



Avaya Solution & Interoperability Test Lab

Application Notes for Eircom SIP Trunk Service with Avaya IP Office Release 9.0 – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Session Initiation Protocol (SIP) trunking between Eircom SIP Trunk service and Avaya IP Office.

The Eircom SIP Trunk service provides PSTN access via a SIP trunk connected to the Eircom Voice Over Internet Protocol (VoIP) network as an alternative to legacy Analogue or Digital trunks. Eircom 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 Eircom SIP Trunk service and Avaya IP Office. Customers using this Avaya SIP-enabled enterprise solution with Eircom'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 Eircom SIP Trunk service. 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 G.711A and G.729A codecs
- Fax calls to/from a group 3 fax machine to a PSTN-connected fax machine using T.38
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- Inbound and outbound PSTN calls to/from an IP Office Softphone clients
- Various call types including: local, long distance, international, toll free (outbound) and directory assistance
- Caller ID presentation and Caller ID restriction
- Voicemail navigation for inbound and outbound calls
- User features such as hold and resume, transfer, and conference
- Off-net call forwarding and twinning

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for Eircom's SIP Trunk service with the following observations:

- 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 with the Emergency Services Operator.

2.3. Support

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

For technical support on Eircom products please contact Eircom Customer Care at:

- Telephone: 1800 255 255
- Telephone: +353 1 4688530
- Email: servicedesk@eircom.ie

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Eircom SIP Trunk service. Located at the enterprise site is an Avaya IP Office 500 v2. Endpoints include Avaya 1600 Series IP Telephones (with H.323 firmware), Avaya 9600 Series IP Telephones (with SIP firmware), Avaya 1140e SIP Telephones, Avaya 2420 Digital Telephone, Avaya Analogue Telephone and fax machine. The site also has a Windows XP PC running Avaya IP Office Manager to configure the Avaya IP Office as well as an IP Office Softphone client and Flare Experience for Windows for mobility testing. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been changed to a private format and all phone numbers have been obscured beyond the city code.

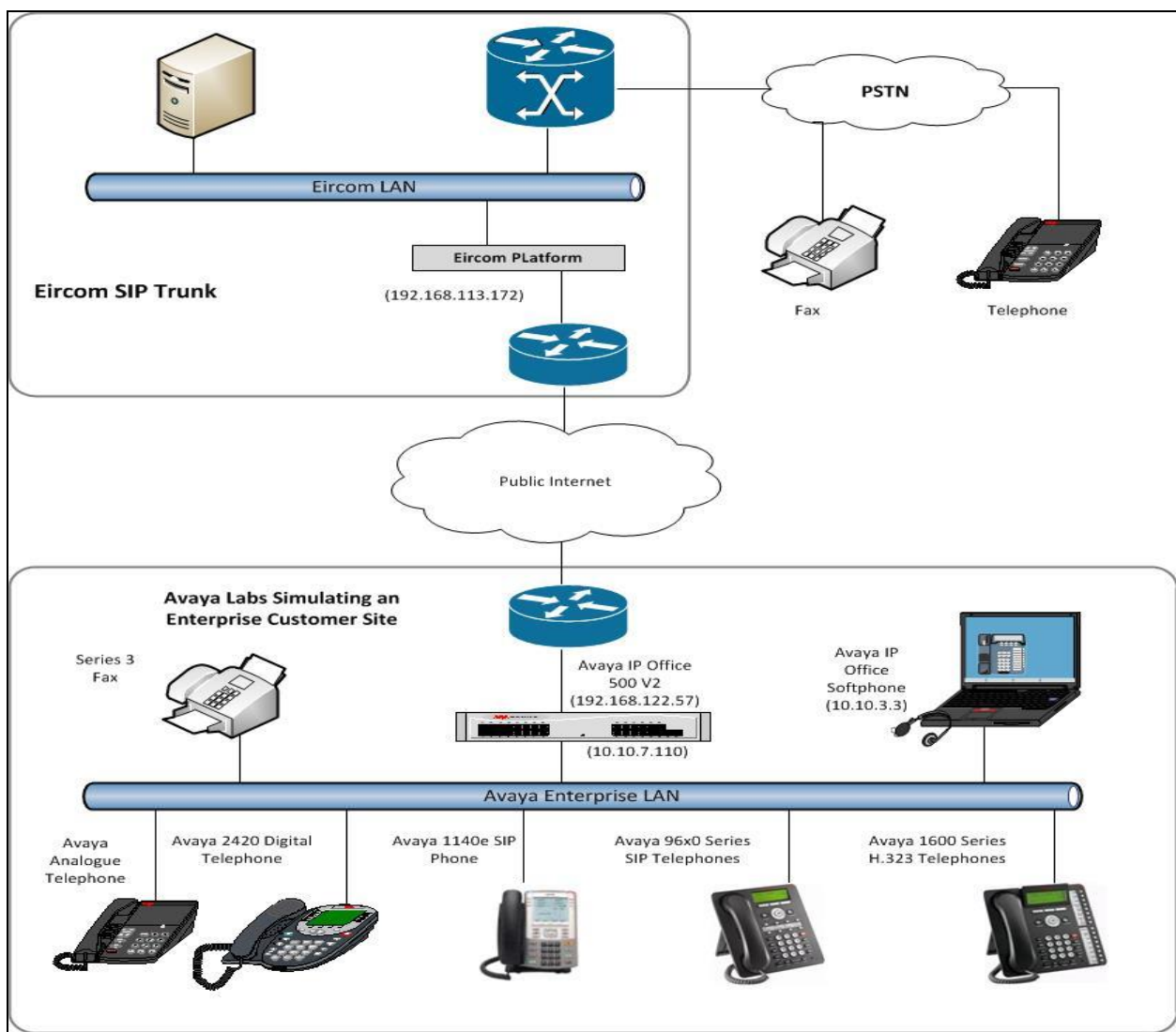


Figure 1: Test Setup Eircom SIP Trunk service to Simulated Avaya Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya	
Avaya IP Office 500 V2	Version 9.0.3.0 build 941
Avaya 1608 Phone (H.323)	1.3.5
Avaya 9600 Series Phone (SIP)	6.3.0
Avaya SoftPhone (SIP)	3.056516
Avaya Flare Experience for Windows (SIP)	1.1.3.14
Avaya 1140e (SIP)	FW: 04.04.10.00.bin
Avaya 2420 Digital Phone	R6.0
Avaya 98390 Analogue Phone	N/A
Eircom	
Eircom SIP Trunk Service	Broadsoft Broadworks rel 19SP1 Ericsson IMS rel 13A AcmePacket SD running on 4500 platform, software release 6.4

