



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya IP Office Release 10.0 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.

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. 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 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:

- 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.
- 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.

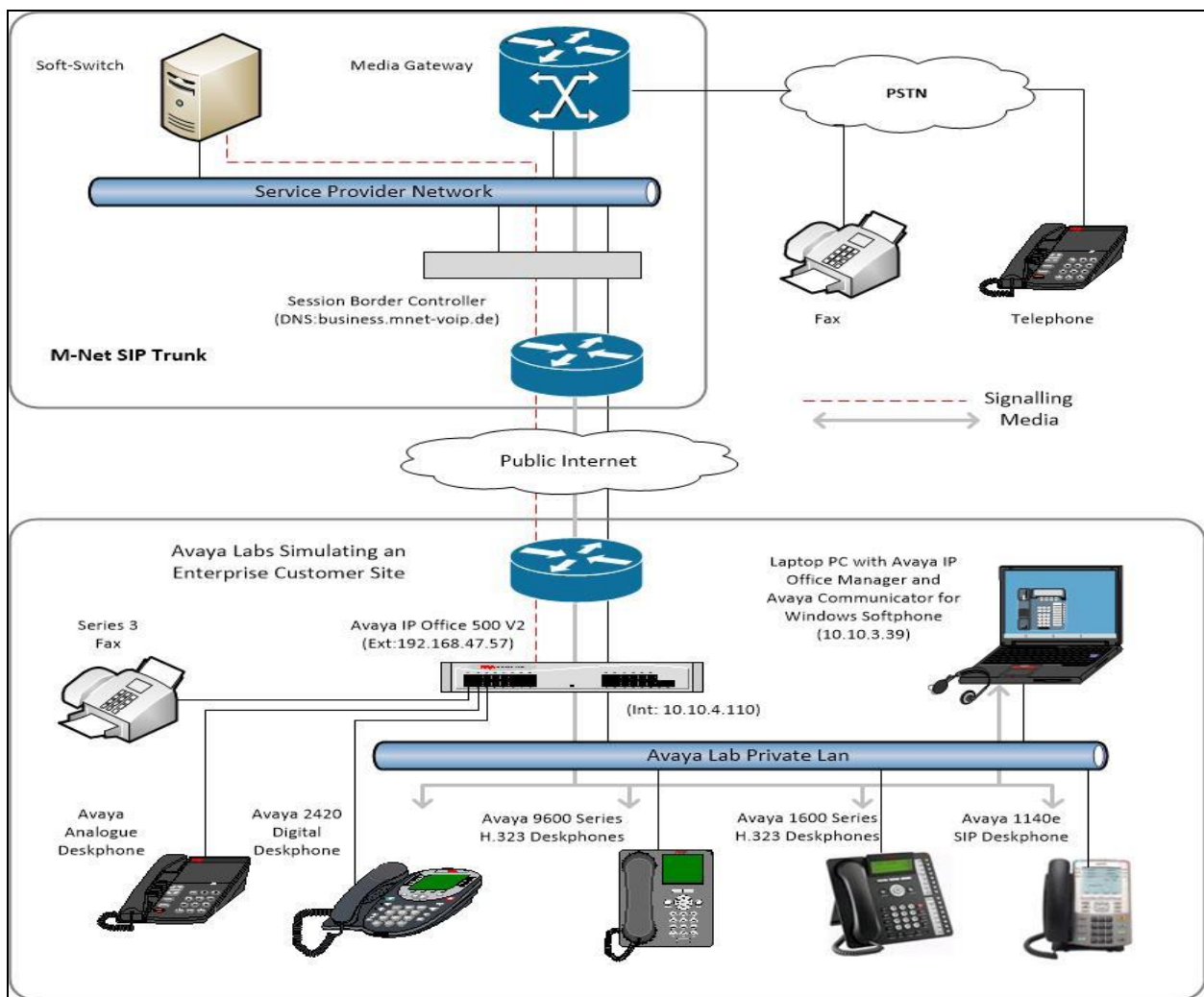


Figure 1: Test setup M-net Premium SIP Trunk to simulated Avaya Enterprise

4. Equipment and Software Validated

