Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the PAETEC Broadsoft based 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 PAETEC Broadsoft based SIP Trunking solution and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

PAETEC is a domestic service provider offering affordable, flexible and reliable integrated voice and data services to small and medium-sized business in nearly 500 cities throughout the Midwest, Rocky Mountain, Southwest and Northwest regions.

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 Broadsoft based SIP Trunking service and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

PAETEC[1] is a domestic service provider offering affordable, flexible and reliable integrated voice and data services to small and medium-sized business in nearly 500 cities throughout the Midwest, Rocky Mountain, Southwest and Northwest regions.

SIP Trunking from PAETEC combines local and long-distance voice service, secure data networking and broadband Internet access on one performance-guaranteed connection.

Customers using this Avaya IP Office telephony solution with PAETEC Broadsoft based SIP Trunking solution is 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.

Customers looking for a turnkey access solution can select a Managed Service option, where PAETEC provides and manages the router that interfaces with customer equipment.
PAETEC's SIP Trunking solution offers the following capabilities:

- Outbound domestic calling to local and long distance services (voice and fax[2])
- Outbound international calling
- Incoming Direct Inward Dial (DID) service
- Dynamically reroute inbound calls from one location to another location on the PAETEC MPLS network, without long distance charges
- Call restriction, by area code, by number, or by call type

Figure 1 illustrates a sample Avaya IP telephony solution connected to PAETEC's SIP Trunking solution. This configuration was utilized for compliance testing.
Note: Generic IP addresses are used in this document to hide the real IP addresses used during 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

[1] The service providers of PAETEC and McLeodUSA have recently merged; therefore, references to both may be found throughout this document.
[2] For proper FAX operation, appropriate network engineering should be done to ensure QoS. Also, see **Section 6** for specific FAX limitations.

- Avaya 6210 Analog Telephone
- Avaya Voice Mail Pro

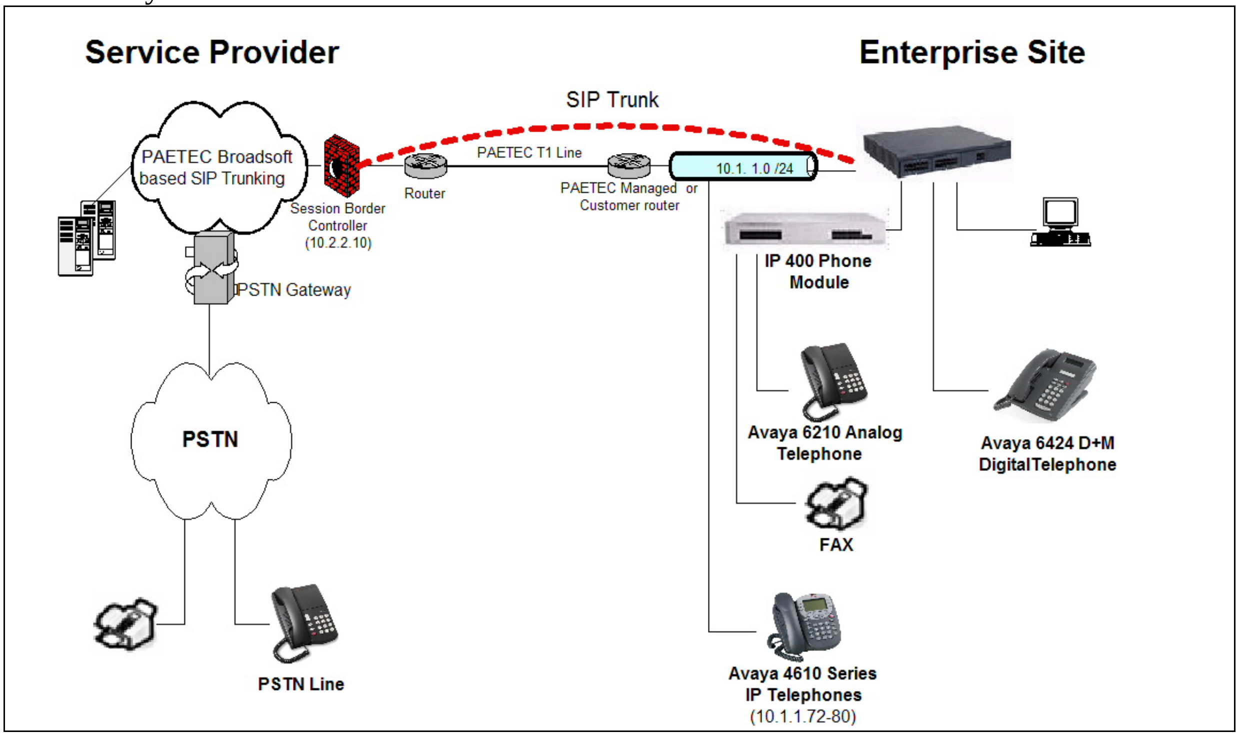


Figure 1: Avaya IP Telephony Network using PAETEC Broadsoft based SIP Trunking Solution

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 500	R 4.1.9
Avaya IP Office Manager (Windows PC)	R 6.1 (09)
Avaya Voice Mail Pro	R 4.1 (27)
Avaya 4610SW IP Telephone	R2.8 – H.323 – a10d01b2_8.bin
Avaya 6424D+M Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
PAETEC Broadsoft based SIP Trunking Solution Components	
PAETEC Application Server	R14
PAETEC Network Server	R14
PAETEC Media Server	R14
PAETEC PSTN Gateway (LCS)	v3.10.1.7.xp19
PAETEC SBC	V3.3.4

Table 1: Equipment and Software Tested

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.1.9.