5. Configure Avaya IP Office

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

5.1. Verify System Capacity

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

License		Remote Server				
Feature	License Key	Instances	Status	Expiry Date	Source	
Office Worker	WtX400gbtXrB3O3Wqt_3rRbPd9_rdrx5	255	Valid	Never	ADI Nodal	
one-X Portal for IP Office	dTna3Bo4d5j@qq6q3fmc5uwyLMdWv1L9	255	Valid	Never	ADI Nodal	
Phone Manager Pro IP Audio Enabled...	MnzlkOhxVs50htJyK5pkoxgLYLMpYm21	255	Valid	Never	ADI Nodal	
Phone Manager Pro	JaBHzCy0Xd_hXk@gNmVxBhf1GMYg1xH	255	Valid	Never	ADI Nodal	
Phone Manager Pro (per seat)	bt95ih5rLV8F3OQBaUopgCjVNgLGypiz	255	Valid	Never	ADI Nodal	
IP Office Dealer Support - Profession...	kHr2Eo5vO233UeIkHmjBQdnA9_n582z	255	Valid	Never	ADI Nodal	
IP Office Dealer Support - Standard E...	jkM9f_d75GfjJNWFkB79xwSFLGxwv1VM	255	Valid	Never	ADI Nodal	
RAS LRQ Support (Rapid Response)	GKOlzGBLDdPdt0_pGqmf9JkZQ5N7LEse	255	Valid	Never	ADI Nodal	
Report Viewer	eveekN6mEIO0ZzeIBulJcn8u5lrYEErM	255	Valid	Never	ADI Nodal	
CCC Agent Rostering	PIn_3dL@tvY@yYwU74rjsDp1dnwLbC	255	Valid	Never	ADI Nodal	
Customer Service Agent	rU02FO5PVX8DNfCjfmue5@j61M7Y6s9	255	Valid	Never	ADI Nodal	
CCR CCC UPG	sTJh@vdhQ58c@HYkg@_35IRb9LLjGjJB	255	Valid	Never	ADI Nodal	
CCR Designer	faW7akviXUiz37rsADc7995AIQuIsp0c	255	Valid	Never	ADI Nodal	
CCR SUP	8U288A6IXTIFRrh32pRLJhhIZMWE7x5	255	Valid	Never	ADI Nodal	
Advanced Small Community Networking	eT@t6ISTtO942yxYwI7gBIG8A0oIw_8B	255	Obsolete	Never	ADI Nodal	
SIP Trunk Channels	unXMBE6x9dJKGKJ73uEpf7JrpF4smme	255	Valid	Never	ADI Nodal	
Small Office Edition VCM (channels)	eABRzdgr9vhDAe9YG0uwrpqHEGuLJueM	255	Obsolete	Never	ADI Nodal	
Small Office Edition WIFI	2XQ66z5GdVEFD6e61deHwEu9LNi5QuW	255	Obsolete	Never	ADI Nodal	
Small Site Software Upgrade 255	IGXCiQSLMtVPa4TK6H7qwwReU0EJAWMj	1	Valid	Never	ADI Nodal	
Receptionist	wGM8o16nddQm2mr@UMm9LwA@XRFhgM0e	255	Valid	Never	ADI Nodal	
Software Upgrade 255	wqxeh6@XLO1ZV@MFkjo6BCY@IdFyggZu	1	Valid	Never	ADI Nodal	
CCC Spectrum Wallboards	MAXOapb3VvFZdn614ep_kkR2GEZV5w@C	255	Valid	Never	ADI Nodal	

5.2. LAN2 Settings

In the sample 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_IPO2** 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).

GSSCP_IP09*

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

LAN Settings VoIP Network Topology

IP Address192 · 168 · 122 · 57

IP Mask255 · 255 · 255 · 128

Primary Trans. IP Address0 · 0 · 0 · 0

Firewall Profile<None>

RIP ModeNone

☐ Enable NAT

Number Of DHCP IP Addresses200

DHCP Mode

☐ Server ☐ Client ☐ Dialin ☒ Disabled

Advanced

On the **VoIP** tab in the Details Pane, check the **SIP Trunks Enable** box to enable the configuration of SIP trunks. The IP Office Softphone uses SIP. If Softphone 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 LAN 2 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 VoIP tab active. The configuration is as follows:

- System** tab: LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, Codecs.
- VoIP** tab:
 - ☒ H323 Gatekeeper Enable
 - ☐ Auto-create Extn
 - ☐ Auto-create User
 - ☐ H323 Remote Extn Enable
 - ☒ 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** section:
 - Port Number Range: Minimum 20000, Maximum 30000
 - Port Number Range (NAT): Minimum 49152, Maximum 53246
 - ☐ Enable RTCP Monitoring on Port 5005
 - Keepalives:
 - Scope: RTP-RTCP
 - Periodic timeout: 5
 - Initial keepalives: Enabled
- DiffServ Settings** section:
 - DSCP (Hex): 88, Video DSCP (Hex): FC, DSCP Mask (Hex): 88, SIG DSCP (Hex): 88
 - DSCP: 46, Video DSCP: 46, DSCP Mask: 63, SIG DSCP: 34

Select the **Network Topology** tab as shown in the following screen. In the sample configuration, the default settings were used and the **Use Network Topology Info** in the **SIP Line** was set to “None” in **Section 5.6.2**. It is important that the **Binding Refresh Time** is set to the correct value. Avaya IP Office sends SIP OPTIONS messages periodically to determine if the SIP connection is active. Below is a sample configuration. 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 settings:

- STUN Server Address: [Empty text box]
- STUN Port: 3478 (spin box)
- Firewall/NAT Type: Open Internet (dropdown menu)
- Binding Refresh Time (seconds): 200 (spin box)
- Public IP Address: 0 . 0 . 0 . 0 (IP address field)
- Run STUN button and Cancel button

Below the main settings is a 'Public Port' section with three spin boxes for UDP, TCP, and TLS, all set to 0.

