AVAYA

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Club Communication SIP Trunking Service and an Avaya IP Office Telephony Solution – Issue 1.1

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Club Communications VoIP Network and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office and Avaya H.323, digital, and analog endpoints. Club Communications ltd is a member of the Fixed Service Providers Association and a member of the Internet Telephony Service Providers Association.

Club Communications provides Internet Protocol (IP) services. Enterprise customers with an Avaya IP Office SIP-based network can connect to the Club Communication VoIP Network over the Internet and access the PSTN by subscribing to Club Communication VoIP services.

Club Communications is a member of the Avaya DevConnect Service Provider program. 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 steps to configure Session Initiation Protocol (SIP) trunking between the Club Communications VoIP Network and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office and Avaya H.323, digital, and analog endpoints. Club Communication is an international service provider for customers with a SIP based network.

Customers using this Avaya IP Office telephony solution with Club Communications VoIP Network services are able to place and receive PSTN calls via a dedicated broadband Internet connection using the SIP protocol. This converged network solution is an alternative to more traditional PSTN trunks such as T1 or ISDN PRI.

Club Communication can connect directly to an IP phone system as well as an external router. Club Communications VoIP Network service offerings are as follows:
- Outbound calling to local services
- Outbound international calling
- Outbound toll free and emergency calling
- Incoming Direct Inward Dial service

Figure 1 illustrates a sample Avaya IP Office telephony solution connected to the Club Communications VoIP Network solution. This configuration was utilized for compliance testing.

Note: The IP Addresses used for testing, shown in **Figure 1** and throughout this document were masked for security purposes.

The following equipment comprised the Avaya IP Office telephony solution and simulated a customer site:
- Avaya IP Office 406v2
- Avaya 5621 SW IP and T3 IP Classic Telephone (H.323 protocol)
- Avaya T3.14 Classic Digital Telephone
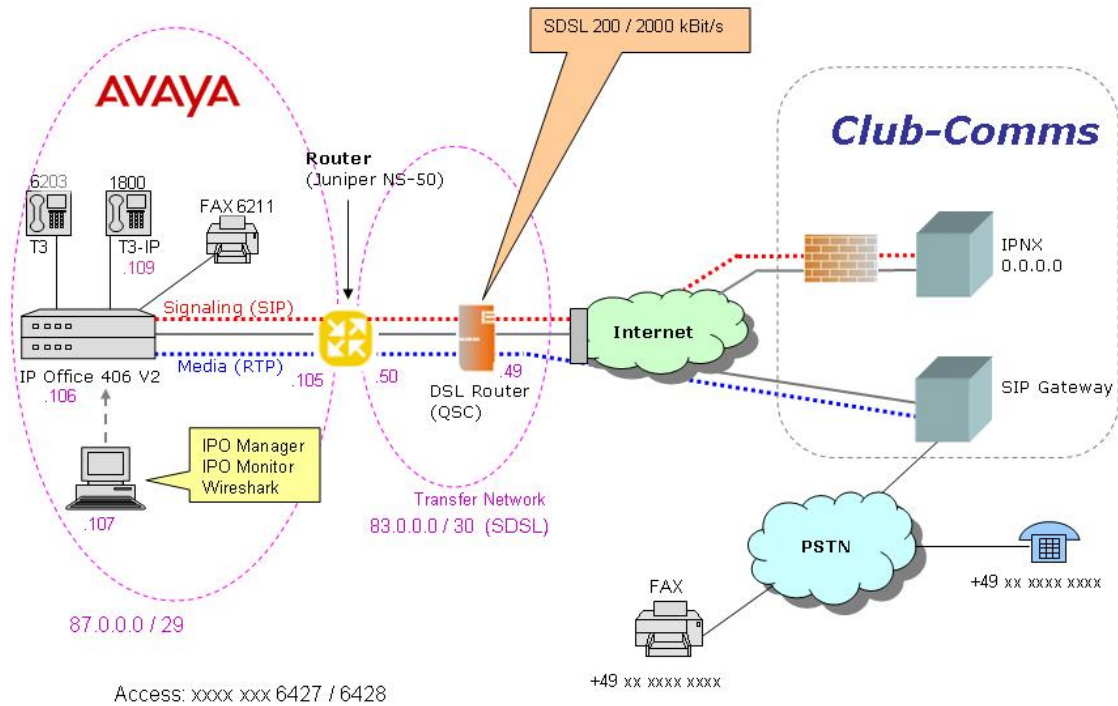- Avaya T3 Standard Analog Telephone

Figure 1: Avaya IP Office Telephony Network using Global Club Communications VoIP
Network solution

2. Equipment and Software Validated

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

Avaya IP Office 406v2	R 4.1.12
Avaya IP Office Manager (Windows PC)	R 6.1.12
Avaya 5621 SW IP Telephone	H.323 – i20d01a2184e.bin
Avaya T3 14 Classic	V02_01
Avaya T3 IP II Classic	H.323 - T233_0DE
Avaya T3-Standard Analog Telephone	n/a

Club Communications SIP Trunking Solution Components	
IPNX	n/a

Table 1: Equipment and Software Tested

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.1.12.

3. Configure Avaya IP Office

This section describes the steps for configuring a SIP trunk on IP Office.

IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start → Programs → IP Office → Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

1. Verify that there is a valid SIP Trunk Channels License. Double-click on **License** in the left panel. Check that there is a **SIP Trunk Channels** entry.

