Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the COLT VoIP Access 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 COLT VoIP Access 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 COLT VoIP Access SIP trunking network and 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. A TCP link was used for communication between Avaya IP Office and COLT VoIP Access SIP trunking network for the tests which were performed, however a UDP link can be used as well.

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 COLT. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

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.
- Supervised Call Transfer / Blind Call Transfer to local extension, PSTN, and GSM endpoints.
- Conference Call with local, GSM, and PSTN endpoints, also with mixed codecs.
- Short and long Fax Send / Receive using T.38, using both the G.711 and G.729 codecs.
- Simultaneous Calls.
- Long Calls.
- Calls with both ends muted.
- Recovery from both trunk and phone connection failure.

1.2. Support

Telephone support is available on a national basis as shown in the following table, which shows the hotline number for each country where support is available, as well as a toll-free number if available.

Country	Hot Line	Toll-Free
Austria	(+43) 1 20 500 500	0800 880 990
Belgium	(+32) 2 790 16 29	0800 50701
Switzerland	(+41) 44 560 0720	0800 560 560
Denmark	(+45) 70 27 35 59	
France	(+33) 1 70995600	0800 948 888
Germany	(+49) 69 56606 3115	0800 855 4444
Ireland	(+34) 9355 02568	1800 944040
Italy	(+39) 0230 329 550	0800 909 377
Netherlands	(+31) 20 888 2433	0800 265 8023
Portugal	(+351) 211 200 222	808 780 222
Spain	(+34) 913 206018	901 888400
Sweden	(+46) 8781 8333	
UK	(+44) 203 140 2023	0800 136 166

2. Reference Configuration

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

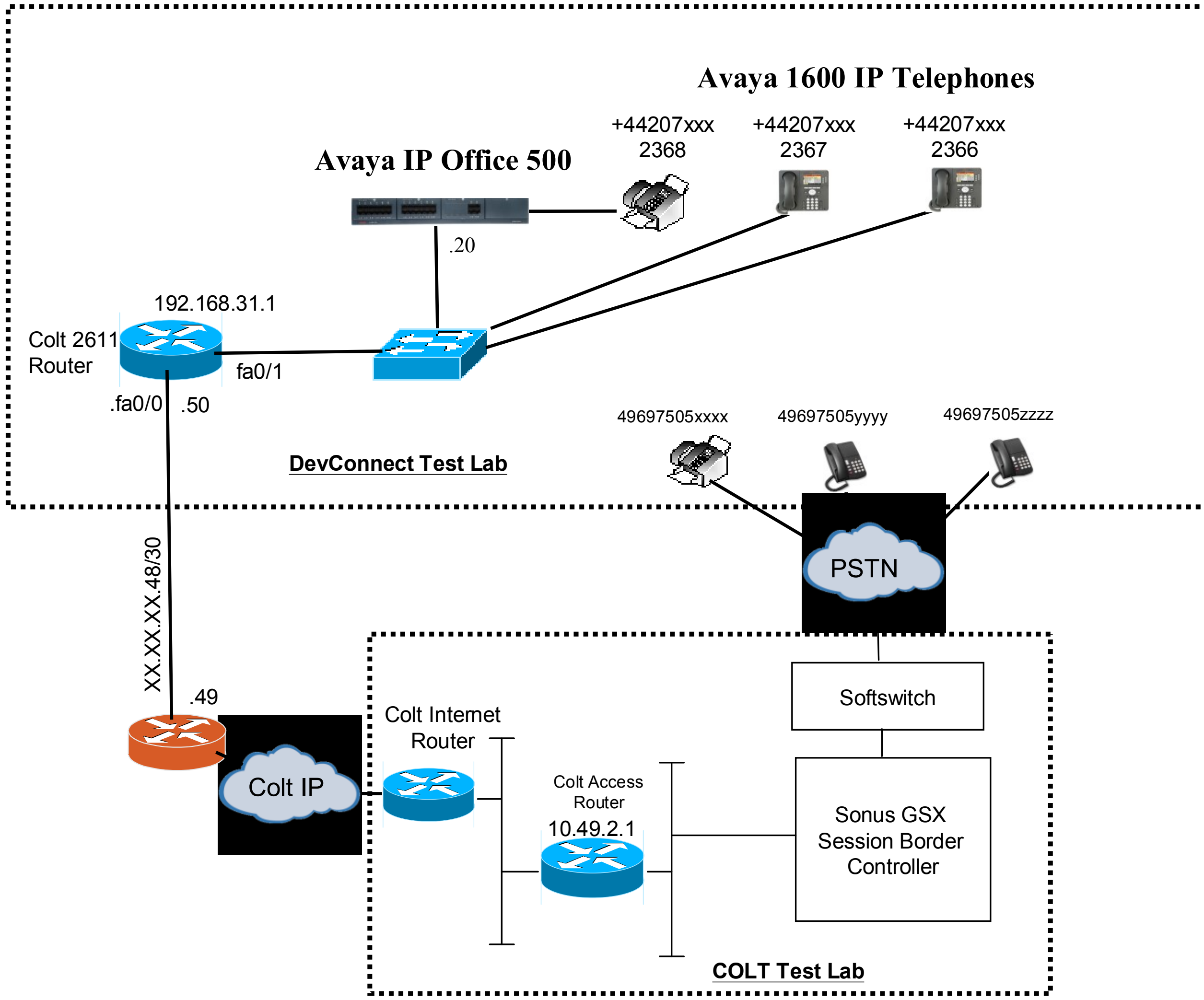

Figure 1: Reference Configuration

In the above diagram, Avaya H323 IP Telephones are registered with the Avaya IP Office.

