Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between TelePacific SmartVoice SIP Connect and an Avaya IP Office Telephony Solution – 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the TelePacific SmartVoice SIP Connect service and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

TelePacific offers a flexible VoIP solution for customers with a SIP based network. TelePacific Communications is a facility based competitive carrier that serves customers throughout California and Nevada. Headquartered in Los Angeles, the Company is the leading competitive carrier in its markets, with customer care centers in Los Angeles and Stockton in California and Las Vegas in Nevada.

TelePacific 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 TelePacific SmartVoice SIP Connect service and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

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Customers using the Avaya IP Office telephony solution with TelePacific's SmartVoice SIP Connect solution are able to place and receive PSTN calls using the SIP protocol via a dedicated broadband Internet connection. This converged network solution is an alternative to more traditional PSTN trunks such as T1 or ISDN PRI.

TelePacific can connect directly to an IP phone system as well as an external router. TelePacific's SmartVoice SIP Connect solution offers the following capabilities:

- Outbound domestic calling to local and long distance services
- Outbound international calling
- Incoming Direct Inward Dial (DID) service

Figure 1 illustrates a sample Avaya IP telephony solution connected to TelePacific's SmartVoice SIP Connect solution. This configuration was utilized for compliance testing.

Note: The IP addresses shown in **Figure 1** and throughout this document are not the actual public addresses used during testing. They are used as an example for these Application Notes. The real IP addresses are not revealed for security purposes.

The following equipment comprised the Avaya IP telephony solution and simulated a customer site:

- Avaya IP Office 406v2
- Avaya 4610SW IP Telephone (H.323)
- Avaya 6424D+M Digital Telephone
- Avaya 6210 Analog Telephone

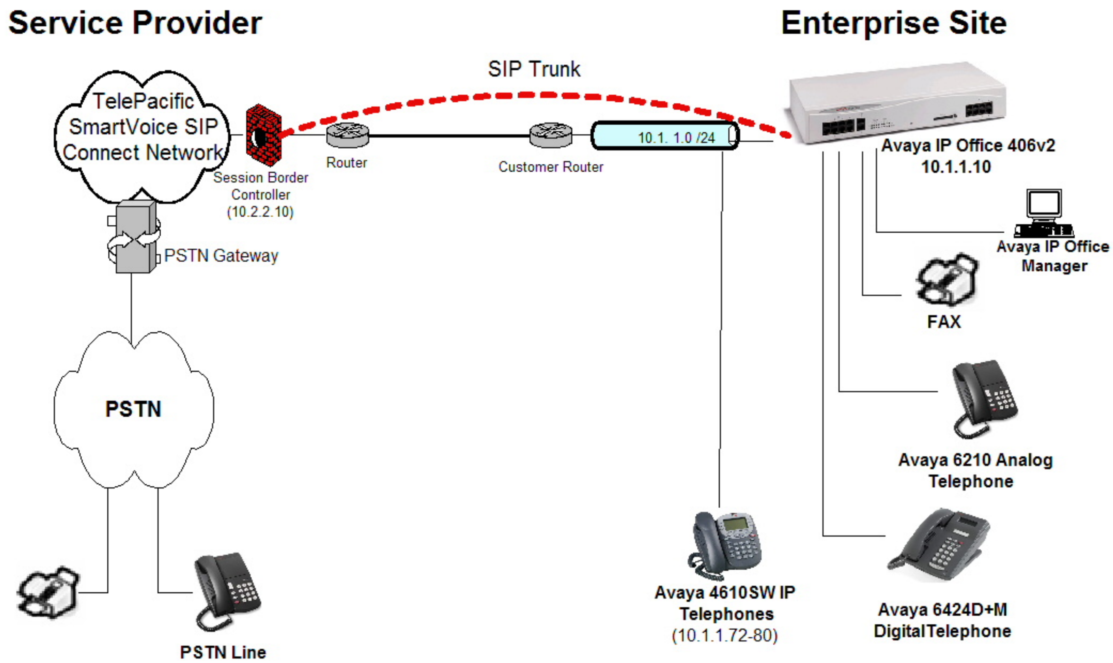


Figure 1: Avaya IP Telephony Network using TelePacific SIP Trunking Solution

2. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Avaya IP Office 406v2	R 4.0.10
Avaya IP Office Manager (Windows PC)	R 6.0 (10)
Avaya 4610SW IP Telephone	R2.3 – H.323 – a10d01b2_3.bin
Avaya 6424D+M Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
TelePacific SIP Trunking Solution Components	
SmartVoice SIP Connect	R1.0

Table 1: Equipment and Software Tested

This solution is compatible with all other Avaya IP Office platforms running Avaya IP Office software release 4.0.10.