   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.
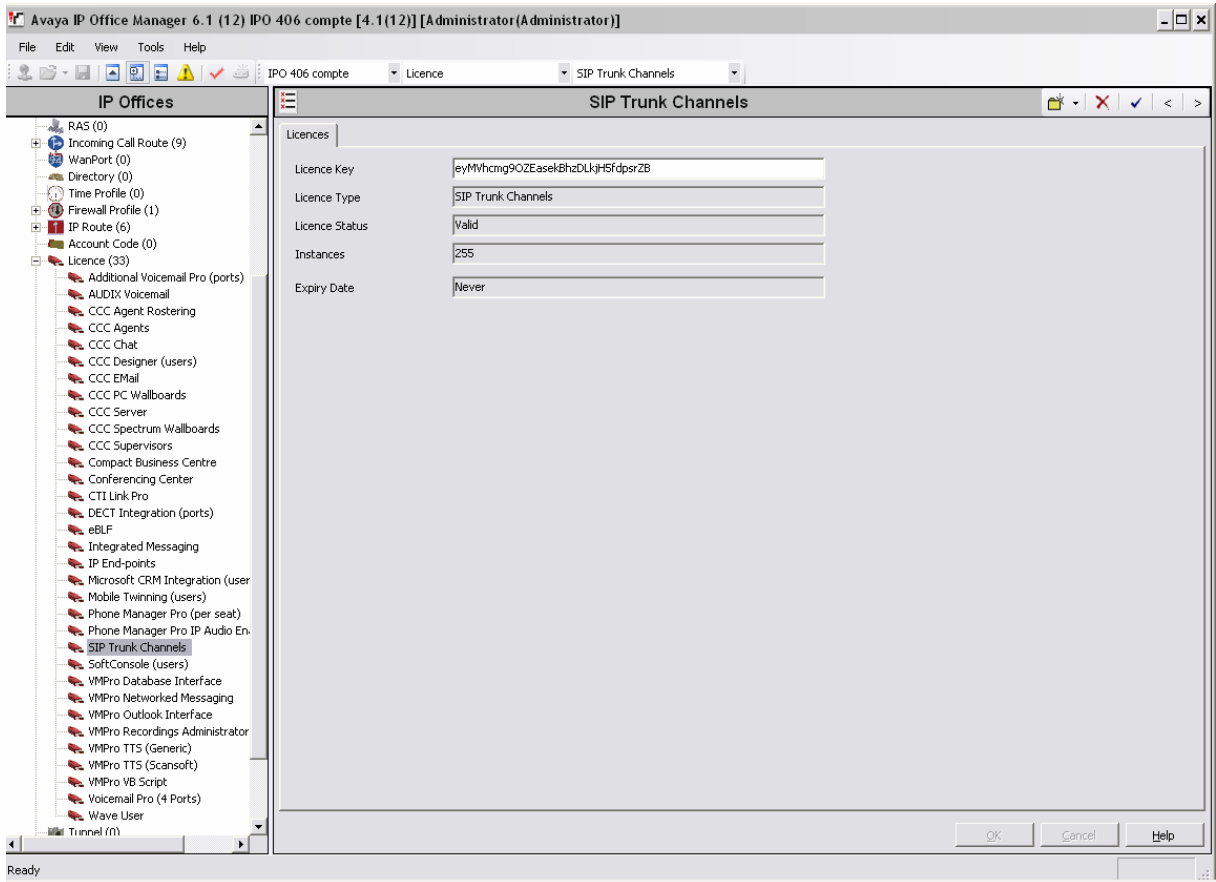
2. Enable SIP Proxy Functionality. Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper tab,** and check the **SIP Proxy Enable** box. Press the **OK** button. Note: Some changes may require a system reboot to take effect.
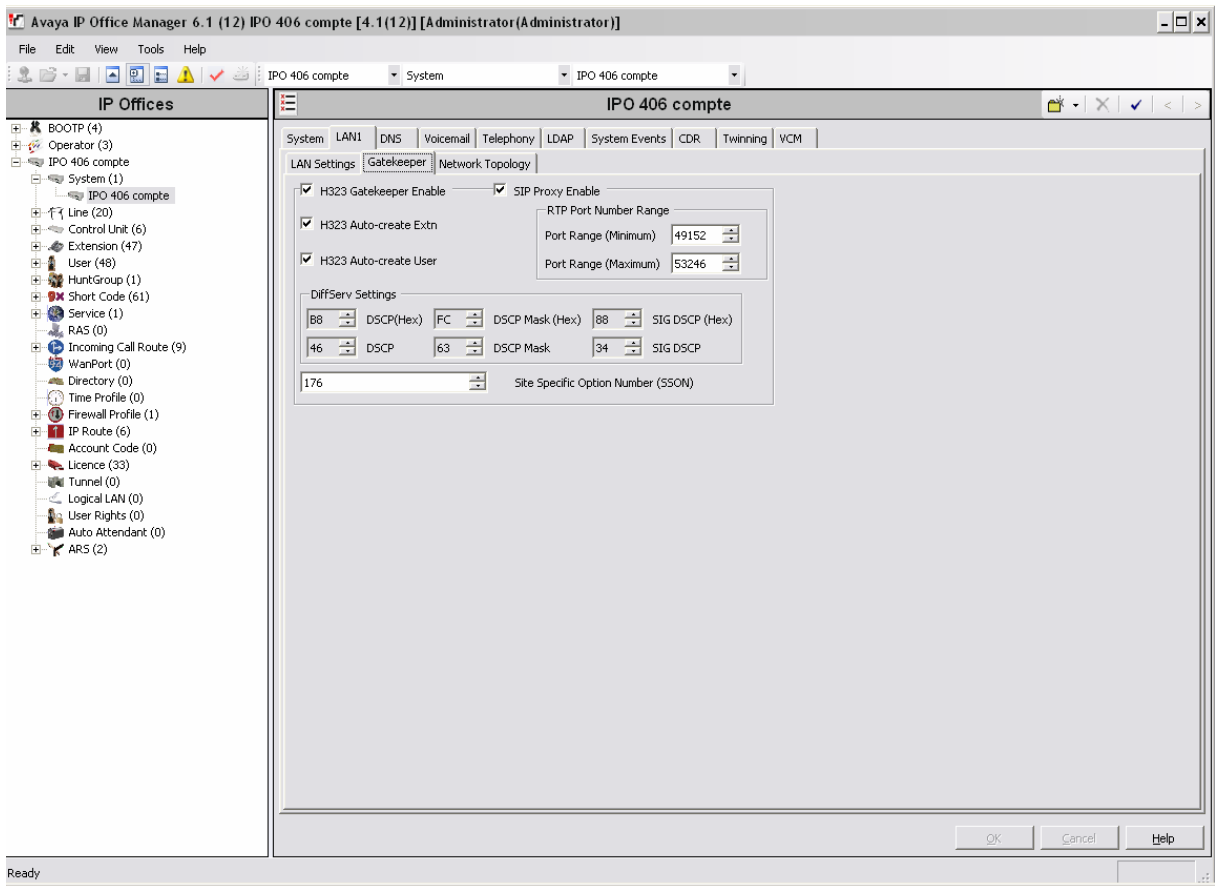
3. Create the SIP line for the Club Communications service. Select **Line** in the left panel. Right-click and select **New → SIP Line**.