At the bottom, there is a checkbox labeled 'Run STUN on startup' which is currently unchecked.

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 shows the GSSCP_IP09* configuration window with the 'Telephony' tab selected. The 'Companding Law' section is highlighted with a red box, showing 'A-Law' selected for both 'Switch' and 'Line'. The 'Inhibit Off-Switch Forward/Transfer' checkbox is also highlighted with a red box and is unchecked. Other settings visible include 'Default Outside Call Sequence' set to 'Normal', 'Default Inside Call Sequence' set to 'Ring Type 1', 'Default Ring Back Sequence' set to 'Ring Type 2', and 'Restrict Analogue Extension Ringer Voltage' set to 'No'.

5.4. 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 Eircom SIP Trunk.

The screenshot shows the GSSCP_IP09 configuration window with the 'Twining' tab selected. The 'Send original calling party information for Mobile Twinning' checkbox is unchecked, and the 'Calling party information for Mobile Twinning' field is empty.

5.5. 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.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** 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, CCR, 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, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' section is further divided into 'Unused' and 'Selected' sub-sections. The 'Unused' section contains G.711 ULAW 64K, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' section, which is highlighted with a red box, contains G.729(a) 8K CS-ACELP and G.711 ALAW 64K. Between the 'Unused' and 'Selected' sections are four buttons: '>>', '<<', '<<', and '>>', used for moving codecs between the lists.

5.6. SIP Line

A SIP line is needed to establish the SIP connection between Avaya IP Office and the Eircom SIP Trunking service. The recommended method for configuring a SIP Line is to use the template associated with these Application Notes. The template is an .xml file that can be used by IP Office Manager to create a SIP Line. Follow the steps in **Section 5.6.1** to create the SIP Line from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration shown in **Section 5.6.2**.

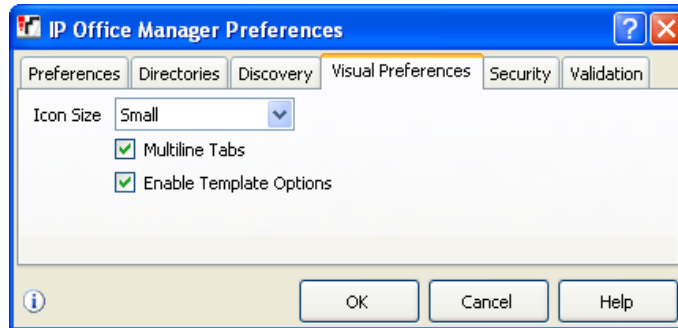
Also, the following SIP Line settings are not supported on Basic Edition:

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

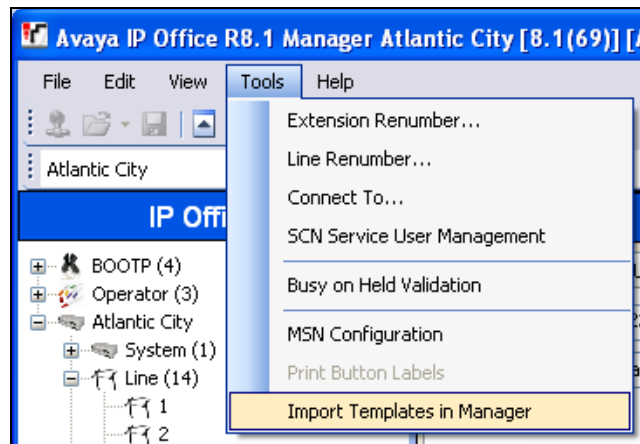
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Section 5.6.2**.

5.6.1. SIP Line From Template

1. Copy the template file to the computer where IP Office Manager is installed. Rename the template file to **IE_Eircom_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**.

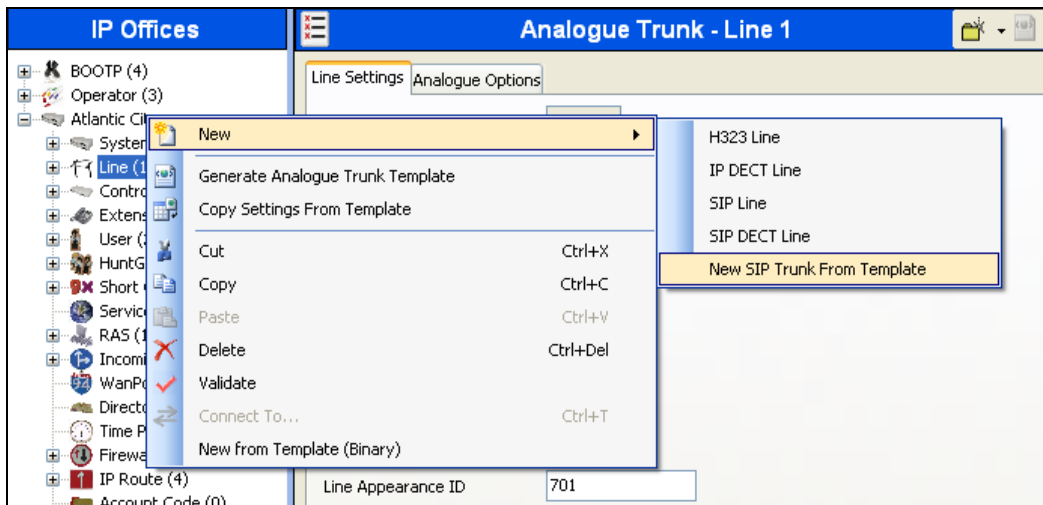


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools** → **Import Templates in Manager**. This action will copy the template file into the IP Office template directory and make the template available in the IP Office Manager pull-down menus in **Step 5**. The default template location is **C:\Program Files\Avaya\IP Office\Manager\Templates**.



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 **Ireland** from the **Country** pull-down menu and select **Eircom** from the **Service Provider** pull-down menu as shown below. These values correspond to parts of the file name (**IE_Eircom_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 5.6.2**.

5.6.2. SIP Line – SIP Line Tab

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 **ngv.eircom.net**.
- Ensure the **In Service** box is checked.
- Set **REFER Supported** to Auto.
- Set **Method for Session Refresh** to **Reinvite**.
- Ensure the **Check OOS** box is checked.
- Default values may be used for all other parameters.

