Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Belgacom SIP Service and an Avaya IP Office Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the Belgacom SIP Service and Avaya IP Office. The Avaya solution consists of Avaya IP Office and various IP Telephones.

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 procedure for configuring Session Initiation Protocol (SIP) trunking between the Belgacom SIP trunking network and an Avaya SIP telephony solution consisting of Avaya IP Office and Avaya H.323 IP telephones. Avaya digital and analog telephones can be used as well, but were not included in the test configuration.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [3] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by Belgacom. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

Note that FAX transmission was done without T.38 using in-band transmission via the G.711A codec.

1.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing:

- Incoming & outgoing basic calls, including busy, no answer, calling party hang-up, called party hang-up.
- Outbound calls to domestic and international PSTN and GSM national and international endpoints.
- Codec support and priority selection.
- DTMF tone generation and recognition using RFC 2833 [4].
- Calling Party Number and Called Party Number presentation and restriction for incoming and outgoing calls.
- Call forwarding unrestricted / busy /no answer to local extension, PSTN, and GSM endpoints.
- Call forwarding to busy endpoint.
- Supervised Call Transfer / Blind Call Transfer to local extension, PSTN, and GSM endpoints.
- Conference Call with local and PSTN endpoints.
- Fax Send / Receive using the G.711 codec.
- Simultaneous Calls.
- Long Calls.
- Calls with both ends muted.

1.2. Support

Support is available at:
http://www.belgacom.be/private/hbsres/jsp/dynamic/homepage.jsp

Prior registration is required, which can be done at
http://www.belgacom.be/private/en/jsp/dynamic/productCategory.jsp?dcrName=hbsres_cockpit

2. Reference Configuration

The following diagram illustrates the configuration which was used for testing.

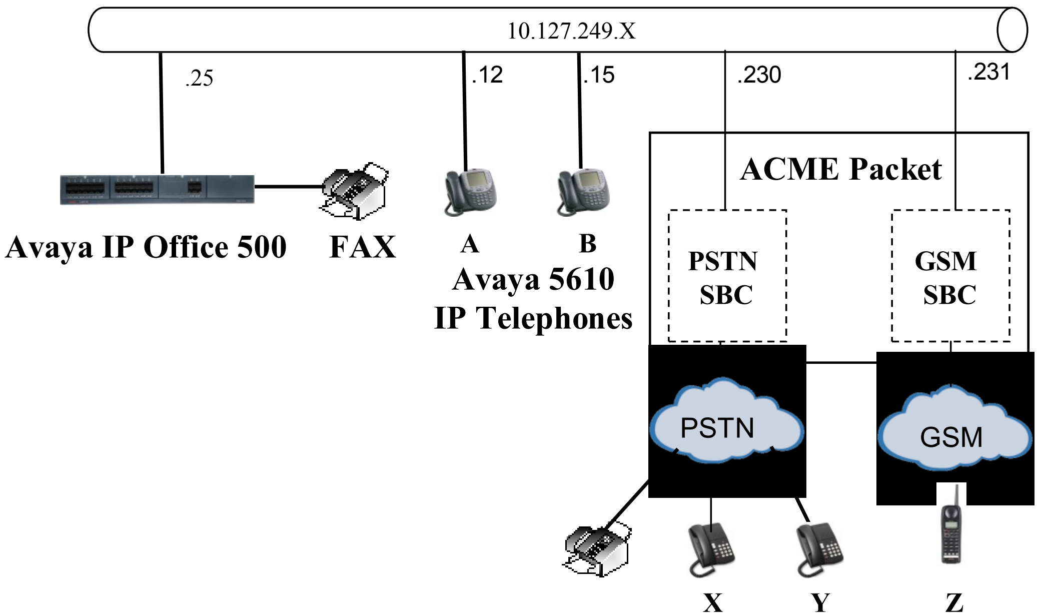

Figure 1: Reference Configuration

The following endpoints were used for testing:

Endpoint	Type	Extension	Number
A	5610SW IP	332	02xxxx 332
B	5610SW IP	333	02xxxx 333
FAX	FAX	331	02xxx 331
X	PSTN	--	02xxx 9040
Y	PSTN	--	02xxx 3025
Z	GSM	--	04xxxxxxx

Table 1: Endpoints Used for Testing

3. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office 500	5.0 (18)
Avaya ANALOG POTS8V2	7.0 (18)
Avaya 5610 IP Telephones	2.9.1
ACME Packet SD4250 SBC	6.1.0 M3P1

Table 2: Equipment and Software Validated

4. Configuration

All configuration steps for Avaya IP Office were performed using the IP Office Manager application. This application presents the administrator with a hierarchy of icons for the various components which can be configured, as shown below.

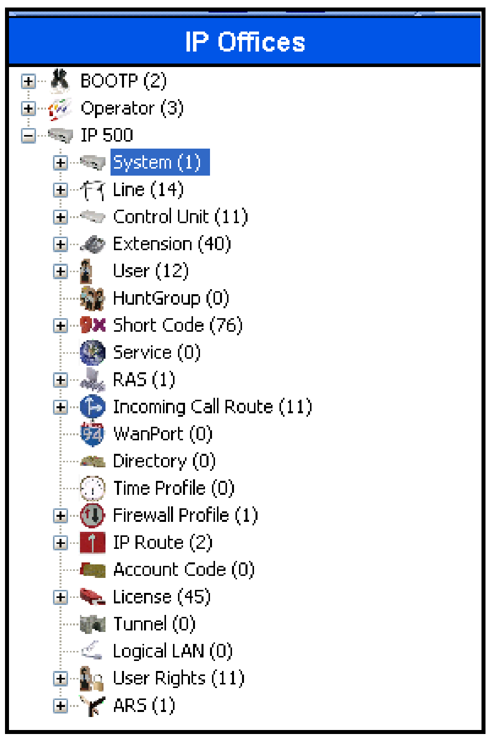

Figure 2: IP Office Manager Top Level Presentation

4.1. Licenses

A license is required for SIP Trunk Channels, which can be confirmed by selecting the "License" icon shown in **Figure 2**.

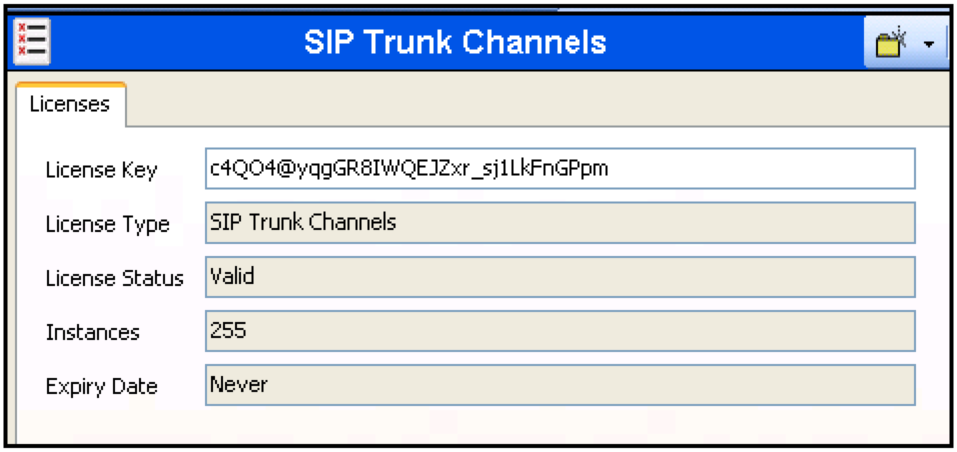

Figure 3: IP Office License for SIP Trunk Channels

4.2. System

Select the "System" icon shown in **Figure 2** and enter the parameters shown in the following table.