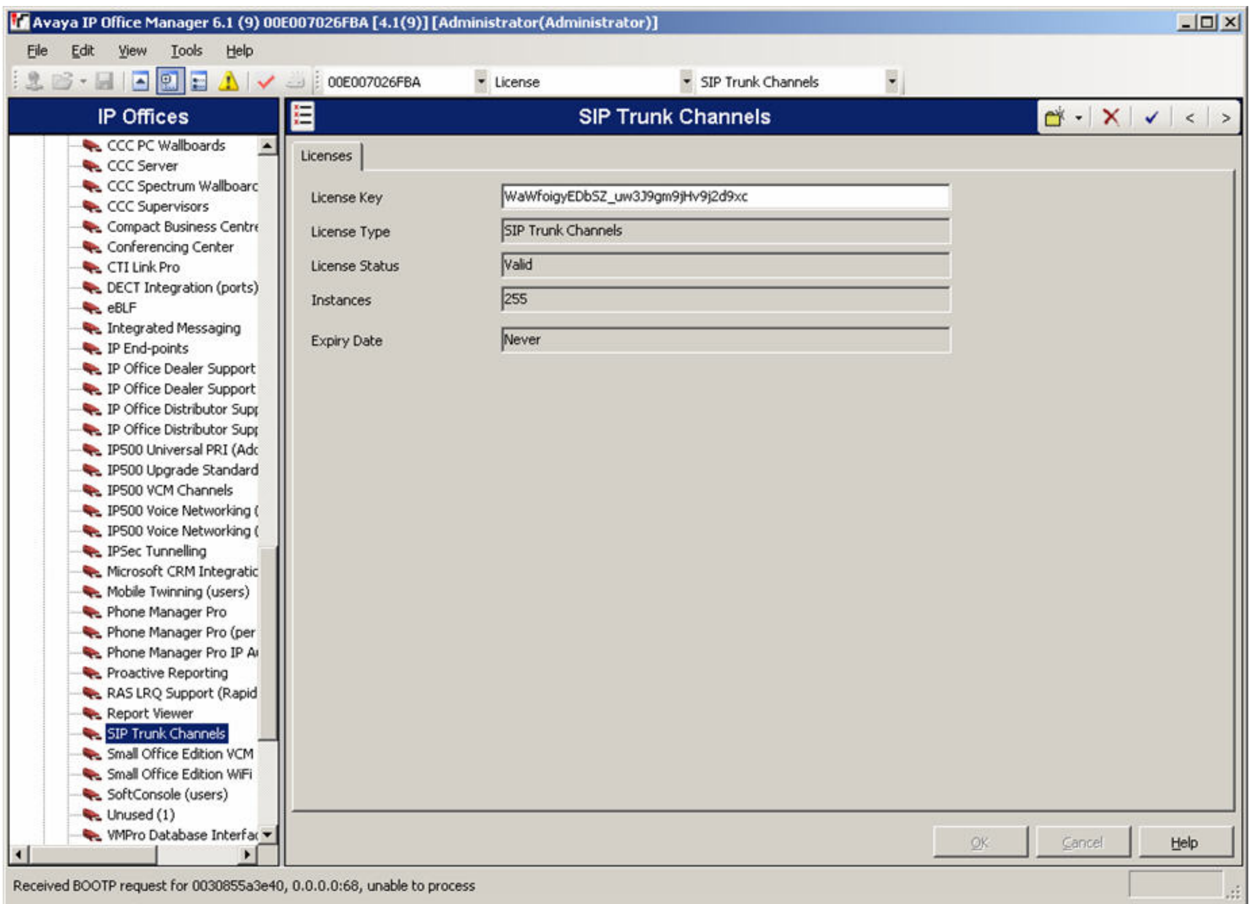
3. Configure Avaya IP Office

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

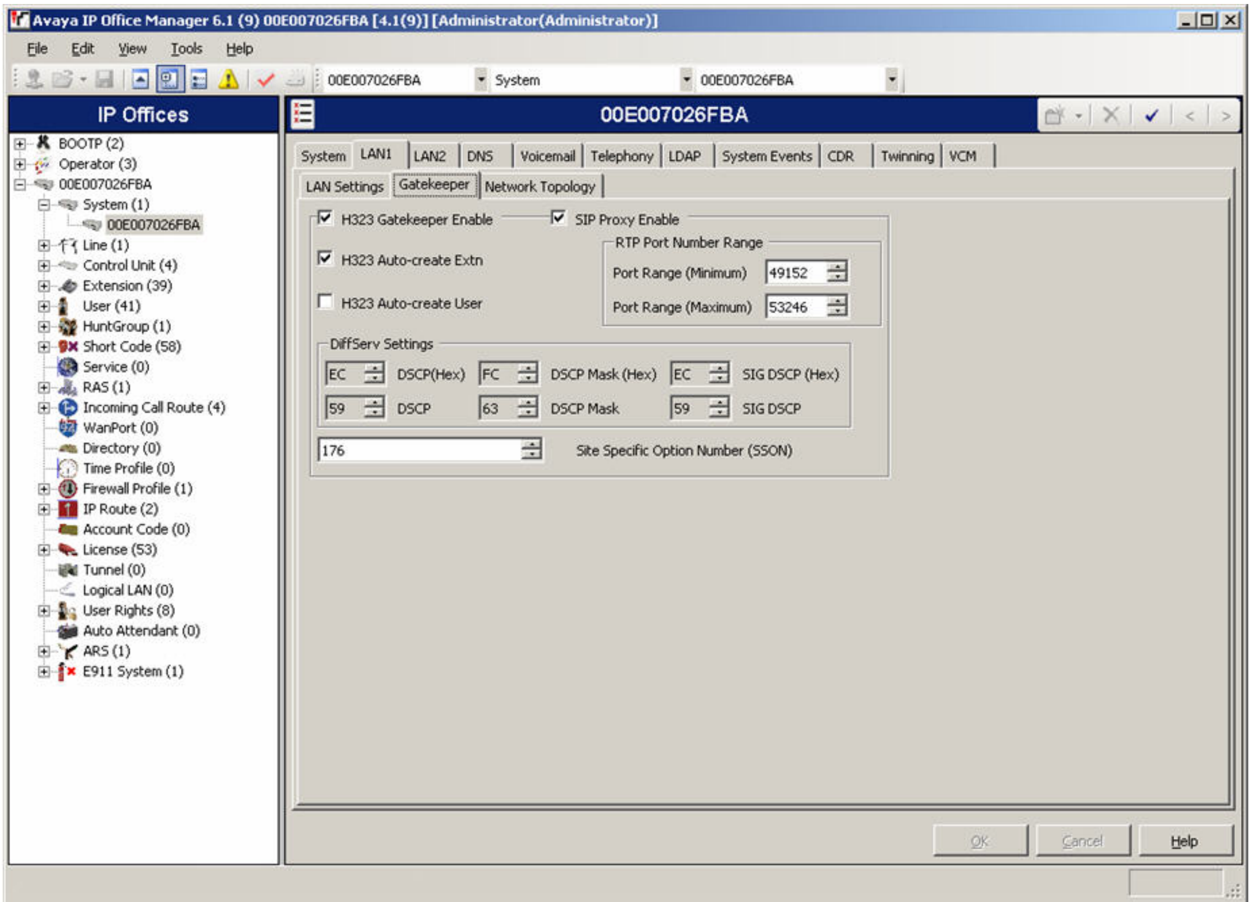
IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

