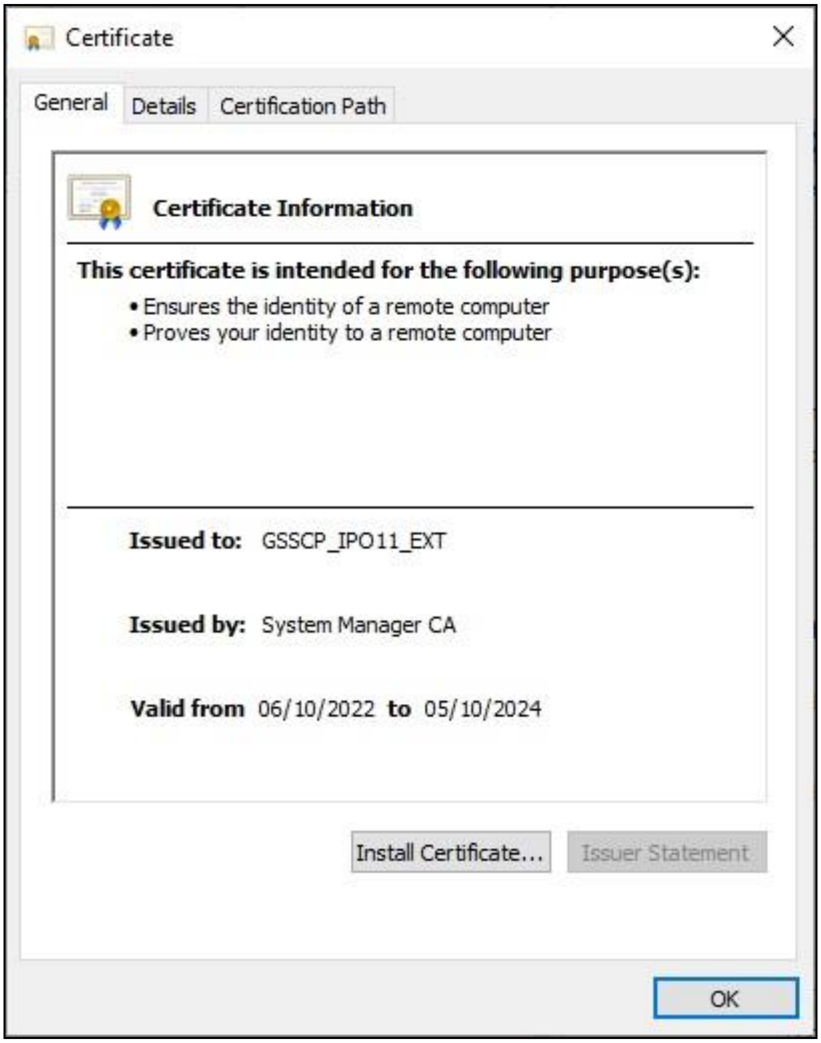
For the compliance test, TLS signalling was used internally to the enterprise wherever possible. Testing was done using identity certificates signed by a local certificate authority **System Manager CA**. The generation and installation of these certificates are beyond the scope of these Application Notes.

To view the certificate currently installed on IP Office, navigate to **File → Advanced → Security Settings**. In the Security Settings window, navigate to **Security → System** and select the **Certificates** tab.

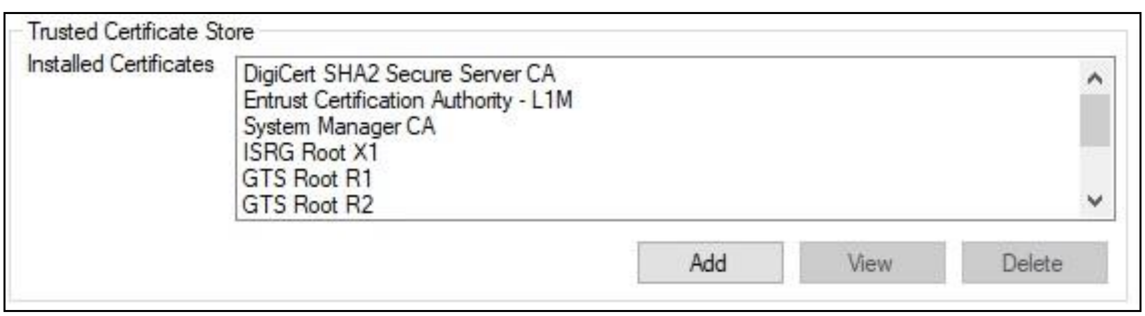
To verify the identity certificate, locate the **Identity Certificate** section and click **View** to see the details of the certificate.



A pop-up window displays the certificate that is issued to the Avaya IP Office (**GSSCP_IPO_SE**) and issued by **System Manager CA**. Click **OK** to close the pop-up window.



To verify the trusted certificates, return to the **Security → System →Certificates** tab and scroll down to the **Trusted Certificate Store** section. Verify that **System Manager CA** is displayed as an **Installed Certificates**.

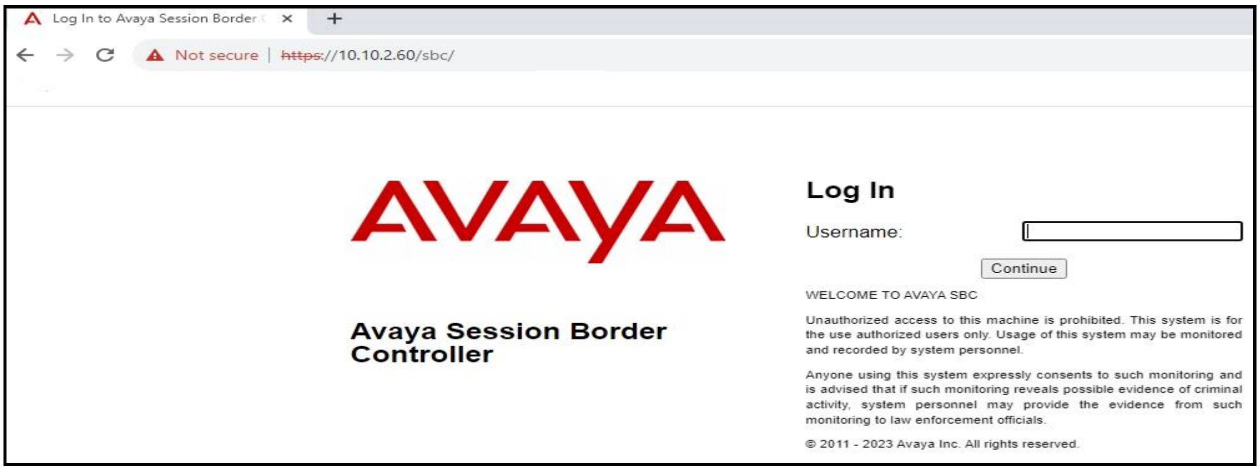


6. Configure Avaya Session Border Controller

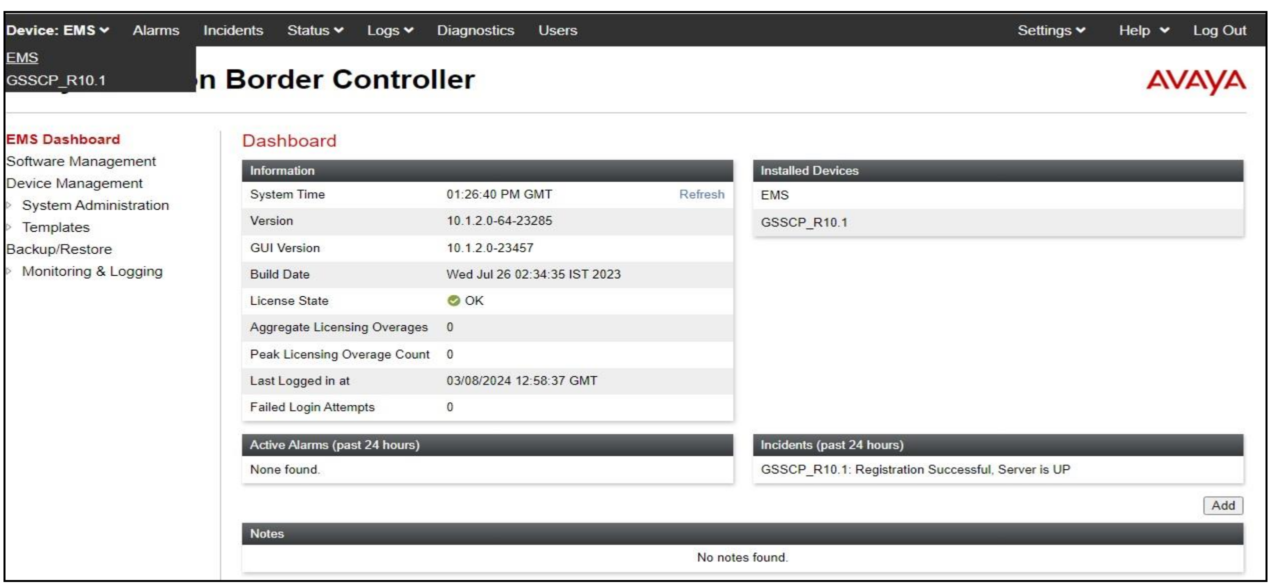
This section describes the configuration of the Session Border Controller (Avaya SBCE). The Avaya SBCE provides security and manipulation of signalling to provide an interface to the Service Provider's SIP Trunk that is standard where possible and adapted to the Service Provider's SIP implementation where necessary.

6.1. Accessing Avaya Session Border Controller

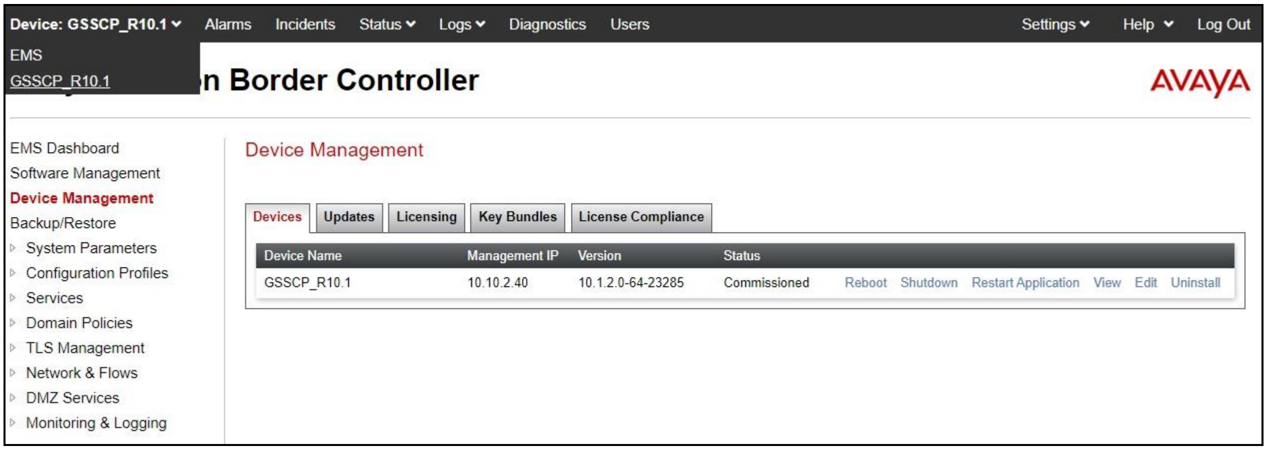
Access the Avaya SBCE using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the management IP address configured at installation and enter the **Username** and **Password**.



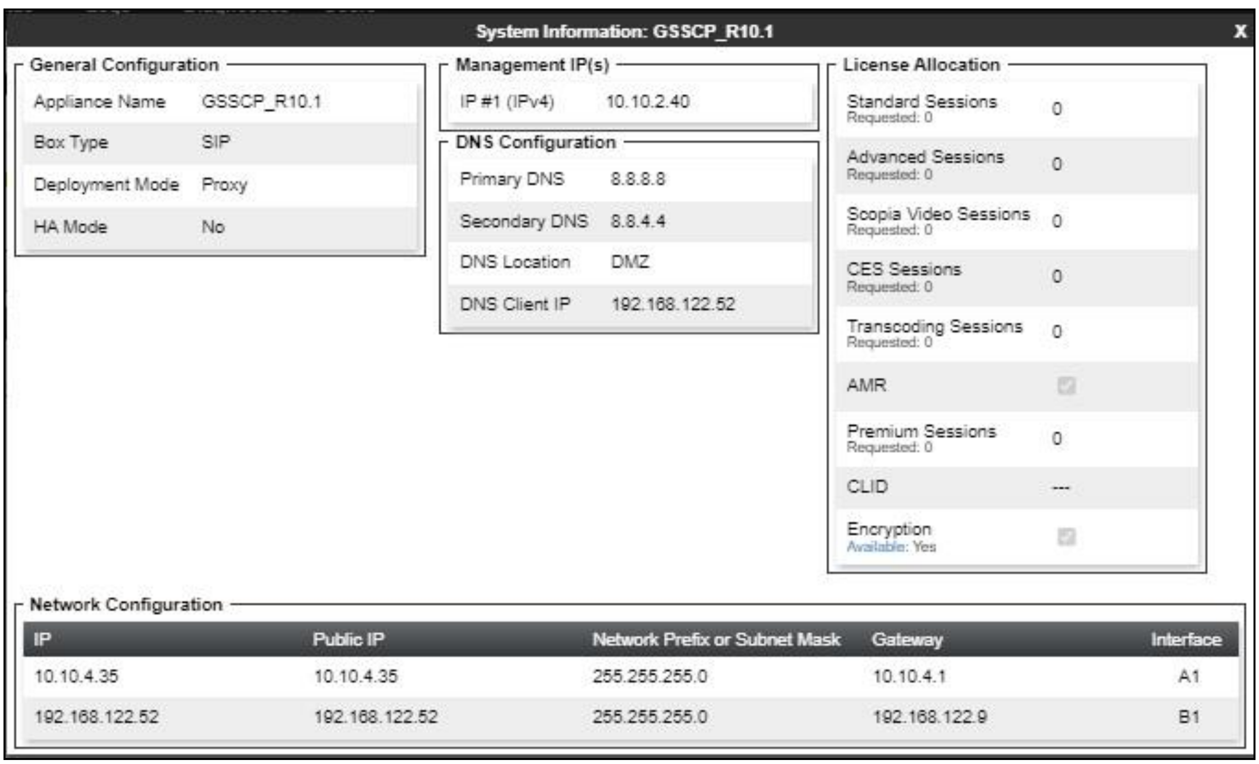
Once logged in, on the top-left of the screen, under **Device:** select the required device from the drop-down menu. with a menu on the left-hand side. In this case, **GSSCP_R10.1** is used as a starting point for all configuration of the Avaya SBCE.



To view system information that was configured during installation, navigate to **Device Management**. A list of installed devices is shown in the right pane. In the case of the sample configuration, a single device named **GSSCP_R10.1** is shown. To view the configuration of this device, click **View** (the third option from the right).



The **System Information** screen shows the **General Configuration**, **Device Configuration**, **License Allocation**, **Network Configuration**, **DNS Configuration** and **Management IP** information.

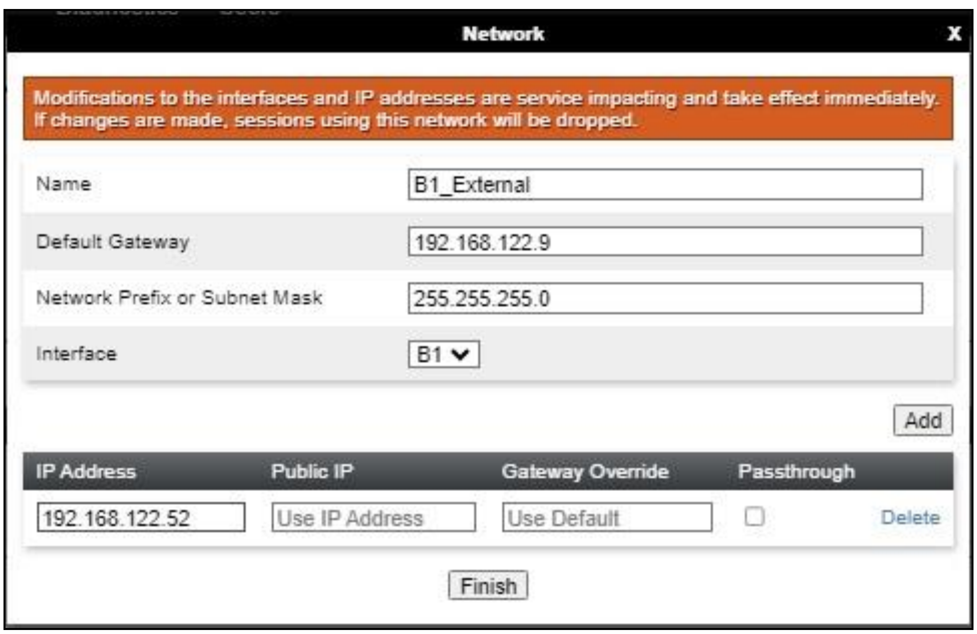


6.2. Define Network Management

Network information is required on the Avaya SBCE to allocate IP addresses and masks to the interfaces. Note that only the **A1** and **B1** interfaces are used, typically the **A1** interface is used for the internal side and **B1** is used for external. Each side of the Avaya SBCE can have only one physical interface assigned.

