



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Broadvox GO! SIP Trunking and Avaya IP Office – Issue 1.2

Abstract

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

Broadvox 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 Broadvox GO! SIP Trunking services 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 Broadvox.

Broadvox GO! SIP Trunking delivers inexpensive local, toll-free, domestic and international long distance service through the Broadvox national footprint covering North America with per trunk and per minute rates much lower than that of traditional service. Also, with phone numbers from over 7,000 rate centers, Broadvox can provide new or port existing numbers nationwide.

Customers using this Avaya IP telephony solution with Broadvox GO! SIP Trunking 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.

Broadvox GO! SIP Trunking services include the following capabilities:

- Outbound PSTN calling to local, long distance and international services
- Incoming Direct Inward Dial (DID) service
- Incoming Toll-Free service

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 Broadvox GO! SIP Trunking 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:

- Trunk Registration/Authentication with Broadvox GO! SIP Trunking services.
- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Broadvox GO! SIP Trunking services.
- Outgoing calls from the enterprise site were completed via Broadvox GO! SIP Trunking service 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 μ LAW codecs.
- Fax routing to ensure G.711 μ LAW use for fax calls.
- 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 for Broadvox GO! SIP Trunking services, contact Broadvox Customer Support by telephone at 800-273-4320 or by sending email to info@broadvox.com.

2. Reference Configuration

Figure 1 illustrates an example Avaya IP telephony solution connected to Broadvox GO! SIP Trunking 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 1603SW IP Telephone (H.323 protocol)
- Avaya 4610SW IP Telephone (H.323 protocol)
- Avaya 2420 Digital Telephone
- Avaya 5420 Digital Telephone
- Avaya 6210 Analog Telephone

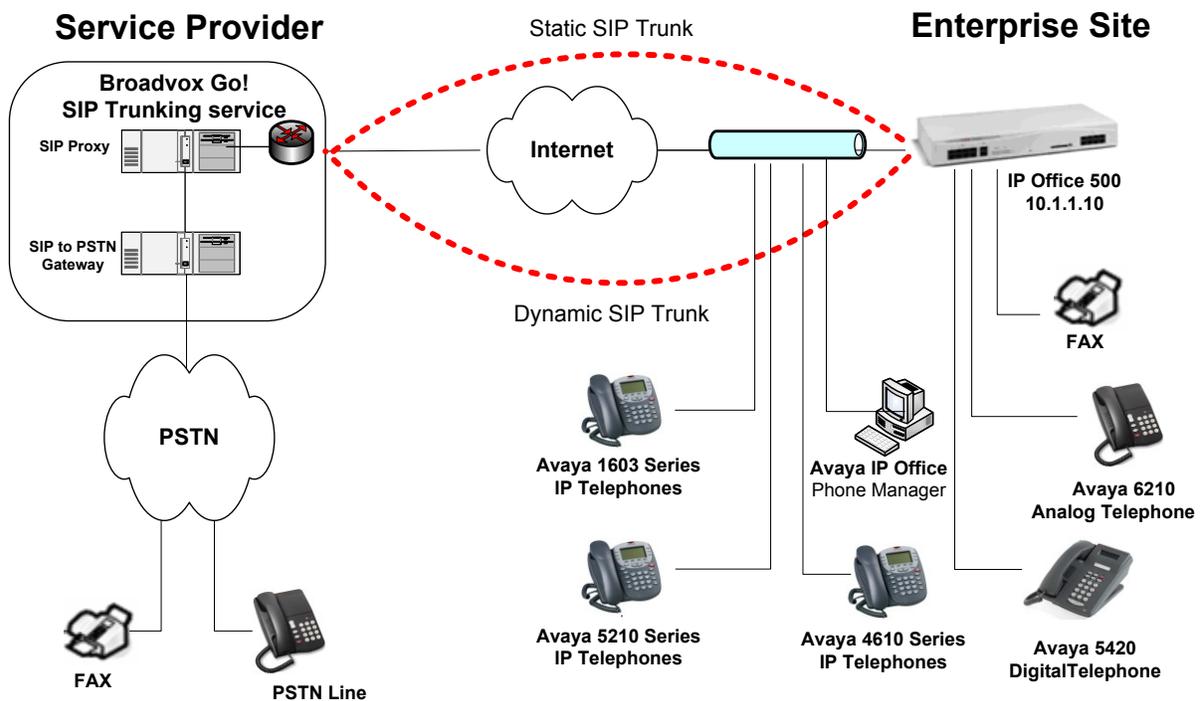


Figure 1: Avaya IP Telephony Network connected to Broadvox GO! SIP Trunking service

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 (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 1603 IP Telephone	R10492
Avaya 4610SW IP Telephone	R8.016 – H.323
Avaya 5620 IP Telephone	R2.2.3
Avaya 6210 Analog Telephone	n/a
Avaya 5420 Digital Phone	R5 Firmware
Avaya IP Office Phone Manager	R4.2.19
Broadvox GO! SIP Trunking service Solution Components	
NextPoint MSX Static	N/A
NextPoint MSX Dynamic	N/A

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

4. Configure the Avaya IP Office

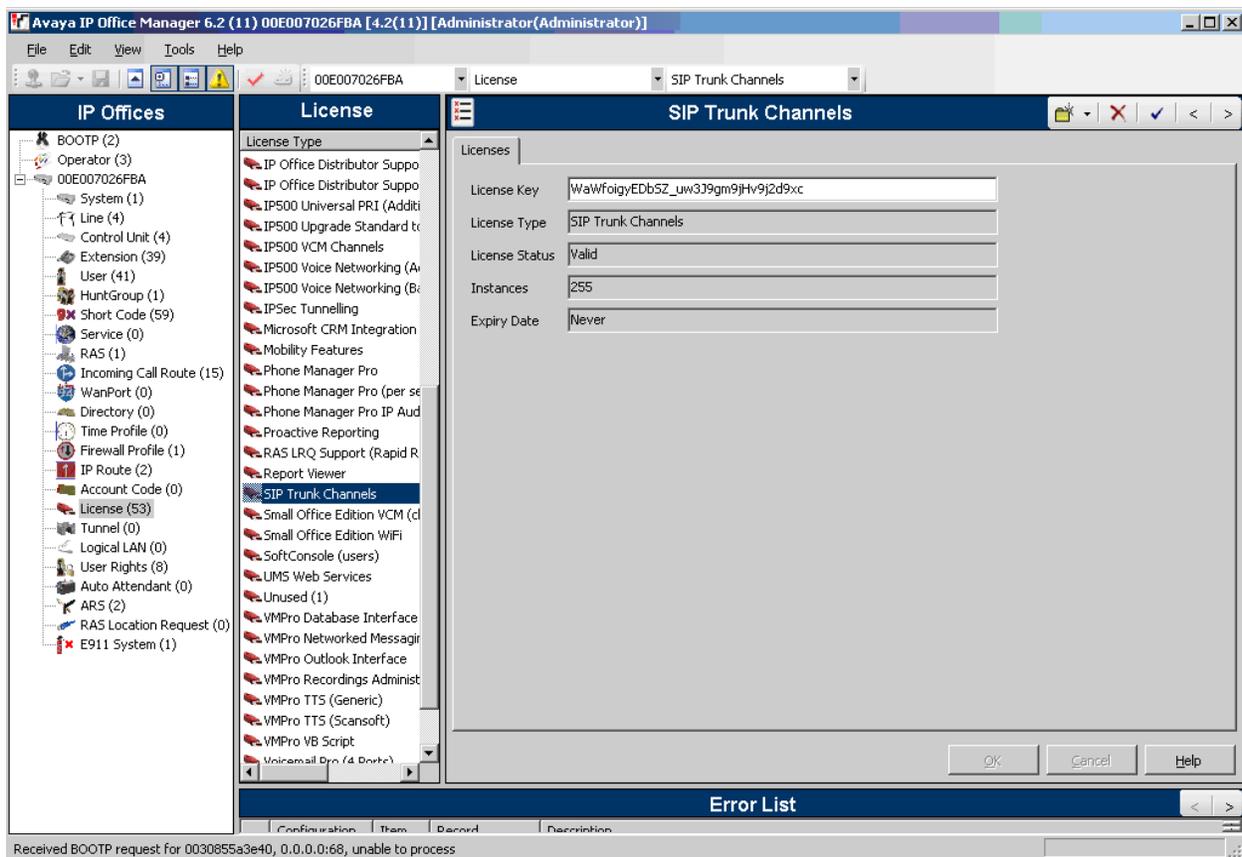
This section describes the steps for configuring a static and dynamic SIP trunk on IP Office.