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

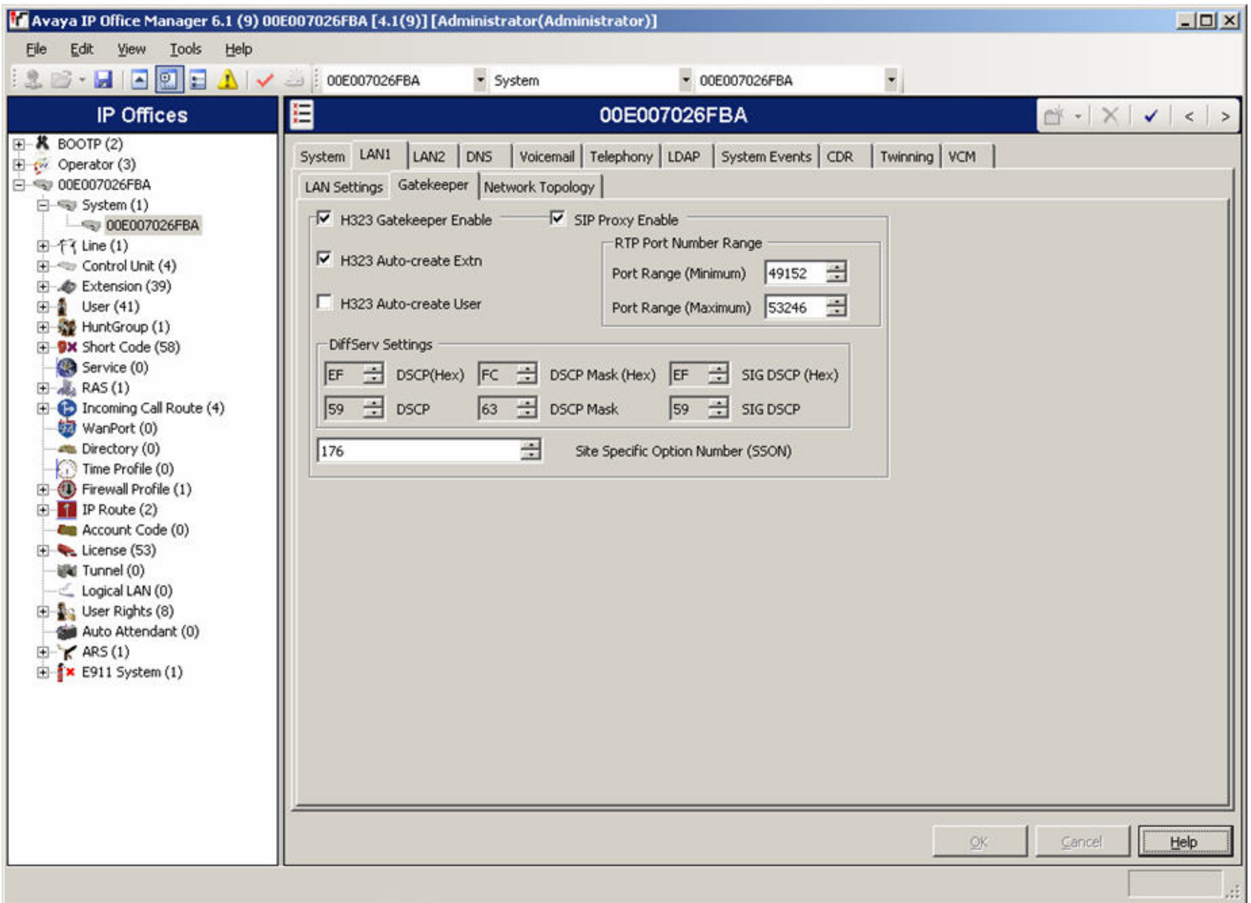
   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



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.



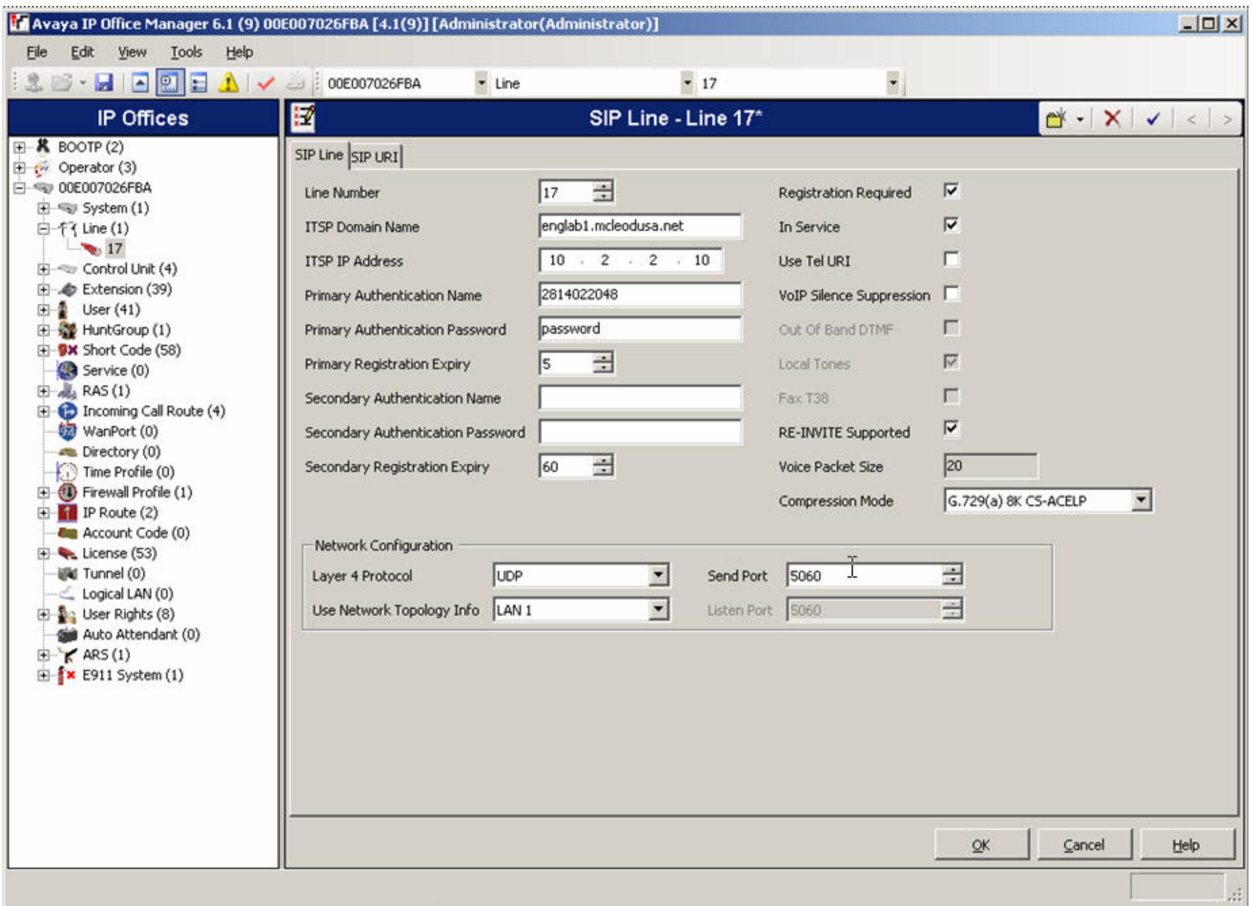
3. *Configure DiffServ Settings according to PAETEC requirements.* Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper** tab. Under **DiffServ Settings**, enter **EF** into the **DSCP(Hex)** and **SIG DSCP(Hex)** text boxes. This allows the voice and signaling packets to get the highest priority in PAETEC's network.



