Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Cincinnati Bell Any Distance eVantage and Avaya IP Office – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between Cincinnati Bell Any Distance eVantage and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

Cincinnati Bell, Inc. 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 Cincinnati Bell Any Distance (CBAD) eVantage solution and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office, and Avaya H.323, digital and analog endpoints.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [5] is the primary specification governing this protocol. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between the Avaya IP Office and the network services offered by Cincinnati Bell Any Distance eVantage solution.

The CBAD eVantage solution is a turn-key business trunking solution for customers. eVantage provides customers with a single IP connection that converges voice and data services to drive optimization, reduce costs, and offer enhanced features not typically available in the traditional PSTN network. Voice services, such as local, long distance, and toll free calling, as well a high speed data and Internet services, are the primary applications of the eVantage solution.

1.1. Interoperability Compliance Testing

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 CBAD eVantage solution. 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 Cincinnati Bell.
- Outgoing calls from the enterprise site were completed via CBAD eVantage 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, and toll free calls.
- Calls using the G.729(a) and G.711 ULAW codecs.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation.
- Telephone features such as hold, transfer, conference, and call forwarding.
- Mobility Features: Mobile twinning to a mobile phone.
- IP Office Phone Manager in either Telecommuter or Softphone mode

1.2. Support

For technical support on Cincinnati Bell Any Distance eVantage solution, customers can call 1-866-914-9474.

2. Reference Configuration

Figure 1 illustrates an example Avaya IP telephony solution connected to the Cincinnati Bell Any Distance eVantage 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 1603SW IP Telephone (H.323 protocol)
- Avaya 4610SW IP Telephone (H.323 protocol)
- Avaya 5620 Digital Telephone
- Avaya 5420 Digital Telephone
- Avaya 6210 Analog Telephone
- Avaya IP Office Phone Manager

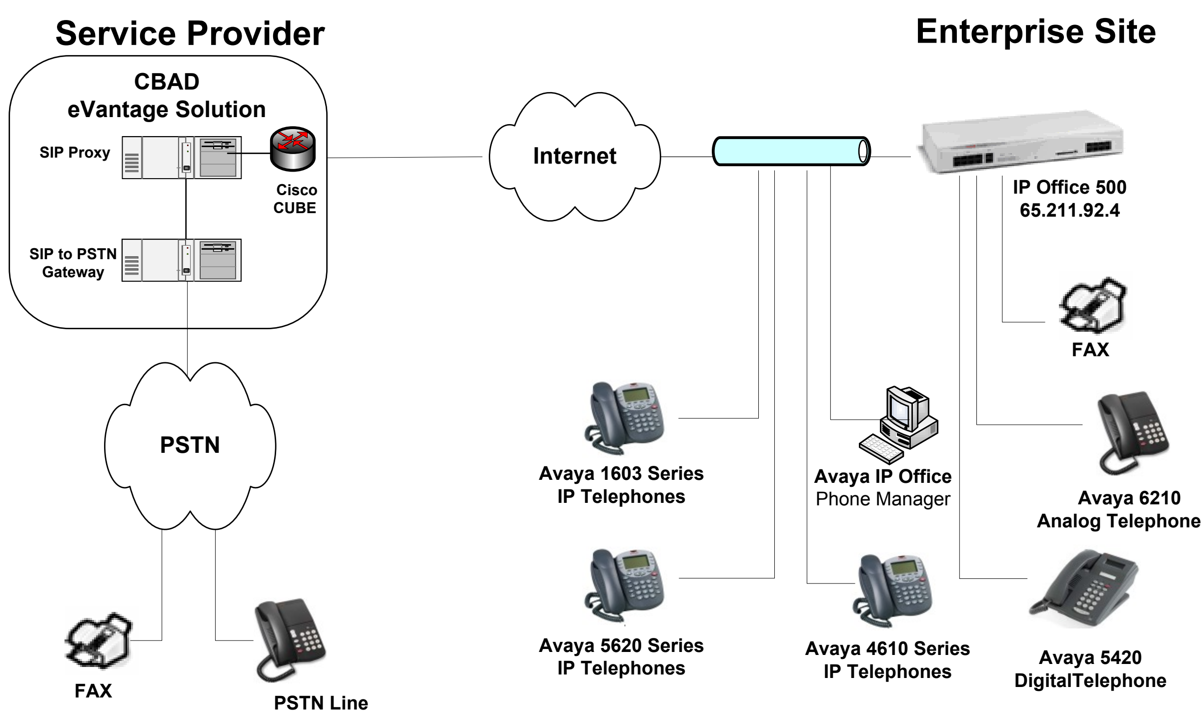


Figure 1: Avaya IP Telephony Network CBAD eVantage Solution

3. 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 (17)
Avaya IP 400 Analog POTS 30+	R 6.2 (17)
Avaya IP Office Manager (Windows PC)	R 6.2 (17)
Avaya IP Office Voicemail Pro	R 4.2.(30)
Avaya 1603 IP Telephone	R10483
Avaya 4610SW IP Telephone	R2.9.1
Avaya 5620 IP Telephone	R2.9.1
Avaya 6210 Analog Telephone	n/a
Avaya 5420 Digital Phone	R5 Firmware
Avaya IP Office Phone Manager	R4.2.23
Cincinnati Bell SIP Trunk Service Solution Components	
Cisco CUBE	12.4(15)T8

This solution is compatible with all other Avaya IP Office platforms running IP Office software release 4.2 (17).