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.

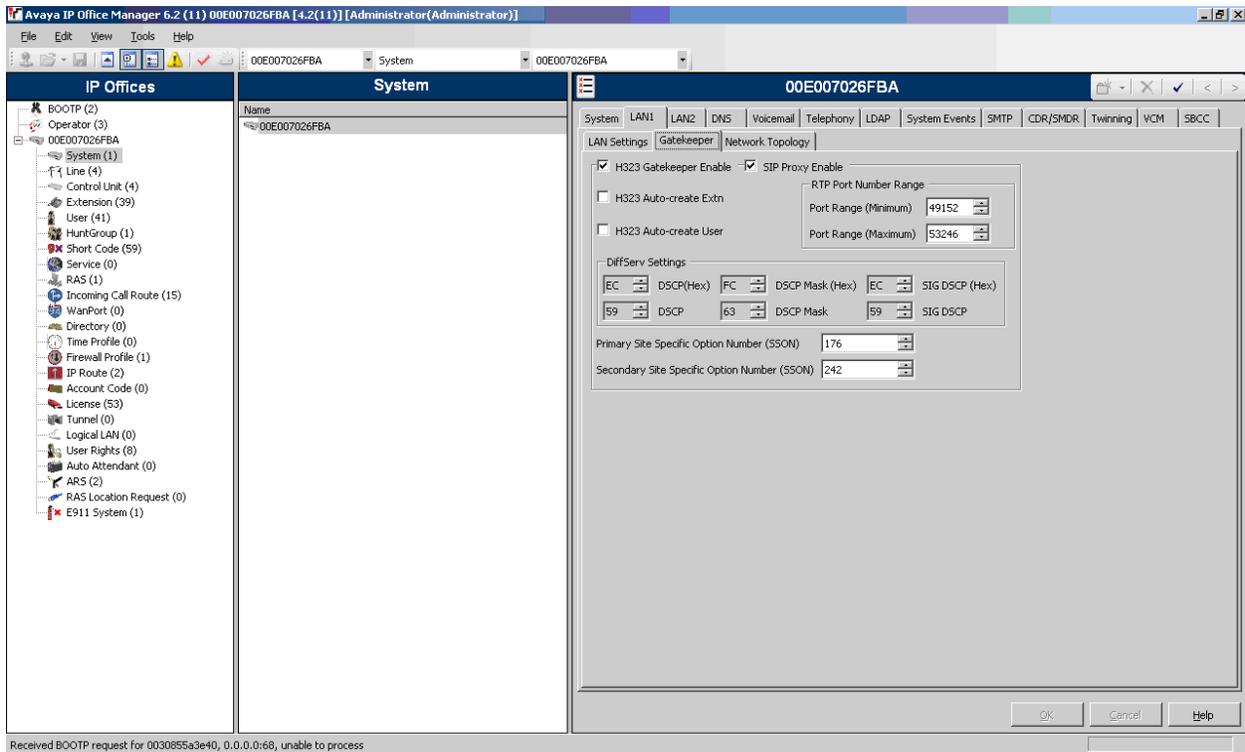
Click on **License** in the left panel. Confirm that there is a **SIP Trunk Channels** entry.

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

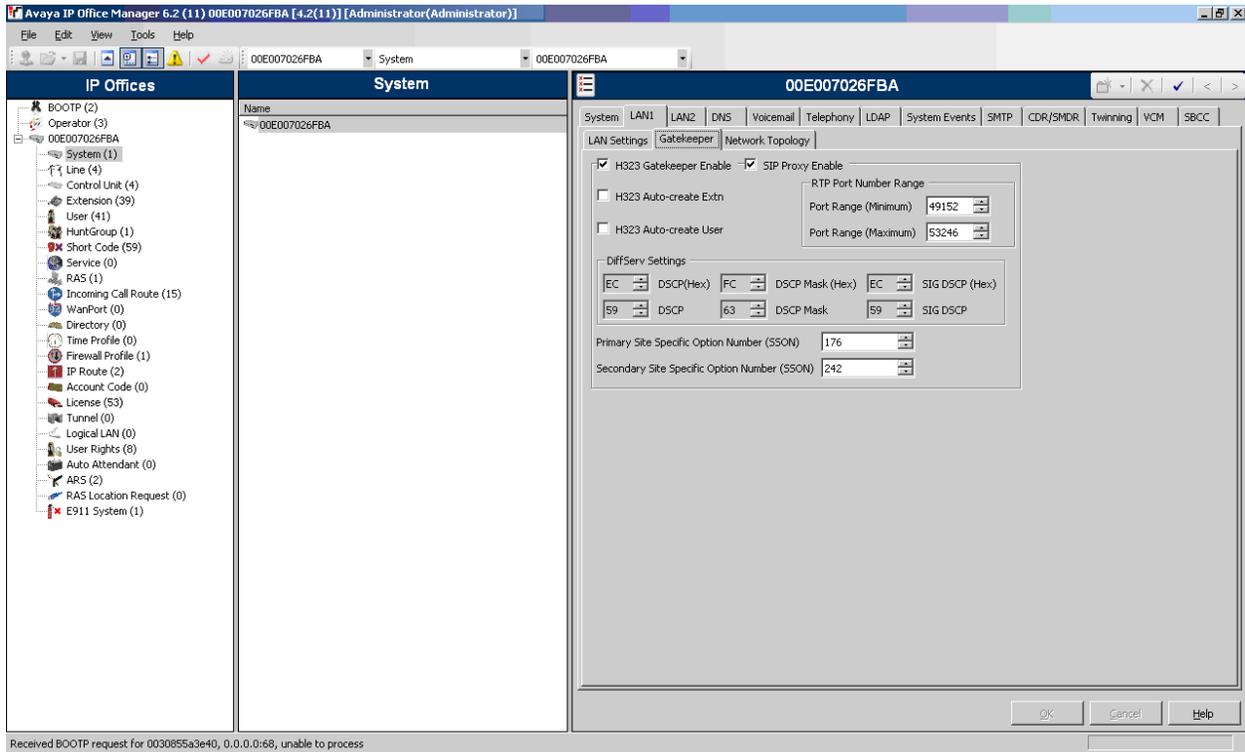


2. *Enable SIP Proxy Functionality.*

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.



