Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Global Crossing SIP Trunking Service and an Avaya IP Office Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Global Crossing VoIP Network and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office and Avaya H.323, digital, and analog endpoints.

Headquartered in Hamilton, Bermuda, Global Crossing Limited provides Internet Protocol (IP) and legacy telecommunications services worldwide. Enterprise customers with an Avaya IP Office SIP-based network can connect to the Global Crossing VoIP Network over the Internet and access the PSTN by subscribing to Global Crossing Enterprise VoIP services: DID, Toll Free and VTERM.

Global Crossing 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 Global Crossing's VoIP Network and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office and Avaya H.323, digital, and analog endpoints.

Global Crossing is an international service provider offering a wide variety of telecommunications and data services such as:

- Domestic and international long-distance traffic
- Direct-inward-dialing transport
- Toll-free enhanced routing services via traditional TDM or VoIP interconnections
- Converged IP services, IP virtual private network (VPN) services Internet access services
- Frame Relay and Asynchronous Transfer Mode (ATM) services

Customers using this Avaya IP Office telephony solution with Global Crossing's VoIP Network services 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.

Global Crossing can connect directly to an IP phone system as well as an external router. Global Crossing's VoIP Network service offerings are as follows:
- **DID** – IP Inbound to customer
- **Toll Free** – IP Inbound to customer
- **VTERM** – IP Outbound from customer

Figure 1 illustrates a sample Avaya IP Office telephony solution connected to Global Crossing's VoIP Network solution. This configuration was utilized for compliance testing.

Note: The IP Addresses shown in **Figure 1** and throughout this document are not the actual public addresses used during testing. They are simply used as an example for these Application Notes. The real IP Addresses are not revealed for security purposes.

The following equipment comprised the Avaya IP Office telephony solution and simulated a customer site:
- Avaya IP Office 406v2
- Avaya 4610 SW IP Telephone (H.323 protocol)
- Avaya 6424D+M Digital Telephone
- Avaya 6210 Analog Telephone
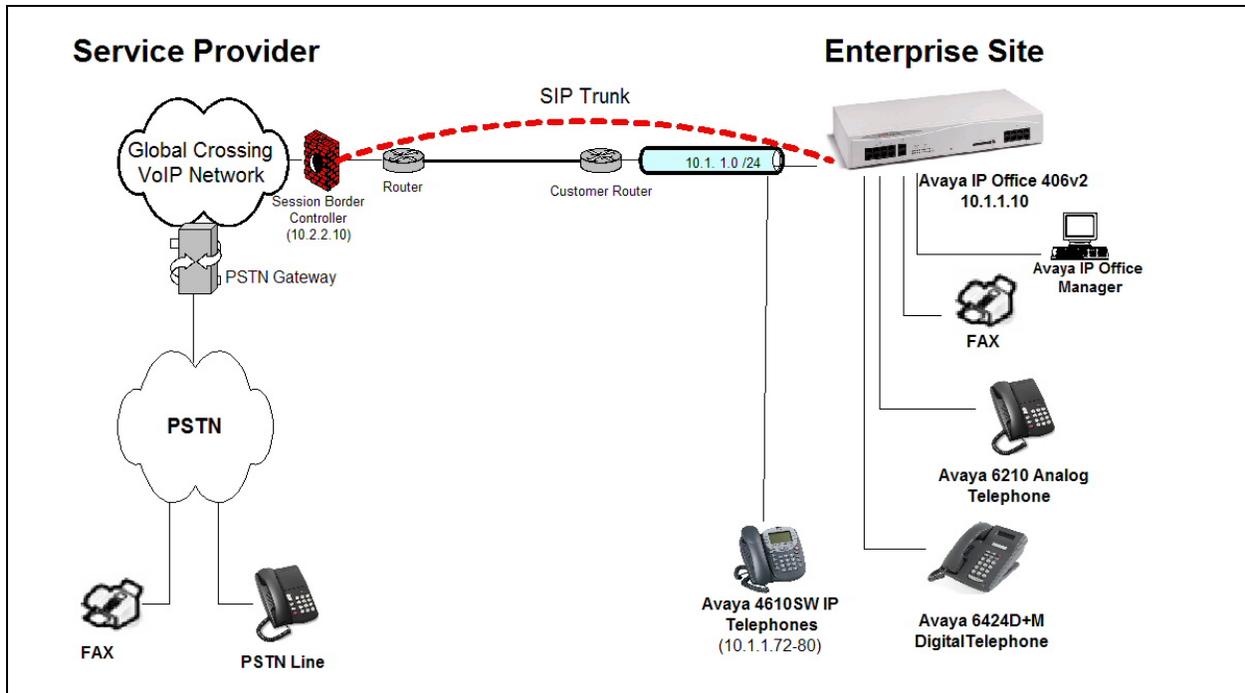
Figure 1: Avaya IP Office Telephony Network using Global Crossing's VoIP Network solution

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya IP Office telephony solution Components	
Avaya IP Office 406v2	R 4.1.9
Avaya IP Office Manager (Windows PC)	R 6.1 (9)
Avaya 4610SW IP Telephone	H.323 – a10d01b2824.bin
Avaya 6424D+M Digital Telephone	n/a
Avaya 6210 Analog Telephone	n/a
Global Crossing SIP Trunking Solution Components	
Acme Packet Net-Net Session Director	4.1.1 P56a
Sonus NBS	V06.00.25 R000