4. *Create a SIP line for the PAETEC service*. Right-click on **Line** and select **New → SIP line**.

Select the following:

- For the **ITSP Domain Name field**, enter the PAETEC Domain Name. SIP registration messages will use this name (Please refer to the Installation Package provided by PAETEC for actual settings.)
- For the **ITSP IP Address field**, enter the IP address of the PAETEC SIP Proxy
- For **Registration Required**, check the box to enable
- For **Primary Authentication Name**, use the Trunk ID assigned by PAETEC
- For **Primary Authentication Password**, use the password assigned by PAETEC
- For **Compression Mode**, select the **G729a 8K CS-ACELP** for all voice calls[3].
- 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



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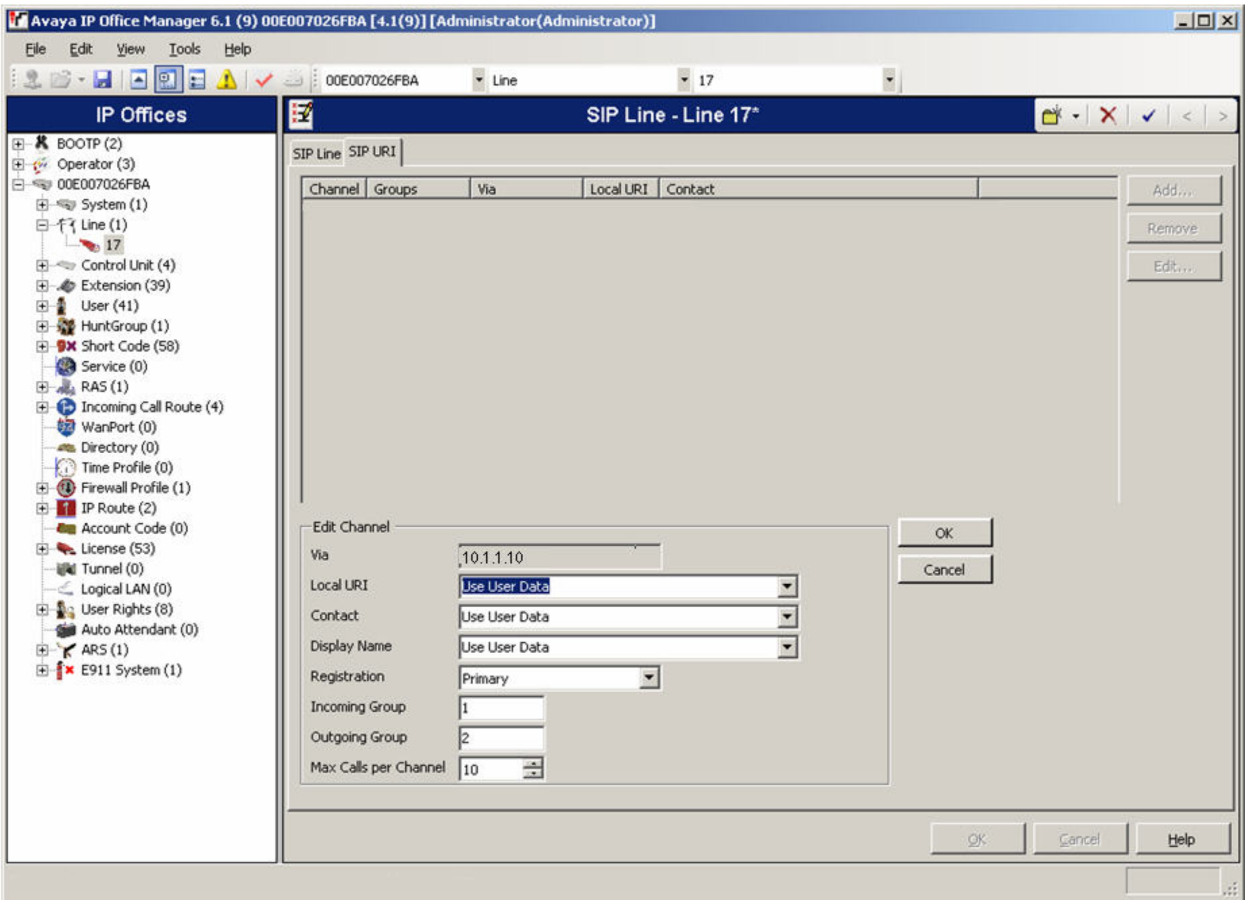
[3] To ensure proper interoperability between Avaya IP Office and PAETEC for voice calls, G.729a is required. For FAX, PAETEC will designate specific DIDs which will be configured to use only the G.711mu or G.711a codec types. This will allow FAX calls using these DIDs to negotiate to a G.711 codec even though G.729a is selected as the preferred codec type on the IP Office SIP Line.