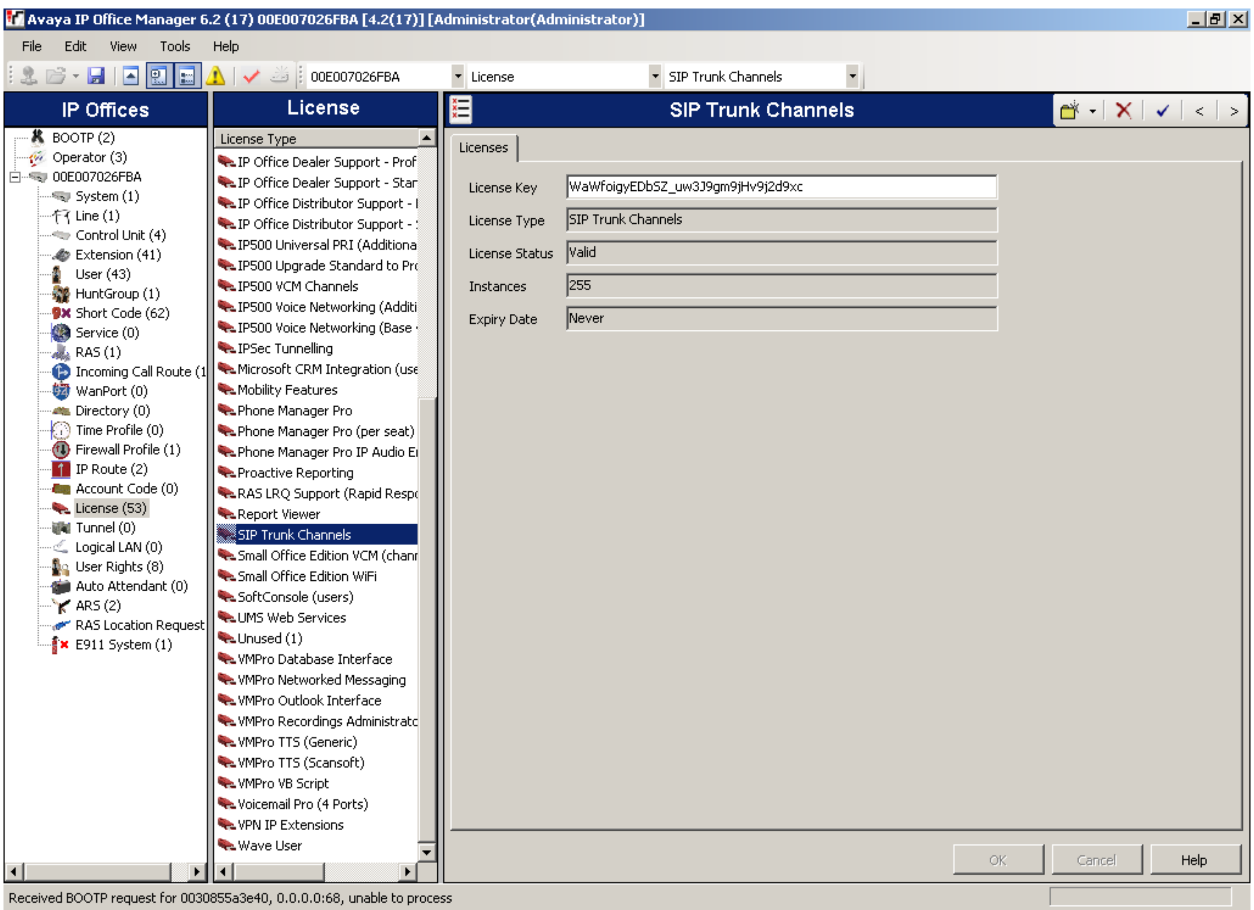
4. Configure the Avaya IP Office

This section describes the steps for configuring a static 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.*
   Click on **License** in the left panel. Confirm that there is a **SIP Trunk Channels** entry.

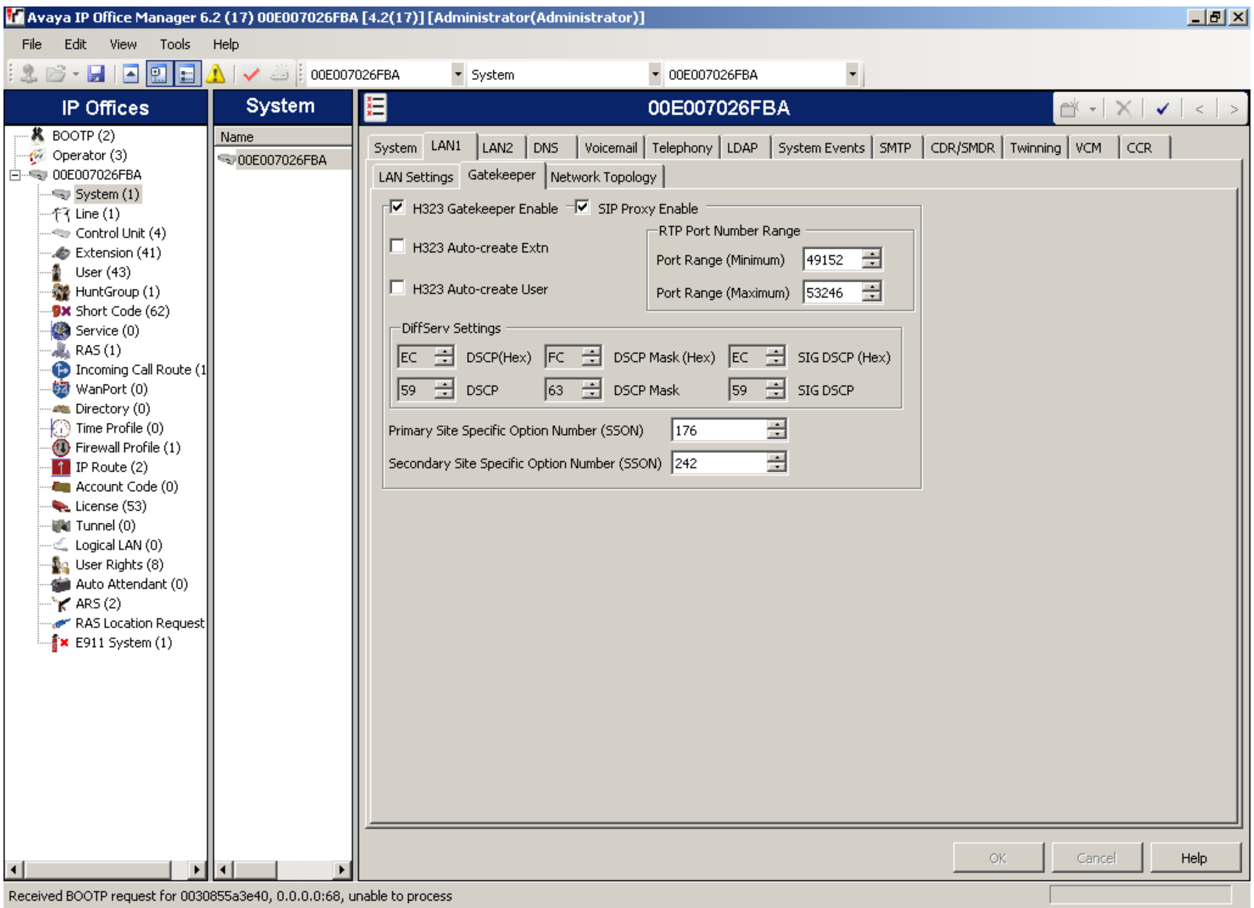
   If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



2. *Enable SIP Proxy Functionality and DiffServ Settings.*
   Select **System** in the left panel. Click the **LAN1** tab and select the **Gatekeeper tab.**

   Modify the following:
   - Check the **SIP Proxy Enable** box.
   - 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.
   - Use default values for other fields.

   Click the **OK** button.

