Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Bandwidth.com 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 Bandwidth.com 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.

Bandwidth.com is a national provider covering over 300 US markets offering complete business communications with services such as advanced VoIP, Internet Services and Managed Network Services to small and medium businesses.

Bandwidth.com 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.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Bandwidth.com 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.

Bandwidth.com is a national provider covering over 300 US markets offering complete business communications with services such as advanced VoIP, Internet Services and Managed Network Services to small and medium businesses.

SIP Trunking from Bandwidth.com delivers local and long-distance voice service over either a Bandwidth.com provided Internet connection or a customer provided Internet connection.

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

Customers looking for a turnkey access solution can select a Managed Service option, where Bandwidth.com provides and manages the router that interfaces with customer equipment.
A Bandwidth.com SIP Trunking 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 a Bandwidth.com SIP 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 6210 Analog Telephone
- Avaya Voice Mail Pro

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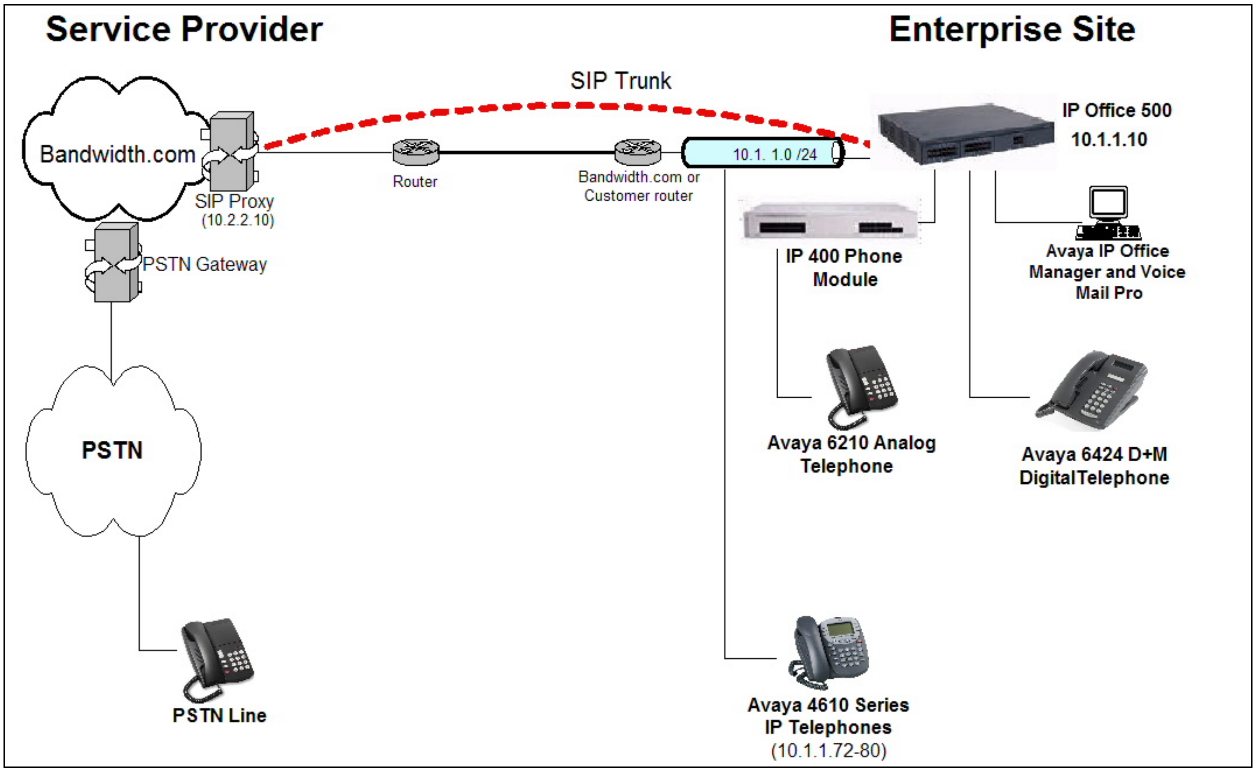

Figure 1: Avaya IP Telephony Network using Bandwidth.com SIP Trunking Solution

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 4.1.12
Avaya IP Office Manager (Windows PC)	R 6.1 (12)
Avaya Voice Mail Pro	R 4.1 (27)
Avaya 4610SW IP Telephone	R2.8.3 – H.323
Avaya 6424D+M Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Bandwidth.com SIP Trunking Solution Components	
Bandwidth.com Network Version	1.3.1 gold trm

Table 1: Equipment and Software Tested

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.1 (12).