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

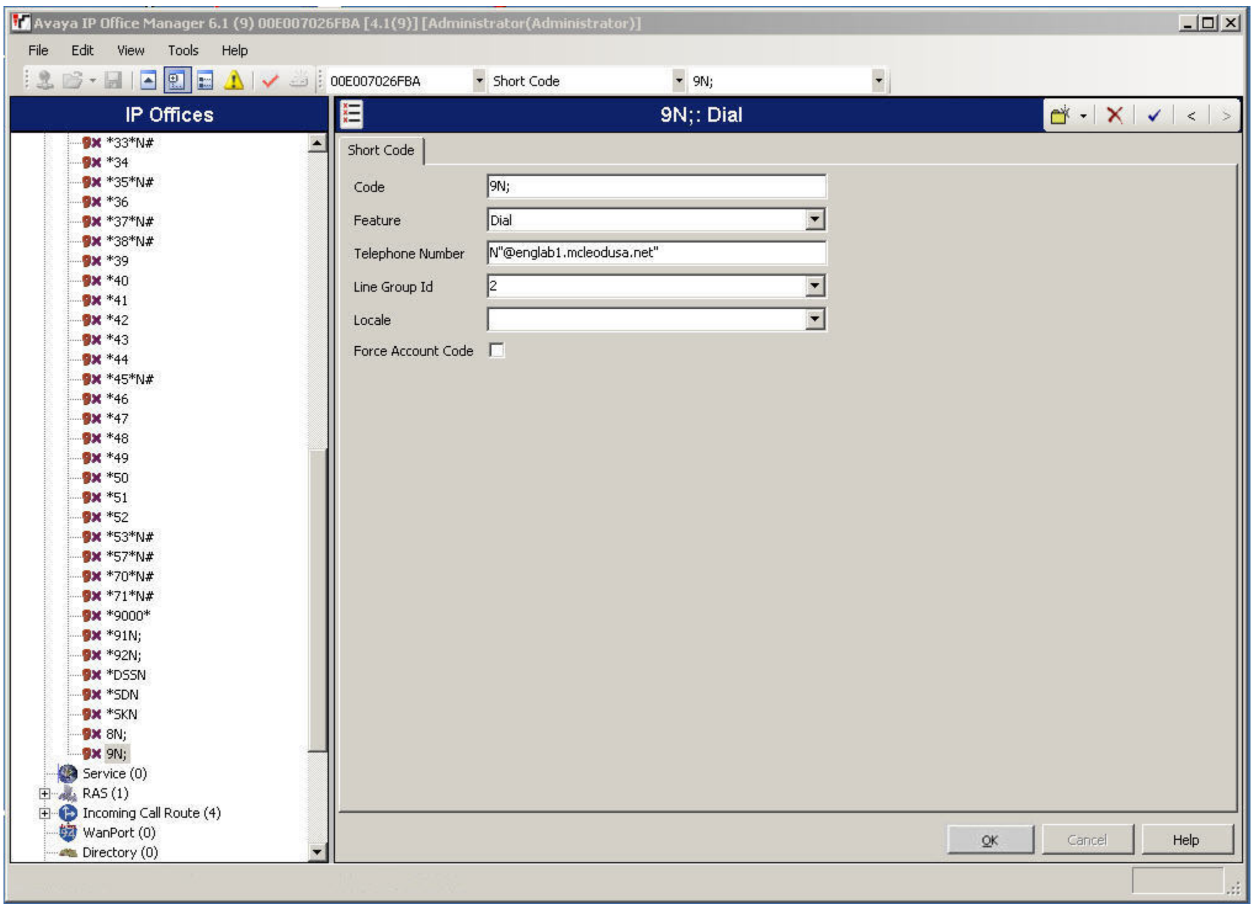
   Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. Select **Use User Data** for the **Contact**, **Local URI** and **Display Name** fields, this tells the system to use the information configured on the SIP tab for each individual user (see **Step 9**). Use defaults for all other fields. Press the **OK** button.



6. *Configure a short code to route calls to PAETEC'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 **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, N can be a 10-digit Direct Inward Dial (DID) number, operator assistance, 411, information service, etc.