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 **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.
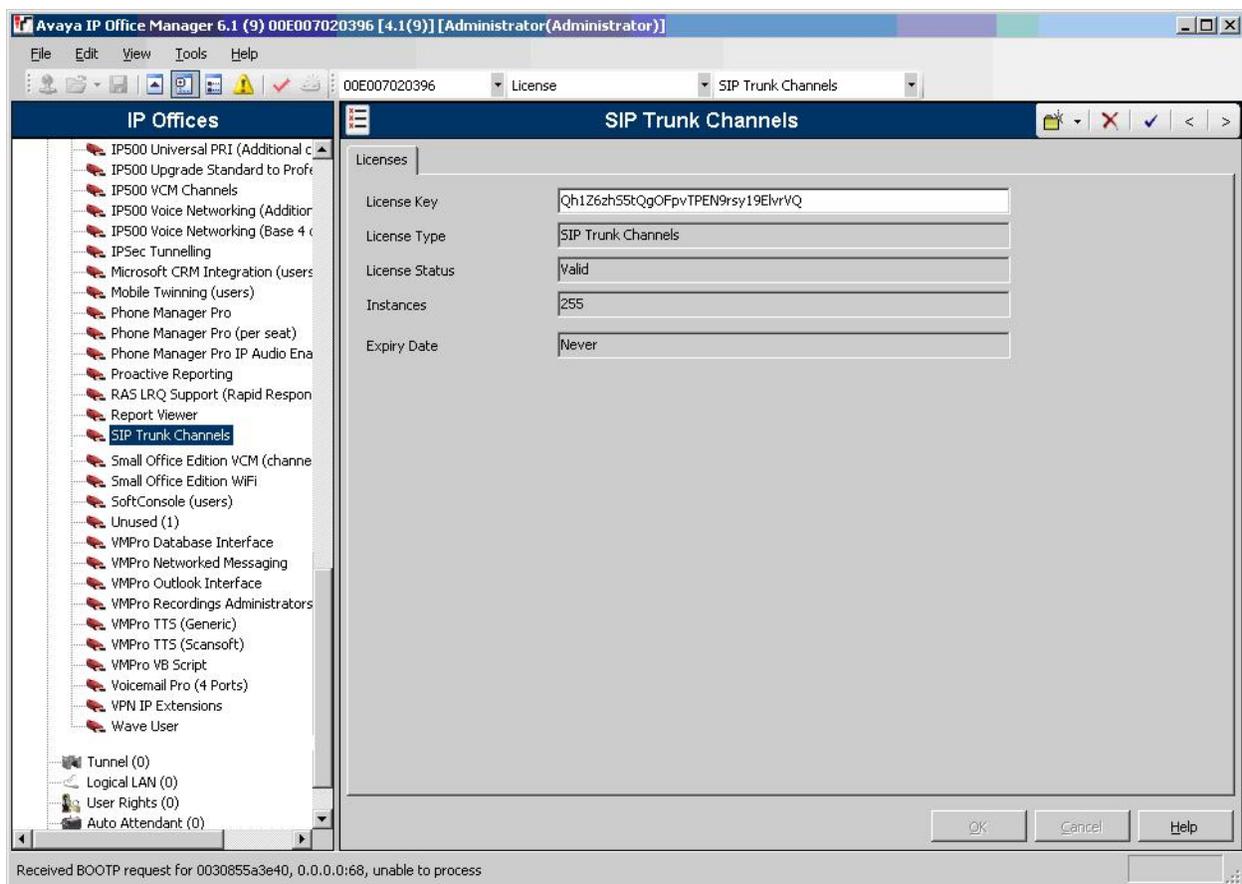
2. *Enable SIP Proxy Functionality.* Select **System** in the left panel. In the **LAN1** tab, select the **Gatekeeper tab,** and check the **SIP Proxy Enable** box. Press the **OK** button. Note: Some changes may require a system reboot to take effect.
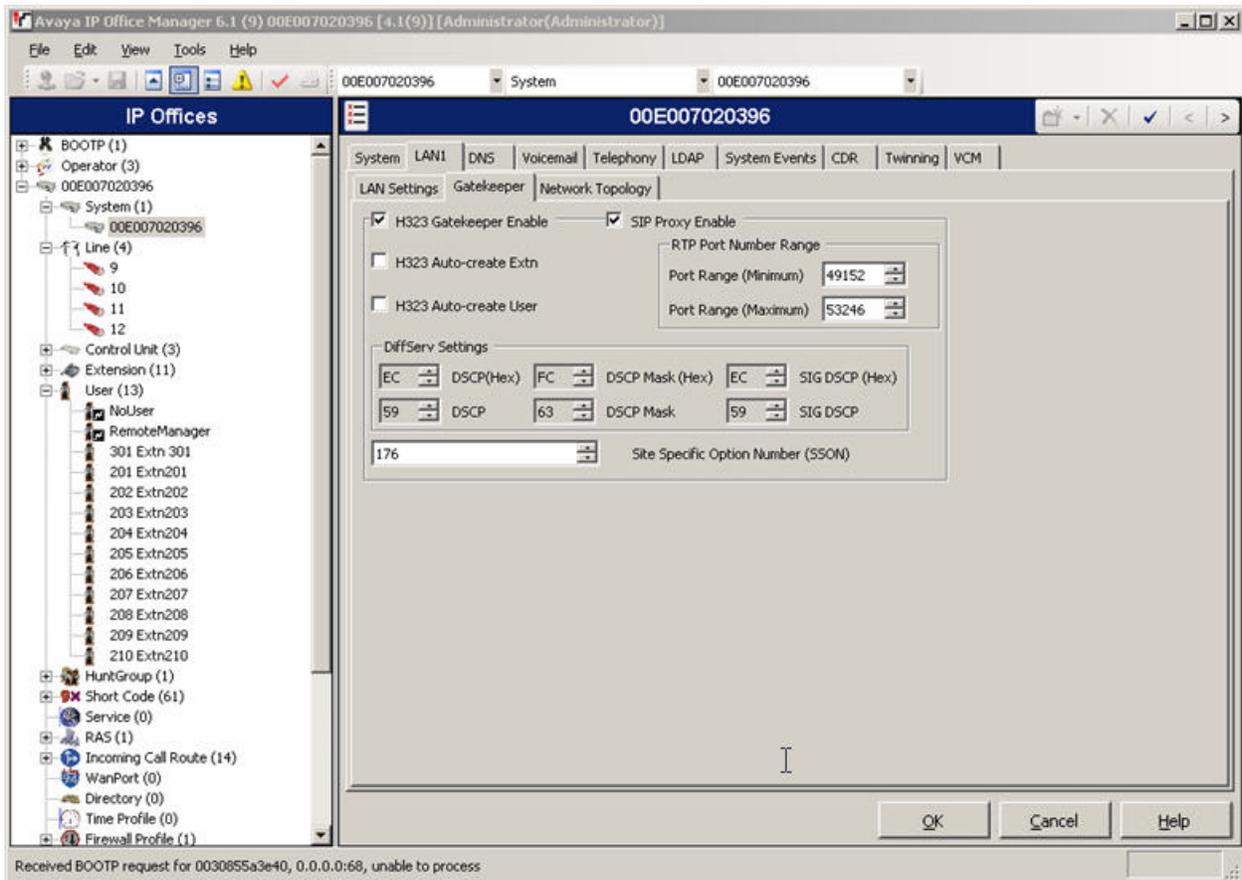
3. *Create the SIP line for the Global Crossing service*. Select **Line** in the left panel. Right-click and select **New → SIP Line**.