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

The screenshot shows the 'SIP Line - Line 18*' configuration window. It has tabs for 'SIP Line', 'Transport', 'SIP URI', 'VoIP', 'T38 Fax', and 'SIP Credentials'. The 'SIP Line' tab is active. The configuration is divided into two columns. The left column contains: 'Line Number' (18), 'ITSP Domain Name' (ngv.eircom.net), 'Prefix' (empty), 'National Prefix' (0), 'Country Code' (empty), 'International Prefix' (empty), 'Send Caller ID' (None), 'Association Method' (By Source IP address), 'REFER Support' (checked), 'Incoming' (Auto), 'Outgoing' (Auto), and 'Method for Session Refresh' (Reinvite). The right column contains: 'In Service' (unchecked), 'URI Type' (SIP), 'Check OOS' (checked), 'Call Routing Method' (Request URI), 'Originator number for forwarded and twinning calls' (empty), 'Name Priority' (System Default), 'Caller ID from From header' (unchecked), 'Send From In Clear' (unchecked), 'User-Agent and Server Headers' (empty), 'Service Busy Response' (486 - Busy Here), and 'Action on CAC Location Limit' (Allow Voicemail).

Parameter	Value
Line Number	18
ITSP Domain Name	ngv.eircom.net
Prefix	
National Prefix	0
Country Code	
International Prefix	
Send Caller ID	None
Association Method	By Source IP address
REFER Support	Checked
Incoming	Auto
Outgoing	Auto
Method for Session Refresh	Reinvite
In Service	Unchecked
URI Type	SIP
Check OOS	Checked
Call Routing Method	Request URI
Originator number for forwarded and twinning calls	
Name Priority	System Default
Caller ID from From header	Unchecked
Send From In Clear	Unchecked
User-Agent and Server Headers	
Service Busy Response	486 - Busy Here
Action on CAC Location Limit	Allow Voicemail

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

- Set **ITSP Proxy Address** to the IP address of the Eircom SIP proxy
- Set **Layer 4 Protocol** to **UDP**
- Set **Send Port** to **5060** and **Listen Port** to **5060**
- Set **Network Topology Info** to **None**

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

The screenshot shows the 'SIP Line - Line 18*' configuration window with the 'Transport' tab selected. A red box highlights the 'ITSP Proxy Address' field (192.168.113.172) and the 'Network Configuration' section. The 'Network Configuration' section includes 'Layer 4 Protocol' set to 'UDP', 'Send Port' set to '5060', 'Use Network Topology Info' set to 'None', and 'Listen Port' set to '5060'. Below this, there are fields for 'Explicit DNS Server(s)' (0 . 0 . 0 . 0 . 0 . 0) and a checked 'Calls Route via Registrar' checkbox. A 'Separate Registrar' field is also present.

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. A table with columns 'Channel', 'Groups', 'Via', 'Local URI', 'Contact', 'Display Name', 'PAI', 'Credential', and 'Max Calls' is visible. To the right of the table are three buttons: 'Add...', 'Remove', and 'Edit...'. The 'Add...' button is highlighted with a red box.

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 5.8**.
- For **Registration**, select **1: <pxxxxxxxxx@ngv.eircom.net>** from the pull-down menu since this configuration uses SIP registration.
- Associate this line with an incoming line group by entering a line group number in the **Incoming Group** field. This line group number will be used in defining incoming call routes for this line. Similarly, associate the line to an outgoing line group using the **Outgoing Group** field. The outgoing line group number is used in defining short codes for routing outbound traffic to this line. For the compliance test, a new incoming and outgoing group **18** was defined that was associated to a single line (line 18).
- Set **Max Calls per Channel** to the number of simultaneous SIP calls that are allowed using this SIP URI pattern.

The screenshot displays the 'SIP Line - Line 18*' configuration window. The 'SIP URI' tab is selected, showing a table with columns: Channel, Groups, Via, Local URI, Contact, Display Name, and PAI. The table contains one entry with Channel 1 and Groups 18 18. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table, the 'Edit Channel' dialog box is open, showing fields for 'Via' (set to '<None>'), 'Local URI', 'Contact', 'Display Name', and 'PAI' (all set to 'Use Internal Data'), 'Registration' (set to '1: pxxxxxxxx_TG1@ng'), 'Incoming Group' (set to 18), 'Outgoing Group' (set to 18), and 'Max Calls per Channel' (set to 10). The dialog box has 'OK' and 'Cancel' buttons.

Channel	Groups	Via	Local URI	Contact	Display Name	PAI
1	18 18	<...>				

Buttons: Add..., Remove, Edit...

Edit Channel

Via: <None>

Local URI: Use Internal Data

Contact: Use Internal Data

Display Name: Use Internal Data

PAI: Use Internal Data

Registration: 1: pxxxxxxxx_TG1@ng

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 10

Buttons: 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 **Custom** from the drop-down menu.
- Select **G.729(a) 8K CS-ACELP** and **G.711 ALAW 64K** codecs.
- Select the **Fax Transport Support** box to **T.38**.
- 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 Eircom 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 shows 'Custom' selected in the dropdown. The 'Unused' list contains 'G.711 ULAW 64K', 'G.722 64K', and 'G.723.1 6K3 MP-MLQ'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP' and 'G.711 ALAW 64K'. The 'Fax Transport Support' is set to 'T38'. The 'Location' is set to 'Cloud'. The 'Call Initiation Timeout (s)' is set to '4'. The 'DTMF Support' is set to 'RFC2833'. On the right, the 'VoIP Silence Suppression' checkbox is unchecked, 'Allow Direct Media Path' is unchecked, 'Re-invite Supported' is checked, 'Codec Lockdown' is unchecked, 'PRACK/100rel Supported' is checked, 'Force direct media with phones' is unchecked, and 'G.711 Fax ECAN' is unchecked.

Select the **T.38 Fax** tab, to set the T.38 parameters for the line. Un-check the Use Default Values box (not shown) and select **0** from the **T38 Fax Version** drop down menu. Set the **Max Bit Rate (bps)** to **14400**. All other field may retain their default values. On completion, click the **OK** button (not shown).