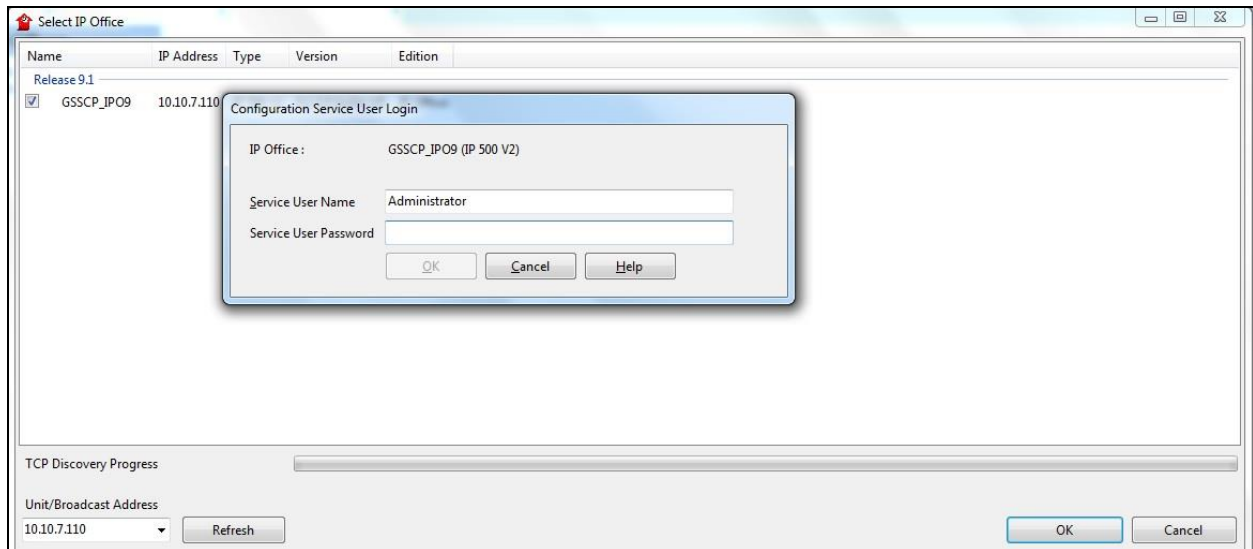
The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Version 10.0.0.2.0 build 10
Avaya Voicemail Pro Client	Version 10.0.0.2.0 build 29
Avaya IP Office Manager	Version 10.0.0.2.0 build 10
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	
Metaswitch Perimeta SBC and IPX (Class 4 Switch/Router and SBC)	4.0.40
Metaswitch CFS (Class 5 Switch)	9.2

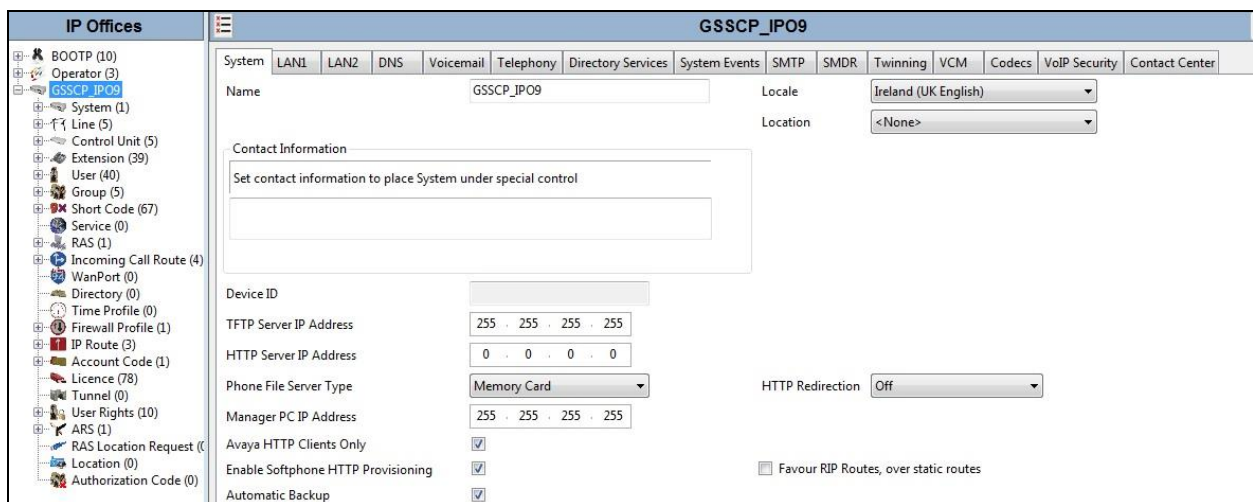
Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

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.