    Select the following:
    - For the **ITSP IP Address field**, enter the IP address of the Global Crossing SIP Proxy
    - For **Compression Mode**, select the **G.711 ULAW 64K** or **G729a 8K CS-ACELP** for all voice calls. (Global Crossing may support additional codecs upon request.)
    - For **Layer 4 Protocol**, use **UDP**
    - For **Send Port**, use **5060**
    - For **Listen Port** use **5060**
    - Use defaults for other fields
    Press the **OK** button.
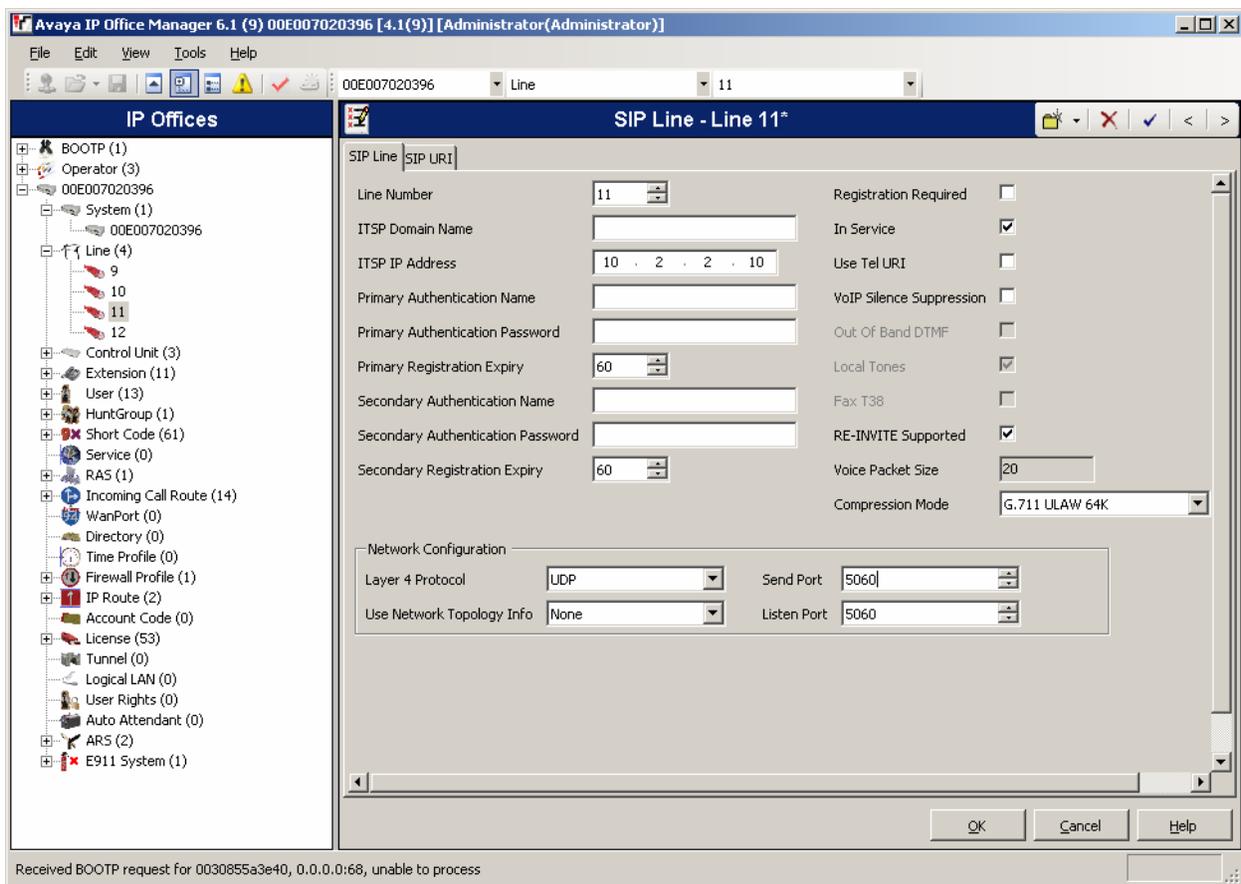
4. *Configure URI parameters for the line*. Select the **SIP URI** Tab. Press the **Add** button.