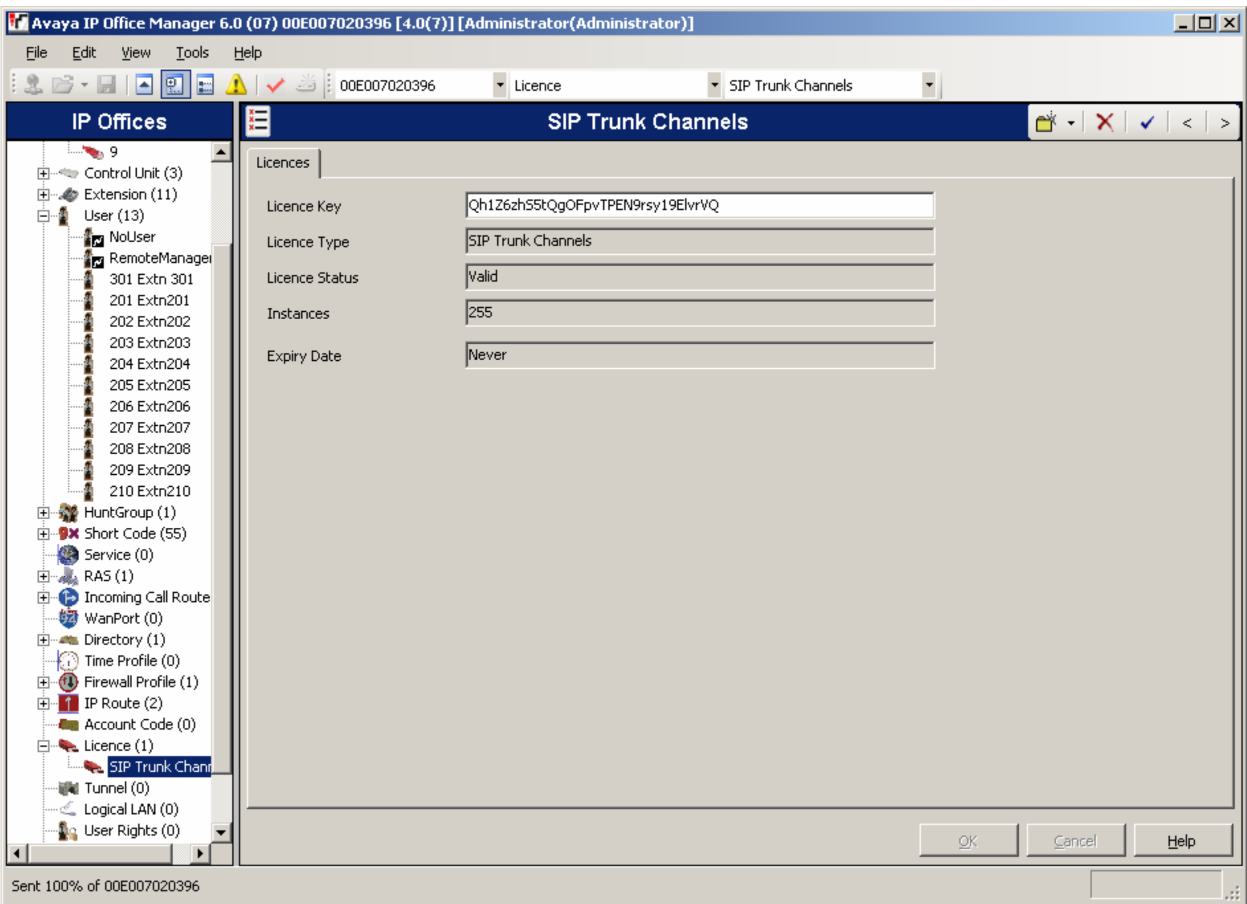
3. Configure Avaya IP Office

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

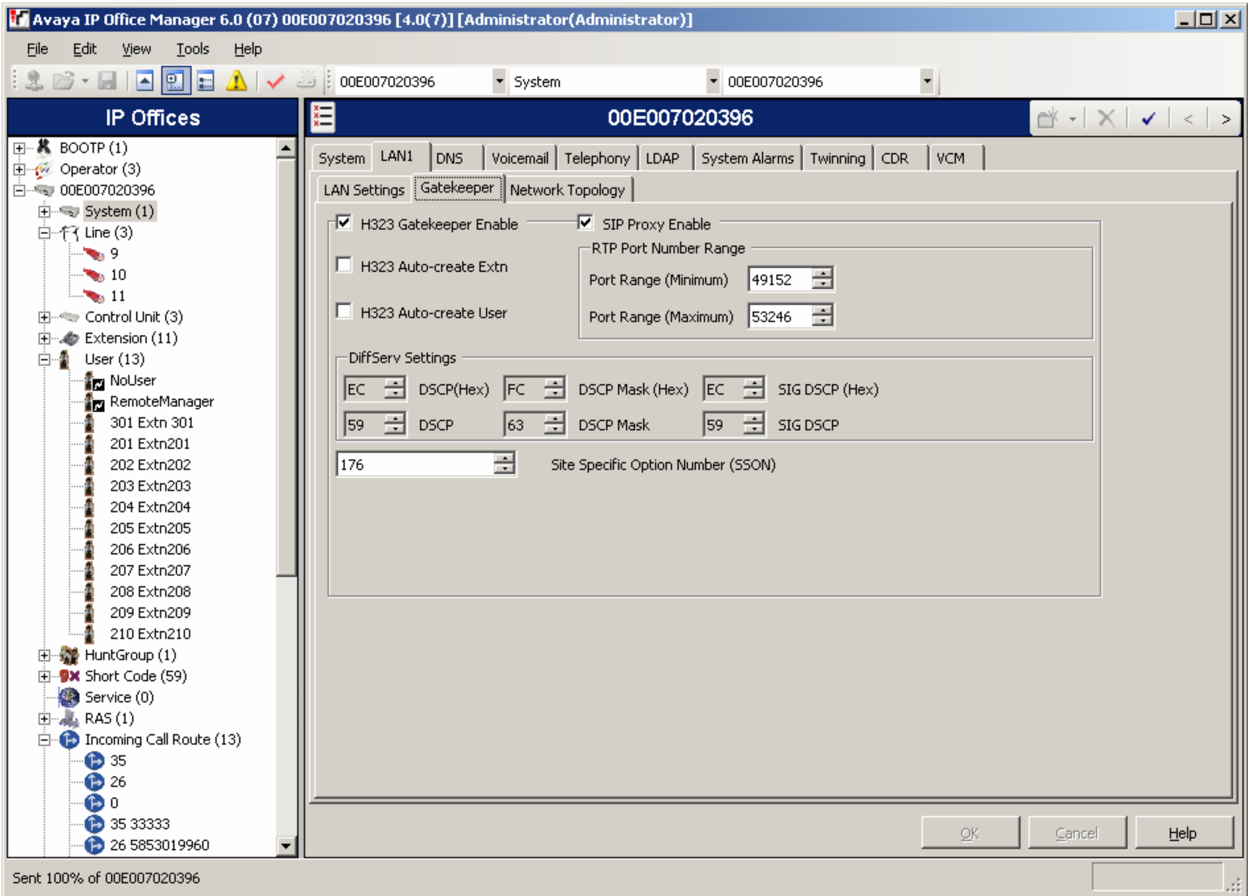
Avaya IP Office is configured via the Avaya IP Office Manager program. Log into the Avaya IP Office Manager PC, and select **Start → Programs → IP Office → Manager** to launch the Avaya IP Office Manager application. Log into the Avaya IP Office Manager application using the appropriate credentials.

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

   If the SIP Trunk Channels license is not enabled or there is insufficient capacity in the **Instances** field, contact an authorized Avaya sales representative to make the appropriate changes.