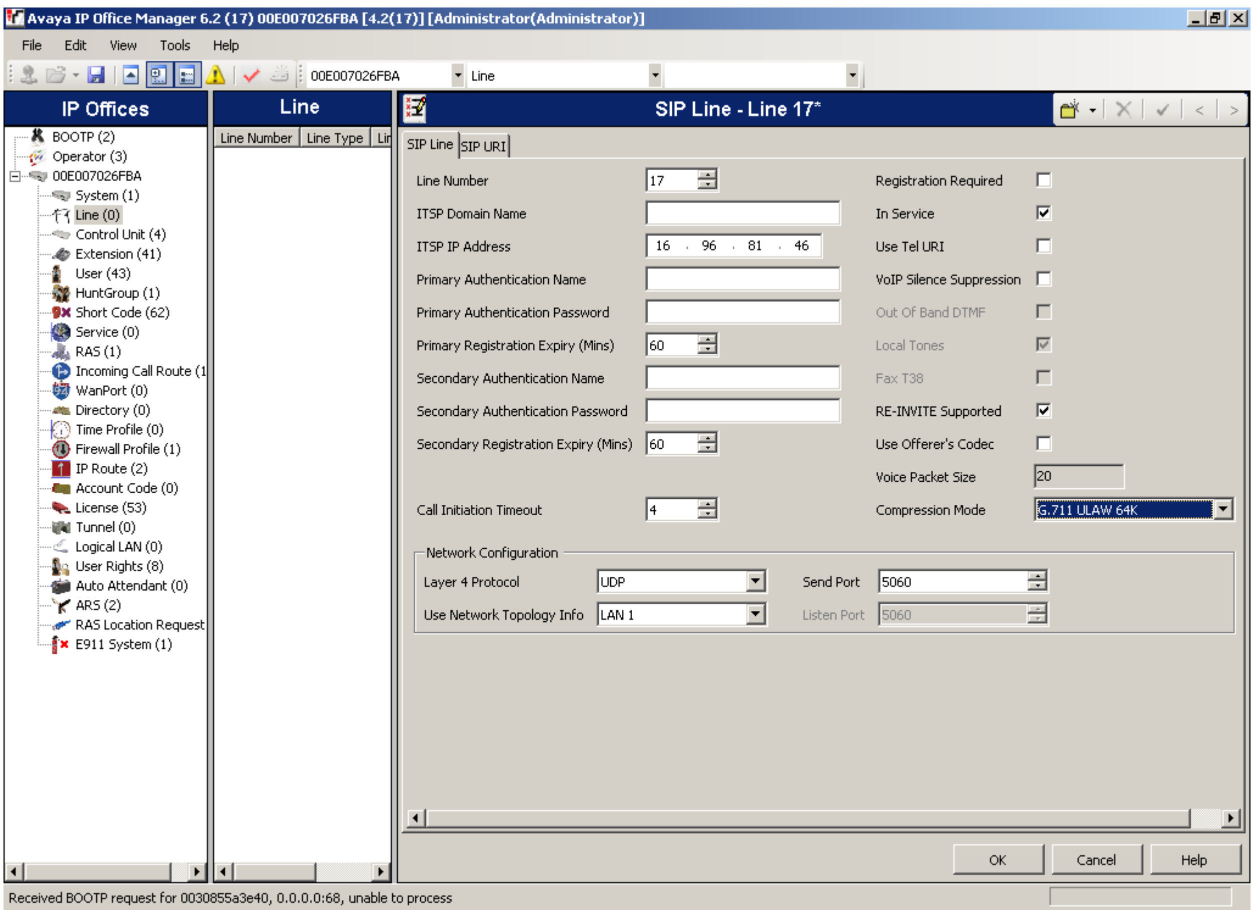


3. *Create the static SIP line*.
Select **Line** in the left panel. Right-click and select **New** → **SIP Line**.

Configure the following:

- For the **ITSP IP Address** field, enter the IP address of the CBAD eVantage solution service SIP Proxy.
- For **Compression Mode**, select the **G.729(a) 8K CS-ACELP** or **G.711 ULAW 64K** for voice calls. In order to use the G.729(a) codec for this solution, see Appendix B for configuration details.
- Click on **RE-INVITE Supported**.
- Use default values for other fields.

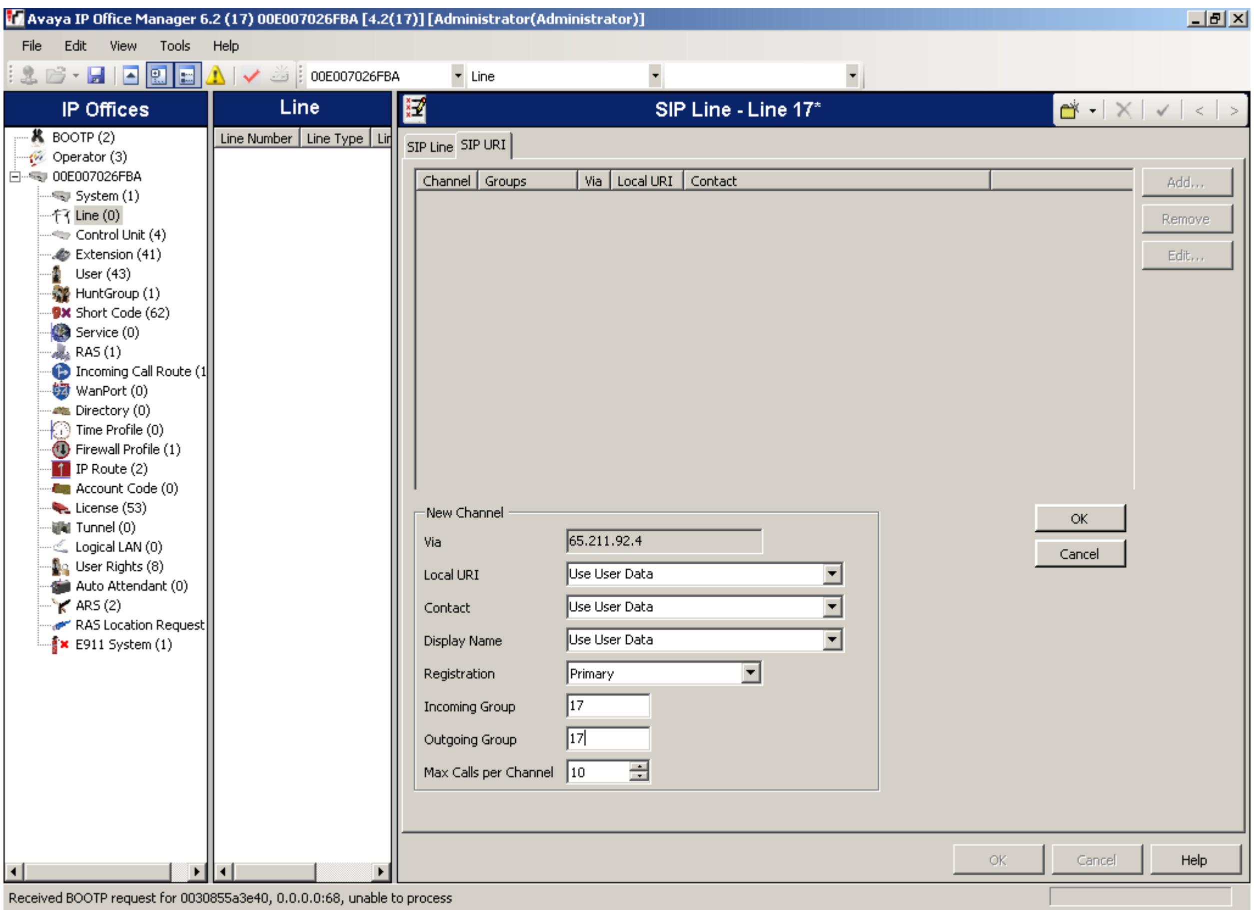
Click the **OK** button.



4. *Configure SIP URI parameters for the SIP Line*.
   Select the **SIP URI** Tab. Click the **Add** button.

   Configure the following:
   - 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 8).
   - 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 7. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in step 6.
   - Use default values for all other fields.

   Click the **OK** button.