5. 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 7**). Use defaults for all other fields. Press the **OK** button.
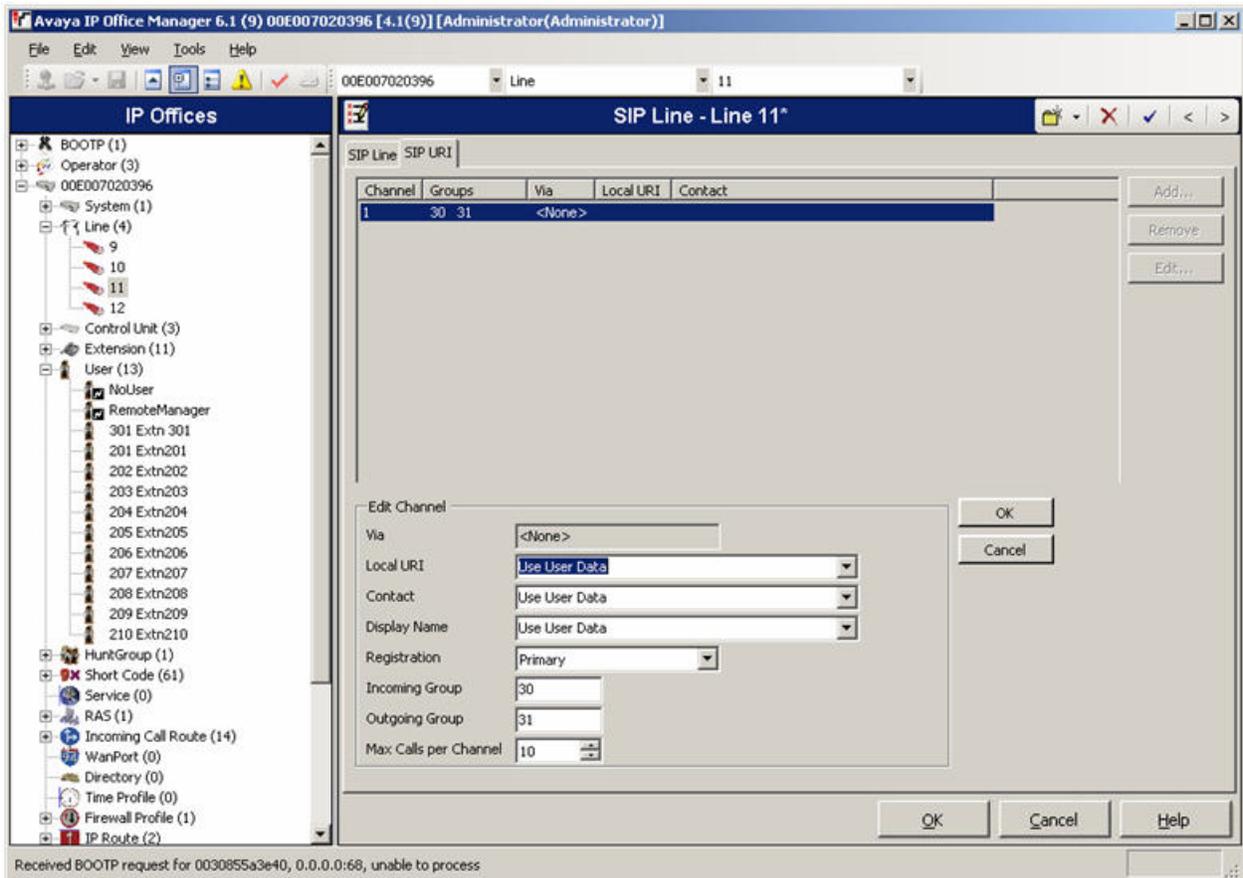
6. *Configure a short code to route calls to Global Crossing's SIP Proxy Server.* Select **Short Code** in the left panel. Right click and select **Add**. Enter **[x]N;**, where **[x]** is a valid number, in the **Code** text box. The number **9** is used for **[x]** in the below example. This code requires 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, a 10-digit Direct Inward Dial (DID) number, operator assistance, 411, information service etc.