    Select the following:
    - For the **ITSP IP Address field**, enter the IP address of the Club Communications SIP Proxy. The IP address in the figure is masked to 000.000.000.000 .
    - For **Compression Mode**, select the **G729a 8K CS-ACELP** for all voice calls. (To ensure proper interoperability between Avaya IP Office and Club Communication for voice calls, G.729a is required. G.711a / T38 is reserved for Fax calls only, but the IP Office doesn't support T38)
    - For **Layer 4 Protocol**, use **UDP**
    - For **Send Port**, use **5060**
    - For **Listen Port** use **5060**
    - Use defaults for other fields

    Press the **OK** button.
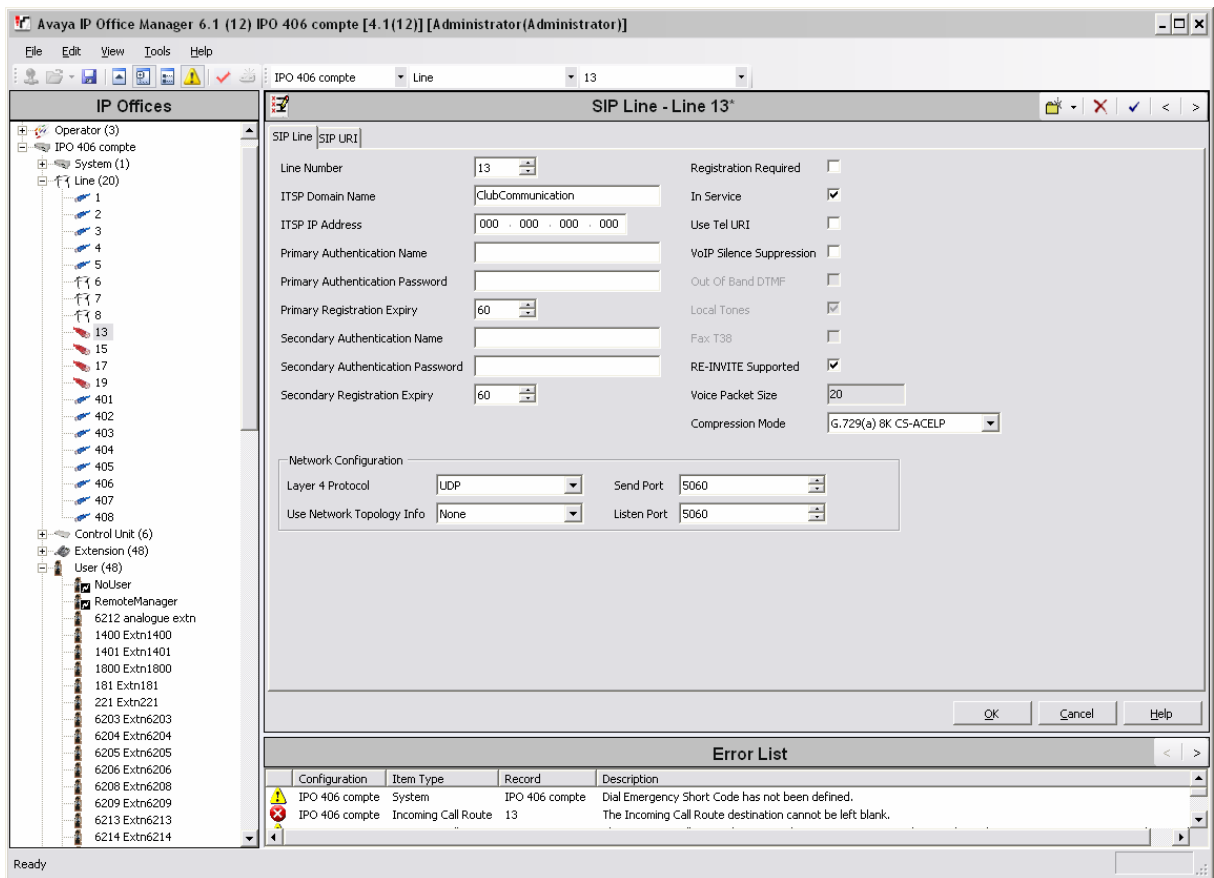
