Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring SIP Trunking Using PAETEC Communications Dynamic IP SIP Trunk Service 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 PAETEC Communications Dynamic IP SIP Trunk Service and an Avaya IP Office telephony solution. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a Genband platform in the network. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital, and analog endpoints.

PAETEC 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 PAETEC Communications Dynamic IP SIP Trunk Service and an Avaya IP Office telephony solution. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a Genband platform in the network. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital, and analog endpoints.

PAETEC can offer the Dynamic IP SIP Trunk Service using several different platform technologies in the PAETEC network. These Application Notes correspond to the Dynamic IP SIP Trunk Service offered using a Genband platform in the network.

Customers using this Avaya IP telephony solution with the PAETEC Dynamic IP SIP Trunk Service 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 PAETEC Dynamic IP SIP Trunk Service at the time of writing these Application Notes. Please consult PAETEC for the most current description of capabilities. PAETEC serves 82 of the top 100 Metropolitan Statistical Areas, and offers data, voice, and value-added services throughout the United States. From local and long distance to VoIP, PAETEC offers a full spectrum of traditional and next-generation voice services, each predicated on vast industry expertise and the world-class technology of partners.

PAETEC Dynamic IP SIP Trunk Service includes the following capabilities:

- Outbound PSTN calling to local, long distance and international services
- Incoming Direct Inward Dial (DID) service
- Incoming Toll-free service
- Operator, Directory Assistance and Calling Card Service
- Converged IP access via a private IP MPLS Network

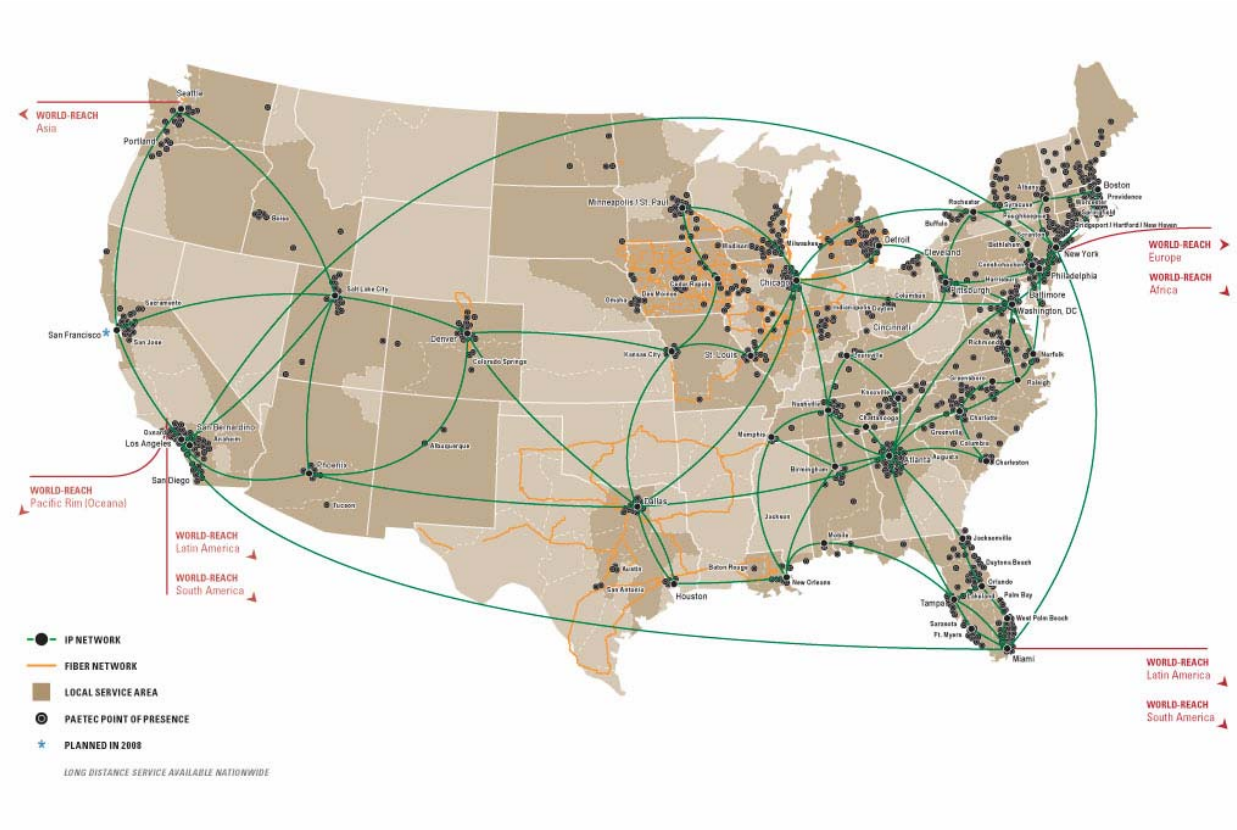



**Figure 1** illustrates an example Avaya IP telephony solution connected to the PAETEC Dynamic IP SIP Trunk Service. This configuration was utilized for compliance testing.