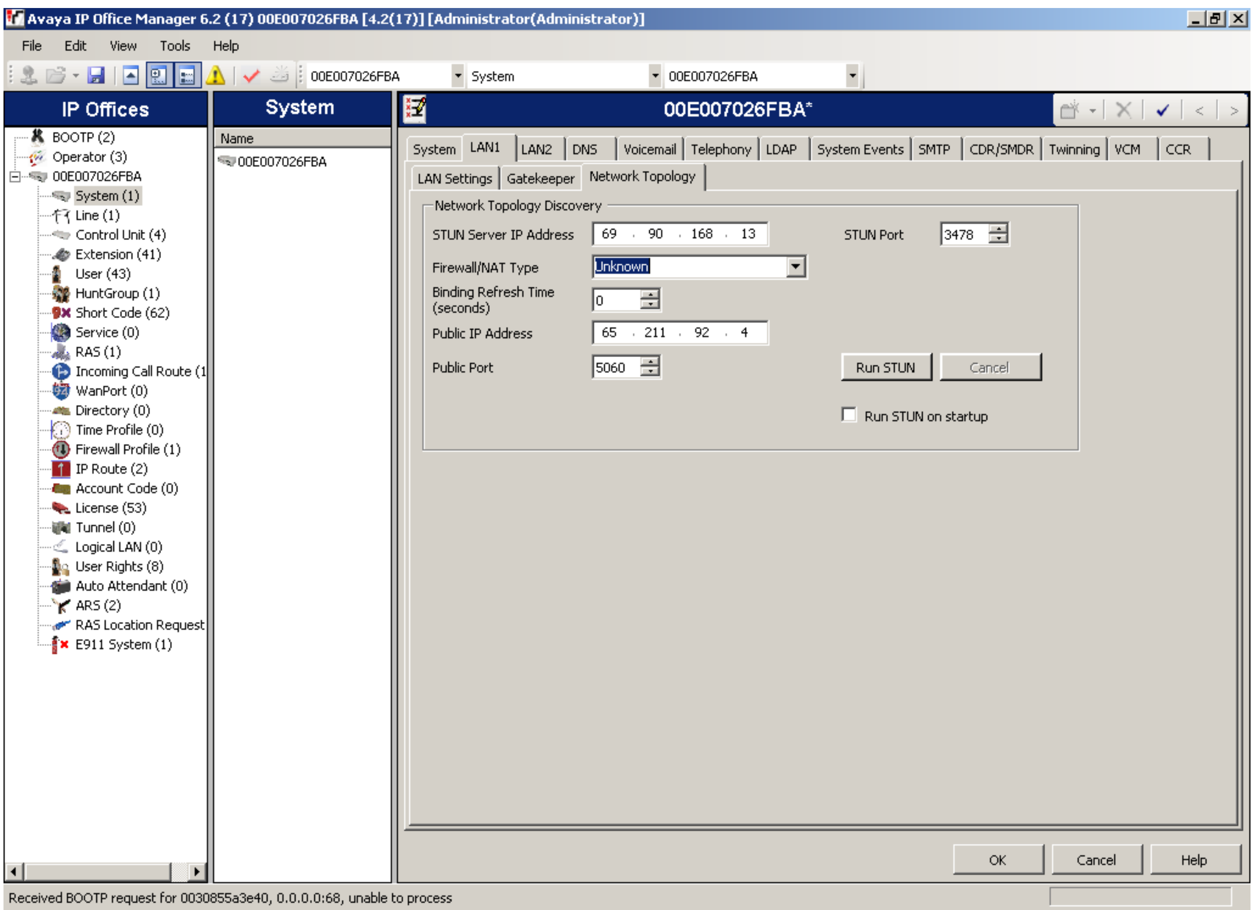
5. *Configure SIP OPTIONS timer on Network Topology Tab for "keep alive" function.*
Select **System** in the left panel. Select the **LAN1** tab and select the **Network Topology** tab.

Configure the following:

- Set the **Binding Refresh Time** to the desired interval which determines the frequency with which OPTIONS messages will be sent to CBAD eVantage solution. The solution was tested with no OPTIONS messages sent.
- For **Public IP Address**, enter the public or private IP address to reach Avaya IP Office.
- Confirm that **Public Port** is set to 5060.
- Use default values for all other fields.

Click the **OK** button.

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].

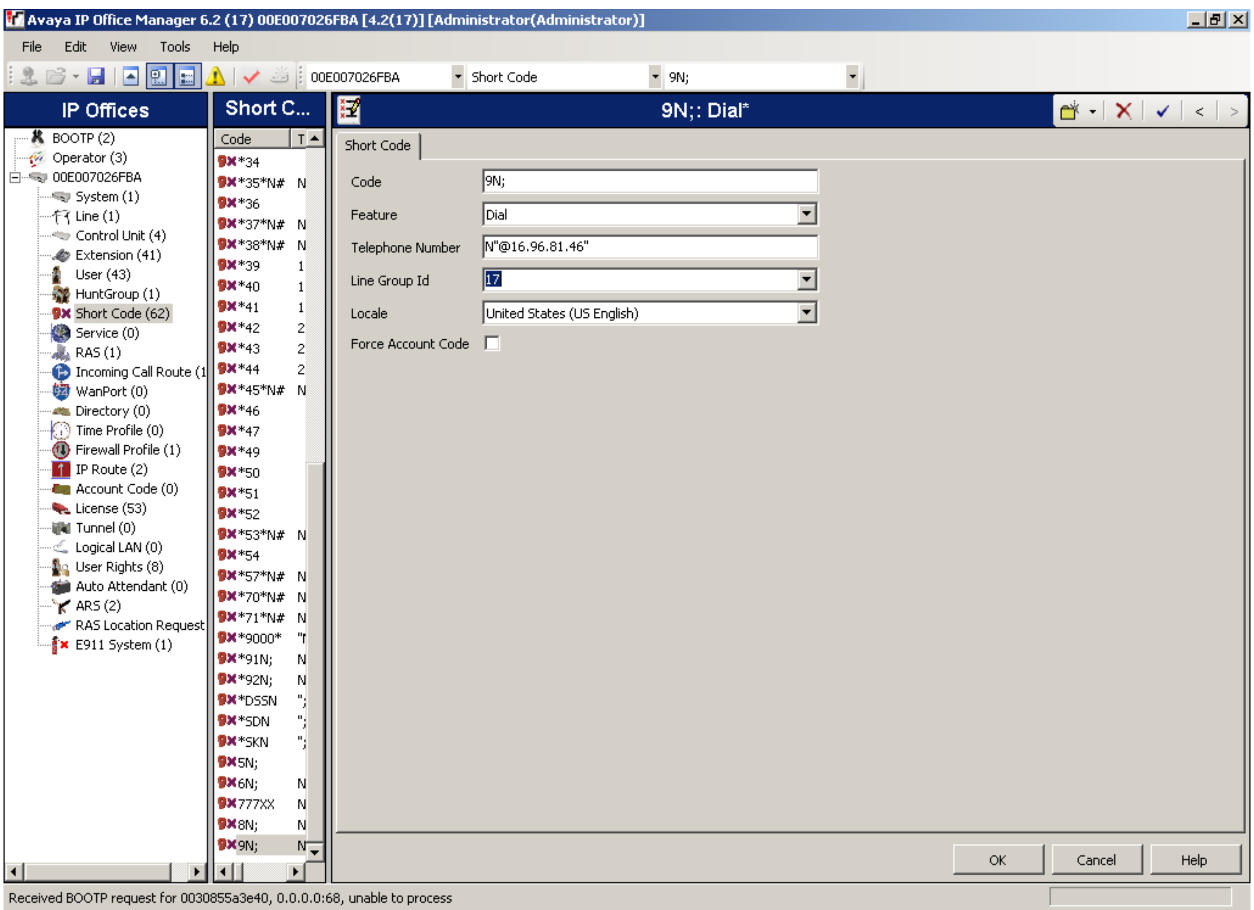


6. *Configure a short code to route calls to CBAD eVantage solution*.
Select **Short Code** in the left panel. Right click and select **Add**.