3. Configure Avaya IP Office

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

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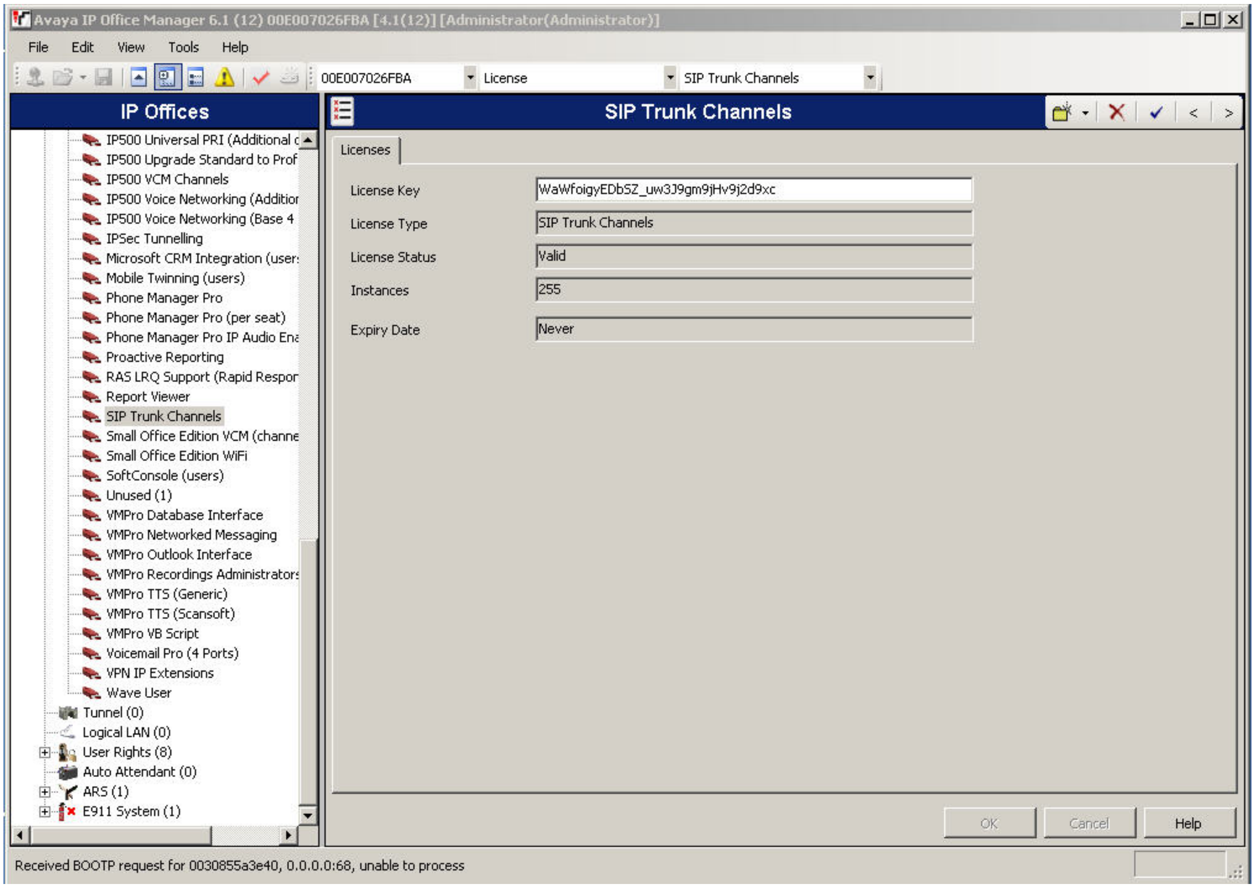
IP Office is configured via the IP Office Manager program. Log into the IP Office Manager PC and select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Log into the Manager application using the appropriate credentials.

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



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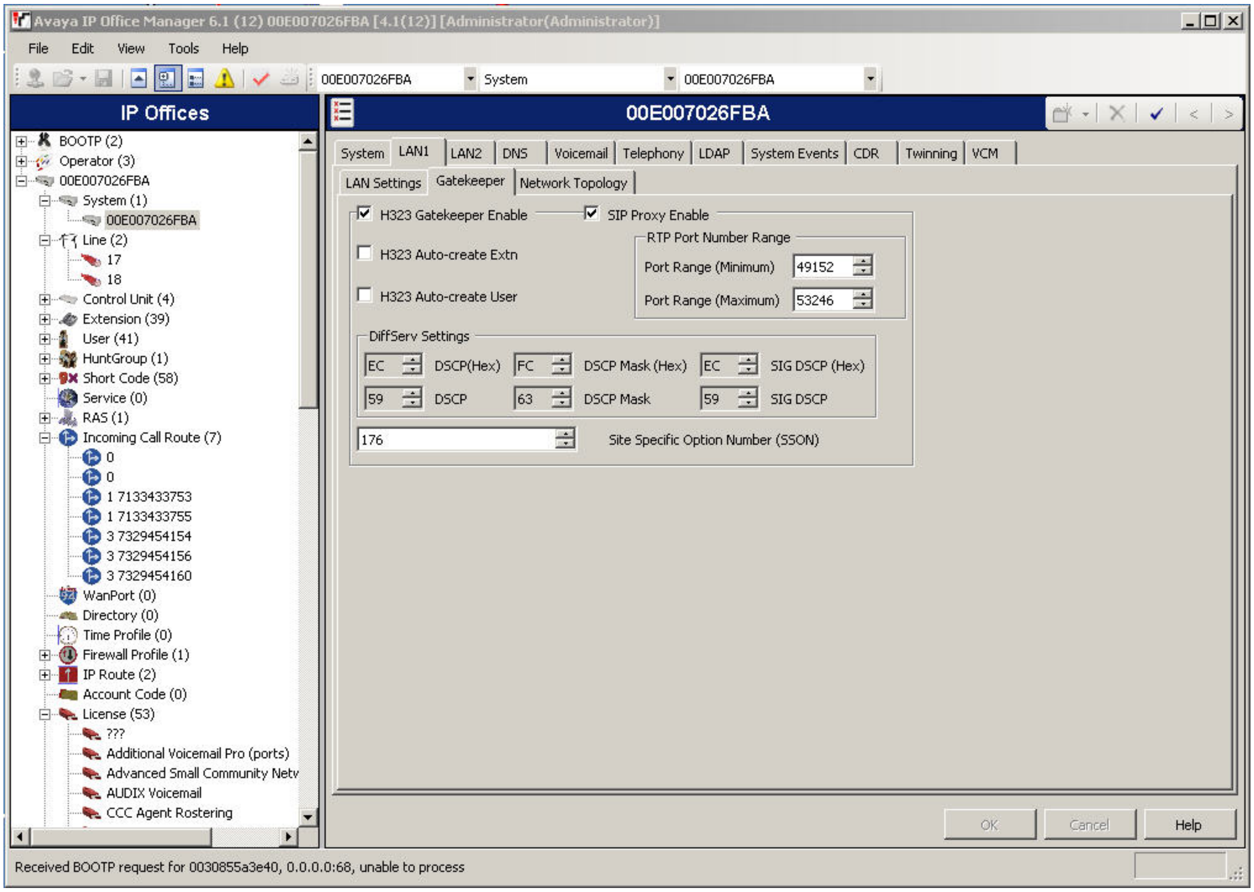
2. *Enable SIP Proxy Functionality.* Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper tab,** and check the **SIP Proxy Enable** box.