Tab	Parameter	Usage
System	Name	Enter a name to be assigned to IP Office for identification purposes.
	Locale	Select the locale for the installation from the drop-down menu.
LAN1	IP Address	Enter the IP address assigned to IP Office.
	IP Mask	Enter the network mask assigned to IP Office.
Telephony	Dial Delay Time	Enter the inter-digit dial delay time. A value of "5" seconds was used for the test.
	Dial Delay Count	Enter "0".
	Automatic Codec Preference	Select "G.711 ALAW 64K".

Table 3: IP Office System Parameters

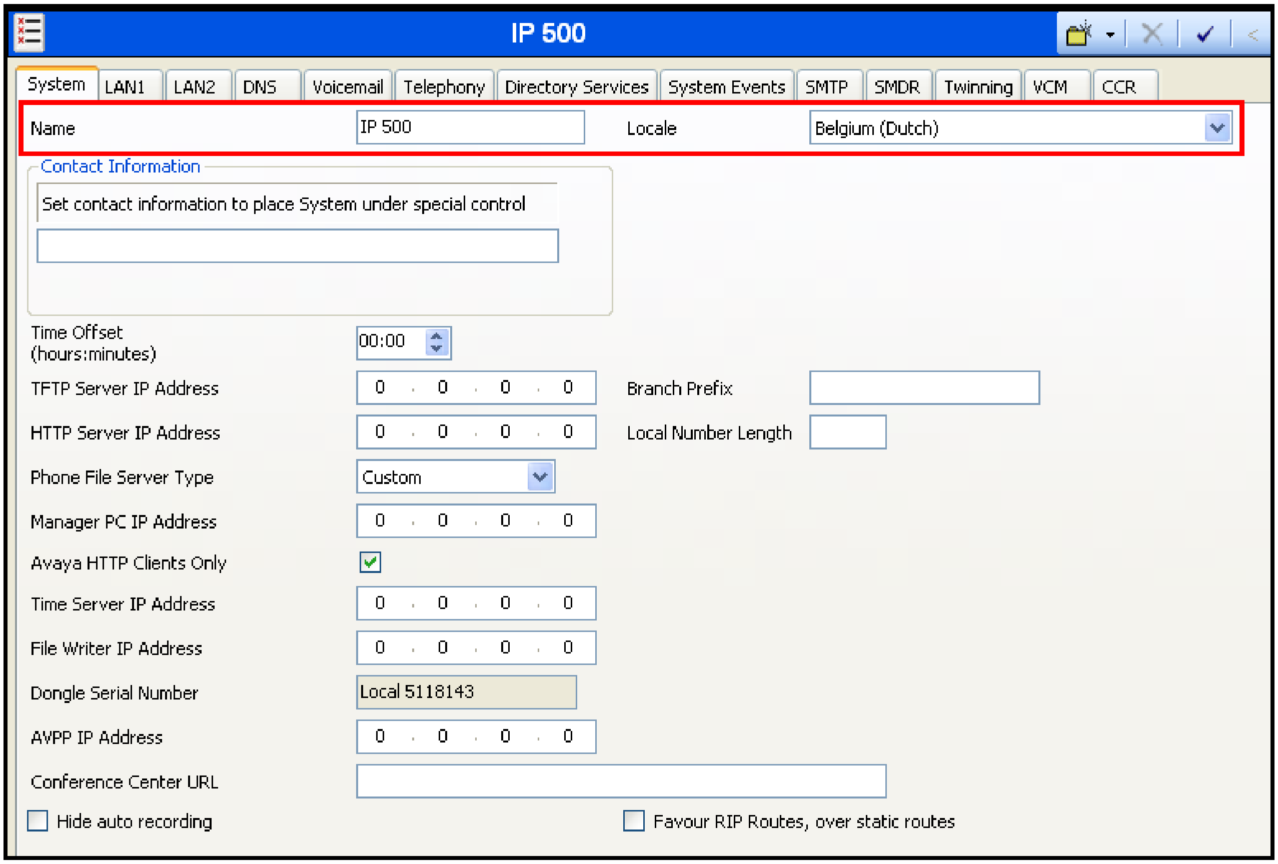

Figure 4: IP Office System: System Tab

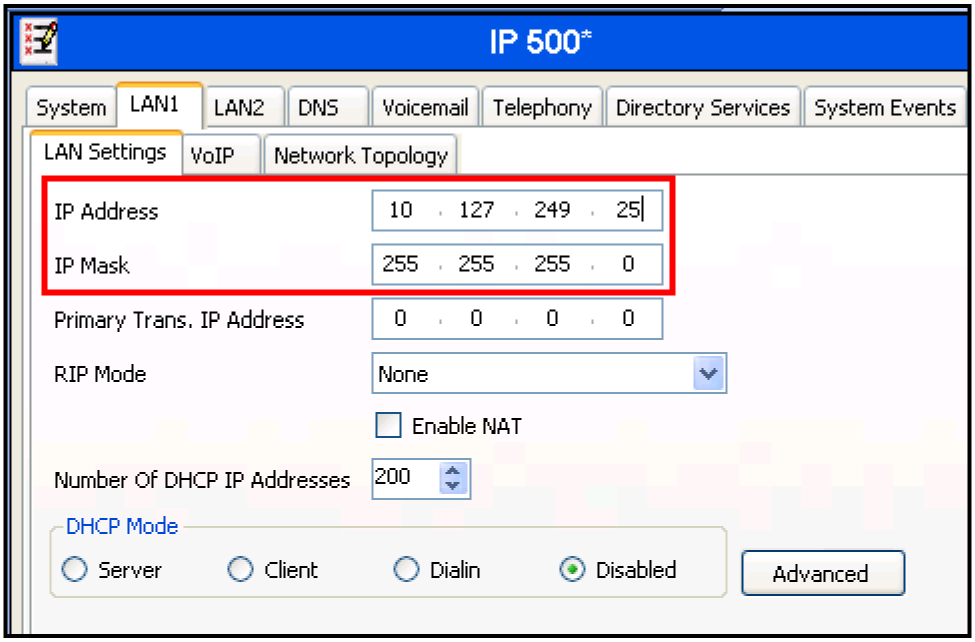

Figure 5: IP Office System: LAN Settings Tab

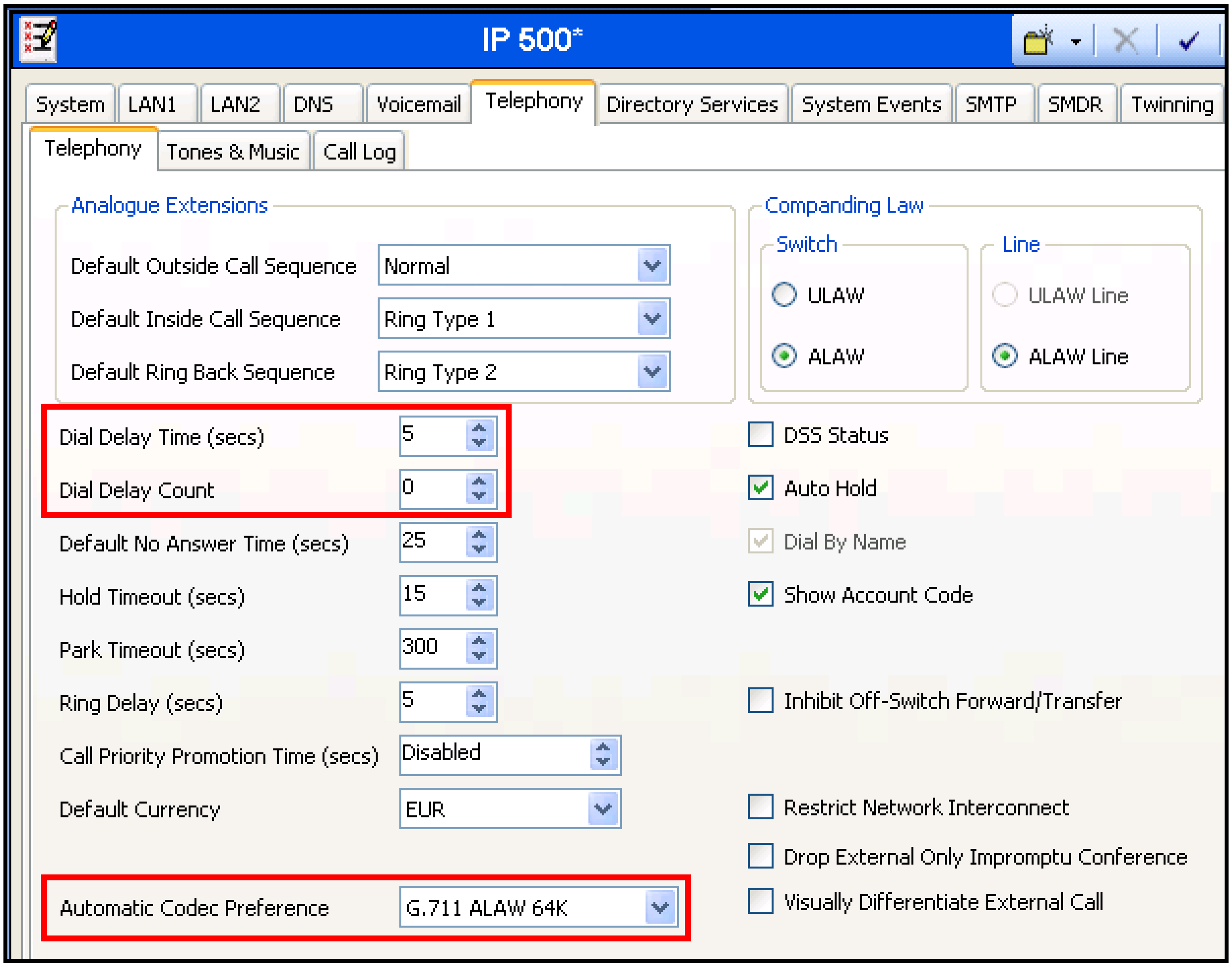

Figure 6: IP Office System: Telephony Tab

4.3. Default Gateway

Select the "IP-Route" icon shown in **Figure 2** and create a default route with the parameters shown in the following table.

Parameter	Usage
IP Address	Enter "0.0.0.0".
IP Mask	Enter "0.0.0.0".
Gateway IP Address	Enter the IP address of the default gateway used by IP Office.
Destination	Select "LAN1" from the drop-down list.

Table 4: IP Office Route: Default Gateway Parameters

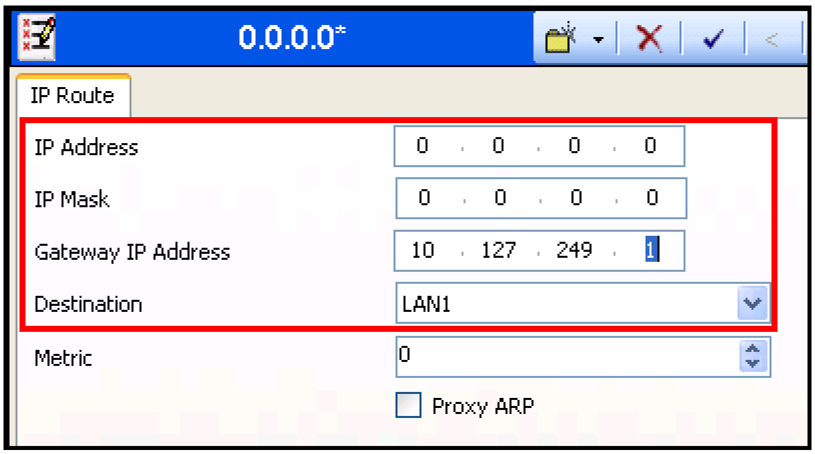

Figure 7: IP Office Route: Default Gateway

4.3.1. SIP Lines