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 (15)
Avaya ANALOG POTS8V2	7.0 (15)
Avaya 1600 IP Telephones	2100
Sonus GSX Session Border Controller	7.02.R004

Table 1: Equipment and Software Validated

4. Configuration

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

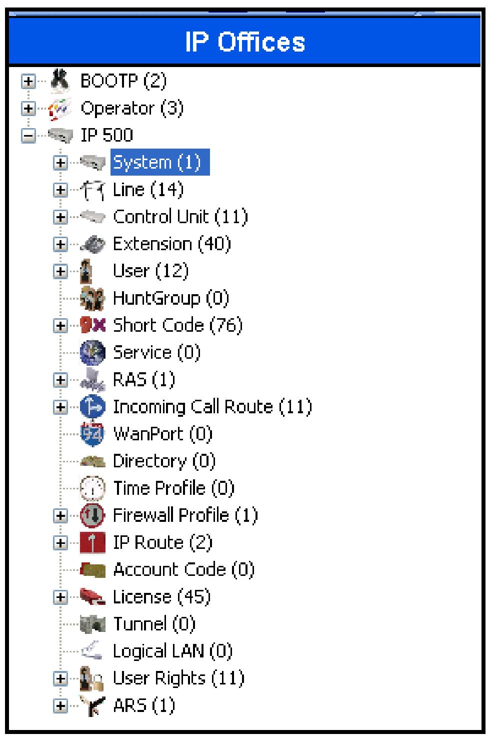

Figure 2: IP Office Manager Top Level Presentation

4.1. Licenses

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

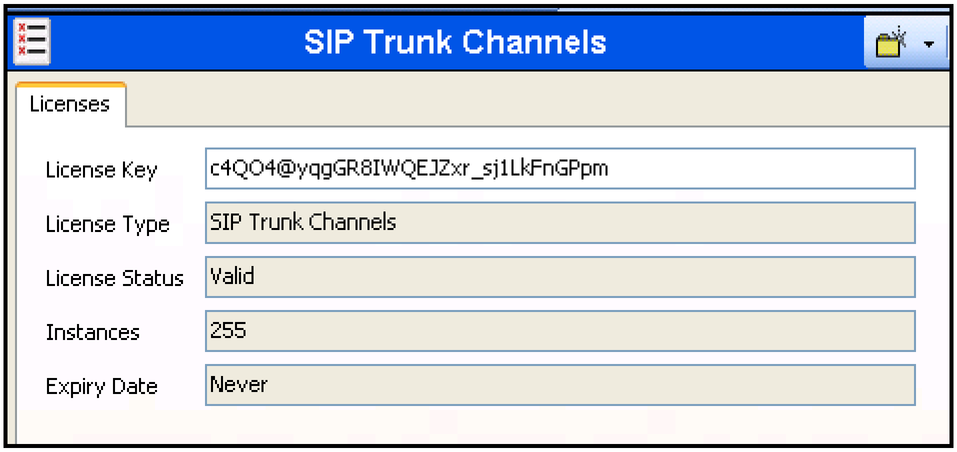

Figure 3: IP Office License for SIP Trunk Channels

4.2. System

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

Tab	Parameter	Usage
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.729(a) 8K CS-ACELP".