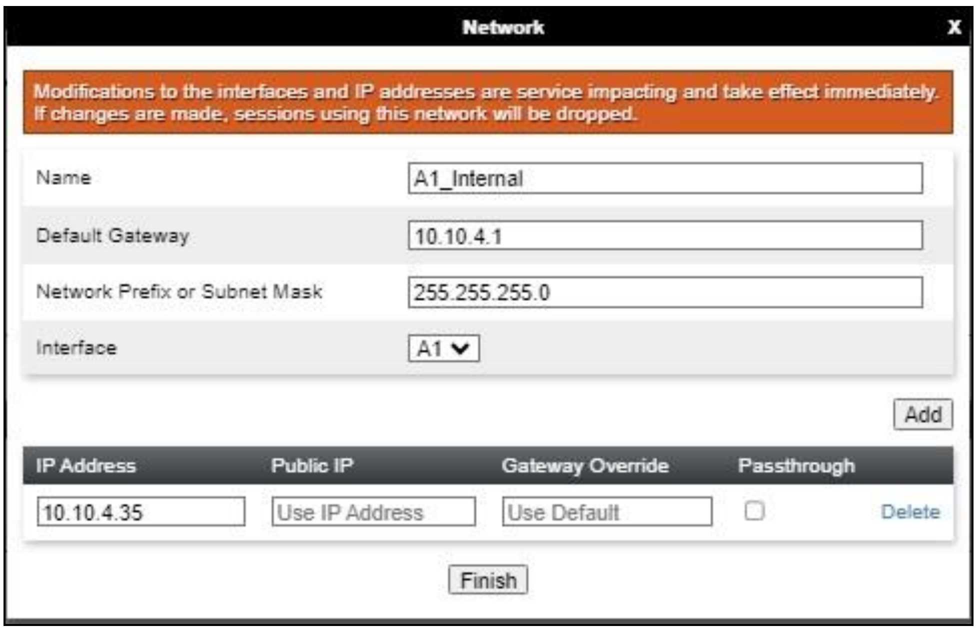
To define the network information, navigate to **Network & Flows → Network Management** in the main menu on the left-hand side and click on **Add**. Enter details for the external interfaces in the dialogue box:

- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the external interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the external physical interface to be used from the **Interface** drop down menu. In the test environment, this was **B1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the external IP address of the Avaya SBCE on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.

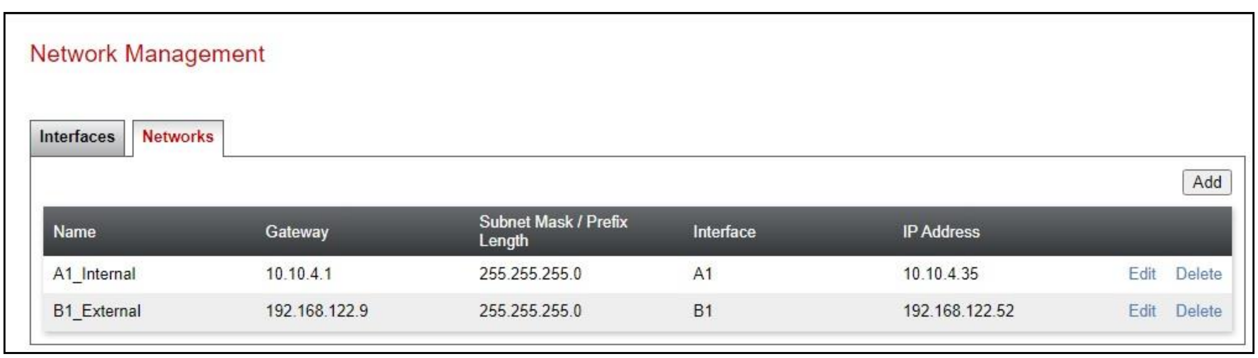


Click on **Add** to define the internal interfaces or Edit if it was defined during installation of the Avaya SBCE. Enter details in the dialogue box:

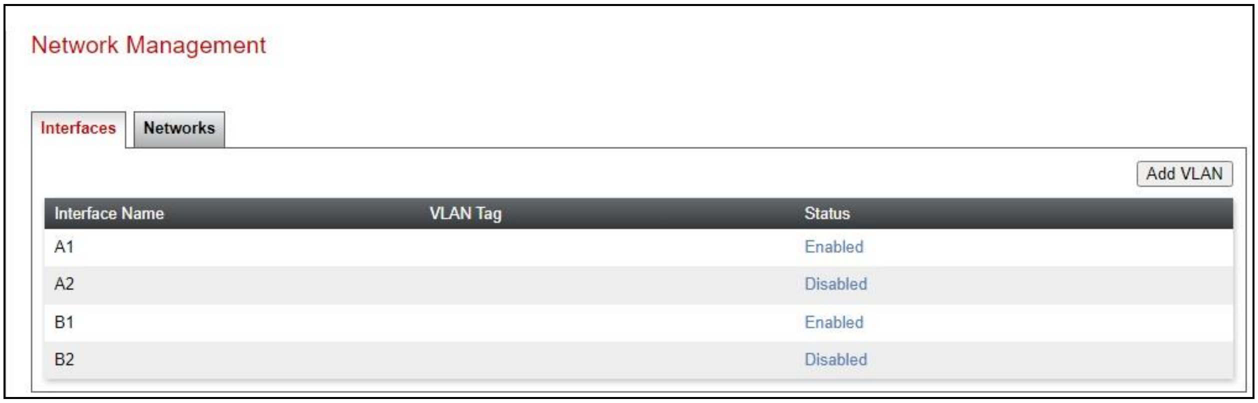
- Enter a descriptive name in the **Name** field.
- Enter the default gateway IP address for the internal interfaces in the **Default Gateway** field.
- Enter the subnet mask in the **Network Prefix or Subnet Mask** field.
- Select the internal physical interface to be used from the **Interface** drop down menu. In the test environment, this was **A1**.
- Click on **Add** and an additional row will appear allowing an IP address to be entered.
- Enter the internal IP address of the Avaya SBCE on the SIP trunk in the **IP Address** field and leave the **Public IP** and **Gateway Override** fields blank.
- Click on **Finish** to complete the interface definition.



The following screenshot shows the completed Network Management configuration:



Select the **Interfaces** tab and click on the **Status** of the physical interface to toggle the state. Change the state to **Enabled** where required.



Note: to ensure that the Avaya SBCE uses the interfaces defined, the Application must be restarted.

- Click on **Device Management** in the main menu (not shown).
- Select **Restart Application** indicated by an icon in the status bar (not shown).

A status box will appear that will indicate when the restart is complete.

6.3. Define TLS Profiles

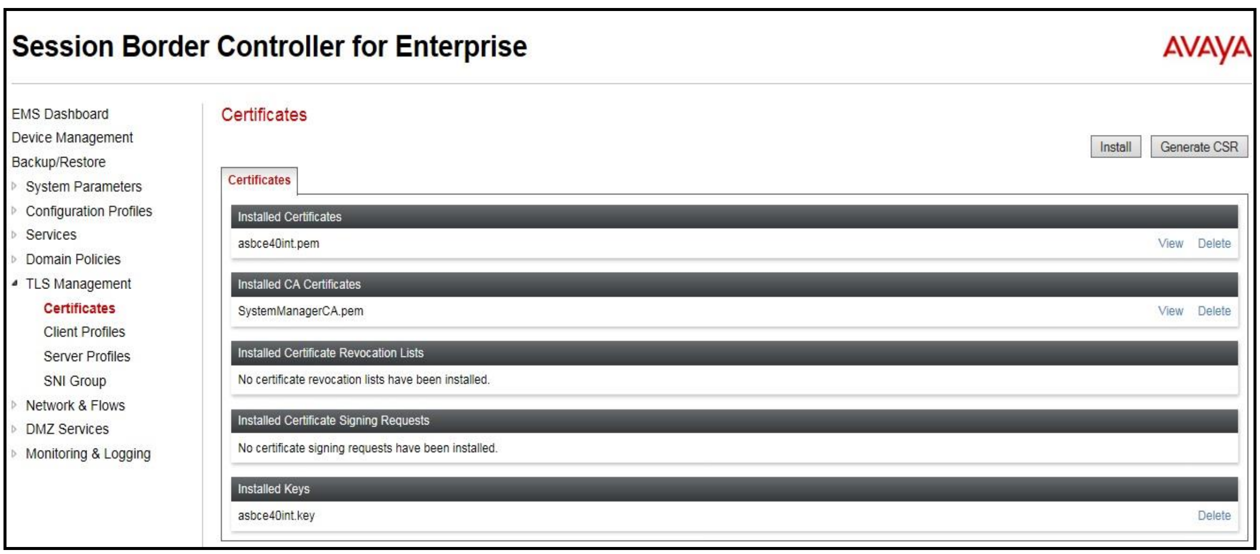
TLS management is required to install certificates and define client and server profiles so that the Avaya SBCE can connect securely with other network elements. For the compliance test, TLS transport is used for signalling on the SIP trunk between IP Office and the Avaya SBCE. Compliance testing was done using identity certificates signed by a local certificate authority. The generation and installation of these certificates are beyond the scope of these Application Notes.

The following procedures show how to view the certificates and configure the Client and Server profiles to support the TLS connection.

6.3.1. Certificates

To view the certificates currently installed on the Avaya SBCE, navigate to **TLS Management → Certificates**:

- Verify that an Avaya SBCE identity certificate (**asbce40int.pem**) is present under **Installed Certificates**.
- Verify that certificate authority root certificate (**SystemManagerCA.pem**) is present under **Installed CA certificates**.
- Verify that private key associated with the identity certificate (**asbce40int.key**) is present under **Installed Keys**.

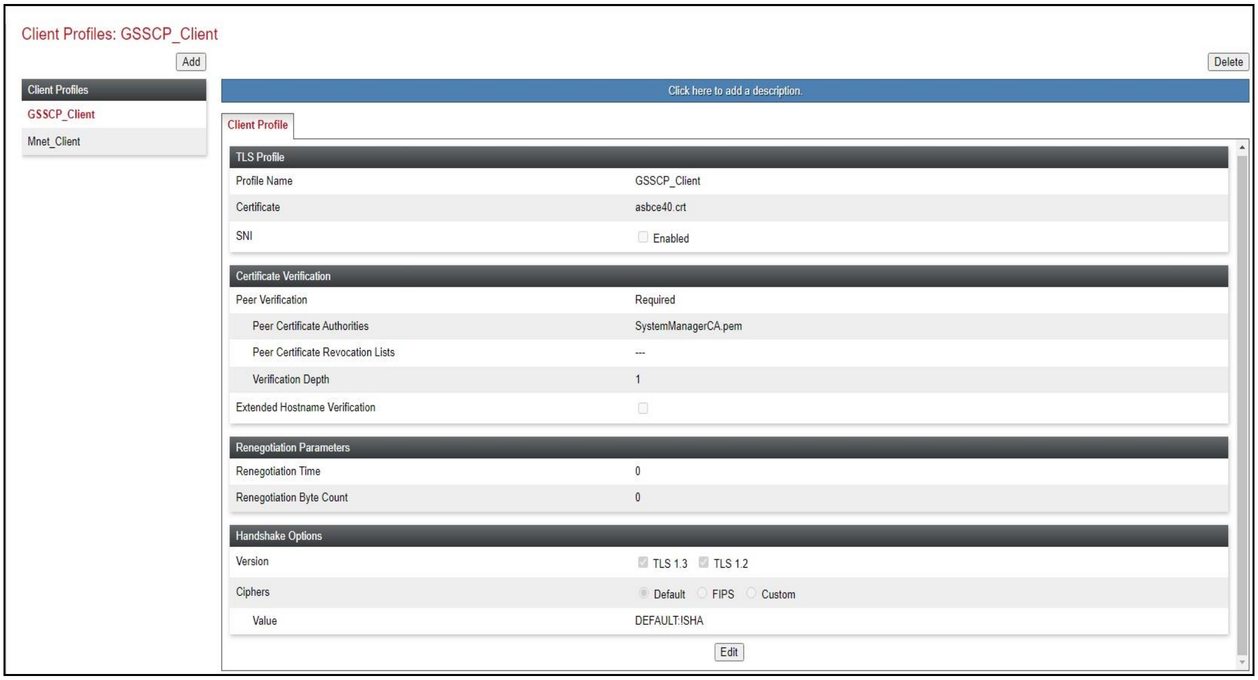


6.3.2. Client Profile

To create a new client profile, navigate to **TLS Management → Client Profile** in the left pane and click **Add** (not shown).

- Set **Profile Name** to a descriptive name. **GSSCP_Client** was used in the compliance testing.
- Set **Certificate** to the identity certificate **asbce40int.pem** used in the compliance testing.
- **Peer Verification** is automatically set to **Required**.
- Set **Peer Certificate Authorities** to the **SystemManagerCA.pem** identity certificate.
- Set **Verification Depth** to **1**.

Click **Next** to accept default values for the next screen and click **Finish** (not shown).

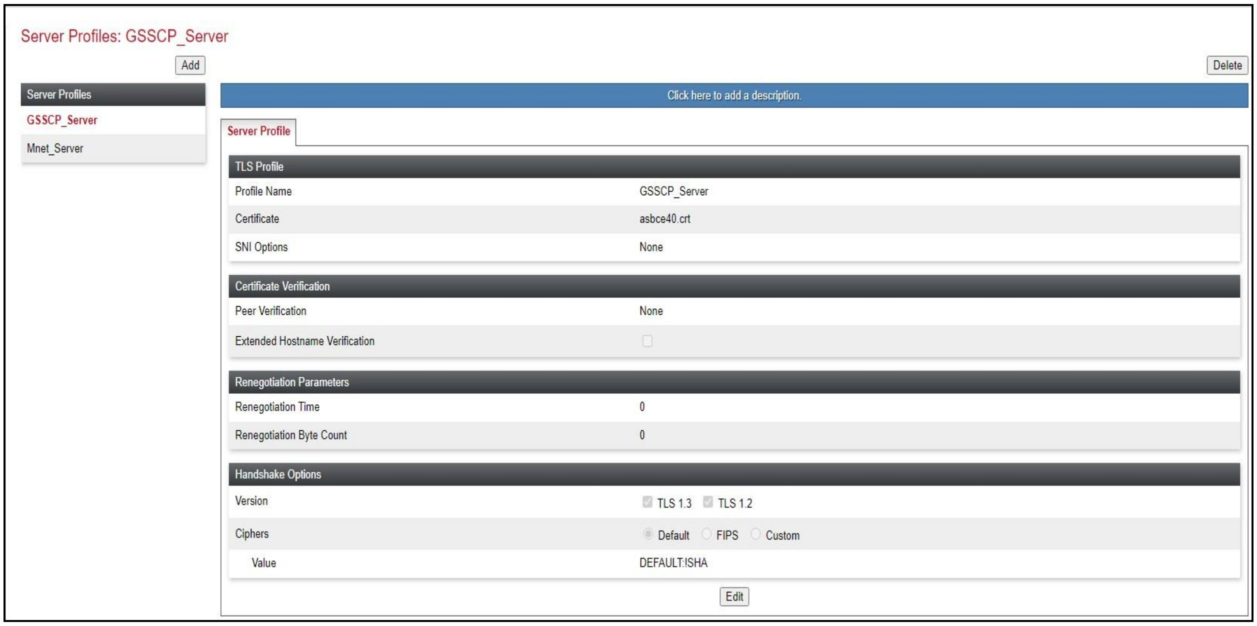


6.3.3. Server Profile

To create a new server profile, navigate to **TLS Management → Server Profile** in the left pane and click **Add** (not shown).