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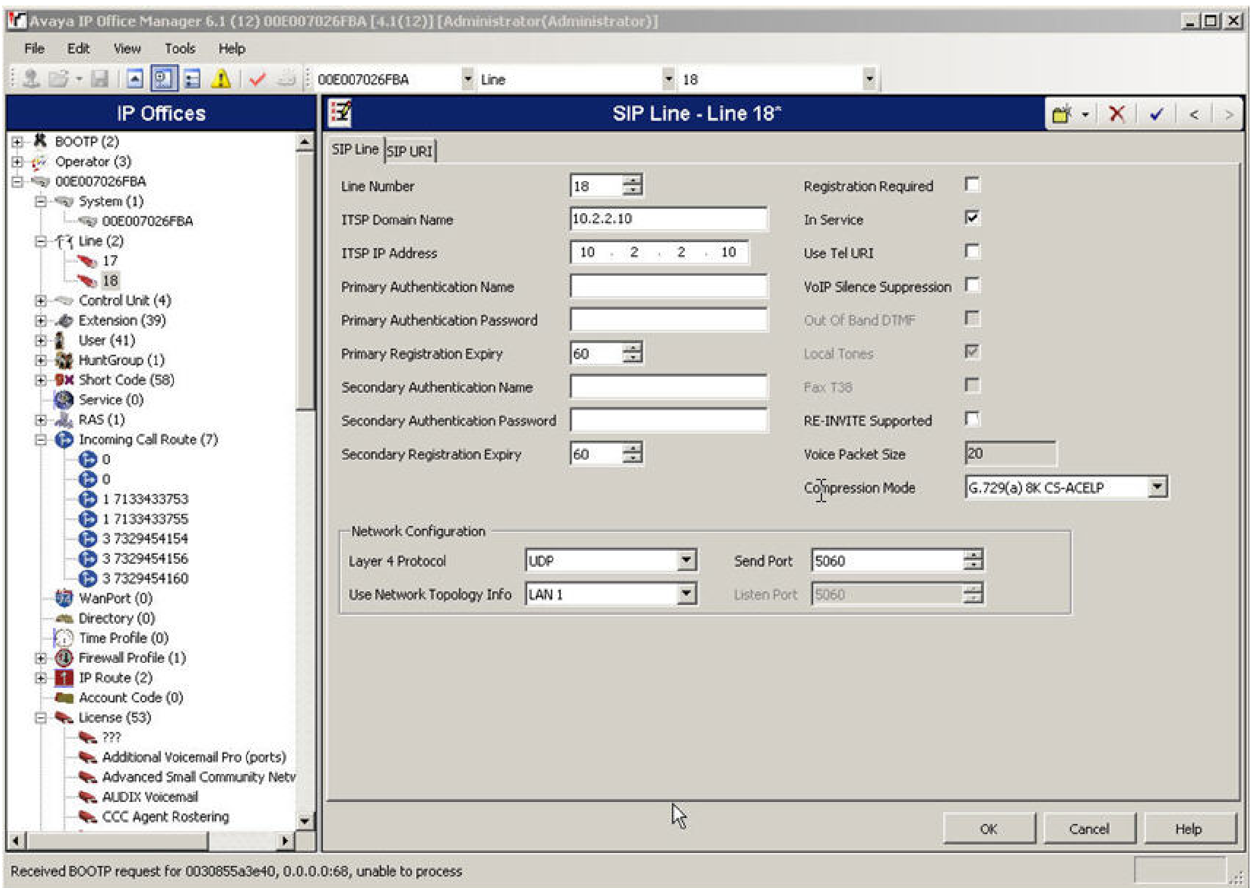
3. *Create a SIP line for the Bandwidth.com service.* Right-click on **Line** and select **New → SIP line**.

