



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.

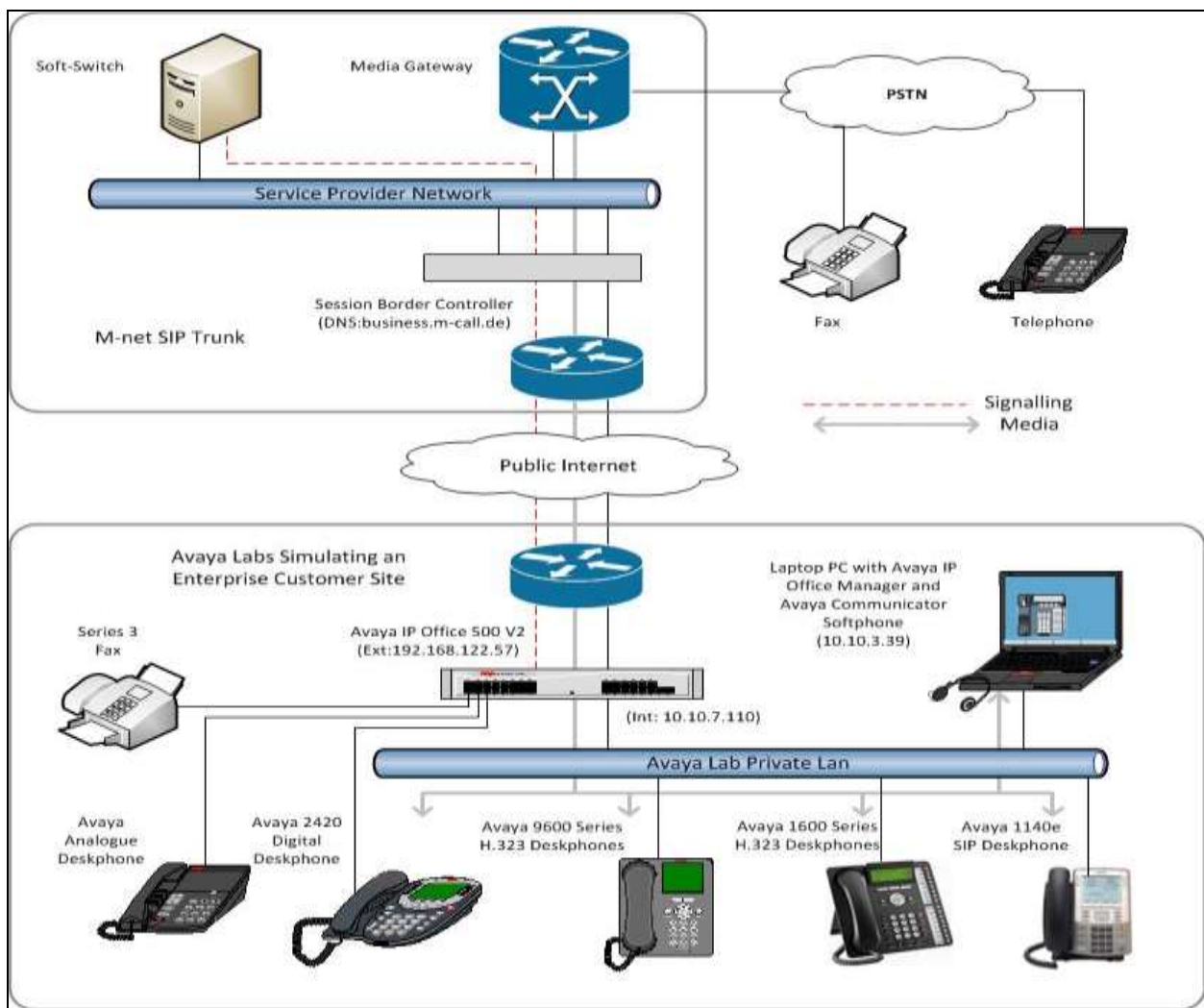


Figure 1: Test setup M-net Premium SIP Trunk to simulated Avaya Enterprise

4. Equipment and Software Validated

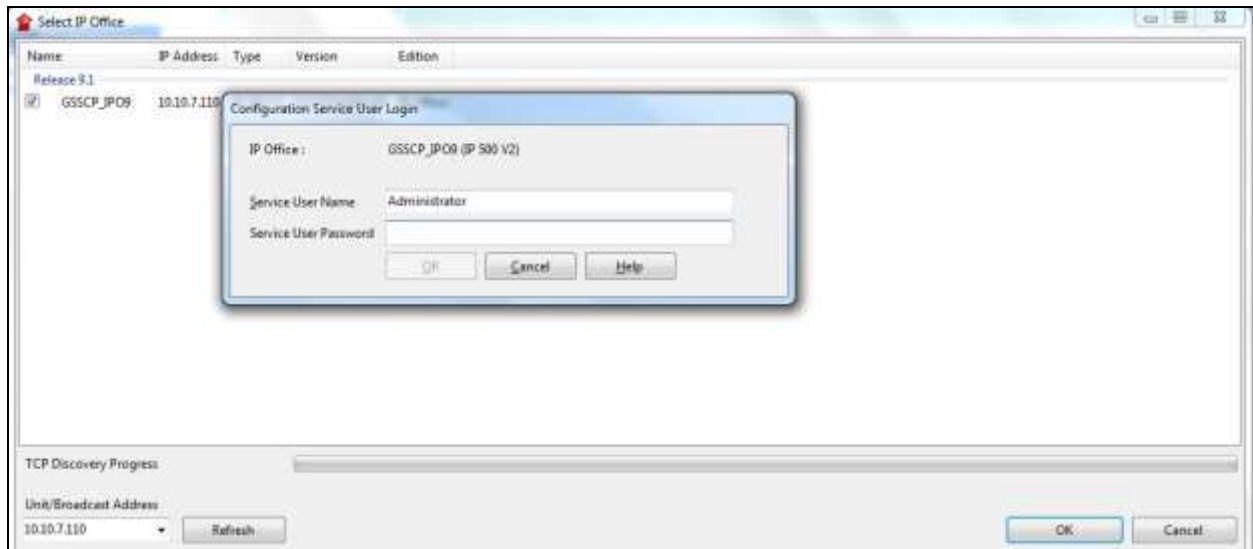
The following equipment and software were used for the sample configuration provided:

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.

The screenshot shows the 'License Remote Server' window. It displays license information for 'License Normal' version 9.1 with serial number 1311049777. Below this is a table of features:

Feature	Key	Instances	Status	Expiry Date
CCR SUP	8U288A6iXXTIFRh32pRLJhhZMWE7x5	255	Obsolete	Never
Advanced Small Community Netw...	eT@t6l5TtO942yxYwI7gBIG8A0olw_8B	255	Obsolete	Never
SIP Trunk Channels	unXMBE6x9dJKGKJ73uEpoF7JrpF4smme	255	Valid	Never
Small Office Edition VCM (channels)	eABRzdgr9vhDAe9YGOuwrpqHEGuLjueM	255	Obsolete	Never

Buttons for 'Add...', 'Remove', 'OK', 'Cancel', and 'Help' are visible on the right and bottom.

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).

The screenshot shows the 'GSSCP_IPO9' configuration window with the 'LAN1' tab selected. Under 'LAN Settings', the 'VoIP' sub-tab is active. The configuration fields are as follows:

- IP Address: 10 . 10 . 7 . 110
- IP Mask: 255 . 255 . 255 . 0
- Primary Trans. IP Address: 0 . 0 . 0 . 0
- RIP Mode: None (dropdown menu)
- ☐ Enable NAT
- Number Of DHCP IP Addresses: 200 (spinner)
- DHCP Mode: ☐ Server ☐ Client ☐ Dialin ☒ Disabled