Table 2: System-Parameters Features Parameters

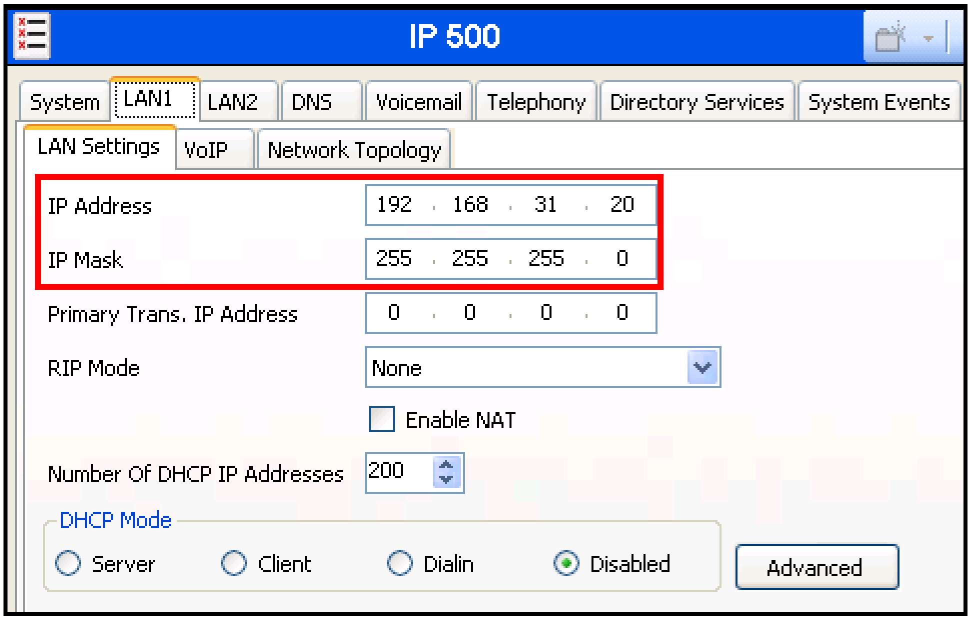

Figure 4: IP Office System: LAN Settings Tab

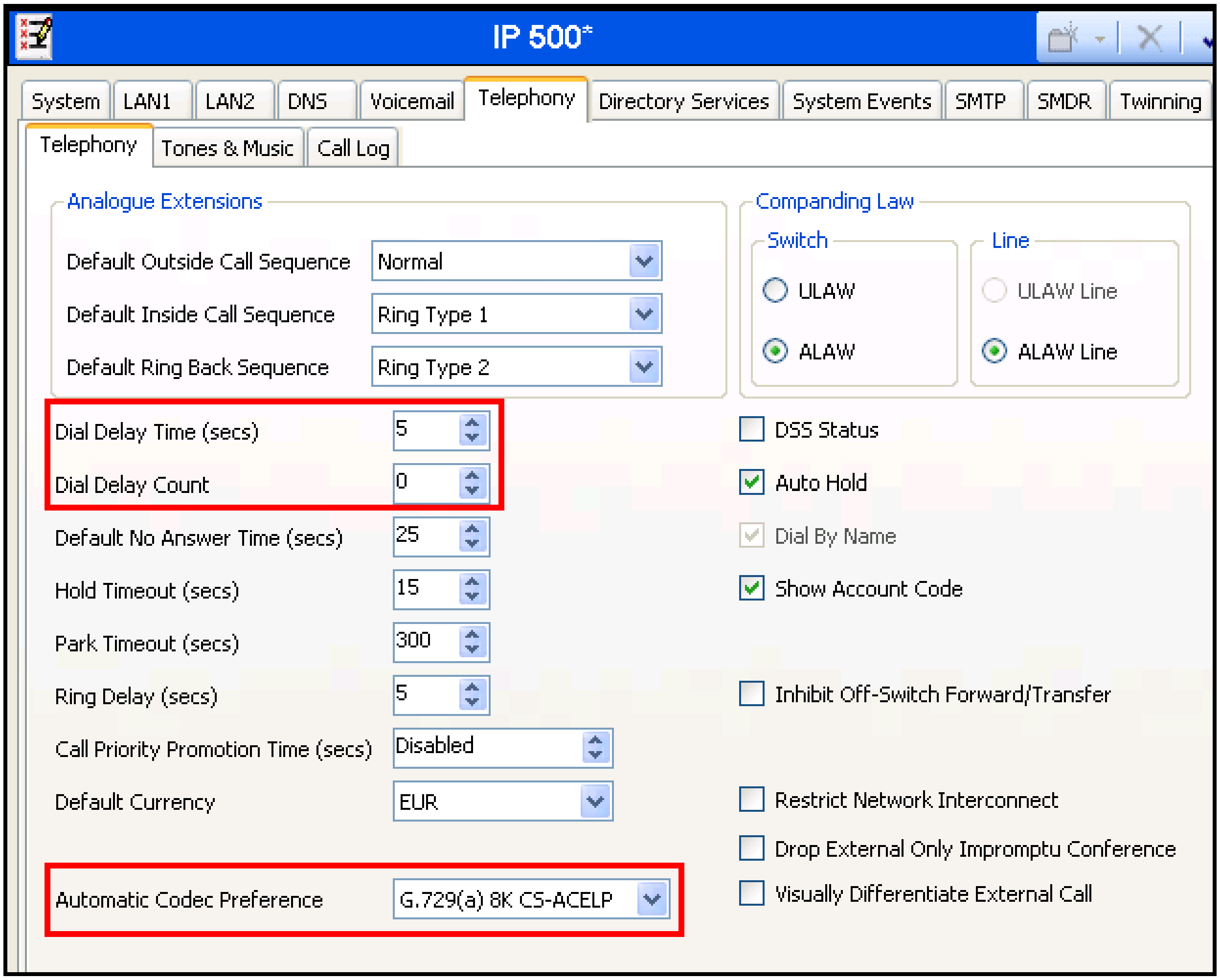

Figure 5: IP Office System: Telephony Tab

4.3. Default Gateway

Select the "IP-Route" icon shown in **Figure 2** and create a 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 address of the router which is used to attach IP Office to the COLT VoIP Network.
Destination	Select "LAN1" from the drop-down list.

Table 3: System-Parameters Features Parameters

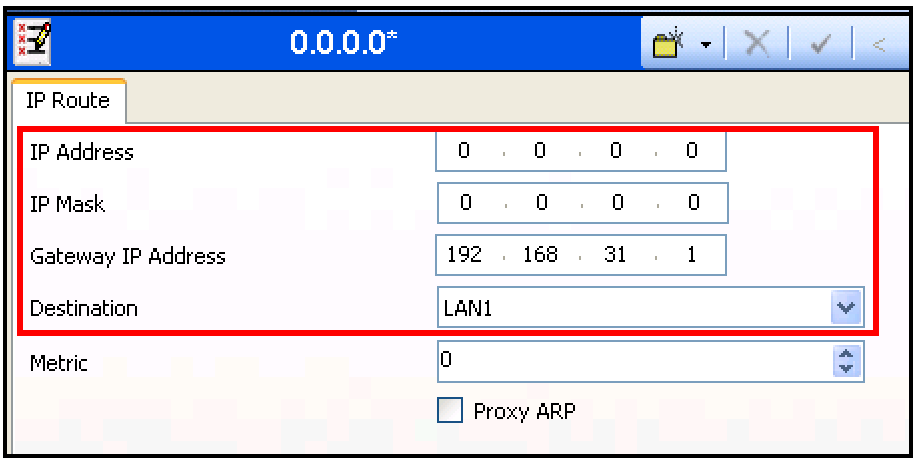

Figure 6: IP Office Route: Default Gateway

4.4. SIP Line

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.

- If the “VoIP Silence Suppression” box in **Figure 9** is checked, the IP Office selects the G.729 codec with “annex b” set to the default value “on” for outgoing calls. This is not supported by the COLT VoIP Network, which then accepts the call using its secondary codec, G.711A.
- If the “VoIP Silence Suppression” box in **Figure 9** is unchecked, the IP Office selects the G.729 codec with “annex b” set to “off” for outgoing calls. This is supported by the COLT VoIP Network, which then accepts the call using its primary codec, G.729.

Tab	Parameter	Usage
SIP Line	ITSP Domain Name	Enter the domain name or IP address to be used to identify the COLT VoIP Network
	ITSP IP Address	Enter the IP address of COLT VoIP Network session border controller.
SIP URI	Incoming Group	Enter the number of an otherwise unassigned incoming group.
	Outgoing Group	Enter the number of an otherwise unassigned outgoing group.
VoIP	Compression Mode	Select G.729(a) followed by G.711 ALAW from the drop-down box.
	VoIP Silence Suppression	Check this box.
	Fax Transport Support	Check this box.
	Re-invite Supported	Check this box.

