Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Sotel IP Services SIP Edge Advanced SIP Trunking Solution and an Avaya IP Office Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Sotel IP Services SIP Edge Advanced Trunking solution and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, Avaya H.323, digital, and analog endpoints.

Headquartered in Maryland Heights, Missouri, Sotel IP Services provides Internet Protocol (IP) telecommunications services worldwide. Enterprise customers with an Avaya IP telephony SIP-based network can connect to the Sotel IP Services VoIP Network over the Internet and access the PSTN by subscribing to Sotel IP Services SIP Edge Advanced SIP product.

Sotel IP Services 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 Sotel IP Services SIP Edge Advanced SIP Trunk service and an Avaya IP telephony solution. The Avaya solution consists of Avaya SIP Enablement Services, Avaya Communication Manager, and various Avaya H.323, digital, and analog endpoints.

Customers using this Avaya IP telephony solution with Sotel IP Services SIP Edge Advanced product are able to place and receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI.

The text and coverage diagram below summarizes the Sotel IP Services SIP Edge Advanced trunk Service at the time of writing these Application Notes. Please consult Sotel IP Services for the most current description of capabilities.

Sotel IP Services SIP Edge Advanced solution provides SIP Trunking voice services including DID, Out-bound, Long Distance, and Toll-Free. 24 X 7 Support is provided by a dedicated account team. On-Line tools are available to customers providing secure access and fast turn around. Sotel IP Services is co-located on the Level (3) network, is E911 compliant, provides flexible billing options, and provides customized pricing to meet Enterprise requirements.

Sotel IP Services SIP Edge Advanced product offers the following:

- Tier 1 pricing without commitments
- Toll quality voice
- Sotel is co-located with level 3
- Over 5 nines of availability since inception
- Includes LNP (Local Number Portability)
- Well defined change management process

Figure 1 illustrates a sample Avaya IP telephony solution connected to Sotel IP Services SIP Edge Advanced trunking solution. This configuration was utilized for compliance testing.

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

- Avaya IP Office 500
- Avaya IP 400 Phone Expansion Module
- Avaya 4610SW IP Telephone (H.323 protocol)
- Avaya 6424D+M Digital Telephone
- Avaya 2420 Digital Telephone
- Avaya 6210 Analog Telephone

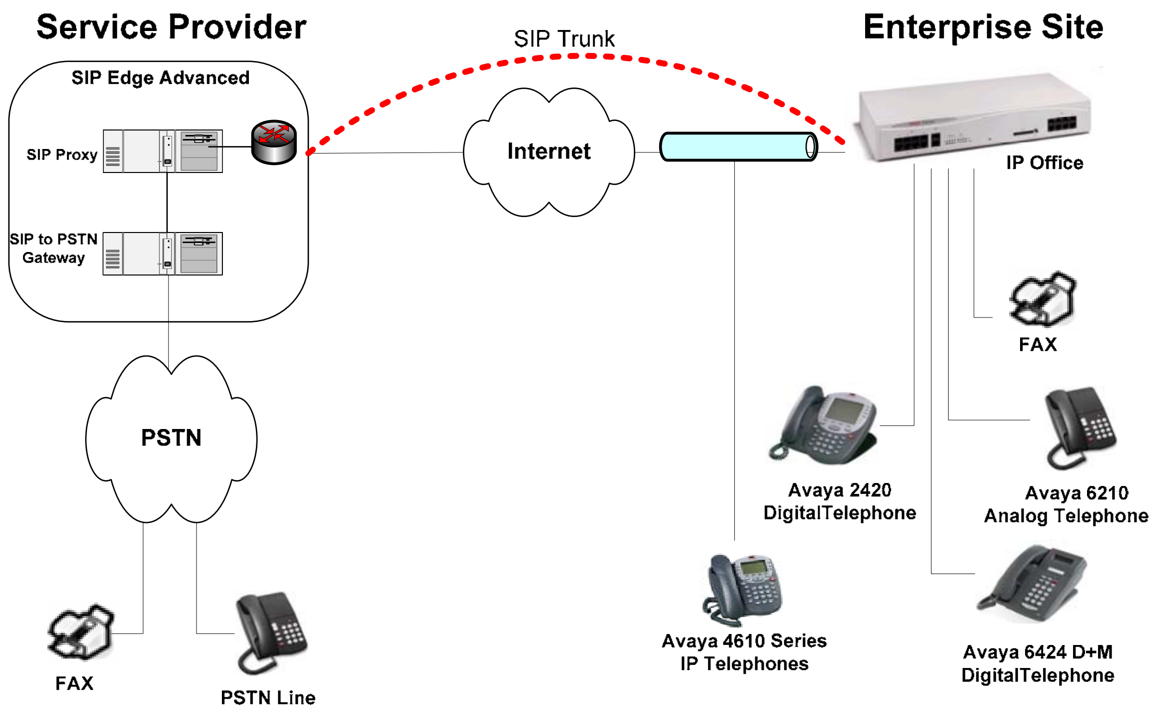


Figure 1: Avaya IP Telephony Network using Sotel IP Services SIP Edge Advanced Trunking Solution

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 500	R 4.2 (49501)
Avaya IP 400 Analog POTS 30+	R 6.2 (4)
Avaya IP Office Manager (Windows PC)	R 6.2 (4)
Avaya IP Office Voicemail Pro	R 4.2.19
Avaya 4610SW IP Telephone	R2.8.3 – H.323
Avaya 6424D+M Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Avaya 2420	R5 Firmware
Sotel IP Services SIP Edge Advanced Trunking Solution Components	
CarrierClass.net Release 1.1	R1.1

Table 1: Equipment and Software Tested

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.2 (49501).