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

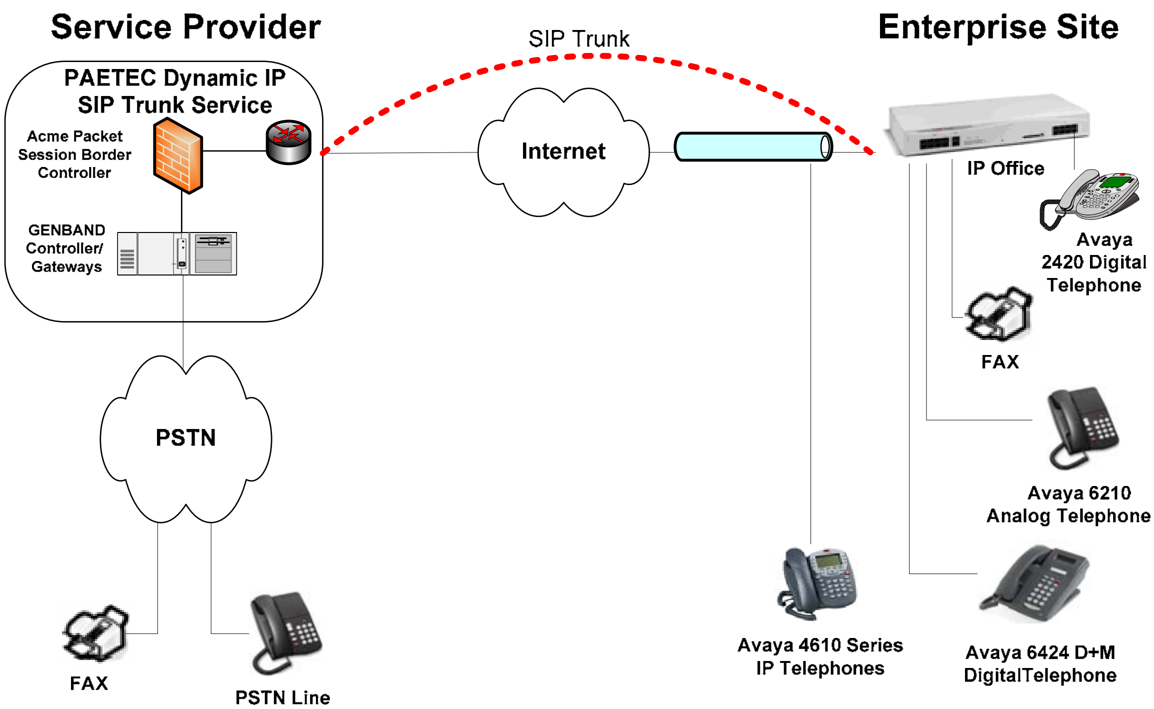



**Figure 1: Avaya IP Telephony Network PAETEC Dynamic IP SIP Trunk Service**

## 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 (11) |
| Avaya IP 400 Analog POTS 30+ | R 6.2 (11) |
| Avaya IP Office Manager (Windows PC) | R 6.2 (11) |
| Avaya IP Office Voicemail Pro | R 4.2.(24) |
| Avaya 4610SW IP Telephone | R8.016 – H.323 |
| Avaya 6424D+M Digital Telephone | n/a |
| Avaya 6210 Analog Telephone | n/a |
| Avaya 2420 Digital Telephone | R5 Firmware |
| **PAETEC Dynamic IP SIP Trunk Service Solution Components** | |
| | |
| GENBAND<br>• C3 Signaling Controller<br>• G9 Converged Gateway<br>• 8000 Media Gateway | 6.0.60.60 |
| Acme Packet Session Border Controller | 2.0.1P64 |

**Table 1: Equipment and Software Tested**

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.2 (11).

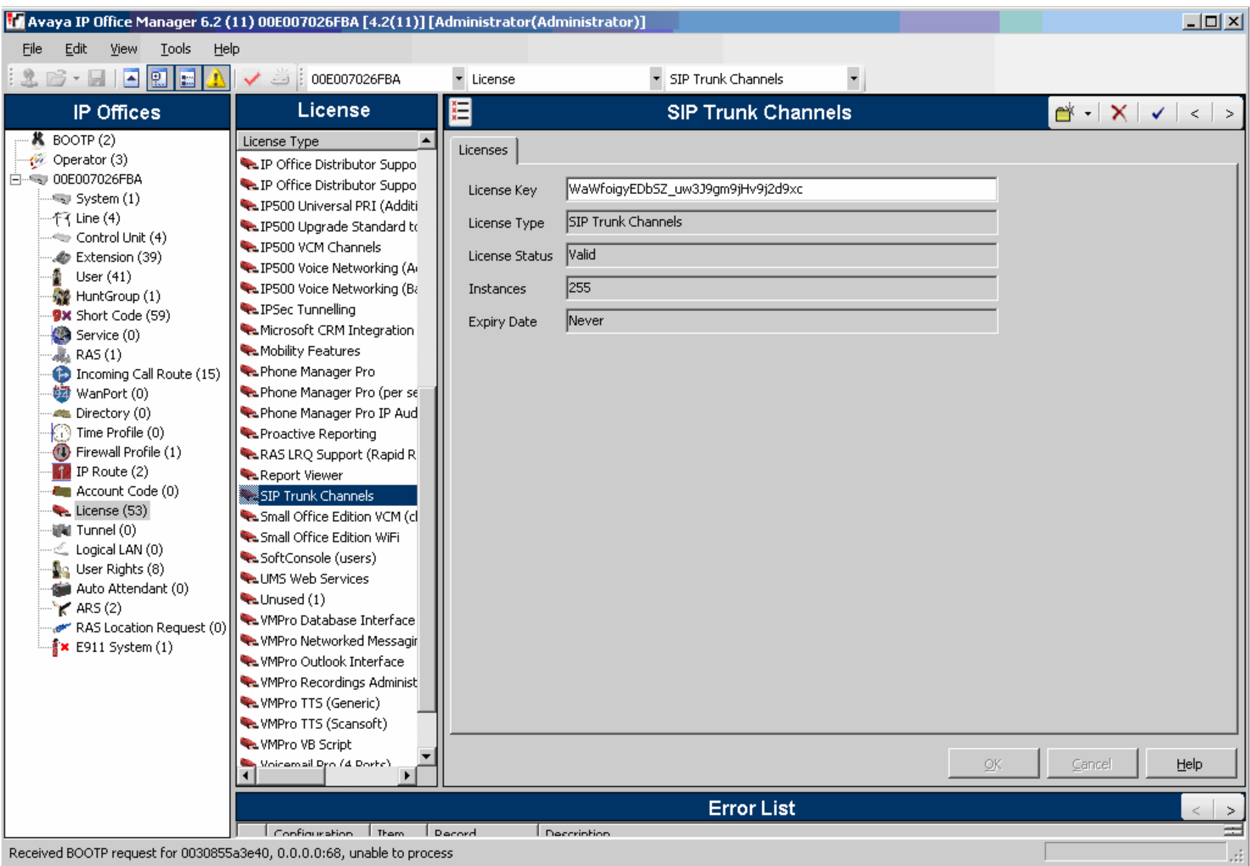
# 3. Configure Avaya IP Office

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

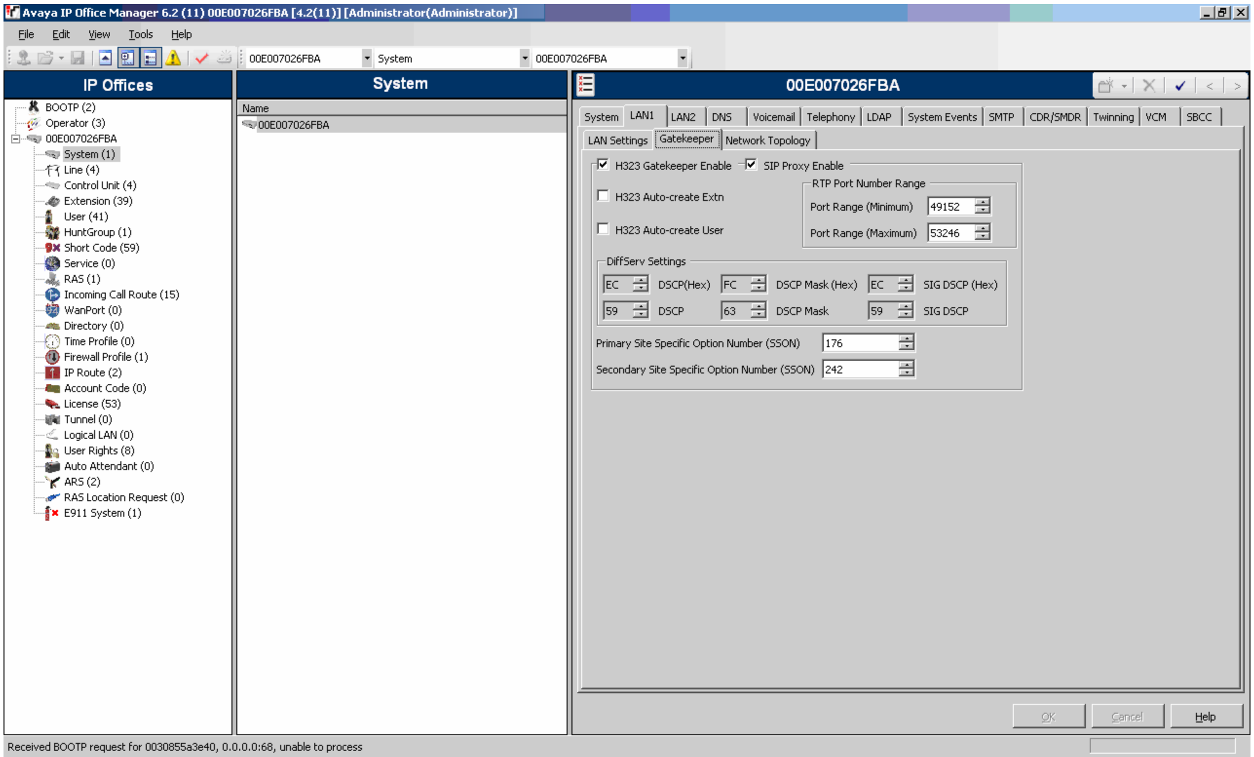
IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