An 'Advanced' button is located at the bottom right.

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.

The screenshot shows the GSSCP_IPO9 configuration window with the VoIP tab selected. The interface includes a top navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and Codecs. Below this, there are sub-tabs for LAN Settings, VoIP, and Network Topology. The VoIP tab is active, displaying various configuration options:

- H323 Gatekeeper Enable**: Checked.
- Auto-create Extn**: Unchecked.
- Auto-create User**: Unchecked.
- H323 Remote Extn Enable**: Unchecked.
- Remote Call Signalling Port**: 1720.
- SIP Trunks Enable**: Checked.
- SIP Registrar Enable**: Checked.
- Auto-create Extn/User**: Unchecked.
- SIP Remote Extn Enable**: Unchecked.
- Domain Name**: avaya.com.
- Layer 4 Protocol**:
 - UDP**: Checked, **UDP Port**: 5060, **Remote UDP Port**: 5060.
 - TCP**: Checked, **TCP Port**: 5060, **Remote TCP Port**: 5060.
 - TLS**: Unchecked, **TLS Port**: 5061, **Remote TLS Port**: 5061.
- Challenge Expiry Time (secs)**: 10.

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).

The screenshot shows the 'GSSCP_IPO9*' configuration window with the 'LAN Settings' tab selected. The 'LAN2' sub-tab is also active. The configuration fields are as follows:

Field	Value
IP Address	192 . 168 . 122 . 57
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dialin <input checked="" type="radio"/> Disabled

An 'Advanced' button is located at the bottom right of the configuration area.

