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.

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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 IP telephones using the H.323 protocol as endpoints. The communication between Avaya IP Office and COLT VoIP Access SIP trunking network is via the UDP protocol.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [4] 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.

The COLT VoIP Access SIP trunking network has also been tested with Avaya Communication Manager. The following is a list of significant behavioral differences between these products with respect to their operation with the COLT VoIP Access SIP trunking network:

- Avaya IP Office does not support direct IP connections between local endpoints, and those attached to the COLT VoIP Access SIP trunking network (shuffling). However, it does support RTP Relay, which allows Avaya IP Office to not allocate a Voice Compression Module (VCM) channel for calls between Avaya IP Telephones and the COLT VoIP Access SIP trunking network once a call is established, when matching codecs are used.
- Avaya IP Office does not support T.38 FAX via its SIP trunk. FAX operation was tested using the G.711A codec, and performed well receiving at 14.4kbps and transmitting at 9.6kbps.
- Avaya IP Office cannot configure individual devices to use different codecs. Thus, if FAX operation via the SIP trunk is required, the SIP trunk must be configured to use the G.711A codec.
- If local stations are diverted via the SIP trunk, the Avaya IP Office signals diverted calls to called parties with the diverting party indicated as being the caller.

1.1. System Configuration

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

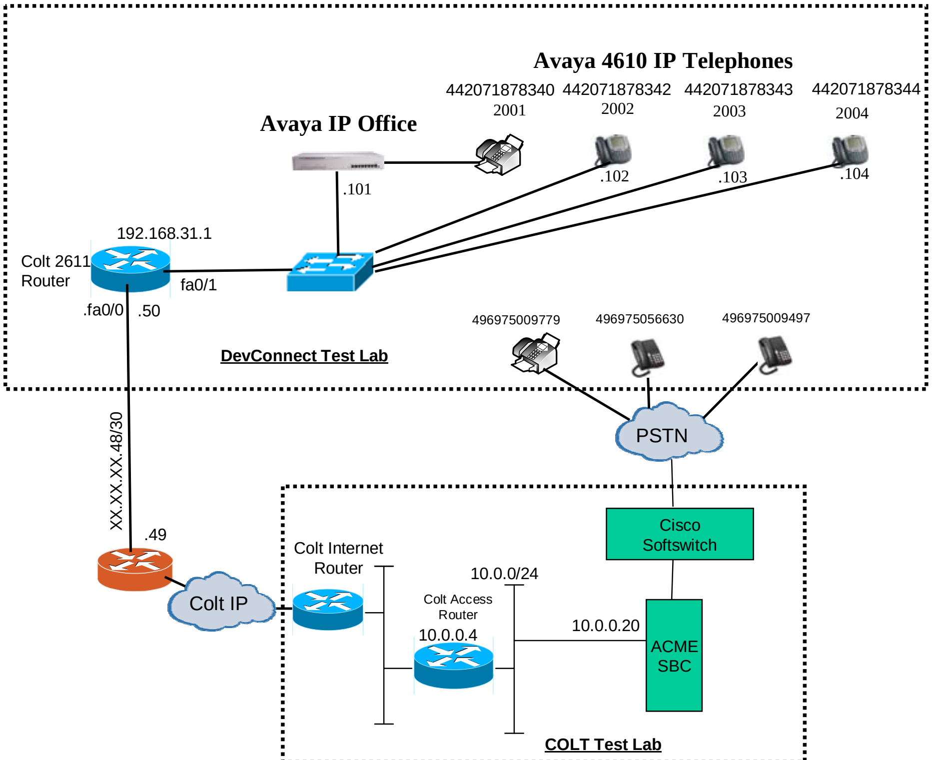

Figure 1: System Configuration

In the above diagram, Avaya IP Telephones are registered with the Avaya IP Office. Avaya 4610 IP Telephones configured for H.323 were used for testing.

The following table shows the extensions which were used for testing.

Endpoint Type	Local Extension	DID
FAX	2001	4420xxx8340
IP Telephone	2002	4420xxx8342
IP Telephone	2003	4420xxx8343
IP Telephone	2004	4420xxx8344
FAX		49xxxx9779
PSTN Telephone		
PSTN Telephone		

Table 1: Extensions Used for Testing

1.2. Call Routing

The telephone numbers received from the COLT VoIP Access SIP Service have the format 00<country code><national number> for both the called and the calling party, including numbers for those calls which originate in the local dialing area. The same number sequence can be used for dialing outgoing calls. Alternatively, calls to national destinations can be made by dialing 0<national number>.

2. Equipment and Software Validated

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

Equipment	Software
Avaya IP Office	4.2 (49501)
Avaya 4610 IP Telephones	2.8.3
ACME Session Border Controller	4.1.4 p24
Cisco Soft Switch	PGW 2200 version 9.7.3
Cisco Voice Gateway AS5400	12.4(9)

Table 2: Equipment and Software Validated

3. Configuration

All configuration steps for Avaya IP Office were performed using the IP Office Manager program.

3.1. Licenses

A license is required for SIP Trunk Channels, which can be confirmed by selecting the “Licenses” icon.

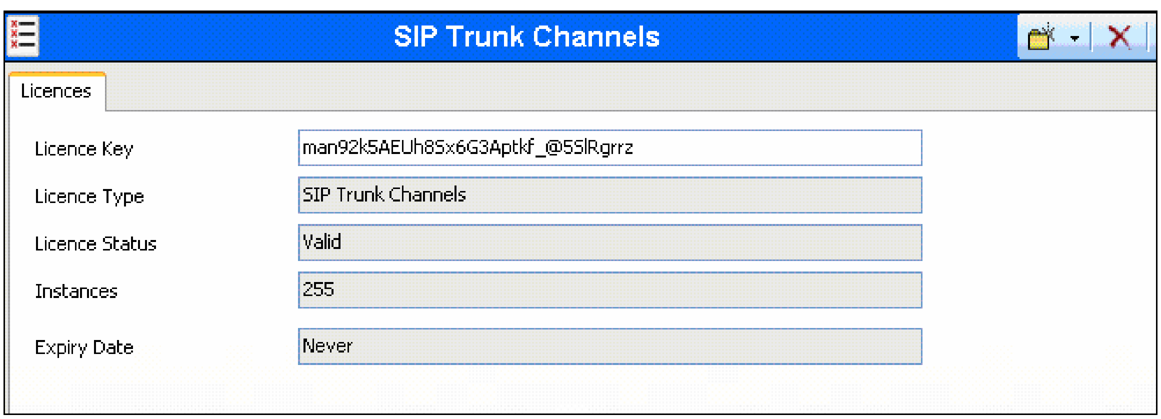

Figure 2: IP Office License for SIP Trunk Channels

3.2. System

Select the “System” icon and enter the parameters shown in the following table.

Tab	Parameter	Usage
System	TFTP Server IP Address	Enter the IP address assigned to IP Office.
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”.