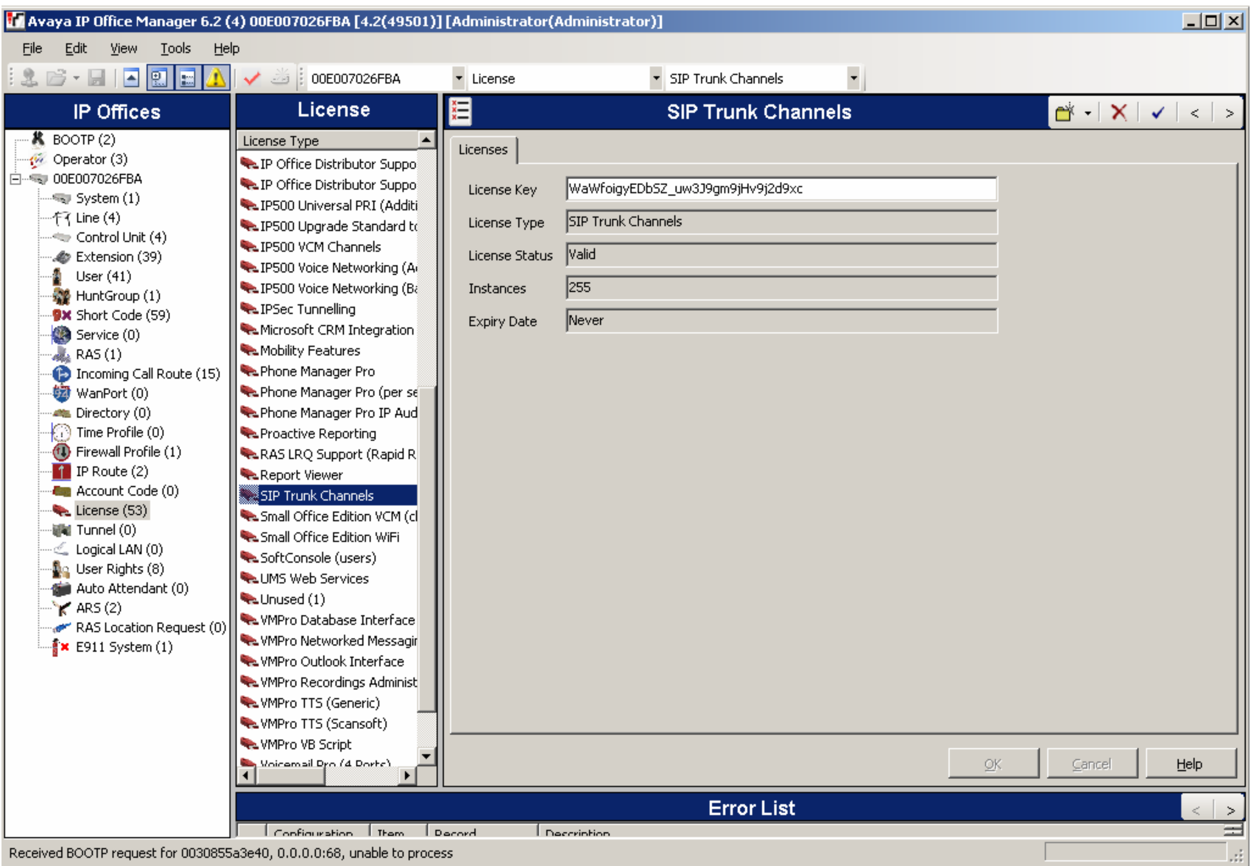
3. Configure Avaya IP Office

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

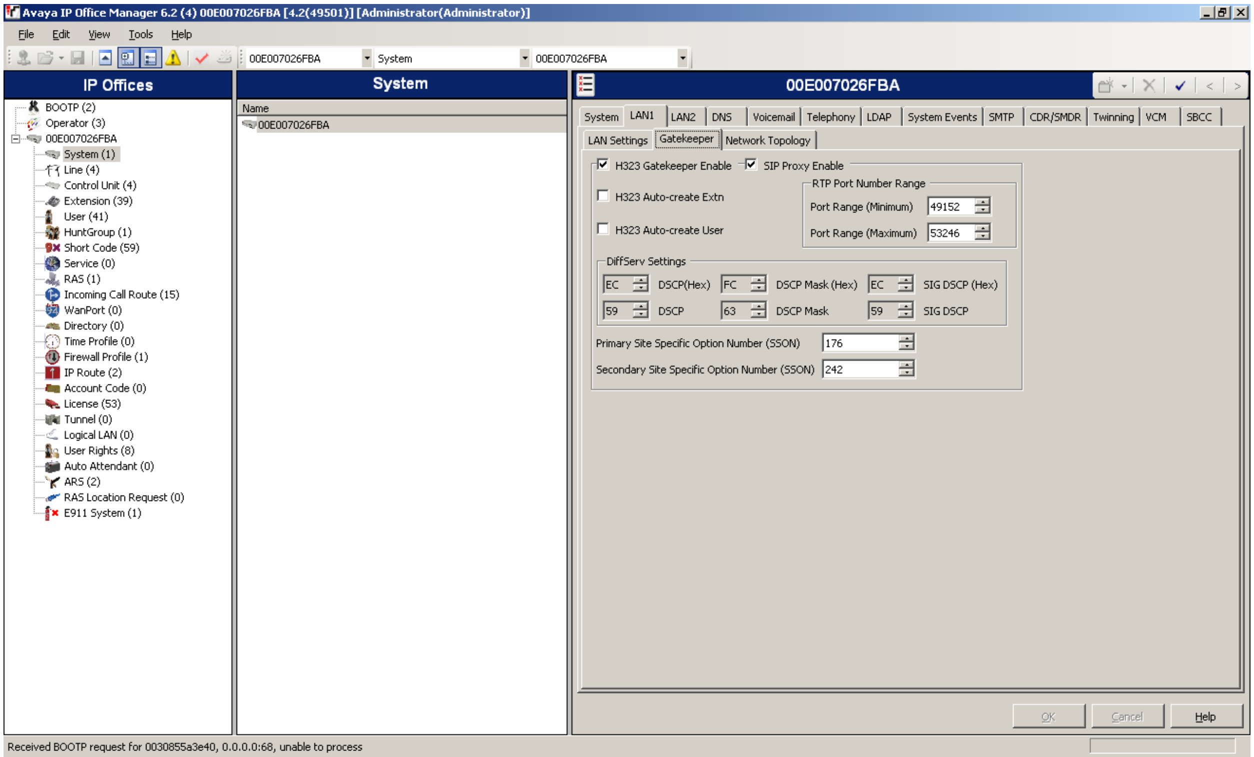
IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