Table 4: System-Parameters Features Parameters

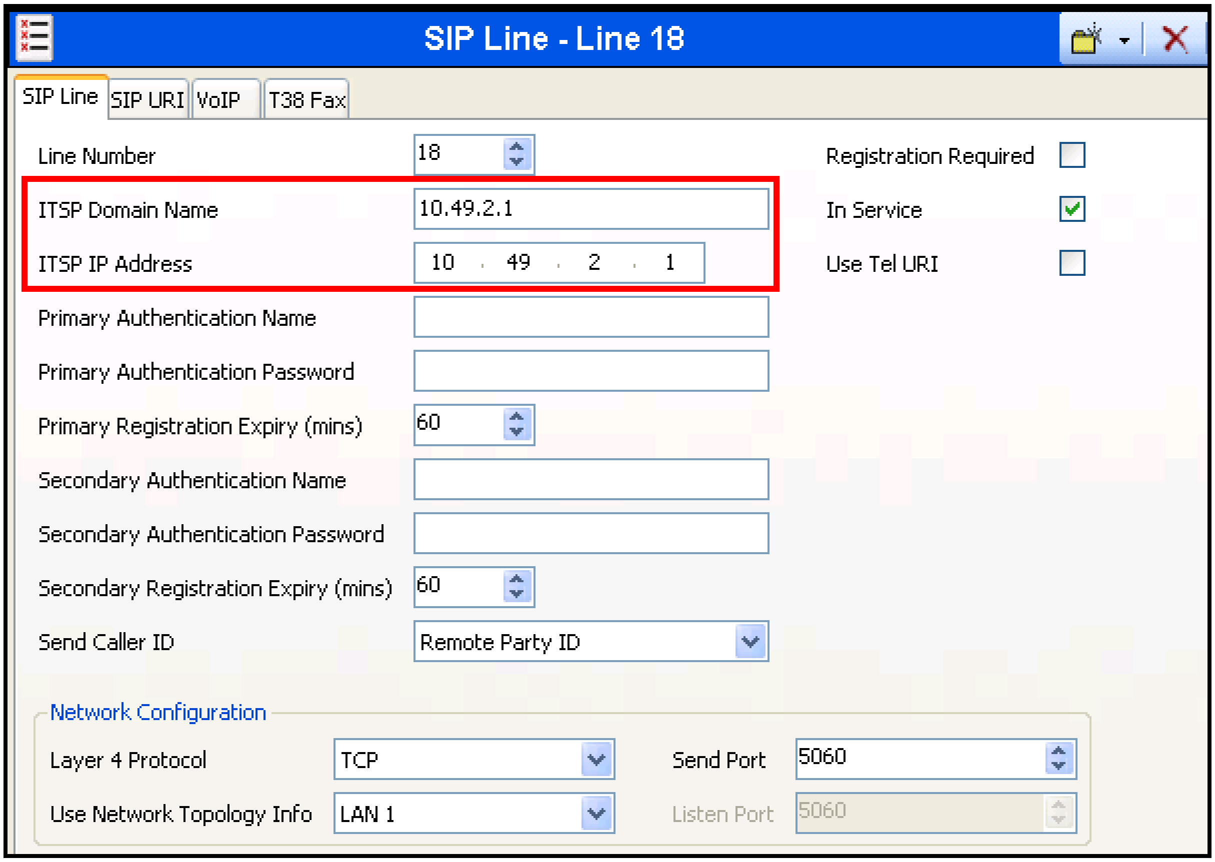

Figure 7: IP Office SIP Line: SIP Line Tab

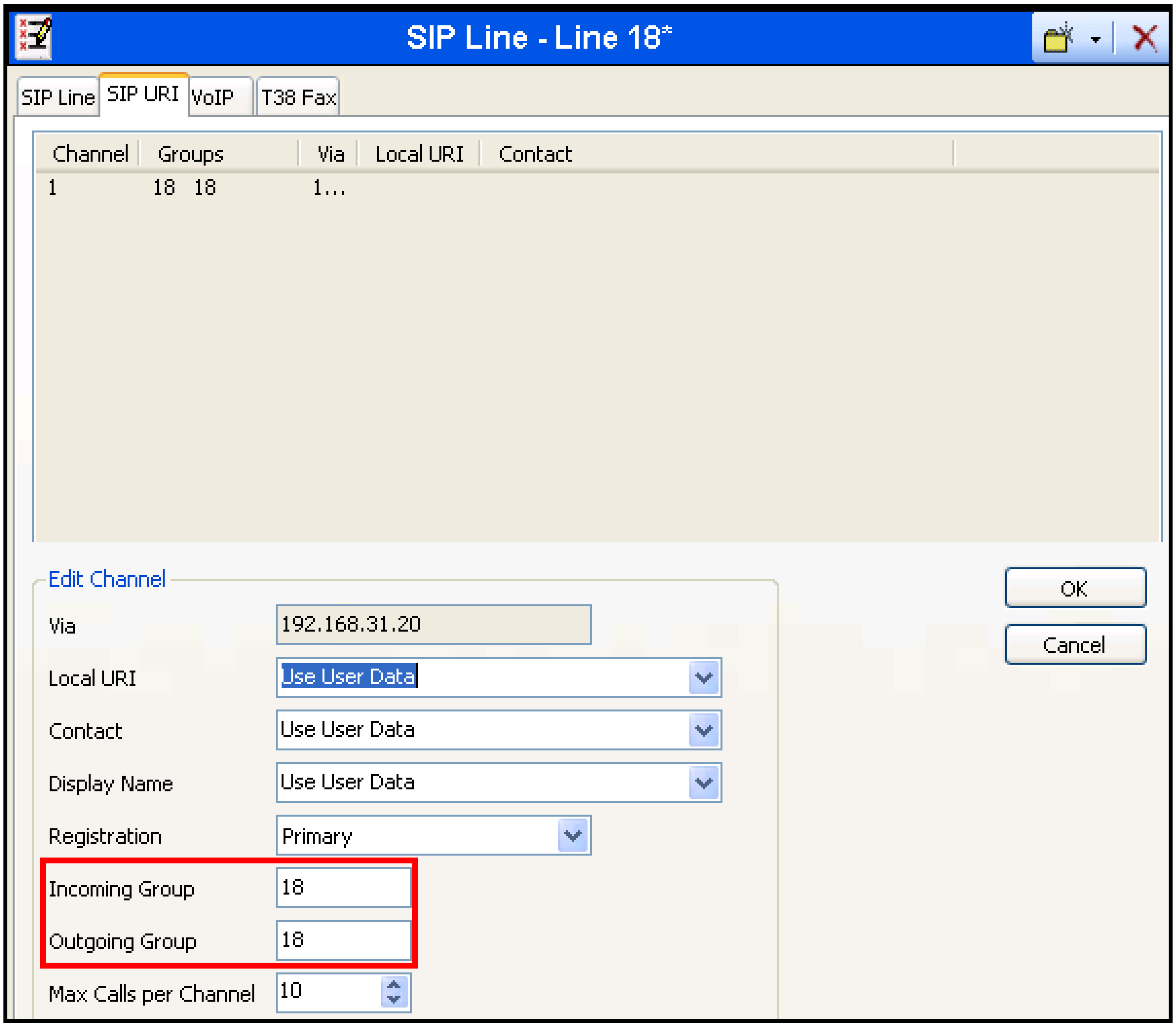

Figure 8: IP Office SIP Line: SIP URI Tab

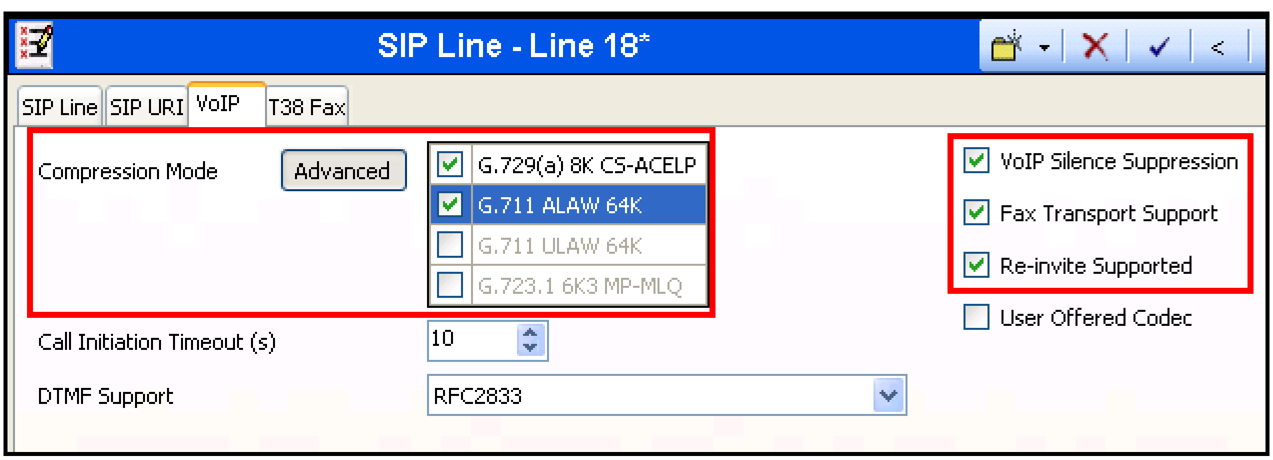

Figure 9: IP Office SIP Line: VoIP Tab

With the "Users" icon shown in **Figure 2** selected for the user with the name "NoUser", select the "Source Numbers" tab and add the item "FORCE_ANNEXB_NO_SPACES", so that IP Office sends