Table 3: System-Parameters Features Parameters

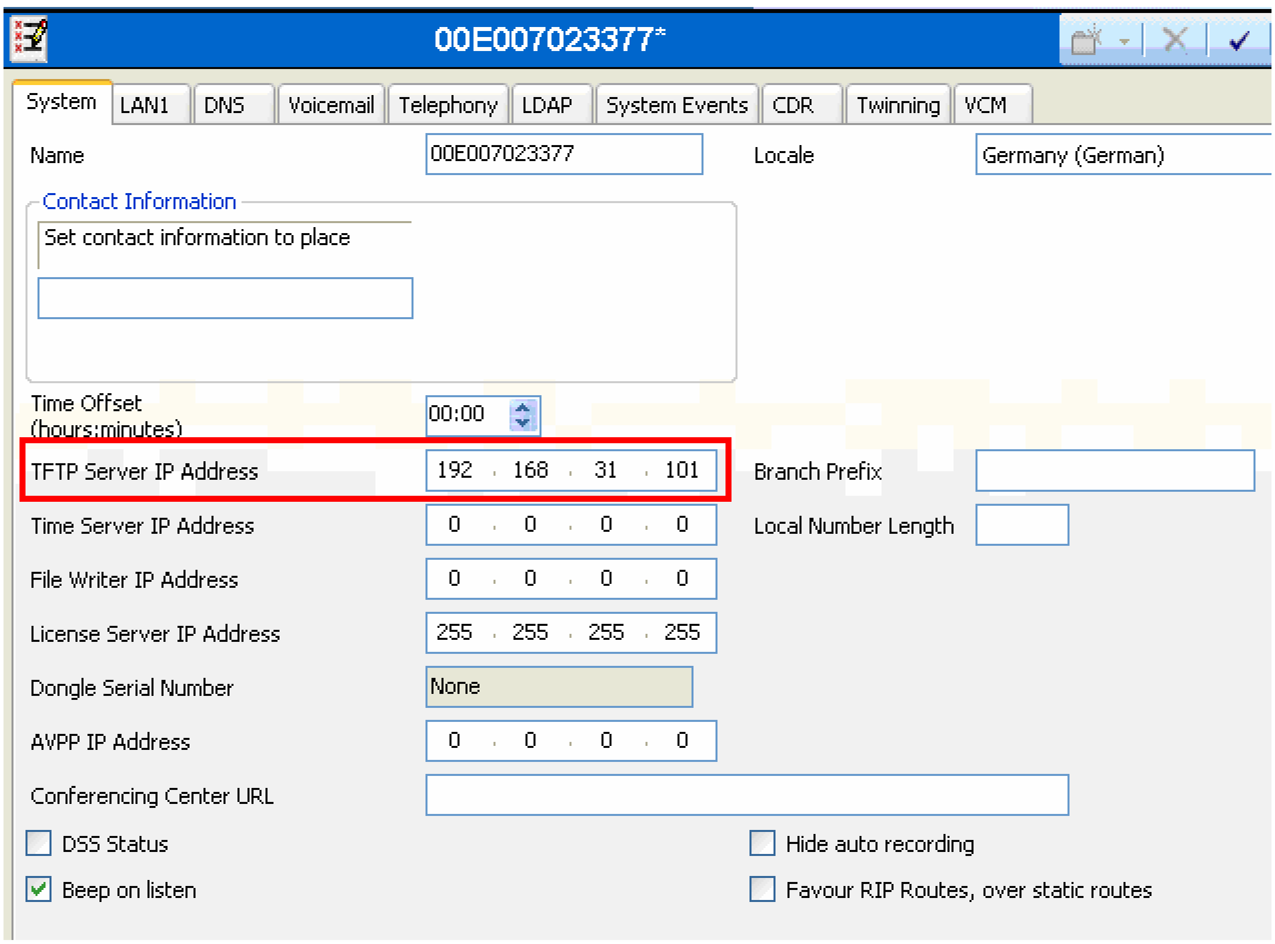

Figure 3: IP Office System: System Tab

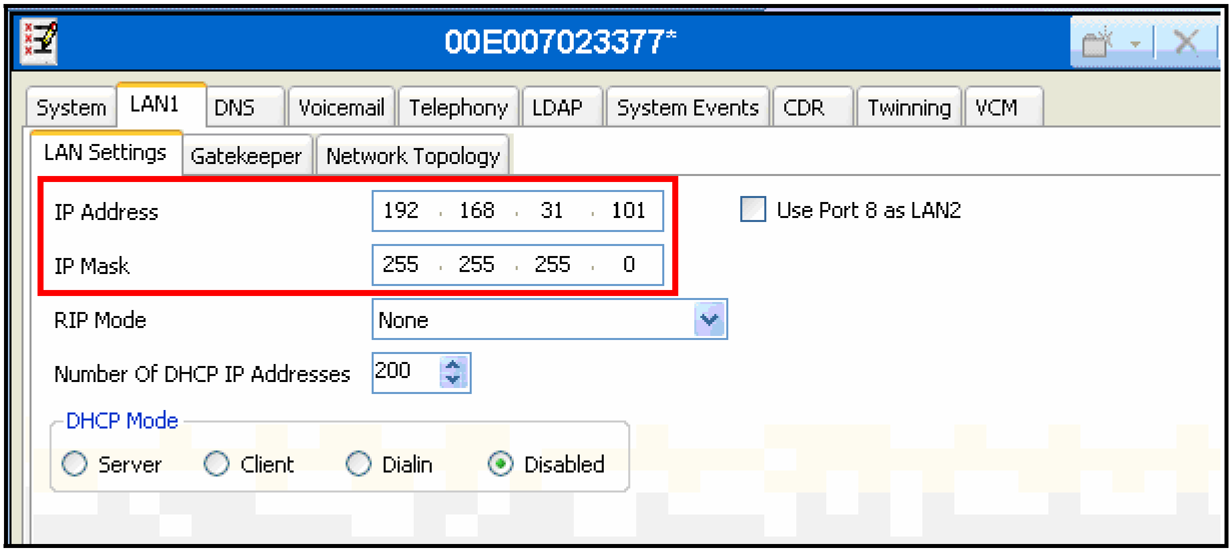

Figure 4: IP Office System: LAN Settings Tab

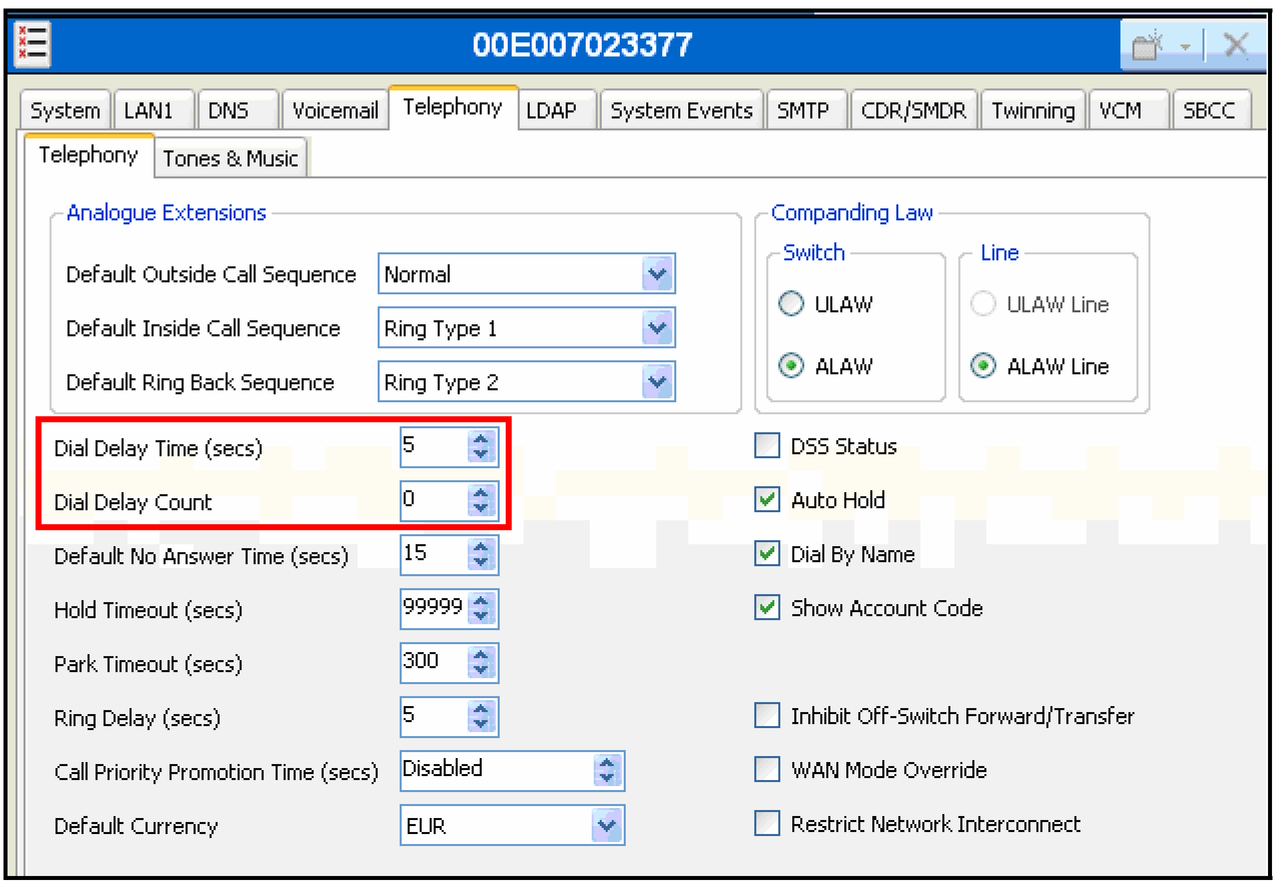

Figure 5: IP Office System: Telephony Tab

3.3. Default Gateway

Select the “IP-Route” icon 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 number	Select “LAN1” from the drop-down list.

Table 4: System-Parameters Features Parameters

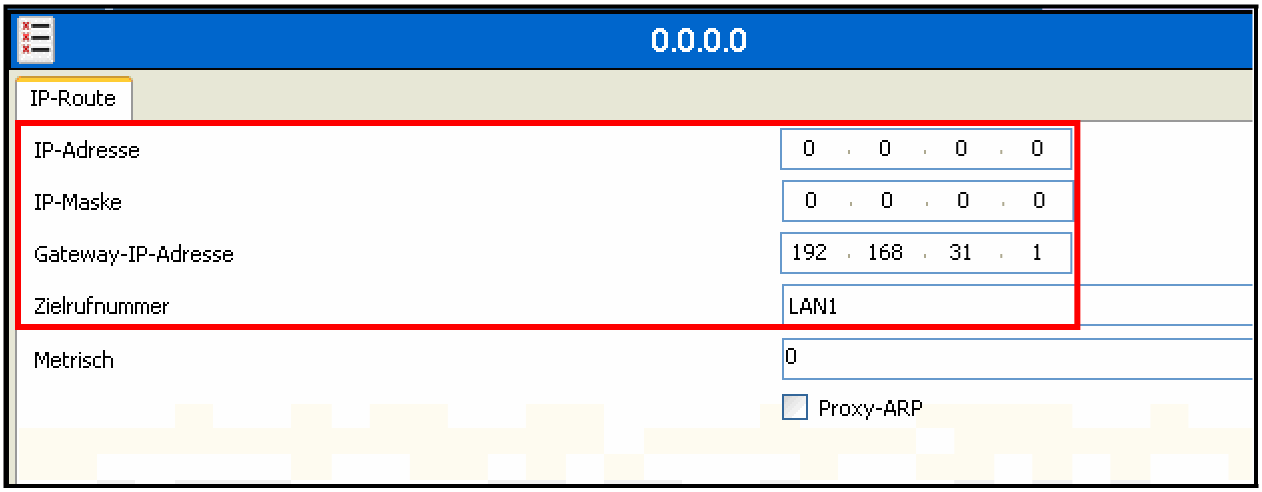

Figure 6: IP Office Route: Default Gateway

3.4. SIP Line

Select the “Line” icon 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 is checked, the IP Office selects the G.729 codec with “annex b” set to the default value of “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 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 to be used to identify the IP Office in SIP messages that it sends to the COLT VoIP Network.
	ITSP IP Address	Enter the IP address of COLT VoIP Network session border controller.
	VoIP Silence Suppression	Check this box if the G.711 codec is to be used. Uncheck it if the G.729 codec is to be used.
	RE-INVITE Supported	Select this box to enable RTP Relay.
	Compression Mode	Select G.729 from the drop-down box.
SIP URI	Incoming Group	Enter the number of an otherwise unassigned incoming group.
	Outgoing Group	Enter the number of an otherwise unassigned outgoing group.