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

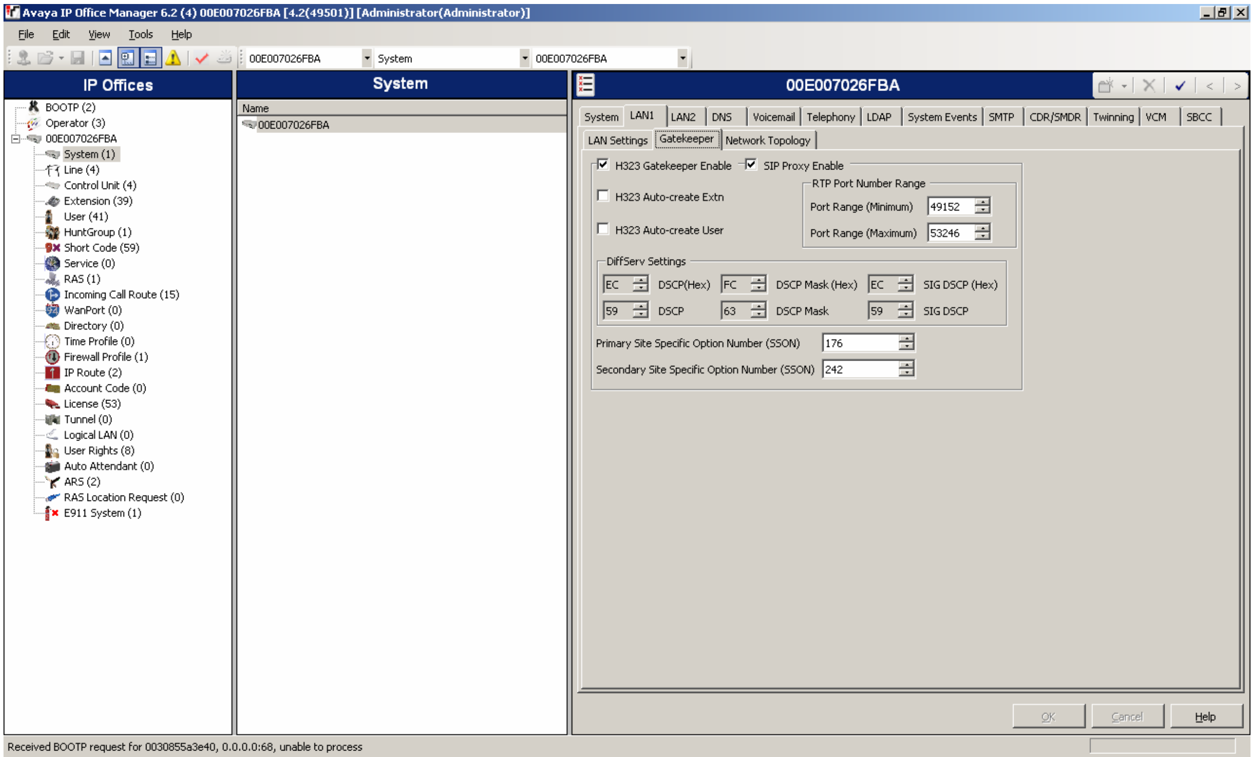
   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



2. *Enable SIP Proxy Functionality*. Select **System** in the left panel. Click the **LAN1** tab. In the **LAN1** tab, select the **Gatekeeper tab,** and check the **SIP Proxy Enable** box. Click the **OK** button.



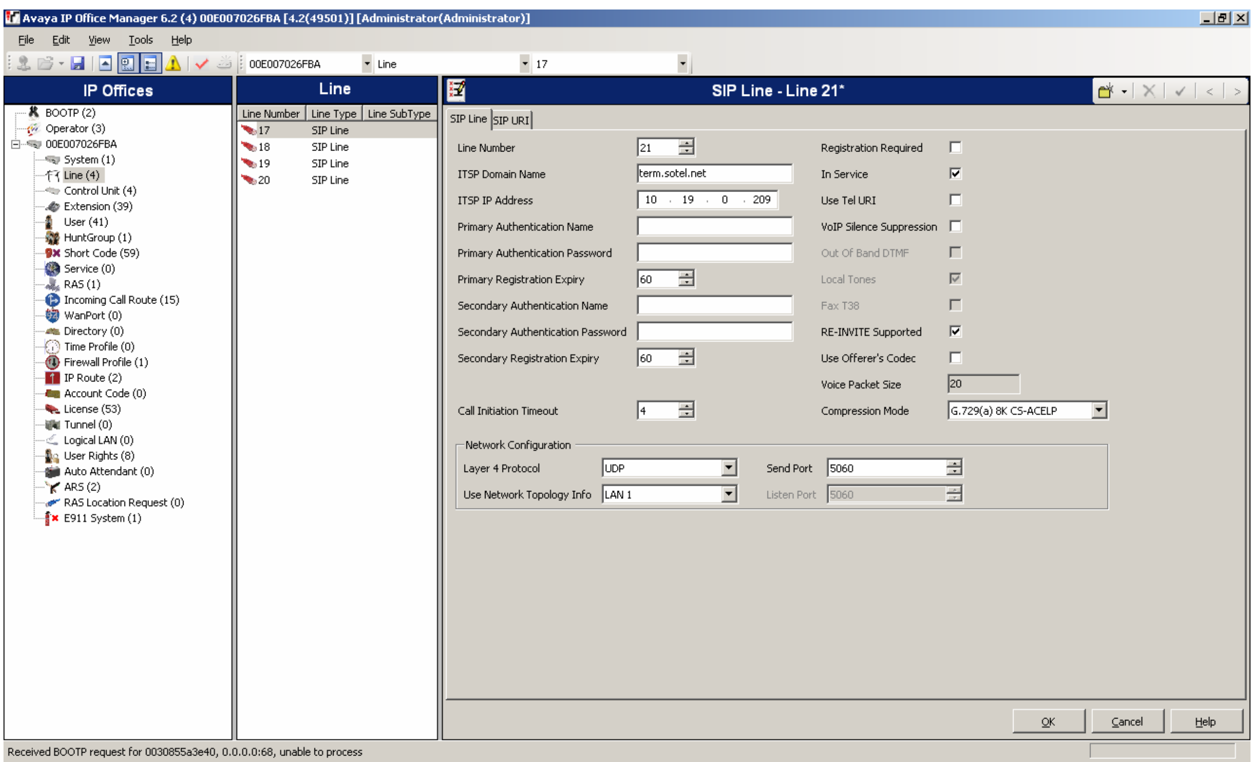
3. *Configure DiffServ Settings* according to Sotel IP Services requirements. Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper** tab (same as mentioned in step 2). Under **DiffServ Settings**, enter **EC** into the **DSCP(Hex)** and **SIG DSCP(Hex)** text boxes by clicking the increment tab. These are the values used during the test. Click the **OK** button.



4. *Create the SIP line* for the Sotel IP Services SIP Edge Advanced service. Select **Line** in the left panel. Right-click and select **New → SIP Line**.