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

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

5.7. Short Codes

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 **0N** which will insert the prefix **0** before a dialled number **N** which is a requirement by Eircom for all outbound calls to terminate successfully. 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 5.6**

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

9N;; Dial	
Short Code	
Code	9N;
Feature	Dial
Telephone Number	0N
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

For incoming calls from mobility extension to FNE features hosted by IP Office to provide dial tone or mobile callback functionalities, Short Code **FNE00** was created.

- In the **Code** field, enter the FNE feature code as **FNE00**.
- Set **Feature** to **FNE Service**.
- Set **Telephone Number** to **00** for **FNE00**.
- Set the **Line Group Id** to **18** which is the outgoing line group number defined on the SIP URI tab on the SIP Line in **Section 5.6**.

The screenshot shows a configuration window titled "FNE00: FNE Service". On the left is a sidebar with a "Short Code" tab selected. The main area contains a form with the following fields:

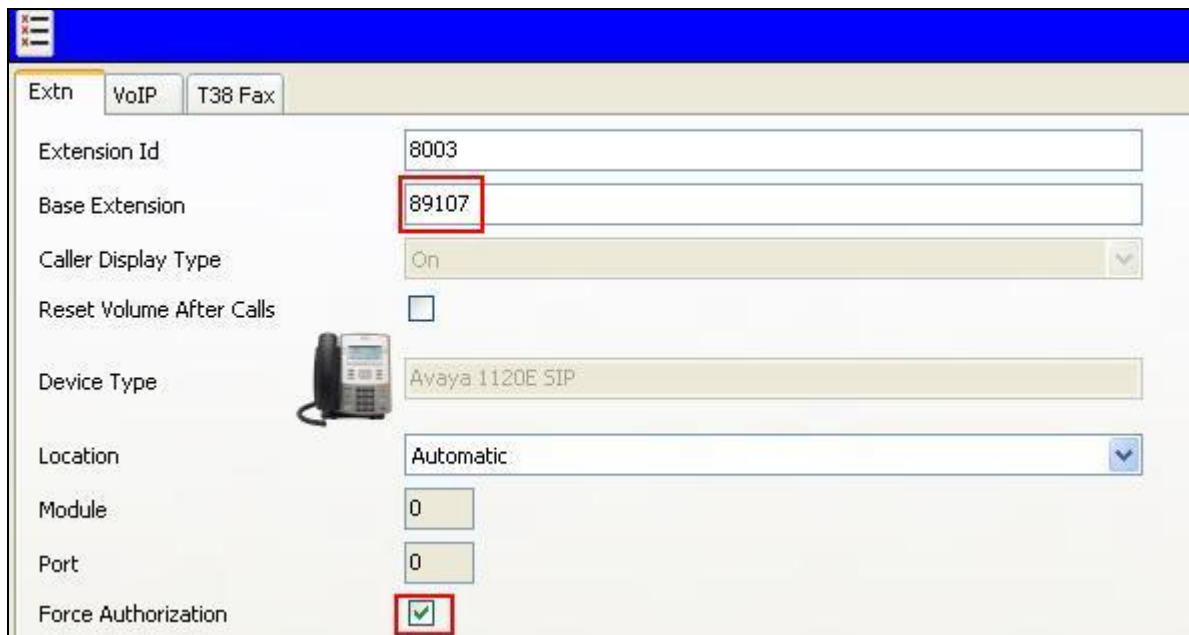
Field	Value
Code	FNE00
Feature	FNE Service
Telephone Number	00
Line Group ID	18
Locale	
Force Account Code	<input type="checkbox"/>

A red rectangular box highlights the "Code", "Feature", "Telephone Number", and "Line Group ID" fields.

5.8. User and Extensions

In this section, examples of IP Office Users, Extensions, and Hunt Groups will be illustrated. In the interests 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** 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 Authorization** box is checked.



The screenshot displays the configuration window for a SIP extension. The 'Extn' tab is active, showing various fields for configuration. The 'Base Extension' field is highlighted with a red box and contains the value '89107'. The 'Force Authorization' checkbox is also highlighted with a red box and is checked. Other fields include 'Extension Id' (8003), 'Caller Display Type' (On), 'Device Type' (Avaya 1120E SIP), 'Location' (Automatic), 'Module' (0), and 'Port' (0).

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. Check the **Reserve Avaya IP endpoint license** box. The new **Codec Selection** parameter may retain the default setting “System Default” to follow the system configuration shown in **Section 5.5**. 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.

SIP Extension: 8003 89107

Extn | **VoIP** | T38 Fax

IP Address: 0 . 0 . 0 . 0

Codec Selection: System Default

Unused

G.711 ULAW 64K
G.722 64K

Selected

G.711 ALAW 64K
G.729(a) 8K CS-ACELP

>>
↑
<<
↓
>>

Reserve License: None

Fax Transport Support: None

TDM->IP Gain: Default

IP->TDM Gain: Default

DTMF Support: RFC2833

☐ VoIP Silence Suppression

☐ Local Hold Music

☒ Allow Direct Media Path

☒ Re-invite Supported

☐ Codec Lockdown

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 5.6**. 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.

Extn89107: 89107*

User | Voicemail | DND | Short Codes | Source Numbers | Telephony | Forwarding | Dial In | Voice Recording | Button Programming

Name: Extn89107

Password: *****

Confirm Password: *****

Account Status: Enabled

Full Name: Ext 89107

Extension: 89107

Email Address:

Locale:

Priority: 5

System Phone Rights: None

Profile: Power User

☐ Receptionist

☐ Enable Softphone

☐ Enable one-X Portal Services

☒ Enable one-X TeleCommuter

☒ Enable Remote Worker

☐ Enable Flare

☐ Enable Mobile VoIP Client

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.

Ext89107: 89107*

User Voicemail DND Short Codes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

Call Settings **Supervisor Settings** Multi-line Options Call Log TUI

Login Code *****

Login Idle Period (secs)

Monitor Group <None>

Coverage Group <None>

Status on No-Answer Logged On (No change)

Reset Longest Idle Time

☒ All Calls

☐ External Incoming

After Call Work Time (secs) System Default (10)

☐ Force Login

☐ Force Account Code

☐ Incoming Call Bar

☐ Outgoing Call Bar

☐ Inhibit Off-Switch Forward/Transfer

☐ Can Intrude

☒ Cannot be Intruded

☐ Can Trace Calls

☐ CCR Agent

☐ Automatic After Call Work

☐ 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 incoming notifications and transfer operations.

