



## Avaya Solution & Interoperability Test Lab

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# **Application Notes for Configuring SIP Trunking between Global Crossing Voice over IP services including VoIP On-Net Plus™, VoIP Outbound™, VoIP Local Service, and Global Crossing VoIP Toll Free Service with Avaya Communication Manager Branch Edition – Issue 1.0**

## **Abstract**

These Application Notes describe the steps for configuring SIP trunking between Global Crossing Voice over IP services and an Avaya Communication Manager Branch Edition (Release 1.2), formerly known as Avaya Distributed Office using various Avaya telephony endpoints.

Global Crossing SIP trunks are managed Voice over IP communication solutions using SIP trunks to provide inbound and outbound local, long distance, international, and toll-free services for US sites.

Global Crossing is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction and Scope

These Application Notes describe the steps for configuring SIP trunking between the Global Crossing SIP trunk service offer and the Avaya Communication Manager Branch Edition (Release 1.2), formerly known as Avaya Distributed Office using various Avaya telephony endpoints.

SIP (Session Initiation Protocol) is a standards-based VoIP communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [13] 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, media connection host address, RTP payload type, etc. Within these Applications Notes, SIP is used as the signaling protocol between the Avaya Communication Manager Branch Edition and the network services offered by Global Crossing.

The Global Crossing offerings are Voice over IP communication services using SIP trunks to provide inbound and outbound local, long distance, international and toll-free services for US sites. Below are the Global Crossing Offers utilized as part of the interoperability testing:

- **Global Crossing VoIP On-Net Plus™[9]**  
The Global Crossing VoIP On-Net Plus™ service offer allows customers to connect locations to Global Crossing via a dedicated facility, and send IP or TDM originated voice traffic for onnet completions without traditional or international long distance charges
- **Global Crossing VoIP Outbound™[10]**  
Global Crossing accepts enterprise outbound IP voice traffic for national long distance and international long distance calling to more than 240 countries worldwide
- **Global Crossing VoIP Local Service™[11]**  
Global Crossing provides local origination for nationwide local numbers Direct Dial-In/Direct Inward Dialing (DDI/DID) through a single IP point of interconnection. This service eliminates traditional time division multiplexing, private line and foreign exchange service fees by providing a single IP connection alternative that can support all originating markets services by Global Crossing
- **Global Crossing VoIP Toll Free Service™[12]**  
Global Crossing VoIP Toll Free service accepts originating Toll Free traffic from the public switched telephone network covered under the North American dialing plan, and converts it to IP for transport across Global Crossing's private VoIP platform for IP completion to customer's location within the U.S. and Canada

In the remainder of this document, “Global Crossing services” will be used to generically refer to all Global Crossing offers.