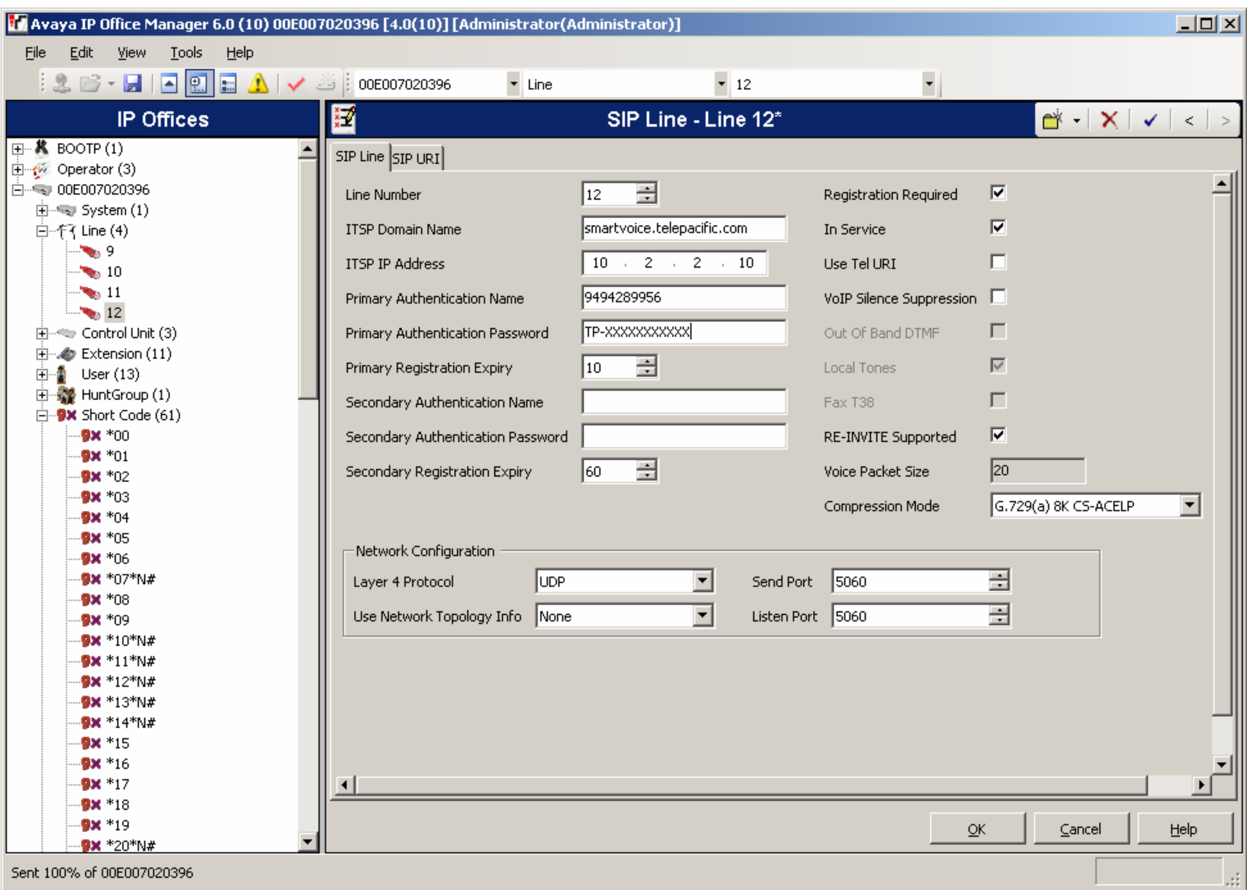
2. *Enable SIP Proxy Functionality*. Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper tab**, and check the **SIP Proxy Enable** box. All other fields can be left with their default settings.



3. *Create the SIP line for the TelePacific service*. Right-click Line in the left panel, and select **New → SIP Line**.

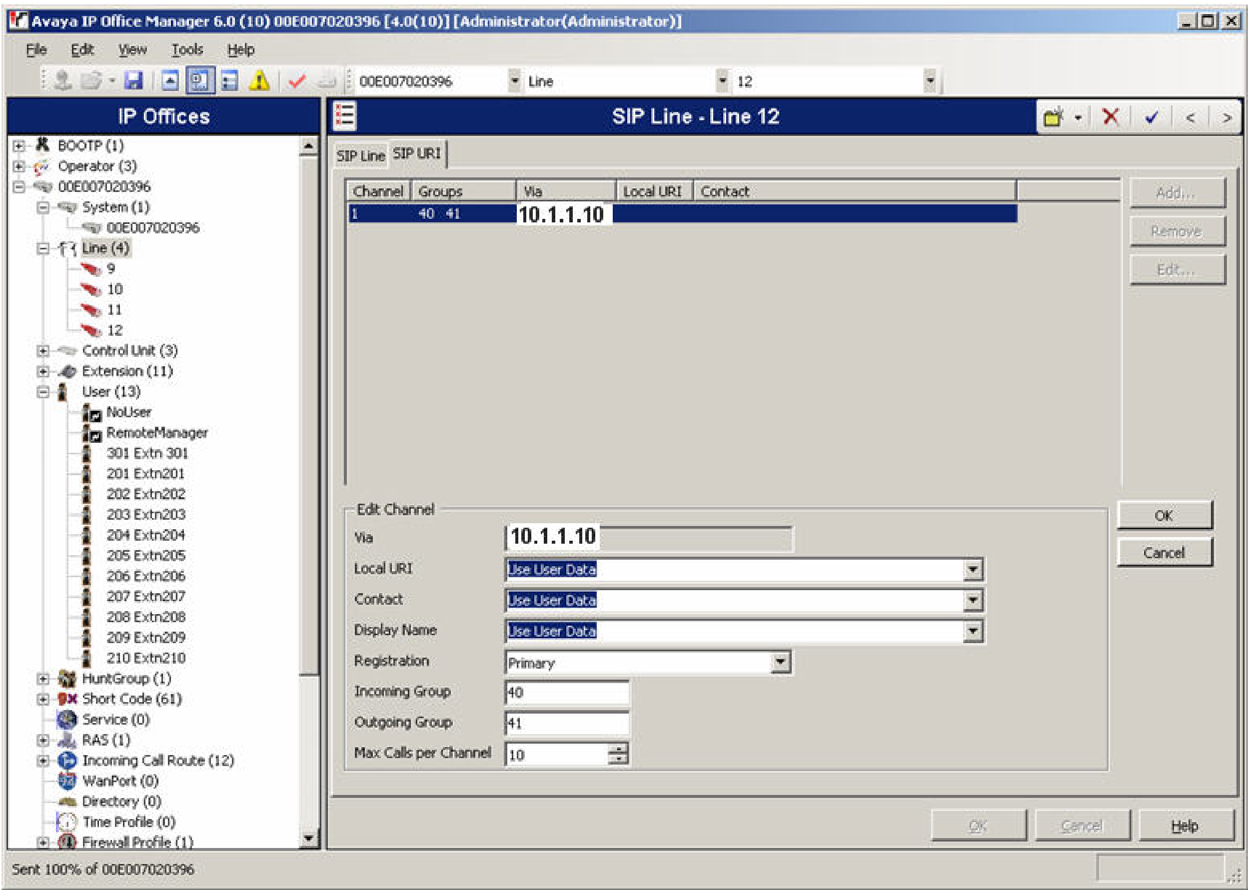
Select the following:

- For the **ITSP Domain Name** field, enter the TelePacific Domain Name. SIP registration messages from Avaya IP Office will use this name (please refer to the Installation Package provided by TelePacific for actual settings).
- For the **ITSP IP Address** field, enter the IP address of the TelePacific SIP Proxy
- For **Registration Required**, check the box to enable
- For **Primary Authentication Name**, enter the DID assigned by TelePacific
- For **Primary Authentication Password**, enter the password assigned by TelePacific
- For **Primary Registration Expiry**, set the value to according to TelePacific instructions
- For **Compression Mode**, select the **G729(a) 8K CS-ACELP** for all voice calls. (To ensure proper interoperability between Avaya IP Office and TelePacific for voice calls, G.729a is required. G.711MU is reserved for Fax calls only.)
- For **Layer 4 Protocol**, use **UDP**
- For **Send Port**, use **5060**
- For **Listen Port** use **5060**
- Use defaults for other fields



4. *Configure URI parameters for the line*. Select the **SIP URI** tab. Press the **Add** button.

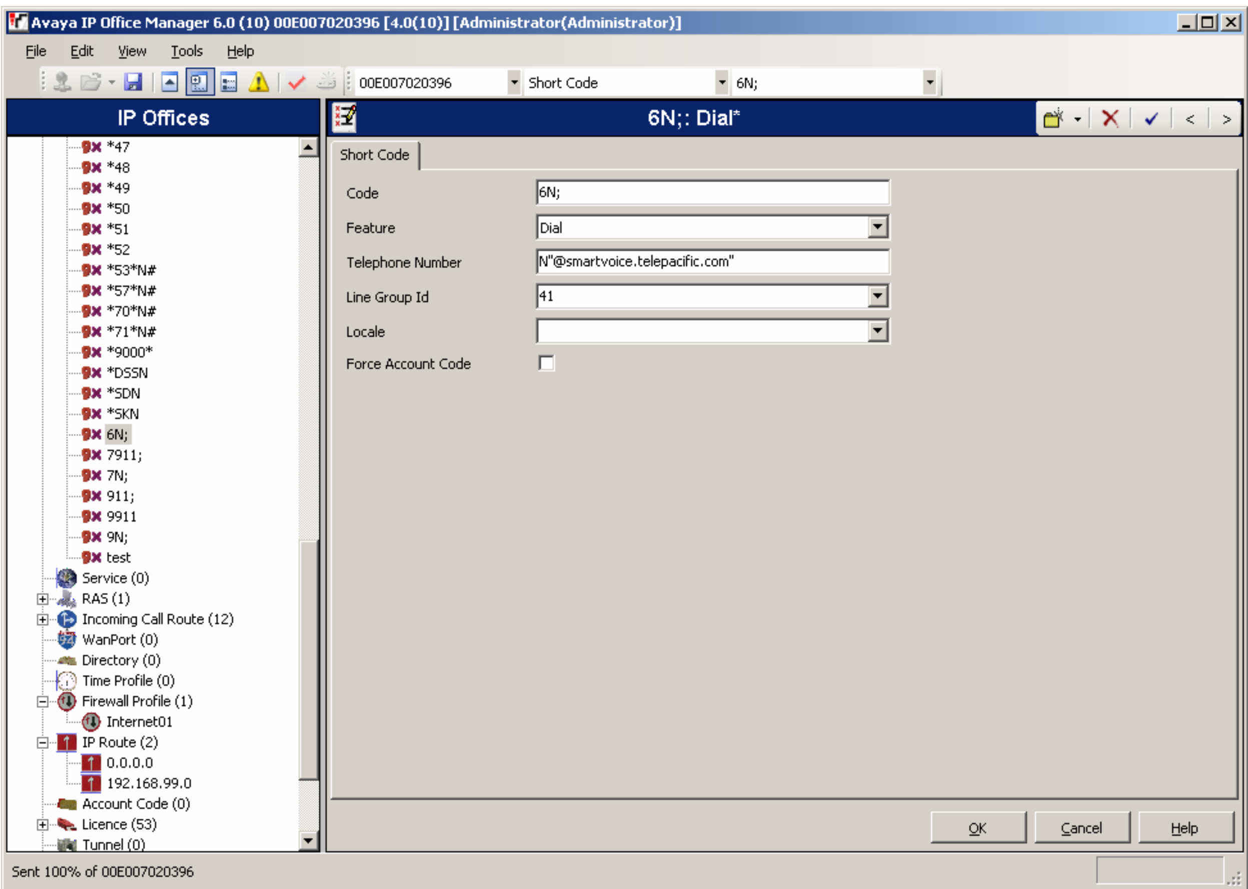
   Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. Select **Use User Data** for the **Contact**, **Local URI** and **Display Name** fields. This tells the system to use the information configured on the **SIP** tab for each individual **User** (see **Step 7**). Use defaults for all other fields. Press the **OK** button.