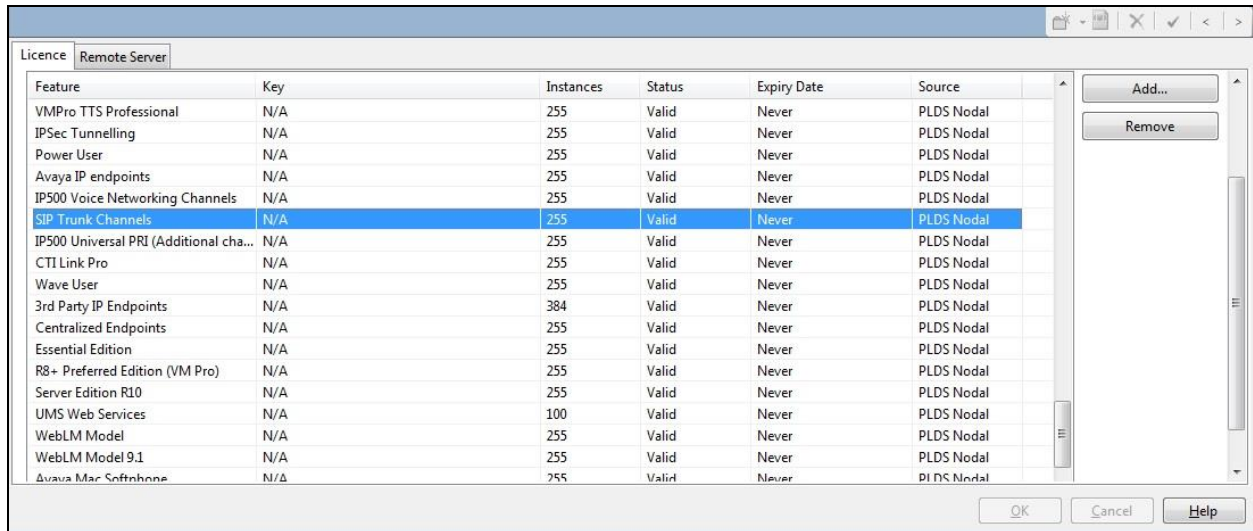


A management window will appear similar to the one in the next section. All the Avaya IP Office configurable components are shown in the left pane known as the Navigation Pane. The pane on the right is the Details Pane. These panes will be referenced throughout the Avaya IP Office configuration. All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



Feature	Key	Instances	Status	Expiry Date	Source
VMPro TTS Professional	N/A	255	Valid	Never	PLDS Nodal
IPSec Tunnelling	N/A	255	Valid	Never	PLDS Nodal
Power User	N/A	255	Valid	Never	PLDS Nodal
Avaya IP endpoints	N/A	255	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	N/A	255	Valid	Never	PLDS Nodal
SIP Trunk Channels	N/A	255	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	N/A	255	Valid	Never	PLDS Nodal
CTI Link Pro	N/A	255	Valid	Never	PLDS Nodal
Wave User	N/A	255	Valid	Never	PLDS Nodal
3rd Party IP Endpoints	N/A	384	Valid	Never	PLDS Nodal
Centralized Endpoints	N/A	255	Valid	Never	PLDS Nodal
Essential Edition	N/A	255	Valid	Never	PLDS Nodal
R8+ Preferred Edition (VM Pro)	N/A	255	Valid	Never	PLDS Nodal
Server Edition R10	N/A	255	Valid	Never	PLDS Nodal
UMS Web Services	N/A	100	Valid	Never	PLDS Nodal
WebLM Model	N/A	255	Valid	Never	PLDS Nodal
WebLM Model 9.1	N/A	255	Valid	Never	PLDS Nodal
Avaya Mac Softphone	N/A	255	Valid	Never	PLDS Nodal

5.2. LAN2 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 **LAN2** interface was used to connect Avaya IP Office to the M-net Premium SIP platform.

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 displays the configuration interface for GSSCP_IPO9*. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, and VoIP. The LAN2 tab is selected, and the LAN Settings sub-tab is active. The configuration fields are as follows:

Field	Value
IP Address	192 . 168 . 47 . 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).

The screenshot displays the GSSCP_IPO9 configuration window with the following settings:

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

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 5.5.2**. 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).

GSSCP_IPO9

System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning VCM Codecs

LAN Settings VoIP Network Topology

Network Topology Discovery

STUN Server Address 0.0.0.0 STUN Port 3478

Firewall/NAT Type Open Internet

Binding Refresh Time (seconds) 30

Public IP Address 0 . 0 . 0 . 0

Public Port

UDP 0

TCP 0

TLS 0

☐ Run STUN on startup

Run STUN Cancel

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).

The screenshot displays the 'GSSCP_IPO9' configuration window, specifically the 'Telephony' tab. The interface is divided into several sections:

- Analogue Extensions:** Includes dropdowns for 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type 1), and 'Default Ring Back Sequence' (Ring Type 2). There is also a checkbox for 'Restrict Analogue Extension Ringer Voltage'.
- Dialing Parameters:** A list of numeric values with up/down arrows: 'Dial Delay Time (secs)' (2), 'Dial Delay Count' (0), 'Default No Answer Time (secs)' (15), 'Hold Timeout (secs)' (0), 'Park Timeout (secs)' (300), 'Ring Delay (secs)' (5), 'Call Priority Promotion Time (secs)' (Disabled), 'Default Currency' (EUR), and 'Default Name Priority' (Favour Trunk).
- Companding Law:** A section with two sub-sections: 'Switch' and 'Line'. Each has radio buttons for 'U-Law' and 'A-Law'. 'A-Law' is selected for both.
- Advanced Settings:** A list of checkboxes: 'DSS Status', '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).

5.4. 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. 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 the tabs, there is a section for 'RFC2833 Default Payload' with a dropdown menu set to '101'. The main area is divided into three panels: 'Available Codecs', 'Default Codec Selection', and 'Selected'. The 'Available Codecs' panel contains a list of codecs with checkboxes: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' panel is further divided into 'Unused' and 'Selected' sub-sections. The 'Unused' section lists G.711 ULAW 64K, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' section lists G.711 ALAW 64K and G.729(a) 8K CS-ACELP. Between the 'Unused' and 'Selected' sections are four buttons: '>>>', '<<<', '<<<', and '>>>', along with up and down arrow buttons.