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.

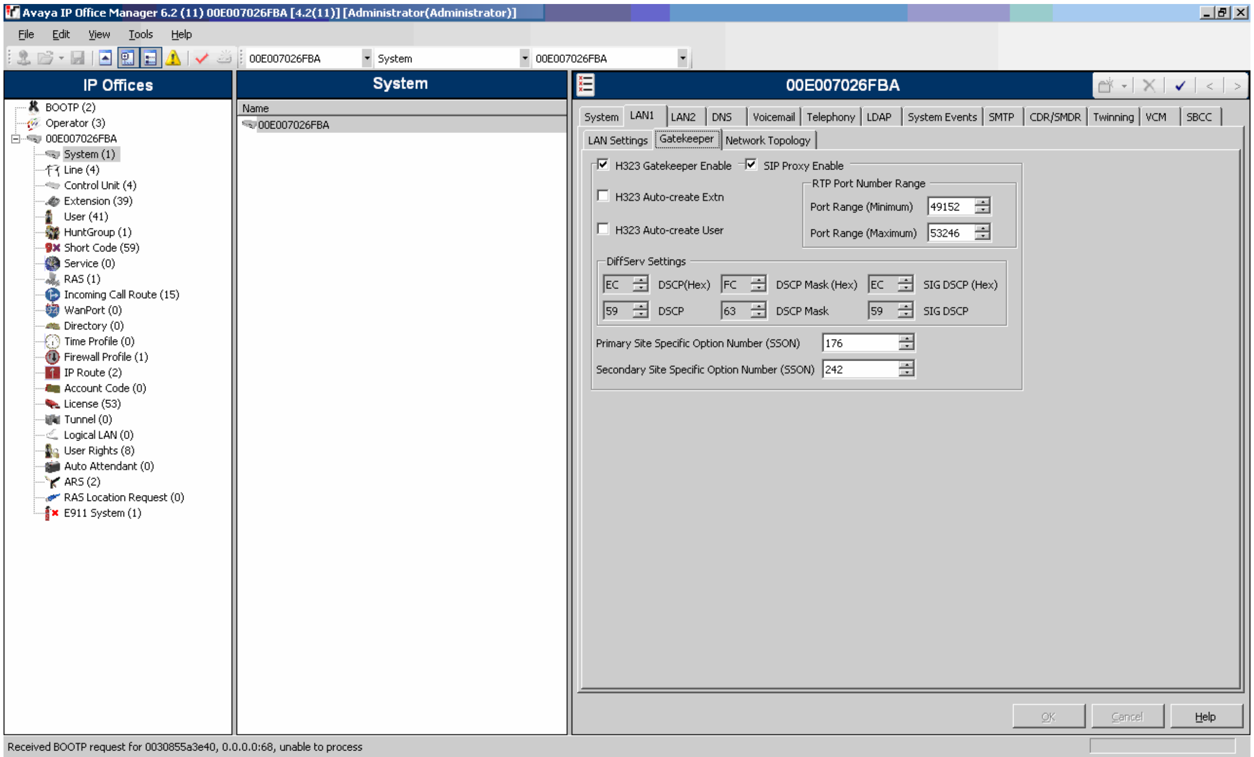
   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



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.



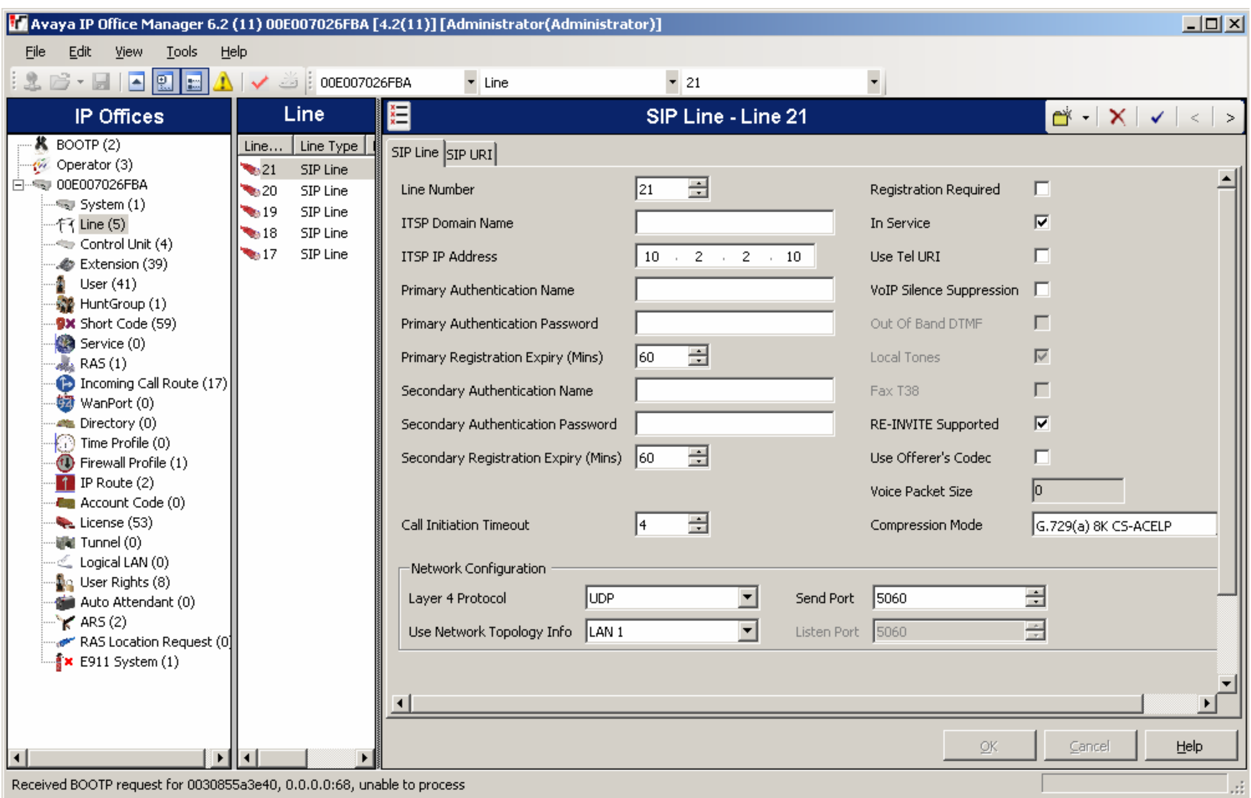
3. *Configure DiffServ Settings* according to PAETEC IP Services requirements. Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper** tab (same as mentioned in step 2). Under **DiffServ Settings**, enter **EC** into the **DSCP(Hex)** and **SIG DSCP(Hex)** text boxes by clicking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