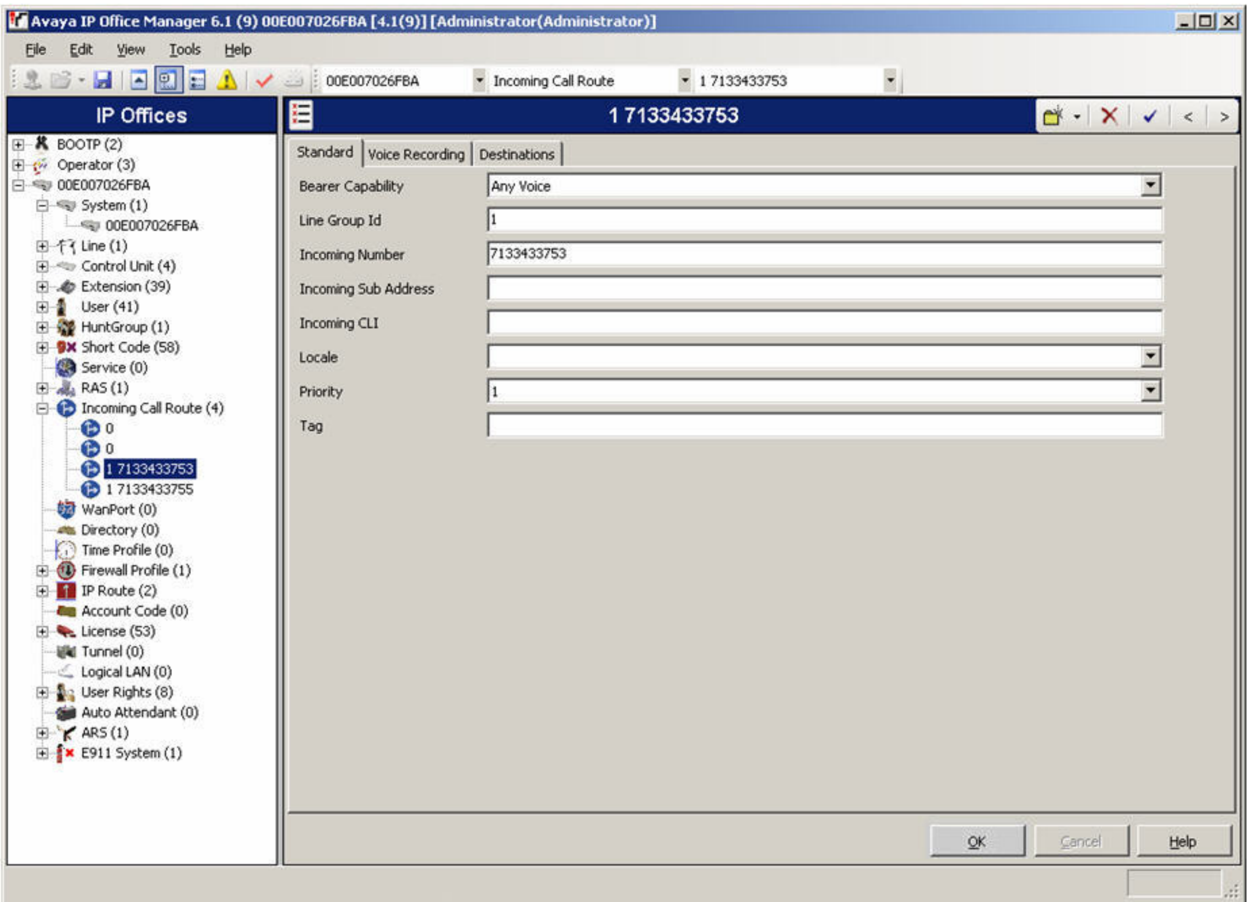
   Select **Dial** for the **Feature**. Enter the **Outgoing Group Id** created in **Step 5** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<Domain Name of PAETEC >"** for the **Telephone Number** field. The **Telephone Number** field is used to construct the **To** field's 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.



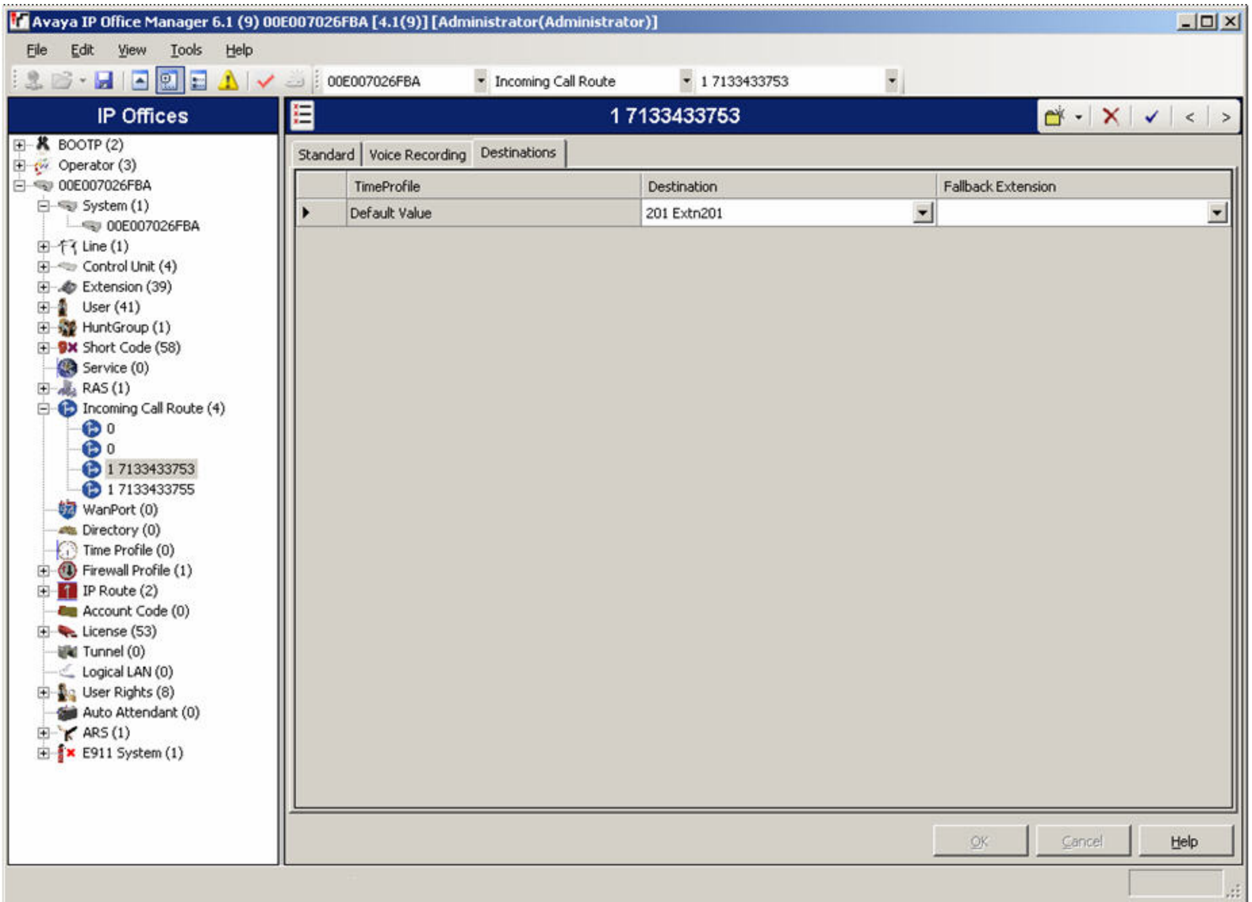
7. *Create an Incoming Call Route for the Inbound SIP calls*. Select **Incoming Call Route** in the left panel. Right-click and select **New**.

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. Press the **OK** button.



- Next, navigate to the **Destinations** tab and select the desired local extension number from the drop down menu.
- Press the **OK** button.