the SDP payload attribute "annexb=on" without spaces on either side of the "=" character, as required by the COLT VoIP Network.

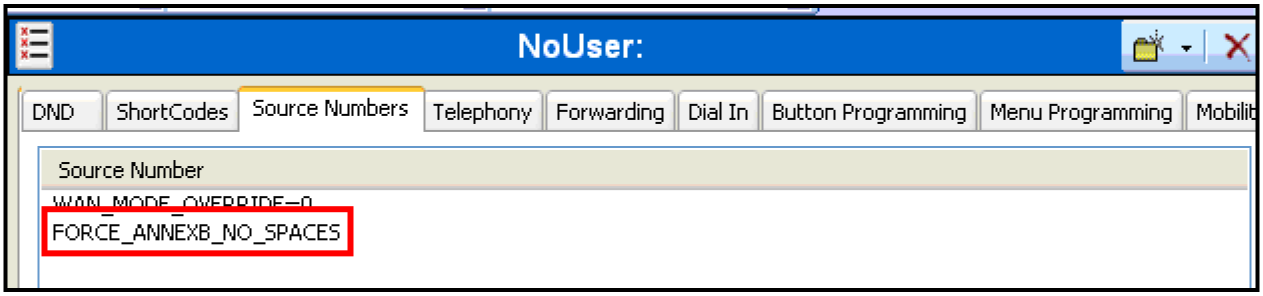

Figure 10: IP Office NoUser: Source Numbers Tab

4.5. FAX

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

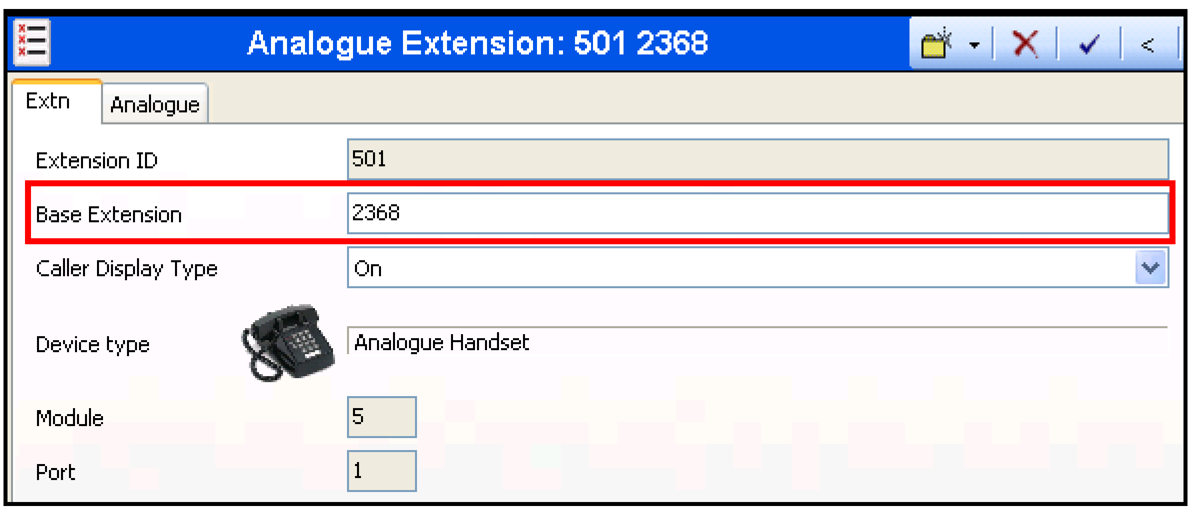

Figure 11: IP Office FAX Extension: Extn Tab

From the "Users" tab 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 5: System-Parameters Features Parameters