Select the following:

- For the **ITSP Domain Name** field, enter the IP address of the Bandwidth.com SIP Proxy.
- For the **ITSP IP Address** field, enter the IP address of the Bandwidth.com SIP Proxy
- For **Registration Required**, leave this disabled.
- For **Compression Mode**, select the **G729a 8K CS-ACELP** or **G.711mu** for voice calls.
- 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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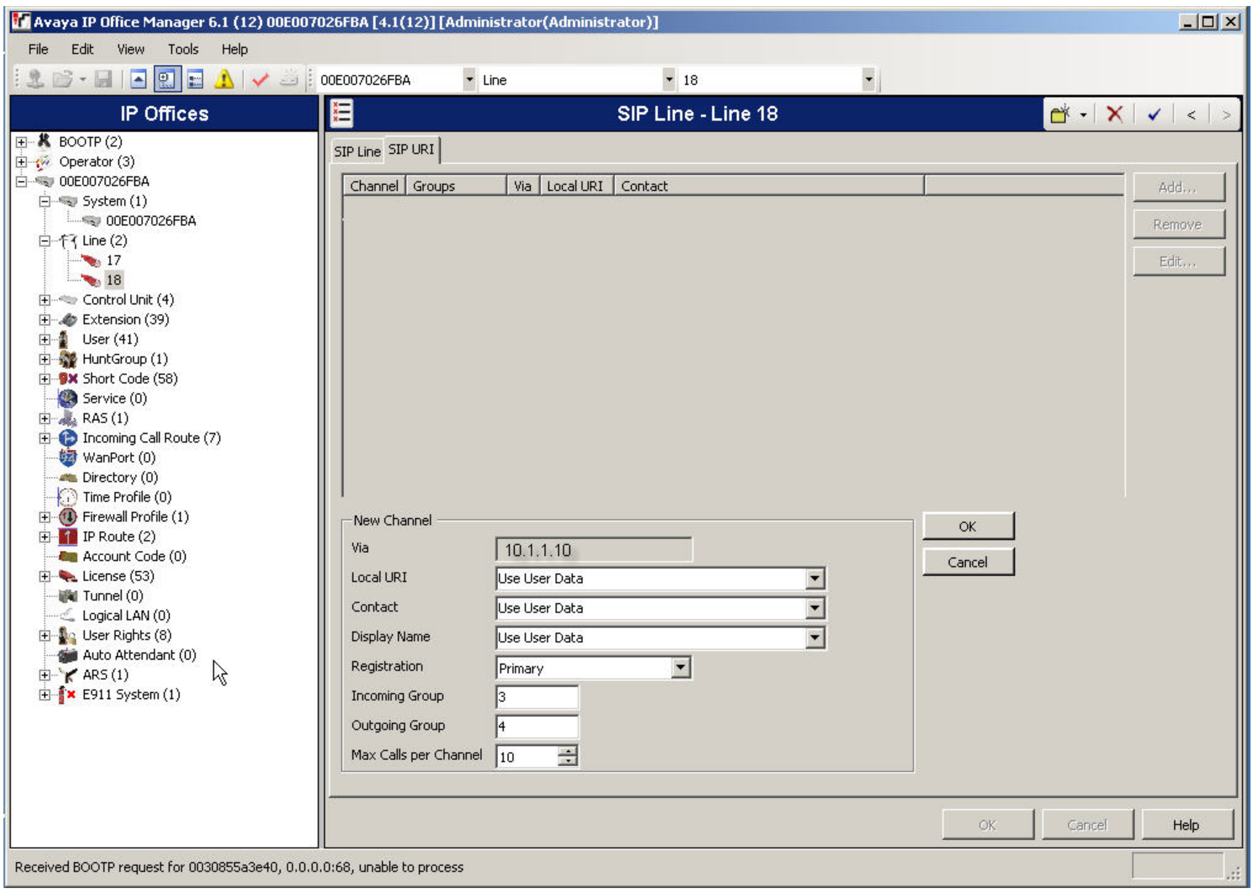
4. *Configure URI parameters for the line*. Select the **SIP URI** Tab. Press the **Add** button.

   Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP Trunk to local stations, see **Step 6**. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in **Step 5**. 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 when building the SIP headers (see **Step 7**). Use defaults for all other fields. Press the **OK** button. *Note: The **Incoming Group** and **Outgoing Group** fields can share the same value if so desired.*