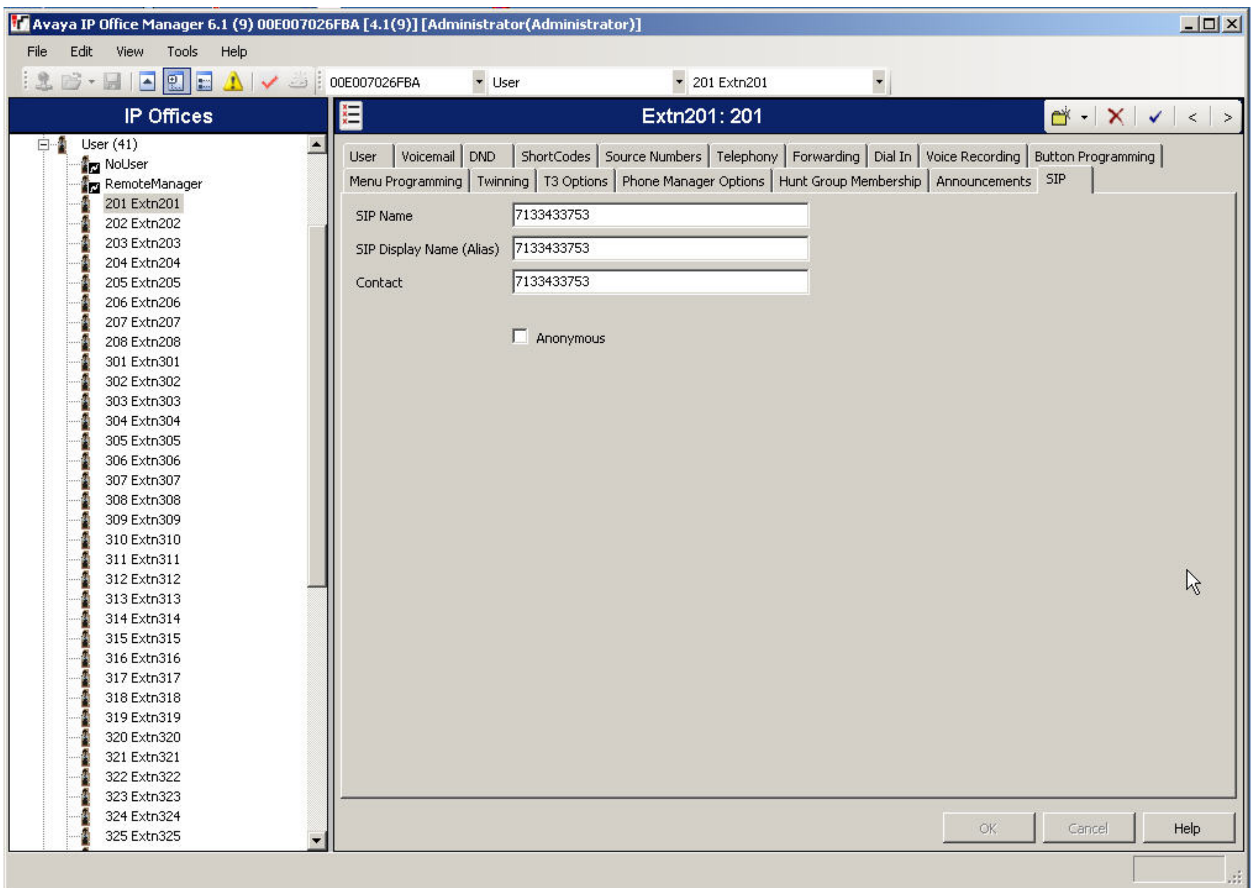
8. *Configure Users' SIP names*. Select **User** in the left panel. Select the desired user by double-clicking on an entry in the left panel. Select the **SIP** tab all the way at the end.

Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to the DID number provided by PAETEC 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.)

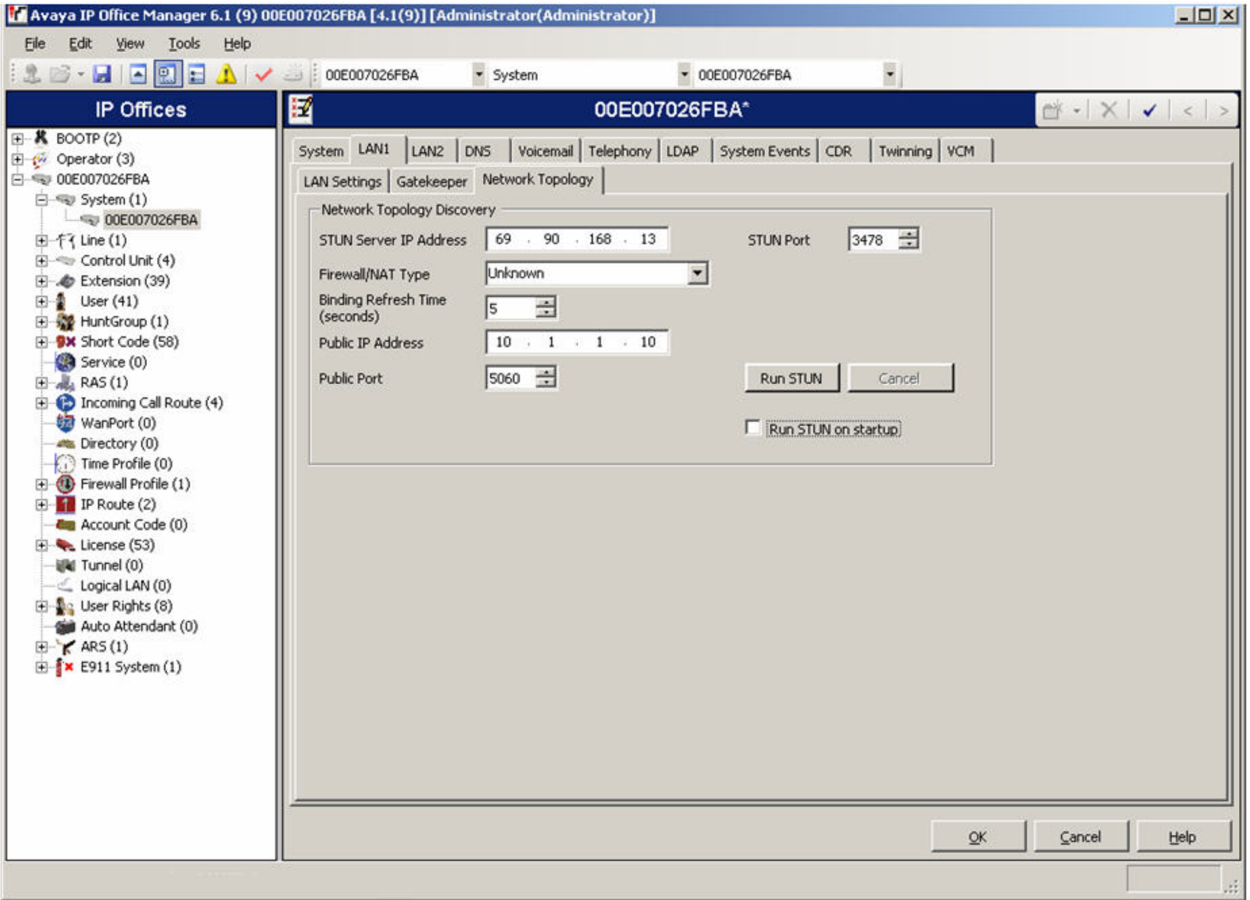
The other fields can be left as defaults. Press the **OK** button.