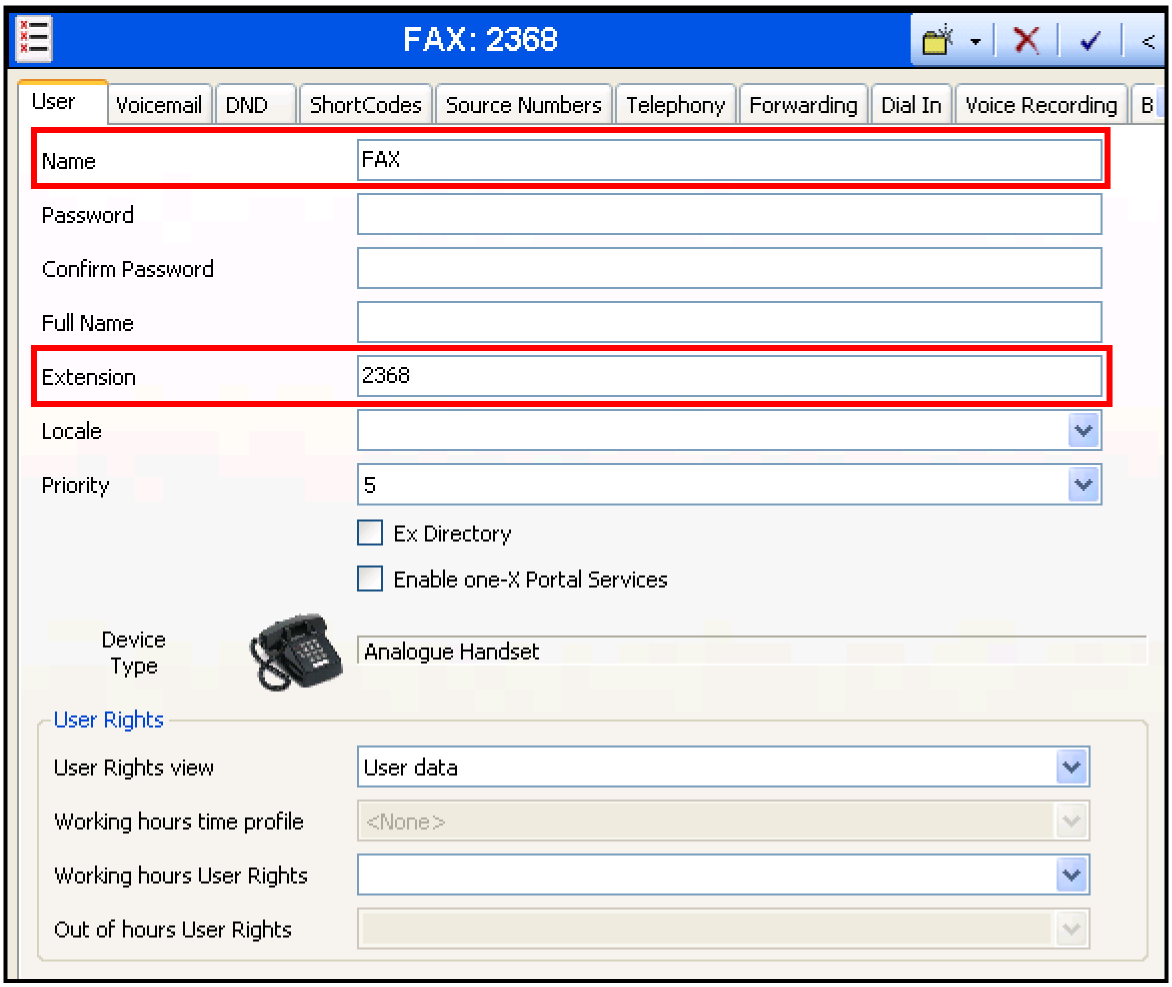

Figure 12: IP Office FAX User: User Tab

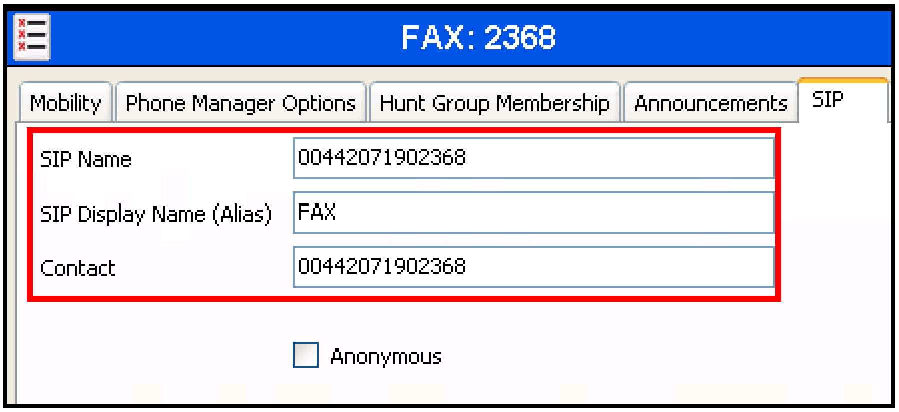

Figure 13: IP Office FAX User: SIP Tab

4.6. Local Telephone

From 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 **Figure 1.**

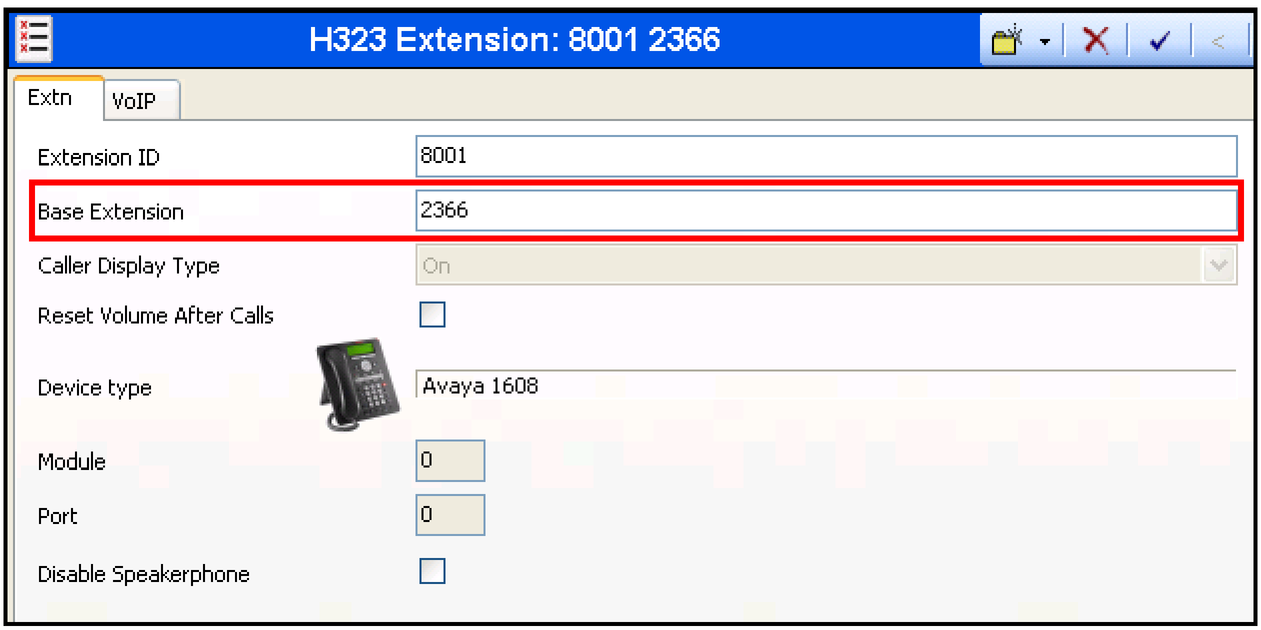

Figure 14: IP Office Local Telephone Extension: Extn Tab

From the “Users” tab of the “Users” icon shown in **Figure 2**, add a new user for each of the local telephones show 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 6: System-Parameters Features Parameters