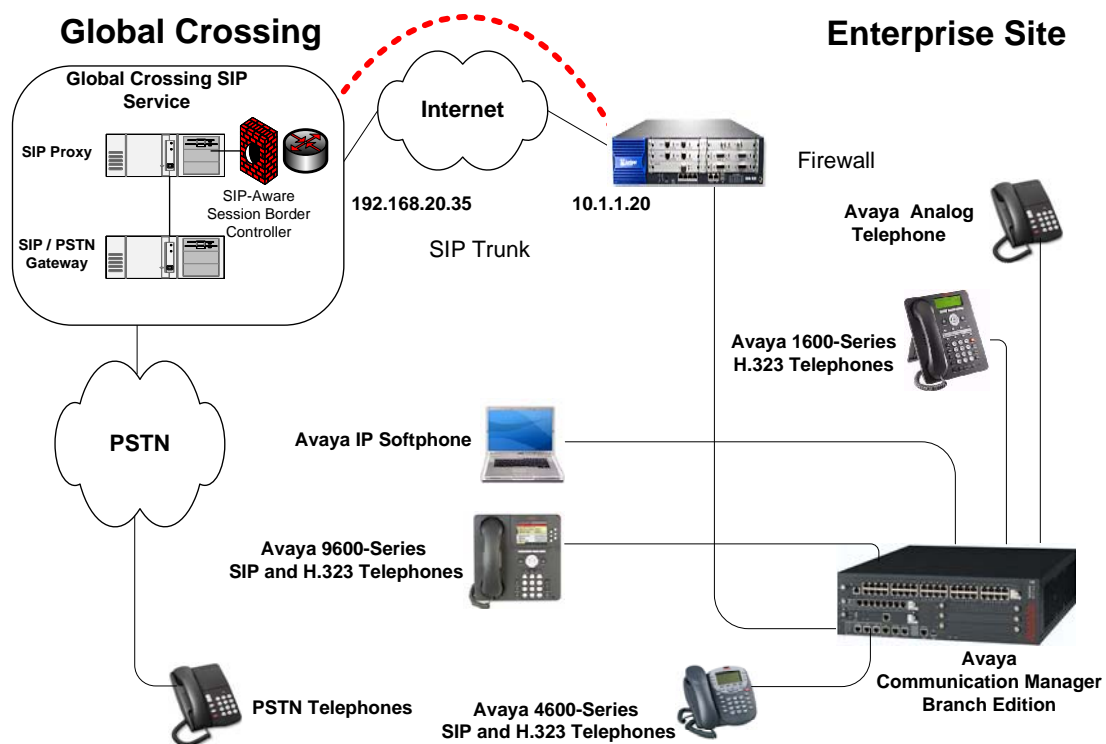
## 2. Network Topology

**Figure 1** illustrates a typical customer location using an Avaya Communication Manager Branch Edition with SIP trunking to Global Crossing’s network. This configuration includes:

- Avaya Communication Manager Branch Edition i120 providing the communication services for this customer location
- Avaya telephones including SIP, H.323, and analog
- IP routing and data network infrastructure to support IP connectivity between the enterprise location and the Global Crossing services

The SIP trunking configuration described within this document assumes a previously configured system capable of station to station calling. These Application Notes only cover configuration steps relevant to this interface. Certain customer configurations are beyond the scope of these Application Notes. Below are examples of administration not shown:

- The initial installation and administration of the Avaya Communication Manager Branch Edition to provide basic telephony services
- Detailed administration being performed individually for the i120 and i40 (i40 was not used during the test) platforms since the administration is identical on each platform
- Administration of trunks other than SIP trunks
- Configuration of stations in Avaya Communication Manager Branch Edition
- IP Network Address Translation (NAT), firewalls, Application Layer Gateway (ALG), and/or Session Border Controller (SBC) devices may exist between the Global Crossing services and the Avaya Communication Manager Branch Edition within a customer’s communications infrastructure and should be generally SIP-aware. When configured correctly, they are transparent to the Avaya communications infrastructure



**Figure 1: Network configuration used to validate the Global Crossing services with Avaya Communication Manager Branch Edition.**

### 3. Hardware and Software Validated

Table 1 lists the equipment and software/firmware versions used to validate the solution.

Network Component	Version Information
<b>Avaya</b>	
Avaya Communication Manager Branch Edition (i120)	Version 1.2.0_24.05
Avaya 4610 IP Phone (H.323)	Version 2.9
Avaya 1608 IP Phone (H.323)	Version 1.0.2
Avaya 9620 one-X™ SIP Phone	Version 2.0.4
Avaya IP Softphone	Release 6.0 Service Pack 4
Avaya 6211 Analog Phone	N/A
brother IntelliFAX 1360	N/A
Venta Fax & Voice Fax Application	Release 2.8
Juniper SSG-520M Firewall	Version 6.1.0r1.0
<b>Global Crossing</b>	
Global Crossing VoIP On-Net Plus™	N/A
Global Crossing VoIP Outbound™	N/A
Global Crossing VoIP Local Service™	N/A
Global Crossing VoIP Local Service™	N/A
Sonus GSX (NBSE)	Version V06.04.10 R006

**Table 1 – Equipment Version Information**

Table 2 lists the Global Crossing Provisioning information used to validate the solution.

<b>Global Crossing Provisioning Information</b>	<b>Illustrated Values in this Compliance Test</b>
Global Crossing Border Element IP Address(es)	192.168.20.35 192.168.20.36 (media)
G.711U, G.711A, G.729A, G.729AB Codec Support	Yes
RFC 2833 (DTMF Event) Supported	Yes
Via Header Routing	Yes
Maximum Concurrent Calls (specified by customer during service ordering)	10
Assigned Direct Inward Dial Numbers	602-466-72xx (where x is any digit)
DID Digits Passed in SIP Request URI	602-466-72xx
DID Digits Passed in SIP To Header	Same as SIP Request URI
Incoming Toll-Free Number	888-246-9143
Incoming Toll-Free Digits Passed in SIP Request URI	888-246-9143
Incoming Toll-Free Digits Passed in SIP To Header	888-246-9143