9. Repeat **Steps 7** and **8** for all users that will be sending/receiving SIP calls on the system.

10. *Firewall/NAT Considerations*

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 [3]. For this testing, a NAT device was not used and therefore the **Firewall/NAT Type** was left as the default value of **Unknown**. Press the **OK** button.



4. PAETEC Services Configuration

The PAETEC Broadsoft based SIP Trunking solution offers the growing number of business customers with IP-based phone systems, many different configurations that can be scaled based on specific voice, data and configuration needs.

Customers can choose to terminate service directly to their IP phone system, to an external router provided by the customer, or to an Integrated Access Device (IAD) with the standard and premium options provided by PAETEC. If the customer requires an Ethernet hand-off to their equipment, the PAETEC standard or premium equipment option is required.

The standard product solution is available in bandwidths ranging from 1.5 Mbps up to 12.0 Mbps in single DS1 increments. Solutions with PAETEC-provided equipment are limited to 6.0 Mbps. Each solution includes a minimum of 6 simultaneous call paths, with the ability to scale to the customer's specific business need in increments of 6 call paths. The standard configuration includes 30 call paths for each 1.5 Mbps of bandwidth, although up to 48 call paths for each 1.5 Mbps of bandwidth can be accommodated if "non-voice" bandwidth is not required.

For customers selecting the standard or premium equipment option, PAETEC also includes access to 3 FXS ports (per 1.5MB) on the IAD for customer use, typically for analog device support. Customers must identify all numbers that are associated with analog devices so that voice compression is disabled.

All telephone numbers associated with the PAETEC Broadsoft based SIP Trunking solution must be in a PAETEC-served rate center within approximately 1,000 central offices in a 20-state footprint. This includes numbers that will be ported to the service or numbers requested as new in conjunction with the installation of service. Specific availability is available at www.mcleodusa.com

The product solution is sold by PAETEC field sales executives and by other Direct and Indirect Sales channels such as Master Agents, their sub Agents, Lead Agents and through Referral Agents. Companies interested in becoming a PAETEC Business Partner may visit www.mcleodusa.com/IndirectSales.do for more information.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between a PAETEC Broadsoft based SIP Trunking solution 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 Broadsoft based SIP Trunking solution provided by PAETEC. 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.
- Outgoing calls from the enterprise site were completed via PAETEC 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, and directory assistance calls.
- Calls using G.729a codec types.
- Fax routing to ensure G.711 use for fax calls.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with G.729a.
- Telephone features such as hold, transfer, conference.

6. Test Results

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

The following observations were noted:

- Inbound calls to IP Office using the G.711 codec will drop after 5 minutes. Therefore, as stated in **Step 4**, G.729a should be configured as the preference on the SIP line. This will ensure that all voice call traffic will use this codec. However, inbound FAX calls will be subject to this limitation.

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 PAETEC'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.

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 questions regarding a customer installation, please call the PAETEC toll-free number provided in the customer information packet. For general, non-technical questions, please contact PAETEC Customer Care at 1-800-593-1177.

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to PAETEC's service. PAETEC 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 4.1 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 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 PAETEC 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 PAETEC to Avaya IP Office:

```
INVITE sip:2814022048@10.1.1.10:5060;eplid=10.1.1.10:5060:5060;elid=10.1.1.10:5060;evlid=16;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bK00004fc60000e6a60002
From: "AVAYA INC C/O T" <sip:7328521637@10.2.2.10;user=phone>;tag=177731643-1184614393685-
To: "7133433752 7133433752" <sip:7133433752@generic.paetec.net;eplid=10.1.1.10:5060:5060;elid=10.1.1.10:5060;evlid=16>
Call-ID: 4c4c4143-6400004fc6@10.2.2.10
CSeq: 675451819 INVITE
Contact: <sip:10.2.2.10:5060;transport=udp>
supported:
max-forwards: 10
Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS, PRACK, REFER, NOTIFY
Content-Type: application/sdp
Accept: multipart/mixed, application/media_control+xml, application/sdp
Content-Length: 00363

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): BroadWorks 96025 1 IN IP4 10.2.2.11
Session Name (s): -
Connection Information (c): IN IP4 10.2.2.11
Time Description, active time (t): 0 0
Media Description, name and address (m): audio 39584 RTP/AVP 18 2 0 8 4 101
Media Attribute (a): sendrecv
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute (a): rtpmap:2 G726-32/8000
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute (a): rtpmap:8 PCMA/8000
Media Attribute (a): rtpmap:4 G723/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15
Media Attribute (a): fmtp:18 annexb=no
Media Attribute (a): fmtp:4 annexa=no
Media Attribute (a): fmtp:4 bitrate=6.3
```

Sample SIP INVITE Message from Avaya IP Office to PAETEC:

INVITE sip:17324501327@generic.paetec.net SIP/2.0
Via: SIP/2.0/UDP 10.1.1.10:5060;rport;branch=z9hG4bK11b89950ec7beb5f4d4902db759396f2
From: 7133433752 <sip:7133433752@generic.paetec.net>;tag=92803ca513e5344d
To: <sip:17324501327@generic.paetec.net>
Call-ID: ede688606659867da20684367c77f2b2@10.1.1.10
CSeq: 1762978545 INVITE
Contact: 7133433752 <sip:7133433752@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 244833631 3701624698 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.