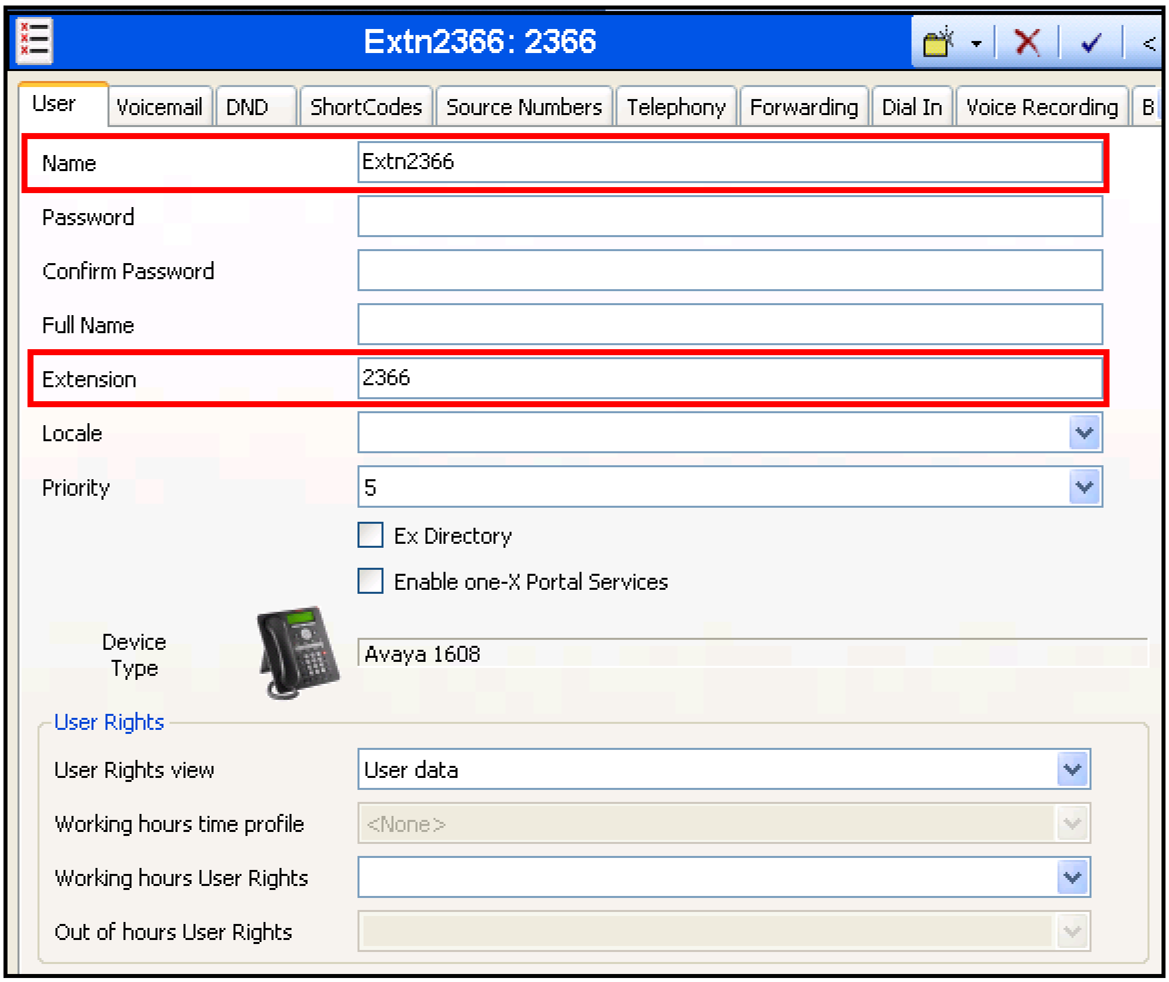

Figure 15: IP Office Local Telephone User: User Tab

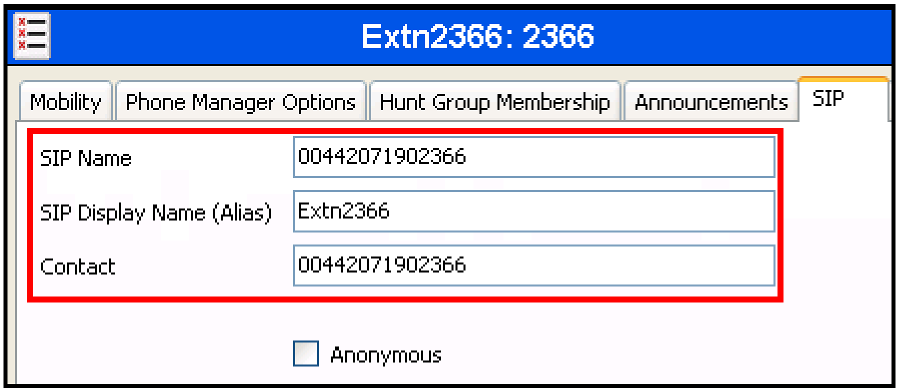

Figure 16: IP Office Local Telephone User: SIP Tab

4.7. Call Routing

From the “Short Code” icon shown in **Figure 2** create a shortcode to route outgoing calls from the Avaya IP Office to the COLT VoIP Network. From the “Shortcode” icon, 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”@<adr>”, where <adr> 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 8**.

Table 7: System-Parameters Features Parameters

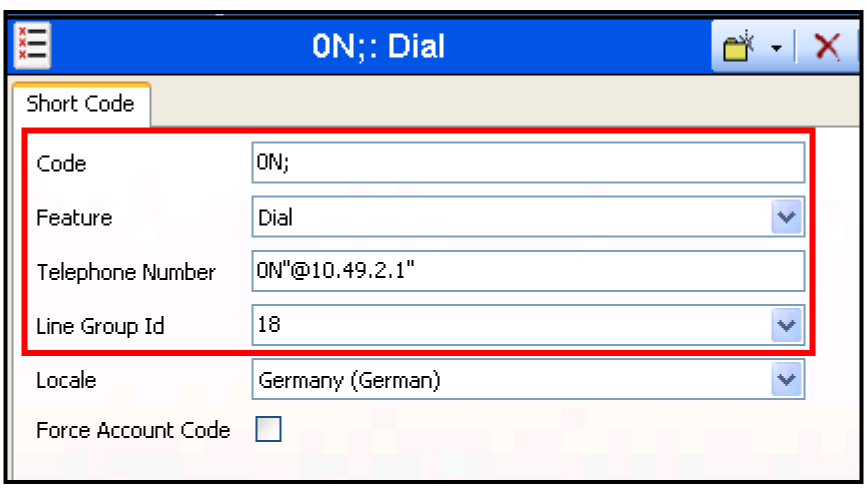

Figure 17: IP Office Outgoing Call Shortcode

From 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**.