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
<div> <div>LAN Settings</div> <div>VoIP</div> <div>Network Topology</div> </div>												
<div> <input checked="" type="checkbox"/> H323 Gatekeeper Enable <div> <input type="checkbox"/> Auto-create Extn <input type="checkbox"/> Auto-create User </div> <div> <input type="checkbox"/> H323 Remote Extn Enable Remote Call Signalling Port <input type="text" value="1720"/> </div> </div>												
<div> <input checked="" type="checkbox"/> SIP Trunks Enable <input checked="" type="checkbox"/> SIP Registrar Enable <input type="checkbox"/> Auto-create Extn/User <input type="checkbox"/> SIP Remote Extn Enable </div>												
<div> <div>Domain Name</div> <div>avaya.com</div> </div>												
<div> <div> <input checked="" type="checkbox"/> UDP UDP Port <input type="text" value="5060"/> Remote UDP Port <input type="text" value="5060"/> </div> <div> <input checked="" type="checkbox"/> TCP TCP Port <input type="text" value="5060"/> Remote TCP Port <input type="text" value="5060"/> </div> <div> <input type="checkbox"/> TLS TLS Port <input type="text" value="5061"/> Remote TLS Port <input type="text" value="5061"/> </div> </div>												
<div> Challenge Expiry Time (secs) <input type="text" value="10"/> </div>												
<div> <div>RTP</div> <div> <div>Port Number Range</div> <div> <div>Minimum</div> <div><input type="text" value="49152"/></div> <div>Maximum</div> <div><input type="text" value="53246"/></div> </div> </div> <div> <div>Port Number Range (NAT)</div> <div> <div>Minimum</div> <div><input type="text" value="49152"/></div> <div>Maximum</div> <div><input type="text" value="53246"/></div> </div> </div> <div> <input type="checkbox"/> Enable RTCP Monitoring on Port 5005 RTCP collector IP address for phones: <input type="text" value="0 . 0 . 0 . 0"/> </div> <div> <div>Keepalives</div> <div> <div>Scope</div> <div><input type="text"/></div> <div>Periodic timeout</div> <div><input type="text" value="5"/></div> </div> <div> <div>Initial keepalives</div> <div><input type="text"/></div> </div> </div> </div>												
<div> <div>DiffServ Settings</div> <div> <div><input type="text" value="B8"/></div> <div>DSCP(Hex)</div> <div><input type="text" value="B8"/></div> <div>Video DSCP(Hex)</div> <div><input type="text" value="FC"/></div> <div>DSCP Mask (Hex)</div> <div><input type="text" value="88"/></div> <div>SIG DSCP (Hex)</div> </div> <div> <div><input type="text" value="46"/></div> <div>DSCP</div> <div><input type="text" value="46"/></div> <div>Video DSCP</div> <div><input type="text" value="63"/></div> <div>DSCP Mask</div> <div><input type="text" value="34"/></div> <div>SIG DSCP</div> </div> </div>												

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).

The screenshot shows the 'GSSCP_IPO9' configuration window with the 'Network Topology' tab selected. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** 0.0.0.0
- STUN Port:** 3478
- Firewall/NAT Type:** Open Internet (selected from a dropdown menu)
- Binding Refresh Time (seconds):** 30
- Public IP Address:** 0 . 0 . 0 . 0
- Public Port:**
 - UDP: 0
 - TCP: 0
 - TLS: 0
- ☐ Run STUN on startup

Buttons for 'Run STUN' and 'Cancel' are located at the bottom right of the configuration area.

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).

The screenshot shows the 'GSSCP_IPO9' configuration window with the 'Telephony' tab selected. The 'Telephony' sub-tab is also active. The 'Analogue Extensions' section includes dropdowns for 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type 1), and 'Default Ring Back Sequence' (Ring Type 2), along with a checkbox for 'Restrict Analogue Extension Ringer Voltage'. The 'Dial Delay Time (secs)' is set to 2, 'Dial Delay Count' to 0, 'Default No Answer Time (secs)' to 15, 'Hold Timeout (secs)' to 0, 'Park Timeout (secs)' to 300, 'Ring Delay (secs)' to 5, 'Call Priority Promotion Time (secs)' to Disabled, 'Default Currency' to EUR, and 'Default Name Priority' to Favour Trunk. The 'Companding Law' section has two columns: 'Switch' and 'Line'. In the 'Switch' column, 'A-Law' is selected. In the 'Line' column, 'A-Law Line' is selected. Other settings include 'DSS Status' (unchecked), 'Auto Hold' (checked), 'Dial By Name' (checked), 'Show Account Code' (checked), 'Inhibit Off-Switch Forward/Transfer' (unchecked), 'Restrict Network Interconnect' (unchecked), 'Include location specific information' (unchecked), 'Drop External Only Impromptu Conference' (unchecked), 'Visually Differentiate External Call' (unchecked), and 'Unsupervised Analog Trunk Disconnect Handling' (unchecked).

5.5. System Twinning Settings

To view or change Twinning settings, select the **Twining** 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.

The screenshot shows the 'GSSCP_IPO9' configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked. Below it, the 'Calling party information for Mobile Twinning' field is empty.

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.

The screenshot shows the 'GSSCP_IPO9' configuration window with the 'Codecs' tab selected. At the top, there is a navigation bar with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, and Codecs. Below the navigation bar, the 'RFC2833 Default Payload' is set to '101'. The main area is divided into three sections: 'Available Codecs', 'Default Codec Selection', and 'Selected'. The 'Available Codecs' section contains a list of codecs with checkboxes: G.711 ULAW 64K (checked), G.711 ALAW 64K (checked), G.722 64K (checked), G.729(a) 8K CS-ACELP (checked), and G.723.1 6K3 MP-MLQ (checked). The 'Default Codec Selection' section is further divided into 'Unused' and 'Selected' sub-sections. The 'Unused' sub-section contains a list of codecs: G.711 ULAW 64K, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' sub-section contains a list of codecs: G.711 ALAW 64K and G.729(a) 8K CS-ACELP. Between the 'Unused' and 'Selected' sub-sections are five buttons: '>>>', '<<<', '<<<', '>>>', and '>>>'. The first and last buttons are for moving all codecs between the two sub-sections, while the middle three are for moving individual codecs.