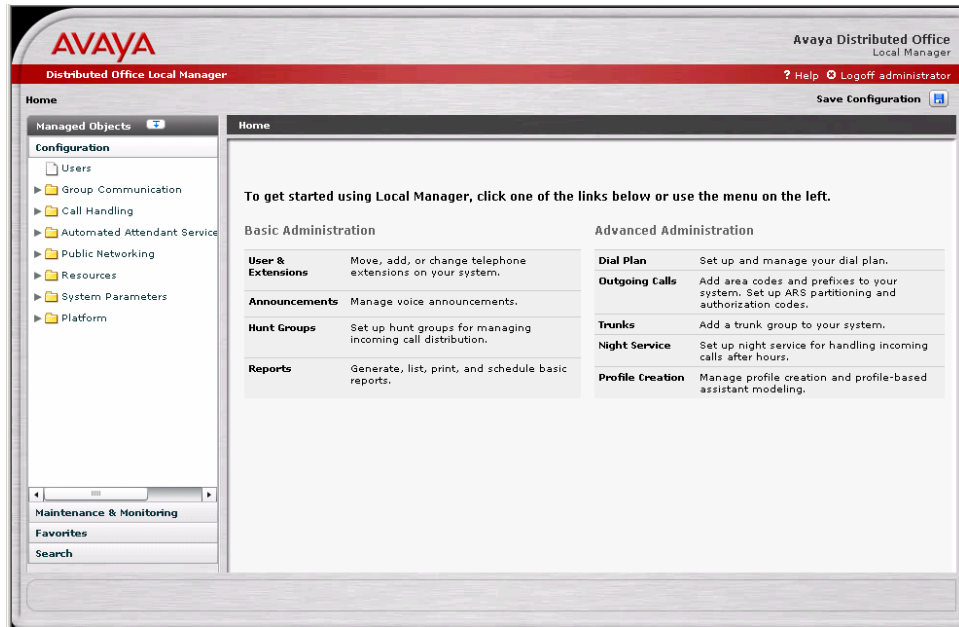
**Table 2 – Global Crossing Provisioning Information**

## **4. Configuring Avaya Communication Manager Branch Edition**

Initially, the Avaya Communication Manager Branch Edition i120 was provisioned for extension calling. The following is the additional provisioning performed to support the Global Crossing SIP trunking configuration. Initial provisioning details are not within scope of this application note.

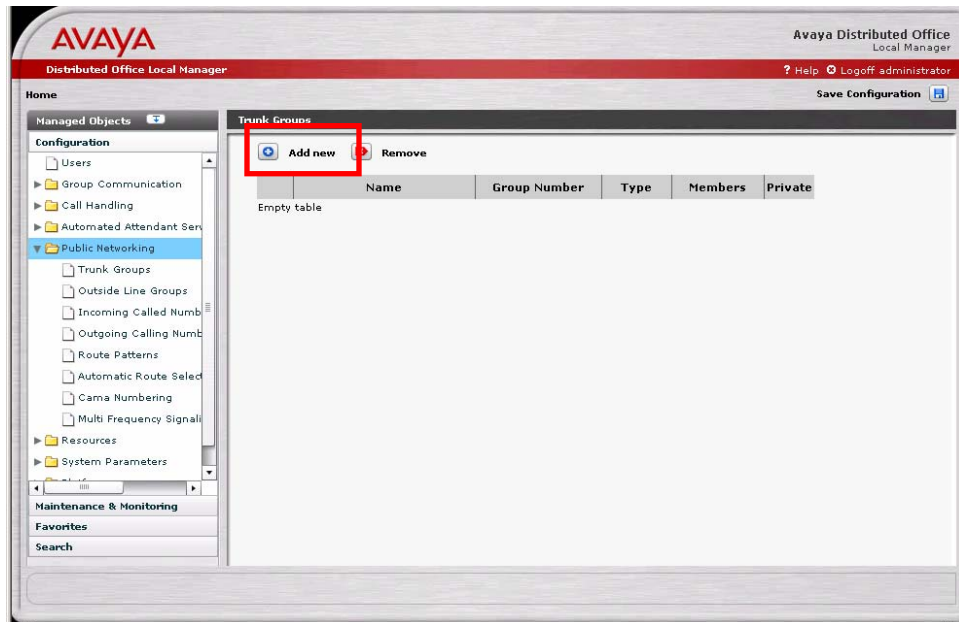
## 4.1. Accessing the Avaya Communication Manager Branch Edition

Launch a web browser and type the Host IP Address of the Avaya Communication Manager Branch Edition Local Manager in the web browser. After successfully logging in, The **Local Manager** appears as shown:



## 4.2. Adding a SIP trunk to connect to the Global Crossing services

From the **Configuration** menu in the left pane, expand the **Public Networking** option and click **Trunk Groups**. The **Trunk Groups** screen will be displayed. Click **Add new** to display the Add Trunk Group page.