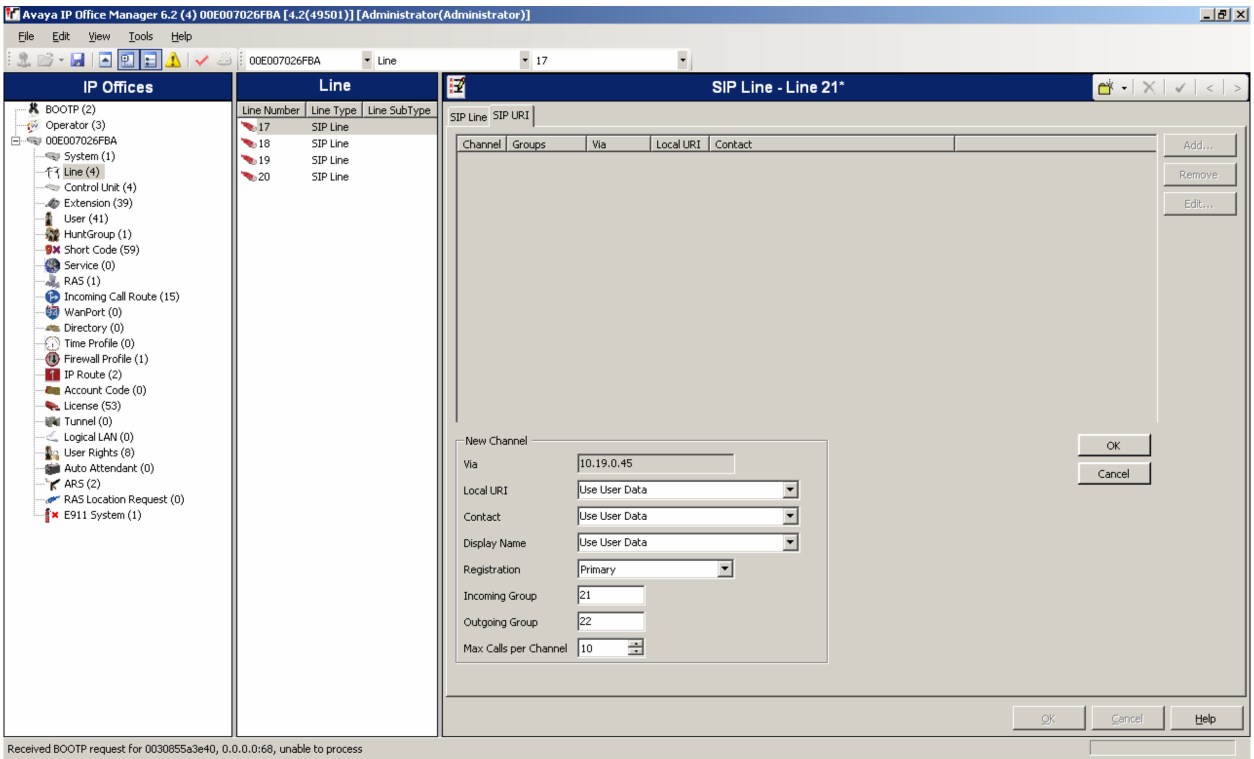
Select the following:

- For the **ITSP Domain Name field**, enter the Sotel IP Services Domain Name. SIP registration messages will use this name. Sotel IP Services will provide the actual settings
- For the **ITSP IP Address field**, enter the IP address of the Sotel IP Services SIP Proxy
- For **Registration Required**, check the box to enable
- For **Primary Authentication Name**, use the Trunk ID assigned by Sotel IP Services
- For **Primary Authentication Password**, use the password assigned by Sotel IP Services
- For **Compression Mode**, select the **G729a 8K CS-ACELP** or **G.711 ULAW 64K** for voice calls. (** NOTE ** If FAX is to be used, **G.711 ULAW 64K** must be selected for proper operation)
- For **Layer 4 Protocol**, use **UDP**
- For **Send Port** and **Listen Port**, use **5060**
- For **Line Network Topology Info** use **LAN 1**
- Use defaults for other fields
- Click the **OK** button



5. *Configure URI parameters for the line*. Select the **SIP URI** Tab. Click the **Add** button.

Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations configured in **Step 8**. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in **Step 7.** Select **Use User Data** for the **Local URI, Contact**, and **Display Name** fields. This tells the system to use the information configured on the SIP tab for each individual user (see **Step 9**). Use defaults for all other fields. Click the **OK** button.

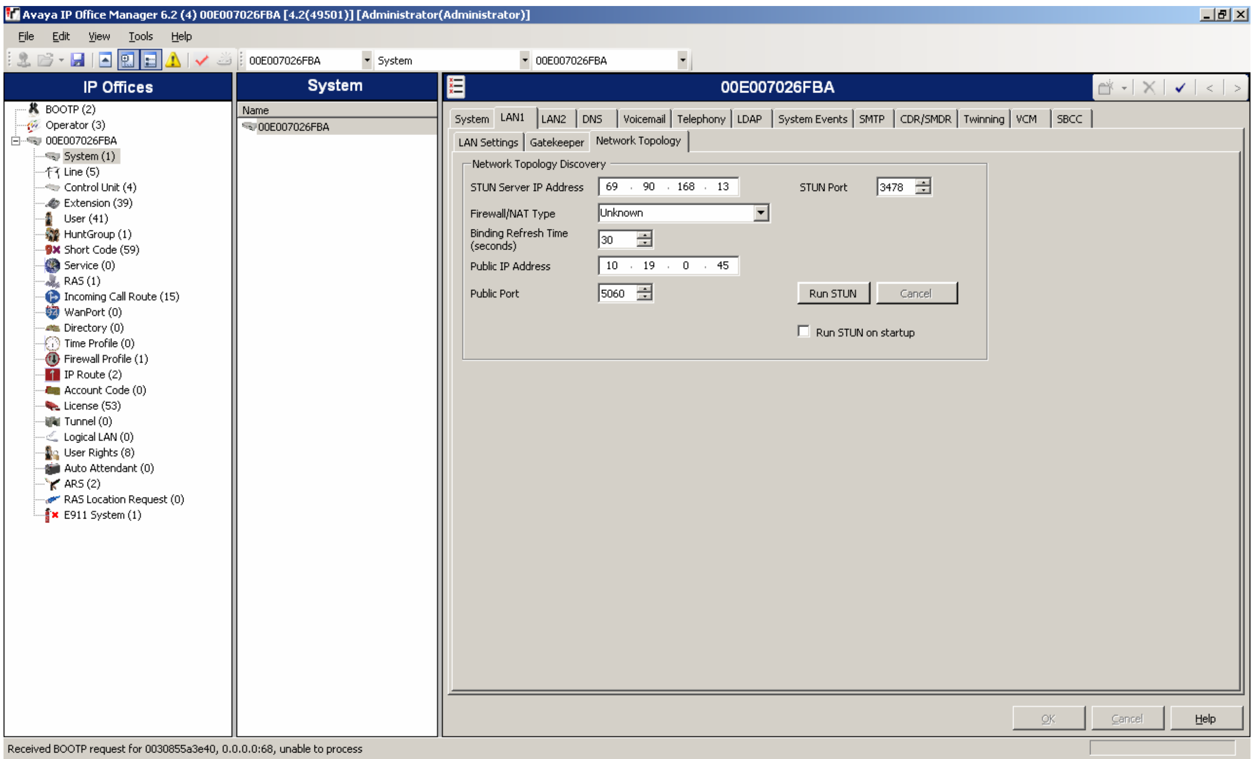


Currently, Sotel IP Services uses two border elements. A second Line is required to be configured containing the second border element's IP address. Other than the new line number and ITSP IP Address, all other administration for the SIP line and SIP URI are identical to the previously administered SIP line and SIP URI.