5.5. 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 5.5.1** 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 5.5.2**.

Also, the following SIP Line settings are not supported on Basic Edition:

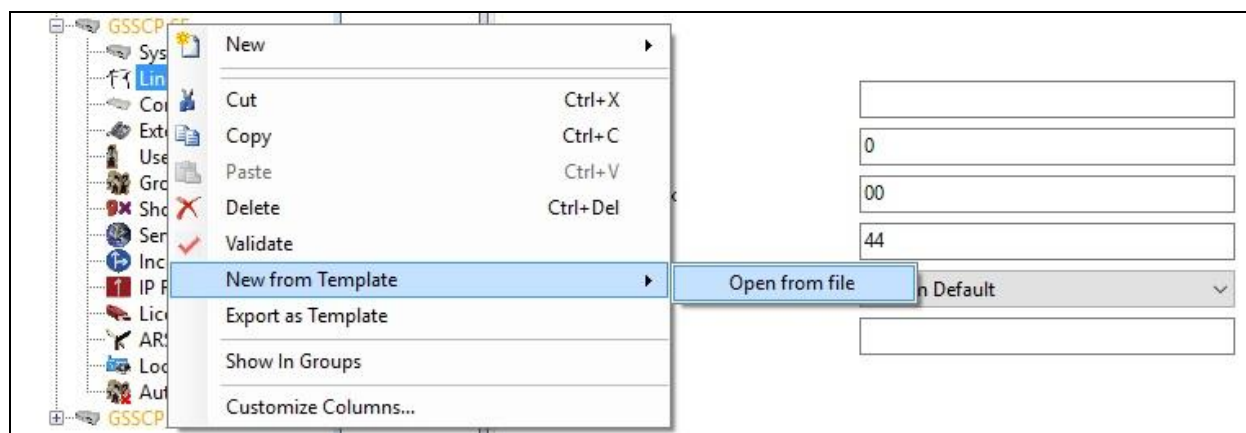
- SIP Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Required

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 5.5.2**.

5.5.1. 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 the template file to the 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**.



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 5.5.2**.

5.5.2. 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.
- Leave **Prefix blank** and set the, **National Prefix** and **International Prefix** to those used in Germany. This ensures that Calling Party Numbers are presented on IP Office extensions in diallable format. It also removes the prefixes on outgoing dialled numbers for conversion to E.164 format.
- Set **Country Code** to **49** for Germany, this prefixes the country code on outgoing dialled numbers for conversion to E.164 format
- Ensure the **In Service** box is checked.
- Ensure the **Check OOS** box is checked.
- Set **Refresh Method** to **Auto**.
- 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 17*							
SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials	SIP Advanced	Engineering
Line Number	17			In Service	<input checked="" type="checkbox"/>		
ITSP Domain Name	business.mnet-voip.de			Check OOS	<input checked="" type="checkbox"/>		
Local Domain Name				Session Timers			
URI Type	SIP			Refresh Method	Auto		
Location	Cloud			Timer (seconds)	On Demand		
Prefix				Redirect and Transfer			
National Prefix	0			Incoming Supervised REFER	Auto		
International Prefix	00			Outgoing Supervised REFER	Auto		
Country Code	49			Send 302 Moved Temporarily	<input type="checkbox"/>		
Name Priority	System Default			Outgoing Blind REFER	<input type="checkbox"/>		
Description							

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 17' 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 '0 . 0 . 0 . 0' and '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. The 'Separate Registrar' field 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 table below has columns: Channel, Groups, Via, Local URI, Contact, Display Name, PAI, Credential, and Max Calls. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
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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** and **Display Name** to **Use Internal Data**. This will use the DDI number applied to the specific extension in the **User** settings described in **Section 5.7**. It is the default setting when no SIP Credentials are specified.
- Set **Identity** to **Use Internal Data** and leave the Header at default **P Asserted ID**.
- Leave the **Originator Number** for **Forwarding and Twinning** blank so that the DDI number for the User is sent as the calling party number. Select **Diversion Header** as the **Send Caller ID**.
- Select **None** in the **Diversion Header** drop down menu.
- For **Registration**, select **1: <+4989xxxxxx30>** from the pull-down menu.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. For the compliance test, a new incoming group **17** was defined that was associated to a single line (line 17).
- Associate this line with an outgoing line group by entering a line group number in the **Outgoing Group** field. For the compliance test, a new outgoing group **17** was defined that was also associated to line 17.
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

SIP Line - Line 17							
SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials	SIP Advanced	Engineering
Local URI	Use Internal Data						
Contact	Use Internal Data						
Display Name	Use Internal Data						
Identity	Use Internal Data						
Header	P Asserted ID						
Forwarding And Twinning							
Originator Number							
Send Caller Id	Diversion Header						
Diversion Header	None						
Registration	1: +4989xxxxxx30						
Incoming Group	17						
Outgoing Group	17						
Max Sessions	10						

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 interface includes several tabs: SIP Line, Transport, SIP URI, VoIP, T38 Fax, SIP Credentials, SIP Advanced, and Engineering. The 'Codec Selection' section features a dropdown menu set to 'System Default'. Below this are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.711 ALAW 64K' and 'G.729(a) 8K CS-ACELP'. Navigation buttons (right arrow, up arrow, left arrow, down arrow, and right arrow) are positioned between the lists. On the right side, there are checkboxes for '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, there are dropdown menus for '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.