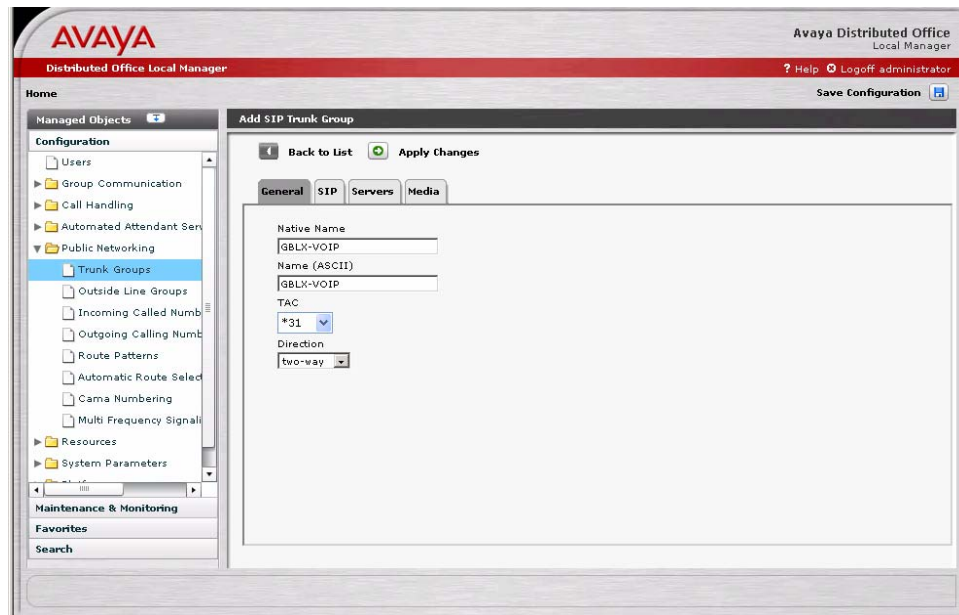
On the **Add Trunk Group** screen do the following:

1. Set the **Trunk Type** to “SIP”.
2. Set the **Native Name** field to any descriptive string (e.g. “GBLX-SIP”).
3. The **Name (ASCII)** field defaults to the entry in the Native Name field. This can be modified if desired.
4. Click the **Continue** button.

The screenshot shows the Avaya Distributed Office Local Manager interface. The title bar includes the Avaya logo and the text 'Avaya Distributed Office Local Manager'. The main window has a red header bar with 'Distributed Office Local Manager' and links for 'Help' and 'Logoff administrator'. A 'Home' button is in the top right. On the left is a 'Managed Objects' tree with categories like 'Configuration', 'Maintenance & Monitoring', 'Favorites', and 'Search'. Under 'Configuration', 'Trunk Groups' is selected. The main area is titled 'Add Trunk Group' and contains the instruction: 'Specify the following required information and then click **Continue**.' Below this are four fields: 'Trunk Type' (a dropdown menu showing 'SIP'), 'Trunk Group Number' (a text box with '31'), 'Native Name' (a text box with 'GBLX-VOIP'), and 'Name (ASCII)' (a text box with 'GBLX-VOIP'). At the bottom of the form are 'Cancel' and 'Continue' buttons.