Select the "Line" icon shown in **Figure 2** and create an entry for the SIP trunk with the parameters shown in the following table. Note that the "VoIP Silence Suppression" parameter is used to select the codec via the mechanism described below. Perform this for both the PSTN SIP line and the GSM SIP line using the appropriate the ITSP IP Address parameter for each.

Tab	Parameter	Usage
SIP Line	Line Number	This value is assigned automatically when the line is created. For the PSTN SIP line, the value for this configuration is 18. For the GSM SIP line, the value for this configuration is 19.
	ITSP Domain Name	Enter the domain name or IP address associated with the Belgacom Network
	ITSP IP Address	Enter the IP address of the Belgacom Network session border controller. For the PSTN SIP line, this value 10.127.249.230. For the GSM SIP line, this value 10.127.249.231 (see **Figure 1**).
SIP URI	Local URI	Select "Use User Data".
	Contact	Select "Use User Data".
	Display Name	Select "Use User Data".
	Incoming Group	Enter the number used as "Line Number" in **Figure 8**.
	Outgoing Group	Enter the same number selected for "Incoming Group".
VoIP	Compression Mode	Select G.711 ALAW followed by G.729(a) from the drop-down box.
	VoIP Silence Suppression	Uncheck this box.
	Fax Transport Support	Uncheck this box.
	Re-invite Supported	Check this box.