4. Configure URI parameters for the line. Select the **SIP URI** Tab. Press the **Add** button. Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. Select **Use User Data** for the **Contact**, **Local URI** and **Display Name** fields. This tells the system to use the information configured on the SIP tab for each individual User (see **Step 7**). Use defaults for all other fields. Press the **OK** button.
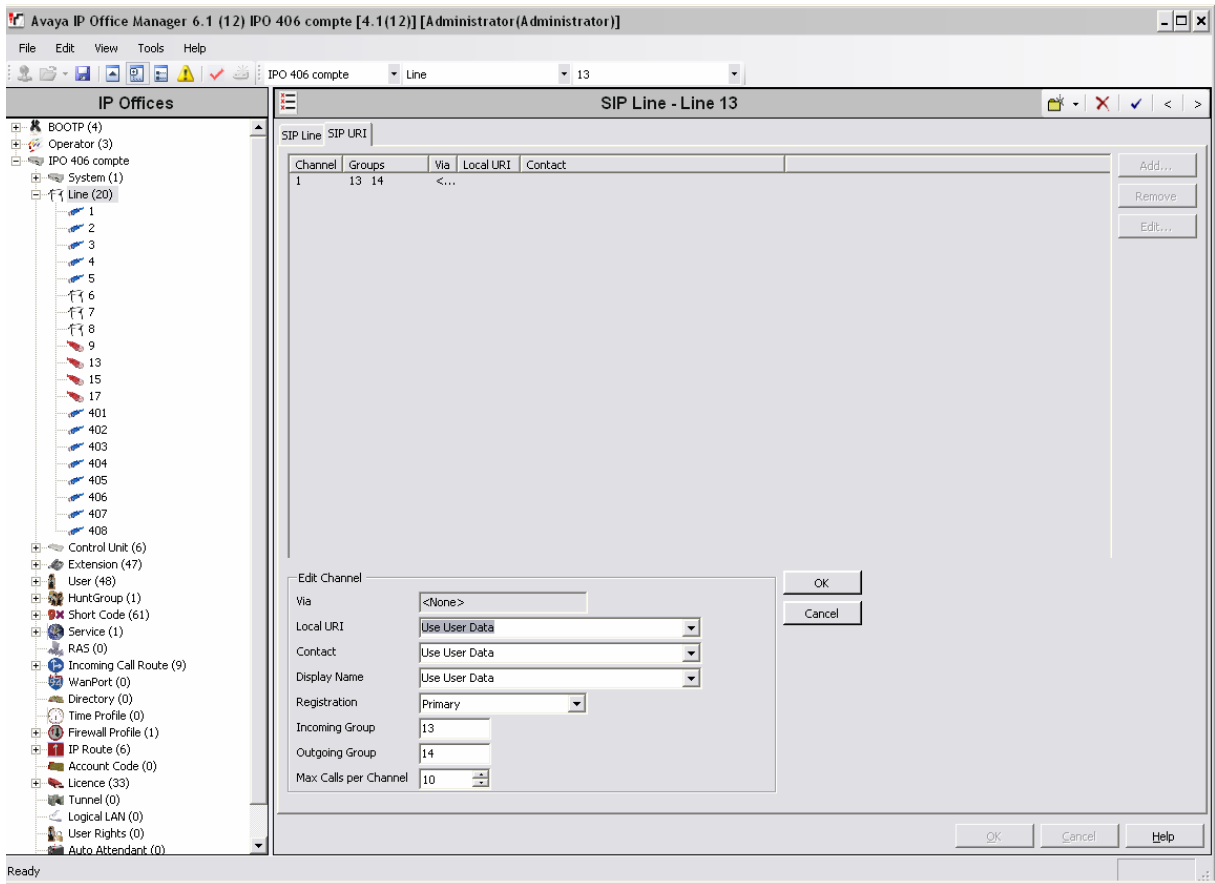
5. Configure a short code to route calls to Club Communications SIP Proxy Server. Select **Short Code** in the left panel. Right click and select **Add,** enter **0N** in the Code text box. This code requires the user to dial the digit 0 followed by the destination's telephone number symbolized by **N** in order to route the call out the SIP Trunk.

   Note: **N** can be any number other than a local IP Office extension.
   Instead of **0**, you can use any other digit.