Configure the following:

- Enter **[x]N;**, where **[x]** is a valid number, in the **Code** text box. The number **9** is used for **[x]** in the example below. 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 (Ex: 411), information service etc.
- Select **Dial** for the **Feature**.
- Enter the dialed number **N** followed by **"@<Domain Name of CBAD eVantage solution >"** 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).
- Enter the **Outgoing Group Id** created in Step 4 for the **Line Group Id** field.
- Use default values for all other fields.

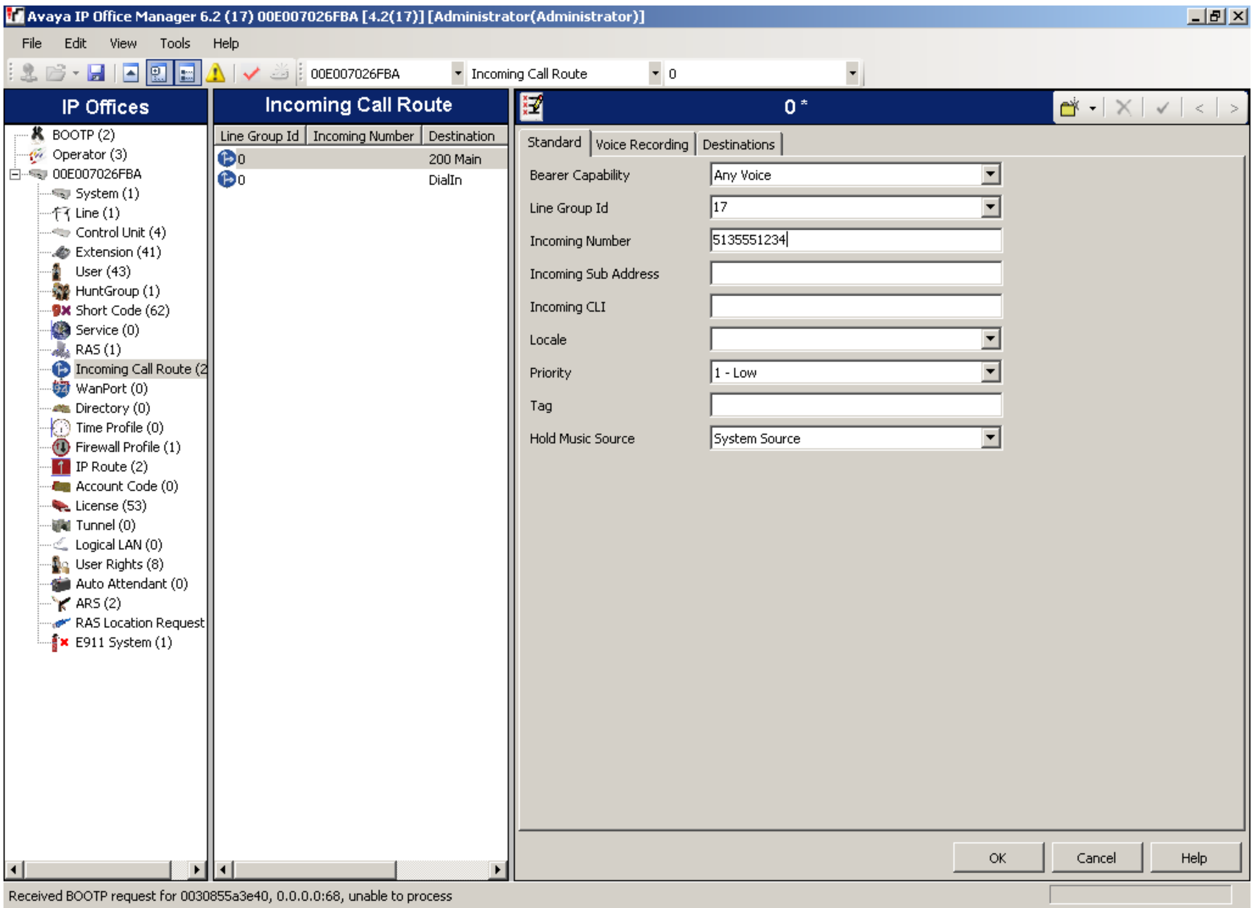
Click the **OK** button.



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

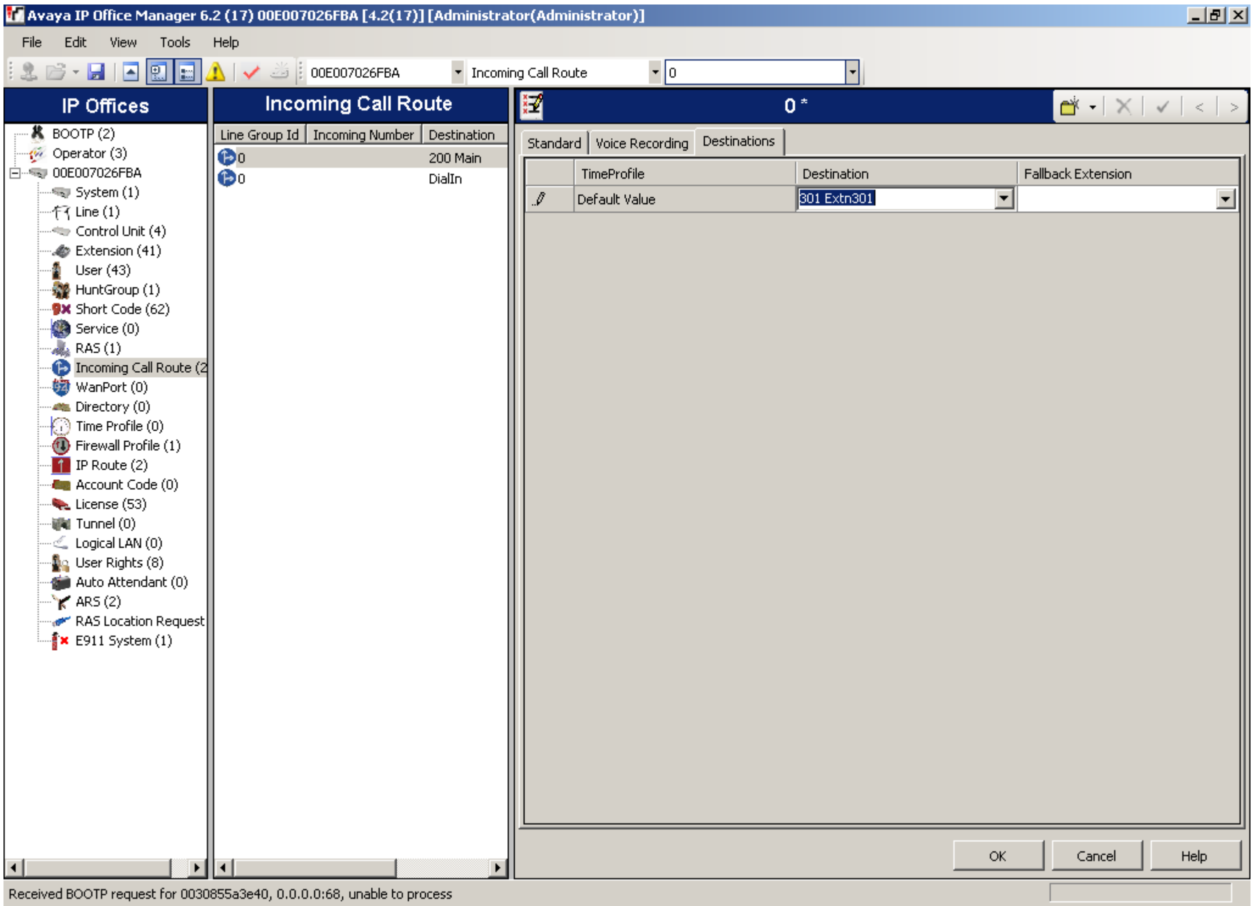
Enter the following:

- **Any Voice** for the **Bearer Capability** field.
- The Incoming Group created for the URI in Step 4 in the **Line Group Id** field.
- The 10 digit DID provided by Cincinnati Bell, 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.