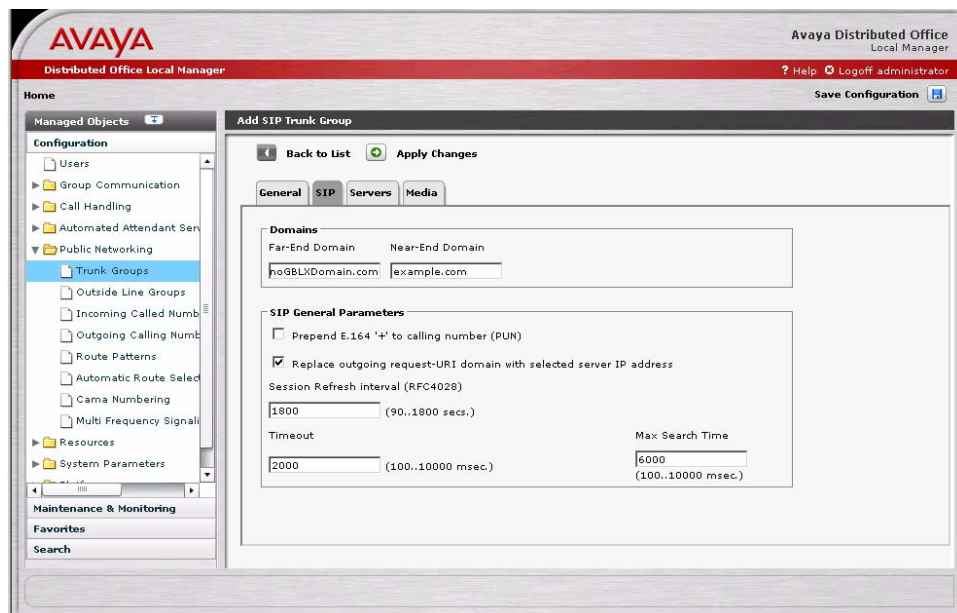
The **Add SIP Trunk Group** screen is displayed. Select the **General** tab.

1. Select the desired Trunk Access Code (TAC) or accept the default value.
2. In the **Direction** field select “two-way” to allow both incoming and outgoing calling on this trunk group.
3. Click the **SIP** tab to enter the **SIP** screen.



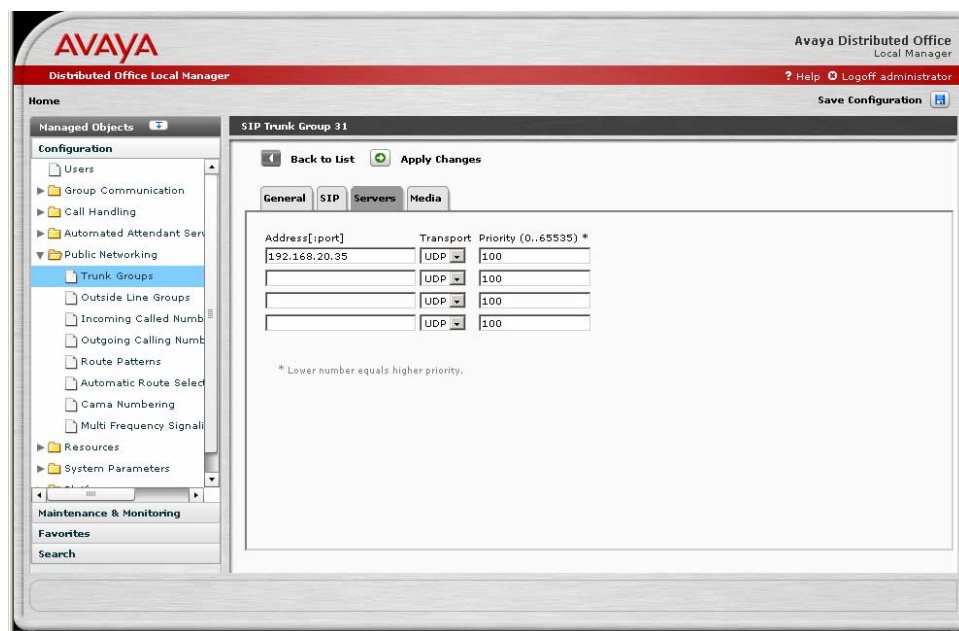
At the **SIP** screen:

1. Enter a **Far-End Domain** value for the Global Crossing services. Since Global Crossing uses a dotted IP address format rather than Domain names, enter “noGBLXDomain.com” as a placeholder. This domain is not required to be resolvable in DNS.
2. Enter the customer’s SIP domain in the **Near-End Domain** field. “example.com” was used during the verification testing. This domain is not required to be resolvable for the Global Crossing SIP trunking.
3. Check the **Replace outgoing request-URI domain with the selected server IP address** box.
4. Enter “1800” in the **Session Refresh interval (RFC4028)** field.
5. Set the **Timeout** field to “2000”.
6. Set the **Max Search Time** field to “6000”.
7. Click the **Servers** tab to enter the **Servers** screen.