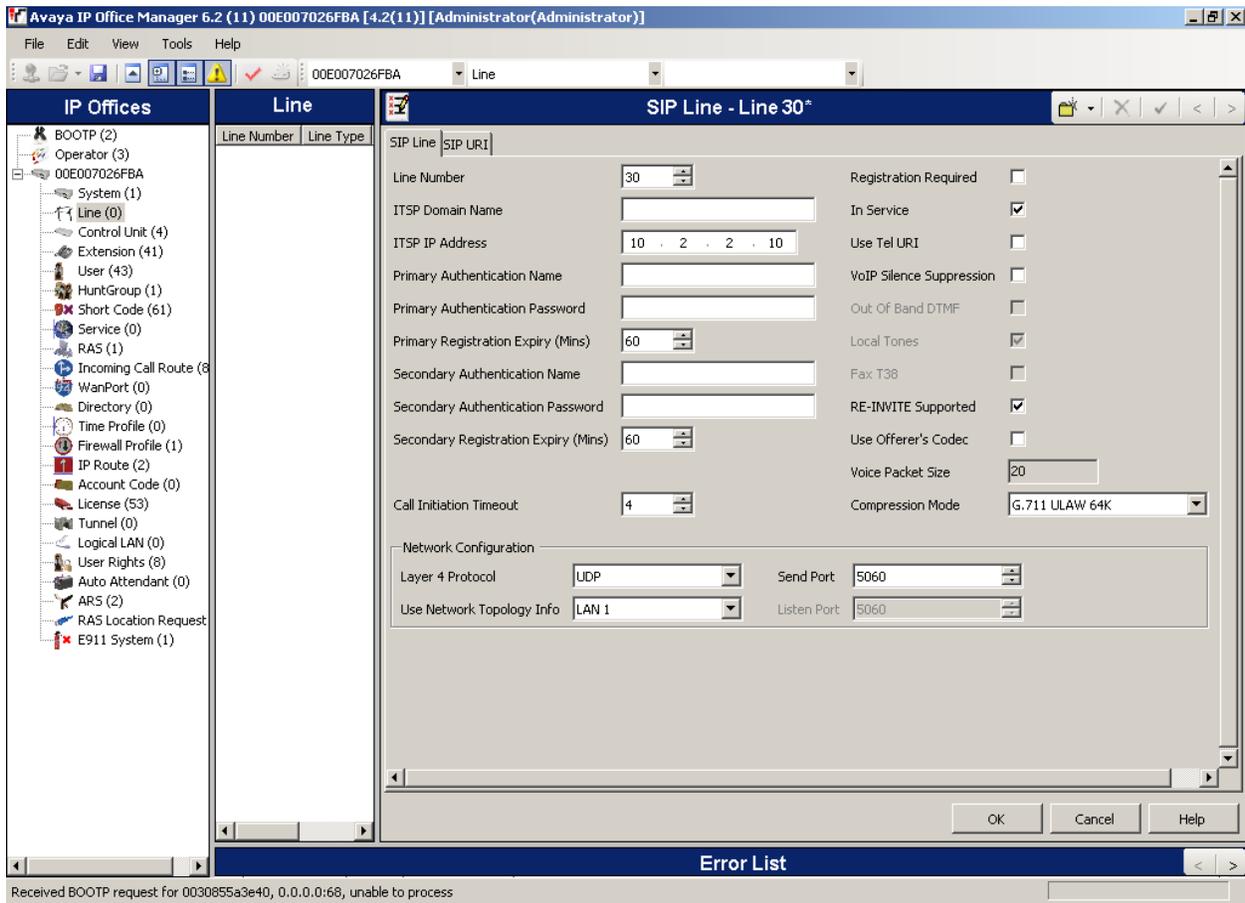
3. *Configure DiffServ Settings according to Broadvox GO! SIP Trunking services requirements.* Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper** tab (same as mentioned in step 2). Under **DiffServ Settings**, enter **EC** into the **DSCP(Hex)** and **SIG DSCP(Hex)**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



4. Create the static SIP line for the Broadvox GO! SIP Trunking service SIP proxy. Select **Line** in the left panel. Right-click and select **New** → **SIP Line**.

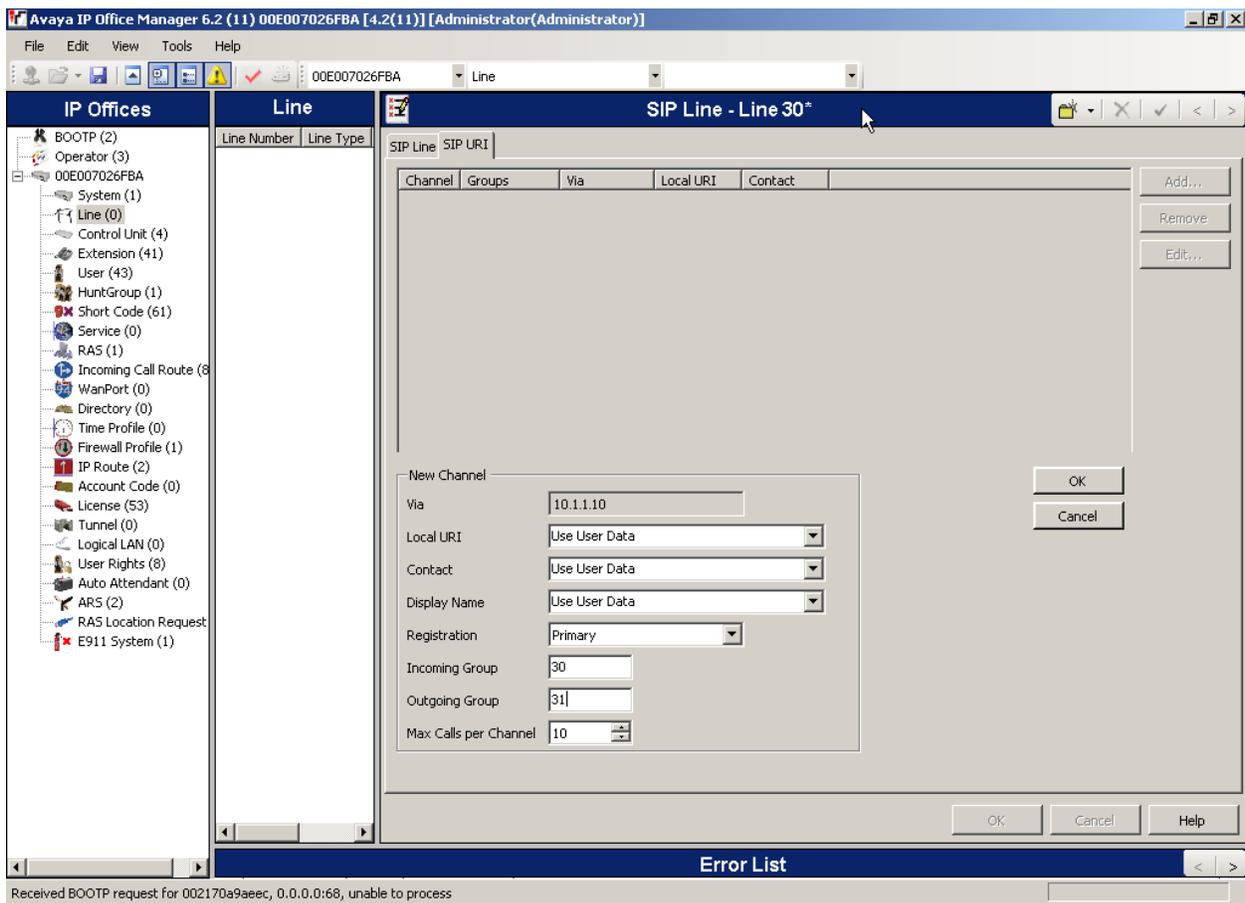
Configure the following:

- For the **ITSP IP Address** field, enter the IP address of the Broadvox GO! SIP Trunking 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).
- Click on **RE-INVITE Supported**.
- Use default values for other fields.



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

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). Enter a unique number for the **Incoming Group** and **Outgoing Group** fields. The **Incoming Group** field will be used for mapping inbound calls from the SIP trunk to local stations configured in step 8. The **Outgoing Group** will be used for routing calls externally via the Short Code configured in step 7. Use default values for all other fields. Click the **OK** button.