   Select **Dial** for the **Feature**. Enter the **Outgoing Group** created in **Step 4** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<IP Address of Club Communications >"** for the **Telephone Number** field. In the figure the IP Address is masked to xx.xx.xx.xx. xx.xx.xx.xx is the IP Address from the ITSP (0.0.0.0). The **Telephone Number** field is used to construct the To field's SIP URI in the outgoing SIP INVITE message (see Appendix A for examples of SIP INVITE messages). Use default values for all other fields. Press the **OK** button.
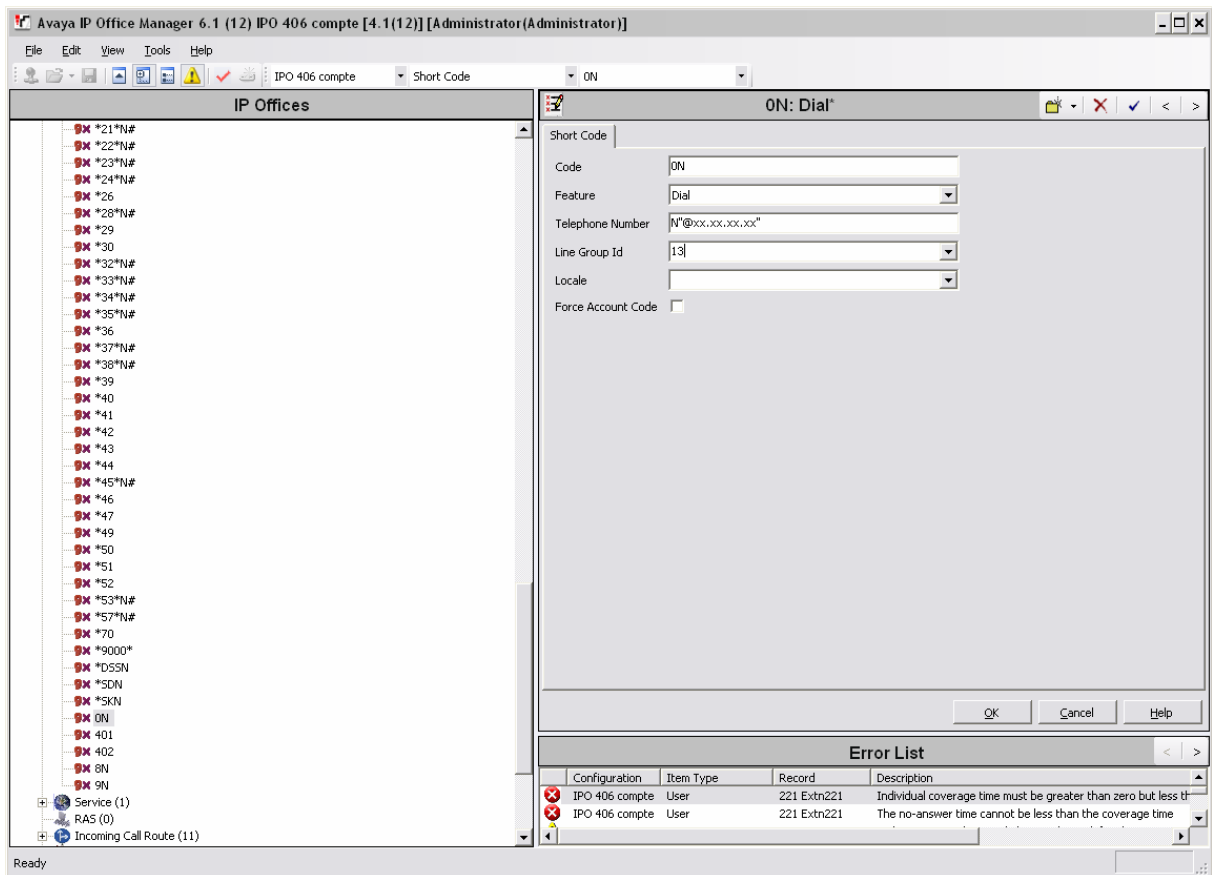
6. Create an Incoming Call Route for the Inbound SIP calls. Select **Incoming Call Route** in the left panel. Right-click and select **New**.