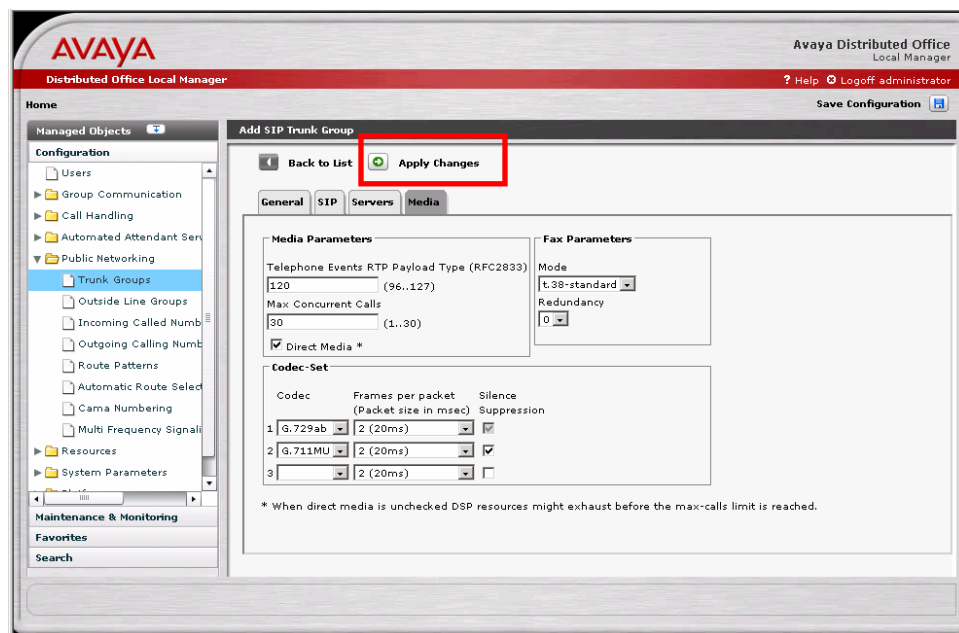
At the **Servers** screen:

1. Enter the IP address of the primary Global Crossing Border Element provided by Global Crossing in the **Address[:port]** field. In this compliance test, 192.168.20.35 is used as noted in **Section 3**. Specifying the port is not required since the UDP default port “5060” is used.
2. In the **Transport** field select “UDP”.
3. The default **Priority** field setting “100” was used.
4. Repeat **Steps 1 - 3** for additional Border Element addresses if needed.
5. Click the **Media** tab to enter the **Media** screen.



At the **Media** screen:

1. Set the **Telephone Events RTP Payload Type (RFC2833)** field to match the value used by Avaya 96xx series SIP phones. During compliance testing “120” was used. This value matches Avaya 96xx series SIP phone default.<sup>1</sup>
2. Set the **Max Concurrent Calls** field to the number of simultaneous calls supported. This value is specified by the customer when ordering the Global Crossing services. It is a function of the bandwidth of the VoIP network access, codec choices, and Global Crossing service limits.
3. Check the **Direct Media** option. Checking this allows the media path to be routed directly to IP and SIP telephones thus conserving DSP resources.
4. Select **Codec** field in row 1 to use “G.729A” or “G.729AB” to be used as the preferred codec choice.<sup>2</sup>
5. Select **Codec** field in row 2 to use “G.711MU” as the second codec choice.
6. Select the “2 (20 ms)” **Frames per packet** field choice for both codecs.
7. Check the **Silence Suppression** field for both codecs.
8. In the **Mode** field select “t.38-standard” and in the **Redundancy** field select “0”.
9. Click **Apply Changes** to apply the settings.



<sup>1</sup> The value used by the 96xx telephones can be modified by changing the SET DTMF\_PAYLOAD\_TYPE value within the 46xxsettings.txt file used during telephone initialization. Details regarding this administration are beyond the scope of these Application Notes (but are found in Reference [8]).

<sup>2</sup> Coordinate codec selection with Global Crossing to assure proper codec compatibility and negotiation.