Table 5: IP Office SIP Line Parameters

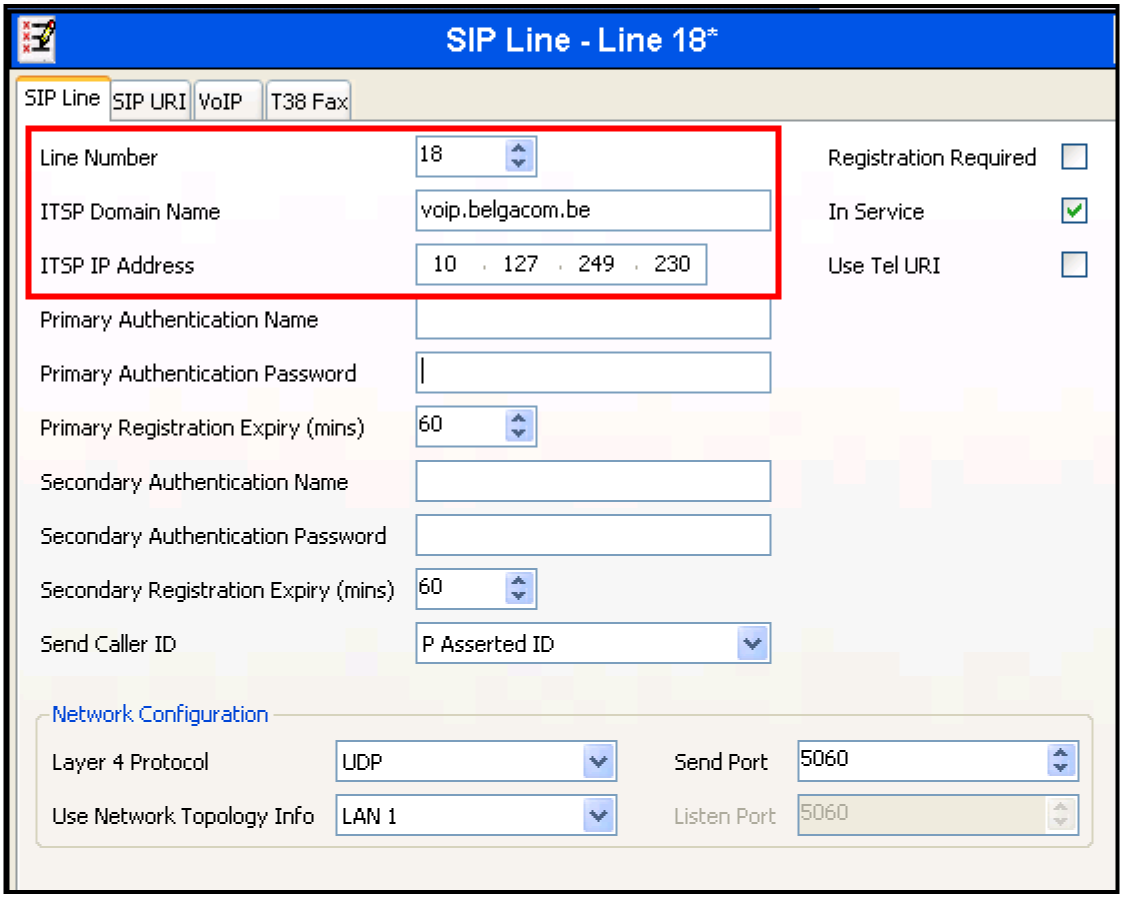

Figure 8: IP Office SIP Line: SIP Line Tab

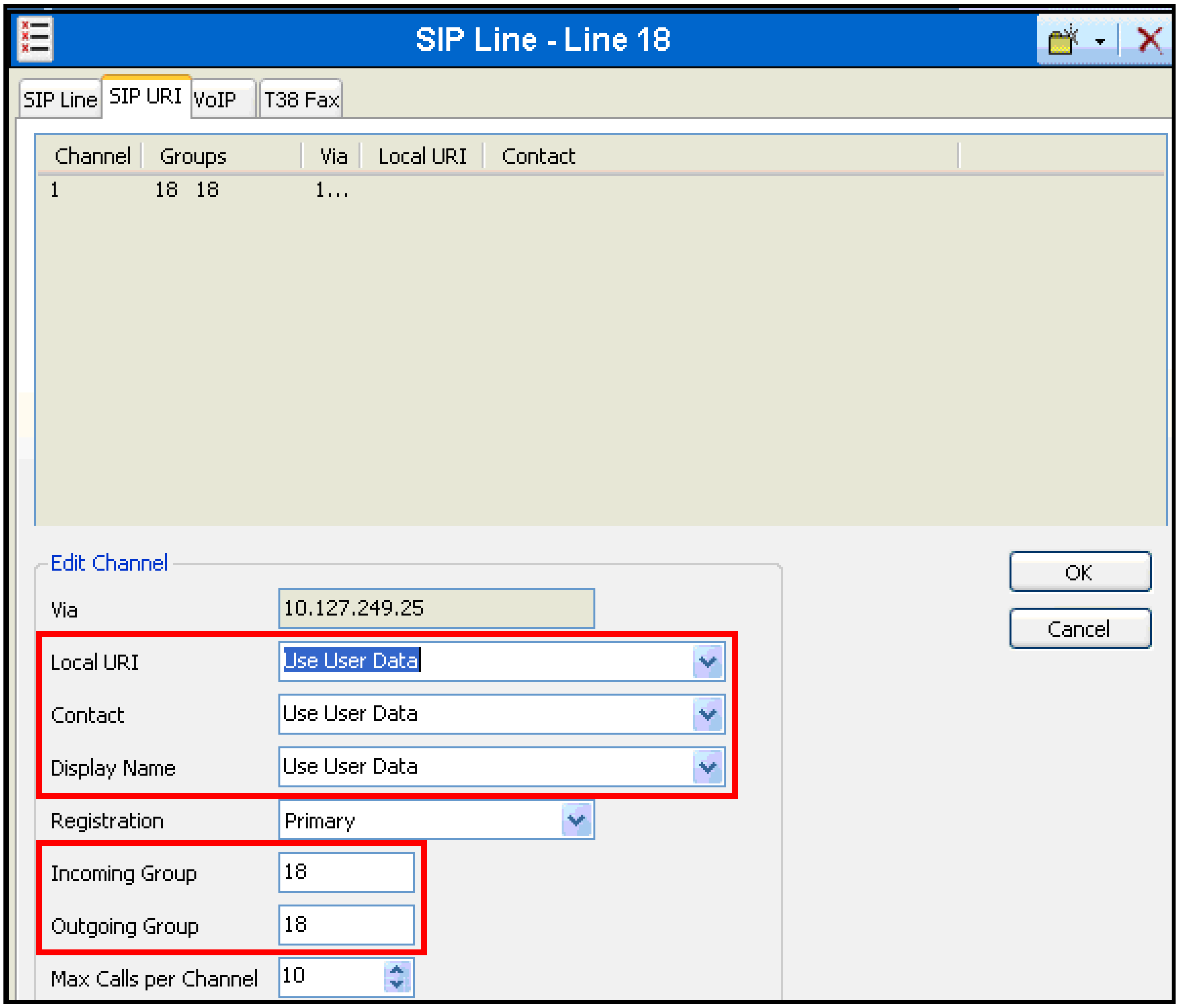

Figure 9: IP Office SIP Line: SIP URI Tab

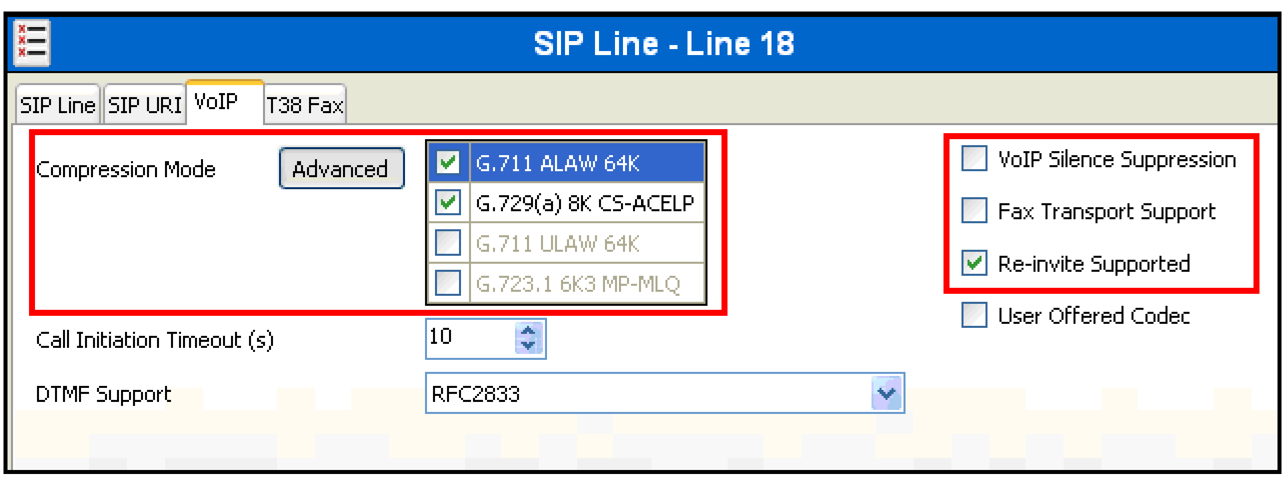

Figure 10: IP Office SIP Line: VoIP Tab

4.4. FAX

Select the "Extensions" icon shown in **Figure 2**, create an extension for the FAX device, and enter the extension in the "Base Extension" field.