Tab	Parameter	Usage
Standard	Line Group Id	Enter the Line Group Id assigned to the SIP trunk in **Figure 8**.
	Incoming Number	Enter the DID assigned to the telephone or FAX.
Destinations	Destination	Select the User to which the call is to be routed from the drop-down list.

Table 8: System-Parameters Features Parameters

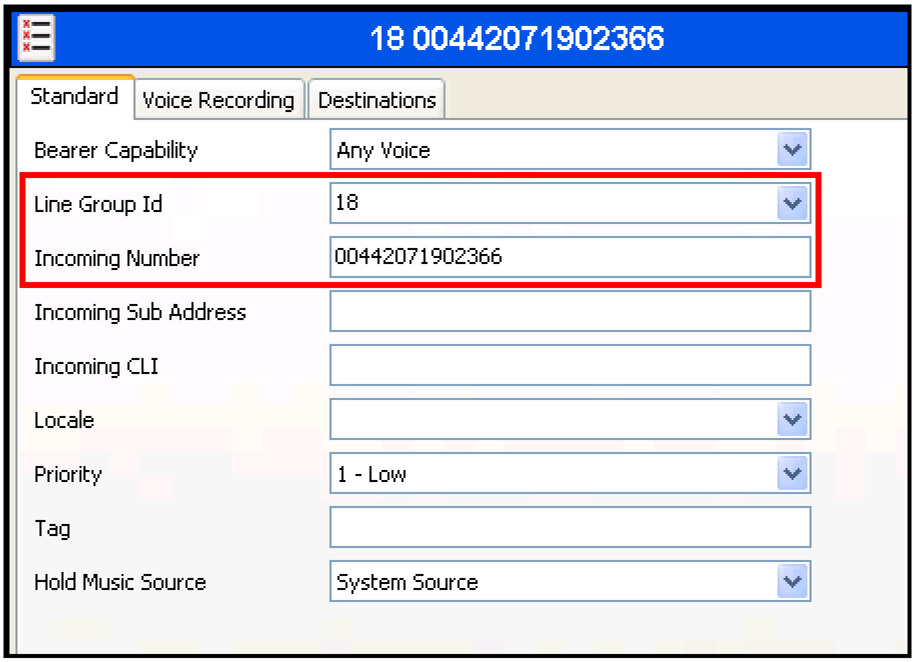

Figure 18: IP Office Incoming Call Route: Standard Tab

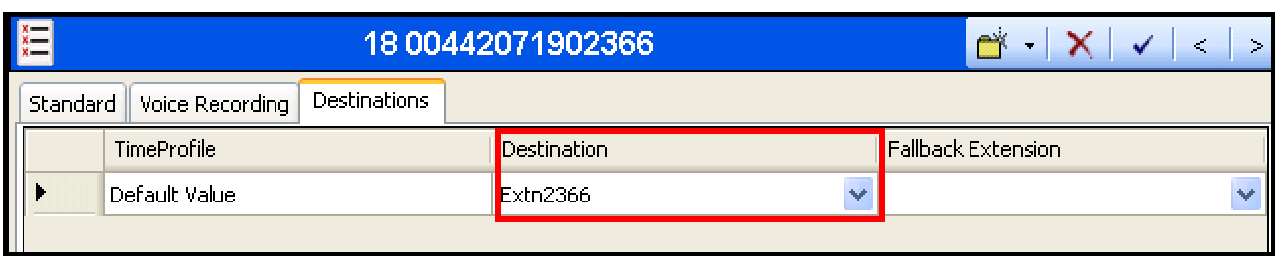

Figure 19: IP Office Incoming Call Route: Destinations Tab

5. General Test Approach and Test Results

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

The following issues were encountered during testing:

- If a PSTN endpoint calls to an IPO extension while the IPO is disconnected from network, the caller receives no feedback that there is a problem, and the call is cleared after several minutes.
- If a PSTN endpoint calls an IPO extension which has forwarded all calls to a busy PSTN endpoint, the IPO does not report the busy condition to the original caller. This problem has been reported to the Avaya Product Team (MRDB00043127).

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 Avaya IP Office Server can ping the Sonus GSX Session Border Controller address.
- 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 COLT Office Voice SIP trunk. All test cases passed with exceptions 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

These traces were made at the connection between the Avaya IP Office and the COLT 2611 router.

Incoming call:

```
INVITE sip:00442071902366@192.168.31.20:5060 SIP/2.0
Via: SIP/2.0/TCP 10.49.2.1:5060;branch=z9hG4bK0fB24915bf4ca7ba846
From: <sip:00496975050@10.49.2.1>;tag=gK0f28d5d3
To: <sip:00442071902366@192.168.31.20>
Call-ID: 420457899_66954515@10.49.2.1
CSeq: 7586 INVITE
Max-Forwards: 70
Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS,MESSAGE,PUBLISH
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Contact: <sip:00496975050@10.49.2.1:5060;transport=tcp>
P-Asserted-Identity: <sip:00496975050@10.49.2.1:5060>
Supported: timer,100rel
Session-Expires: 1800
Min-SE: 90
Content-Length:  302
Content-Disposition: session; handling=optional
Content-Type: application/sdp

v=0
o=Sonus_UAC 23062 11062 IN IP4 10.49.2.1
s=SIP Media Capabilities
c=IN IP4 10.49.2.10
t=0 0
m=audio 18124 RTP/AVP 18 8 2 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:8 PCMA/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=ptime:20
```

Outgoing call:

```
INVITE sip:00496975056174@10.49.2.1 SIP/2.0
Via: SIP/2.0/TCP 192.168.31.20:5060;rport;branch=z9hG4bK0cfde21023113eace9472466e3f17dff
From: "Extn2366" <sip:00442071902366@10.49.2.1>;tag=4915bf13cc967b5d
To: <sip:00496975056174@10.49.2.1>
Call-ID: 4956e0b60daf096ae99dcd0e3283db19@192.168.31.20
CSeq: 1951080416 INVITE
Contact: "Extn2366" <sip:00442071902366@192.168.31.20:5060;transport=tcp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, INFO
Content-Type: application/sdp
Content-Length: 254

v=0
o=UserA 1510400003 58713733 IN IP4 192.168.31.20
s=Session SDP
c=IN IP4 192.168.31.20
t=0 0
m=audio 49154 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

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