   Enter the following in the **Standard** tab:
   • **Any Voice** for the **Bearer Capability** field.
   • The **Incoming Group** created for the URI in **Step 4** in the **Line Group Id** field.
   • The 10 digit number provided by Club Communication, that is mapped back to a local IP Office extension, in the **Incoming Number** field.
   • Use default values for all other fields.
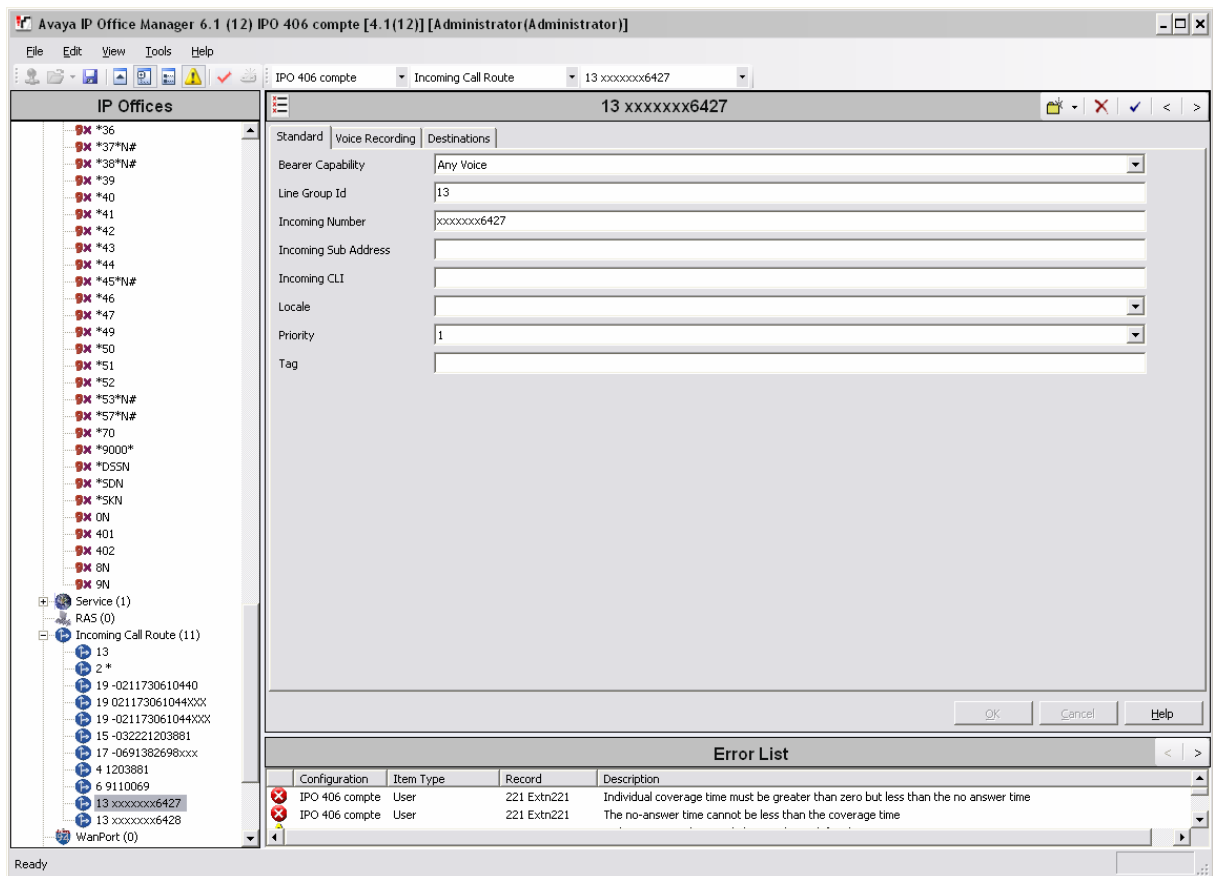
- Next, navigate to the **Destinations** tab, select the desired local extension number from the drop down menu.
- Press the **OK** button.
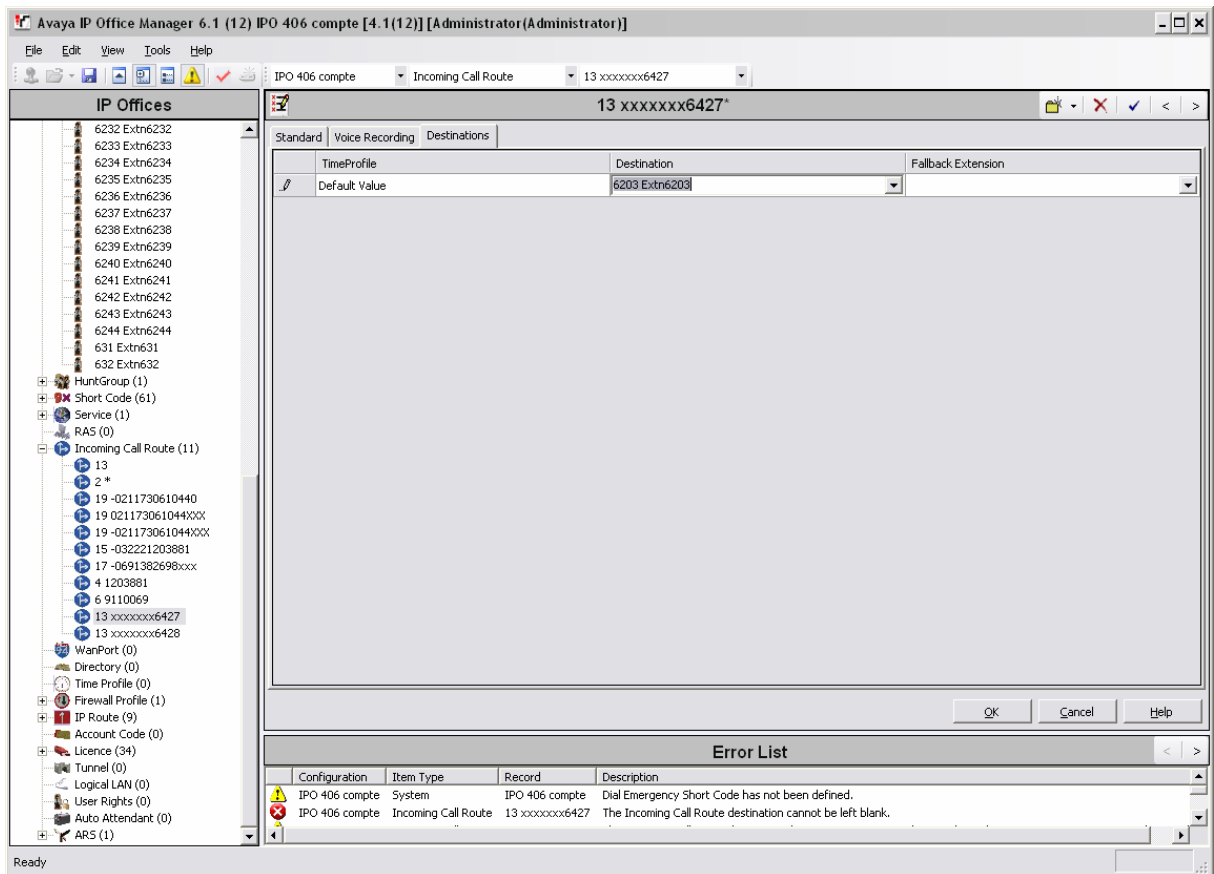
7. Configure Users SIP names with the authenticated DID. Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab all the way at the end.

> Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to one of the numbers provided by Club Communication**.** This valid number provided by Club Communication is used as authentication for calls coming into their VoIP network. This number should match the number used in the **Incoming Call Route**, see **Step 6**. These settings instruct the system to use this DID to construct the:
>
> - user part of the SIP URI in the From header of an outgoing SIP INVITE message
> - user part of the SIP URI in the Contact header of an outgoing SIP INVITE message
>
> Club Communications SIP equipment then verifies that the user part of the SIP URI in the INVITE's From header from IP Office is an authenticated user. (see Appendix: A for examples of SIP INVITE messages.)