Ext89107: 89107*

User Voicemail DND Short Codes Source Numbers **Telephony** Forwarding Dial In Voice Recording Button Programming

Call Settings Supervisor Settings Multi-line Options Call Log TUI

Outside Call Sequence Default Ring

Inside Call Sequence Default Ring

Ringback Sequence Default Ring

No Answer Time (secs) System Default (15)

Wrap-up Time (secs) 2

Transfer Return Time (secs) Off

Call Cost Mark-Up 100

☒ **Call Waiting On**

☐ Answer Call Waiting On Hold

☐ Busy On Held

☐ 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 5.6**). As such, these fields should be set to one of the DDI numbers assigned to the enterprise from Eircom.

The screenshot shows the configuration interface for a SIP line. The title bar reads "Ext89107: 89107*". Below the title bar is a "Personal Directory" section with a row of tabs: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, Mobility, Group Membership, Announcements, and SIP. The SIP tab is selected. In the main area, there are three text input fields: "SIP Name" with the value "+35376xxxxx10", "SIP Display Name (Alias)" with the value "+35376xxxxx10", and "Contact" with the value "+35376xxxxx10". These three fields are enclosed in a red rectangular box. Below these fields is an "Anonymous" checkbox, which is currently unchecked.

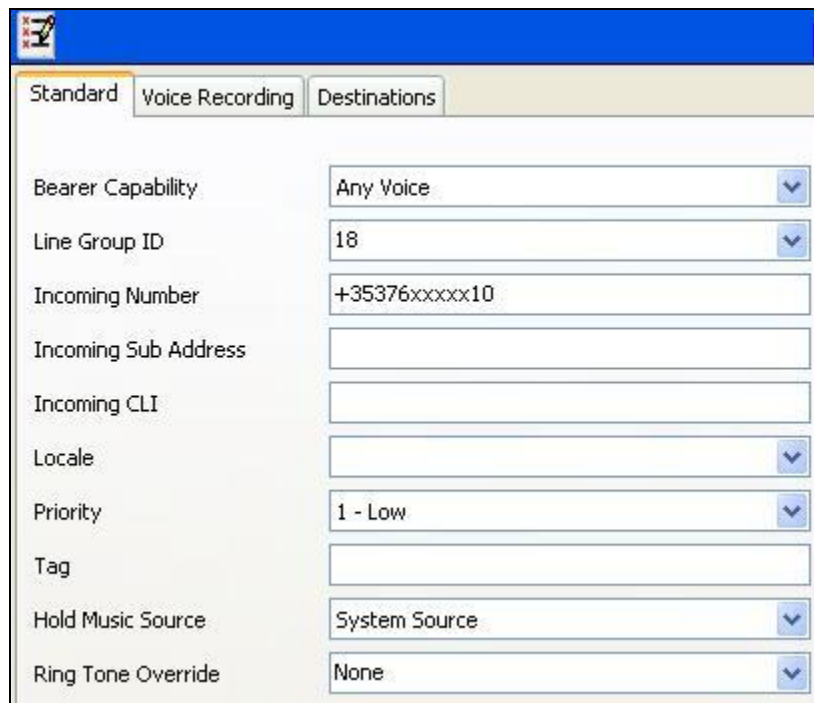
The following screen shows the Mobility tab for user 89101. The **Mobility Features** and **Mobile Twinning** are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone over the SIP Trunk. Other options can be set accordingly to customer requirements.

The screenshot shows the configuration interface for a Mobility line. The title bar reads "Ext89101: 89101*". Below the title bar is a "Personal Directory" section with a row of tabs: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, and Mobility. The Mobility tab is selected. In the main area, there are several sections. The "Internal Twinning" section is collapsed. The "Mobility Features" section is expanded and contains a "Mobile Twinning" sub-section, which is also expanded. The "Mobile Twinning" sub-section contains several fields: "Twinned Mobile Number (including dial access code)" with the value "9+353894xxxxx1", "Twinning Time Profile" with a dropdown menu showing "<None>", "Mobile Dial Delay (secs)" with the value "3", and "Mobile Answer Guard (secs)" with a value of "0". There are also four checkboxes: "Twin Bridge Appearances", "Twin Coverage Appearances", "Twin Line Appearances", and "Hunt group calls eligible for mobile twinning", all of which are unchecked. At the bottom of the "Mobile Twinning" sub-section, there are three checkboxes: "one-X Mobile Client" (unchecked), "Mobile Call Control" (checked), and "Mobile Callback" (checked). The entire "Mobile Twinning" sub-section is enclosed in a red rectangular box.

5.9. Incoming Call Routing

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

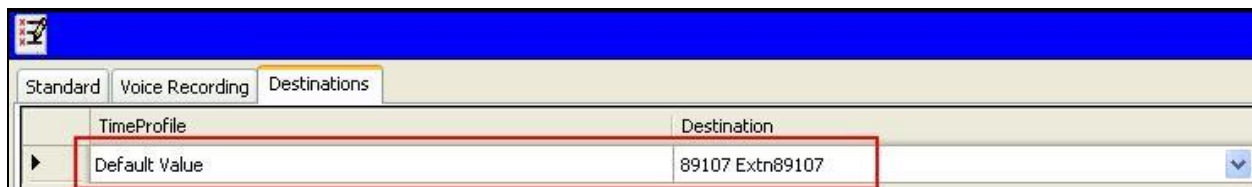
- Set the **Bearer Capacity** to **Any Voice**.
- Set the **Line Group Id** to the incoming line group of the SIP line defined in **Section 5.6**.
- Set the **Incoming Number** to the incoming number that this route should match on. Matching is right to left.
- Default values can be used for all other fields.