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

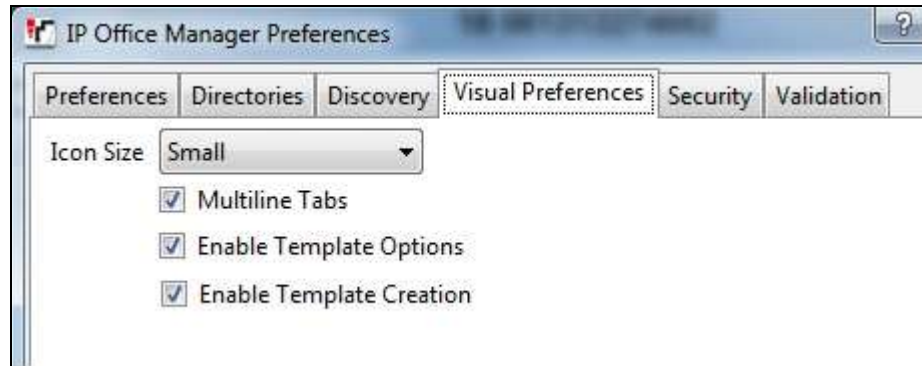
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.7.2**.

5.7.1. SIP Line From Template

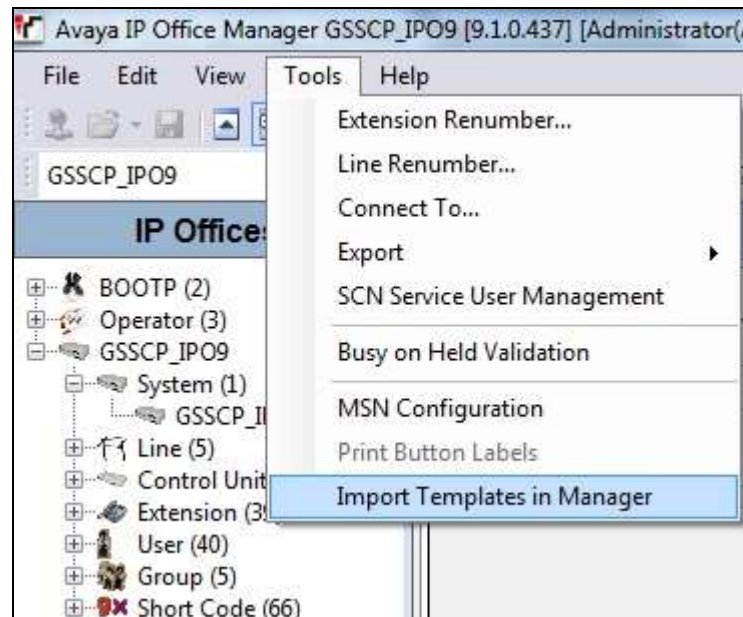
DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (500 V2) and 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**.

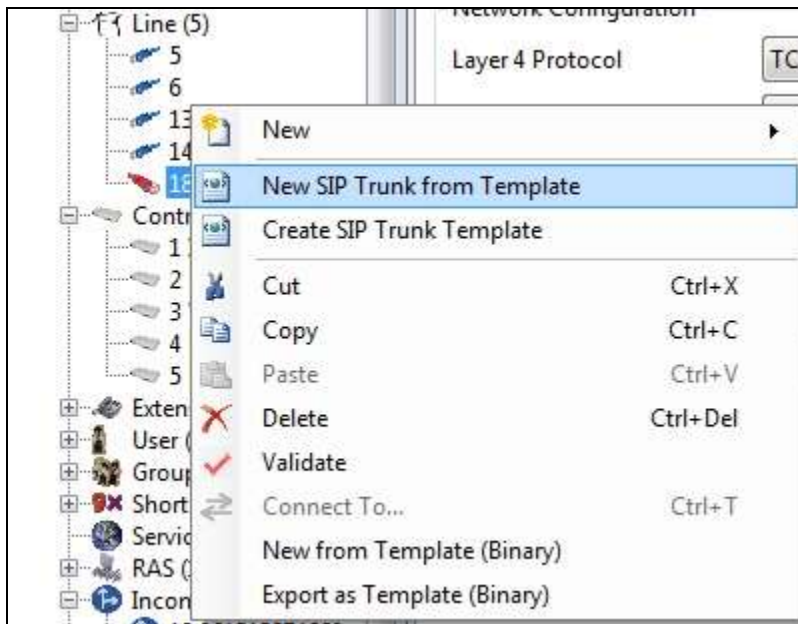


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**.



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).