4. *Create the SIP line* for the PAETEC Dynamic IP SIP Trunk Service. Select **Line** in the left panel. Right-click and select **New** → **SIP Line**.

Select the following:

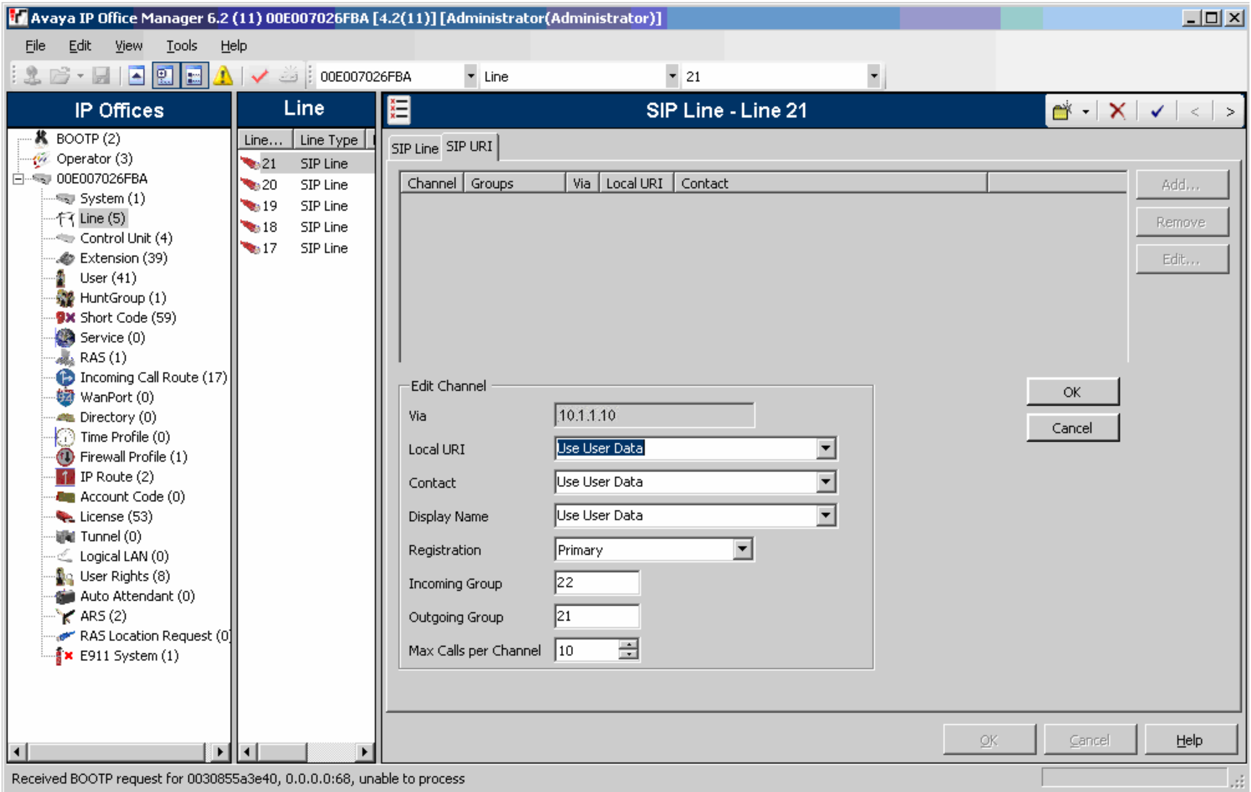
- For the **ITSP IP Address field**, enter the IP address of the PAETEC Dynamic IP SIP Trunk service SIP Proxy
- For **Compression Mode**, select the **G.729(a) 8K CS-ACELP** or **G.711 ULAW 64K** for voice calls. (** NOTE ** If FAX is to be used, **G.711 ULAW 64K** must be selected for proper operation).
- 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 unique numbers 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

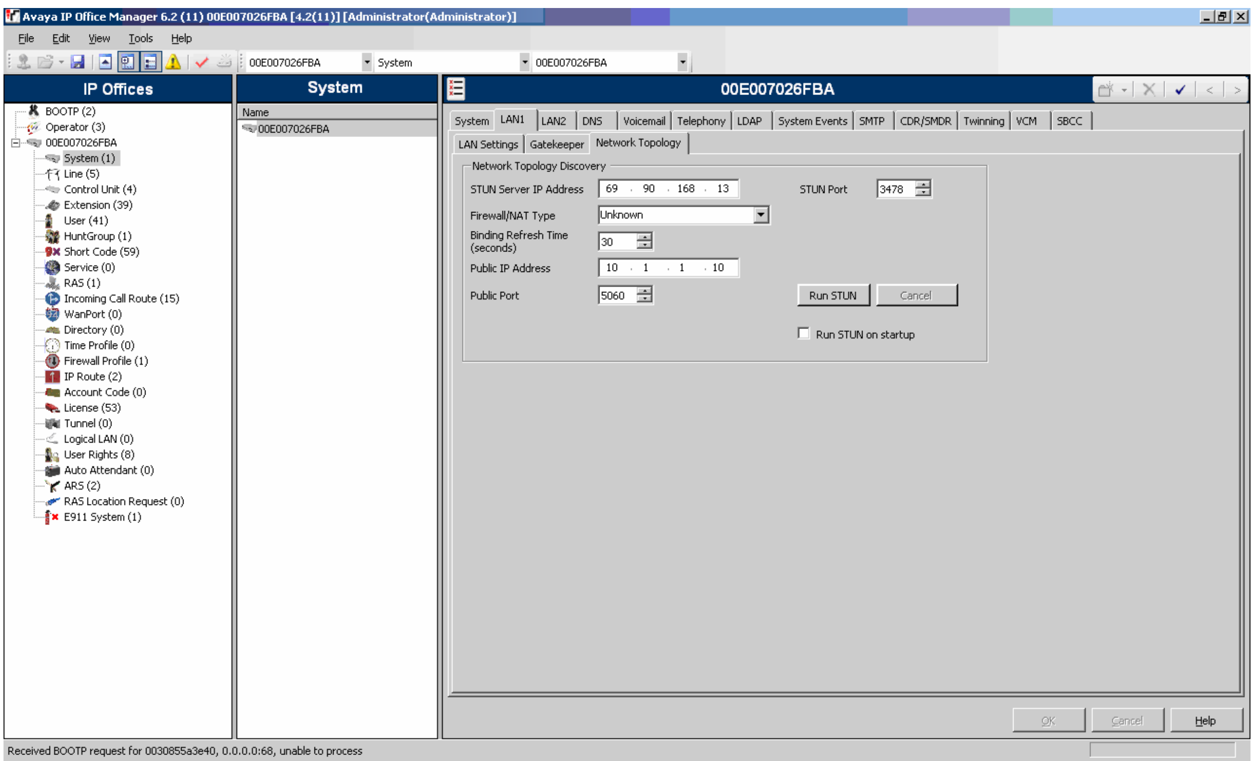
**Display Name** fields. This tells the system to use the information configured on the SIP tab for each individual user (see **Step 9**). Use defaults for all other fields. Click the **OK** button.