The screenshot shows the 'Standard' tab of the Incoming Call Routes configuration window. The fields and their values are as follows:

Field	Value
Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	+35376xxxxx10
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

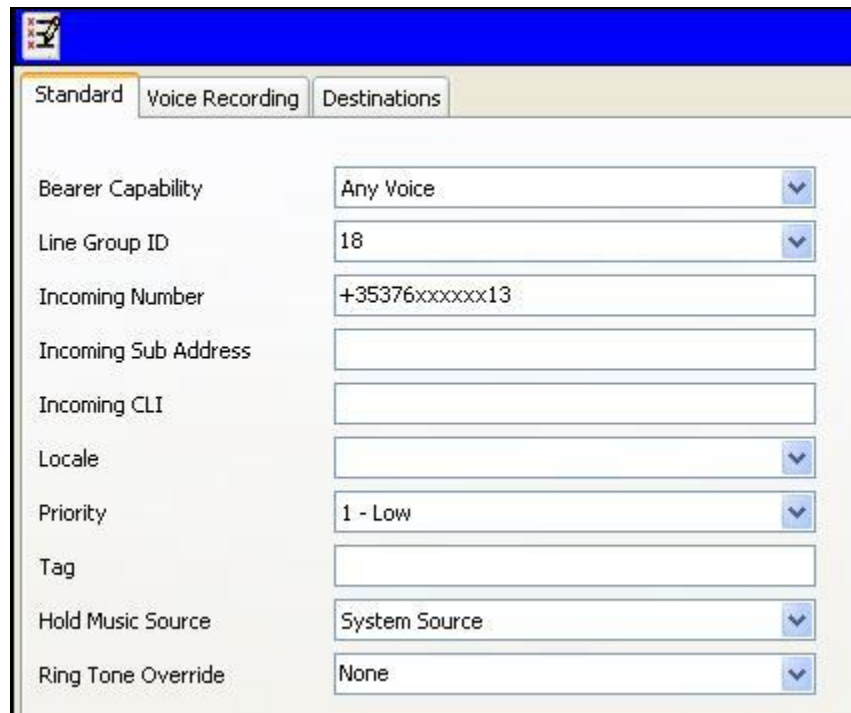
On the **Destinations** tab, select the destination extension from the pull-down menu of the **Destination** field. On completion, click the **OK** button (not shown). In this example, incoming calls to the test DDI number on line 18 are routed to extension 89107.



The screenshot shows the 'Destinations' tab of the Incoming Call Routes configuration window. The fields and their values are as follows:

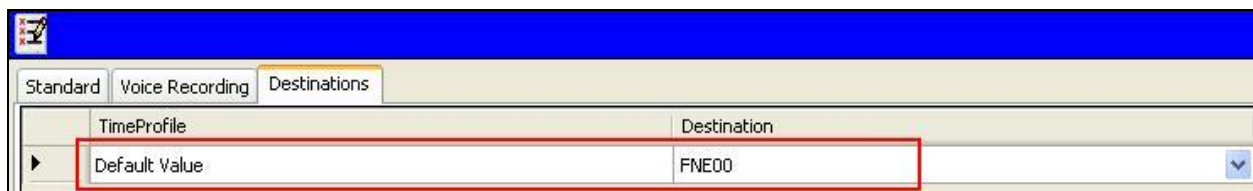
Field	Value
TimeProfile	Default Value
Destination	89107 Extn89107

Incoming Call Routes for other direct mappings of DDI numbers to IP Office users or FNE short codes etc. can be configured in the same fashion. In the screenshot below, the incoming call route for **+35376xxxxxx13** mapped to a shortcode **FNE** is illustrated.



Bearer Capability	Any Voice
Line Group ID	18
Incoming Number	+35376xxxxxx13
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

The **Destinations** tab contains the Destination field FNE00 which was entered manually. An incoming call to **+35376xxxxxx13** will be delivered directly to internal dial tone allowing the caller to perform dialing actions to both internally and external calls. The incoming caller ID must match the Twinned Mobile Number entered in User Mobility tab (**Section 5.8**); otherwise, IP Office responds with a 486 Busy Here and busy tone.



TimeProfile	Destination
Default Value	FNE00

5.10. Privacy / Anonymous Calls

There are multiple methods for a user to withhold outgoing identification:

- Dialing the short code *67 to access the SIP Line.
- Specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user.
- Avaya Telephones equipped with a “Features” button can also request privacy for a specific call, without dialing a unique short code, using **Features** → **Call Settings** → **Withhold Number**, on the phone itself.

To configure IP Office to include the caller’s DID number in the P-Asserted-Identity SIP header, required to admit an otherwise anonymous caller to the network, the following procedure may be used.

From the Navigation pane, select **User**. From the Group pane, scroll down past the configured users and select the user named **NoUser**. From the NoUser Details pane, select the tab **Source Numbers**. Press the **Add** button to the right of the list of any previously configured Source Numbers. In the **Source Number** field, type **SIP_USE_PAI_FOR_PRIVACY**. Click **OK**.



The source number **SIP_USE_PAI_FOR_PRIVACY** should now appear in the list of Source Numbers as shown below.



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.

6. Eircom SIP Trunk Service Configuration

Eircom is responsible for the configuration of the SIP Trunk service. The customer will need to provide the public IP address used to reach the Avaya IP Office at the enterprise. Eircom will provide the customer the necessary information to configure the SIP connection to the SIP Trunking service including:

- IP address of SIP Trunking SIP proxy
- Network SIP Domain
- Supported codecs
- DDI numbers
- All IP addresses and port numbers used for signalling or media that will need access to the enterprise network through any security devices.