### 4.3. Configuring the Outgoing Calling Number

The entries in the Outgoing Calling Number fields determine the calling number that will be sent in the **SIP From** header for Avaya Communication Manager Branch Edition stations. In this compliance test the extension numbers used are in the range of 20000 to 20004. The outgoing calling numbers will be the following:

X20000 – 2014720000

X20001 – 7322020001

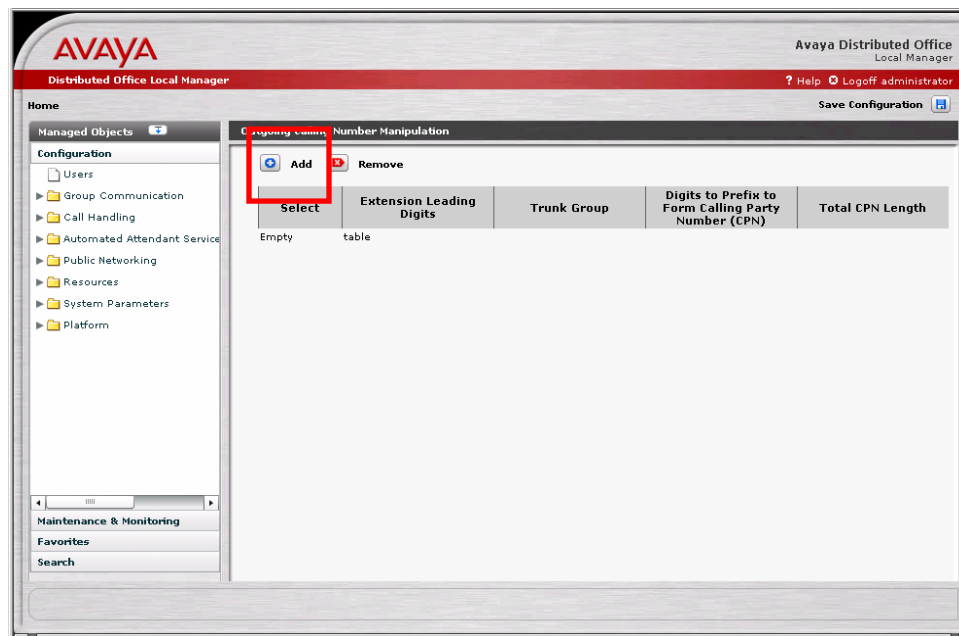
X20002 – 7320220002

X20003 – 2144520003

X20004 – 9999920004

From the left hand **Configuration** menu, expand the **Public Networking** option and click **Outgoing Calling Number**. The **Outgoing Calling Number Manipulation** screen will be displayed.

1. Click **Add** to enter the **Outgoing Calling Number Manipulation** listing screen.



At the **Outgoing Calling Number Manipulation** screen:

1. In the **Extension Leading Digits** field, enter the leading digit(s) of the desired extension or extension range.
2. In the **Trunk Group** field select the trunk group configured in **Section 4.2**.
3. In the **Digits to Prefix to Form Calling Party Number (CPN Prefix)** field enter the leading digits to append to the extension.
4. In the **Total CPN Length** field enter the total length of the digit string.
5. Click **Apply Changes** to save the entries and return to the **Outgoing Calling Number Manipulation** summary screen.

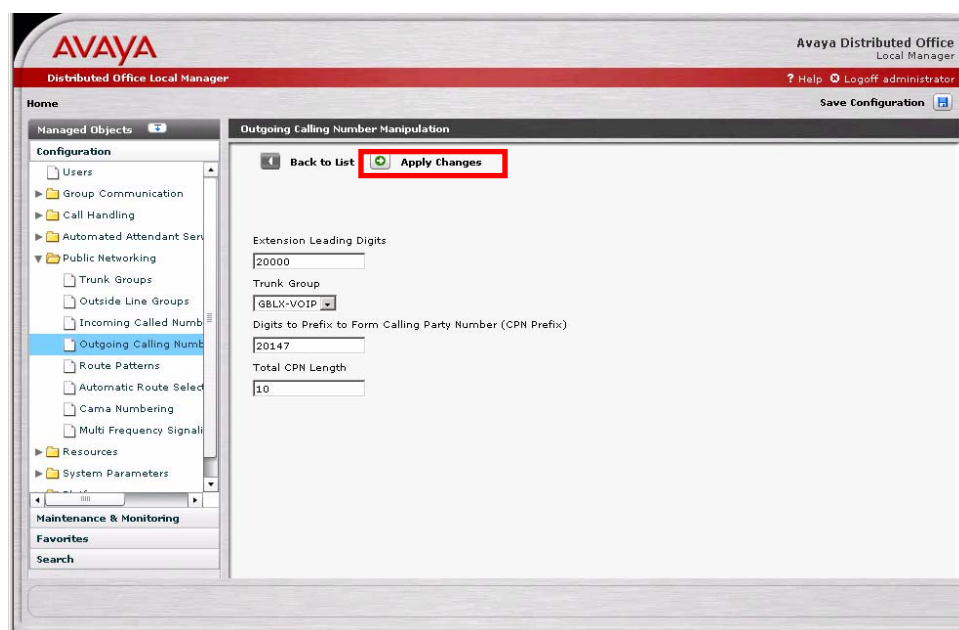
The following values were used for this compliance test:

Extension leading Digits – 20000

Digits to Prefix to Form Calling Party Number (CPN Prefix) – 20147

Total CPN Length - 10

Will produce the following number in the SIP From header – 2014720000



The **Outgoing Calling Number Manipulation** summary screen is displayed reflecting the changes.