6. *Configure SIP OPTIONS timer on Network Topology Tab for "keep alive" function with the PAETEC Dynamic IP SIP trunk Service.* 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 PAETEC Dynamic IP SIP Trunk Service 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.

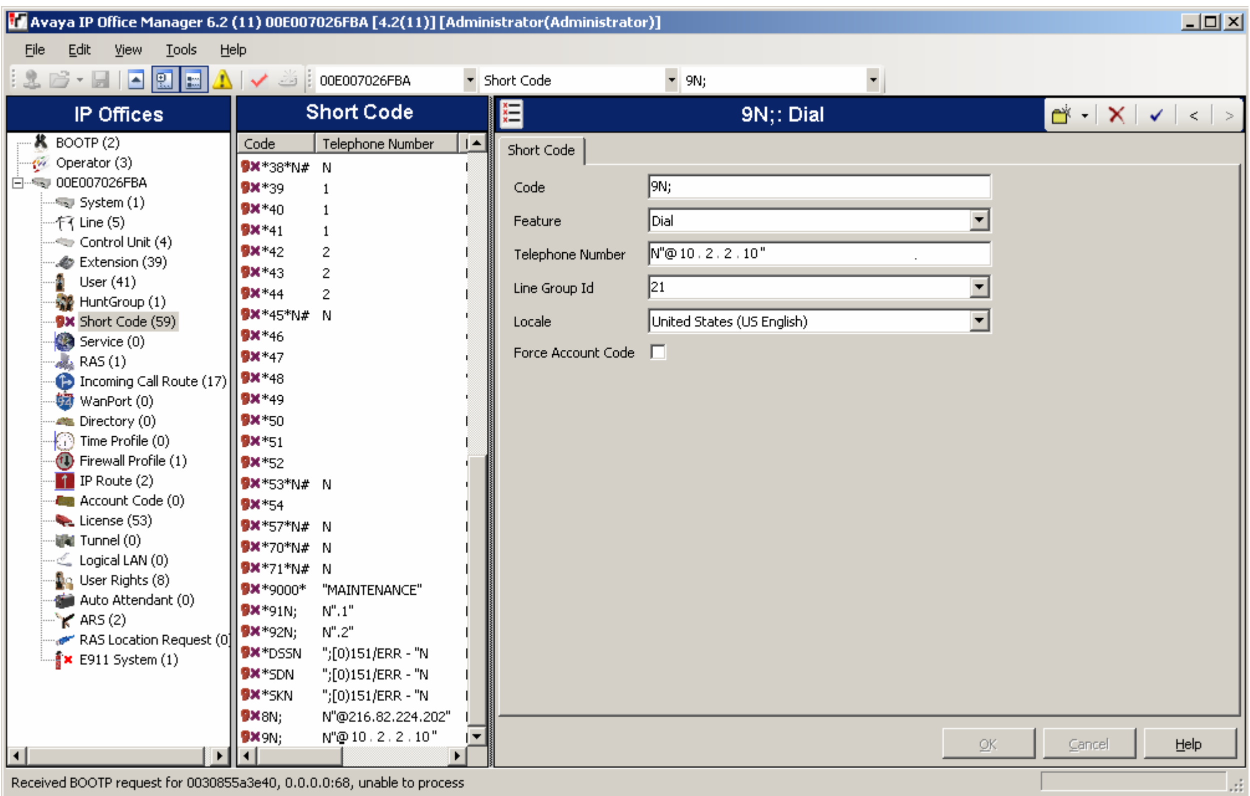
   **Note:** Depending upon what type of firewall or Network Address Translation device is being used at the customer premise, it may be necessary to set the **Firewall/NAT Type** setting to the appropriate setting as defined [2]. Click the **OK** button.



7. *Configure a short code to route calls to PAETEC 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.

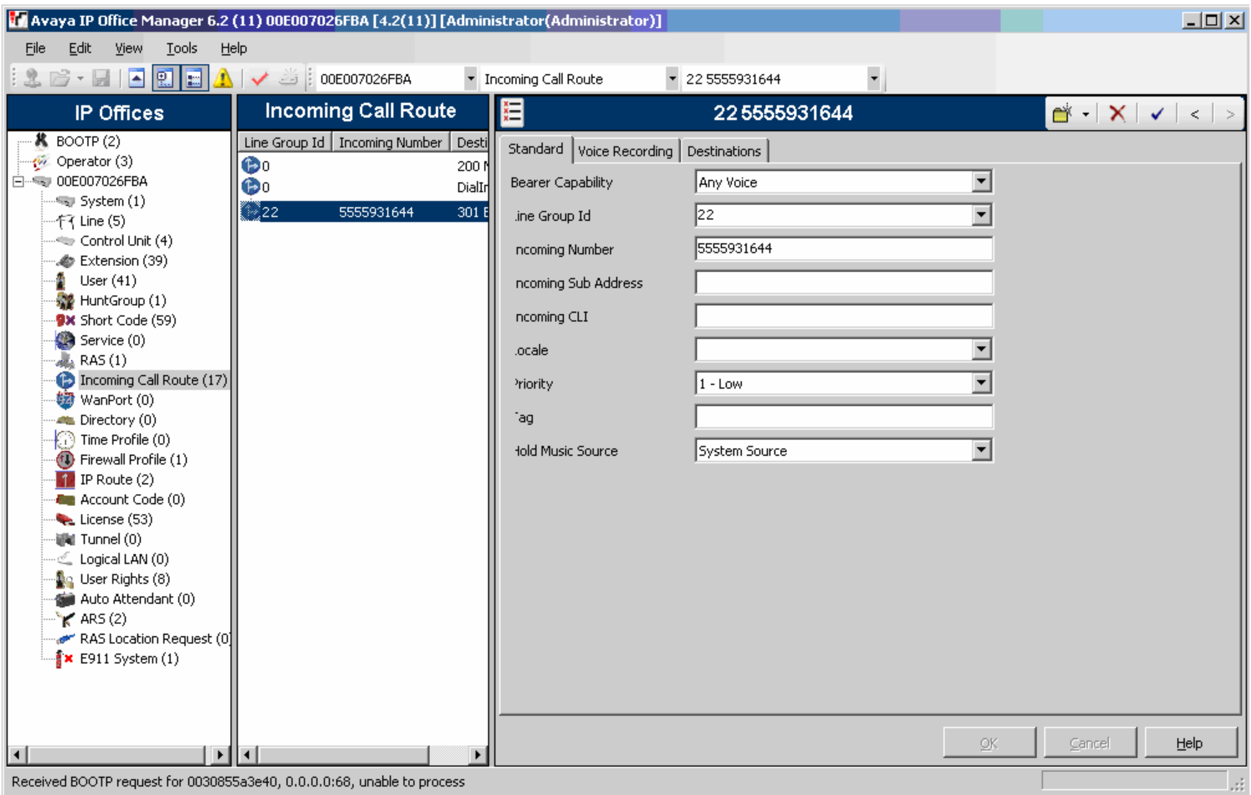
   **Note: N** can be any number other than a local IP Office extension. For example, a 10-digit number, a toll free number, directory assistance (e.g., 411), information service etc.