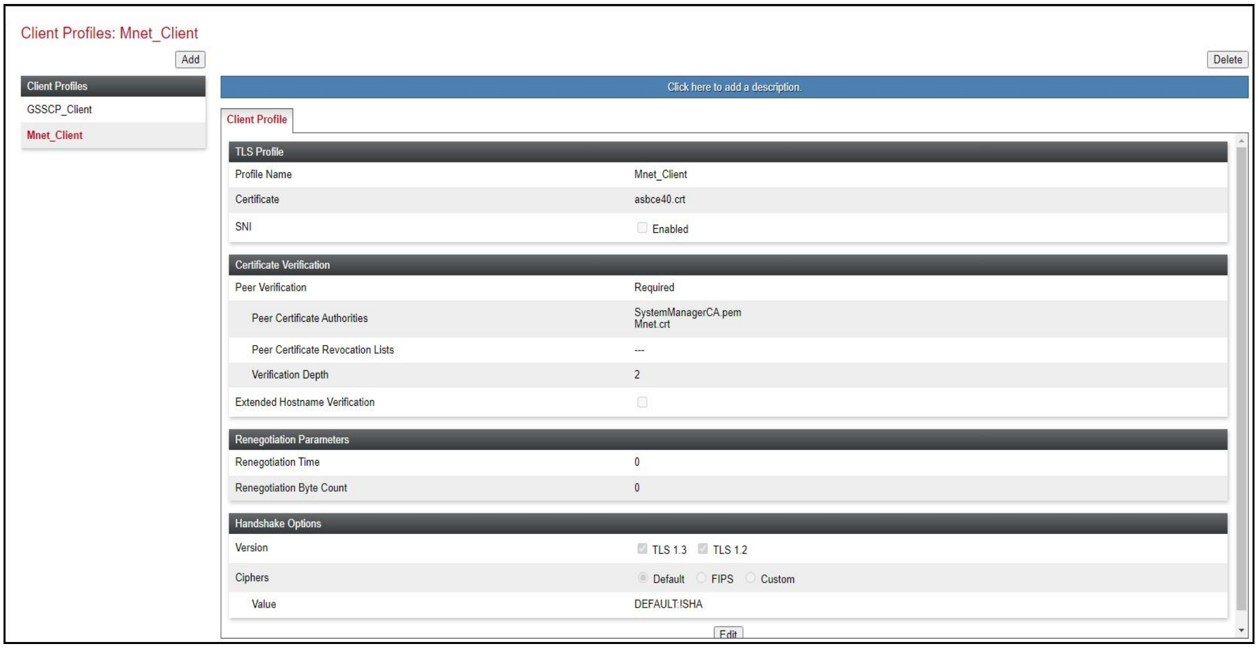
- Set **Profile Name** to a descriptive name. **GSSCP_Server** was used in the compliance testing
- Set **Certificate** to the identity certificate **asbce40int.pem** used in the compliance testing.
- Set **Peer Verification** to **None**.

Click **Next** to accept default values for the next screen and click **Finish** (not shown).

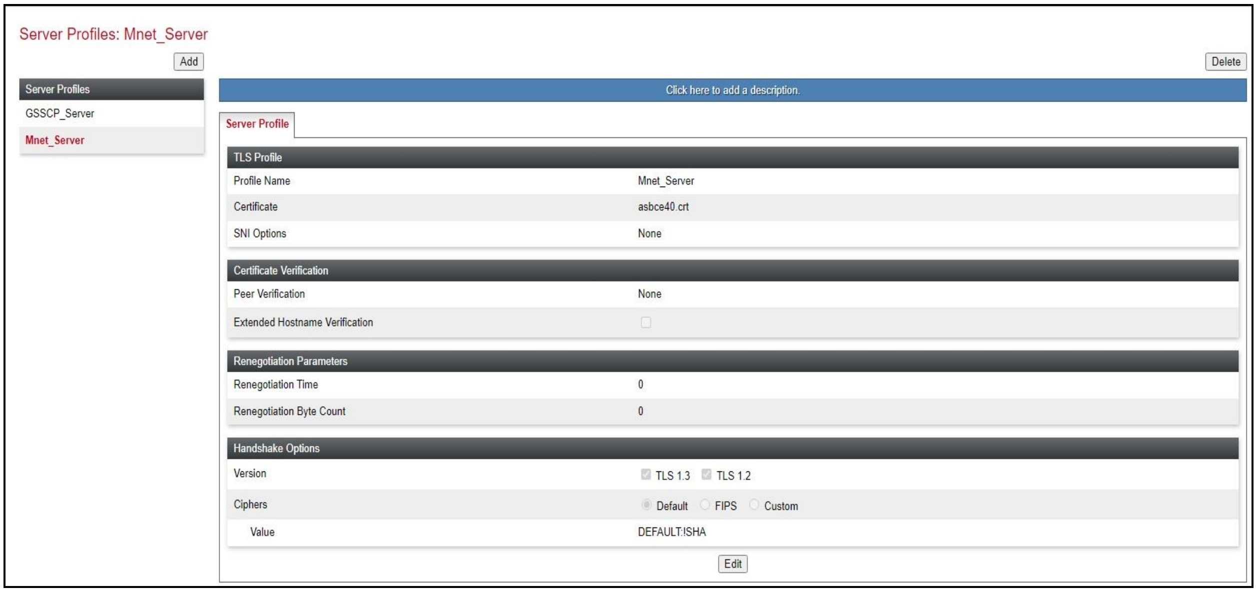


Note: Please contact a M-net representitive to obtain the necessary security certificates and installation information about applying certs to the Avaya SBCE. During compliance testing, test certificates were issued by M-net in order to encrypt the SIP trunk connection between the Avaya and M-net SIP platforms. The Client and Server Profiles details created with the M-net test certs are detailed in the screen shots below.

The following screen shows the Client Profile configured for M-net.



The following screen shows the Server Profile configured for M-net.



6.4. Define Interfaces

When the IP addresses and masks are assigned to the interfaces, these are then configured as signalling and media interfaces.

6.4.1. Signalling Interfaces

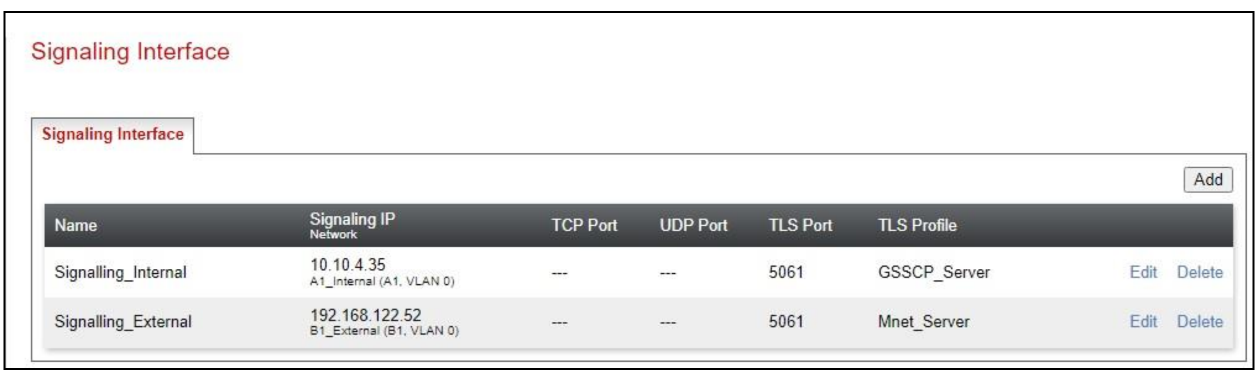
To define the signalling interfaces on the Avaya SBCE, navigate to **Network & Flows → Signaling Interface** from the menu on the left-hand side. Details of transport protocol and ports for the internal and external SIP signalling are entered here.

To enter details of transport protocol and ports for the SIP signalling on the internal interface:

- Select **Add** and enter details of the internal signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the interface.
- For **Signaling IP**, select the **A1_Internal** signalling interface IP addresses defined in **Section 6.2**.
- Select **TLS** port number, **5061** is used for IP Office.
- Select a **TLS Profile** defined in **Section 6.3.3** from the drop-down menu.
- Click **Finish**.

To enter details of transport protocol and ports for the SIP signalling on the external interface:

- Select **Add** and enter details of the external signalling interface in the pop-up menu (not shown).
- In the **Name** field enter a descriptive name for the external signalling interface.
- For **Signaling IP**, select the **B1_external** signalling interface IP address defined in **Section 6.2**.
- Select **TLS** port number, **5061** is used for the M-net Premium SIP Trunk.
- Select a **TLS Profile** defined in **Section 6.3.3** from the drop-down menu.
- Click **Finish**.


Signaling Interface

Signaling Interface

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile		
Signalling_Internal	10.10.4.35 A1_Internal (A1, VLAN 0)	---	---	5061	GSSCP_Server	Edit	Delete
Signalling_External	192.168.122.52 B1_External (B1, VLAN 0)	---	---	5061	Mnet_Server	Edit	Delete

6.4.2. Media Interfaces

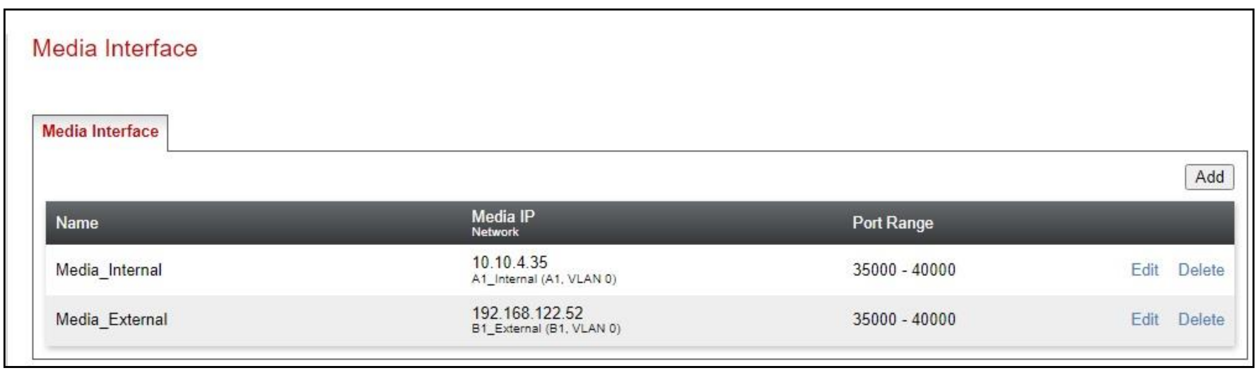
To define the media interfaces on the Avaya SBCE, navigate to **Network & Flows → Media Interface** from the menu on the left-hand side. Details of the RTP and SRTP port ranges for the internal and external media streams are entered here. The IP addresses for media can be the same as those used for signalling.

To enter details of the media IP and RTP port range for the internal interface to be used in the server flow:

- Select **Add Media Interface** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the internal media interface.
- For **Media IP**, select the **A1_Internal** media interface IP address defined in **Section 6.2**.
- For **Port Range**, enter **35000-40000**.
- Click **Finish**.

To enter details of the media IP and RTP port range on the external interface to be used in the server flow.

- Select **Add Media Interface** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the external media interface.
- For **Media IP**, select the **B1_External** media interface IP address defined in **Section 6.2**.
- Select **Port Range**, enter **35000-40000**.
- Click **Finish**.



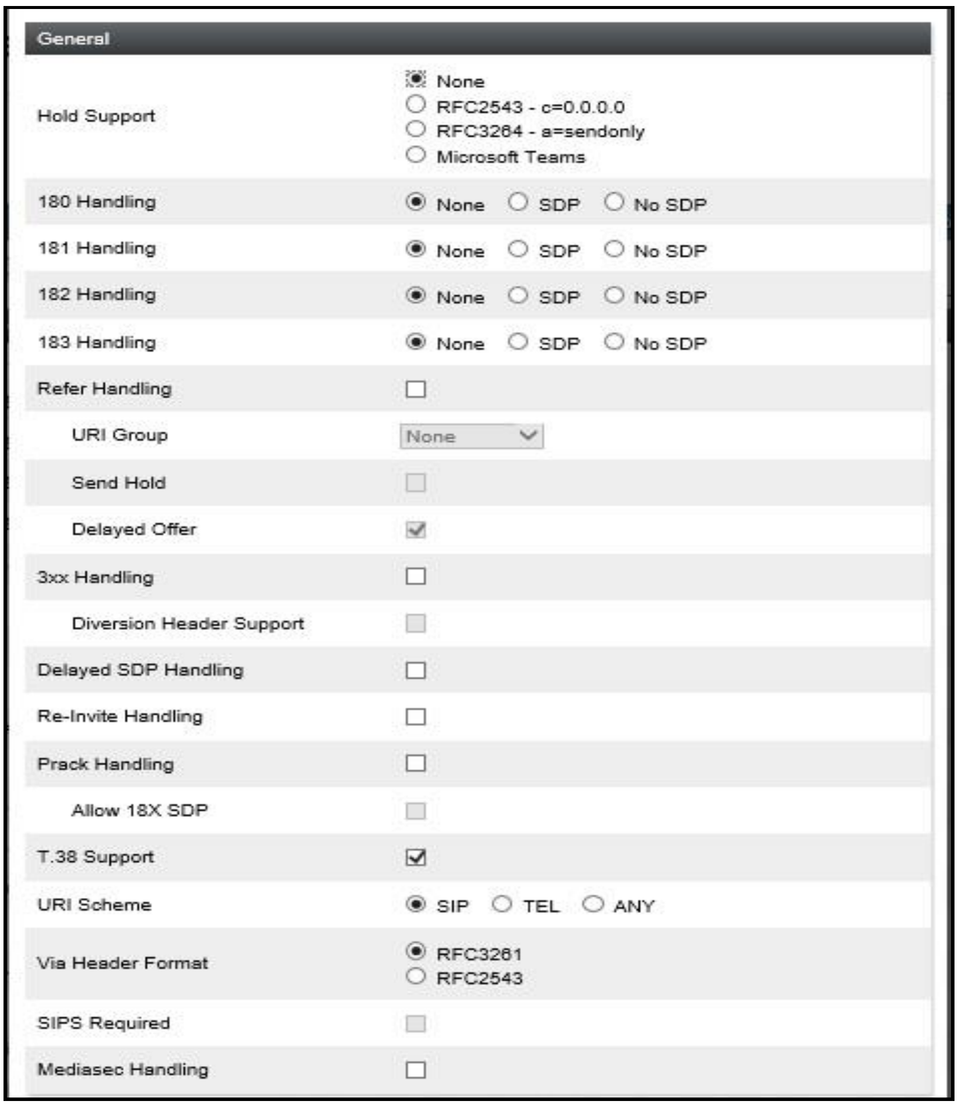
6.5. Define Server Interworking

Server interworking is defined for each server connected to the Avaya SBCE. In this case, M-net is connected as the Trunk Server and the IP Office is connected as the Call Server.

6.5.1. Server Interworking Avaya

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Configuration Profiles → Server Interworking** and click on **Add**.