5. *Configure a short code to route calls to TelePacific's SIP Proxy Server*. Select **Short Code** in the left panel. Right click and select **Add**. Enter **[x]N;**, where **[x]** is a valid number, in the **Code** text box. The number **6** is used for **[x]** in the below example. This code requires the user to dial the digit **6** 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 Avaya IP Office extension. For example, a 10-digit DID number, operator assistance, 411, information service etc.

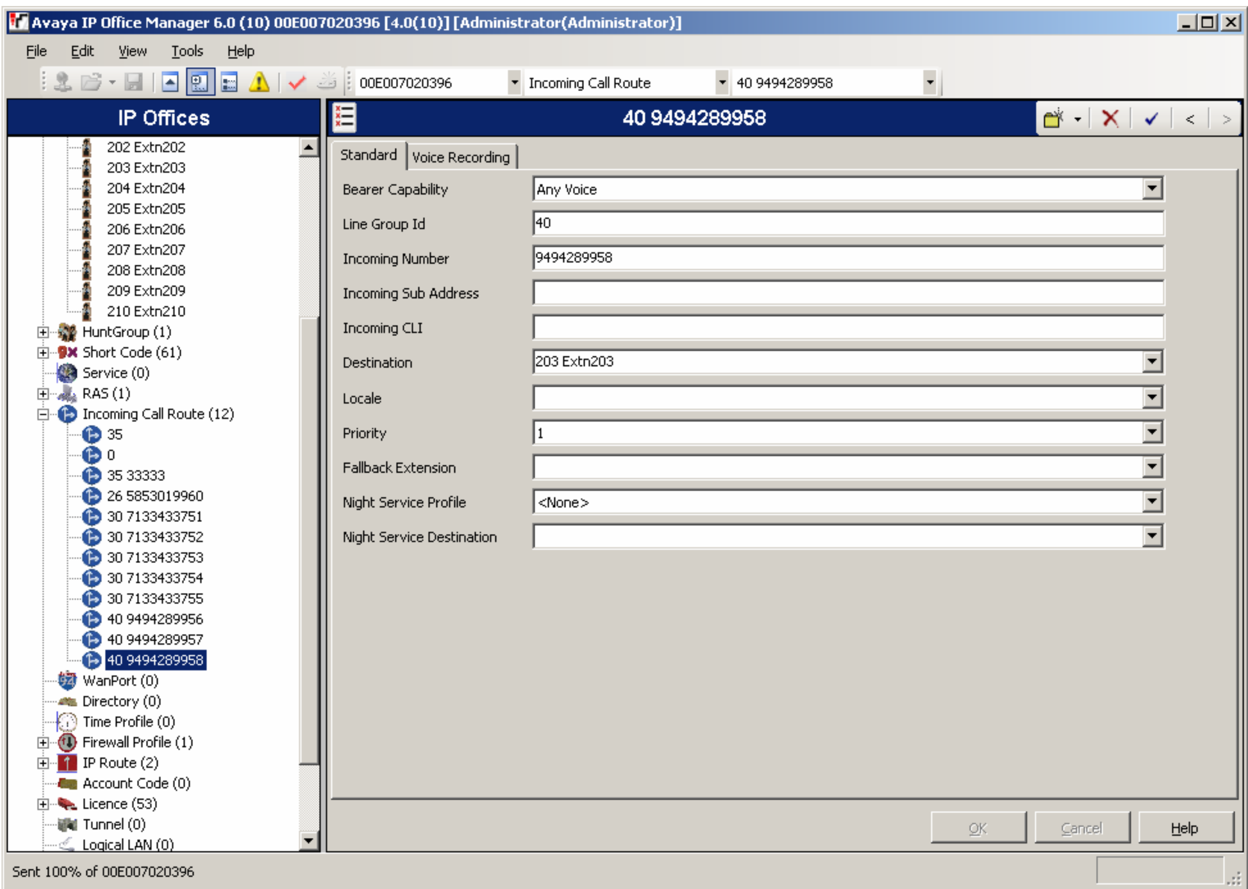
   Select **Dial** for the **Feature** field. Enter the **Outgoing Group** created in **Step 4** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<Domain Name of TelePacific >"** for the **Telephone Number** field. The **Telephone Number** field is used to construct the To 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:

- **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 DID provided by TelePacific, that is mapped back to a local Avaya IP Office extension, in the **Incoming Number** field.
- In the **Destination** field, select the desired local extension number from the drop down menu.
- Use default values for all other fields. Press the **OK** button.