Table 5: System-Parameters Features Parameters

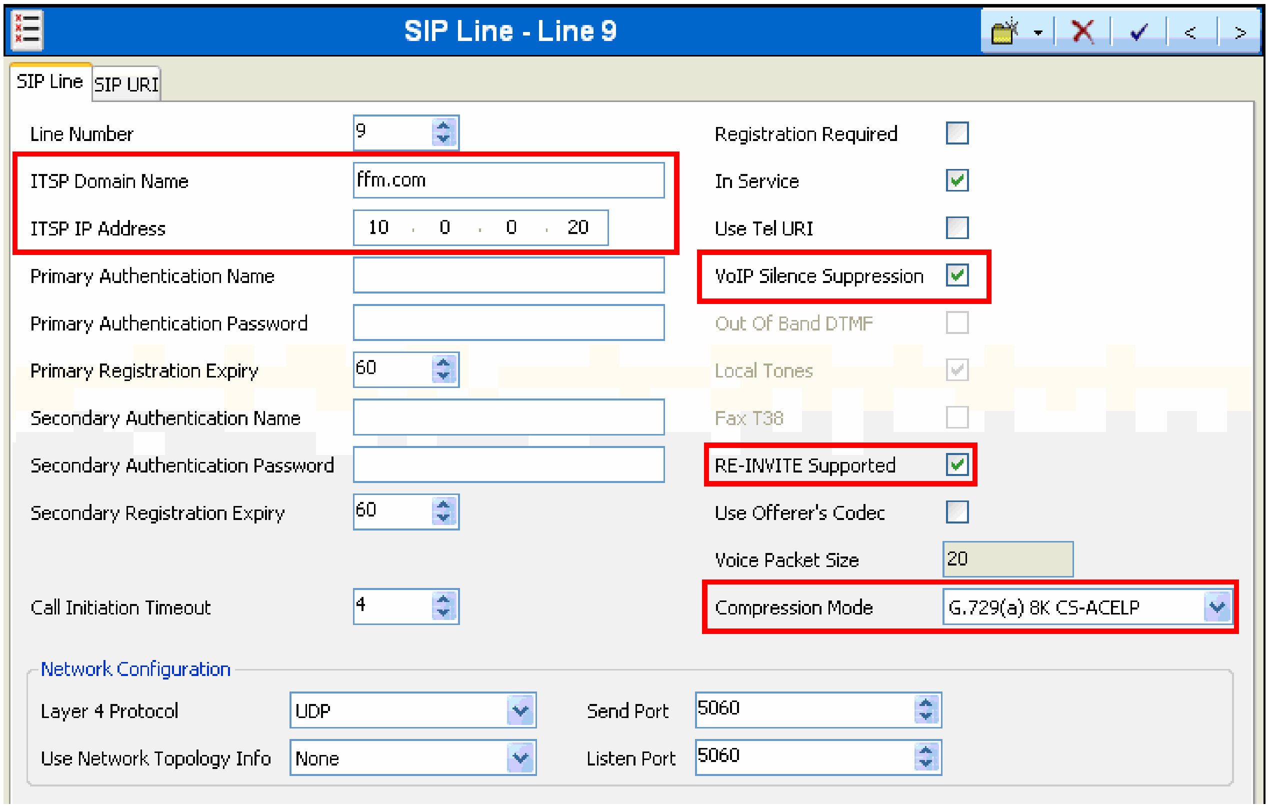

Figure 7: IP Office SIP Line: SIP Line Tab

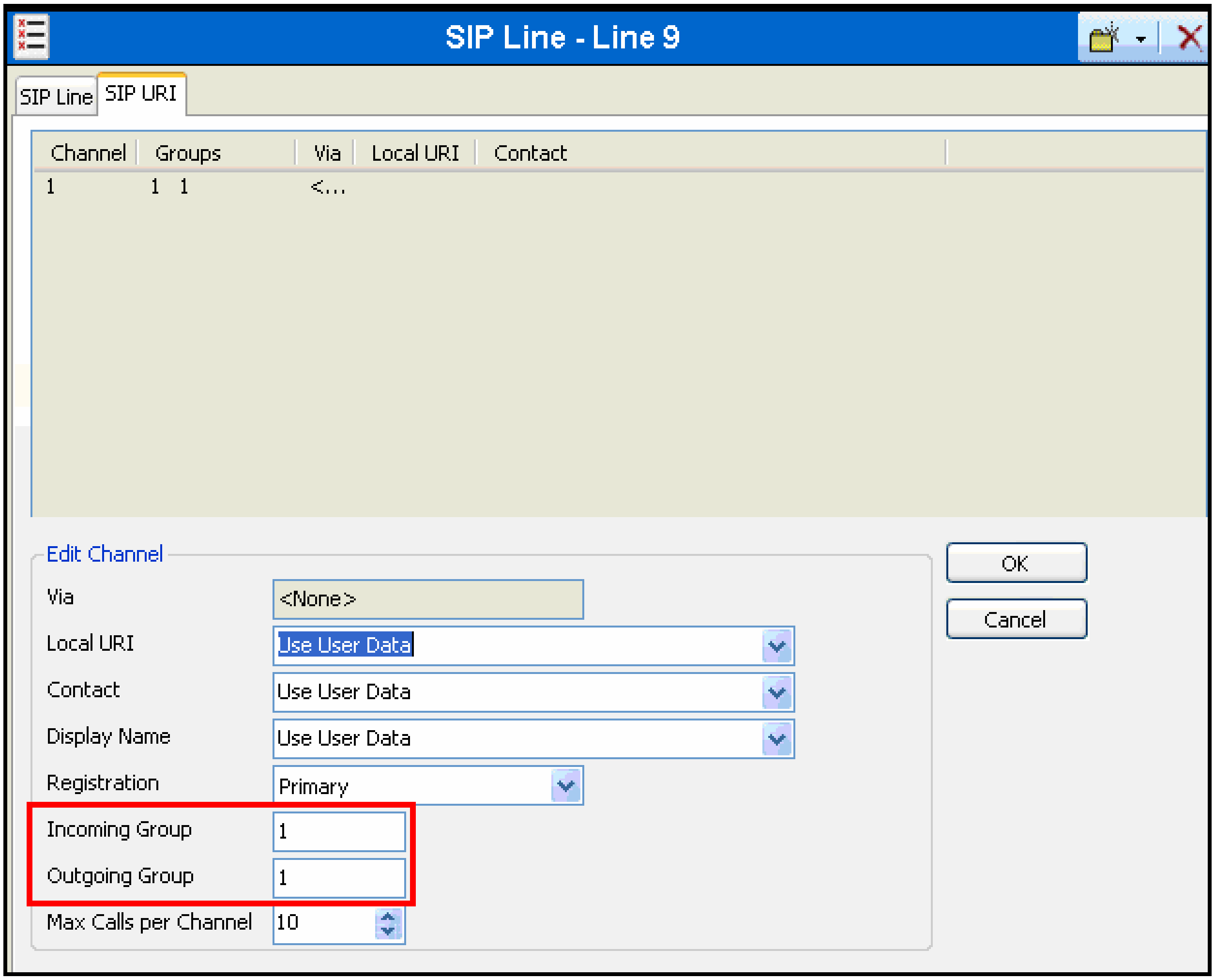

Figure 8: IP Office SIP Line: SIP URI Tab

With the "Users" icon 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.