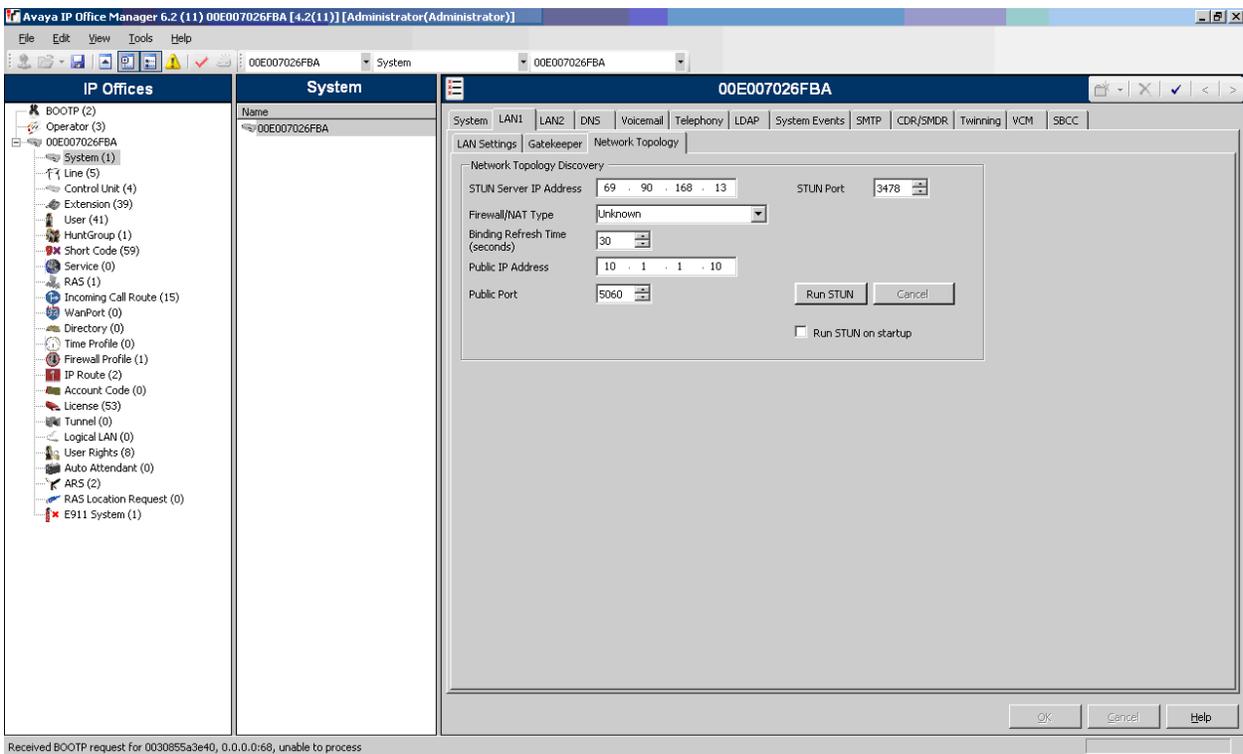


6. *Configure SIP OPTIONS timer on Network Topology Tab for “keep alive” function with the Broadvox GO! SIP Trunking service.*

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 Broadvox GO! SIP Trunking service. For **Public IP Address**, enter the Avaya IP Office system IP address. Confirm that **Public Port** is set to 5060 and accept the 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].

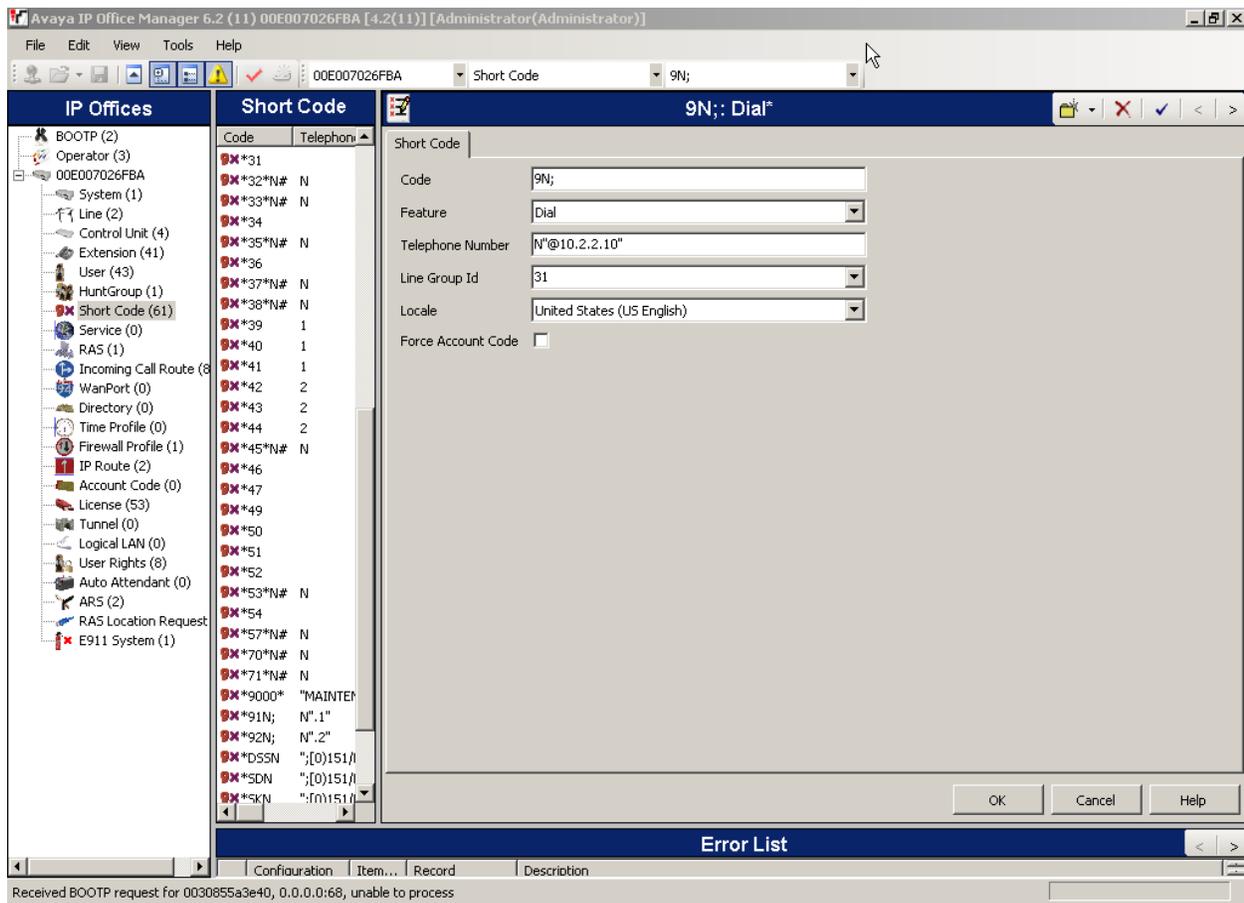


7. Configure a short code to route calls to Broadvox SIP Proxy Server.

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 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 (e.g., 411), information service, etc.