SIP Line - Line 17*

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

Edit...

Edit SIP Credentials

User name: +49xxxxxxxx30

Authentication Name: +49xxxxxxxx30

Contact: +49xxxxxxxx30

Password: ••••••••

Confirm Password: ••••••••

Expiry (mins): 60

Registration required: ☒

OK

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 shows the 'SIP Line - Line 17*' configuration window with the 'SIP Advanced' tab selected. The configuration is as follows:

Section	Field	Value
Addressing	Association Method	By Source IP address
	Call Routing Method	Request URI
	Suppress DNS SRV Lookups	<input type="checkbox"/>
Identity	Use "phone-context"	<input type="checkbox"/>
	Add user=phone	<input checked="" type="checkbox"/>
	Use + for International	<input checked="" type="checkbox"/>
	Use PAI for Privacy	<input checked="" type="checkbox"/>
	Use Domain for PAI	<input type="checkbox"/>
	Swap From and PAI/Diversion	<input type="checkbox"/>
	Caller ID from From header	<input type="checkbox"/>
	Send From In Clear	<input type="checkbox"/>
	Cache Auth Credentials	<input checked="" type="checkbox"/>
	User-Agent and Server Headers	
Send Location Info	Never	
Media	Allow Empty INVITE	<input checked="" type="checkbox"/>
	Send Empty re-INVITE	<input type="checkbox"/>
	Allow To Tag Change	<input type="checkbox"/>
	P-Early-Media Support	None
	Send SilenceSupp=Off	<input type="checkbox"/>
	Force Early Direct Media	<input type="checkbox"/>
Media Connection Preservation	Media Connection Preservation	Disabled
	Indicate HOLD	<input checked="" type="checkbox"/>
Call Control	Call Initiation Timeout (s)	4
	Call Queuing Timeout (m)	3
	Service Busy Response	486 - Busy Here
	on No User Responding Send	408-Request Timeout
	Action on CAC Location Limit	Allow Voicemail
	Suppress Q.850 Reason Header	<input type="checkbox"/>
	Emulate NOTIFY for REFER	<input type="checkbox"/>
	No REFER if using Diversion	<input type="checkbox"/>

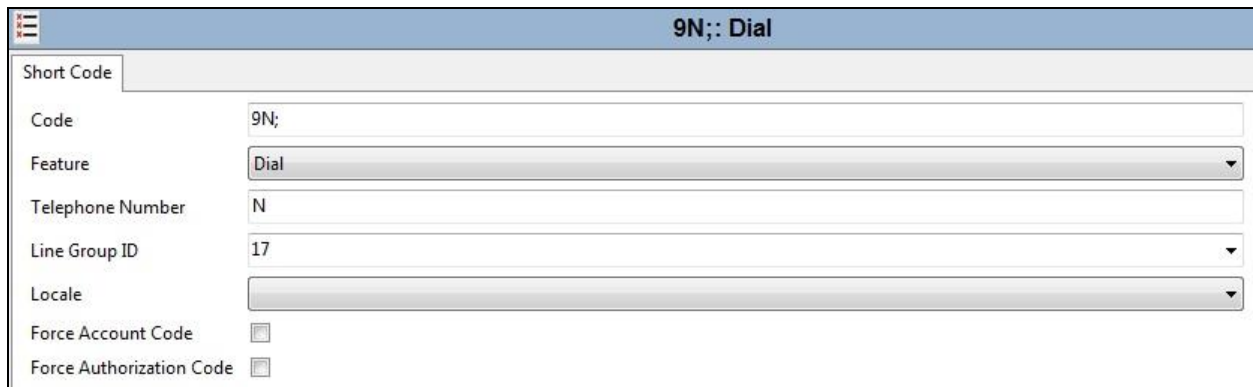
Note: It is advisable at this stage to save the configuration as described in **Section 5.9**.

5.6. 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 5.5.2**.