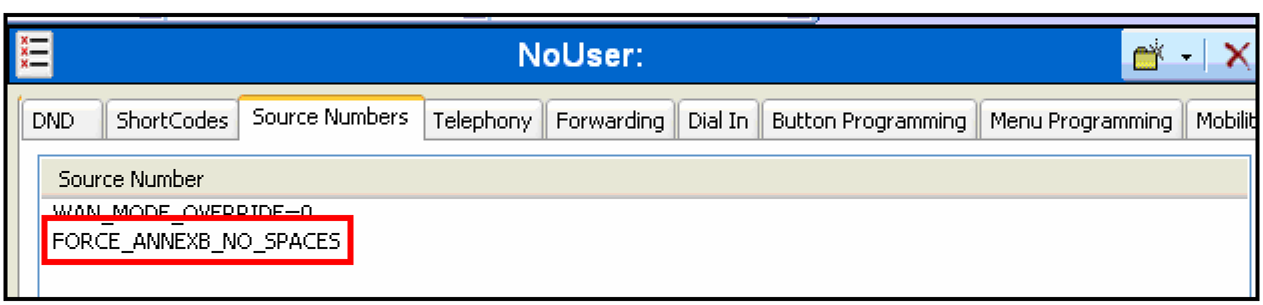

Figure 9: IP Office NoUser: Source Numbers Tab

3.5. FAX

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

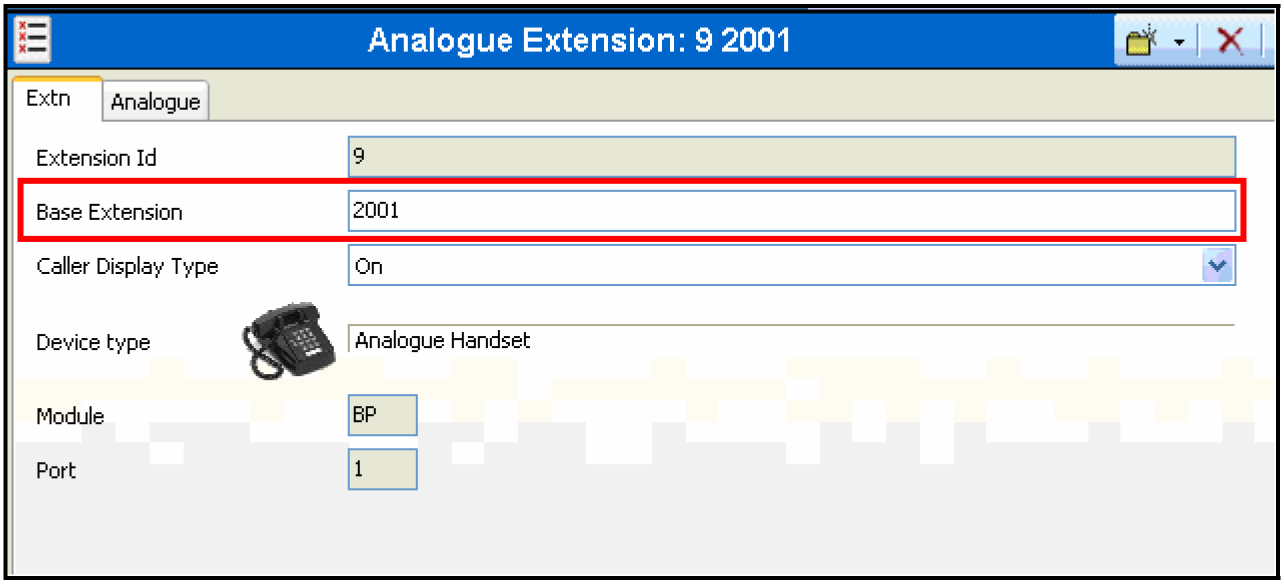

Figure 10: IP Office FAX Extension: Extn Tab

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

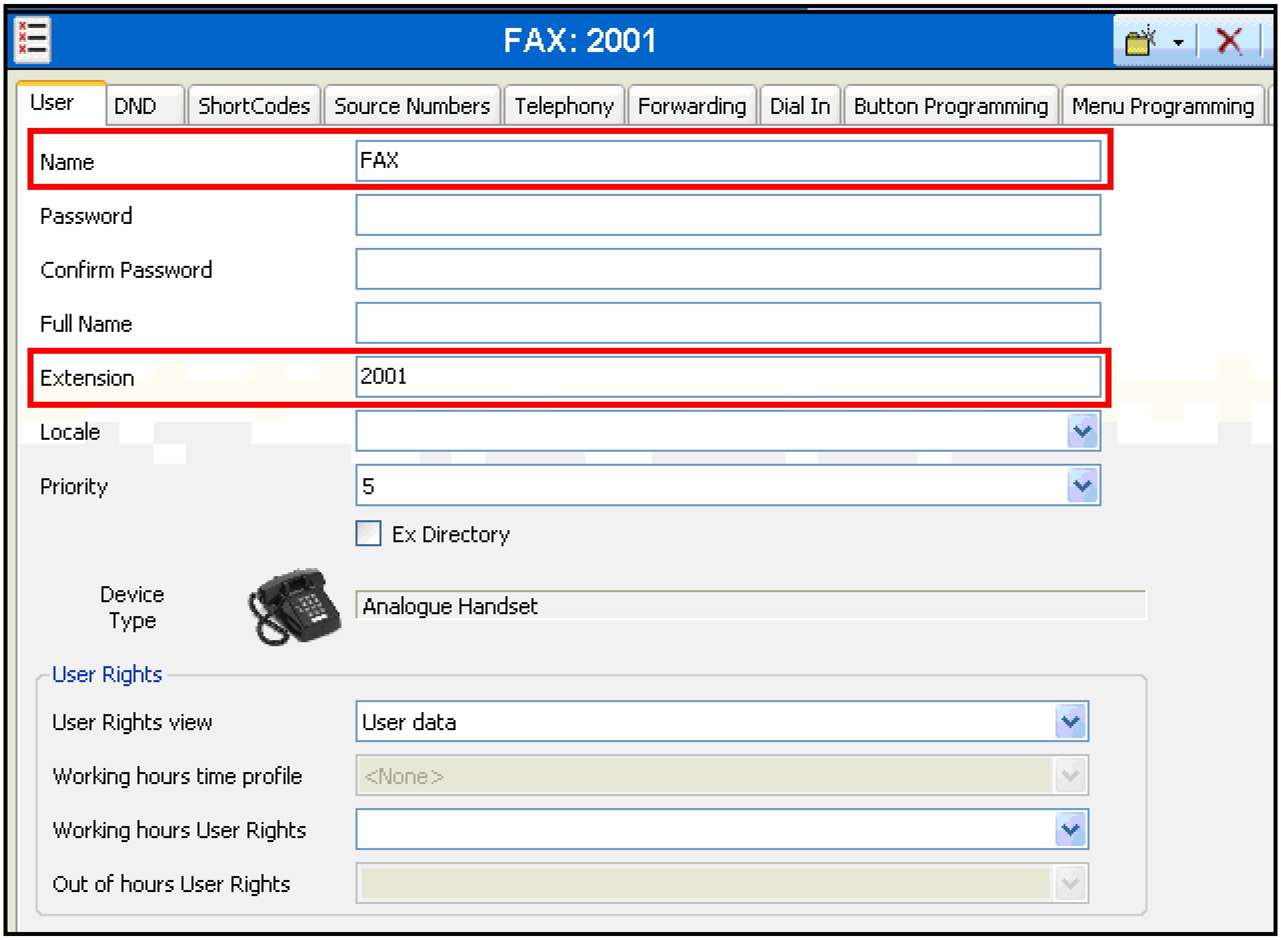

Figure 11: IP Office FAX User: Extn Tab

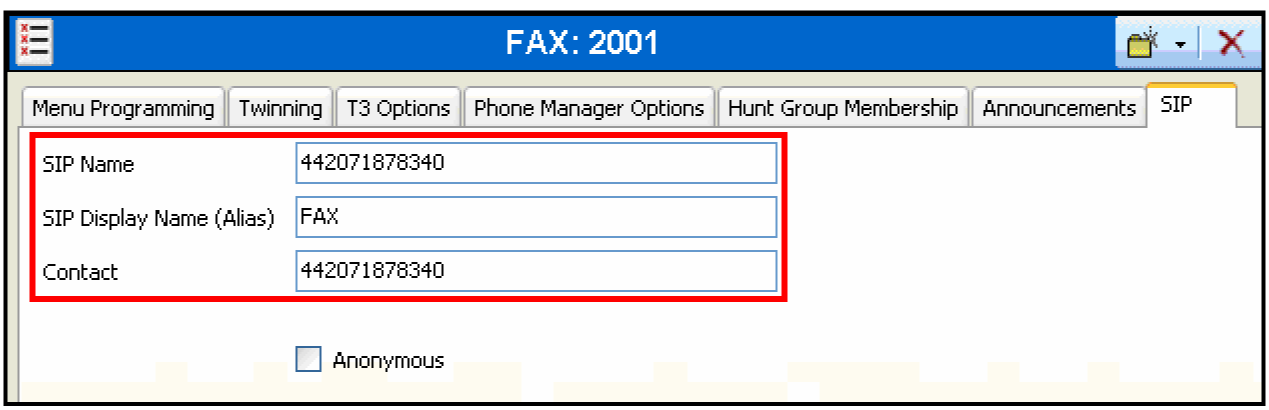

Figure 12: IP Office FAX User: SIP Tab

3.6. Local Telephone

From the "Extensions" icon, 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**.