6. *Configure SIP OPTIONS timer on Network Topology Tab for "keep alive" function with Sotel IP Services.* Select **System** in the left panel. In the **LAN1** tab, select the **Network Topology** tab.

   Set the **Binding Refresh Time** to the desired interval which determines the frequency with which OPTIONS messages will be sent to Sotel IP Services SIP proxy. For **Public IP Address**, enter the Avaya IP Office system IP address. Confirm that **Public Port** is set to 5060 and take defaults for all other fields.

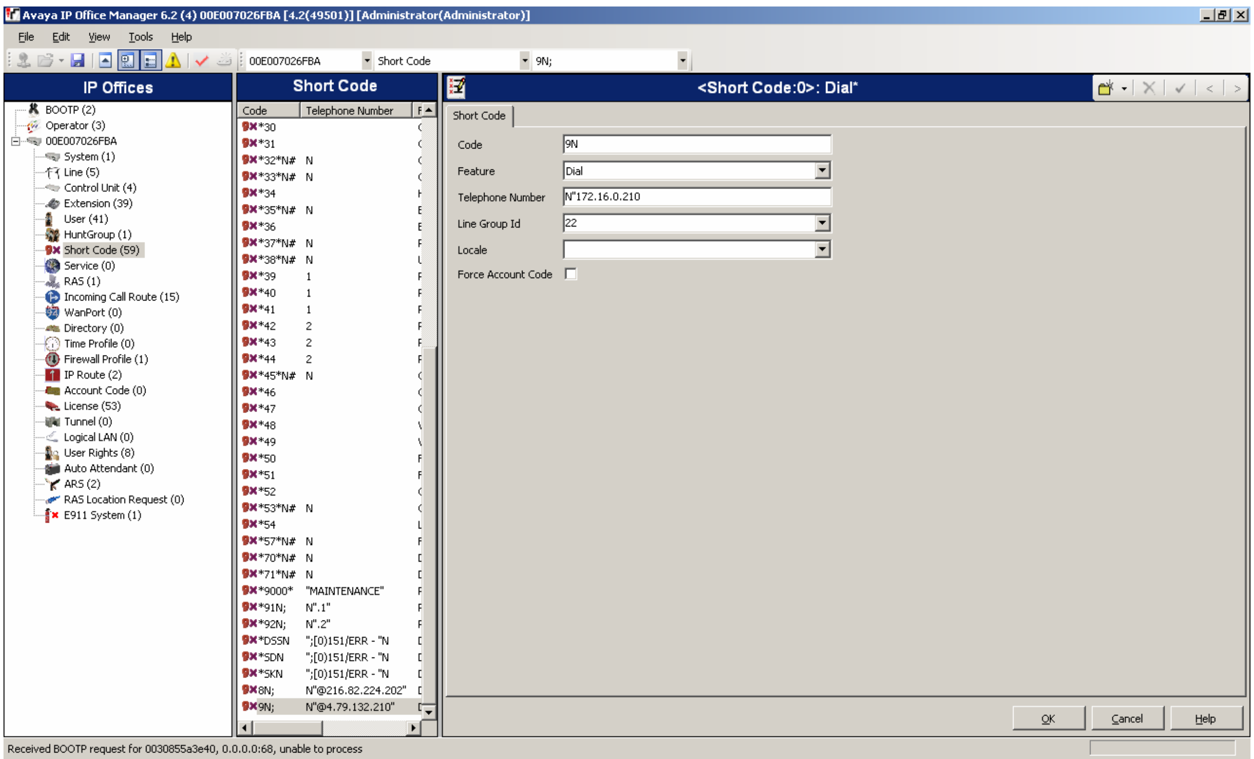
   Note: Depending upon what type of firewall or Network Address Translation device is being used at the customer premise, it may be necessary to set the **Firewall/NAT Type** setting to the appropriate setting as defined [2].



7. *Configure a short code to route calls to Sotel IP Services 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 **9** is used for **[x]** in the below example. This code requires the user to dial the digit **9** 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. For example, a 10-digit number, a toll free number, directory assistance (e.g: 411), information service etc.

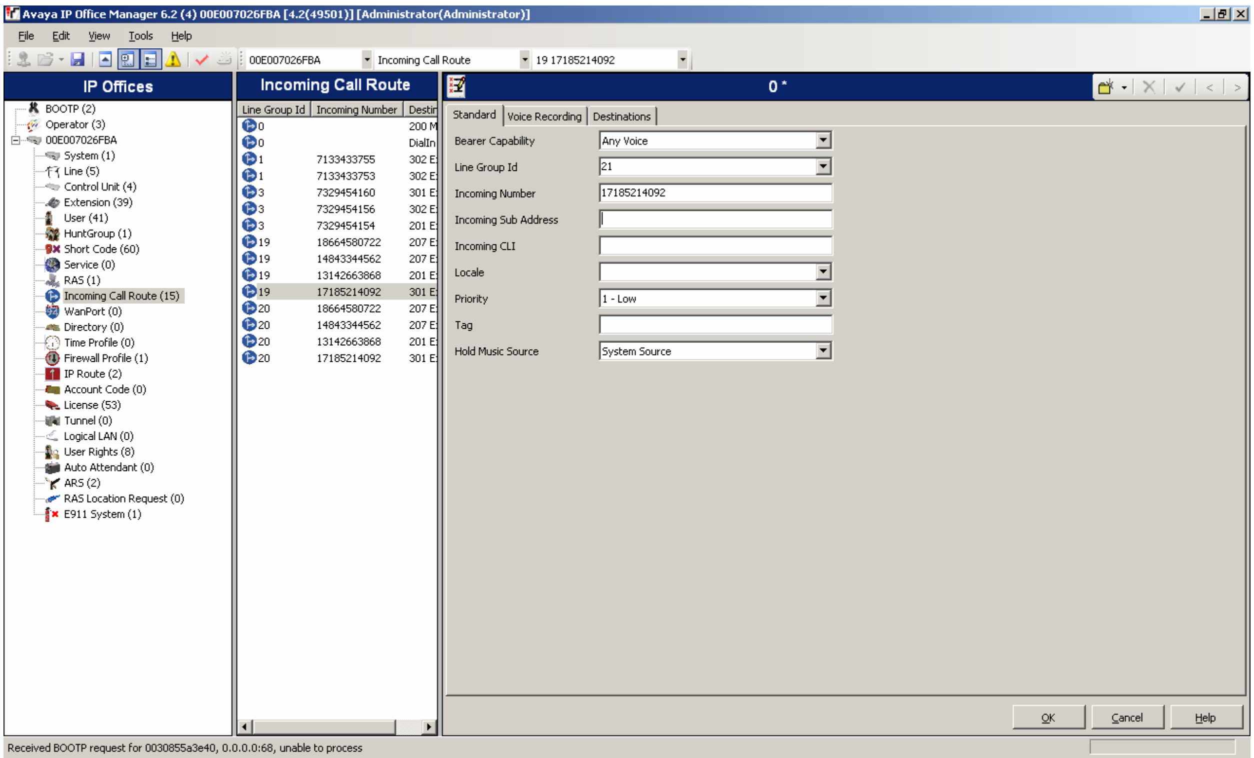
   Select **Dial** for the **Feature**. Enter the **Outgoing Group Id** created in **Step 5** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<Domain Name of** Sotel IP Services **>"** for the **Telephone Number** field. The **Telephone Number** field is used to construct 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. Click the **OK** button.