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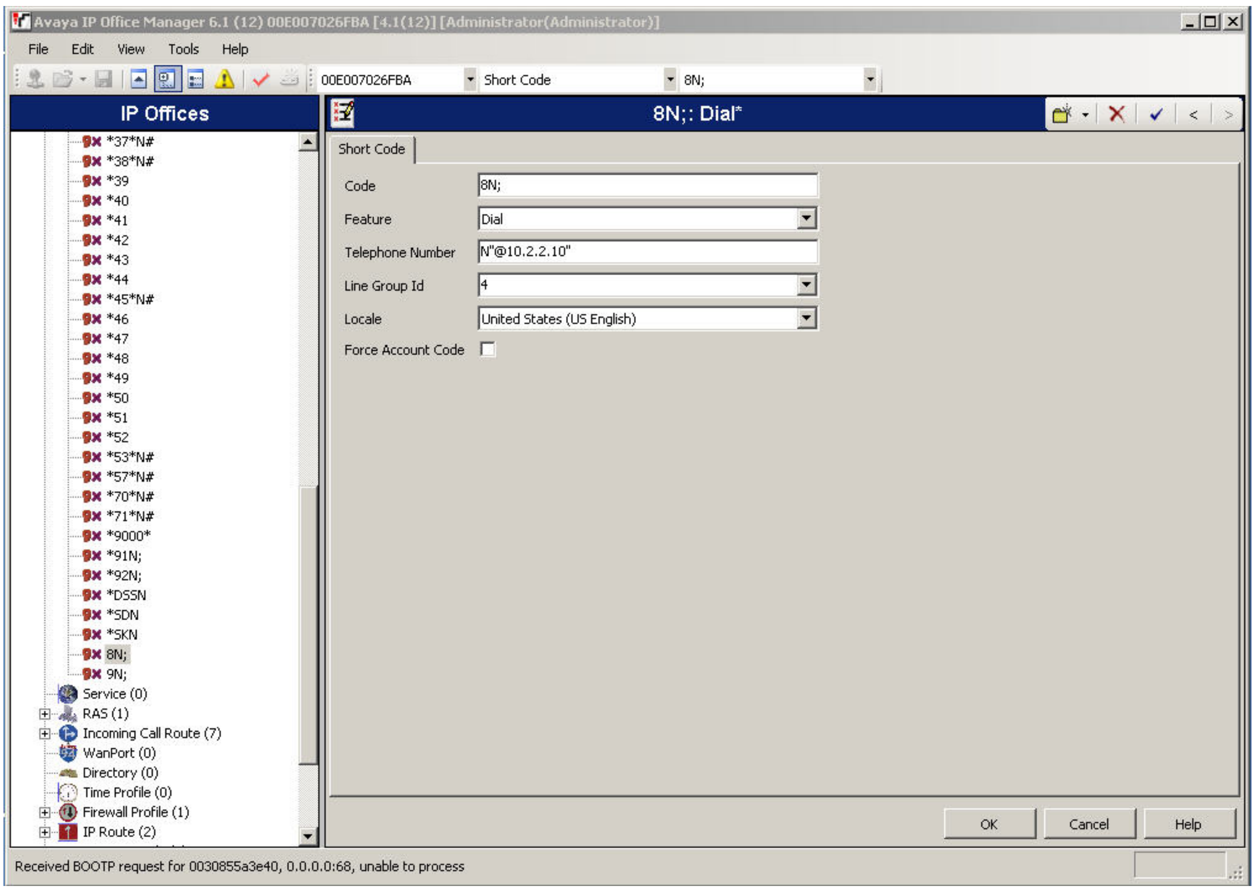
5. *Configure a short code to route calls to the Bandwidth.com SIP Proxy Server*. Select **Short Code** in the left panel. Right click and select **Add**. Enter **[x]N;**, where **[x]** is a valid number, in the **Code** text box. The number **8** is used for **[x]** in the below example. This code requires the user to dial the digit **8** 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 PSTN telephone number, operator assistance, 411, information service etc.

   Select **Dial** for the **Feature**. Enter the **Outgoing Group Id** created in **Step 4** for the **Line Group Id** field. Enter the dialed number **N** followed by **"@<IP Address of bandwidth's SIP Proxy >"** 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.



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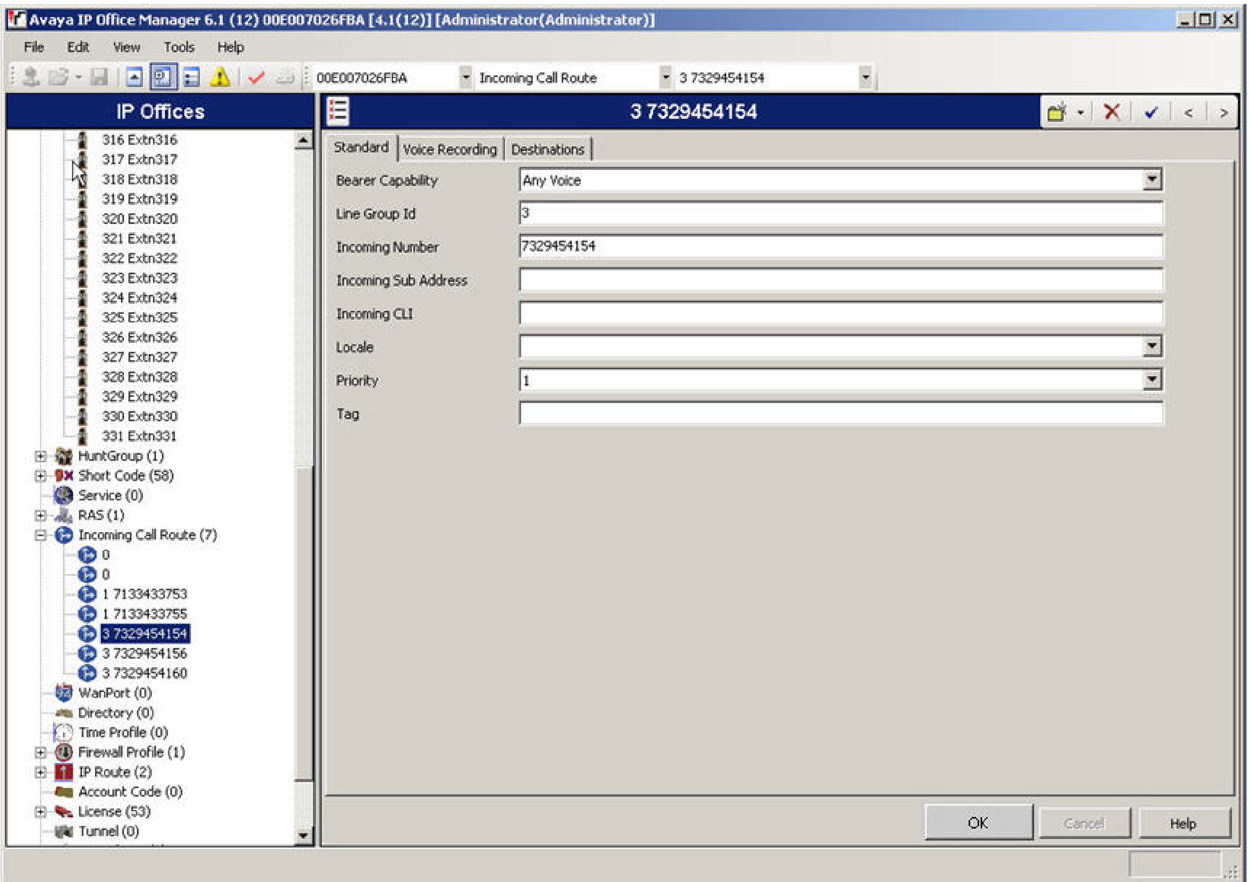
6. *Create an Incoming Call Route for the Inbound SIP calls.* Select **Incoming Call Route** in the left panel. Right-click and select **New**.