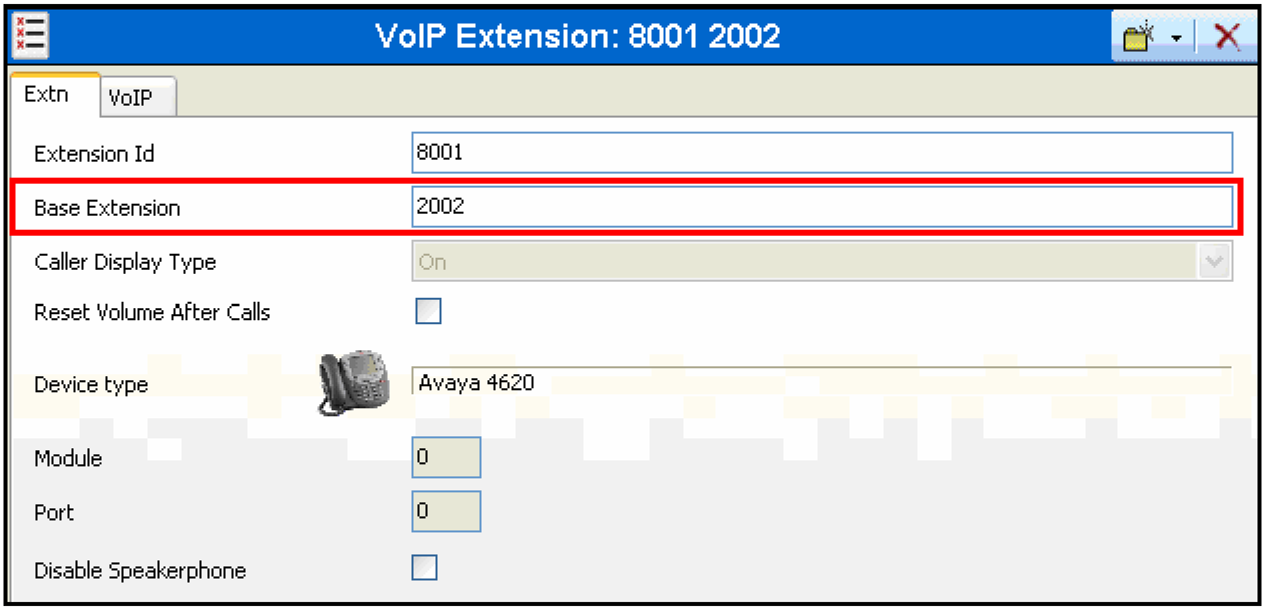

Figure 13: IP Office Local Telephone Extension: Extn Tab

From the "Users" tab, add a new user for each of the local telephones show in **Table 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: System-Parameters Features Parameters