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



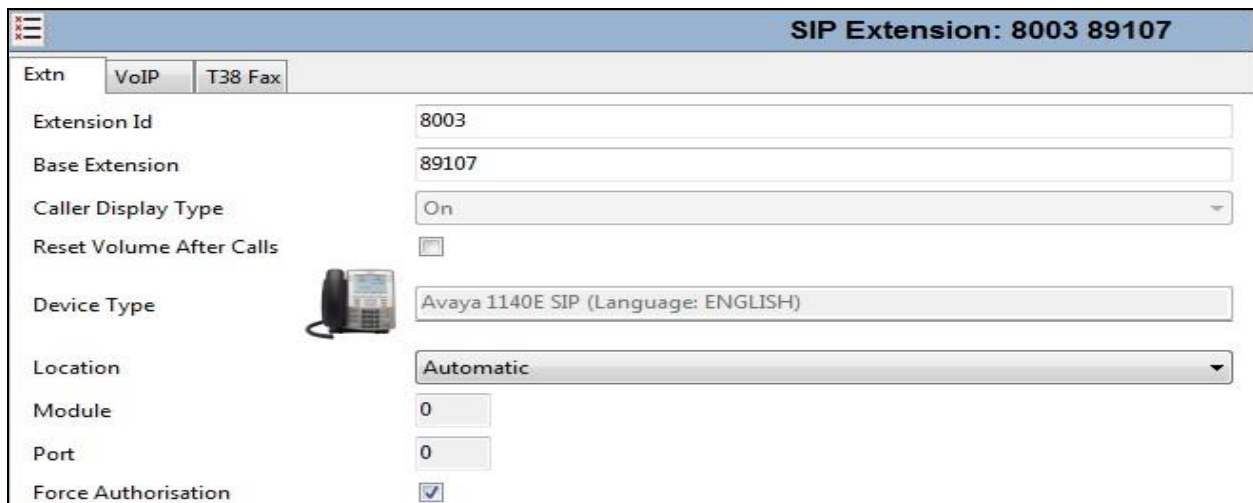
The screenshot shows a configuration window titled "9N;; Dial". The "Short Code" tab is selected. The fields are configured as follows:

Field	Value
Code	9N;
Feature	Dial
Telephone Number	N
Line Group ID	17
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.7. 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 navigation pane on the left with tabs for 'Extn', 'VoIP', and 'T38 Fax'. The 'Extn' tab is active, showing a list of configuration fields on the left and their corresponding values on the right. A small icon of a telephone is positioned between the 'Device Type' and 'Location' fields. The 'Force Authorisation' field at the bottom has a checked checkbox.

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 5.4**. 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 displays the configuration page for SIP Extension 8003 89107. The interface includes tabs for 'Extn', 'VoIP', and 'T38 Fax', with 'VoIP' currently 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: 'Unused' containing 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'; and 'Selected' containing 'G.711 ALAW 64K' and 'G.729(a) 8K CS-ACELP'. Between these lists are navigation buttons: '>>>', an up arrow, '<<<', a down arrow, and '>>>'. To the right of these lists 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 Licence' (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 5.5.2**. 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.

SIP		Personal Directory		Web Self-Administration					
User	Voicemail	DND	ShortCodes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming
Name	Extn89107								
Password	••••••								
Confirm Password	••••••								
Conference PIN									
Confirm Conference PIN									
Account Status	Enabled ▼								
Full Name	Ext 89107								
Extension	89107								
Email Address									
Locale	▼								
Priority	5 ▼								
System Phone Rights	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 the extension number 'Extn89107: 89107' and a navigation bar with tabs for SIP, Personal Directory, Web Self-Administration, User, Voicemail, DND, ShortCodes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, and Button Programming. Below the navigation bar, the 'Supervisor Settings' sub-tab is selected, showing fields for Login Code, Confirm Login Code, Login Idle Period (secs), Monitor Group, Coverage Group, Status on No-Answer, and a 'Reset Longest Idle Time' section with radio buttons for 'All Calls' and 'External Incoming'. On the right side, there are several checkboxes: 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.

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 header and navigation bar are the same as in the previous screenshot. The 'Call Settings' sub-tab displays fields for Outside Call Sequence, Inside Call Sequence, Ringback Sequence, No Answer Time (secs), Wrap-up Time (secs), Transfer Return Time (secs), and Call Cost Mark-Up. 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 5.5.2**). 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 'Anonymous' checkbox which is unchecked.

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 '900353894xxxx1'. 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.

5.8. Incoming Call Routing

An incoming call route maps an inbound DDI number on a specific line to an internal extension. To create an incoming call route, right-click **Incoming Call Routes** in the Navigation Pane and select **New**. On the **Standard** tab of the Details Pane, enter the parameters as shown below:

- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.5.2**.
- 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 titled "17 +4989xxxxxx10". It has three tabs: "Standard", "Voice Recording", and "Destinations". The "Standard" tab is active. The fields and their values are:

Field	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	+4989xxxxxx10
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