SIP Line - Line 18

SIP Line | Transport | SIP URI | VoIP | T38 Fax | SIP Credentials | SIP Advanced | Engineering

Line Number	18	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	business.m-call.de	Check OOS	<input checked="" type="checkbox"/>
URI Type	SIP	Session Timers	
Location	Cloud	Refresh Method	Auto
		Timer (seconds)	On Demand
Prefix		Forwarding and Twinning	
National Prefix		Originator number	
International Prefix		Send Caller ID	None
Country Code		Redirect and Transfer	
Name Priority	System Default	Incoming Supervised REFER	Auto
Description		Outgoing Supervised REFER	Auto
		Send 302 Moved Temporarily	<input type="checkbox"/>
		Outgoing Blind REFER	<input type="checkbox"/>

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).

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field is empty. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', 'Use Network Topology Info' is 'LAN 2', and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '8.8.8.8' and '0.0.0.0'. 'Calls Route via Registrar' is checked. 'Separate Registrar' is empty.

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.

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'SIP URI' tab selected. The main area is empty, and on the right, there are three buttons: 'Add...', 'Remove', and 'Edit...'.

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.

SIP Line - Line 18*

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering

Edit...

Edit Channel

Via: <None>

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 10

OK Cancel

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.

The screenshot shows the 'SIP Line - Line 18' configuration window with the 'VoIP' tab selected. The 'Codec Selection' section features a dropdown menu set to 'System Default'. Below it, two lists of codecs are shown: 'Unused' (G.711 ULAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ) and 'Selected' (G.711 ALAW 64K, G.729(a) 8K CS-ACELP). Navigation buttons (right arrow, up arrow, left arrow, down arrow, and double right arrow) are positioned between the lists. To the right, a series of checkboxes are displayed: 'VoIP Silence Suppression' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), 'PRACK/100rel Supported' (checked), and 'G.711 Fax ECAN' (unchecked). At the bottom, three dropdown menus are visible: 'Fax Transport Support' set to 'G.711', 'DTMF Support' set to 'RFC2833', and 'Media Security' set to 'Disabled'.

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.

The screenshot shows a configuration window titled "SIP Line - Line 18". It has several tabs: "SIP Line", "Transport", "SIP URI", "VoIP", "T38 Fax", "SIP Credentials" (which is selected), "SIP Advanced", and "Engineering". The "SIP Credentials" tab contains a large empty rectangular area at the top. Below this area is a section titled "Edit SIP Credentials" with the following fields and controls:

- User name: +4989xxxxx30
- Authentication Name: +4989xxxxx30
- Contact: +4989xxxxx30
- Password: [masked with dots]
- Confirm Password: [masked with dots]
- Expiry (mins): 60 (with up/down arrows)
- Registration required: ☒

On the right side of the "Edit SIP Credentials" section, there are two buttons: "OK" and "Cancel".

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.

The screenshot displays the 'SIP Line - Line 18' configuration window in Avaya IP Office. The 'SIP Advanced' tab is active, showing the following settings:

- Addressing:**
 - Association Method: By Source IP address
 - Call Routing Method: Request URI
 - Suppress DNS SRV Lookups: ☒
- Identity:**
 - Use Phone Context: ☐
 - Add user=phone: ☒
 - Use + for International: ☒
 - Use PAI for Privacy: ☒
 - Use Domain for PAI: ☐
 - Swap From and PAI: ☐
 - Caller ID from From header: ☐
 - Send From in Clear: ☐
 - Cache Auth Credentials: ☒
 - User-Agent and Server Headers: (empty text box)
- Media:**
 - Allow Empty INVITE: ☐
 - Send Empty re-INVITE: ☐
 - Allow To Tag Change: ☐
 - P-Early-Media Support: None
 - Send SilenceSupp=Off: ☐
 - Force Early Direct Media: ☐
 - Media Connection Preservation: Disabled
- Call Control:**
 - Call Initiation Timeout (s): 4
 - Call Queuing Timeout (m): 5
 - Service Busy Response: 486 - Busy Here
 - on No User Responding Send: 408-Request Timeout
 - Action on CAC Location Limit: Allow Voicemail

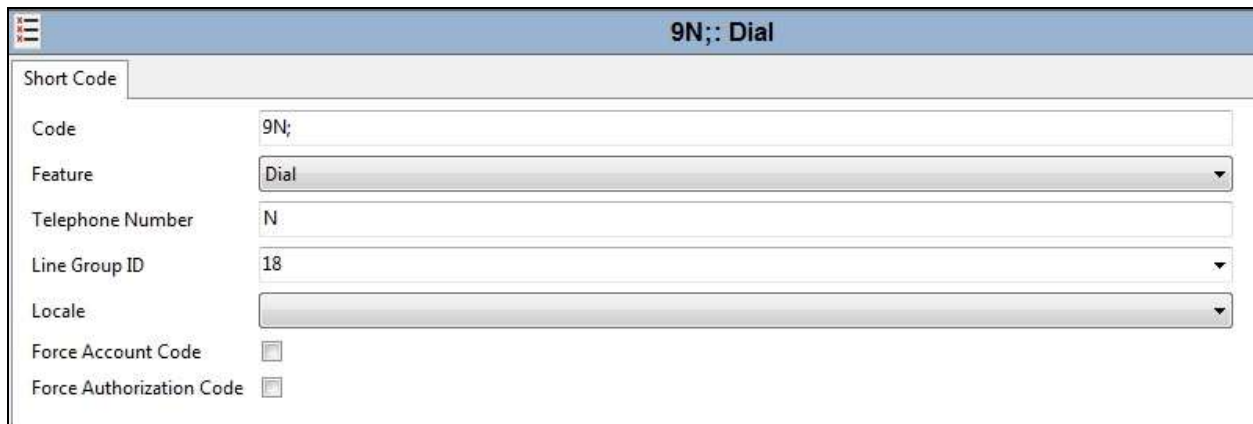
Note: It is advisable at this stage to save the configuration as described in **Section 5.11**.