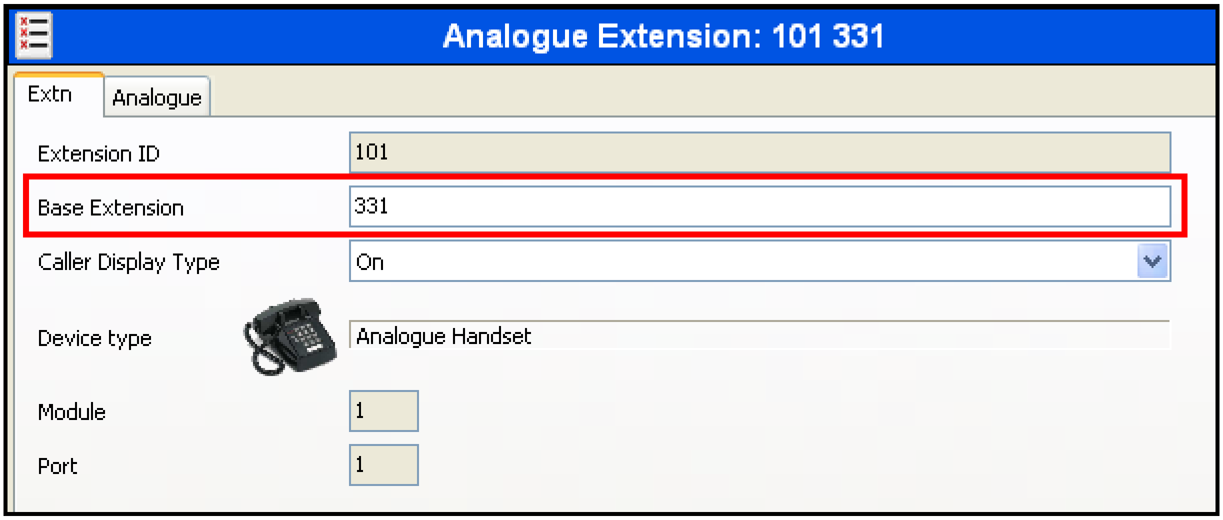

Figure 11: IP Office FAX Extension: Extn Tab

Select the "Users" icon shown in **Figure 2**, add a new user for the FAX device using the parameters shown in the following table.

Tab	Parameter	Usage
User	Name	Enter an appropriate name to be assigned to the FAX device.
	Extension	Enter the local extension to be assigned to the FAX device.
SIP	SIP Name	Enter the DID which is assigned to the FAX device.
	SIP Display Name	Enter an appropriate name to be assigned to the FAX device.
	Contact	Enter the DID which is assigned to the FAX device.