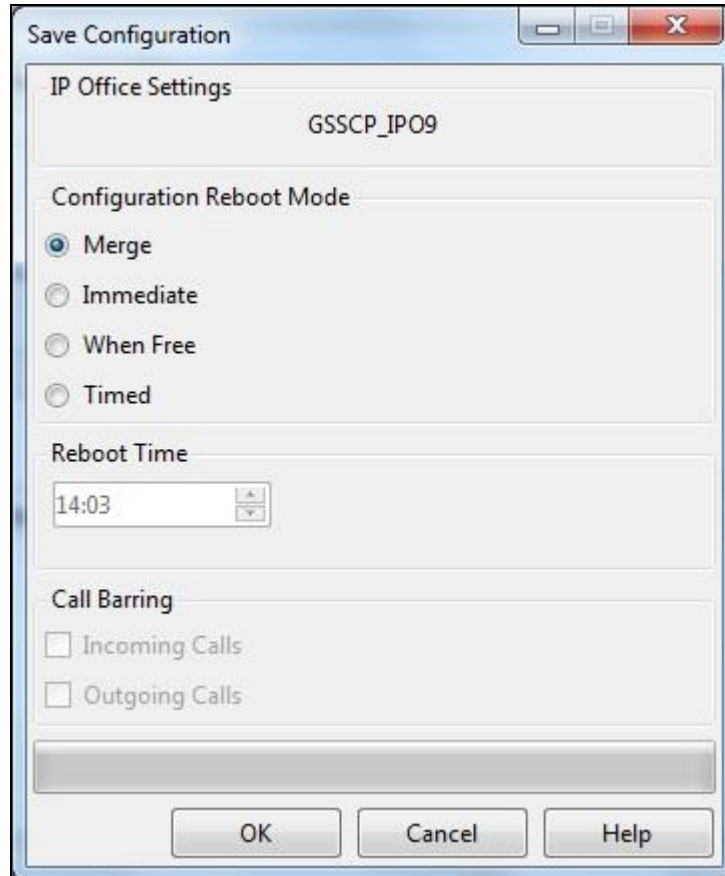
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number **+4989xxxxxx10** on line 18 are routed to extension 89107.

The screenshot shows the "Destinations" tab of the configuration window titled "18 +4989xxxxxx10*". It has three tabs: "Standard", "Voice Recording", and "Destinations". The "Destinations" tab is active. The fields and their values are:

Field	Value
TimeProfile	
Destination	89107 Extn89107

5.9. 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 standard window controls. 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 checkbox options: 'Incoming Calls' and 'Outgoing Calls'. At the bottom of the dialog are 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.

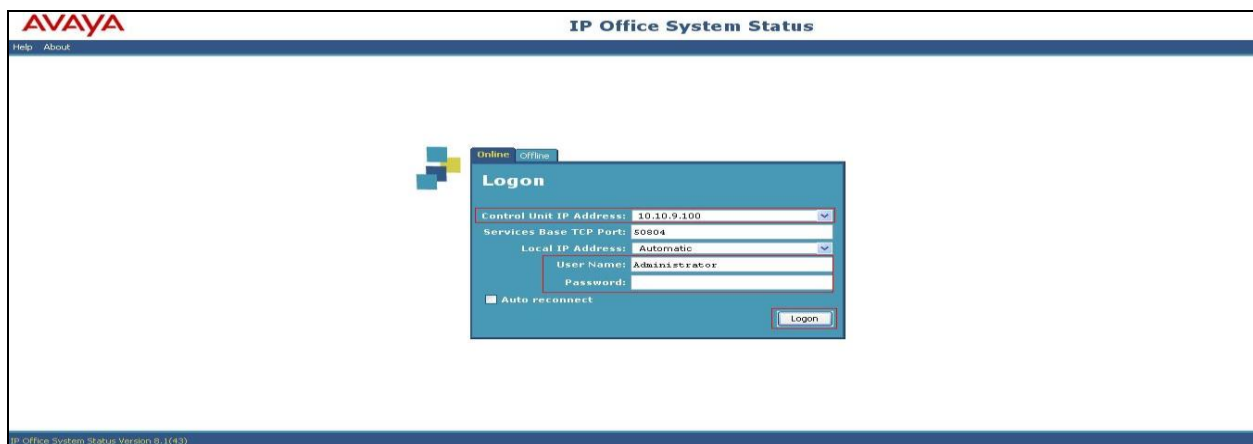
7. Verification Steps

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.

AVAYA

IP Office System Status

Help Snapshot LogOff About

System

Alarms (7)

Extensions (18)

Trunks (9)

Line: 1

Line: 2

Lines: 5 - 8

Line: 9

Line: 10

Line: 18

Active Calls

Resources

Voicemail

IP Networking

Status Utilization Summary Alarms

SIP Trunk Summary

Peer Domain Name: sip://192.168.230.98

Resolved Address: 192.168.230.98

Line Number: 18

Number of Administered Channels: 10

Number of Channels in Use: 0

Administered Compression: G729 A, G711 A

Silence Suppression: Off

SIP Trunk Channel Licenses: Unlimited

SIP Trunk Channel Licenses in Use: 0

SIP Device Features:

Channel Number

URI Gr...