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).

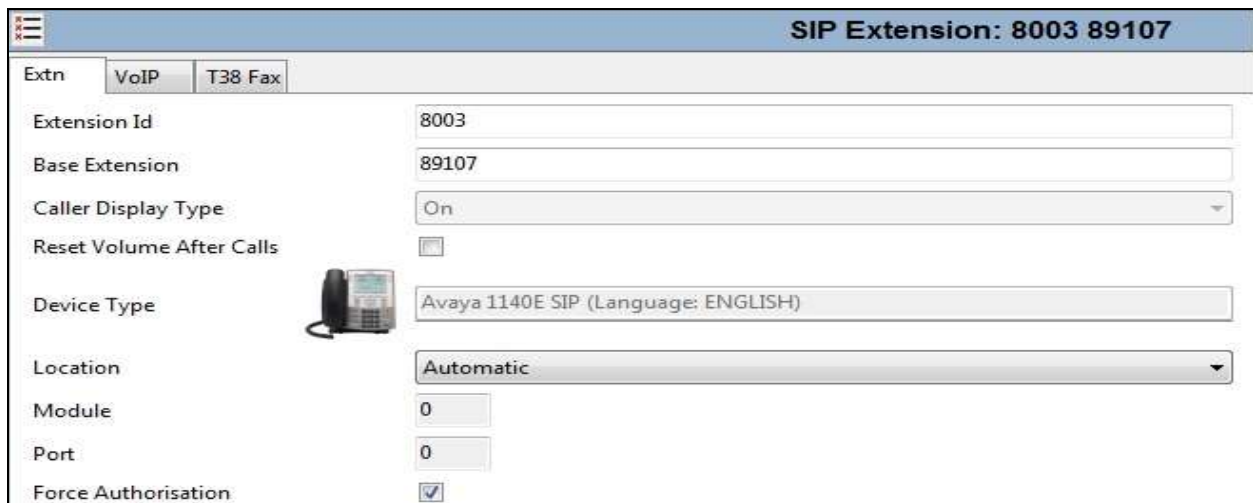


9N;; Dial	
Short Code	
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

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 screenshot displays the 'SIP Extension: 8003 89107' configuration window. It features a tabbed interface with 'Extn', 'VoIP', and 'T38 Fax' tabs. The 'Extn' tab is active, showing the following fields and values:

Field	Value
Extension Id	8003
Base Extension	89107
Caller Display Type	On
Reset Volume After Calls	<input type="checkbox"/>
Device Type	Avaya 1140E SIP (Language: ENGLISH)
Location	Automatic
Module	0
Port	0
Force Authorisation	<input checked="" type="checkbox"/>

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.

The screenshot shows the 'SIP Extension: 8003 89107' configuration page. The 'VoIP' tab is selected. The 'IP Address' field is set to '0 . 0 . 0 . 0'. The 'Codec Selection' dropdown is set to 'System Default'. Below this, there are two lists of codecs: 'Unused' (G.722 64K, G.729(a) 8K CS-ACELP, G.723.1 6K3 MP-MLQ) and 'Selected' (G.711 ALAW 64K, G.711 ULAW 64K). Arrows are used to move codecs between these lists. On the right, there are checkboxes for 'VoIP Silence Suppression', 'Local Hold Music', 'Re-invite Supported' (checked), 'Codec Lockdown', and 'Allow Direct Media Path' (checked). At the bottom, there are dropdown menus for 'Reserve License' (None), 'Fax Transport Support' (None), 'TDM->IP Gain' (Default), 'IP->TDM Gain' (Default), 'DTMF Support' (RFC2833), and '3rd Party Auto Answer' (None).

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.

The screenshot shows the 'Ext89107: 89107' configuration page. The 'SIP' tab is selected. The 'User' sub-tab is active. The 'Name' field is set to 'Ext89107'. The 'Password' and 'Confirm Password' fields are masked with dots. The 'Conference PIN' and 'Confirm Conference PIN' fields are empty. The 'Account Status' dropdown is set to 'Enabled'. The 'Full Name' field is set to 'Ext 89107'. The 'Extension' field is set to '89107'. The 'Email Address' field is empty. The 'Locale' dropdown is empty. The 'Priority' dropdown is set to '5'. The 'System Phone Rights' dropdown is set to 'None'.