7. Select **Dial** for the **Feature**. Enter the **Outgoing Group** created in **Step 4** for the **Line Group Id** field. Enter the dialed number **N** followed by "**@<IP Address of Global Crossing >**" 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.
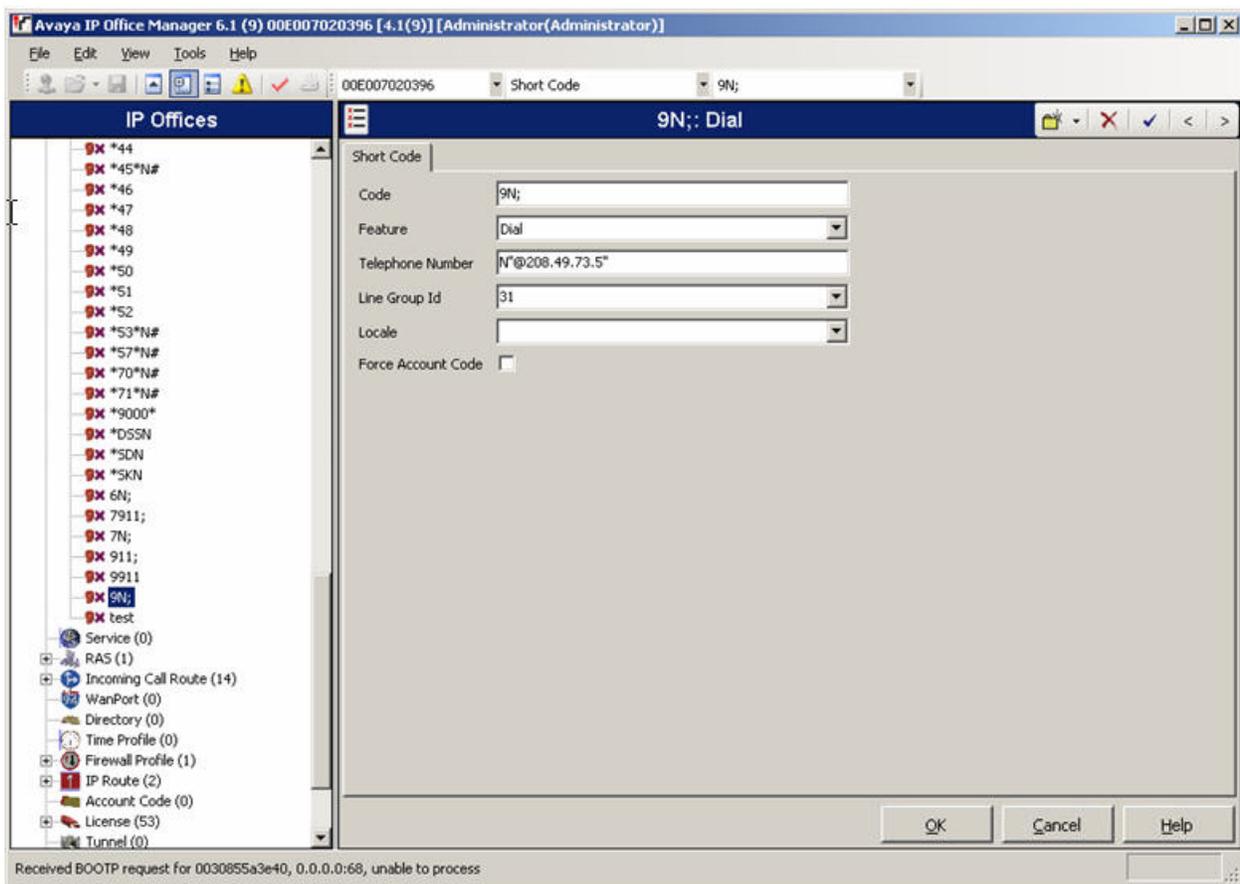
8. *Create an Incoming Call Route for the Inbound SIP calls.* Select **Incoming Call Route** in the left panel. Right-click and select **New**.