   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 Bandwidth.com, 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.



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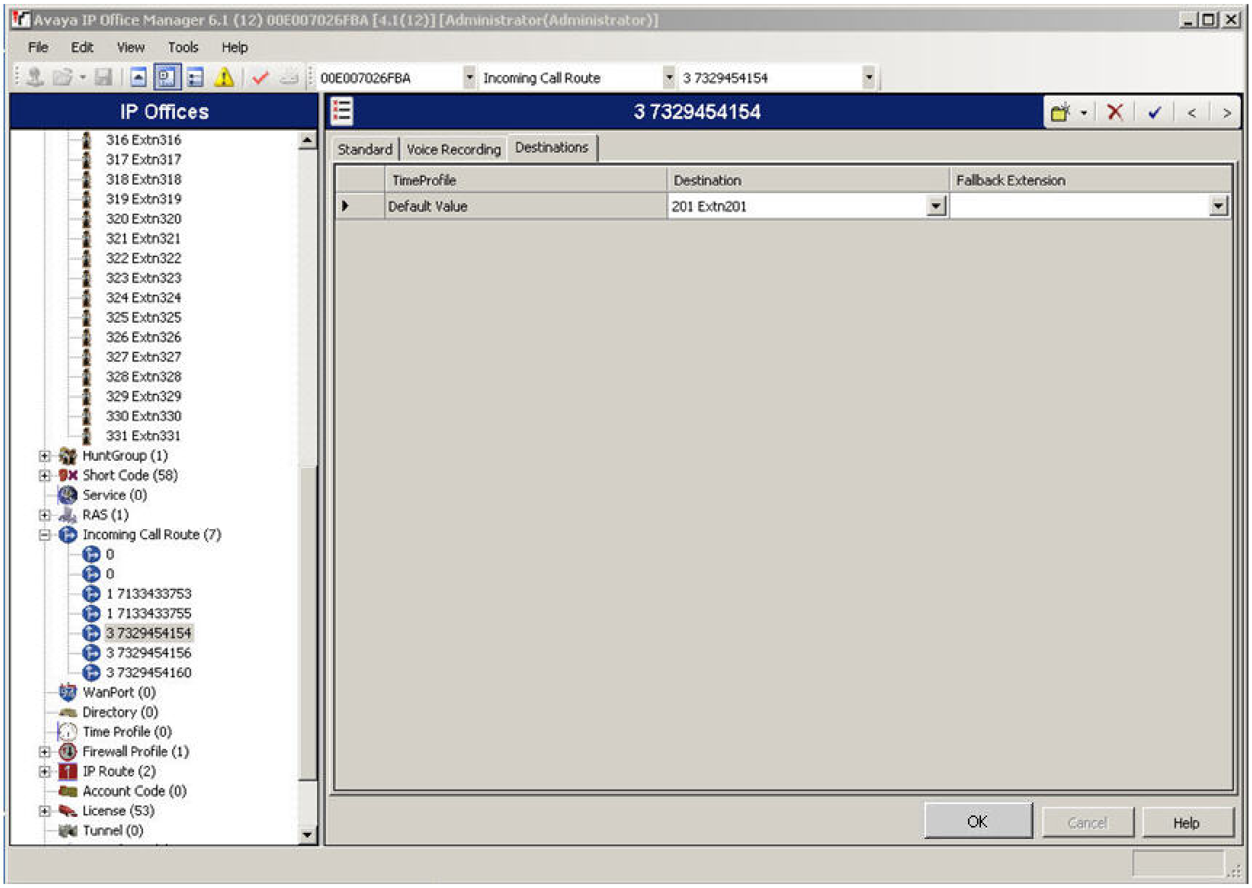
- Next, navigate to the **Destinations** tab, select the desired local extension number from the drop down menu.
- Press the **OK** button.