   Select **Dial** for the **Feature**. Enter the Outgoing Group created in **Step 5** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<Domain Name or IP Address of** PAETEC Dynamic IP SIP Trunk Service>**"** 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.

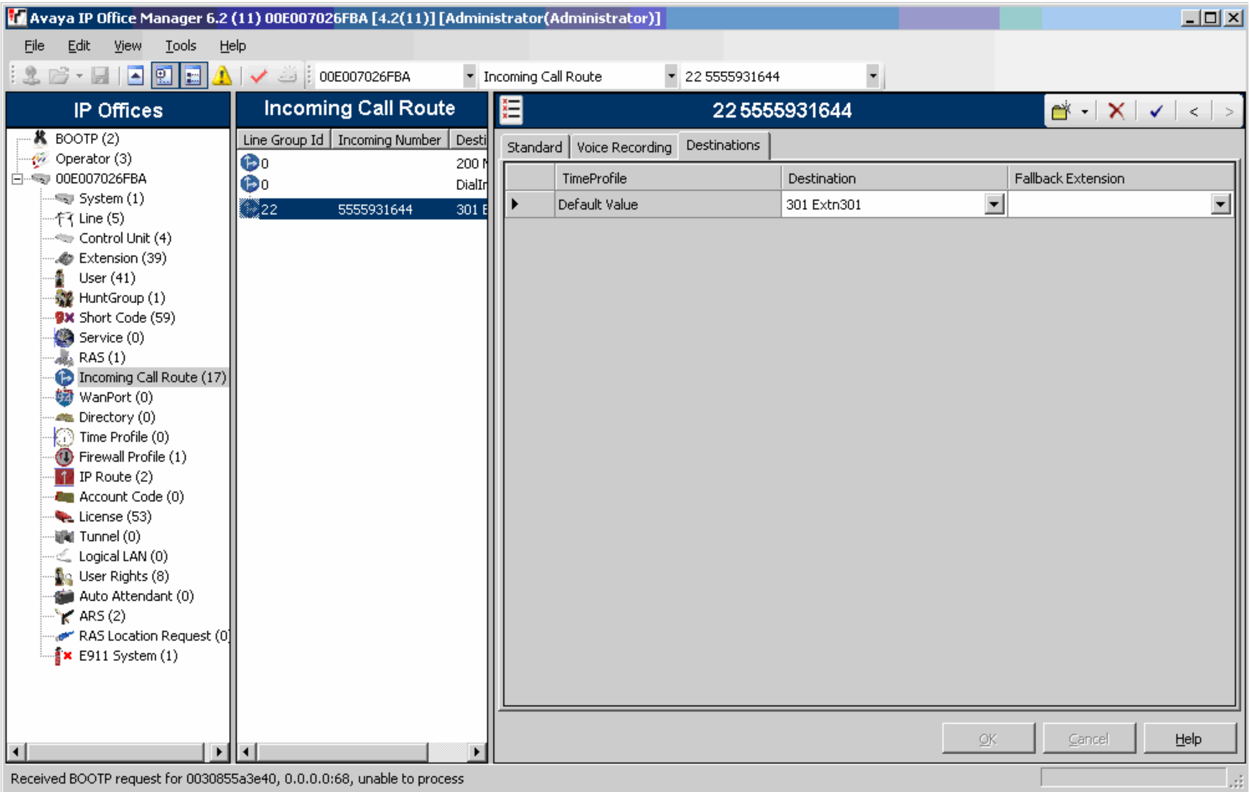


8. *Create an Incoming Call Route for the Inbound SIP calls*. Select **Incoming Call Route** in the left panel. Right-click and select **New**.

   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 PAETEC, 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.



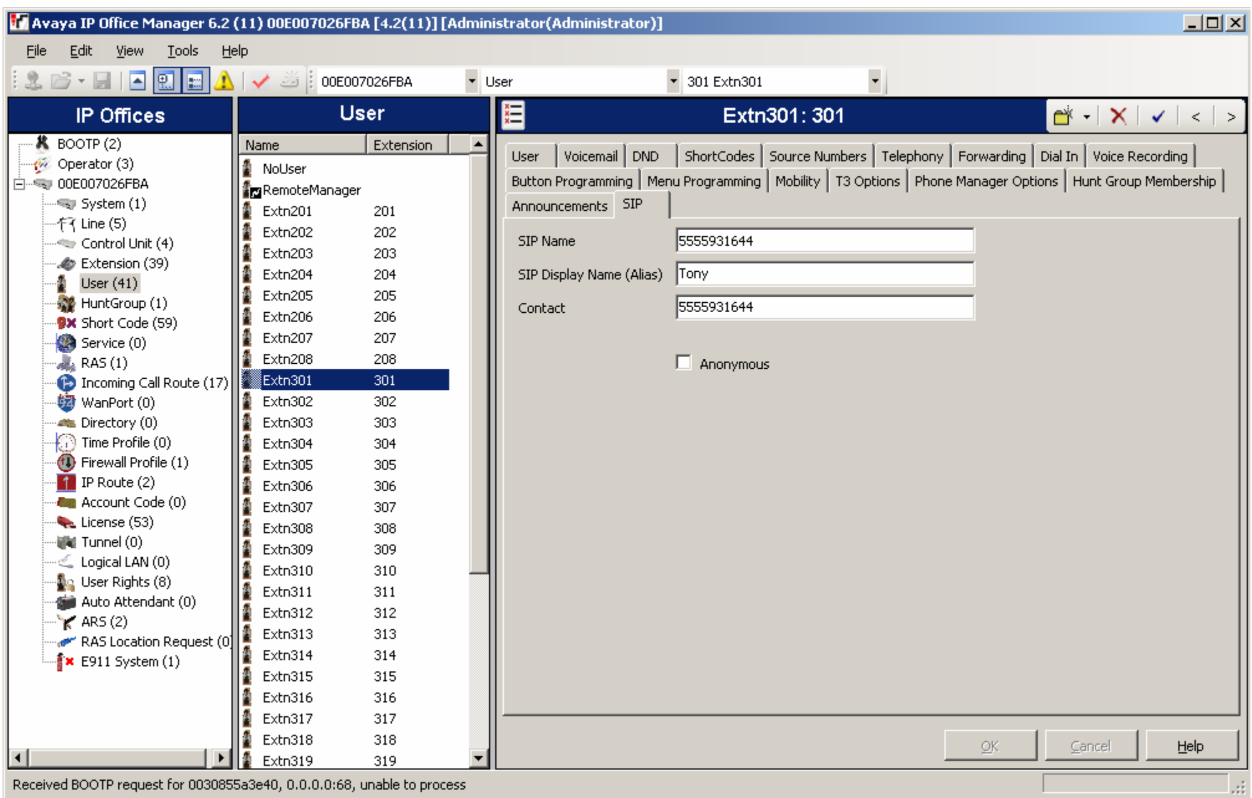
9. *Configure Users' SIP names*. Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab.

Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to the DID number provided by PAETEC's Dynamic IP SIP Trunk Service 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

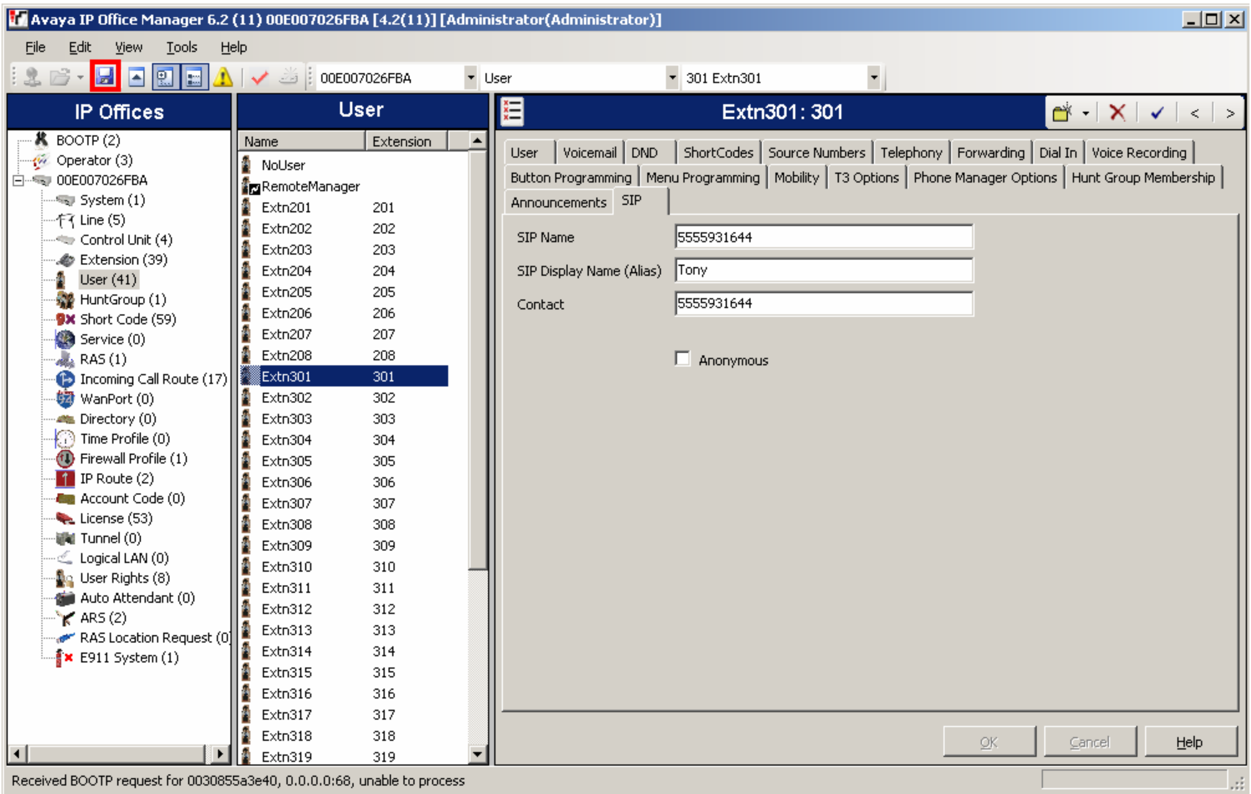
(see Appendix A for examples of SIP INVITE messages.)

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 until this step is completed. ** NOTE ** This may cause a reboot of Avaya IP Office causing service disruption.**




# 4. PAETEC Services Configuration

To use PAETEC Communications Dynamic IP SIP Trunk Service, a customer must request service from PAETEC using their sales processes. The process can be started by contacting PAETEC Communications via the corporate web site at http://www.paetec.com/contact/inforequest.asp and requesting information via the online sales links or telephone numbers.

During the signup process, PAETEC will require that the customer provide the public IP address used to reach the Avaya IP Office server. PAETEC Communications provided the following information for the compliance testing: IP address of the PAETEC Communications SIP proxy/SBC, and Direct Inward Dialed (DID) numbers. This information was used to complete the Avaya IP Office configuration discussed in the previous sections.

# 5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between PAETEC Dynamic IP SIP Trunk 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 SIP Trunking solution provided by PAETEC Dynamic IP SIP Trunk 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 PAETEC Dynamic IP SIP Trunk Services.
- Outgoing calls from the enterprise site were completed via PAETEC Dynamic IP SIP Trunk Service 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.729(a) and G.711 ULAW codecs.
- Fax routing to ensure G.711 ULAW use for fax calls. The PAETEC Dynamic IP SIP Trunk Service does not support T.38.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with G.729a.
- Telephone features such as hold, transfer, and conference.
- Mobility Features: Mobile twinning to a mobile phone.

# 6. 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. T.38 fax is not supported by the PAETEC Dynamic IP SIP Trunk Service.

2. 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.

3. When a PSTN call is placed over a SIP trunk to an enterprise Avaya IP Office phone with mobile twinning enabled, the calling number displayed on the mobile phone is the DID associated with the enterprise Avaya IP Office phone.

# 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 calls and receive inbound calls through PAETEC Dynamic IP SIP Trunk Services.

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.

# 8. Support

For technical support on PAETEC Communications Dynamic IP SIP Trunk services, contact PAETEC Communications Customer Service by calling 877-340-2600 or by sending email to customerservice@PAETEC.com. Include the customer account number in the communication.