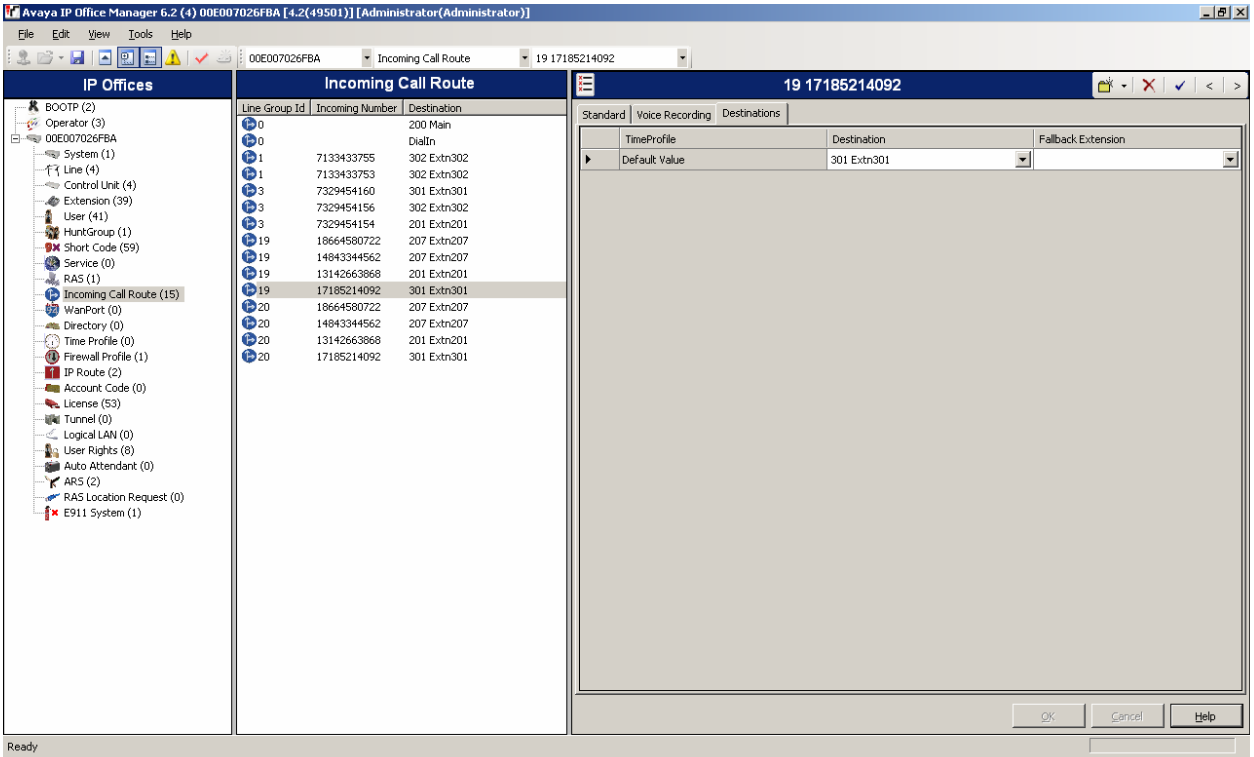
8. *Create an Incoming Call Route for the Inbound SIP calls*. Select **Incoming Call Route** in the left panel. Right-click and select **New**.

Enter the following:

- **Any Voice** for the **Bearer Capability** field
- The Incoming Group created for the URI in **Step 5** in the **Line Group Id** field
- The 10 digit DID provided by Sotel IP Services, 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 and select the desired local extension number from the drop down menu
- Click the **OK** button

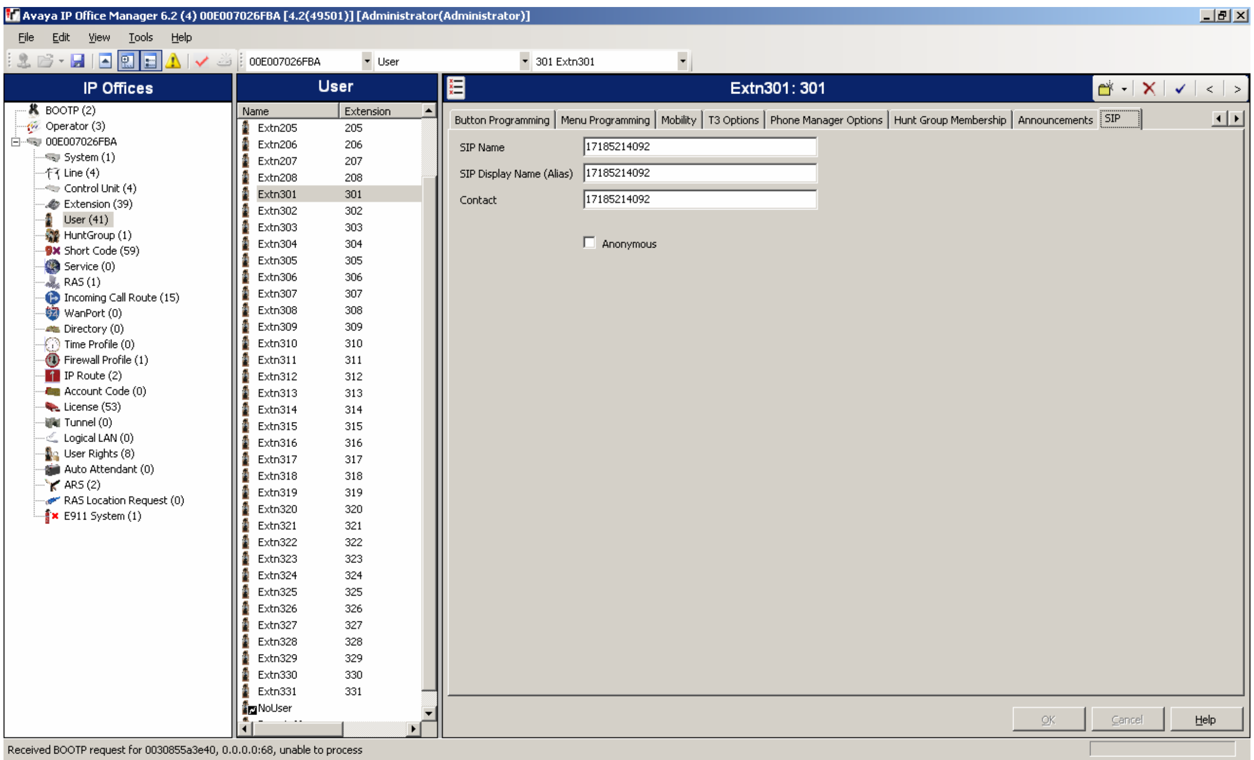


9. *Configure Users' SIP names*. Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab all the way at the end.

   Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to the DID number provided by Sotel IP Services that is used for this particular extension. 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

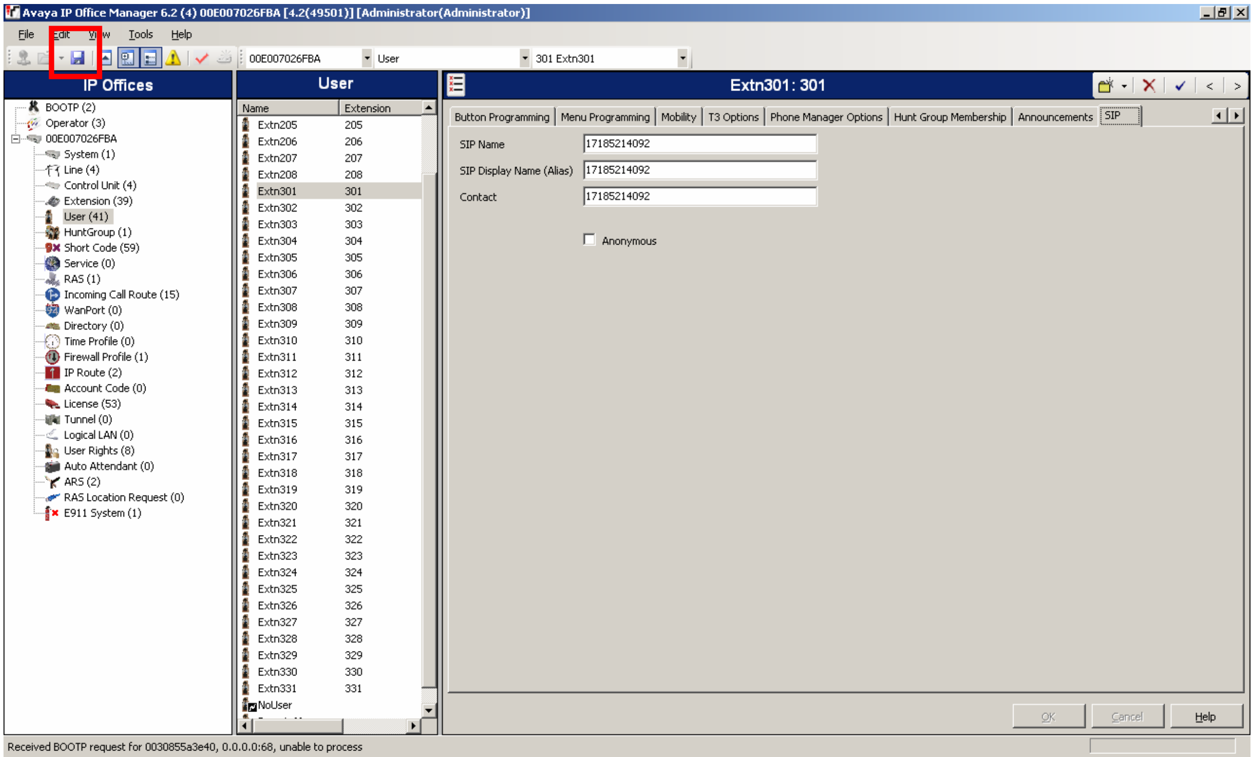
(See Appendix A for examples of SIP INVITE messages.)

The other fields can be left as defaults. Click the **OK** button.



10. Repeat **Steps 8** and **9** for all users that will be sending/receiving SIP calls on the system.

11. After making the changes, click on the floppy disk icon (3rd from left) to push the changes to the IP Office and have them take effect. **Changes will not take effect till this step is completed. ** NOTE ** This may cause a reboot of Avaya IP Office causing service disruption.**



4. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Sotel IP Services SIP Edge Advanced trunking service and an Avaya IP Office Telephony Solution. This section covers the general test approach and the test results.

4.1. 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 SIP Trunking solution provided by Sotel IP Services. 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 Sotel IP Services
- Outgoing calls from the enterprise site were completed via Sotel IP Services SIP Edge Advanced trunking solution to 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, toll free, and directory assistance calls

- Calls using the G.729a and G.711Mu-Law codecs
- Fax routing to ensure G.711 Mu-Law use for fax calls
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with G.729a
- Telephone features such as hold, transfer, and conference

5. Test Results

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

The following observations were noted:

1. For **Compression Mode**, either **G.729a 8K CS-ACELP** or **G.711 ULAW 64K** can be selected for voice calls. However, if FAX is to be used, **G.711 ULAW 64K** must be selected for faxing to work properly. Failure to select G.711 ULAW 64K will result in unsuccessful fax operation.

2. When calling a party that does not answer, Sotel IP Services will send a cancel after 60 seconds and the call will be terminated.

3. Calls were not made to emergency services (911) but this is claimed to be supported by Sotel IP Services.

4. If "anonymous" is selected on the station's SIP page and a call is made to the PSTN, the call will fail. This is due to Sotel IP Service requiring the From username to be numeric. According to Sotel, this is not an issue to their current subscribers.

5. When a call is put on hold there is no indication sent via SIP message to the far end. While on hold, both parties send silence in the RTP stream. This is transparent to the user.

6. If the G.723.1 6K3 MP-MLQ codec is selected, it will not be offered in the SIP INVITE. This will be fixed in a future release of Avaya IP Office.

7. If an inbound call over the SIP trunk terminates to a station that forwards their calls to a station in the PSTN over the SIP trunk, the call alerts but there is no audio after the call is answered. This will be fixed in a future release of Avaya IP Office.

8. When calling internationally, ring back was heard in approximately 20% of the calls. This is thought to be caused by Sotel IP Services service provider using different PSTN routes for the calls. Sotel IP Services is investigating this with their service provider.

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 Sotel IP Services 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.

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.