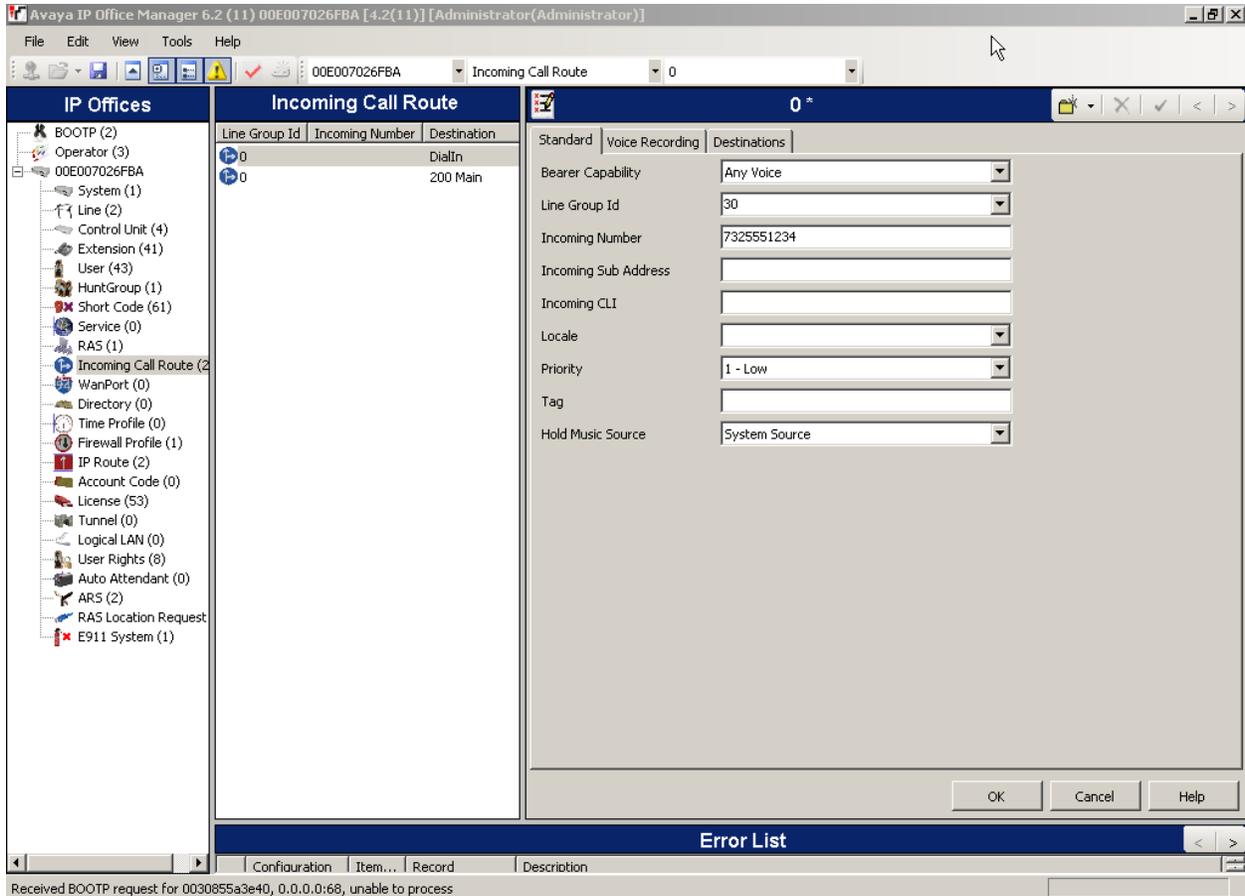
Select **Dial** for the **Feature**. Enter the **Outgoing Group Id** created in step 5 for the **Line Group Id** field. Enter the dialed number **N** followed by "**@<Domain Name of Broadvox GO! SIP Trunking service>**" for the **Telephone Number** field. The **Telephone Number** field is used to construct the SIP URI in the **To** field of 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.



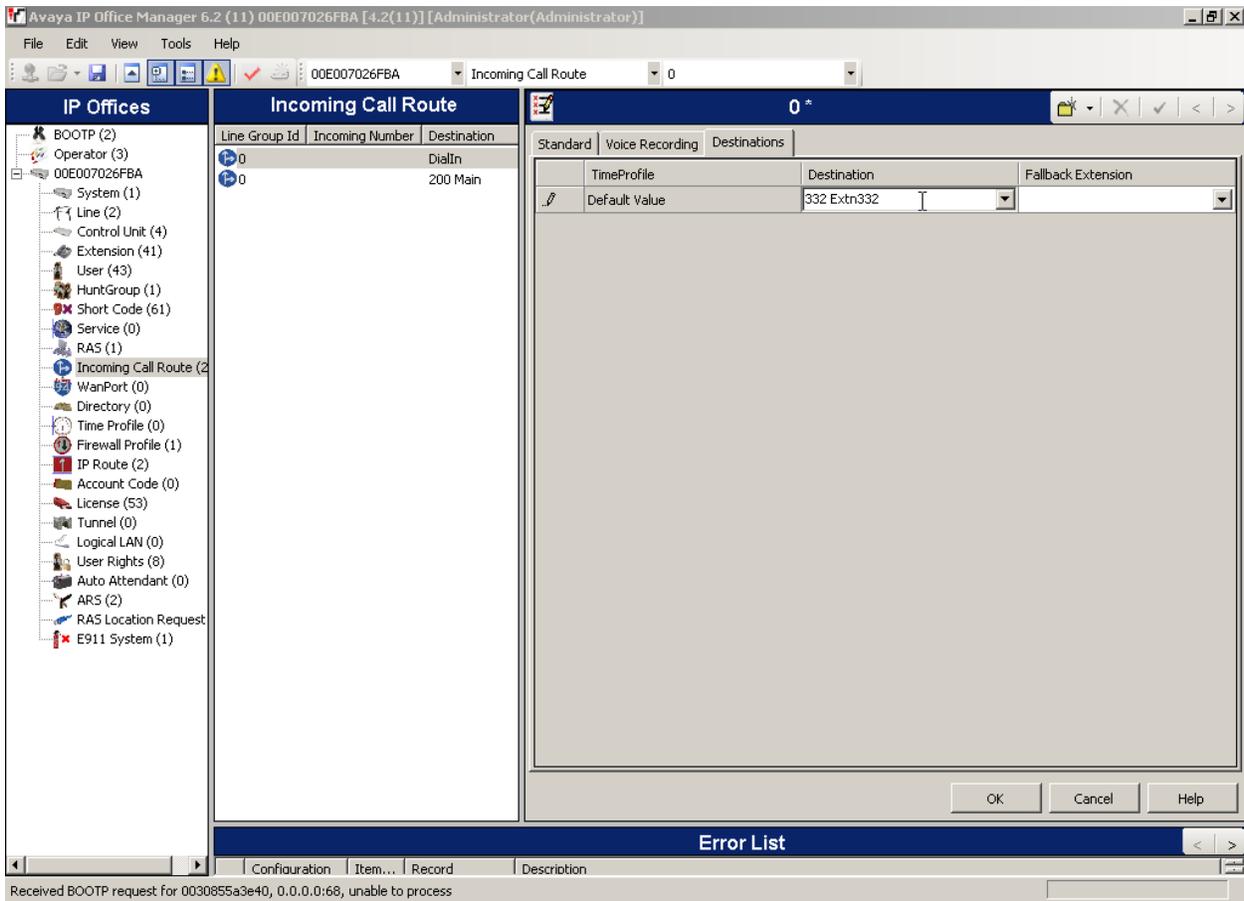
8. *Create an Incoming Call Route for the Inbound SIP calls.*
Select **Incoming Call Route** in the left panel. Right-click and select **New**.

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 number provided by Broadvox 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.