Table 6: IP Office FAX User Parameters

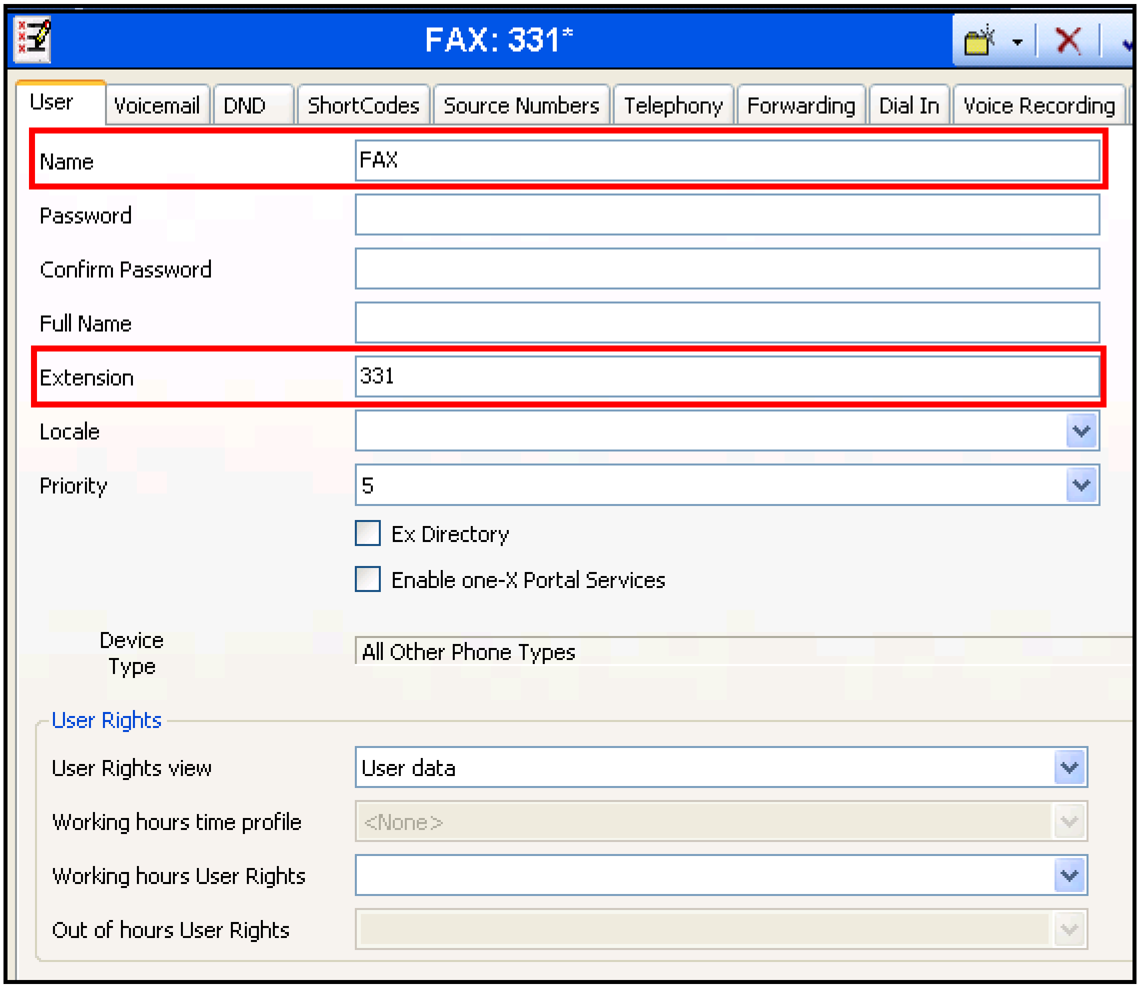

Figure 12: IP Office FAX User: User Tab

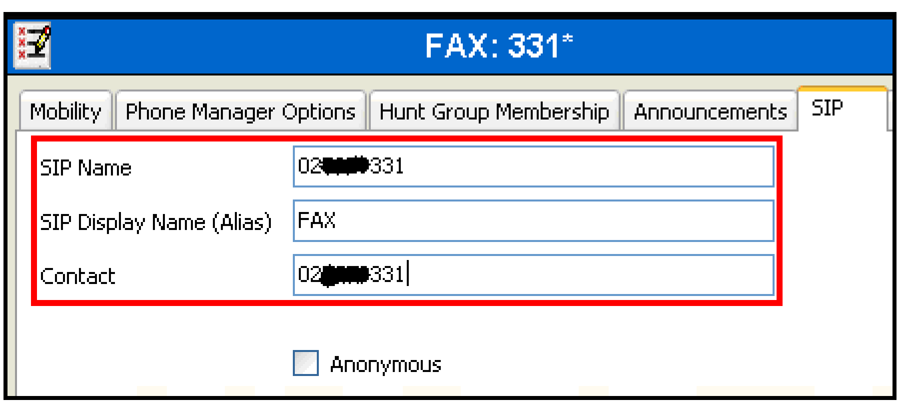

Figure 13: IP Office FAX User: SIP Tab

4.5. Local Telephones

Select the “Extensions” icon shown in **Figure 2**, create an extension for the local telephone, and enter the extension in the “Base Extension” field. Repeat this for each of the extensions shown in **Table 1.**

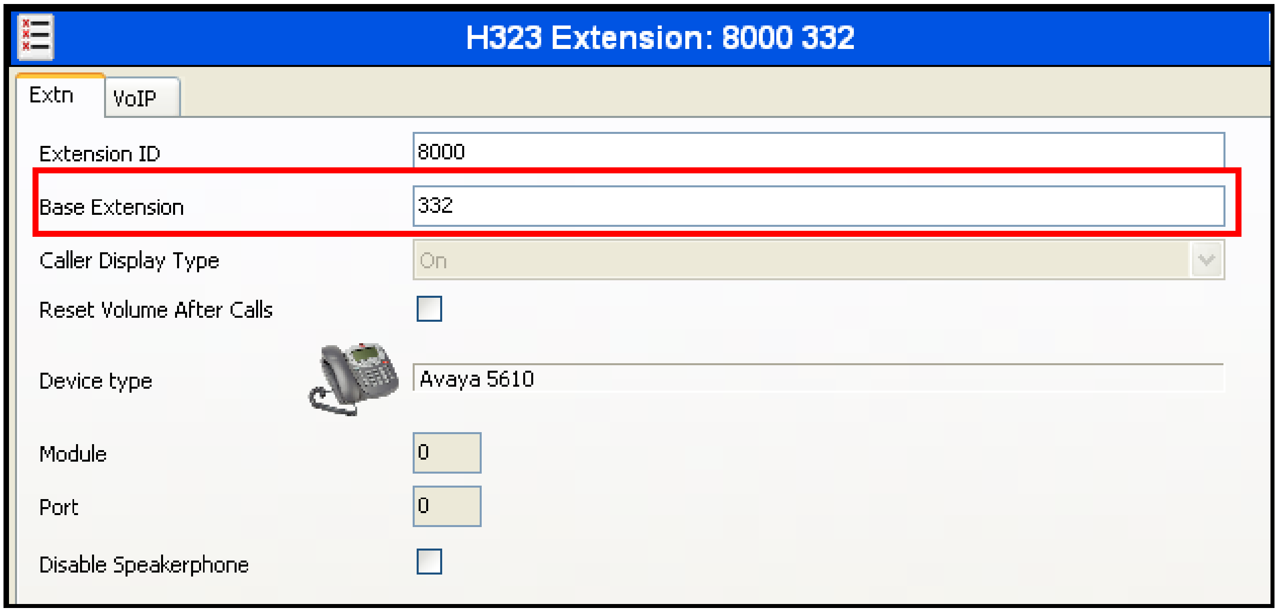

Figure 14: IP Office Local Telephone Extension: Extn Tab

From the “Users” tab of the “Users” icon shown in **Figure 2**, add a new user for each of the local telephones shown in **Figure 1**, using the parameters shown in the following table.

Tab	Parameter	Usage
User	Name	Enter an appropriate name to be assigned to the user.
	Extension	Enter the local extension to be assigned to the user.
SIP	SIP Name	Enter the DID which is assigned to the user.
	SIP Display Name	Enter an appropriate name to be assigned to the user.
	Contact	Enter the DID which is assigned to the user.