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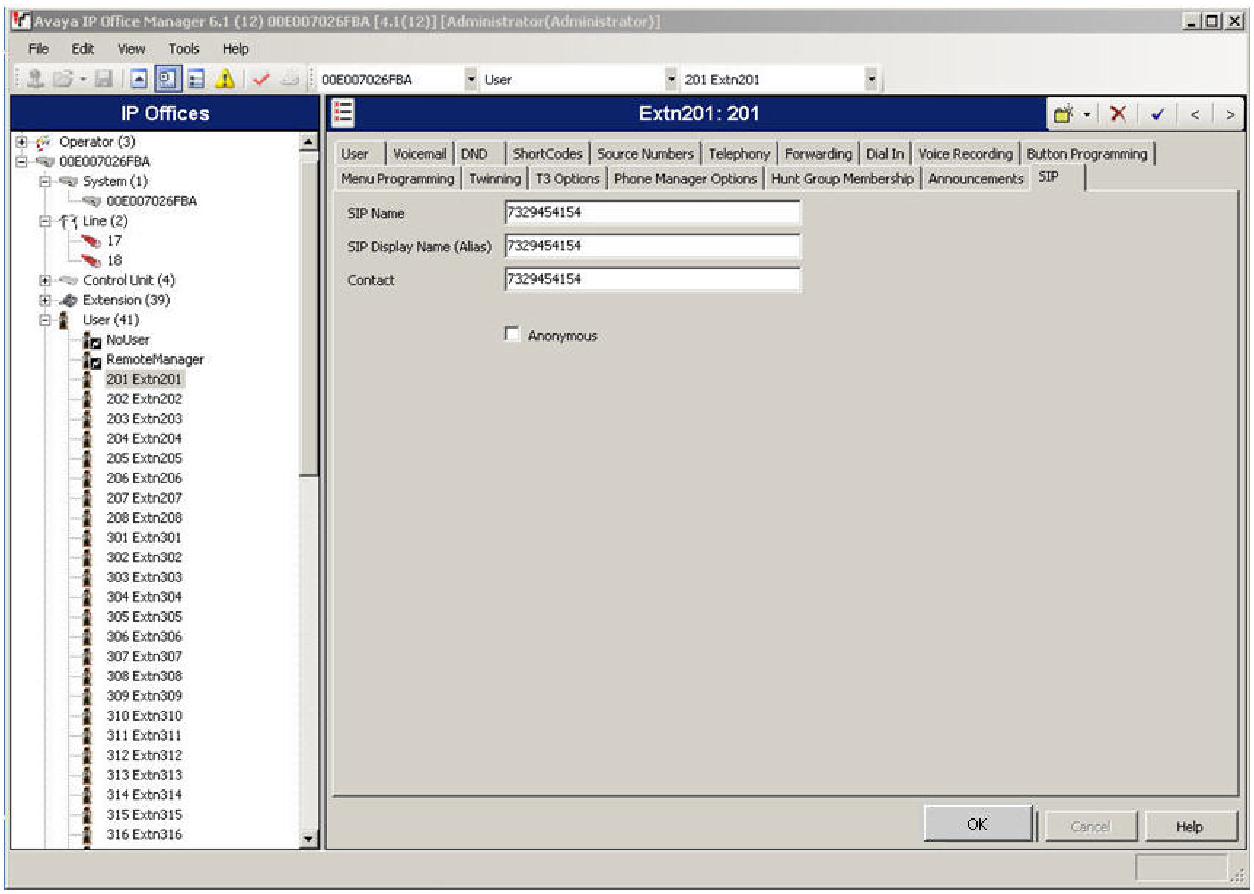
7. *Configure Users' SIP names*. Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab.

   Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to the telephone number provided by Bandwidth.com that is used for this particular extension. These settings instruct the system to use this number 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.



8. Repeat **Steps 6** and **7** for all users that will be sending/receiving external SIP calls to/from the PSTN.

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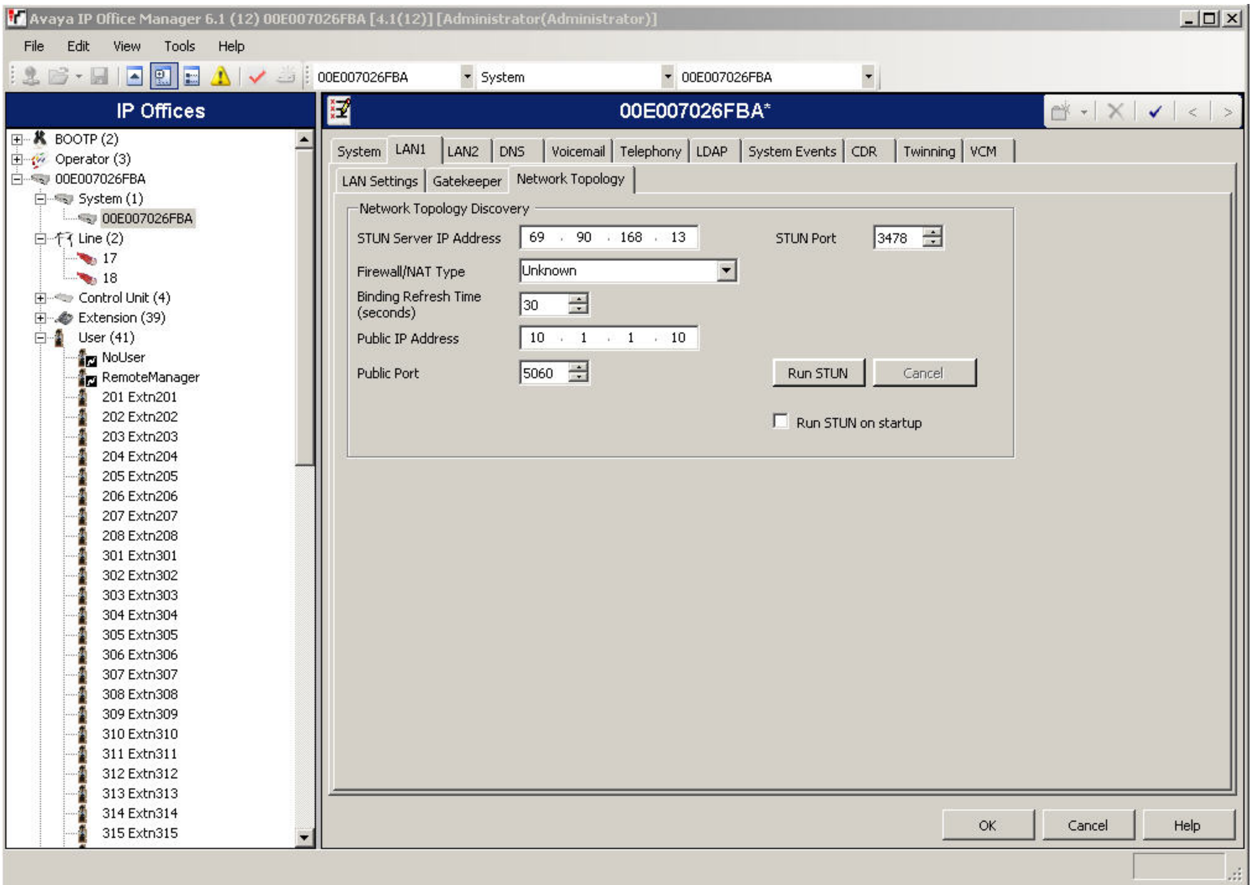
9. Firewall/NAT Considerations