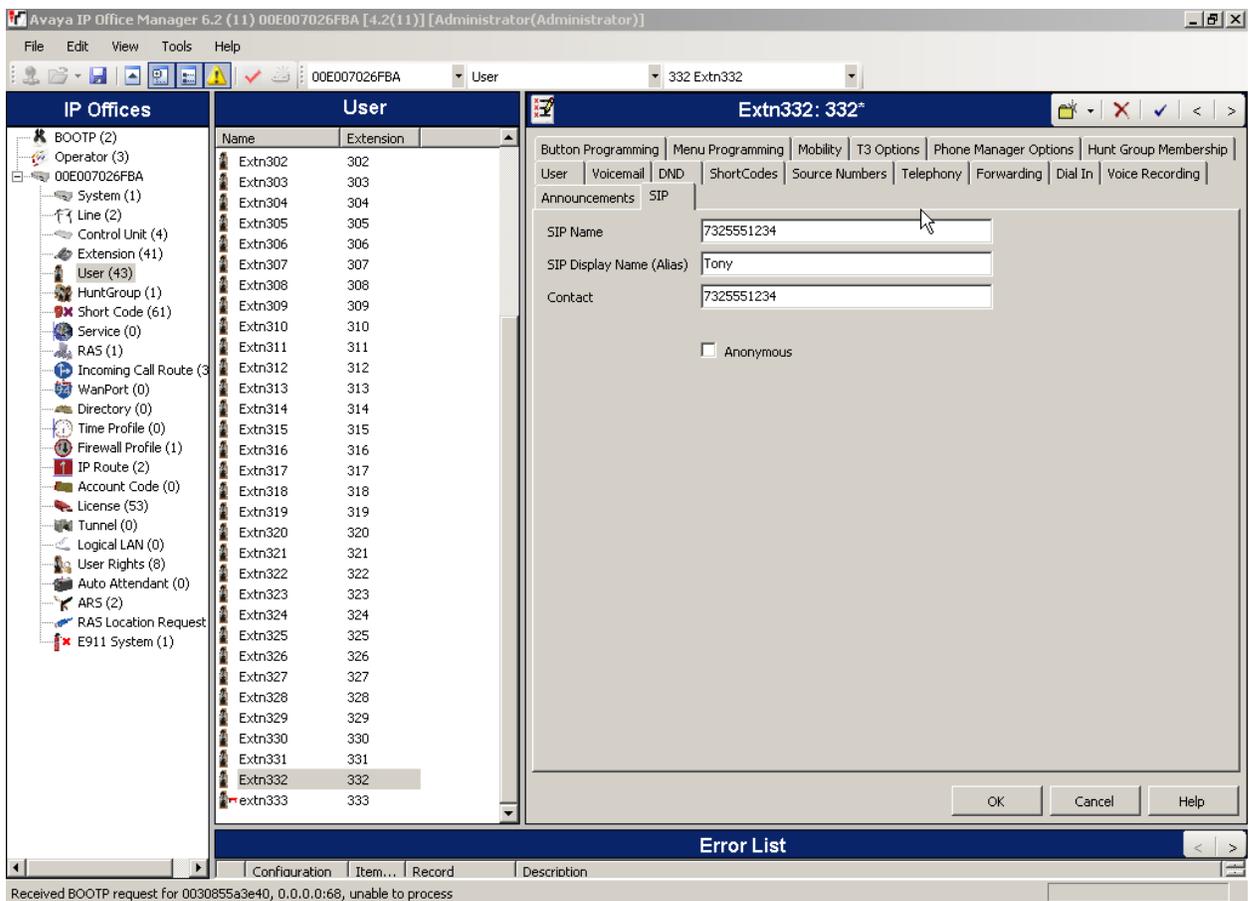
Modify the **SIP Name** and **Contact** fields to the DID number provided by Broadvox that is used for this particular extension. These settings instruct the system to use this DID number to construct the:

- 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

Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info.

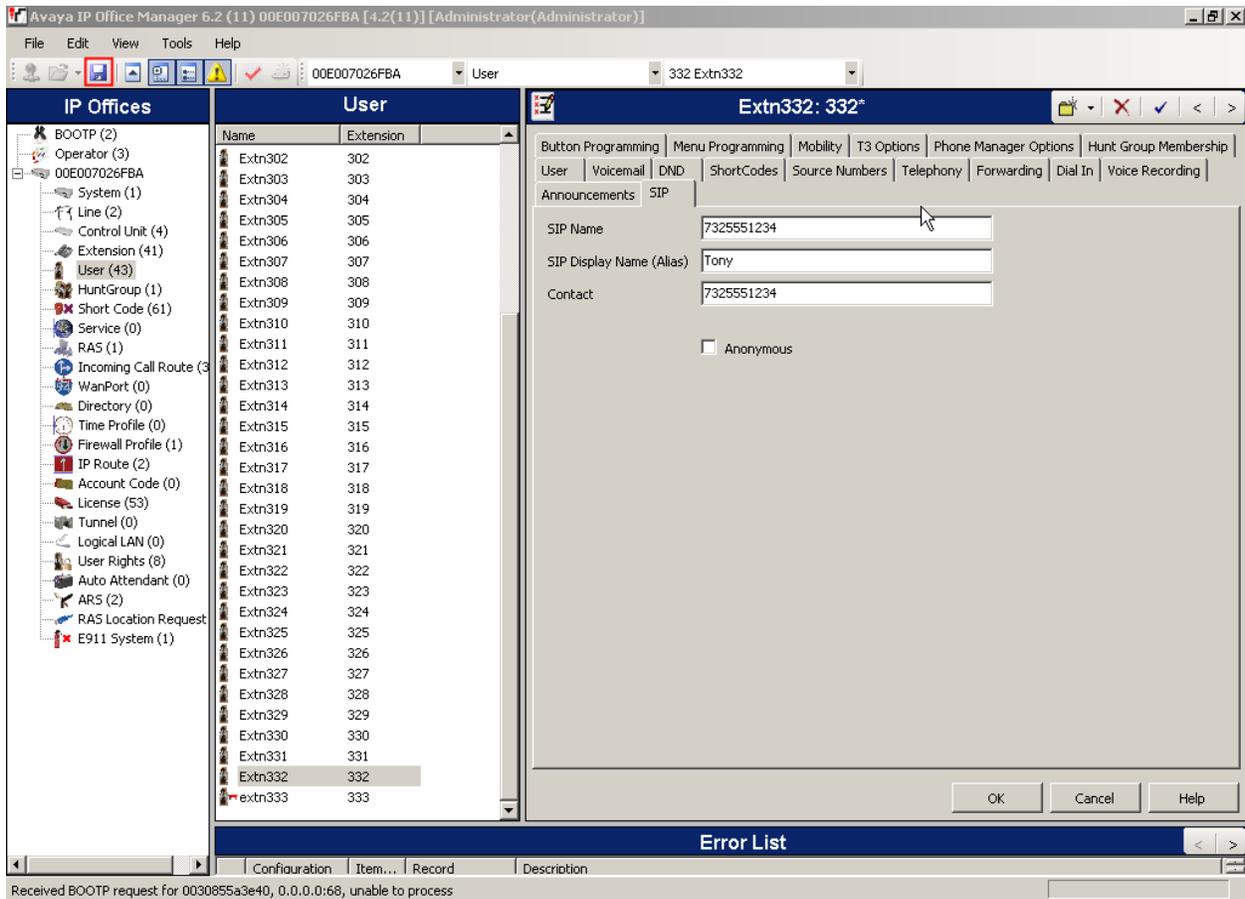
(See Appendix A for examples of SIP INVITE messages.)