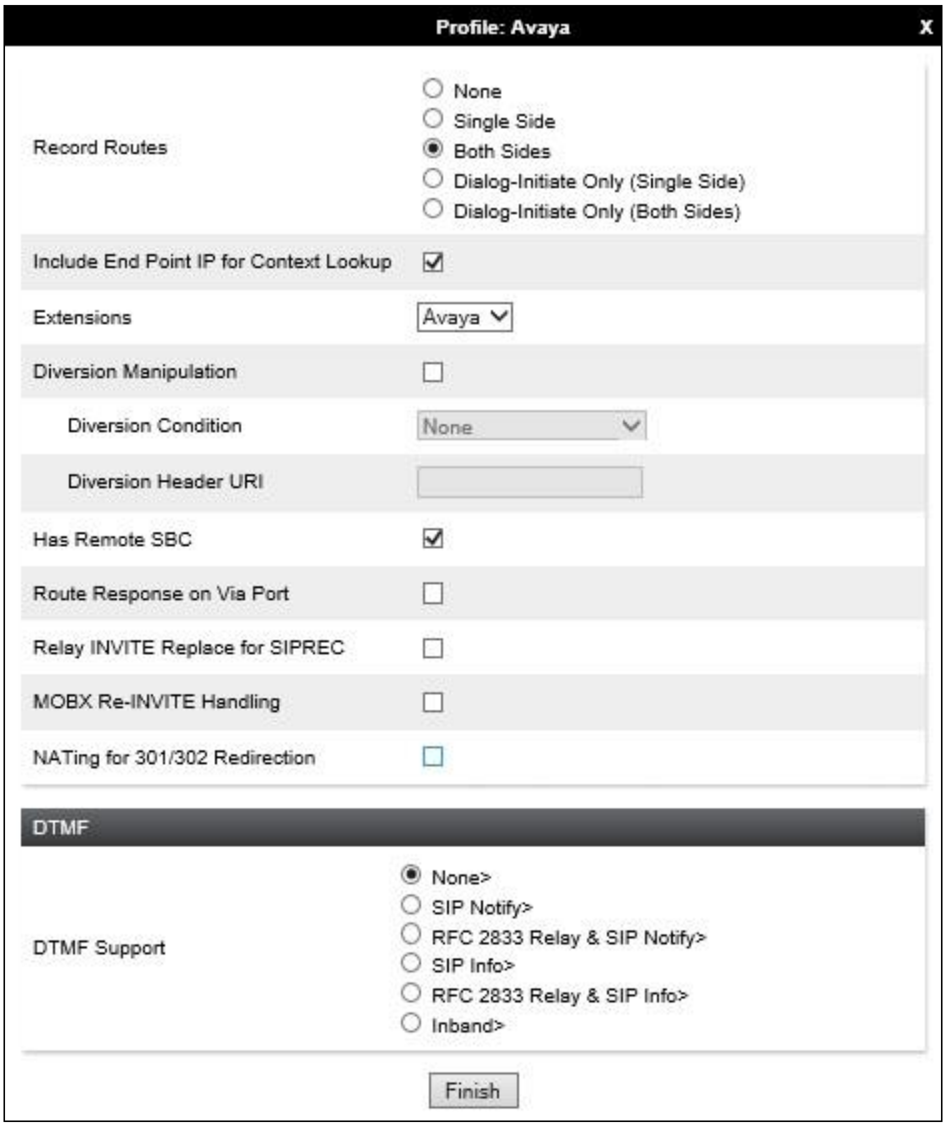
- Enter profile name such as Avaya and click **Next** (Not Shown).
- Check **Hold Support = None**.
- Check **T38 Support**.
- All other options on the **General** Tab can be left at default.



On the **Advanced** Tab:

- Check **Record Routes** = **Both Sides**.
- Ensure **Extensions** = **Avaya**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

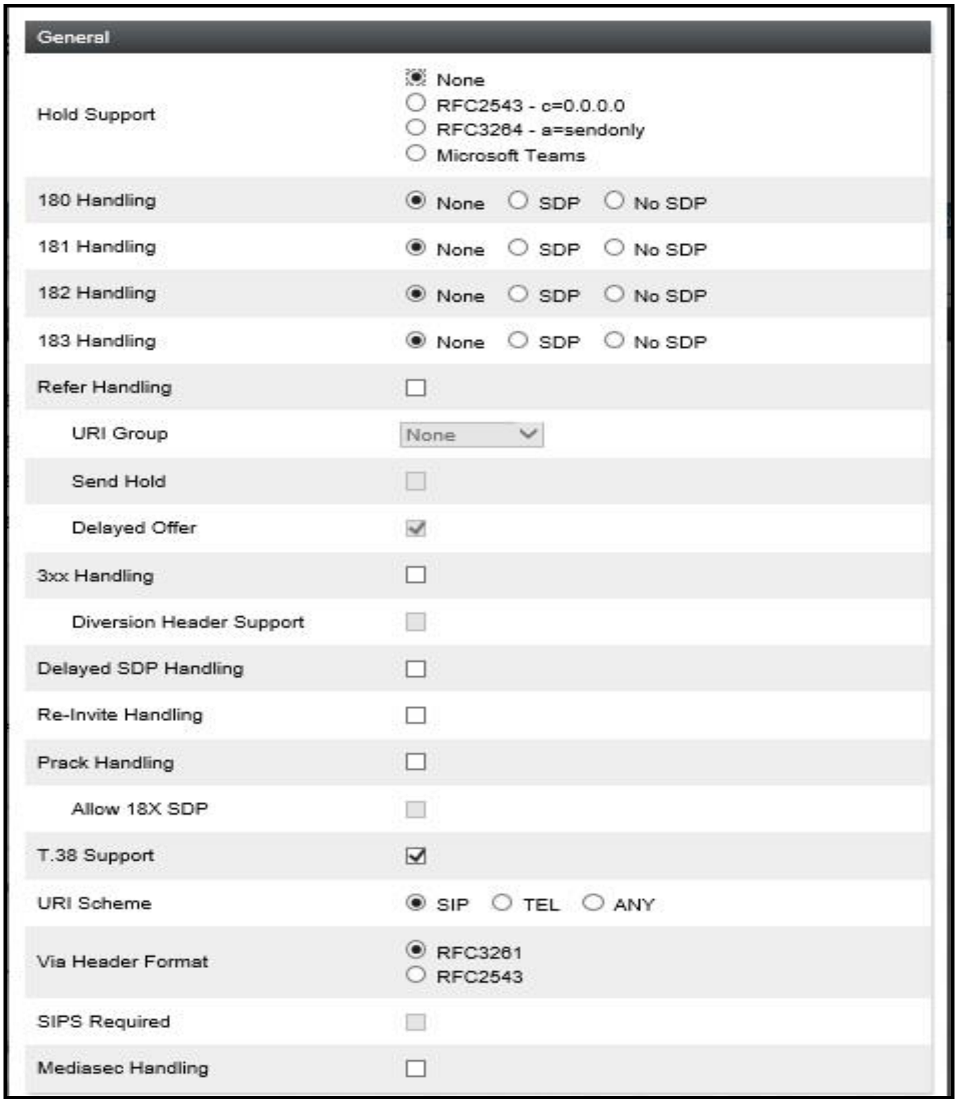
Click **Finish**.



6.5.2. Server Interworking – M-net

Server Interworking allows the configuration and management of various SIP call server-specific capabilities such as call hold and T.38. From the left-hand menu select **Configuration Profiles → Server Interworking** and click on **Add**.

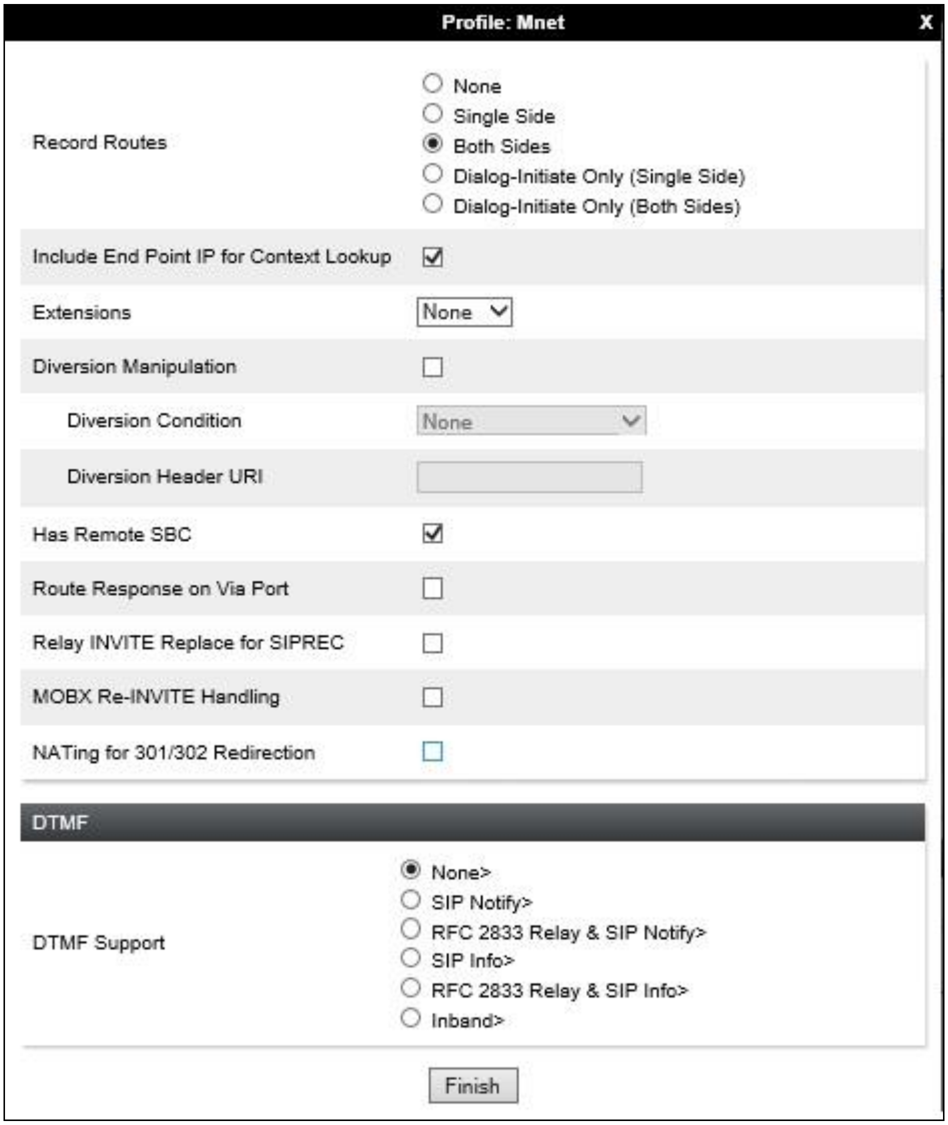
- Enter profile name such as **Mnet** and click **Next** (Not Shown).
- Check **Hold Support = None**.
- Check **T38 Support**.
- All other options on the **General** Tab can be left at default.



On the **Advanced** Tab:

- Check **Record Routes** = **Both Sides**.
- Ensure **Extensions** = **None**.
- Check **Has Remote SBC**.
- All other options on the **Advanced** Tab can be left at default.

Click **Finish**.



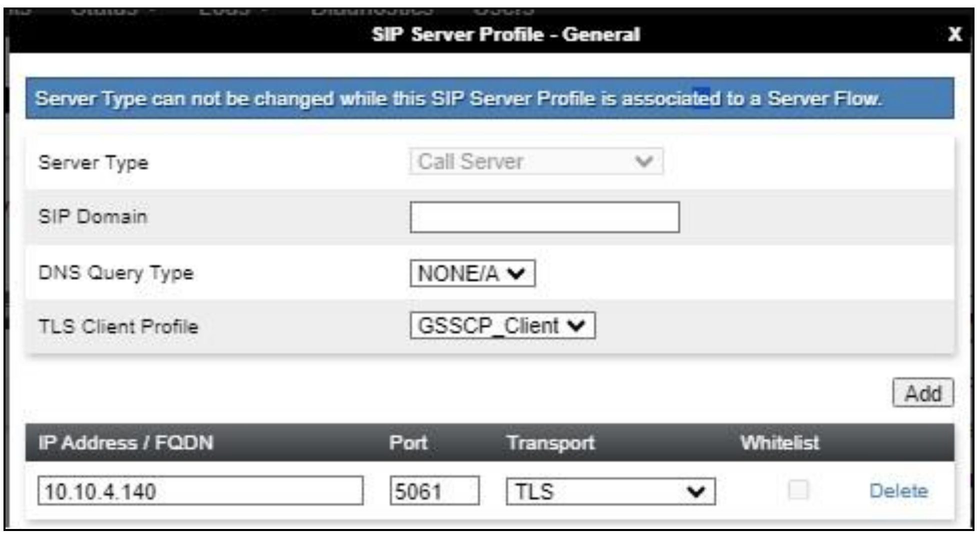
6.6. Define Servers

Servers are defined for each server connected to the Avaya SBCE. In this case, M-net is connected as the Trunk Server and IP Office is connected as the Call Server.

6.6.1. Server Configuration – Avaya

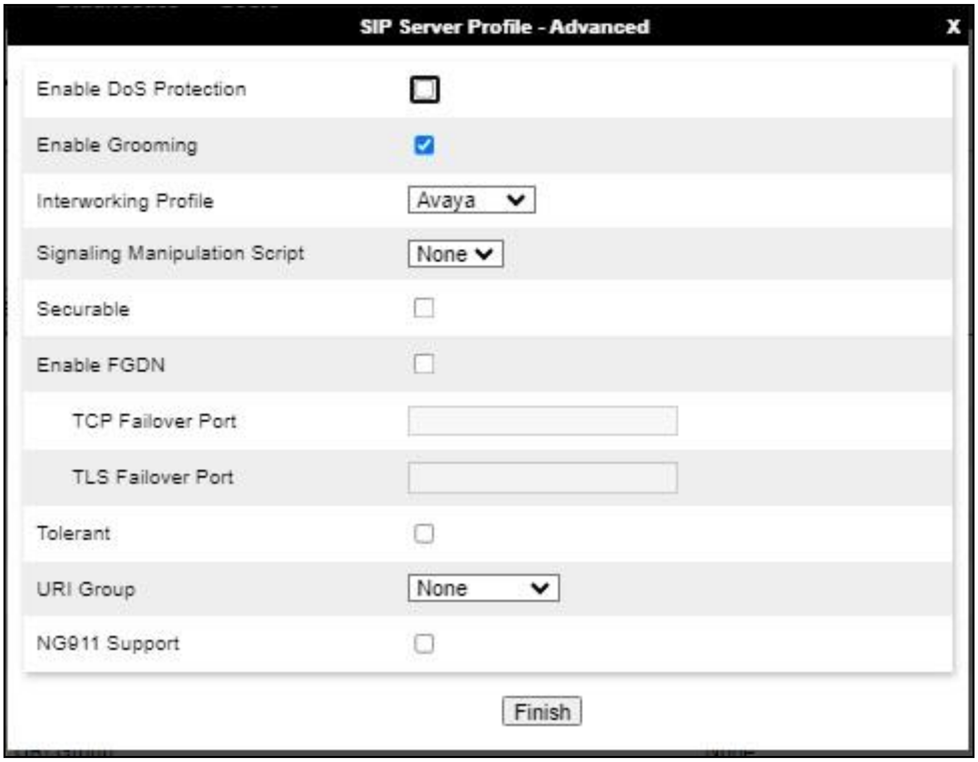
From the left-hand menu select **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profiles** tab, set the following:

- Select **Server Type** to be **Call Server**.
- Select **TLS Client Profile** to be **GSSCP_Client** as defined in **Section 6.3.2**.
- Enter **IP Address / FQDN** to **10.10.4.140** (IP Office IP Address).
- For **Port**, enter **5061**.
- For **Transport,** select **TLS**.
- Click on **Next** (not shown) to use default entries on the **Authentication** and **Heartbeat** tabs.



On the **Advanced** tab:

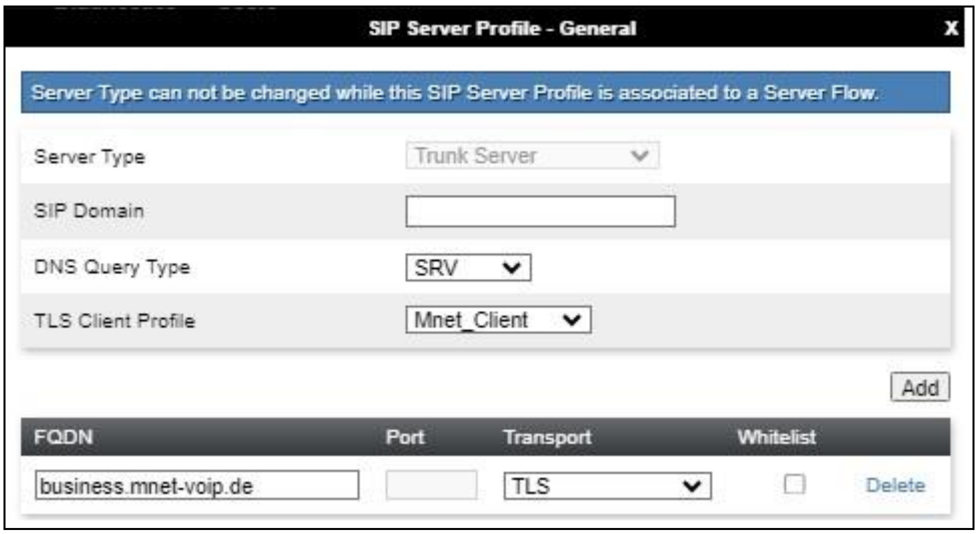
- Check **Enable Grooming**.
- Select **Avaya** for **Interworking Profile**.
- Click **Finish**.