Enter the following in the **Standard** tab:
- **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 Global Crossing, that is mapped back to a local IP Office extension, in the **Incoming Number** field.
- Use default values for all other fields.
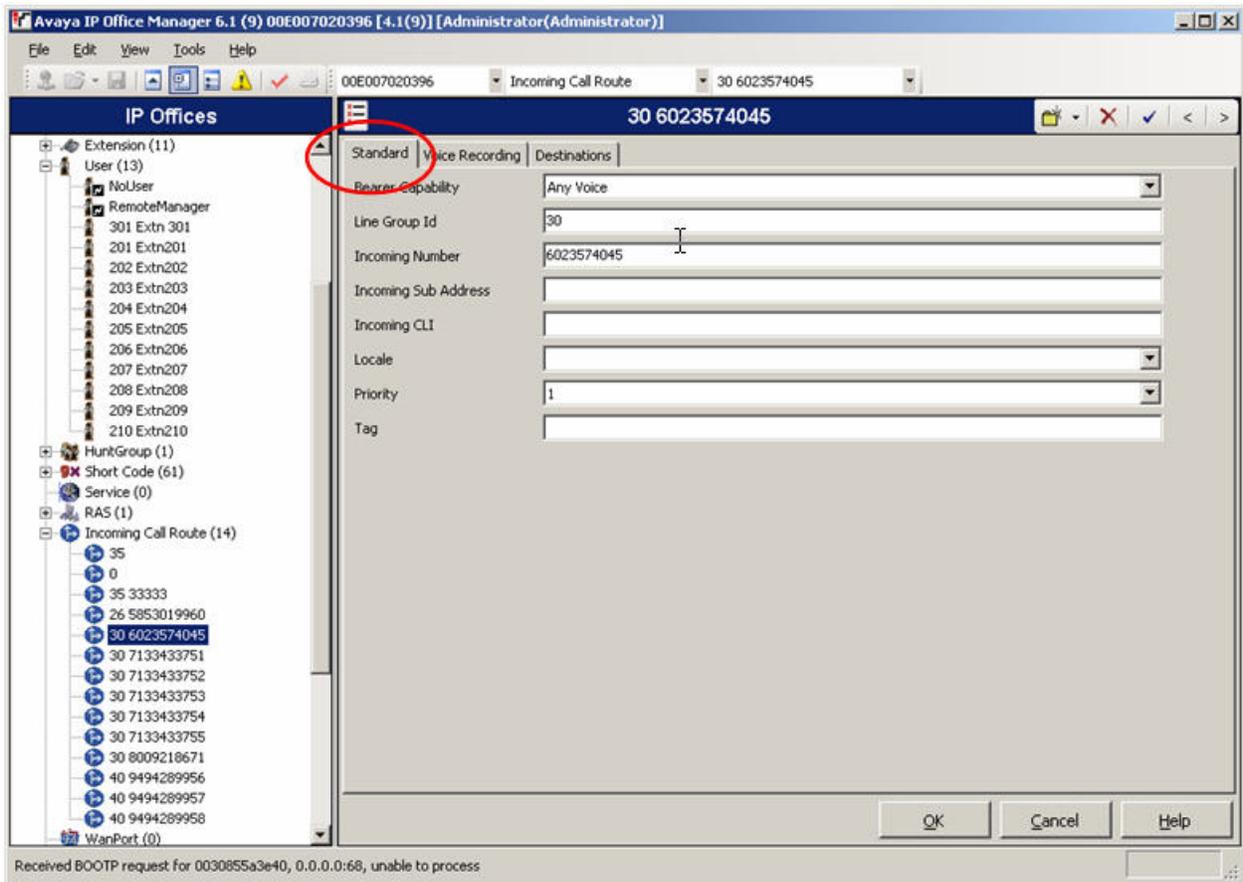
- Next, navigate to the **Destinations** tab, select the desired local extension number from the drop down menu.
- Press the **OK** button.

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GCrossingIPO
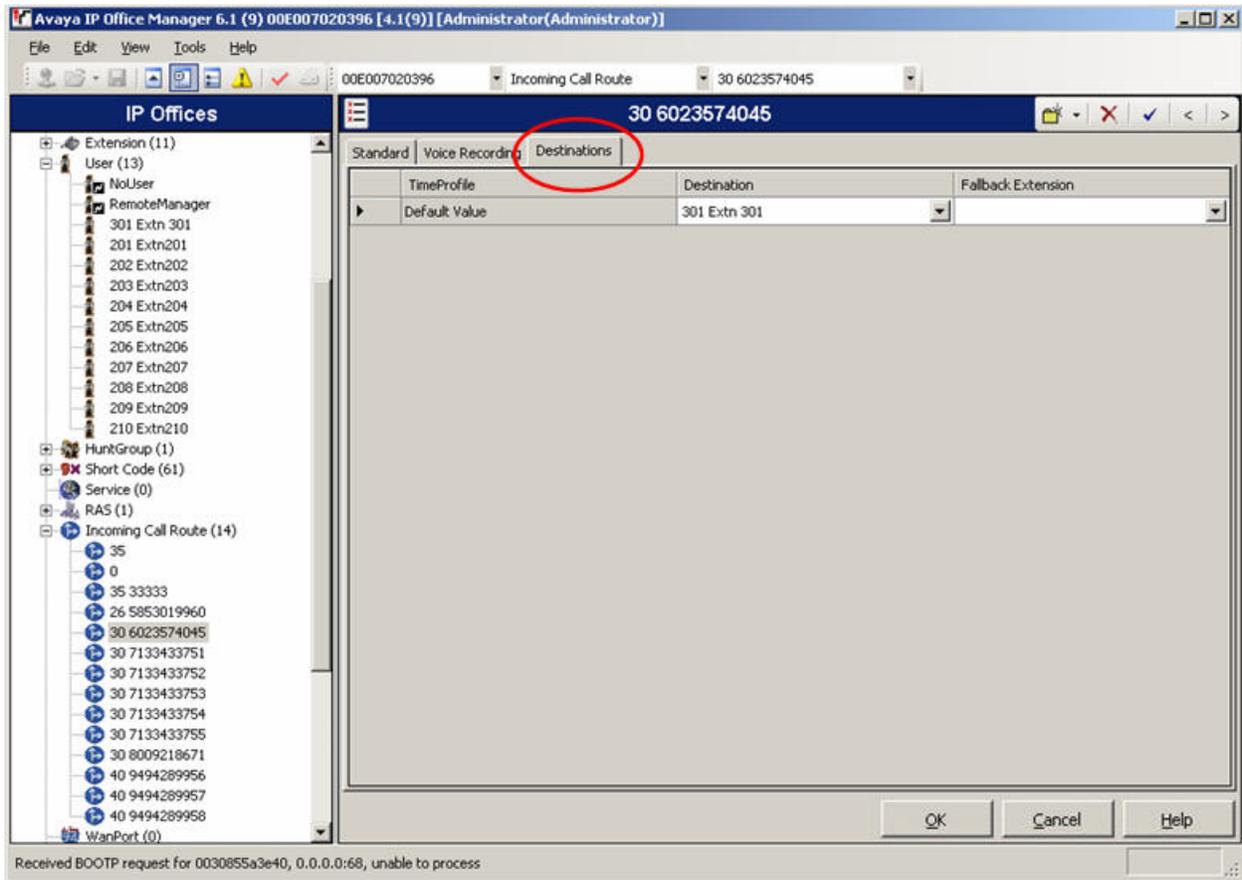
9. *Configure Users' SIP names with the authenticated DID.* Select **User** in the left panel. Select the desired user by double-clicking on an entry in the right panel. Select the **SIP** tab all the way at the end.