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.

The screenshot shows the Avaya web interface for extension 89107. The 'Telephony' tab is selected, and within it, the 'Supervisor Settings' sub-tab is active. The interface includes a header with 'Ext89107: 89107' and a navigation bar with tabs like SIP, Personal Directory, Web Self-Administration, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. The 'Supervisor Settings' sub-tab contains fields for 'Login Code' and 'Confirm Login Code' (both masked with dots), 'Login Idle Period (secs)', 'Monitor Group' (set to '<None>'), 'Coverage Group' (set to '<None>'), and 'Status on No-Answer' (set to 'Logged On (No change)'). There are also checkboxes for 'Force Login', 'Force Account Code', 'Force Authorization Code', 'Incoming Call Bar', 'Outgoing Call Bar', 'Inhibit Off-Switch Forward/Transfer', 'Can Intrude', 'Cannot be Intruded' (checked), 'Can Trace Calls', and 'Deny Auto Intercom Calls'. A 'Reset Longest Idle Time' section has radio buttons for 'All Calls' (selected) and 'External Incoming'.

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.

The screenshot shows the Avaya web interface for extension 89107, with the 'Call Settings' sub-tab selected under the 'Telephony' tab. The interface includes the same header and navigation bar as the previous screenshot. The 'Call Settings' sub-tab contains dropdown menus for 'Outside Call Sequence', 'Inside Call Sequence', and 'Ringback Sequence', all set to 'Default Ring'. It also has input fields for 'No Answer Time (secs)' (set to 'System Default (15)'), 'Wrap-up Time (secs)' (set to '2'), 'Transfer Return Time (secs)' (set to 'Off'), and 'Call Cost Mark-Up' (set to '100'). On the right side, there are checkboxes for 'Call Waiting On' (checked), 'Answer Call Waiting On Hold', 'Busy On Held', and 'Offhook Station'.

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 screenshot shows the configuration page for 'Ext89107: 89107*'. The 'SIP' tab is selected. The 'SIP Name' field contains '+4989xxxxxx10', the 'SIP Display Name (Alias)' field contains 'Ext89107', and the 'Contact' field contains '+4989xxxxxx10'. There is an unchecked checkbox for 'Anonymous'.

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.

The screenshot shows the 'Mobility' tab for 'Ext89107: 89107*'. The 'Maximum Number of Calls' is set to 1. Under 'Twin Bridge Appearances', 'Twin Coverage Appearances', and 'Twin Line Appearances', all checkboxes are unchecked. Under 'Mobility Features', the 'Mobile Twinning' checkbox is checked. The 'Twinned Mobile Number (including dial access code)' field contains '900353894xxxxx1'. The 'Twinning Time Profile' dropdown is set to '<None>'. The 'Mobile Dial Delay (secs)' is set to 2, and the 'Mobile Answer Guard (secs)' is set to 0. Other options like 'Hunt group calls eligible for mobile twinning', 'Forwarded calls eligible for mobile twinning', 'Twin When Logged Out', 'one-X Mobile Client', 'Mobile Call Control', and 'Mobile Callback' are all unchecked.

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.

The screenshot shows a configuration window for an incoming call route. The title bar at the top right displays "18 +4989xxxxxxx10". Below the title bar are three tabs: "Standard", "Voice Recording", and "Destinations". The "Standard" tab is active. It contains several fields with their respective values:

Field	Value
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	+4989xxxxxxx10
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

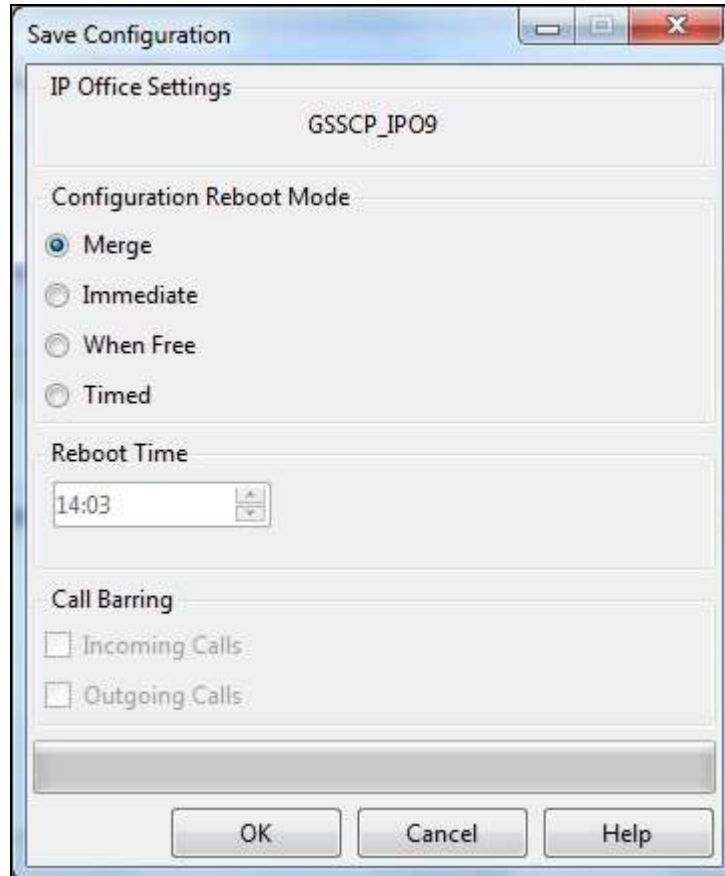
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+4989xxxxxxx10** on line 18 are routed to extension 89107.