Table 7: IP Office Extension User Parameters

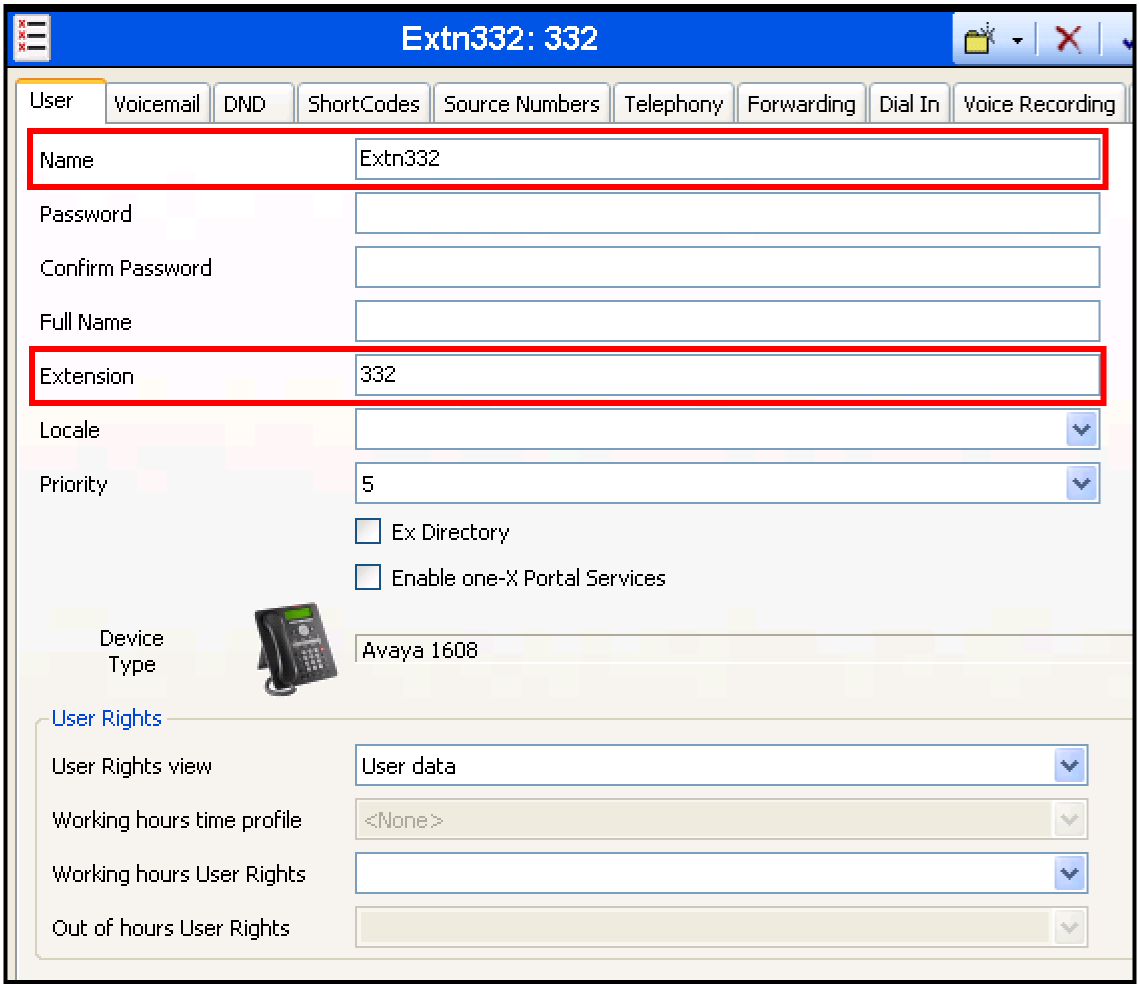

Figure 15: IP Office Local Telephone User: User Tab

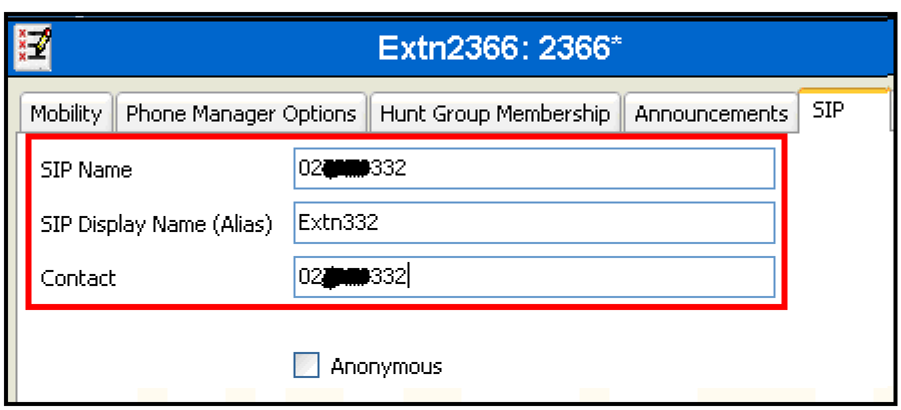

Figure 16: IP Office Local Telephone User: SIP Tab

4.6. Call Routing

Select the "Short Code" icon shown in **Figure 2** to create a shortcode to route outgoing calls from Avaya IP Office to the Belgacom Network. Create a new shortcode with the values shown in the following table.

Parameter	Usage
Code	Enter 0N;
Feature	Select "Dial" from the drop-down menu.
Telephone Number	Enter 0N"@<addr>", where <addr> is IP address of Session Border Controller, as shown in **Figure 1**.
Line Group Id	Enter the line group number assigned to the SIP Line "Outgoing Group" in **Figure 9**.

Table 8: IP Office Outgoing PSTN Call Shortcode Parameters

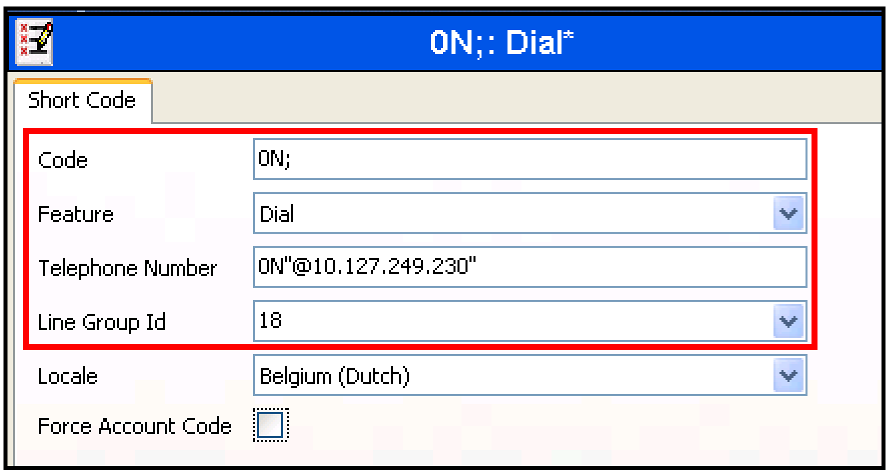

Figure 17: IP Office Outgoing PSTN Call Shortcode

Repeat this procedure to create a shortcode to route outgoing calls from Avaya IP Office to the GSM Network. Create a new shortcode with the values shown in the following table.

Parameter	Usage
Code	Enter 04N; to route calls to all numbers beginning with "04" to the GSM network.
Feature	Select "Dial" from the drop-down menu.
Telephone Number	Enter 0N"@<addr>", where <addr> is IP address of the GSM Session Border Controller, as shown in **Figure 1**.
Line Group Id	Enter the line group number assigned to the SIP Line "Outgoing Group" in **Figure 9**.

Table 9: Table: IP Office Outgoing GSM Call Shortcode Parameters

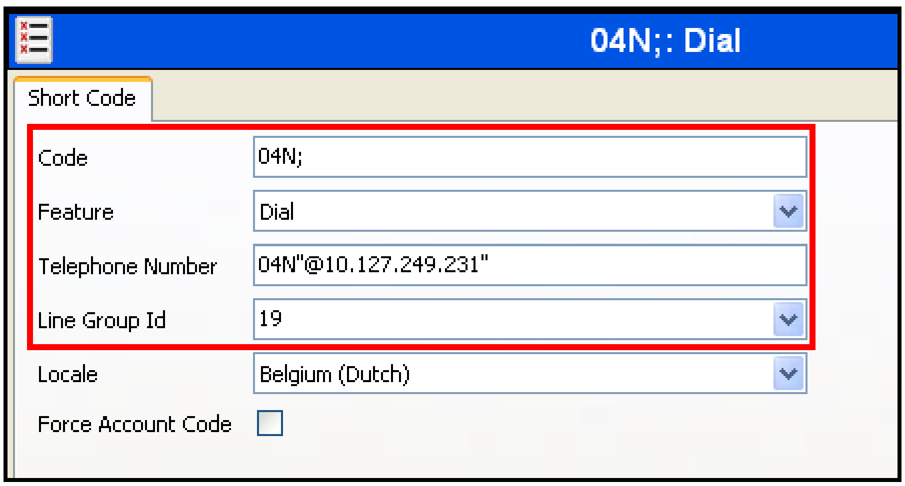

Figure 18: IP Office Outgoing GSM Call Shortcode

Select the "Incoming Call Route" icon shown in **Figure 2**, create a new incoming call route with the values shown in the following table for each of the extension shown in **Figure 1**. Perform this for both the PSTN and GSM trunk.

Tab	Parameter	Usage
Standard	Line Group Id	Enter the Line Group Id assigned to the SIP trunk in **Figure 9**. Perform this for both the PSTN and GSM trunk.
	Incoming Number	Enter the prefix of the number range assigned to the IPO, followed by "xxx" for the extension.
Destinations	Destination	Enter a "#" to designate that the extension occupied by the "xxx" sequence from the "Incoming Number" field should be used as the destination.