> Modify the **SIP Name, SIP Display Name (Alias)** and **Contact** fields to one of the DID numbers provided by Global Crossing. This valid DID number provided by Global Crossing is used as authentication for calls coming into their VoIP network. This number should match the number used in the **Incoming Call Route**, see **Step 6**. 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
>
> Global Crossing's SIP equipment then verifies that the user part of the SIP URI in the INVITE's From header from IP Office is an authenticated user. (see Appendix A for examples of SIP INVITE messages.)
>
> The other fields can be left as defaults. Press the **OK** button.
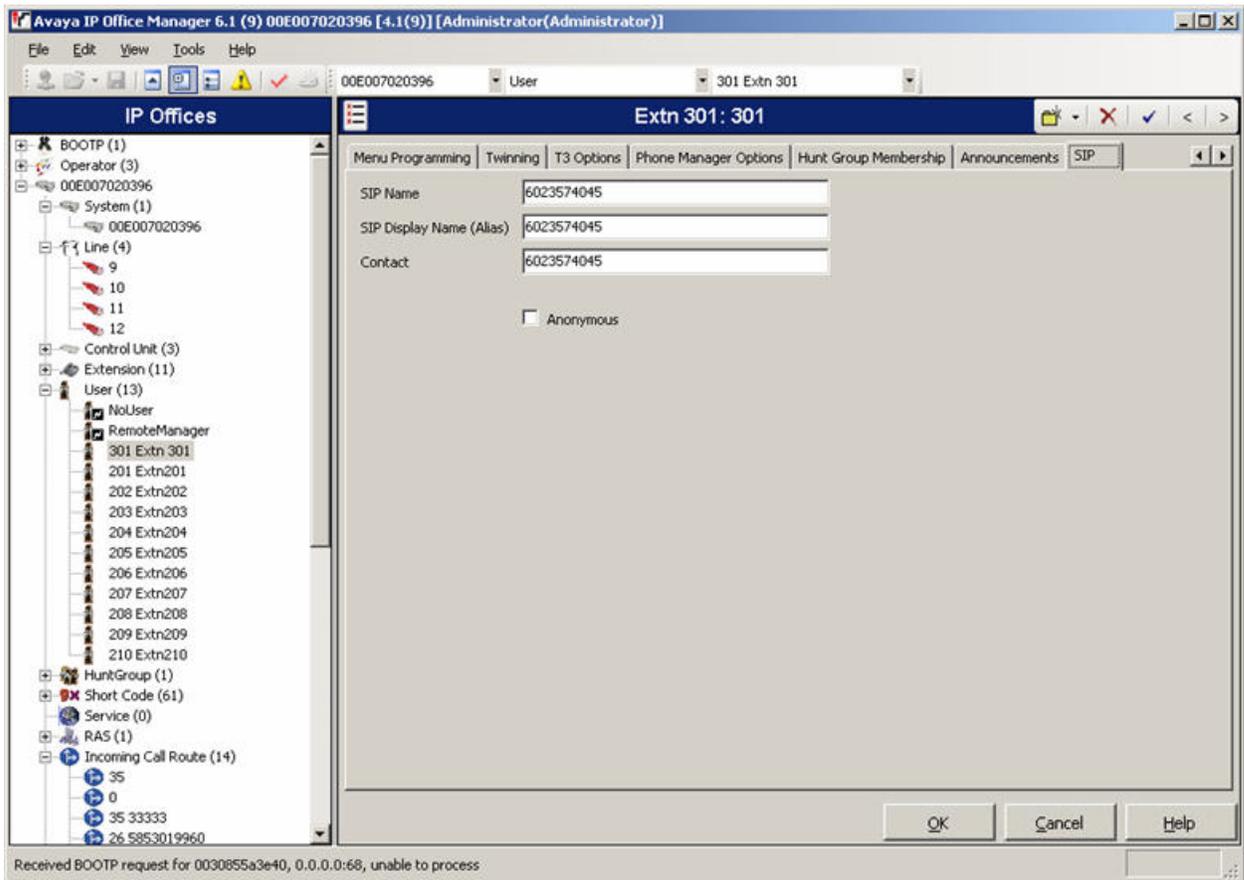
10. Repeat **Steps 6** and **7** for all users that will be sending/receiving external SIP calls on the system.

4. Global Crossing Services Configuration

The following table contains the configuration information, provided by Global Crossing, which was used during the interoperability compliance testing.