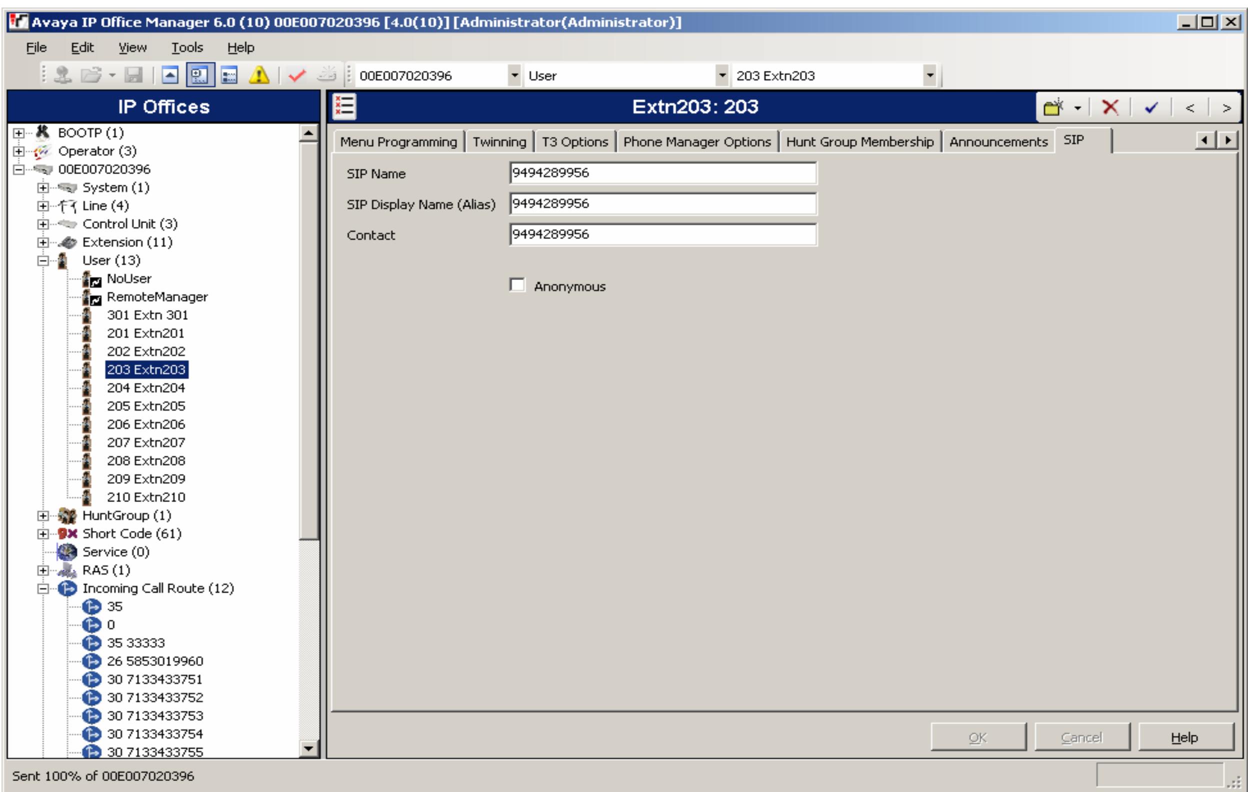


7. *Configure Users' SIP names with the authenticated DID.* Select **User** in the left panel. Select the desired user by double-clicking on an entry in the left panel. Select the **SIP** tab all the way at the end.

   Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to the **Primary Authentication Name** defined in **Step 3.** This name is a valid DID number provided by TelePacific that is used as the authentication name for registration to their SIP Registrar. Since only registered numbers are authorized to place calls through TelePacific's network and Avaya IP Office can only register one user per trunk, all Avaya IP Office Users must use the same authenticated number for the above parameters in order to place outbound calls via TelePacific's service. 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

   TelePacific's SIP equipment then verifies that the user part of the SIP URI in the INVITE's From header from Avaya 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 SIP calls on the system.