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



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4. Bandwidth.com Services Configuration

Bandwidth.com SIP Trunking service is typically sold via the Bandwidth.com BoxSet bundles. Following is a brief description of the BoxSet, for more detailed information please visit: http://www.bandwidth.com/content/boxset.

Bandwidth.com SIP Trunking service can be ordered from any authorized partner or directly from Bandwidth.com.

Available Volumes	vol. 1	vol. 2	vol. 3	vol. UNPLUGGED
All Volumes Include				
Internet Services	SDSL* Up to 1.5Mb	T-1 1.5Mb	Bonded T-1 3.0Mb	Customer Provided**
Voice Lines	5 Up to 8***	5 Up to 16***	16 Up to 30***	5 Up to 30***
Calling Plan	UNLIMITED	UNLIMITED	UNLIMITED	UNLIMITED
Toll-Free Numbers	1 Included	1 Included	1 Included	1 Included

To find a Bandwidth.com Authorized Partner or for information on becoming a partner call: **(888)292-8277.**

For pricing from the Direct Sales team call: **(800) 808-5150**

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5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Bandwidth.com 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 Bandwidth.com. 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 Bandwidth.com.
- Outgoing calls from the enterprise site were completed via Bandwidth.com 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 and G.711mu codec types.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with G.729a and G.711mu.
- Telephone features such as hold, transfer, conference.
- Outbound 911 Emergency calls.

6. Test Results

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

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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 using the Bandwidth.com 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

Bandwidth.com Box Set comes with comprehensive customer care and support tools.

- **24/7/365 Domestic Support**
- **Web-based monitoring of the installation process**
- **Web-based portal for ticketing, account and service management**

Three methods for support:

1. Call **(800) 808-5150**
2. Email customercare@bandwidth.com
3. Portal http://my.bandwidth.com

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to Bandwidth.com's service. Bandwidth.com 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/

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APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by Bandwidth.com 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 Bandwidth.com to Avaya IP Office:

Request-Line: INVITE sip:7329454156@10.1.1.10:5060;transport=udp SIP/2.0
Message Header
Record-Route:
<sip:10.2.2.10;lr;ftag=VPSF506071629460;vsf=AAAAABwCBQcHBAUIAQFxDW4CFhgKGwIbARoJNDg->
Record-Route: <sip:10.3.3.10;lr;ftag=VPSF506071629460>
Via: SIP/2.0/UDP 10.2.2.10;branch=z9hG4bK4ec1.89e4b7b6.0
Via: SIP/2.0/UDP 10.3.3.10;branch=z9hG4bK4ec1.32942ab5.0
Via: SIP/2.0/UDP 10.4.4.10:5060;branch=z9hG4bK506071629460-1206659924318
From: "AVAYA INC   " <sip:7324500819@4.68.250.148>;tag=VPSF506071629460
To: <sip:7329454156@10.3.3.10:5060>
Call-ID: WEEMGC0120080522124736057150@209.244.63.28
CSeq: 1 INVITE
Contact: <sip:+17324500819@10.4.4.10:5060;transport=udp>
Max-Forwards: 67
Content-Type: application/sdp
Content-Length: 173

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 1211460456 1211460457 IN IP4 10.2.210
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 60504 RTP/AVP 0 18 101
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): fmtp:101 0-15

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Sample SIP INVITE Message from Avaya IP Office to Bandwidth.com:

Request-Line: INVITE sip:7324500819@10.2.2.10 SIP/2.0
Via: SIP/2.0/UDP 10.1.1.10:5060;rport;branch=z9hG4bK18b5ad074b2f98dbc8a03a5e248eaffa
From: 7329454156 <sip:7329454156@10.2.2.10>;tag=85c59a7996d4e3a6
To: <sip:7324500819@10.2.2.10>
Call-ID: 3f80c6a2e3e791b197cd69ec107d6426@10.1.1.10
CSeq: 1230656751 INVITE
Contact: 7329454156 <sip:7329454156@10.1.1.10:5060;transport=udp>
Max-Forwards: 70
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
Content-Type: application/sdp
Content-Length: 301

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): UserA 2779165961 1993594873 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

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