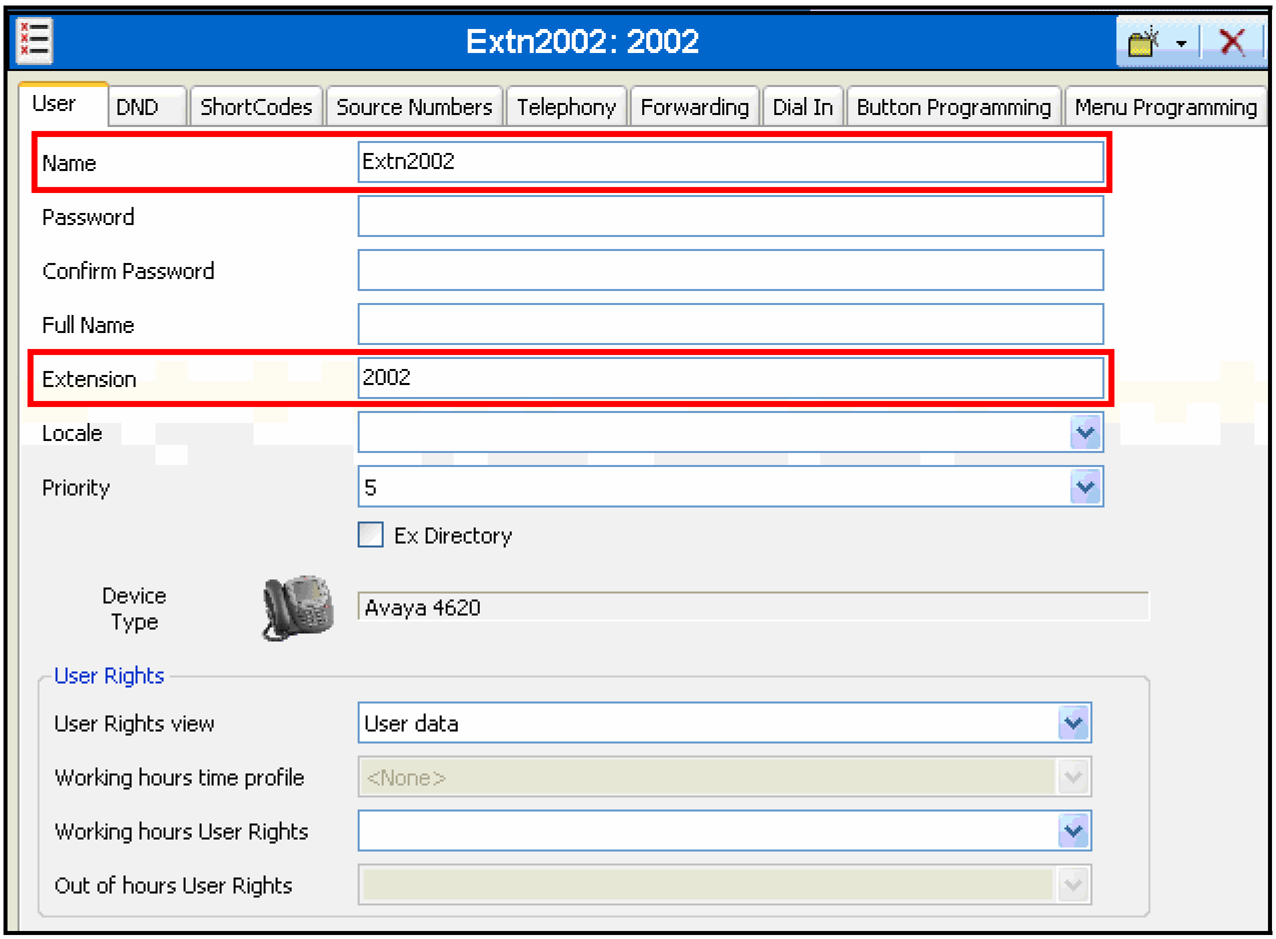

Figure 14: IP Office Local Telephone Extension: Extn Tab

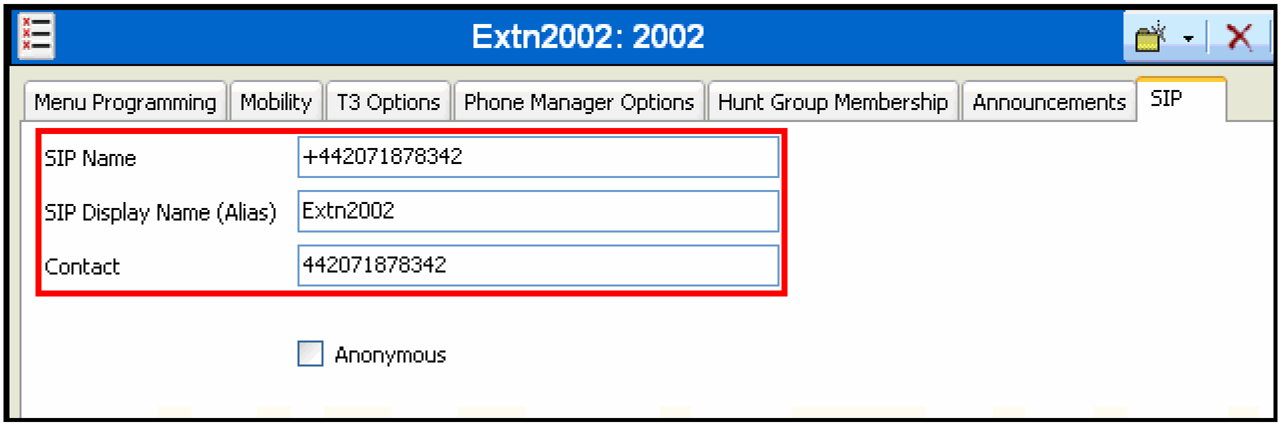

Figure 15: IP Office Local Telephone User: SIP Tab

3.7. Call Routing

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 IP Office.
Line Group Id	Enter the line group number assigned to the SIP Line "Outgoing Group" in **Figure 8**.

Table 8: System-Parameters Features Parameters

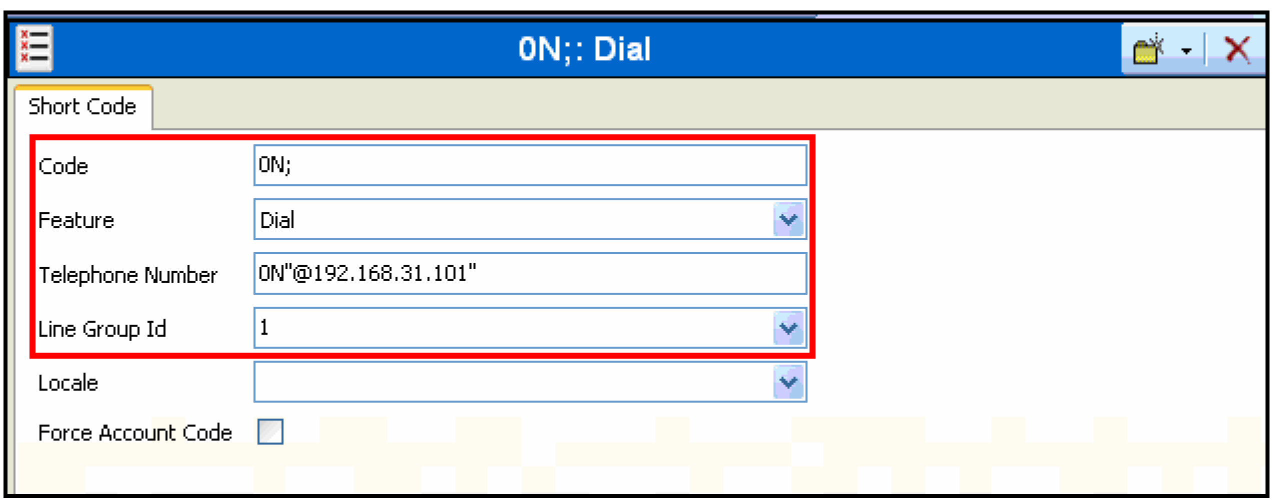

Figure 16: IP Office Outgoing Call Shortcode

From the “Incoming Call Route” icon, create a new incoming call route with the values shown in the following table, for each of the extension shown in **Table 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 9: System-Parameters Features Parameters