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



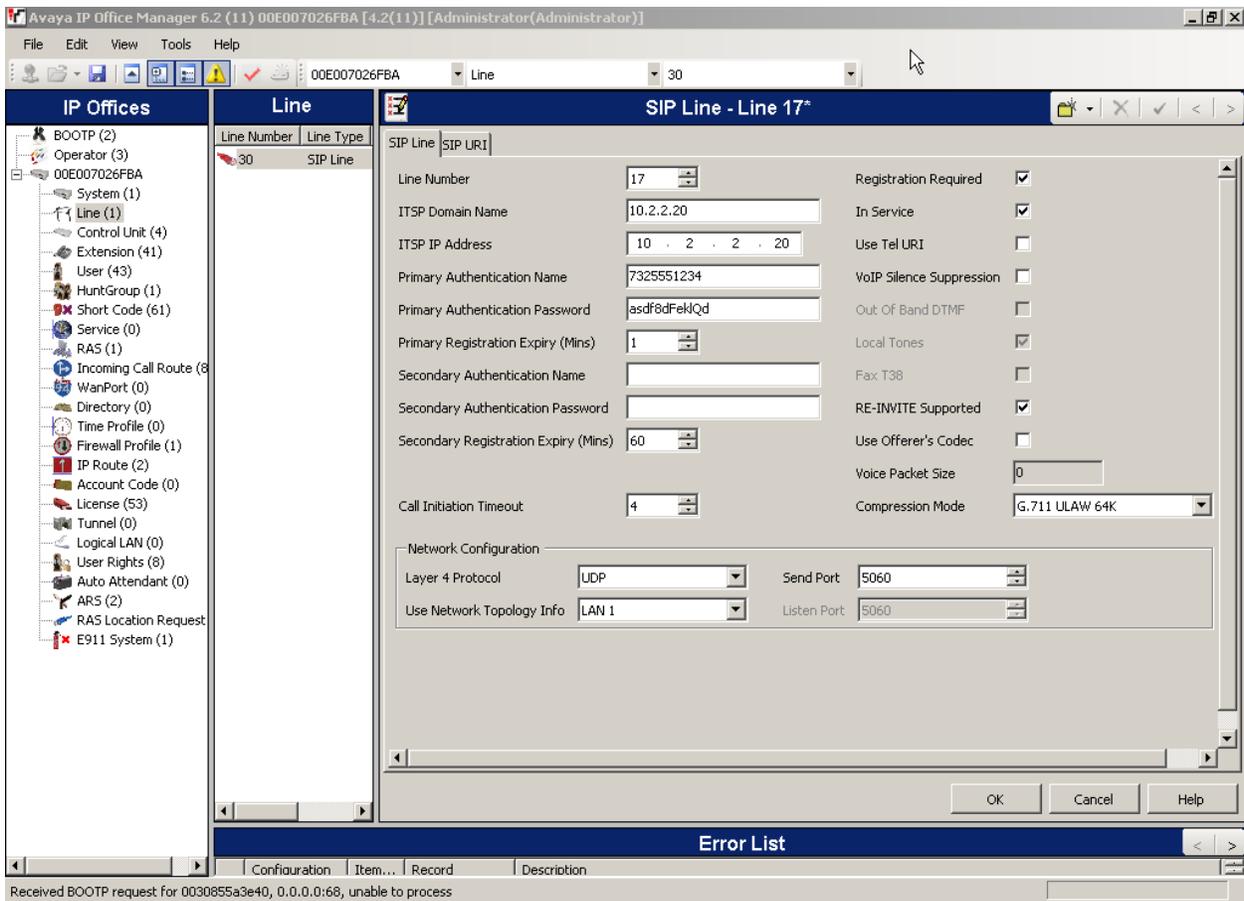
10. Repeat steps 8 and 9 for all users that will be sending/receiving SIP calls on the system.

- 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.**



12. Create the dynamic SIP line for the Broadvox GO! SIP Trunking service SIP proxy. To configure a trunk for Trunk Registration/Authentication, Select **Line** in the left panel. Right-click and select **New** → **SIP Line**. Configure the following:

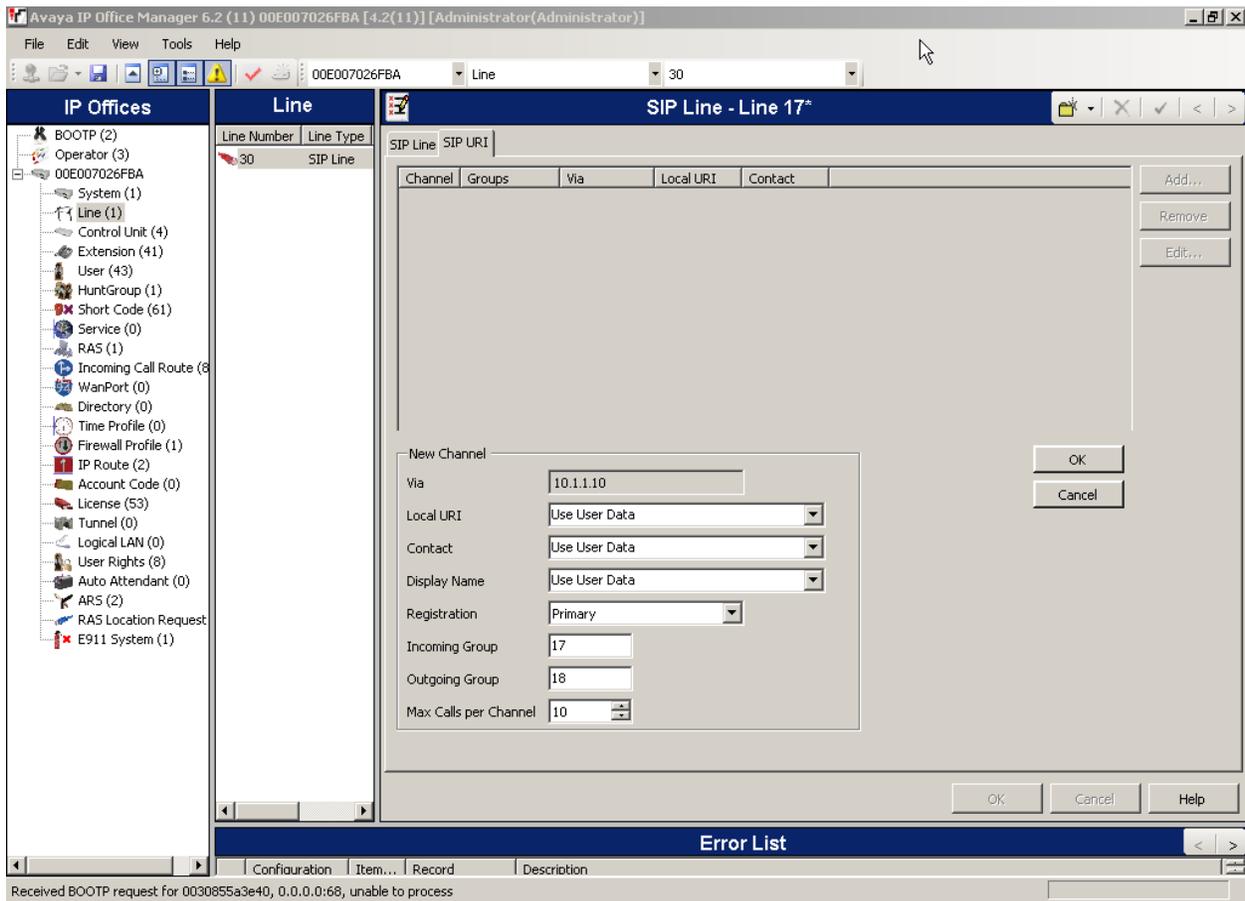
- For the **ITSP Domain Name** and the **ITSP IP Address** fields, enter the IP address of the Broadvox GO! SIP Trunking service SIP Proxy.
- Enter the **Primary Authentication Name** as provided by Broadvox. In this case, Broadvox used the Billing Telephone Number (BTN) as the user ID. Separate DID numbers provided by Broadvox are routed through this SIP trunk for use by IP Office.
- Enter the **Primary Authentication Password** as provided by Broadvox.
- Enter **1** for the **Primary Registration Expiry (Mins)** so that registration must occur every 60 seconds.
- 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).
- Click on **Registration Required**.
- Click on **RE-INVITE Supported**.
- Use default values for other fields.



13. Configure SIP URI parameters for the dynamic SIP trunk.

Select the **SIP URI** Tab. Click the **Add** button.

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). 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. The **Outgoing Group** will be used for routing calls externally via the Short Code. Use default values for all other fields. Click the **OK** button.



Repeat steps 7, 8, and 9 to create new short codes for routing external calls and to route internal calls to the appropriate phones using this dynamic SIP trunk.