# 9. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya IP Office to connect to the PSTN via the PAETEC Dynamic IP SIP Trunk Service. The PAETEC Dynamic IP SIP Trunk Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The PAETEC Dynamic IP SIP Trunk Service provides businesses a flexible, cost-saving 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 4.2 Installation Manual, Issue 19l*, November 2008
Document Number 15-601042
http://support.avaya.com/elmodocs2/ip_office/R4.2/Newissuesept08/eng/ip_office_installation.pdf

[2] *IP Office 4.2 Manager 6.2, Issue 22r*, November 2008
Document Number 15-601011
http://support.avaya.com/elmodocs2/ip_office/R4.2/Newissuesept08/eng/manager_en.pdf

[3] *4600 Series IP Telephone LAN Administrator Guide*, July 2008, Issue 8, Document Number 555-233-507
http://support.avaya.com/japple/css/japple?temp.documentID=344333&temp.productID=107755&temp.bucketID=159898&PAGE=Document

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

[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 the PAETEC Dynamic IP SIP Trunk Service 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 the PAETEC Dynamic IP SIP Trunk Service to Avaya IP Office:**

```
No.     Time        Source                Destination           Protocol Info
      1 0.000000    10.2.2.10         10.1.1.10          SIP/SDP  Request: INVITE
sip:5555931644@10.1.1.10:5060, with session description

Frame 1 (812 bytes on wire, 812 bytes captured)
Ethernet II, Src: Cisco_91:fd:51 (00:18:18:91:fd:51), Dst: AvayaEcs_02:6f:ba
(00:e0:07:02:6f:ba)
Internet Protocol, Src: 10.2.2.10 (10.2.2.10), Dst: 10.1.1.10 (10.1.1.10)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:5555931644@10.1.1.10:5060 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bK08d22n20bo5gtgsoo381.1
            Transport: UDP
            Sent-by Address: 10.2.2.10
            Sent-by port: 5060
            Branch: z9hG4bK08d22n20bo5gtgsoo381.1
        To: <sip:5555931644@10.1.1.10:5060>
            SIP to address: sip:5555931644@10.1.1.10:5060
        From: <sip:7324500819@10.2.2.10>;tag=1016670
            SIP from address: sip:7324500819@10.2.2.10
            SIP tag: 1016670
        Call-ID: 1224793348-1240467@xxx.paetec.com
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        Max-Forwards: 69
        Contact: <sip:7324500819@10.2.2.10:5060;transport=udp>
            Contact Binding: <sip:7324500819@10.2.2.10:5060;transport=udp>
                URI: <sip:7324500819@10.2.2.10:5060;transport=udp>
                    SIP contact address: sip:7324500819@10.2.2.10:5060
        Supported: 100rel
        Expires: 330
        Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, INFO, PRACK, REFER, SUBSCRIBE,
NOTIFY, UPDATE, REGISTER
        Content-Type: application/sdp
        Content-Length: 233
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 287998977 1227104125 IN IP4 10.2.2.10
                Owner Username: -
                Session ID: 287998977
                Session Version: 1227104125
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 10.2.2.10
```

```
Session Name (s): -
Connection Information (c): IN IP4 10.2.2.10
    Connection Network Type: IN
    Connection Address Type: IP4
    Connection Address: 10.2.2.10
Time Description, active time (t): 0 0
    Session Start Time: 0
    Session Stop Time: 0
Session Attribute (a): sendrecv
Media Description, name and address (m): audio 20000 RTP/AVP 18 0 101
    Media Type: audio
    Media Port: 20000
    Media Proto: RTP/AVP
    Media Format: ITU-T G.729
    Media Format: ITU-T G.711 PCMU
    Media Format: 101
Media Attribute (a): rtpmap:18 G729/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 18
    MIME Type: G729
Media Attribute (a): fmtp:18 annexb=no
    Media Attribute Fieldname: fmtp
    Media Format: 18 [G729]
    Media format specific parameters: annexb=no
Media Attribute (a): rtpmap:0 PCMU/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 0
    MIME Type: PCMU
Media Attribute (a): rtpmap:101 telephone-event/8000
    Media Attribute Fieldname: rtpmap
    Media Format: 101
    MIME Type: telephone-event
```

**Sample SIP INVITE Message from Avaya IP Office to the PAETEC Dynamic IP SIP Trunk Service:**

```
No.     Time        Source                Destination           Protocol Info
    135 12.847252   10.1.1.10          10.2.2.10         SIP/SDP  Request: INVITE
sip:17324500819@10.2.2.10, with session description

Frame 135 (820 bytes on wire, 820 bytes captured)
Ethernet II, Src: AvayaEcs_02:6f:ba (00:e0:07:02:6f:ba), Dst: Cisco_91:fd:51
(00:18:18:91:fd:51)
Internet Protocol, Src: 10.1.1.10 (10.1.1.10), Dst: 10.2.2.10 (10.2.2.10)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:17324500819@10.2.2.10 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP
10.1.1.10:5060;rport;branch=z9hG4bKb472a674161fd61cf7b4822bfd8ae49e
            Transport: UDP
            Sent-by Address: 10.1.1.10
            Sent-by port: 5060
            RPort: rport
            Branch: z9hG4bKb472a674161fd61cf7b4822bfd8ae49e
        From: "Tony" <sip:5555931644@10.1.1.10>;tag=fa1035be6783b5c2
            SIP Display info: "Tony"
            SIP from address: sip:5555931644@10.1.1.10
            SIP tag: fa1035be6783b5c2
        To: <sip:17324500819@10.2.2.10>
            SIP to address: sip:17324500819@10.2.2.10
        Call-ID: 584245e4cc320a6ca04e1b716ed37fd5@10.1.1.10
        CSeq: 725596173 INVITE
            Sequence Number: 725596173
            Method: INVITE
        Contact: "Tony" <sip:5555931644@10.1.1.10:5060;transport=udp>
            Contact Binding: "Tony" <sip:5555931644@10.1.1.10:5060;transport=udp>
                URI: "Tony" <sip:5555931644@10.1.1.10:5060;transport=udp>
                    SIP Display info: "Tony"
                    SIP contact address: sip:5555931644@10.1.1.10:5060
        Max-Forwards: 70
        Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
        Content-Type: application/sdp
        Content-Length: 277
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): UserA 3436370381 1943486669 IN IP4
10.1.1.10
                Owner Username: UserA
                Session ID: 3436370381
                Session Version: 1943486669
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 10.1.1.10
            Session Name (s): Session SDP
            Connection Information (c): IN IP4 10.1.1.10
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 10.1.1.10
            Time Description, active time (t): 0 0
                Session Start Time: 0
```

```
        Session Stop Time: 0
    Media Description, name and address (m): audio 49154 RTP/AVP 18 0 8 101
        Media Type: audio
        Media Port: 49154
        Media Proto: RTP/AVP
        Media Format: ITU-T G.729
        Media Format: ITU-T G.711 PCMU
        Media Format: ITU-T G.711 PCMA
        Media Format: 101
    Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 18
        MIME Type: G729
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 0
        MIME Type: PCMU
    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 Developer*Connection* Program at devconnect@avaya.com.