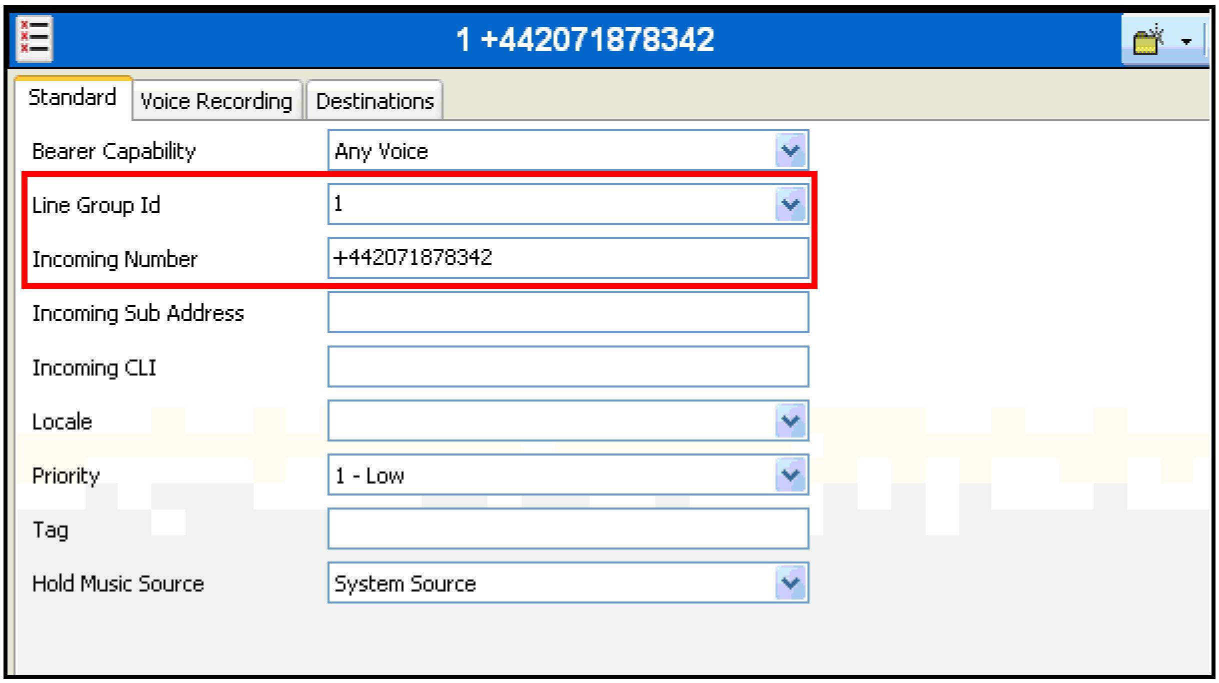

Figure 17: IP Office Incoming Call Route: Standard Tab

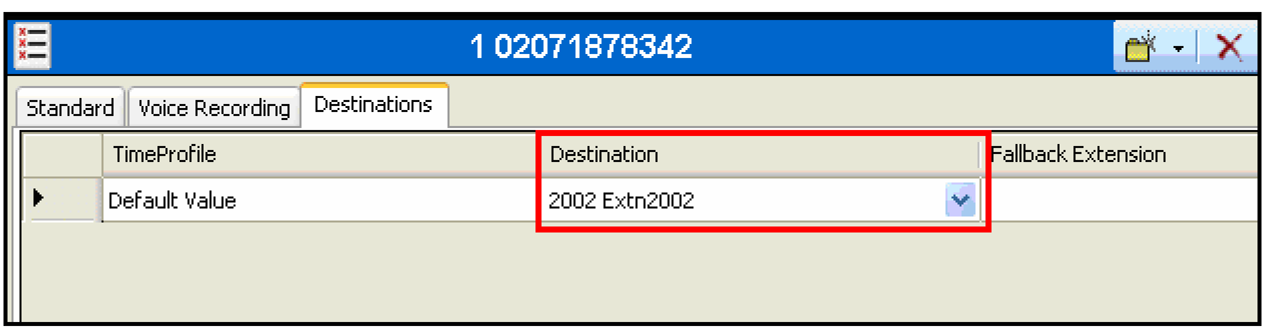

Figure 18: IP Office Incoming Call Route: Destinations Tab

4. Verification Steps

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

- Verify that the local Avaya IP Telephones can call each other.
- Verify that the Avaya IP Office Server can ping the SBC port allocated to Avaya IP Office.
- Verify that locally attached Avaya IP Telephones 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.

5. Interoperability Compliance Testing

5.1. General Test Approach

The following tests were performed:

- Incoming call from PSTN and GSM telephones: verify ringback, codec selection, speech path, calling party number display at called device, connected party number display at calling device, and proper call release after termination.
- Incoming call with no answer, and to busy extension.
- Incoming call to coverage.
- Incoming call to unassigned extension.
- Outgoing call to PSTN and GSM telephones: verify ringback, codec selection, speech path, calling party number display at called device, connected party number display at calling device, and proper call release after termination.
- Outgoing call with no answer, and to busy extension.
- Outgoing call to coverage.
- Outgoing call to unassigned number.
- Proper codec selection for G.711A and G.729 for incoming and outgoing calls.
- Verify the operation of RTP Relay for incoming and outgoing calls for both the G.711A and G.729 codecs.
- DTMF event signaling in both directions for both incoming and outgoing calls.
- Outbound call to domestic and international endpoint.
- Outbound call to internationally roaming GSM endpoint.
- Incoming and outgoing calls with number presentation restriction.
- Call hold/resume.
- Mute.
- Call forward unconditional, busy, no answer.
- Call forward PSTN caller to PSTN destination.
- Consultative and blind transfer local extension to PSTN and GSM endpoints.
- Consultative and blind transfer PSTN caller to local, PSTN, and GSM endpoints.
- Conference with local, PSTN, GSM endpoints.
- Multi-page FAX send/receive using G.711A codec.
- Multiple simultaneous calls.
- Long duration calls.

5.2. Test Results

Of the tested product features, a problem was encountered with DTMF events for incoming calls (from the COLT network to the Avaya IP Office). The Avaya IP Office responds to SDP requests for such calls by accepting a “telephone-event” payload type using a payload number which is different than that which was contained in the offer. The result is ambiguous DTMF signaling from the called to the calling party. This was not deemed to be a major problem, as the DTMF traffic from the called party is not normally of great interest. A fix will be pursued in a future Avaya IP Office release.

6. Conclusion

These Application Notes contain instructions for configuring Avaya IP Office to connect to the COLT SIP network. A list of instructions is provided to enable the user to verify that the various components have been correctly configured.

7. Additional References

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

[1] *Administrator Guide for Avaya IP Office*, January 2008, Issue 4.0, Document Number 03-300509.

[2] *Feature Description and Implementation for Avaya IP Office*, January 2008, Issue 6, Document Number 555-245-205.

[3] *4600 Series IP Telephone LAN Administrator Guide,* October 2007, Issue 7, Document Number 555-233-507

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.

[4] RFC 3261 - *SIP (Session Initiation Protocol),* June 2002, Proposed Standard

[5] 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:

```
Ethernet II, Src: 192.168.31.1 (00:06:28:e2:36:a1), Dst: 192.168.31.101 (00:e0:07:02:33:77)
Internet Protocol, Src: 10.0.0.20 (10.0.0.20), Dst: 192.168.31.101 (192.168.31.101)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:+442071878342@192.168.31.101:5060;user=phone SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 10.0.0.20:5060;branch=z9hG4bKt6eog0209gigec8ov6g1.1
        From: 00496975050 <sip:00496975050@va-test.sip.colt.net:5060;user=phone>;tag=SDj4q2c01-
1003958470
        To: +442071878342 <sip:+442071878342@192.168.31.101:5060;user=phone>
        Call-ID: SDj4q2c01-03c135a4644d483144f084eb37362e92-v300g00
        CSeq: 1 INVITE
        Max-Forwards: 69
        Supported: timer
        Session-Expires: 1800
        Min-SE: 1800
        Contact: 00496975050 <sip:00496975050@10.0.0.20:5060;transport=udp>
        Allow: INVITE,ACK,PRACK,SUBSCRIBE,BYE,CANCEL,NOTIFY,INFO,REFER,UPDATE
        Content-Type: application/sdp
        Content-Length: 399
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 871 0 IN IP4 10.0.0.20
            Session Name (s): Cisco SDP 0
            Connection Information (c): IN IP4 10.0.0.20
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 20054 RTP/AVP 18 8 99 100
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): fmtp:18 annexb=no
            Media Attribute (a): rtpmap:99 telephone-event/8000
            Media Attribute (a): fmtp:99 0-15
            Media Attribute (a): rtpmap:100 X-NSE/8000
            Media Attribute (a): fmtp:100 192-194,200-202
            Media Attribute (a): X-sqn:0
            Media Attribute (a): X-cap: 1 audio RTP/AVP 100
            Media Attribute (a): X-cpar: a=rtpmap:100 X-NSE/8000
            Media Attribute (a): X-cpar: a=fmtp:100 192-194,200-202
            Media Attribute (a): X-cap: 2 image udptl t38
```

Outgoing call:

```
Ethernet II, Src: 192.168.31.101 (00:e0:07:02:33:77), Dst: 192.168.31.1 (00:06:28:e2:36:a1)
Internet Protocol, Src: 192.168.31.101 (192.168.31.101), Dst: 10.0.0.20 (10.0.0.20)
User Datagram Protocol, Src Port: 5060 (5060), Dst Port: 5060 (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:00496975056630@192.168.31.101 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP 192.168.31.101:5060;rport;branch=z9hG4bK82057936565880f79174cd90e4fb803a
        From: "Extn2003" <sip:+442071878343@ffm.com>;tag=b3a9feb6a39feb59
        To: <sip:00496975056630@192.168.31.101>
        Call-ID: e32e0d1a753c85d8d6b7e4a230338537@192.168.31.101
        CSeq: 1119903903 INVITE
        Contact: "Extn2003" <sip:442071878343@192.168.31.101:5060;transport=udp>
        Max-Forwards: 70
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
        Content-Type: application/sdp
        Content-Length: 257
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): UserA 2112644096 932374743 IN IP4 192.168.31.101
            Session Name (s): Session SDP
            Connection Information (c): IN IP4 192.168.31.101
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 49154 RTP/AVP 8 18 0 101
            Media Attribute (a): rtpmap:8 PCMA/8000
            Media Attribute (a): rtpmap:18 G729/8000
            Media Attribute (a): rtpmap:0 PCMU/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.