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.**

5. Configure Broadvox GO! SIP Trunking services

To use Broadvox GO! SIP Trunking service, a customer must request service from Broadvox using their sales processes. The process can be started by contacting Broadvox via the corporate web site at <http://www.broadvox.com> and requesting information via the online sales links or telephone numbers.

During the signup process, Broadvox will require that the customer provide the public IP address used to reach the Avaya IP Office server. Broadvox provided the following information for the compliance testing: IP addresses of the Broadvox Communications SIP proxy/SBC, Direct Inward Dialed (DID) numbers, toll free numbers, and login/password credentials for the dynamic trunk. 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 the Broadvox GO! SIP Trunking service 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 Broadvox GO! SIP Trunking service. The general test approach included the following:

- Inbound Calls – Verify that calls placed from a PSTN telephone to a DID or toll free number are properly routed via the SIP trunk group(s) to the expected extension on Avaya IP Office. Verify the talk-path exists in both directions, that calls remain stable for several minutes 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 configured short code. 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. Broadvox does not support 0 or 0+ dialing from their services.
2. For **Compression Mode** in the SIP line, 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 IP Office.
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 number associated with the enterprise Avaya IP Office phone.
4. 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.
5. Incoming toll free numbers will display the dial string containing ANI II digits as required by law. Broadvox does have the capacity to strip them out on a per-customer basis if the customer requests it.

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 Broadvox GO! SIP Trunking 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.
5. Verify dynamic trunks successfully registered with Broadvox and an outgoing call to the PSTN is successfully completed.
6. Verify incoming DID call routing using dynamic trunks occurred on the number in the SIP INVITE To field, whereas the SIP URI contains the Billing Telephone Number.

8. Conclusion

These Application Notes describe the configuration steps enabling customers using Avaya IP Office to connect to the PSTN via Broadvox GO! SIP Trunking service. Broadvox GO! SIP Trunking service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. Broadvox GO! SIP Trunking service 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/>

10. Change History

Issue	Date	Reason
1.0	6/23/2009	Initial issue
1.1	7/17/2009	Broadvox service naming revisions

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of typical SIP INVITE messages sent between Broadvox GO! SIP Trunking service 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 Broadvox:

```
No.      Time      Source      Destination      Protocol Info
   1 0.000000 10.1.1.10   10.2.2.10        SIP/SDP Request:
INVITE sip:7325550819@10.2.2.10, with session description

Frame 1 (824 bytes on wire, 824 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:7325550819@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=z9hG4bKde98ffa3f76e970b73072b5f9be8472c
  Transport: UDP
  Sent-by Address: 10.1.1.10
  Sent-by port: 5060
  RPort: rport
  Branch: z9hG4bKde98ffa3f76e970b73072b5f9be8472c
  From: "Tony" <sip:7325551234@broadvox.net>;tag=fb717f782bc92d8a
  SIP Display info: "Tony"
  SIP from address: sip:7325551234@broadvox.net
  SIP tag: fb717f782bc92d8a
  To: <sip:7325550819@10.2.2.10>
  SIP to address: sip:7325550819@10.2.2.10
  Call-ID: aba522e3af6bb877f7a7276f31e9e0d4@10.1.1.10
  CSeq: 440180385 INVITE
  Sequence Number: 440180385
  Method: INVITE
  Contact: "Tony" <sip:7325551234@10.1.1.10:5060;transport=udp>
  Contact Binding: "Tony"
<sip:7328891985@10.1.1.10:5060;transport=udp>
  URI: "Tony" <sip:7325551234@10.1.1.10:5060;transport=udp>
  SIP Display info: "Tony"
  SIP contact address: sip:7325551234@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
```

10.1.1.10 Owner/Creator, Session Id (o): UserA 2298741357 1229232207 IN IP4

Owner Username: UserA
 Session ID: 2298741357
 Session Version: 1229232207
 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 49152 RTP/AVP 0 18

8 101

Media Type: audio
 Media Port: 49152
 Media Proto: RTP/AVP
 Media Format: ITU-T G.711 PCMU
 Media Format: ITU-T G.729
 Media Format: ITU-T G.711 PCMA
 Media Format: 101
 Media Attribute (a): rtpmap:0 PCMU/8000
 Media Attribute Fieldname: rtpmap
 Media Format: 0
 MIME Type: PCMU
 Media Attribute (a): rtpmap:18 G729/8000
 Media Attribute Fieldname: rtpmap
 Media Format: 18
 MIME Type: G729
 Media Attribute (a): rtpmap:8 PCMA/8000
 Media Attribute Fieldname: rtpmap
 Media Format: 8
 MIME Type: PCMA
 Media Attribute (a): fmp:18 annexb = no
 Media Attribute Fieldname: fmp
 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): fmp:101 0-15
 Media Attribute Fieldname: fmp
 Media Format: 101 [telephone-event]
 Media format specific parameters: 0-15

Sample SIP INVITE Message from Broadvox 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:7325551234@10.1.1.10:5060;transport=udp, with session description

Frame 1 (1030 bytes on wire, 1030 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:7325551234@10.1.1.10:5060;transport=udp SIP/2.0
Method: INVITE
[Resent Packet: False]

Message Header

Max-Forwards: 69
Session-Expires: 3600;refresher=uac
Supported: timer, 100rel
To: <sip:7325551234@209.249.3.56:5060>
SIP to address: sip:7325551234@209.249.3.56:5060
From: <sip:7325550819@10.2.2.10>;tag=3443898021-537126
SIP from address: sip:7325550819@10.2.2.10
SIP tag: 3443898021-537126
P-Asserted-Identity:<sip:7325550819@4.152.60.7:5060>
Call-ID: 152074-3443898021-537119@NXT02.broadvox.net
CSeq: 1 INVITE

Sequence Number: 1
Method: INVITE

Allow: INVITE, BYE, OPTIONS, CANCEL, ACK, REGISTER, NOTIFY, INFO, REFER, SUBSCRIBE, PRACK, UPDATE

Via: SIP/2.0/UDP

10.2.2.10:5060;branch=z9hG4bKec33b86d10ff83741592076fcd0147e1

Transport: UDP
Sent-by Address: 10.2.2.10
Sent-by port: 5060
Branch: z9hG4bKec33b86d10ff83741592076fcd0147e1

Contact: <sip:7325550819@10.2.2.10:5060>
Contact Binding: <sip:7325550819@10.2.2.10:5060>
URI: <sip:7325550819@10.2.2.10:5060>

SIP contact address: sip:7325550819@10.2.2.10:5060

Call-Info: <sip:10.2.2.10>;method="NOTIFY;Event=telephone-event;Duration=1000"

Content-Type: application/sdp
Content-Length: 251

Message Body

Session Description Protocol

Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): NXT02 19745 20835 IN IP4 10.2.2.10
Owner Username: NXT02
Session ID: 19745
Session Version: 20835
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.2.2.10
Session Name (s): sip call

Connection Information (c): IN IP4 209.249.3.60
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 209.249.3.60
Time Description, active time (t): 0 0
Session Start Time: 0
Session Stop Time: 0
Media Description, name and address (m): audio 44704 RTP/AVP 0 18

101

Media Type: audio
Media Port: 44704
Media Proto: RTP/AVP
Media Format: ITU-T G.711 PCMU
Media Format: ITU-T G.729
Media Format: 101
Media Attribute (a): rtpmap:0 PCMU/8000
Media Attribute Fieldname: rtpmap
Media Format: 0
MIME Type: PCMU
Media Attribute (a): sendrecv
Media Attribute (a): rtpmap:18 G729/8000
Media Attribute Fieldname: rtpmap
Media Format: 18
MIME Type: G729
Media Attribute (a): fmp:18 annexb=no
Media Attribute Fieldname: fmp
Media Format: 18 [G729]
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): fmp:101 0-15
Media Attribute Fieldname: fmp
Media Format: 101 [telephone-event]
Media format specific parameters: 0-15

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