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 Sotel IP Services SIP Edge Advanced Trunk services, contact Sotel IP Services Customer Service by utilizing the contact information found on the following web link: https://sotelips.net/d/?q=support or by calling 1-877-MY-SOTEL (1-877-697-6835).

8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to the Sotel IP Services SIP Edge Advanced trunking service. Sotel IP Services 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 4.2 Installation Manual, Issue 19f*, August 2008
Document Number 15-601042

[2] *IP Office 4.2 Manager 6.2, Issue 22K*, September 2008
Document Number 15-601011

[3] *4600 Series IP Telephone LAN Administrator Guide*, July 2008, Issue 8, Document Number 555-233-507

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

Non-Avaya Documentation:

[5] RFC 3261 *SIP: Session Initiation Protocol* http://www.ietf.org/

[6] 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 Sotel IP Services 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 Sotel IP Services to Avaya IP Office:

```
No.     Time        Source                Destination           Protocol Info
   1876 11.989561   172.16.0.210          10.19.0.45          SIP/SDP  Request: INVITE
sip:17185214092@10.19.0.45, with session description

Frame 1876 (871 bytes on wire, 871 bytes captured)
Ethernet II, Src: Cisco_91:fd:51 (00:18:18:91:fd:51), Dst: NetworkA_02:6f:ba
(00:e0:07:02:6f:ba)
Internet Protocol, Src: 172.16.0.210 (172.16.0.210), Dst: 10.19.0.45 (10.19.0.45)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:17185214092@10.19.0.45 SIP/2.0
    Message Header
        Record-Route: <sip:172.16.0.210:5060;nat=yes;ftag=VPSF50603522629670;lr=on>
        Via: SIP/2.0/UDP 172.16.0.210;branch=z9hG4bK1015.3703f5e5.0
            Transport: UDP
            Sent-by Address: 172.16.0.210
            Branch: z9hG4bK1015.3703f5e5.0
        Via: SIP/2.0/UDP 192.168.0.38:5060;branch=z9hG4bK50603522629670-1192652871567
            Transport: UDP
            Sent-by Address: 192.168.0.38
            Sent-by port: 5060
            Branch: z9hG4bK50603522629670-1192652871567
        From: <sip:17328521640@192.168.0.38>;tag=VPSF50603522629670
            SIP from address: sip:17328521640@192.168.0.38
            SIP tag: VPSF50603522629670
        To: <sip:17185214092@172.16.0.210:5060>
            SIP to address: sip:17185214092@172.16.0.210:5060
        Call-ID: NYCMGC0120081014150734064242@192.175.63.25
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        Contact: <sip:+17328521640@192.168.0.38:5060;transport=udp>
            Contact Binding: <sip:+17328521640@192.168.0.38:5060;transport=udp>
        Max-Forwards: 68
        Content-Type: application/sdp
        Content-Length: 173
        Remote-Party-ID:
<sip:+17328521640@192.168.0.38>;party=calling;screen=no;privacy=off
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1223996854 1223996855 IN IP4
192.168.5.218
                Owner Username: -
                Session ID: 1223996854
                Session Version: 1223996855
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 192.168.5.218
            Session Name (s): -
            Connection Information (c): IN IP4 192.168.5.218
```

```
        Connection Network Type: IN
        Connection Address Type: IP4
        Connection Address: 192.168.5.218
    Time Description, active time (t): 0 0
        Session Start Time: 0
        Session Stop Time: 0
    Media Description, name and address (m): audio 60702 RTP/AVP 0 18 101
        Media Type: audio
        Media Port: 60702
        Media Proto: RTP/AVP
        Media Format: ITU-T G.711 PCMU
        Media Format: ITU-T G.729
        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
```

Sample SIP INVITE Message from Avaya IP Office to Sotel IP Services:

```
No.     Time        Source                Destination           Protocol Info
    150 8.007466    10.19.0.45          172.16.0.209          SIP/SDP  Request: INVITE
sip:7328521640@172.16.0.210, with session description

Frame 150 (828 bytes on wire, 828 bytes captured)
Ethernet II, Src: NetworkA_02:6f:ba (00:e0:07:02:6f:ba), Dst: Cisco_91:fd:51
(00:18:18:91:fd:51)
Internet Protocol, Src: 10.19.0.45 (10.19.0.45), Dst: 172.16.0.209 (172.16.0.209)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:7328521640@172.16.0.210 SIP/2.0
    Message Header
        Via: SIP/2.0/UDP
10.19.0.45:5060;rport;branch=z9hG4bK0d6e7825d69a273a18eb7cc8837dcb55
            Transport: UDP
            Sent-by Address: 10.19.0.45
            Sent-by port: 5060
            RPort: rport
            Branch: z9hG4bK0d6e7825d69a273a18eb7cc8837dcb55
        From: "Drudge" <sip:17185214092@term.sotel.net>;tag=9580dffdf777e635
            SIP Display info: "Drudge"
            SIP from address: sip:17185214092@term.sotel.net
            SIP tag: 9580dffdf777e635
        To: <sip:7328521640@172.16.0.210>
            SIP to address: sip:7328521640@172.16.0.210
        Call-ID: 4215c0eb1141271ba7cef1676e2364eb@10.19.0.45
        CSeq: 2107491922 INVITE
            Sequence Number: 2107491922
            Method: INVITE
        Contact: "Drudge" <sip:Hank@10.19.0.45:5060;transport=udp>
            Contact Binding: "Drudge" <sip:Hank@10.19.0.45:5060;transport=udp>
                URI: "Drudge" <sip:Hank@10.19.0.45:5060;transport=udp>
                    SIP Display info: "Drudge"
                    SIP contact address: sip:Hank@10.19.0.45:5060
        Max-Forwards: 70
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
        Content-Type: application/sdp
        Content-Length: 275
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): UserA 4024978640 52933316 IN IP4 10.19.0.45
                Owner Username: UserA
                Session ID: 4024978640
                Session Version: 52933316
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 10.19.0.45
            Session Name (s): Session SDP
            Connection Information (c): IN IP4 10.19.0.45
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 10.19.0.45
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 49152 RTP/AVP 0 18 8 101
                Media Type: audio
                Media Port: 49152
```

```
    Media Proto: RTP/AVP
    Media Format: ITU-T G.711 PCMU
    Media Format: ITU-T G.729
    Media Format: ITU-T G.711 PCMA
    Media Format: 101
Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 0
    MIME Type: PCMU
Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 18
    MIME Type: G729
Media Attribute (a): rtpmap:8 PCMA/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 8
    MIME Type: PCMA
Media Attribute (a): fmtp:18 annexb = no
    Media Attribute Fieldname: fmtp
    Media Format: 18 [PCMA]
    Media format specific parameters: annexb = no
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
```

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.