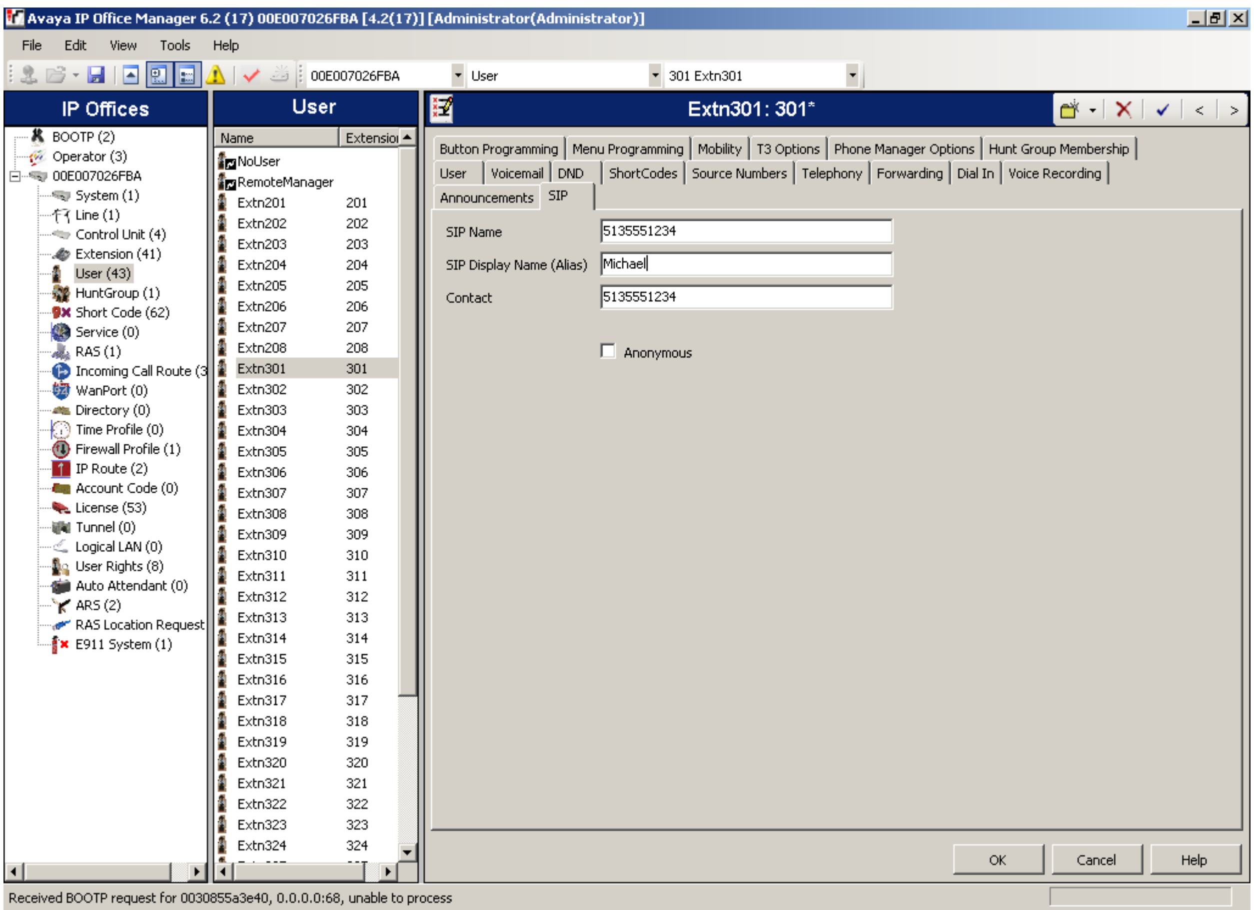
8. *Configure Users' SIP names*.
Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab.

Modify the **SIP Name** and **Contact** fields to the DID number provided by Cincinnati Bell 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

Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info. (See Appendix A for examples of SIP INVITE messages.) The other fields can be left as defaults.

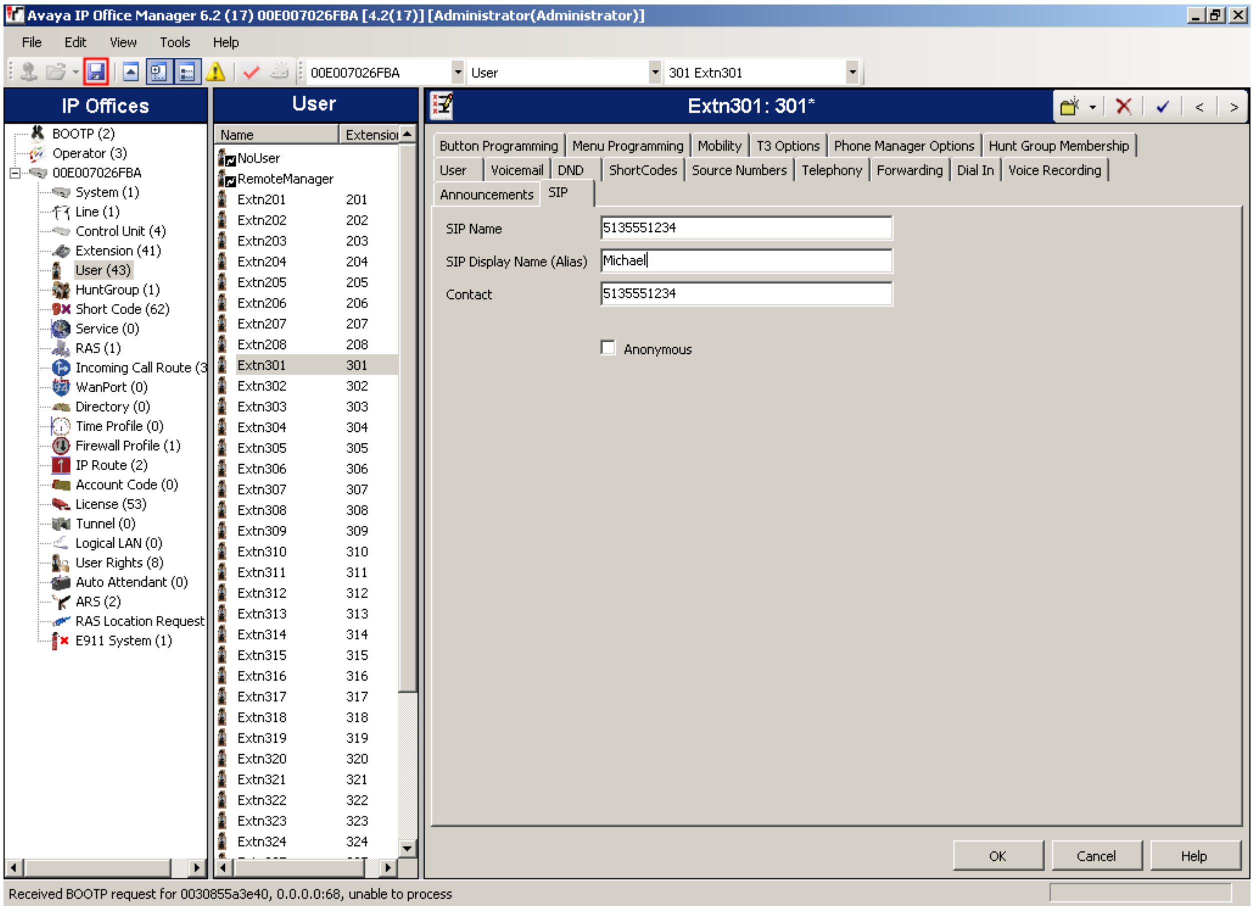
Click the **OK** button.