4. TelePacific Services Configuration

Service is ordered through a TelePacific Account Manager or through a TelePacific Agent. If someone wishes to initiate contact with TelePacific and does not already have an established relationship with an Account Manager, visit http://www.telepacific.com or call 800-399-4925 for more details.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between TelePacific Service 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 SmartVoice SIP Connect solution provided by TelePacific. 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 DID numbers assigned by TelePacific.
- Outgoing calls from the enterprise site were completed via TelePacific to the PSTN destinations.
- Calls using H.323, digital and analog endpoints supported by the Avaya IP Office telephony solution.
- Various call types including: local, long distance, international, emergency and directory assistance calls.
- Calls using the G.729a codec.
- Fax calls using the G.711MU codec.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with the G.729a codec.
- Telephony features such as hold, transfer and conference.

6.2. Test Results

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

The following observations were noted.

- For the Current Release R 4.0.10, only G.729a should be used if DTMF tones are required. As stated in **Step 3 Section 3**, G.729a should be used for all voice calls.
- The full SIP URI is displayed on incoming calls. When an inbound call is placed into Avaya IP Office, the full SIP URI is seen on the telephone's call display instead of just the calling party number.

7. 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 TelePacific's 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.

8. Support

All TelePacific customers have 24 hours access to live technical support. Customer Care may be reached by calling 877-487-TPAC or visiting:
http://www.telepacific.com/contact/customerService.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to TelePacific SmartVoice SIP Connect service. TelePacific 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.

10. 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 4.0 Applications Installation and Administration*, February 2007, Document Number 15-601133

[2] *IP Office 4.0 Manager: 01. Using Manager,* February 2007, Document Number N/A

[3] *IP Office 4.0 Manager: 02. Configuration Settings,* February 2007, Document Number 39DHB0002UKAB

[4] *IP Office 4.0 Manager: 03. Short Codes*, February 2007, Document Number 39DHB0002UKAC

[5] *IP Office 4.0 Manager: 04. Telephony Features,* February 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 at:
http://marketingtools.avaya.com/knowledgebase/

Non-Avaya Documentation:

[8] RFC 3261 *SIP: Session Initiation Protocol* 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 by TelePacific and the Avaya SIP network 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 TelePacific to Avaya IP Office:

INVITE sip:9494289956@10.1.1.10:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bK3306m1001g6h2c8bi4g0.1
From: "AVAYA INC   "<sip:7324500819@10.255.224.100;user=phone>;tag=359664349-1193322795049-
To: "9494289958 9494289958"<sip:9494289958@smartvoice.telepacific.com>
Call-ID: BW073315049251007-1741689833@10.255.224.100
CSeq: 734685205 INVITE
Contact: <sip:7324500819@10.2.2.10:5060;transport=udp>
Contact Binding: <sip:7324500819@10.2.2.10:5060;transport=udp>
Supported: 100rel
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,UPDATE,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Max-Forwards: 9
Content-Type: application/sdp
Content-Length: 303
P-Media-Release: hngl5rp9pu6vivh05pvumhj1r8jof8do6u7g8gsnmkmj2c4kl2v1004092

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): BroadWorks 43581 1 IN IP4 10.2.2.10
Session Name (s): -
Connection Information (c): IN IP4 10.2.2.10
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 52160 RTP/AVP 18 0 2 4 100
Media Attribute (a): rtpmap:18 G729/8000/1
Media Attribute (a): fmtp:18 annexb=no
Media Attribute (a): rtpmap:0 PCMU/8000/1
Media Attribute (a): rtpmap:2 G726-32/8000/1
Media Attribute (a): rtpmap:4 G723/8000/1
Media Attribute (a): rtpmap:100 telephone-event/8000
Media Attribute (a): fmtp:100 0-15
Media Attribute (a): sendrecv

Sample SIP INVITE Message from Avaya IP Office to TelePacific:

INVITE sip:17324500819@smartvoice.telepacific.com SIP/2.0
Via: SIP/2.0/UDP 10.1.1.10:5060;rport;branch=z9hG4bK90bc60dba8836714a2f69f33885d56c3
From: 9494289956 <sip:9494289956@smartvoice.telepacific.com>;tag=3725ff8a0ba3762a
To: <sip:17324500819@smartvoice.telepacific.com>
Call-ID: 4aab26304f1ce3b62c2d9b57994a29bc@10.1.1.10
CSeq: 1976233742 INVITE
Contact: 9494289956 <sip:9494289956@10.1.1.10:5060;transport=udp>
Contact Binding: 9494289956 <sip:9494289956@10.1.1.10:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE
Content-Type: application/sdp
Content-Length: 300

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): UserA 314176889 1247500738 IN IP4 10.1.1.10
Session Name (s): Session SDP
Connection Information (c): IN IP4 10.1.1.10
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 49152 RTP/AVP 18 4 8 0 101
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): fmtp:18 annexb = no
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.