> The other fields can be left as defaults. Press the **OK** button.
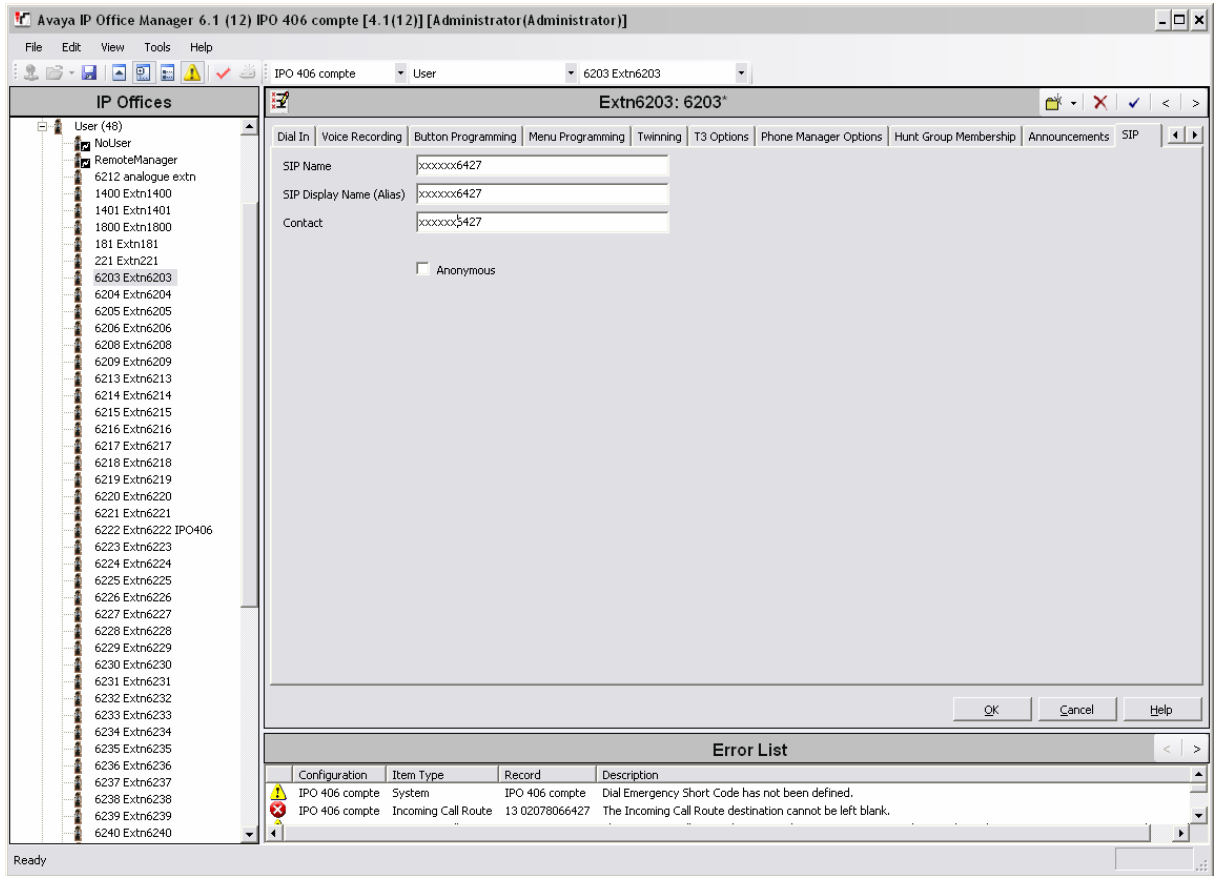
8. Repeat **Steps 6** and **7** for all users that will be sending/receiving external SIP calls on the system.

4. Club Communications Services Configuration

The following table contains the configuration information, provided by Club Communication, which was used during the interoperability compliance testing.

Service / Feature Information	Test Configuration
Club Communication Codec(s): • G.729a • G.711a-law • RFC2833 DTMF	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column.
Club Communication Dial Plan	Two 10-Digit numbers were supported by the test configuration.
Club Communication Direct Inward Dialing Numbers	Direct inward dialing numbers should be assigned to the endpoints at the enterprise site. This allows calls to be delivered from the PSTN. In this configuration, the provided numbers xxx xxx 6427 and xxx xxx 6428 were assigned to the H.323, digital, and analog endpoints in the enterprise network.
Club Communication Proxy IP Address	The IP address in the Club Communication network has to be masked to 0.0.0.0 masked for security purposes in this document.
Customer IP Address of Avaya IP Office	The IP address of Avaya IP Office in the enterprise network was 87.0.0.0 /29. This address must be masked. Club Communication used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya IP Office and Club Communication using UDP and port 5060.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Club Communications VoIP Network and an Avaya IP Office Telephony Solution. This section covers the general test approach and the test results.

5.1. General Test Approach

A simulated enterprise site consisting of an Avaya IP Office telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available DID, Toll Free and

emergency calls provided by Club Communication. This allowed the enterprise site to use SIP trunking for calls to the PSTN.

The following features and functionality were covered during the SIP trunking interoperability compliance test:
- Incoming calls to the enterprise site from the PSTN were routed to the numbers assigned by Club Communication.
- Outgoing calls from the enterprise site were completed via Club Communication to the PSTN destinations.
- Calls using H.323, digital and analog endpoints supported by the Avaya IP Office telephony solution.
- Various call types including: national, international, Toll Free, and emergency calls.
- Outgoing calls from IP Office using the G.729a codec type. Incoming calls can be used the G.729a or the G.711a codec type.
- Fax calls using the G.711a / T.38 codec. See note in Test Results section.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with the G.729a and G.711a codec.
- Telephone features such as hold, transfer, conference, call forwarding.

5.2. Test Results

Interoperability testing of the sample configuration was completed with successful results.

The following observations were noted.