6.6.2. Server Configuration – M-net

To define the M-net Trunk Server, navigate to **Services → SIP Servers** and click on **Add** and enter a descriptive name. On the **Add Server Configuration Profile** tab, set the following:

- Select **Server Type** to be **Trunk Server**.
- Set **DNS Query Type** to **SRV**.
- Select **TLS Client Profile** to be **Mnet_Client** as defined in **Section 6.3.2**.
- Enter **IP Address / FQDN** to **business.mnet-voip.de** (M-net SIP Platform).
- For **Transport,** select **TLS**.
- Click on **Next** (not shown).



In the new Authentication window that appears, enter the following values as M-net require authentication to connect to their network:

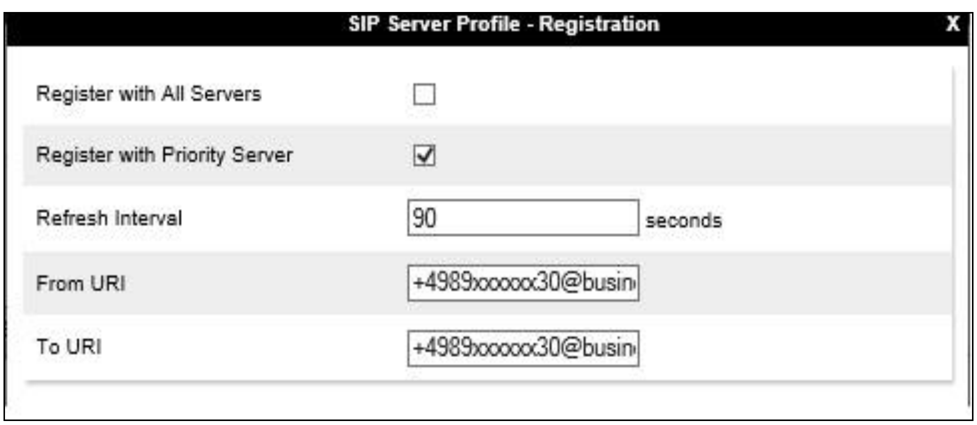
- **Enabled Authentication:** Checked
- **User Name:** Enter username provided by the Service Provider.
- **Realm:** Enter realm details provided by the Service Provider or leave blank to be detected by the server challenge.
- **Password** Enter password provided by the Service Provider.
- **Confirm Password** Re-enter password provided by the Service Provider.

Click **Next** to continue (not shown).

In the new Registration window that appears, enter the following values.

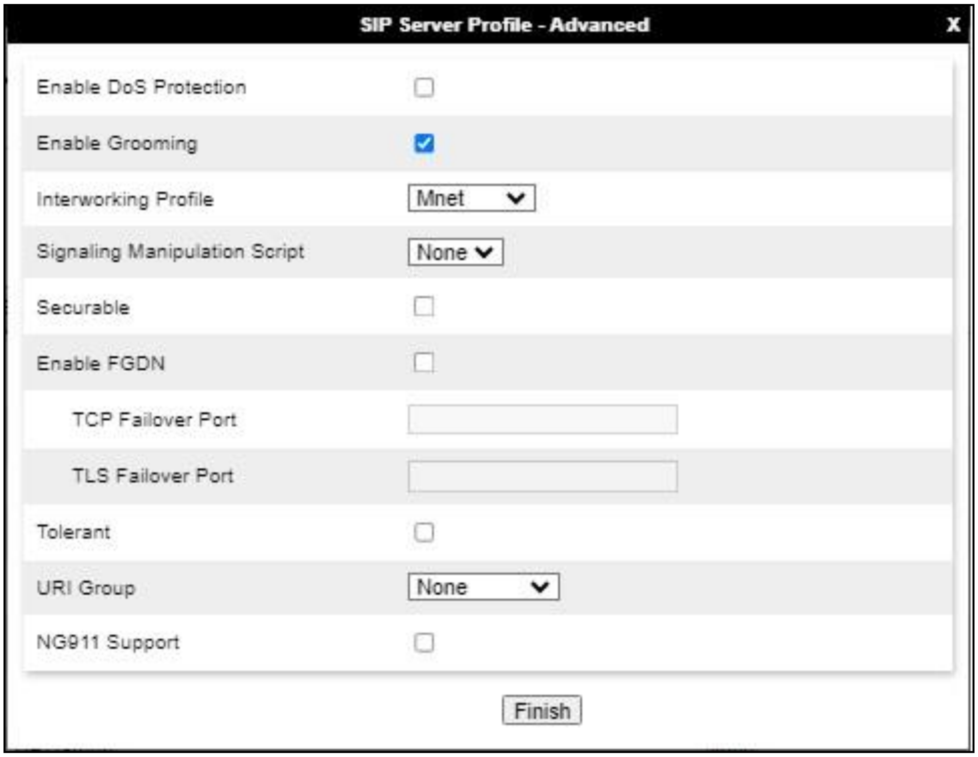
- **Register with Priority Server:** Check.
- **Refresh Interval** Choose the desired frequency in seconds the Avaya SBCE will send SIP REGISTERS.
- **From URI:** Enter an URI to be sent in the FROM header for SIP REGISTERS.
- **TO URI:** Enter an URI to be sent in the TO header for SIP REGISTERS.

Click **Next** to continue (not shown).



On the Advanced tab:

- Select **Mnet** for **Interworking Profile**.
- Check **Enable Grooming**.
- Click **Finish**.



6.7. Routing

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

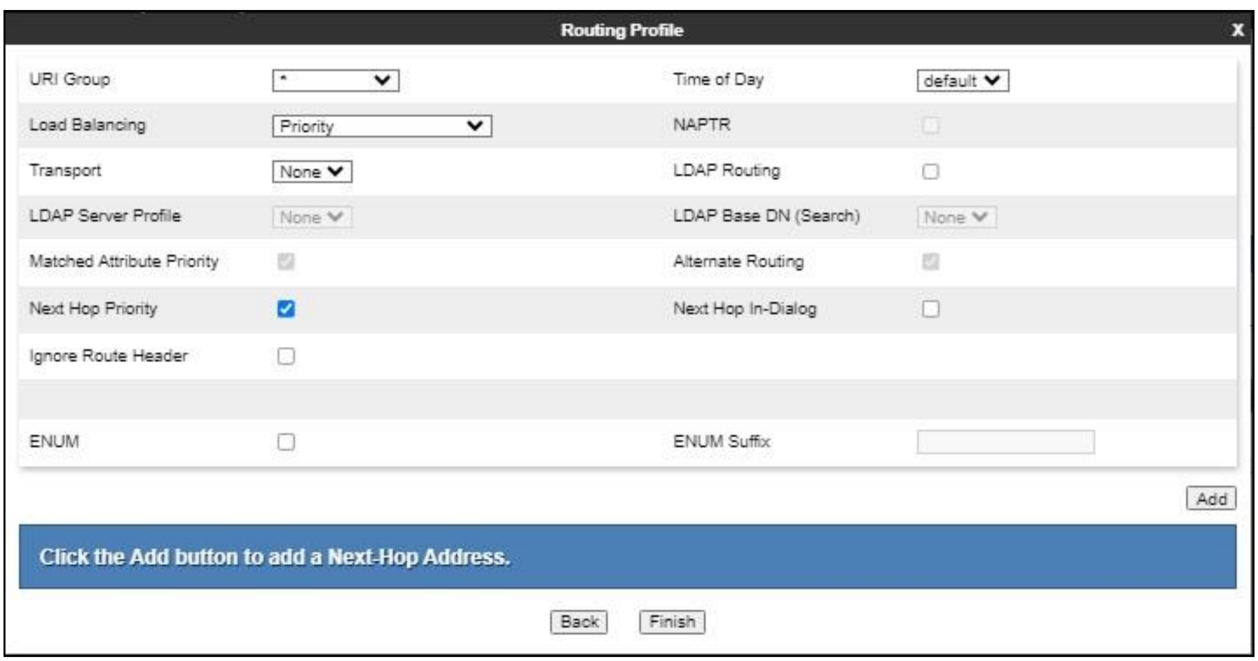
Routing information is required for routing to IP Office on the internal side and M-net address on the external side. The IP addresses and ports defined here will be used as the destination addresses for signalling. If no port is specified in the **Next Hop IP Address**, default 5060 is used.

6.7.1. Routing – Avaya

Create a Routing Profile for IP Office.

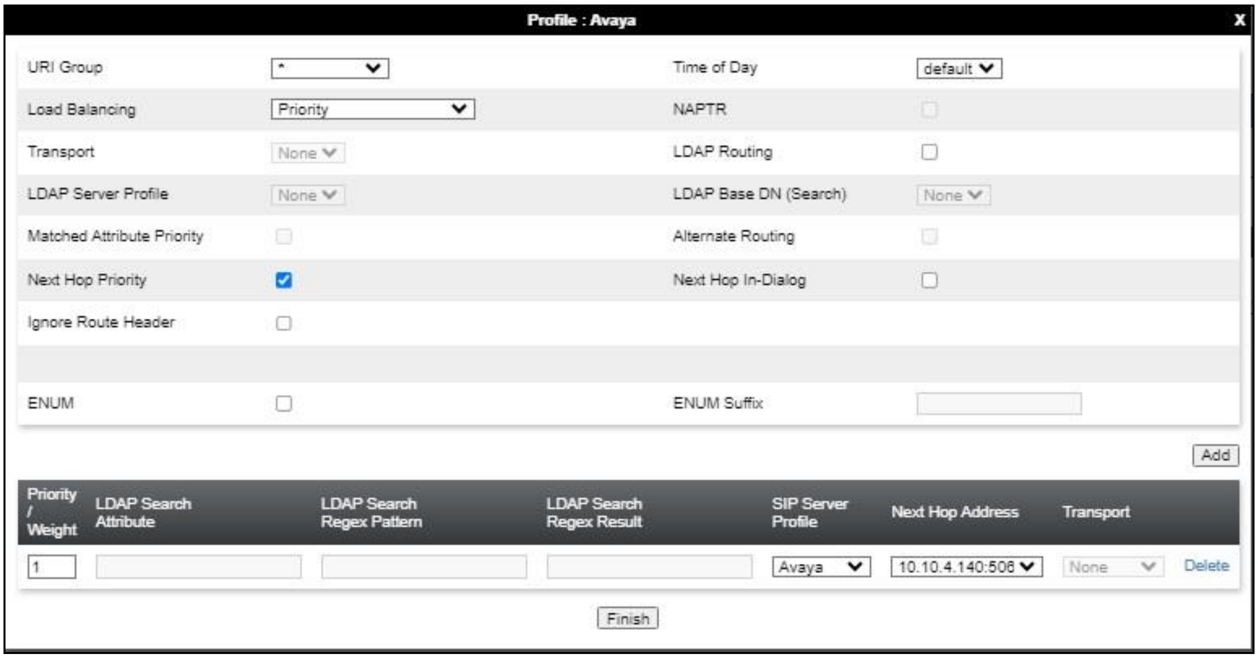
- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

The Routing Profile window will open. Use the default values displayed and click **Add**.



On the **Next Hop Address** window, set the following:

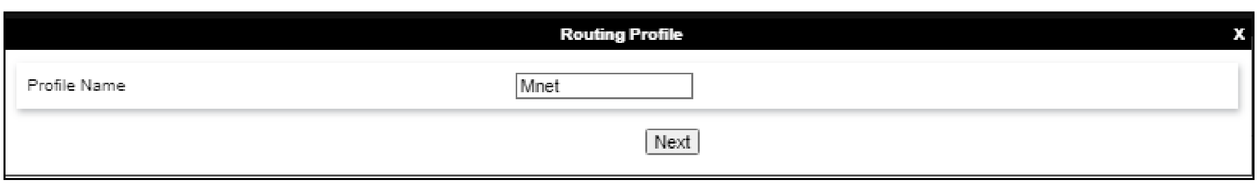
- **Priority/Weight** = **1**.
- **SIP Server Profile** = **Avaya** (**Section 6.6.1**) from drop down menu.
- **Next Hop Address** = Select **10.10.4.140:5061 (TLS)** from drop down menu.
- Click **Finish**.



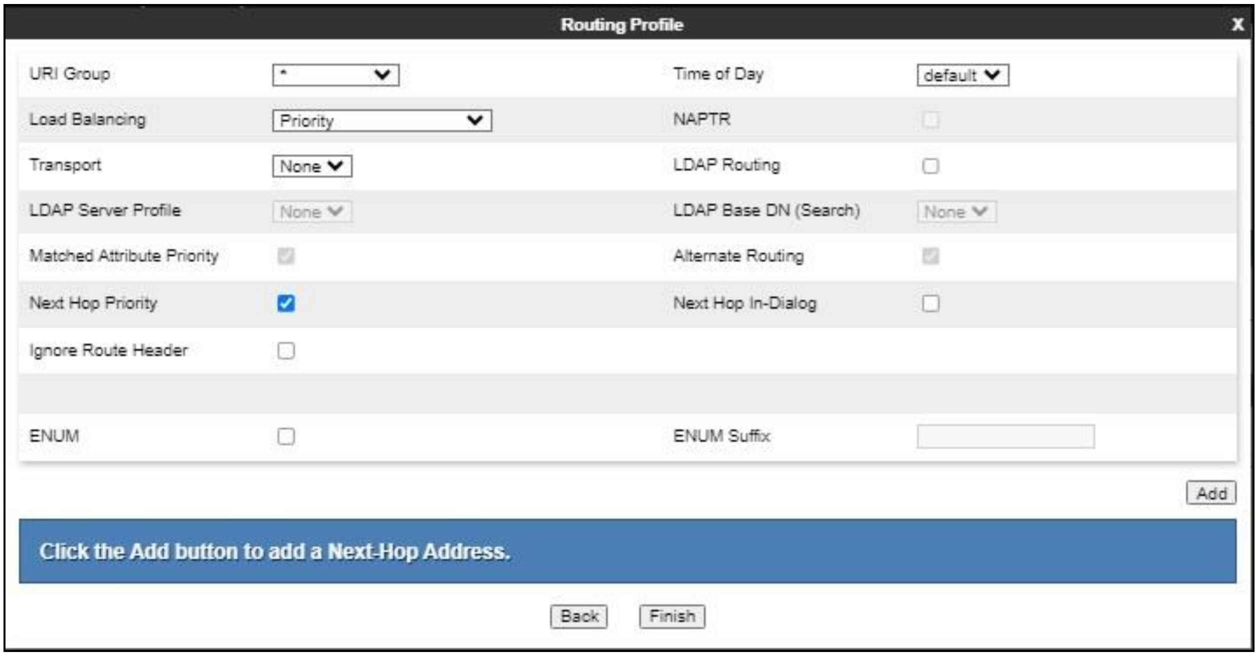
6.7.2. Routing – M-net

Create a Routing Profile for M-net SIP network.

- Navigate to **Configuration Profiles → Routing** and select **Add Profile**.
- Enter a **Profile Name** and click **Next**.

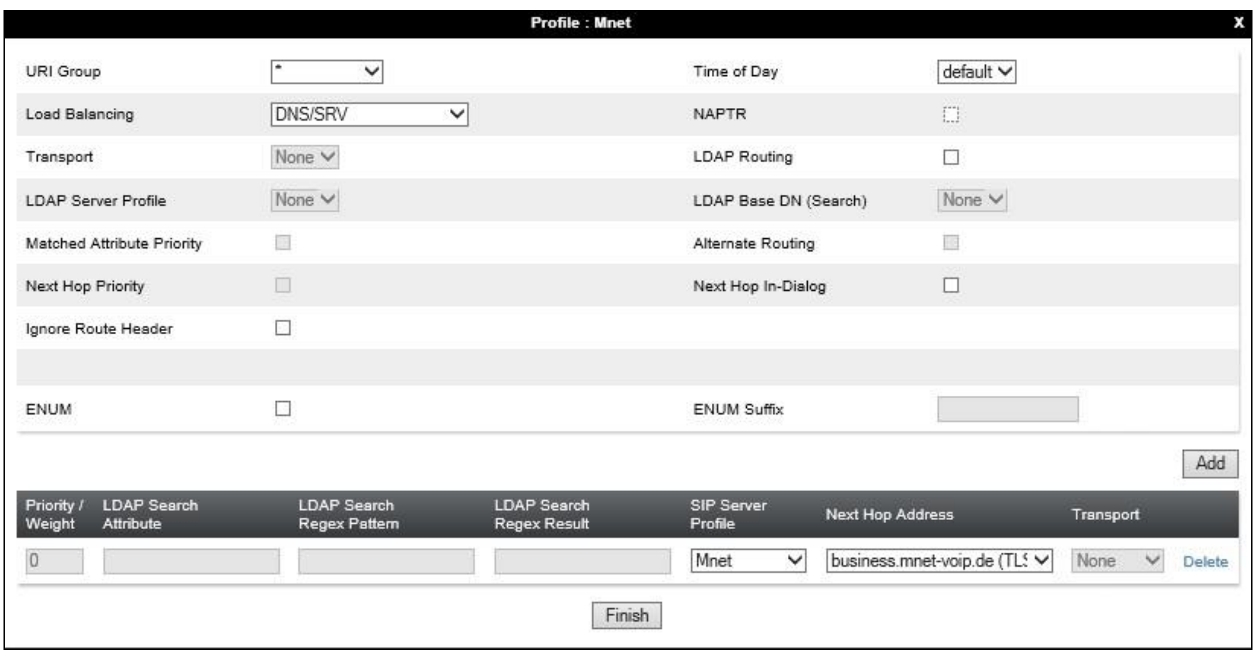


The Routing Profile window will open. Use the default values displayed and click **Add**.