Service / Feature Information	Test Configuration
Global Crossing Enterprise VoIP Service(s): • **DID** – IP Inbound to customer • **Toll Free** – IP Inbound to customer • **VTERM** – IP Outbound from customer	All Global Crossing Enterprise VoIP Services were tested in the Application Notes.
Global Crossing Codec(s): • G.711mu-law • G.729A • RFC2833 DTMF • G.711mu Pass Through facsimile mode	The network configuration described in these Application Notes was tested with all the codecs (payload types) listed in the left column.
Global Crossing Dial Plan	10-Digit North American and International (011 Country Code) were supported by the test configuration. The numbering plan and routing choices are affected by the services used and will be customer dependent.
Global Crossing Direct Inward Dialing Numbers	Direct inward dialing numbers should be assigned to the endpoints at the enterprise site. This allows calls to be delivered from the PSTN. In this configuration, direct inward dialing numbers beginning with area code 602-357 were assigned to the H.323, digital, and analog endpoints in the enterprise network.
Global Crossing Proxy IP Address	The IP address of the Acme Packet Net-Net Session Director in the Global Crossing network was 10.2.2.10.

Customer IP Address of Avaya IP Office	The IP address of Avaya IP Office in the enterprise network was 10.1.1.10. Global Crossing used this IP address for routing calls destined to the listed directory numbers assigned to the enterprise site.
SIP Transport Protocol and Port	SIP signaling was transported between Avaya IP Office and Global Crossing using UDP and port 5060.

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between Global Crossing's VoIP Network and an Avaya IP OfficeTelephony 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 DID, Toll Free and VTERM services provided by Global Crossing. 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 Global Crossing.
- Outgoing calls from the enterprise site were completed via Global Crossing 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, Toll Free, and directory assistance calls.
- Calls using the G.711mu and G.729a codec types.
- Fax calls using the G.711mu codec.
- DTMF tone transmission using RFC 2833 with successful voice mail navigation with the G.711mu and G.729a codec.
- 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.

- Private Calling from the PSTN to Avaya IP Office is not currently operational in Global Crossing's network.

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 Global Crossing'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 support on Global Crossing Enterprise VoIP Services, contact the Global Crossing Customer Center at 866-456-2259 (866-GLOBALX).

9. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony solution to Global Crossing's service. Global Crossing 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 between Global Crossing and the Avaya IP Office Telephony Solution 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 Global Crossing to Avaya IP Office:

Request-Line: INVITE sip:6023574045;npdi=yes@10.1.1.10:5060 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 10.2.2.10:5060;branch=z9hG4bKn3r1nv10bgg0tb0bc1o1.1
    From: <sip:7324500819@10.2.2.10:5060;isup-oli=0>;tag=SDckhn001-gK062f1a95
    To: <sip:6023574045@10.1.1.10:5060>
    Call-ID: SDckhn001-f5408d745600b29903f0896793d2f8ff-v3000i1
    CSeq: 188 INVITE
    Max-Forwards: 69
    Allow: OPTIONS,INVITE,ACK,CANCEL,BYE,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE
    Accept: multipart/mixed, application/sdp, application/isup, application/dtmf, application/dtmf-relay
    Contact: <sip:7324500819@10.2.2.10:5060;transport=udp>
    Remote-Party-ID: <sip:7324500819@10.2.2.10:5060>;privacy=off
    P-Charge-Info: sip:7324500819@10.2.2.10:5060
    Supported: timer
    Session-Expires: 64800
    Min-SE: 64800
    Content-Length: 279
    Content-Disposition: session; handling=required
    Content-Type: application/sdp
  Message body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): Sonus_UAC 873 22593 IN IP4 10.2.2.10
      Session Name (s): SIP Media Capabilities
      Connection Information (c): IN IP4 10.2.2.10
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 50048 RTP/AVP 0 18 100
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:18 G729/8000
      Media Attribute (a): fmtp:18 annexb=no
      Media Attribute (a): rtpmap:100 telephone-event/8000
      Media Attribute (a): fmtp:100 0-15
      Media Attribute (a): sendrecv
      Media Attribute (a): maxptime:20

Sample SIP INVITE Message from Avaya IP Office to Global Crossing:

Request-Line: INVITE sip:17324500819@10.2.2.10 SIP/2.0
  Message Header
    Via: SIP/2.0/UDP 10.1.1.10:5060;rport;branch=z9hG4bKed9d684e1cec823cd6b802f1911bb903
    From: 8009218671 <sip:8009218671@10.1.1.10>;tag=d026801e1e849b65
    To: <sip:17324500819@10.2.2.10>
    Call-ID: a84008b67bc5bc749201afa8c95571fa@10.1.1.10
    CSeq: 1111794815 INVITE
    Contact: 8009218671 <sip:8009218671@10.1.1.10:5060;transport=udp>
    Max-Forwards: 70
    Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
    Content-Type: application/sdp
    Content-Length: 301
  Message body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): UserA 2451659306 1932039530 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 0 18 4 8 101
      Media Attribute (a): rtpmap:0 PCMU/8000
      Media Attribute (a): rtpmap:18 G729/8000
      Media Attribute (a): rtpmap:4 G723/8000
      Media Attribute (a): rtpmap:8 PCMA/8000
      Media Attribute (a): fmtp:18 annexb = no
      Media Attribute (a): rtpmap:101 telephone-event/8000
      Media Attribute (a): fmtp:101 0-15