7. Verification Steps

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

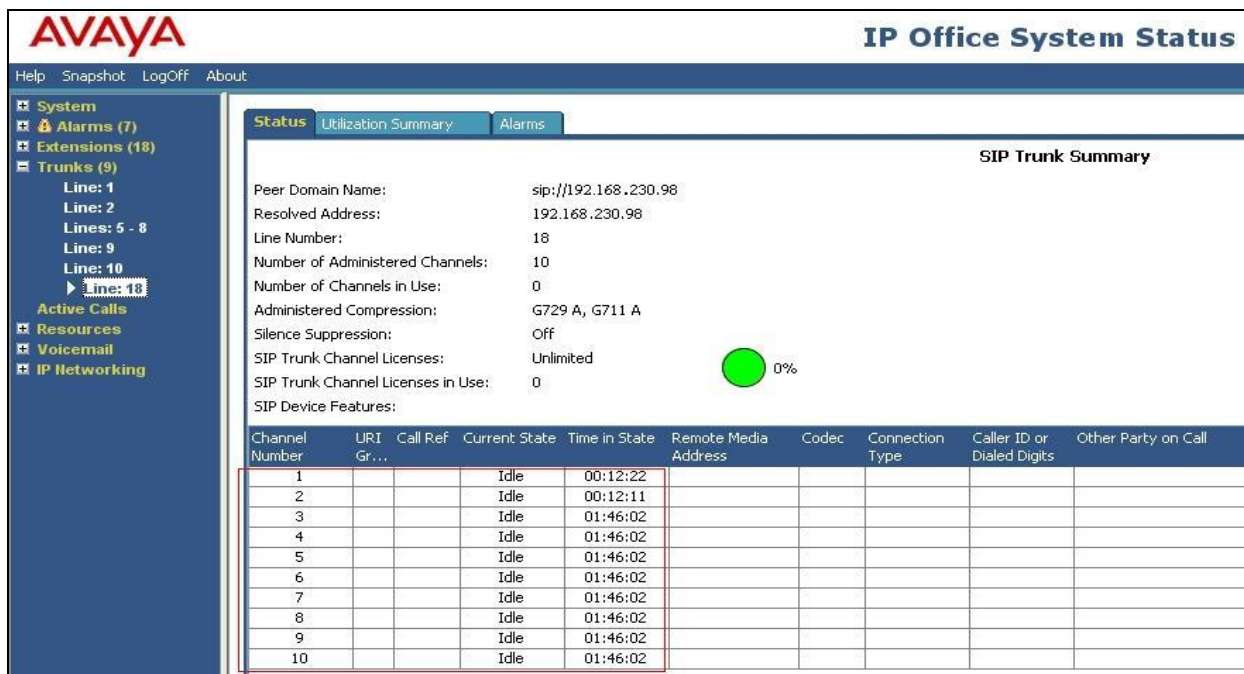
7.1. SIP Trunk status

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 into 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.



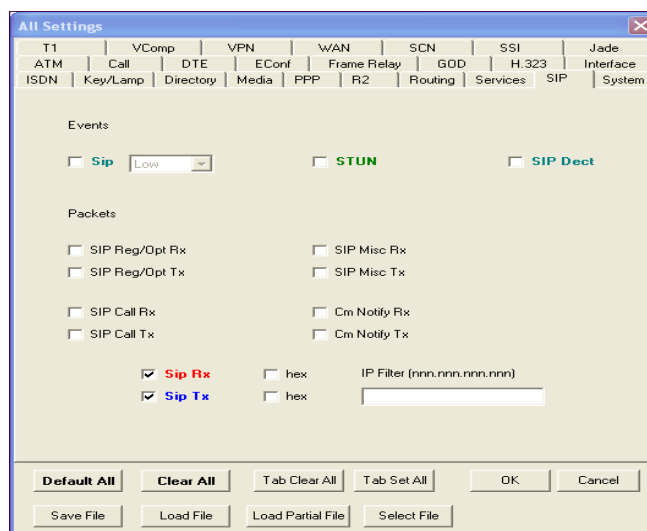
From the left hand menu expand **Trunks** and choose the SIP trunk (**18** in this instance). The status window will show the status as being idle and time in state if the Trunk is operational. IP address has been changed.



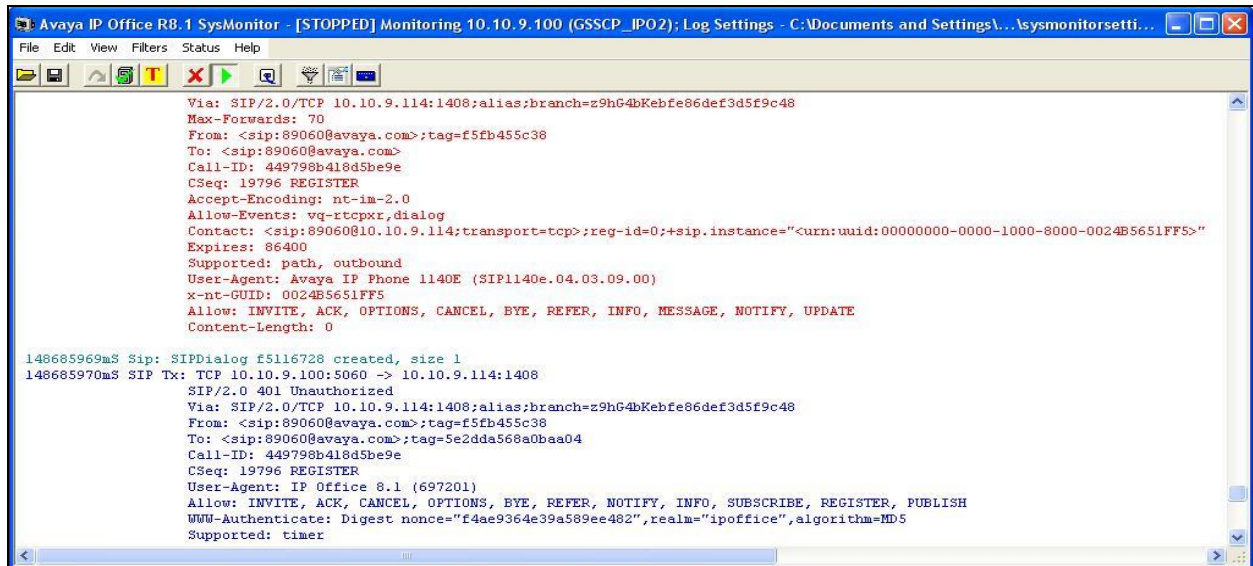
7.2. Monitor

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 button that is third from the right in the screen below, or select **Filters → Trace Options**.

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



As an example, the following shows a portion of the monitoring window for a Registration attempt to the SIP trunk.



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, which includes a WWW-Authenticate header with a Digest nonce and MD5 algorithm.

```
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 and Eircom 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 Eircom's SIP Trunk service. Eircom's 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 9.0* Documentation CD, Jan 2014.
- [2] *IP Office 9.0 Installation Manual*, Document Number 15-601042, Jan 2014.
- [3] *IP Office Manager Manual 10.0*, Document Number 15-601011, Jan 2014
- [4] *IP Office Release 9.0 Implementing Voicemail Pro*, Dec 2013
- [5] *System Status Application*, November 2013
- [6] *IP Office Softphone Installation*, September 2014
- [7] *IP Office SIP Extension Installation*, October 2014
- [8] *Avaya IP Office Knowledgebase*, <http://marketingtools.avaya.com/knowledgebase>
- [9] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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