- CLIP doesn't work to the German PSTN side. An international outgoing call from each endpoint will be always presented with the 0044xxx197009 number at the called party side. That is an incorrect number. It isn't the test number which was provided by Club Communication. The From header in the outgoing invite shows an appropriated number. The called party have to show the xxx6427 or the xxx6428 test number in the display.
  Additional information: If I make an outgoing call to another IP Office extension via sip trunk
  (dialling xxxx xxx 6428), then I have the Clip information correctly (xxxx xxx 6427).

- COLP doesn't work. When an incoming call is answered from another extension, the display of the calling party is not updated. When the call is answered there is no additional SIP signaling to update the display.

- COLR works, because COLP isn't working, but there isn't a signaling in SIP.

- AOC is not supported by Club Communication.

- T.38 for Fax isn't supported by the IP Office. Transmission of a Fax in G.711a is supported by the IP Office but Club Communication only supports G.711a / T.38. Therefore it isn't possible to transmit a fax.

- CLIR doesn't work. An outgoing anonymous call isn't possible. The outgoing invite contains "anonymous" in the From header, but the called party shows a number in the display.

6. Verification Steps

This section provides verification steps that may be performed to verify that the H.323, digital and analog endpoints can place outbound and receive inbound calls through Club Communications service.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers. Verify through the IP Office Monitor and System Trace tools that the call is using the appropriate SIP Line and that the proper SIP messages are exchanged.

2. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds. Verify through the IP Office Monitor and System Trace tools that the call is using the appropriate SIP Line and that the proper SIP messages are exchanged.

3. Verify that the user on the PSTN can terminate an active call by hanging up.

4. Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

7. Support

For technical support on Club Communication VoIP Services, contact the Club Communication Customer Center at 44 (0)1453 760240 or IPT@club-comms.co.uk.

8. Conclusion

These Application Notes describe the configuration steps required to connect customers using
an Avaya IP Office telephony solution to Club Communications VoIP Services. Club Communications offers a flexible VoIP solution for customers with a SIP based network. SIP
trunks use the Session Initiation Protocol to connect private company networks to the public
telephone network via converged IP access, providing an alternative to traditional hardwired
telephony trunk lines.

9. References

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

[1] IP Office 4.1 Installation Manual, October 2007
Document Number 15-601042

[2] IP Office 4.1 Manager:01. Using Manager, November 2007
Document Number 39DHB0002UKAA

[3] IP Office 4.1 Manager: 02. Configuration Settings, October 2007
Document Number 39DHB0002UKAB

[4] IP Office 4.1 Manager: 03. Short Codes, October 2007
Document Number 39DHB0002UKAC

[5] IP Office 4.1 Manager: 04. Telephony Features, October 2007
Document Number 39DHB0002UKAD

[6] 4600 Series IP Telephone R2.8 LAN Administrator Guide, February 2007, Issue 6,
Document Number 555-233-507

[7] Additional IP Office documentation can be found
http://marketingtools.avaya.com/knowledgebase/

Non-Avaya Documentation

[8] RFC 3261 SIP: Session Initiation Protocoll http://www.ietf.org/

[9] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
http://www.ietf.org/

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent between Club Communication and the Avaya IP Office Telephony Solution at the enterprise site. Customers may use these INVITE Messages for comparison and troubleshooting purposes. Differences in these messages may indicate different configuration options selected.

Sample SIP INVITE Message from Club Communication to Avaya IP Office:
```
INVITE sip:xxxxxxx6428@87.0.0.0:5060 SIP/2.0
Via: SIP/2.0/UDP 0.0.0.0:5060;branch=z9hG4bK643705386
From: "+49xxxxxxx" <sip:+49xxxxxx0@0.0.0.0>;tag=1563238898
To: <sip:xxxxxxx6428@87.0.0.0:5060>
Call-ID: 1323573902@0.0.0.0
CSeq: 20 INVITE
Contact: <sip:i@0.0.0.0>
Max-Forwards: 10
Allow: INVITE, ACK, BYE, CANCEL, INFO, UPDATE, REGISTER
Content-Type: application/sdp
Content-Length:   282

v=0
o=i 123456 654321 IN IP4 0.0.0.0
s=-
c=IN IP4 0.0.0.0
t=0 0
m=audio 21830 RTP/AVP 18 8 4 3 101
a=rtpmap:18 G729/8000/1
a=fmtp:18 annexB=no
a=rtpmap:8 PCMA/8000/1
a=rtpmap:4 G723/8000/1
a=rtpmap:3 GSM/8000/1
a=rtpmap:101 telephone-event/8000/1
a=fmtp:101 0-15
```

Sample SIP INVITE Message from Avaya IP Office to Club Communication:
```
INVITE sip:0049xxxxxx6898@00.00.00.00 SIP/2.0
Via: SIP/2.0/UDP 87.0.0.0:5060;rport;
     branch= z9hG4bK147adcf2cc94de254d5ac4ac546e762b
From: xxxxxxx6427 <sip:xxxxxxx6427@0.0.0.0>;tag=fe3fbd3289a396c9
To: <sip:0049xxxxxx6898@0.0.0.0>
Call-ID: 17dddc7e52ec90682eea944ef673d43b@87.0.0.0
CSeq: 2055300330 INVITE
Contact: xxxxxxx6427 <sip:xxxxxxx6427@87.0.0.0:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
```

Content-Type: application/sdp
Content-Length: 302

v=0
o=UserA 188836204 2883257809 IN IP4 87.0.0.0
s=Session SDP
c=IN IP4 87.0.0.0
t=0 0
m=audio 49152 RTP/AVP 8 18 4 0 101
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:0 PCMU/8000
a=fmtp:18 annexb = no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15