Call Ref

Current State

Time in State

Remote Media Address

Codec

Connection Type

Caller ID or Dialed Digits

Other Party on Call

1

Idle

00:12:22

2

Idle

00:12:11

3

Idle

01:46:02

4

Idle

01:46:02

5

Idle

01:46:02

6

Idle

01:46:02

7

Idle

01:46:02

8

Idle

01:46:02

9

Idle

01:46:02

10

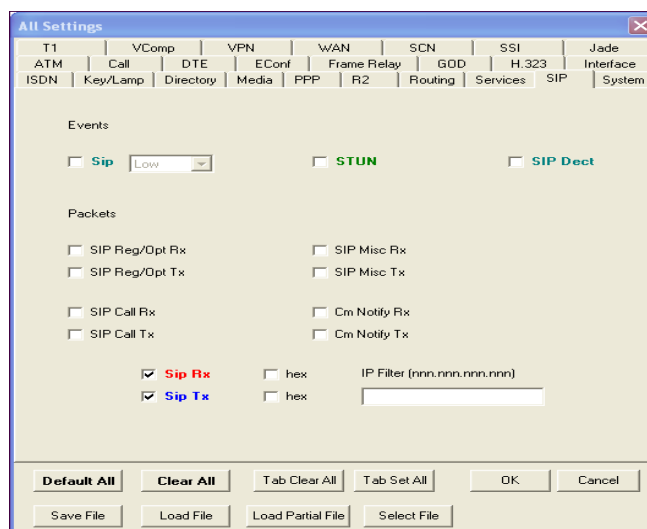
Idle

01:46:02

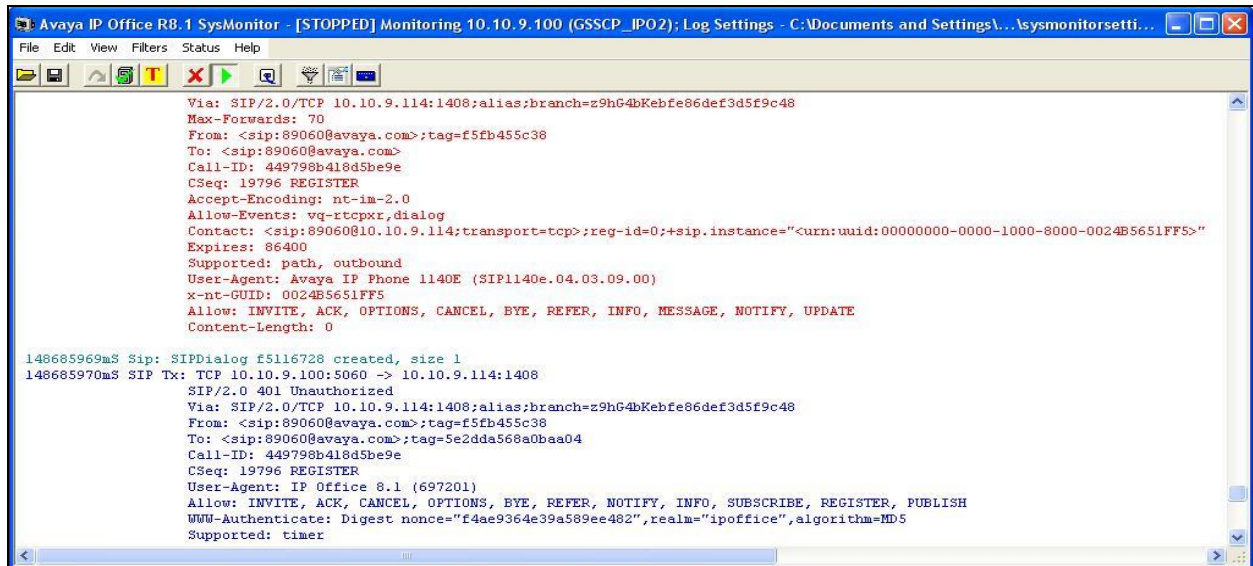
7.2. Monitor

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 10.10.9.114 to 10.10.9.100. The second message is a 401 Unauthorized response from 10.10.9.100 to 10.10.9.114.

```
Via: SIP/2.0/TCP 10.10.9.114:1408;alias=branch=z9hG4bKebfe86def3d5f9c48
Max-Forwards: 70
From: <sip:890608avaya.com>;tag=f5fb455c38
To: <sip:890608avaya.com>
Call-ID: 449798b418d5be9e
CSeq: 19796 REGISTER
Accept-Encoding: nt-im-2.0
Allow-Events: vg-rtcpxr,dialog
Contact: <sip:89060810.10.9.114;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 -> 10.10.9.114:1408
SIP/2.0 401 Unauthorized
Via: SIP/2.0/TCP 10.10.9.114:1408;alias=branch=z9hG4bKebfe86def3d5f9c48
From: <sip:890608avaya.com>;tag=f5fb455c38
To: <sip:890608avaya.com>;tag=5e2dda568a0baa04
Call-ID: 449798b418d5be9e
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 R10.0 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] *Avaya IP Office™ Platform Start Here First*, Release 10, August 2016.
- [2] *IP Office™ Platform 10.0 - Deploying Avaya IP Office Basic Edition*, Document number 15-601042, August 2016.
- [3] *IP Office™ Platform 10.0 Installing and Maintaining the Avaya IP Office™ Platform Application Server*, Document number 15-601011, August 2016.
- [4] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0, August 2016.
- [5] *IP Office™ Platform 10.0 Using System Status*, Document number 15-601758, August 2016.
- [6] *IP Office™ Platform 10.0 Using IP Office System Monitor*, Document number 15-601019, May 2016.
- [7] *Using Avaya Communicator for Windows on IP Office*, Release 10.0, August 2016.
- [8] *IP Office™ Platform 10.0 - Third-Party SIP Extension Installation Notes*, June 2016.
- [9] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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