On the **Next Hop Address** window, set the following:

- **Load Balancing** = **DNS/SRV**.
- **SIP Server Profile** = **Mnet** (**Section 6.6.2**) from drop down menu.
- **Next Hop Address** = Select **business.mnet-voip.de (TLS)** from drop down menu.
- Click **Finish**.



6.8. Topology Hiding

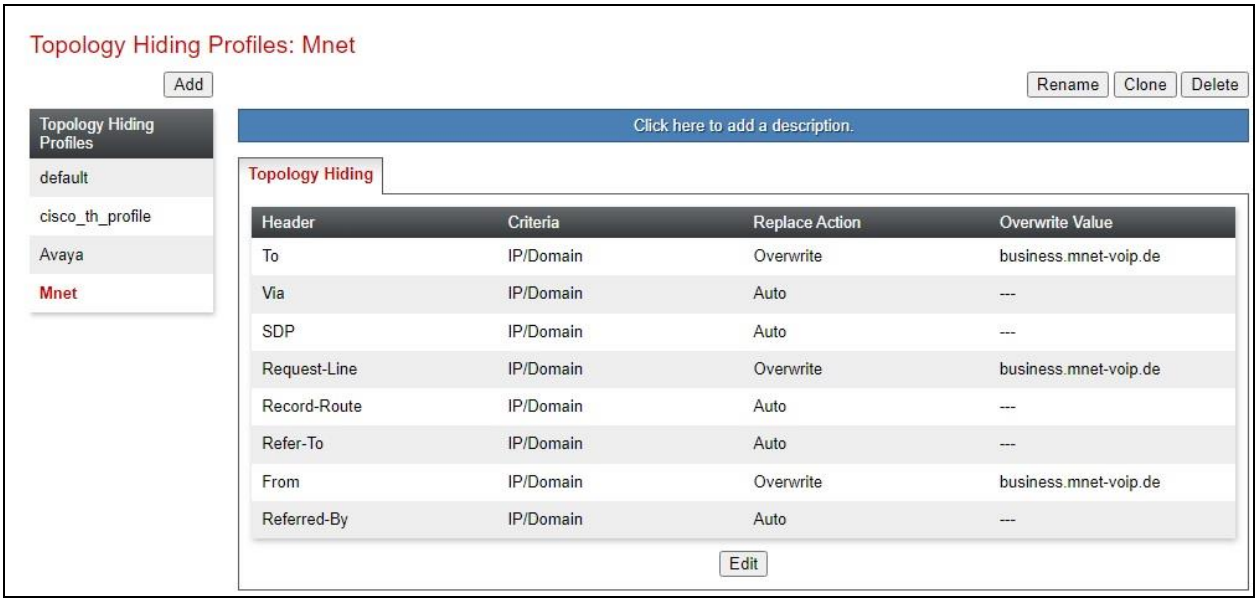
Topology hiding is used to hide local information such as private IP addresses and local domain names. The local information can be overwritten with a domain name or IP addresses. The default **Replace Action** is **Auto**, this replaces local information with IP addresses, generally the next hop. Topology hiding has the advantage of presenting single Via and Record-Route headers externally where multiple headers may be received from the enterprise. In some cases where Topology Hiding can't be applied, in particular the Contact header, IP addresses are translated to the Avaya SBCE external addresses using NAT.

To define Topology Hiding for IP Office, navigate to **Configuration Profiles → Topology Hiding** from menu on the left-hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- Enter a descriptive Profile Name such as **Avaya**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for **To**, **From** and **Request Line**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **avaya.com**.
- Click **Finish** (not shown).

To define Topology Hiding for M-net, navigate to **Configuration Profiles → Topology Hiding** from the menu on the left-hand side. Click on **Add** and enter details in the **Topology Hiding Profile** pop-up menu (not shown).

- In the **Profile Name** field enter a descriptive name for M-net and click **Next**.
- If the required Header is not shown, click on **Add Header**.
- Under the **Header** field for, **From**, select **IP/Domain** under **Criteria** and **Overwrite** under **Replace Action**. For Overwrite value, insert **business.mnet-voip.de**.
- Click **Finish** (not shown).



Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Overwrite	business.mnet-voip.de
Via	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	business.mnet-voip.de
Record-Route	IP/Domain	Auto	---
Refer-To	IP/Domain	Auto	---
From	IP/Domain	Overwrite	business.mnet-voip.de
Referred-By	IP/Domain	Auto	---

6.9. Domain Policies

Domain Policies allow the configuration of sets of rules designed to control and normalize the behavior of call flows, based upon various criteria of communication sessions originating from or terminating in the enterprise. Domain Policies include rules for Application, Media, Signalling, Security, etc.

In the reference configuration, only new Media Rules were defined. All other rules under Domain Policies, linked together on End Point Policy Groups later in this section, used one of the default sets already pre-defined in the configuration. Please note that changes should not be made to any of the defaults. If changes are needed, it is recommended to create a new rule by cloning one the defaults and then make the necessary changes to the new rule.

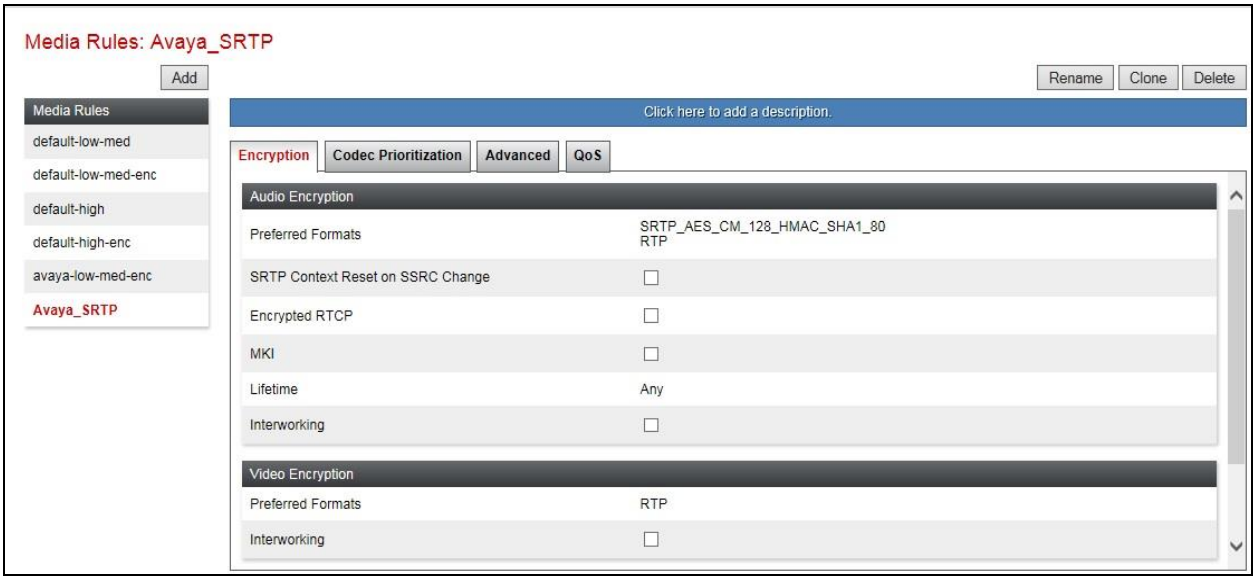
6.9.1. Media Rules

A media rule defines the processing to be applied to the selected media. For the compliance test, media rules were created for both Avaya IP Office and M-net to use SRTP.

To define the Media Rule for IP Office, navigate to **Domain Policies →Media Rules** in the main menu on the left-hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown)

- In the **Rule Name** field enter a descriptive name such as **Avaya_SRTP**.
- Set **Preferred Format #1** to **SRTP_AES_CM_128_HMAC_SHA1_80**.
- Set **Preferred Format #2** to **RTP**.
- Uncheck **Encrypted RTCP**.
- Check **Capability Negotiation** under **Miscellaneous** (not shown).

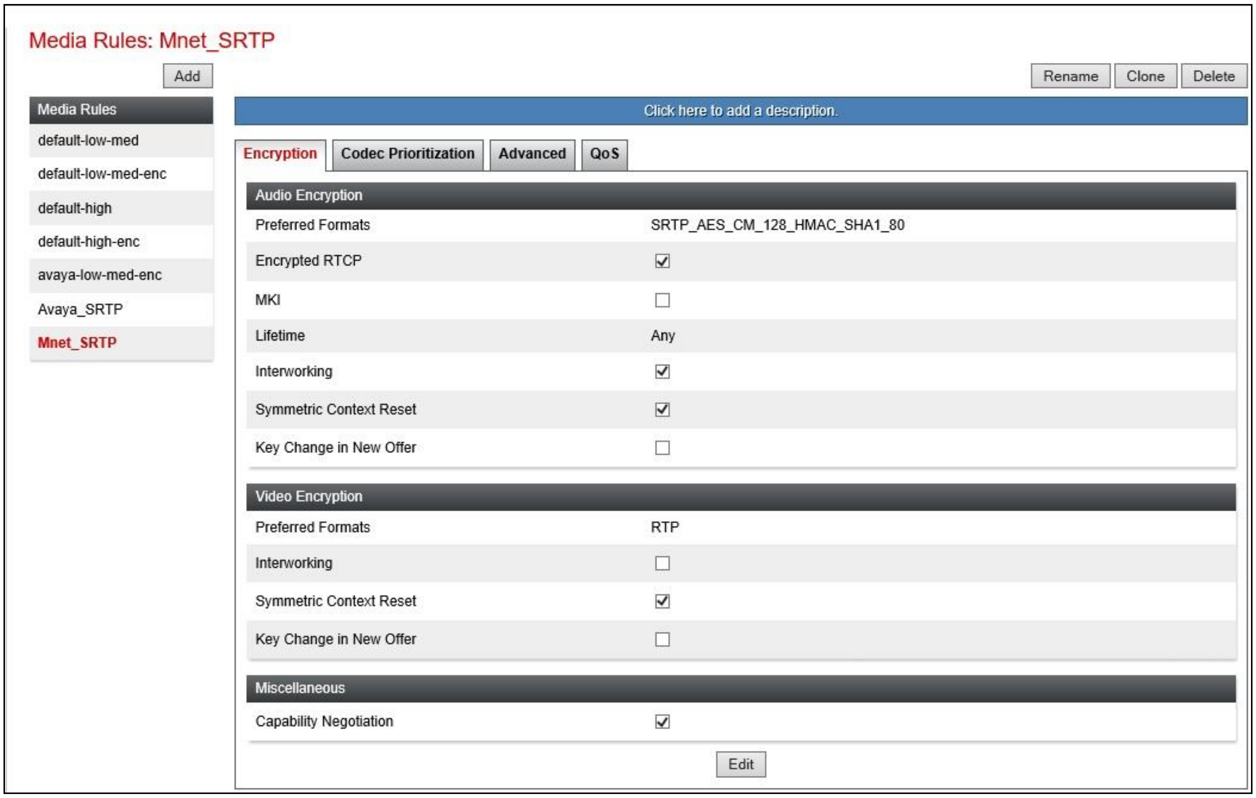
Default values were used for all other fields. Click **Finish** (not shown).



To define the Media Rule for M-net, navigate to **Domain Policies →Media Rules** in the main menu on the left-hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown)

- In the **Rule Name** field enter a descriptive name such as **Mnet_SRTP**.
- Set **Preferred Format #1** to **SRTP_AES_CM_128_HMAC_SHA1_80**.
- Check **Encrypted RTCP**.
- Check **Capability Negotiation** under **Miscellaneous** (not shown).

Default values were used for all other fields. Click **Finish** (not shown).



6.10. End Point Policy Groups

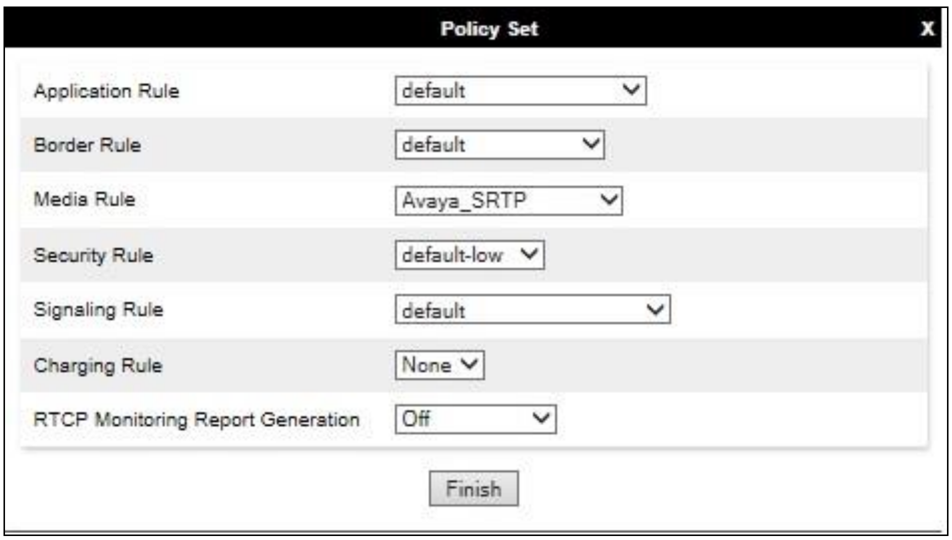
An end point policy group is a set of policies that will be applied to traffic between the Avaya SBCE and a signalling endpoint (connected server). Thus, one end point policy group must be created for Avaya IP Office and another for the M-net Premium SIP trunk. The end point policy group is applied to the traffic as part of the end point flow defined in **Section 6.11**.

6.10.1. End Point Policy Group – Avaya IP Office

To define an End Point policy for IP Office, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left-hand side. Click on **Add** and enter details in the Policy Group pop-up box (not shown).

- In the **Group Name** field enter a descriptive name, in this case **Avaya**, and click **Next** (not shown).
- Leave the **Application Rule**, **Border Rule**, **Security Rule** and **Signalling Rule** fields at their default values.
- In the **Media Rule** drop down menu, select the recently added Media Rule called **Avaya_SRTP**.

Click **Finish**.

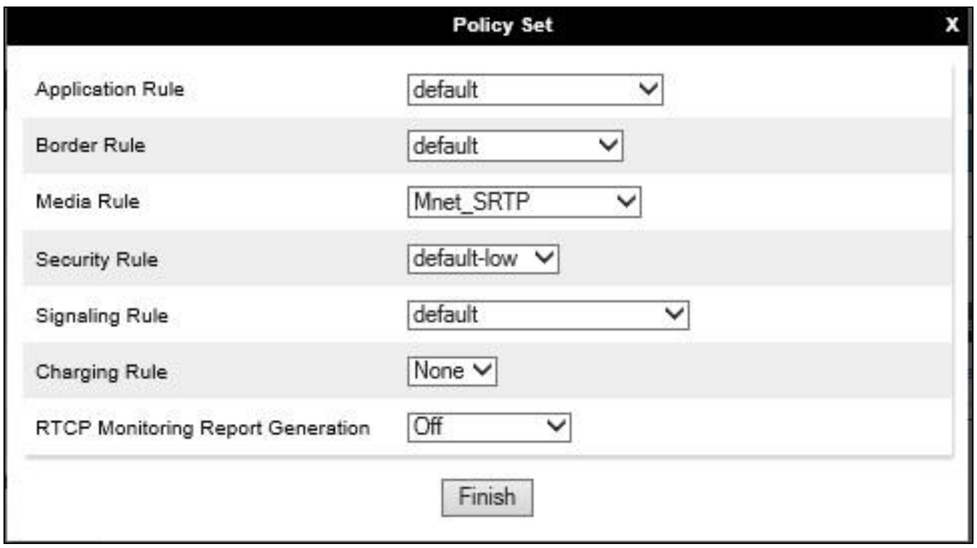


6.10.2. End Point Policy Group – M-net

To define an End Point policy for M-net, navigate to **Domain Policies → End Point Policy Groups** in the main menu on the left-hand side. Click on **Add** and enter details in the Policy Group pop-up box (not shown).