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

10. *Save configuration.*

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.**



5. Configure the Cincinnati Bell Any Distance eVantage Solution

To use Cincinnati Bell Any Distance (CBAD) eVantage solution, a customer must request service from Cincinnati Bell using their sales processes. Sales information for Cincinnati, Dayton, Ohio and Northern Kentucky can be reached at 1-888-CIN-BELL (246-2355). All other areas should call 1-317-816-5100, Option 1.

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

6. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Cincinnati Bell Any Distance eVantage solution and the Avaya IP Office. This section covers the general test approach and the test results.

Avaya IP Office was connected using SIP trunking (via general purpose Internet services) to the CBAD eVantage solution. The general test approach included the following:

- Inbound Calls – Verify that calls placed from a PSTN telephone to the DID or toll free number assigned are properly routed via the SIP trunk group(s) to the expected extension. Verify the talk-path exists in both directions, that calls remain stable for one minute and disconnect properly.

- Outbound Calls – Verify that calls placed to a PSTN telephone are properly routed via the SIP trunk group(s) defined in the ARS route patterns. Verify that the talk-path exists in both directions and that calls remain stable and disconnect properly.

- Inbound DTMF Digit Navigation – Verify inbound DID calls can properly navigate the Avaya IP Office voice mail menus.

- Outbound DTMF Digit Navigation – Verify outbound calls can properly navigate a voice mail or interactive response system reached via a PSTN number.

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

The following observations were noted:

1. Cincinnati Bell does not support 0+ dialing from their services. Users will hear a recorded message that the number dialed is out of service. Users may dial 0 to reach an automated attendant.

2. eVantage does not support FAX capabilities across the SIP trunk to the PBX, but can provide FAX lines via traditional FXS connections.

3. Cincinnati Bell supports incoming toll free numbers by routing the toll free number to a specified DID number.

4. Outbound calling number restriction is not supported by Cincinnati Bell when checking anonymous for a User on the SIP form. However, a support request to Cincinnati Bell to permit the use of dialing *67 for outbound calling number restriction will block calling party name to be sent to the destination.

5. 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.

6. If the G.723.1 6K3 MP-MLQ codec from the pull down menu for the Compression Mode of the SIP Line is selected as the preferred codec, it will not be offered in the SIP INVITE. This will be fixed in a future release of Avaya IP Office.

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 Cincinnati Bell Any Distance eVantage solution.

1. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 1 minute. 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 1 minute.

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. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya IP Office to connect to the PSTN via the Cincinnati Bell Any Distance eVantage solution. The CBAD eVantage solution is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. The CBAD eVantage solution provides businesses a flexible, cost-saving alternative to traditional hardwired telephony trunk lines.

9. Additional 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 typical SIP INVITE messages sent between Cincinnati Bell eVantage solution and Avaya IP Office. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE from Avaya IP Office to Cincinnati Bell:

```
No.     Time        Source                Destination           Protocol Info
      1 0.000000    65.211.92.4          16.96.81.46       SIP/SDP  Request: 
INVITE sip:7325554321@16.96.81.46, with session description

Frame 1 (826 bytes on wire, 826 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: 65.211.92.4 (65.211.92.4), Dst: 16.96.81.46 
(16.96.81.46)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:7325554321@16.96.81.46 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 
65.211.92.4:5060;rport;branch=z9hG4bKf4d236647b7645de9aa39eec77521ee8
            Transport: UDP
            Sent-by Address: 65.211.92.4
            Sent-by port: 5060
            RPort: rport
            Branch: z9hG4bKf4d236647b7645de9aa39eec77521ee8
        From: "Michael" <sip:5135551234@65.211.92.4>;tag=059218cd79b4d500
            SIP Display info: "Michael"
            SIP from address: sip:5135551234@65.211.92.4
            SIP tag: 059218cd79b4d500
        To: <sip:7325554321@16.96.81.46>
            SIP to address: sip:7325554321@16.96.81.46
        Call-ID: 7da245231bc397f8463494c23bd9ea21@65.211.92.4
        CSeq: 1642259807 INVITE
            Sequence Number: 1642259807
            Method: INVITE
        Contact: "Michael" <sip:5135551234@65.211.92.4:5060;transport=udp>
            Contact Binding: "Michael" 
<sip:5135551234@65.211.92.4:5060;transport=udp>
                URI: "Michael" 
<sip:5135551234@65.211.92.4:5060;transport=udp>
                    SIP Display info: "Michael"
                    SIP contact address: sip:5135551234@65.211.92.4:5060
        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 3391009796 1804490086 IN IP4
65.211.92.4
                Owner Username: UserA
                Session ID: 3391009796
                Session Version: 1804490086
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 65.211.92.4
            Session Name (s): Session SDP
            Connection Information (c): IN IP4 65.211.92.4
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 65.211.92.4
            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 18 8
0 101
                Media Type: audio
                Media Port: 49152
                Media Proto: RTP/AVP
                Media Format: ITU-T G.729
                Media Format: ITU-T G.711 PCMA
                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): rtpmap:8 PCMA/8000
                Media Attribute Fieldname: rtpmap
                Media Format: 8
                MIME Type: PCMA
            Media Attribute (a): rtpmap:0 PCMU/8000
                Media Attribute Fieldname: rtpmap
                Media Format: 0
                MIME Type: PCMU
            Media Attribute (a): fmtp:18 annexb = no
                Media Attribute Fieldname: fmtp
                Media Format: 18 [PCMU]
                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
```

Sample SIP INVITE Message from Cincinnati Bell to Avaya IP Office:

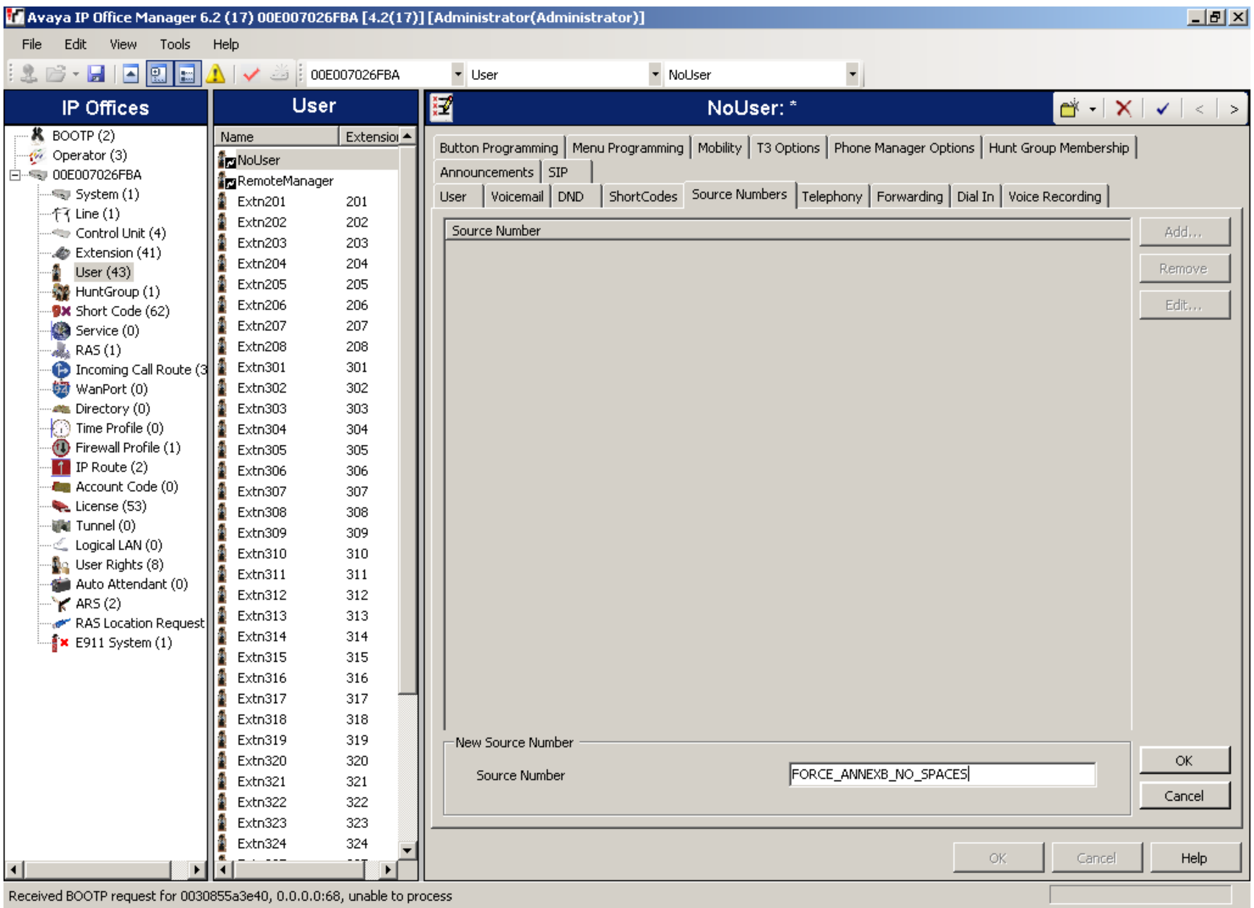
```
No.     Time        Source                Destination           Protocol Info
      1 0.000000    16.96.81.46       65.211.92.4           SIP/SDP  Request: 
INVITE sip:5135551234@65.211.92.4:5060, with session description

Frame 1 (1228 bytes on wire, 1228 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: 16.96.81.46 (16.96.81.46), Dst: 65.211.92.4 
(65.211.92.4)
User Datagram Protocol, Src Port: 58587 (58587), Dst Port: sip (5060)
Session Initiation Protocol
    Request-Line: INVITE sip:5135551234@65.211.92.4:5060 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 16.96.81.46:5060;branch=z9hG4bK2B515AF
            Transport: UDP
            Sent-by Address: 16.96.81.46
            Sent-by port: 5060
            Branch: z9hG4bK2B515AF
        Remote-Party-ID: "REDBANK,NJ" 
<sip:7325554321@as.voip.fuse.net>;party=calling;screen=no;privacy=off
        From: "REDBANK,NJ" <sip:7325554321@as.voip.fuse.net>;tag=228E19D8-2A6
            SIP Display info: "REDBANK,NJ"
            SIP from address: sip:7325554321@as.voip.fuse.net
            SIP tag: 228E19D8-2A6
        To: <sip:5135551234@65.211.92.4>
            SIP to address: sip:5135551234@65.211.92.4
        Date: Thu, 21 May 2009 12:38:59 GMT
        Call-ID: 30B13549-453B11DE-818DF1CE-206E6DAB@as.voip.fuse.net
        Supported: 100rel,timer,resource-priority,replaces
        Min-SE:  1800
        Cisco-Guid: 816799284-1161499102-2173235662-544107947
        User-Agent: Cisco-SIPGateway/IOS-12.x
        Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, 
SUBSCRIBE, NOTIFY, INFO, REGISTER
        CSeq: 101 INVITE
            Sequence Number: 101
            Method: INVITE
        Timestamp: 1242909539
        Contact: <sip:7325554321@16.96.81.46:5060>
            Contact Binding: <sip:7325554321@16.96.81.46:5060>
                URI: <sip:7325554321@16.96.81.46:5060>
                    SIP contact address: sip:7325554321@16.96.81.46:5060
        Expires: 180
        Allow-Events: telephone-event
        Max-Forwards: 8
        Content-Type: application/sdp
        Content-Disposition: session;handling=required
        Content-Length: 256
    Message Body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): CiscoSystemsSIP-GW-UserAgent 4798 
8998 IN IP4 16.96.81.46
```

```
            Owner Username: CiscoSystemsSIP-GW-UserAgent
            Session ID: 4798
            Session Version: 8998
            Owner Network Type: IN
            Owner Address Type: IP4
            Owner Address: 16.96.81.46
        Session Name (s): SIP Call
        Connection Information (c): IN IP4 16.96.81.46
            Connection Network Type: IN
            Connection Address Type: IP4
            Connection Address: 16.96.81.46
        Time Description, active time (t): 0 0
            Session Start Time: 0
            Session Stop Time: 0
        Media Description, name and address (m): audio 19100 RTP/AVP 0
101
            Media Type: audio
            Media Port: 19100
            Media Proto: RTP/AVP
            Media Format: ITU-T G.711 PCMU
            Media Format: 101
        Connection Information (c): IN IP4 16.96.81.46
            Connection Network Type: IN
            Connection Address Type: IP4
            Connection Address: 16.96.81.46
        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
        Media Attribute (a): fmtp:101 0-16
            Media Attribute Fieldname: fmtp
            Media Format: 101 [telephone-event]
            Media format specific parameters: 0-16
        Media Attribute (a): ptime:20
            Media Attribute Fieldname: ptime
            Media Attribute Value: 20
```

APPENDIX B: G.729A Codec Support

To utilize the G.729A codec with the CBAD eVantage solution with the Cisco CUBE router, select **User** in the left panel and then select **NoUser** in the **User** form. Select the **Source Numbers** tab and then click on the **Add..** button. Enter **FORCE_ANNEXB_NO_SPACES** in the **Source Number** field.

Click the **OK** button.



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.