The screenshot shows the same configuration window, but with the "Destinations" tab selected. It displays a table with two columns: "TimeProfile" and "Destination".

TimeProfile	Destination
Default Value	89107 Extn89107

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



The image shows a 'Save Configuration' dialog box with a title bar containing minimize, maximize, and close buttons. The dialog is divided into several sections. The first section, 'IP Office Settings', contains the text 'GSSCP_IP09'. The second section, 'Configuration Reboot Mode', contains four radio button options: 'Merge' (which is selected), 'Immediate', 'When Free', and 'Timed'. The third section, 'Reboot Time', contains a time selection field showing '14:03' with up and down arrow buttons. The fourth section, 'Call Barring', contains two unchecked checkboxes: 'Incoming Calls' and 'Outgoing Calls'. At the bottom of the dialog is a horizontal bar and three buttons: 'OK', 'Cancel', and 'Help'.

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.



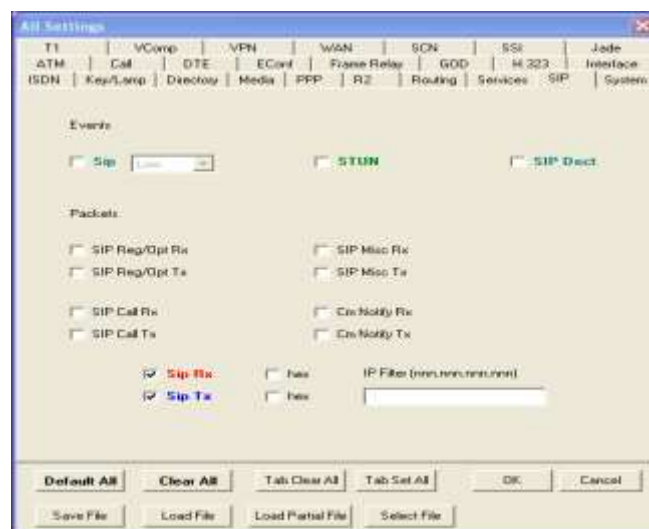
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the **Current State** as being **idle** and **Time in State** if the Trunk is operational. The IP address has been changed for security purposes.



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.



The screenshot shows the Avaya IP Office R8.1 SysMonitor application window. The title bar reads "Avaya IP Office R8.1 SysMonitor - [STOPPED] Monitoring 10.10.9.100 (GSSCP-IPO2): Log Settings: C:\Documents and Settings\... \sysmonitorsetti...". The window contains a log of SIP messages. The first message is a REGISTER request from a handset to the IP Office server. The second message is a 401 Unauthorized response from the server to the handset, indicating that the handset must provide authentication credentials to register.

```
Via: SIP/2.0/TCP 10.10.9.114:1406;alias;branch=z9hG4bKebfe86def3d5f9c48
Max-Forwards: 70
From: <sip:89060@avaya.com>;tag=f5cb455c38
To: <sip:89060@avaya.com>
Call-ID: 449798b416d5be9e
CSeq: 19796 REGISTER
Accept-Encoding: re-ice=2.0
Allow-Events: reg-rtcpt,dialog
Contact: <sip:89060@10.10.9.114:transport=tcp>;reg-id=0;+sip.instance="urn:uuid:00000000-0000-1000-8000-0024b5651ff5"
Expires: 86400
Supported: path, outbound
User-Agent: Avaya IP Phone 1140E (SIP1140e.04.03.09.00)
x-nt-GUID: 0024b5651ff5
Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, REFER, INFO, MESSAGE, NOTIFY, UPDATE
Content-Length: 0

148685969ms Sip: SIPDialog f5116728 created, size 1
148685970ms SIP Tx: TCP 10.10.9.100:5060 -> 10.10.9.114:1406
SIP/2.0 401 Unauthorized
Via: SIP/2.0/TCP 10.10.9.114:1406;alias;branch=z9hG4bKebfe86def3d5f9c48
From: <sip:89060@avaya.com>;tag=f5cb455c38
To: <sip:89060@avaya.com>;tag=5e3dda568a0baa04
Call-ID: 449798b416d5be9e
CSeq: 19796 REGISTER
User-Agent: IP Office 8.1 (697201)
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO, SUBSCRIBE, REGISTER, PUBLISH
WWW-Authenticate: Digest nonce="f4ae9364e39a589ee482",realm="ipoffice",algorithm=MD5
Supported: timer
```


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] *Administering Avaya IP Office with Manager*, December 2015.
- [3] *Administering 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>

©2016 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.