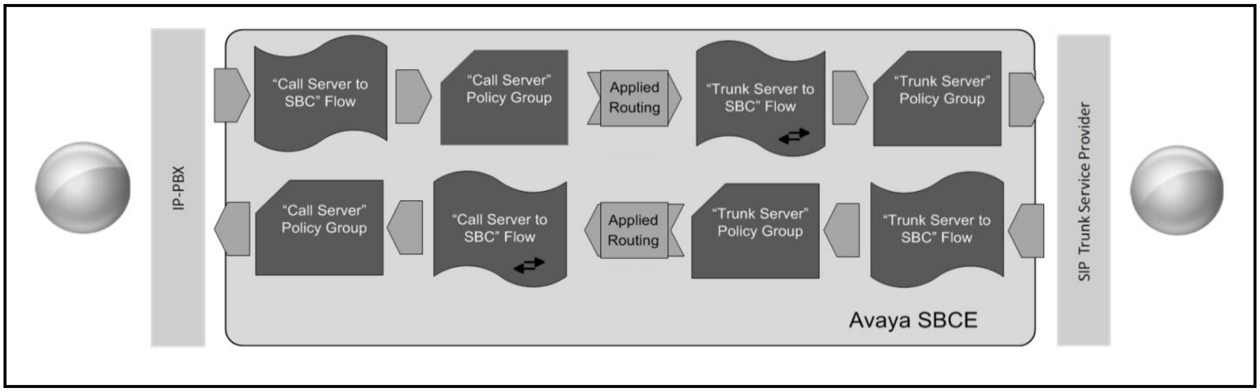
- In the **Group Name** field enter a descriptive name, in this case **Avaya**, and click **Next** (not shown).
- Leave the **Application Rule**, **Border Rule**, **Security Rule** and **Signalling Rule** fields at their default values.
- In the **Media Rule** drop down menu, select the recently added Media Rule called **Mnet_SRTP**.

Click **Finish**.

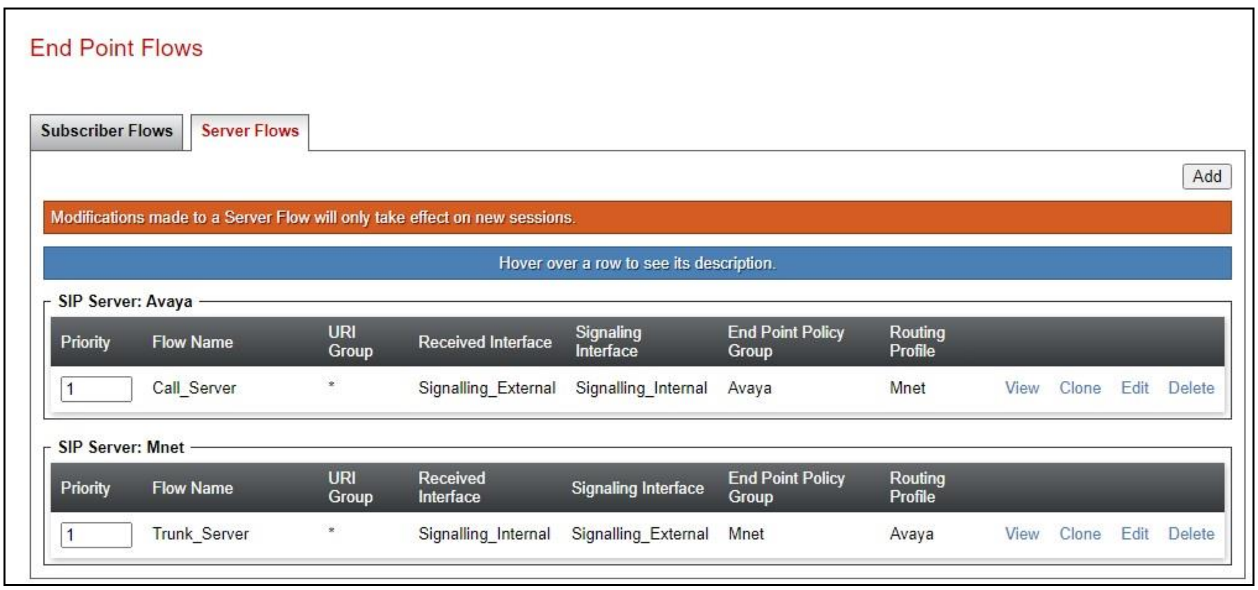


6.11. Server Flows

Server Flows combine the previously defined profiles into outgoing flows from IP Office to M-nets SIP Trunk and incoming flows from M-nets SIP Trunk to IP Office. The following screen illustrates the flow through the Avaya SBCE to secure a SIP Trunk call.

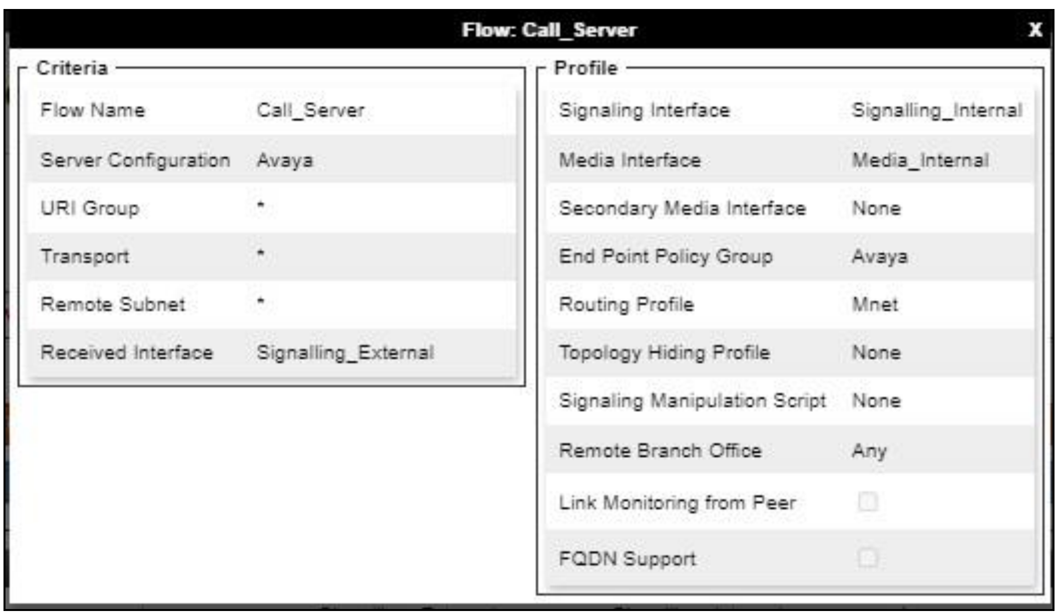


This configuration ties all the previously entered information together so that calls can be routed from IP Office to M-net Premium SIP Trunk and vice versa. The following screenshot shows all configured flows.



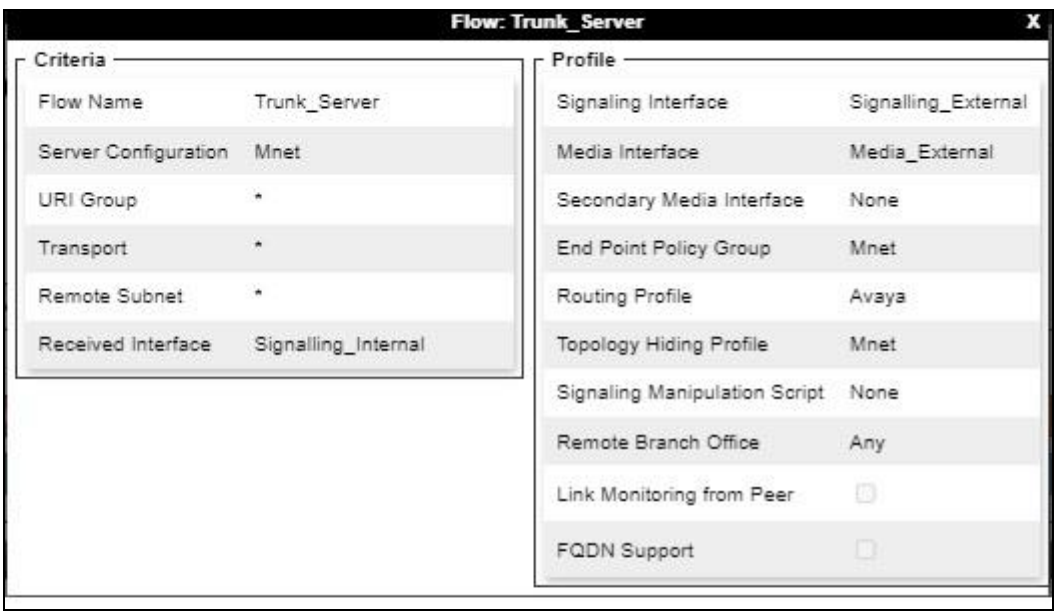
To define a Server Flow for the M-net Premium SIP Trunk, navigate to **Network & Flows → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for M-net Premium SIP Trunk, in the test environment **Trunk_Server** was used.
- In the **Server Configuration** drop-down menu, select the M-net server configuration defined in **Section 6.6.2**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 6.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 6.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **Mnet**.
- In the **Routing Profile** drop-down menu, select the routing profile of the IP Office defined in **Section 6.7.1**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of the M-net Premium SIP Trunk defined in **Section 6.8** and click **Finish** (not shown).



To define an incoming server flow for IP Office from the M-net network, navigate to **Network & Flows → End Point Flows**.

- Click on the **Server Flows** tab.
- Select **Add Flow** and enter details in the pop-up menu.
- In the **Name** field enter a descriptive name for the server flow for IP Office, in the test environment **Call_Server** was used.
- In the **Server Configuration** drop-down menu, select the server configuration for IP Office defined in **Section 6.6.1**.
- In the **Received Interface** drop-down menu, select the internal SIP signalling interface defined in **Section 6.4.1**.
- In the **Signaling Interface** drop-down menu, select the external SIP signalling interface defined in **Section 6.4.1**.
- In the **Media Interface** drop-down menu, select the external media interface defined in **Section 6.4.2**.
- Set the **End Point Policy Group** to the endpoint policy group **Avaya**.
- In the **Routing Profile** drop-down menu, select the routing profile of the M-net SIP Trunk defined in **Section 6.7.2**.
- In the **Topology Hiding Profile** drop-down menu, select the topology hiding profile of IP Office defined in **Section 6.8** and click **Finish** (not shown).



7. 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 as per **Section 2.3**.

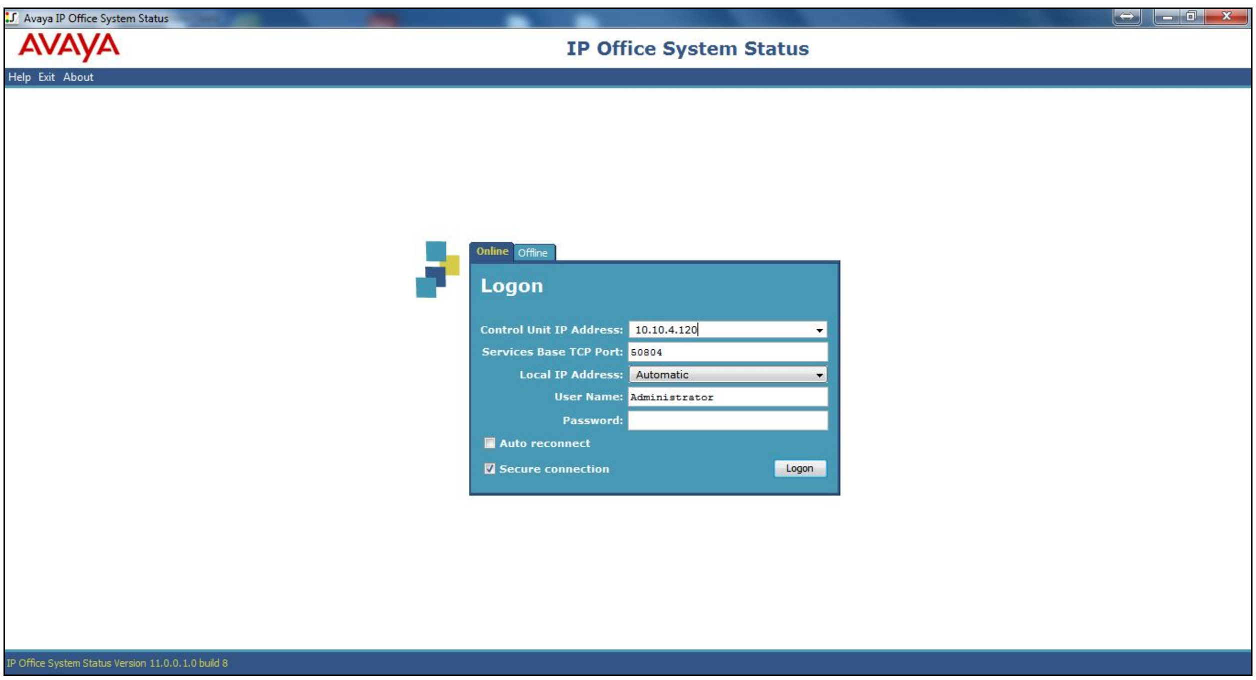
8. Verification Steps

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

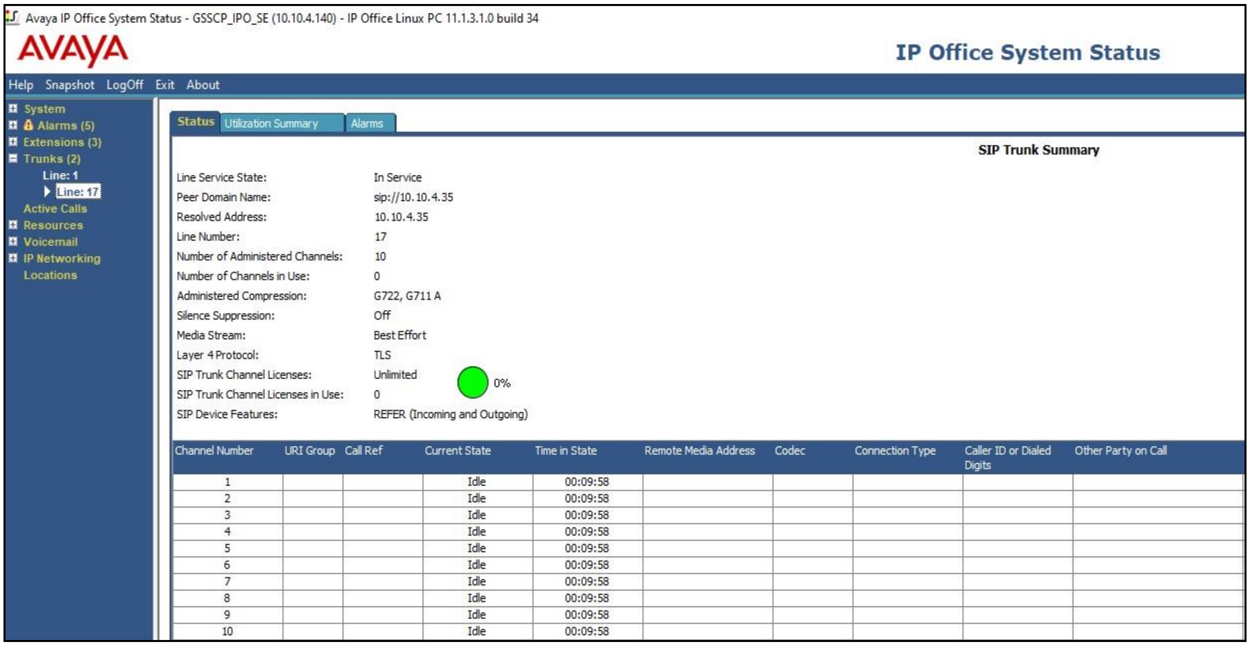
8.1. SIP Trunk status

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.