Table 10: IP Office Incoming Call Route Parameters

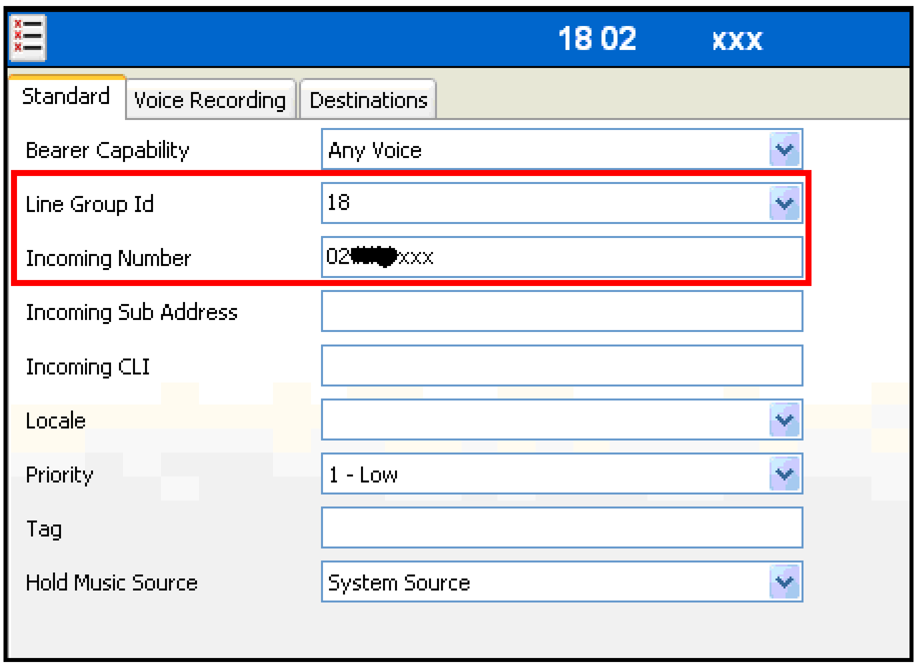

Figure 19: IP Office Incoming Call Route: Standard Tab

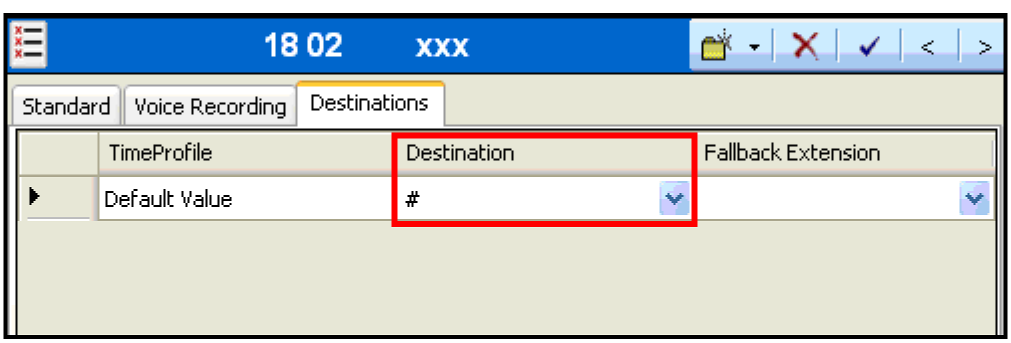

Figure 20: IP Office Incoming Call Route: Destinations Tab

5. General Test Approach and Test Results

The tests listed in **Section 1.1** were performed manually. For each of the tests, correct operation of the endpoints was verified via inspection, and a SIP protocol trace was generated to confirm the expected exchange of SIP protocol messages.

The following issues were encountered during testing:

- If a PSTN endpoint calls an IP Office extension while IP Office is disconnected from network, the caller receives no feedback that there is a problem, and the call is cleared after several minutes.

6. Verification Steps

The correct configuration of the system can be verified by performing the following steps:

- Verify that the local extensions on Avaya IP Office can call and talk to each other.
- Verify that the local extensions on Avaya IP Office and the telephones attached to the PSTN can call each other.
- Verify that it is possible to send FAX messages between the locally attached FAX device and the FAX unit which is attached to the PSTN.

7. Conclusion

These Application Notes contain instructions for configuring Avaya IP Office to connect to the Belgacom SIP network via a SIP trunk. All test cases passed with exception noted in **Section 5**.

8. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *IP Office Installation*, August 2009, Document Number 15-601042.
[2] *IP Office 5.0 Manager,* August 2009, Issue 7, Document Number 15-601011

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[3] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[4] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

Appendix A: Sample SIP INVITE Messages

Below is a sample SIP INVITE message received from the Belgacom network for an incoming call:

```
    Request-Line: INVITE sip:02xxxx333@10.127.130.25:5060 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 26.xx.xx.71:5060;branch=z9hG4bK.iIiIiI.1a01fb36.d819a9cd
        To:  <sip:02xxxx333@voip.belgacom.be;user=phone>
        From: <sip:02xxxx9040@voip.belgacom.be;user=phone>;tag=ca4213d6
        Call-ID: 8b1c007e85a033c-0002-01ff-0000-0000@26.1.251.54
        CSeq:  1 INVITE
        Max-Forwards: 70
        Contact: <sip:02xxxx9040.iIiIiI.1a01fb36.@26.1.251.71>
        Date: Mon, 1 Mar 2010 13:41:11 GMT
        Allow: INVITE, ACK, PRACK, CANCEL, BYE, OPTIONS, MESSAGE, NOTIFY, UPDATE, REGISTER, INFO,
REFER, SUBSCRIBE
        P-Asserted-Identity:  <sip:02xxxx9040@voip.belgacom.be>
        Accept: application/sdp, application/isup, application/xml, application/dtmf-relay
        Content-Type: application/sdp
        Content-Length: 198
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 0 236524 IN IP4 26.xx.xx.38
            Session Name (s): IMSS
            Connection Information (c): IN IP4 26.xx.xx.9
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 26.1.251.9
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 49220 RTP/AVP 8 0 101
                Media Type: audio
                Media Port: 49220
                Media Proto: RTP/AVP
                Media Format: ITU-T G.711 PCMA
                Media Format: ITU-T G.711 PCMU
                Media Format: 101
            Media Attribute (a): rtpmap:101 telephone-event/8000
                Media Attribute Fieldname: rtpmap
                Media Format: 101
                MIME Type: telephone-event
            Media Attribute (a): fmtp:101 0-15
                Media Attribute Fieldname: fmtp
                Media Format: 101 [telephone-event]
                Media format specific parameters: 0-15
            Media Attribute (a): X-sqn: 0
                Media Attribute Fieldname: X-sqn
                Media Attribute Value:  0
            Media Attribute (a): X-cap:  1 image udptl t38
                Media Attribute Fieldname: X-cap
                Media Attribute Value:   1 image udptl t38
```

Below is a sample SIP INVITE message received from the Belgacom network for an outgoing call:

```
    Request-Line: INVITE sip:02xxxx9040@10.127.249.230 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 10.127.249.25:5060;rport;branch=z9hG4bK16f190cc5b0c16c215e31ad7f3ff0a2c
        From: "Extn333" <sip:02xxxx333@belgacom.voip.be>;tag=44e3657f733c8101
        To: <sip:02xxxx9040@10.127.249.230>
        Call-ID: 6919a79d5a1cdd1f4033a45ae7e4fac5@10.127.249.25
        CSeq: 1349782035 INVITE
        Contact: "Extn333" <sip:02xxxx333@10.127.249.25:5060;transport=udp>
        Max-Forwards: 70
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO
        Content-Type: application/sdp
        Content-Length: 256
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): UserA 3213829575 4229209049 IN IP4 10.127.249.25
            Session Name (s): Session SDP
            Connection Information (c): IN IP4 10.127.249.25
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 49154 RTP/AVP 8 0 18 101
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:0 PCMU/8000
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): rtpmap:101 telephone-event/8000
            Media Attribute (a): fmtp:101 0-15
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.