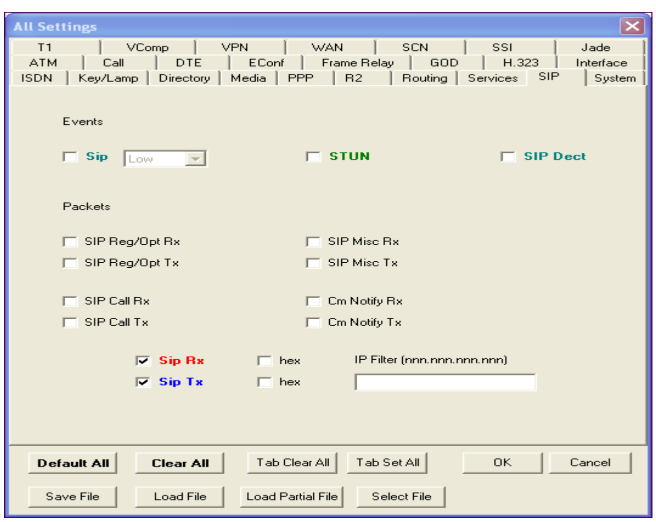


From the left-hand menu expand **Trunks** and choose the SIP trunk (**17** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational.

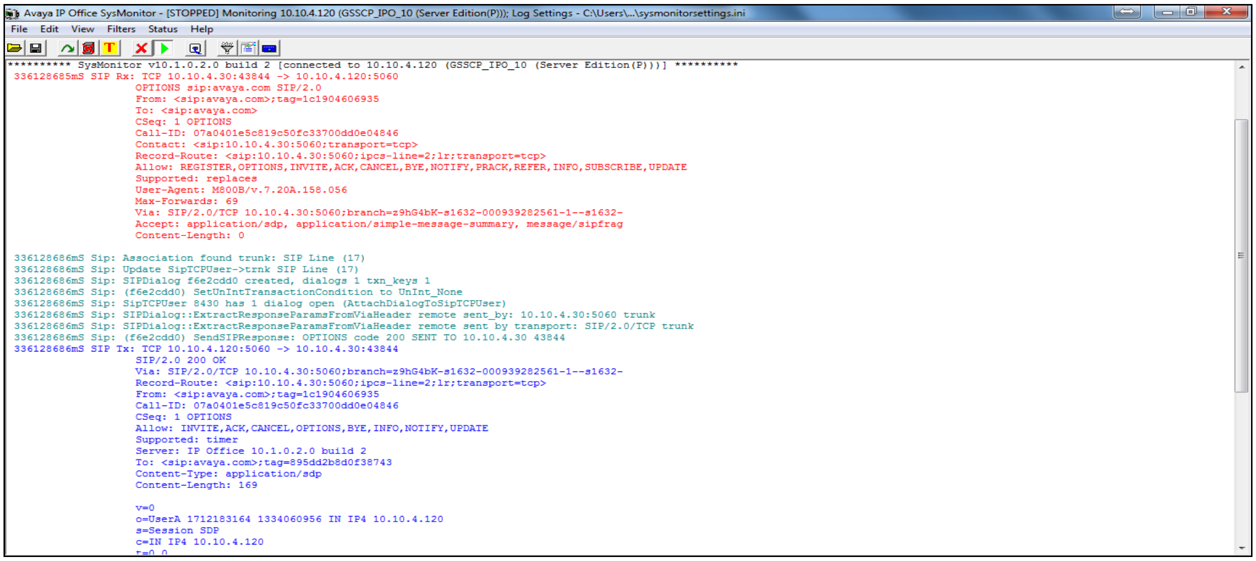


8.1.1. Monitor

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 button that is third from the right in the screen below, 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 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 OPTIONs being sent between IP Office and the Service Provider.

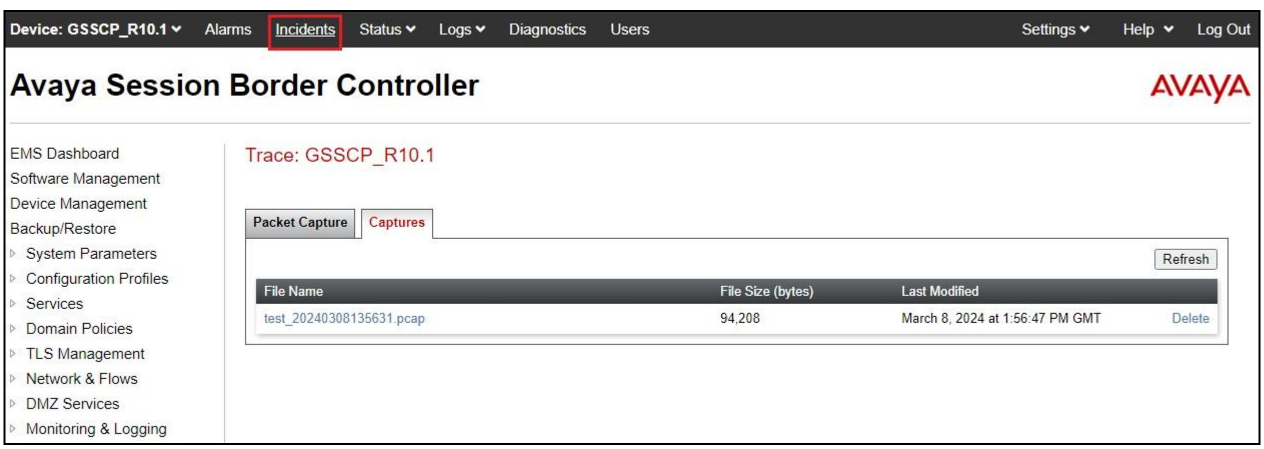


8.2. Avaya SBCE

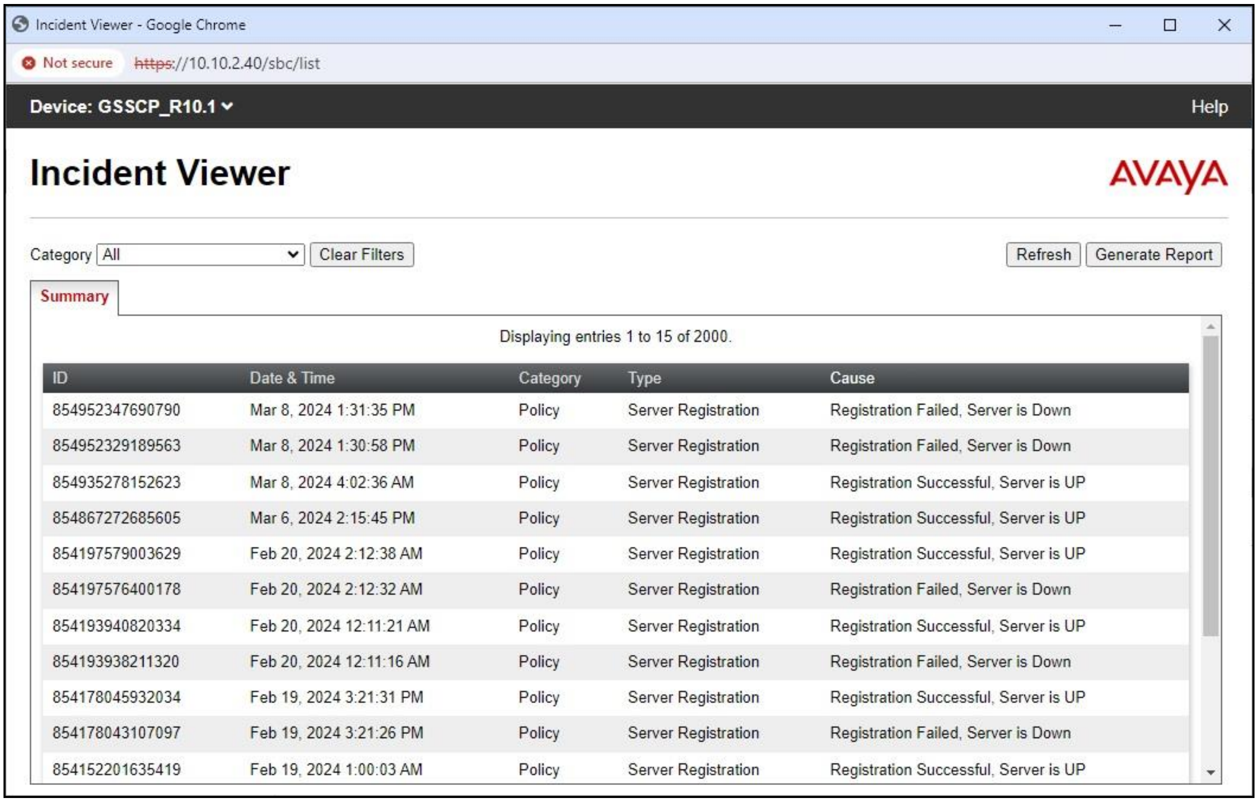
This section provides verification steps that may be performed with the Avaya SBCE.

8.2.1. Incidents

The Incident Viewer can be accessed from the Avaya SBCE dashboard as highlighted in the screen shot below.



Use the Incident Viewer to verify Server Heartbeat and to troubleshoot routing failures.

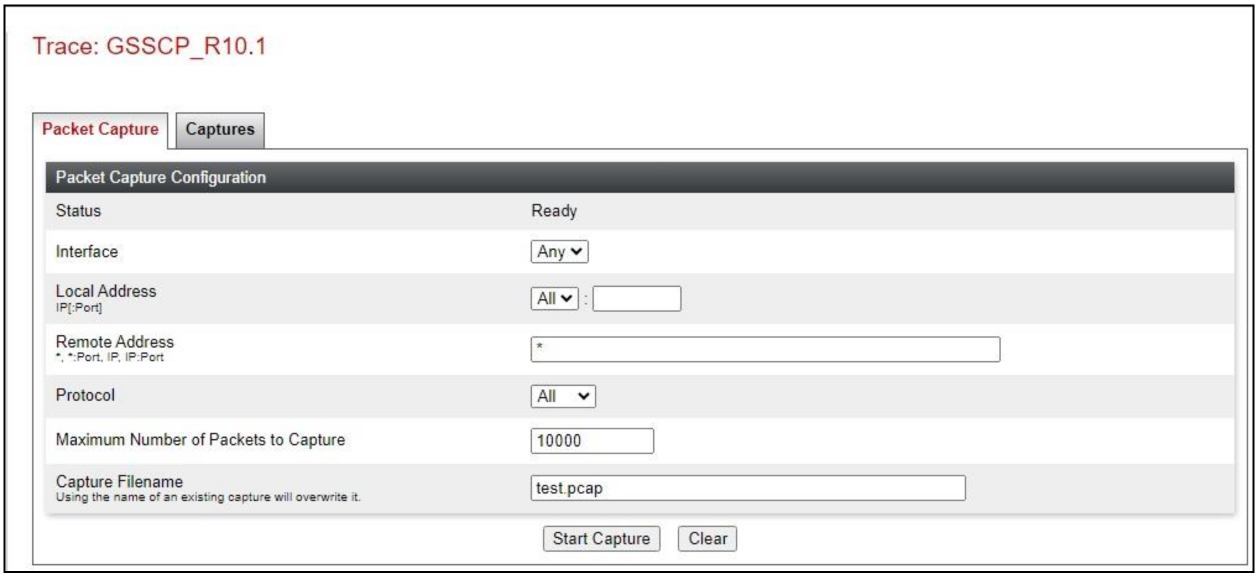


ID	Date & Time	Category	Type	Cause
854952347690790	Mar 8, 2024 1:31:35 PM	Policy	Server Registration	Registration Failed, Server is Down
854952329189563	Mar 8, 2024 1:30:58 PM	Policy	Server Registration	Registration Failed, Server is Down
854935278152623	Mar 8, 2024 4:02:36 AM	Policy	Server Registration	Registration Successful, Server is UP
854867272685605	Mar 6, 2024 2:15:45 PM	Policy	Server Registration	Registration Successful, Server is UP
854197579003629	Feb 20, 2024 2:12:38 AM	Policy	Server Registration	Registration Successful, Server is UP
854197576400178	Feb 20, 2024 2:12:32 AM	Policy	Server Registration	Registration Failed, Server is Down
854193940820334	Feb 20, 2024 12:11:21 AM	Policy	Server Registration	Registration Successful, Server is UP
854193938211320	Feb 20, 2024 12:11:16 AM	Policy	Server Registration	Registration Failed, Server is Down
854178045932034	Feb 19, 2024 3:21:31 PM	Policy	Server Registration	Registration Successful, Server is UP
854178043107097	Feb 19, 2024 3:21:26 PM	Policy	Server Registration	Registration Failed, Server is Down
854152201635419	Feb 19, 2024 1:00:03 AM	Policy	Server Registration	Registration Successful, Server is UP

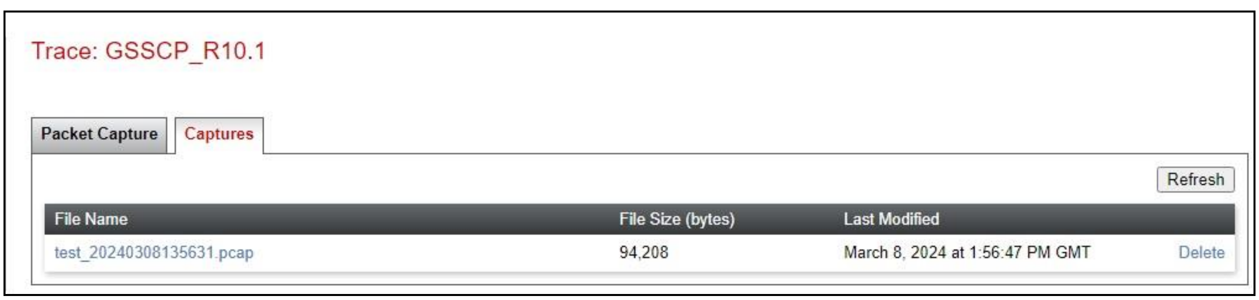
8.2.2. Trace Capture

To define the trace, navigate to **Device Specific Settings →Troubleshooting → Trace** in the menu on the left-hand side and select the **Packet Capture** tab.

- Select the SIP Trunk interface from the **Interface** drop down menu.
- Select **All** from the **Local Address** drop down menu.
- Enter the IP address of the Service Provider's SBC in the **Remote Address** field or enter a ***** to capture all traffic.
- Specify the **Maximum Number of Packets to Capture**, 1000 is shown as an example.
- Specify the filename of the resultant .pcap file in the **Capture Filename** field.
- Click on **Start Capture**.



To view the trace, select the **Captures** tab and click on the relevant filename in the list of traces.



The trace is viewed as a standard .pcap file in Wireshark. If the SIP trunk is working correctly, a SIP response in the form of a 200 OK will be seen from the M-net network.

9. Conclusion

These Application Notes demonstrated how IP Office R11.1 and Avaya Session Border Controller R10.1 can be successfully combined with M-net Premium SIP Trunk Service 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 with Avaya Session Border Controller can be configured to interoperate successfully with M-net Premium SIP Trunk Service using Transport Layer Security (TLS) for signalling and Secured Real-Time Protocol (SRTP) for media encryption. This solution provides IP Office and Avaya Session Border Controller users the ability to access the Public Switched Telephone Network (PSTN) via a SIP trunk with M-net Premium SIP Trunk Service thus eliminating the costs of analog or digital trunk connections previously required to access the PSTN. The service was successfully tested with a number of observations listed in **Section 2.2**.

10. Additional References

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

[1] *Deploying IP Office as Virtual Servers*, Release 11.1, Nov 2021.
[2] *Deploying IP Office Server Edition Servers*, Release 11.1, Nov 2021.
[3] *Deploying an IP500 V2 IP Office System*, Release 11.1, Jul 2022.
[4] *Administering Avaya IP Office with IP Office Web Manager*, Release 11.1, Nov 2021.
[5] *Administering Avaya IP Office with IP Office Manager*, Release 11.1, Nov 2021.
[6] *Using Avaya IP Office System Status*, Nov 2021.
[7] *Using IP Office System Monitor*, Nov 2021.
[8] *Administrating Voicemail Pro*, Release 11.1, Nov 2021.
[9] *Using Avaya Workplace Client for Windows*, Jul 2022.
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[13] *Administering Avaya Session Border Controller Release 10.1*, Apr 2024.
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