The screenshot displays the Avaya Distributed Office Local Manager web interface. The top header features the Avaya logo and the text 'Avaya Distributed Office Local Manager'. Below the header, a red navigation bar contains links for 'Help', 'Logoff administrator', and 'Save Configuration'. The main interface is divided into a left sidebar and a central content area. The sidebar, titled 'Managed Objects', includes a 'Configuration' section with a tree view of system components: Users, Group Communication, Call Handling, Automated Attendant Services, Public Networking (with sub-items: Trunk Groups, Outside Line Groups, Incoming Called Numbers, and 'Outgoing Calling Numbers' which is highlighted), Route Patterns, Automatic Route Selection, Call Numbering, and Multi-Frequency Signaling. Below this are sections for 'Maintenance & Monitoring', 'Favorites', and 'Search'. The central content area is titled 'Outgoing Calling Number Manipulation' and contains an 'Add' button, a 'Remove' button, and a table. The table has five columns: 'Select', 'Extension Leading Digits', 'Trunk Group', 'Digits to Prefix to Form Calling Party Number (CPN)', and 'Total CPN Length'. A single row is visible in the table with the following values: an unchecked checkbox in the 'Select' column, '20000' in the 'Extension Leading Digits' column, 'GBLX-VOIP' in the 'Trunk Group' column, '20147' in the 'Digits to Prefix to Form Calling Party Number (CPN)' column, and '10' in the 'Total CPN Length' column. At the bottom of the interface, a status bar indicates 'Changes applied successfully'.

Select	Extension Leading Digits	Trunk Group	Digits to Prefix to Form Calling Party Number (CPN)	Total CPN Length
<input type="checkbox"/>	20000	GBLX-VOIP	20147	10



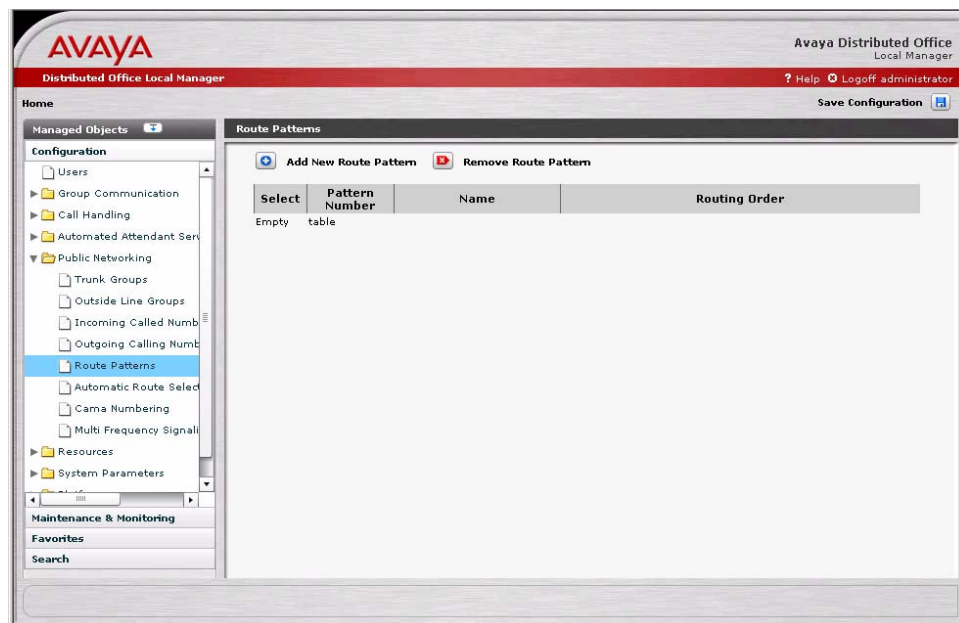
## 4.4. Configuring Call Routing

### 4.4.1. Outbound Calls

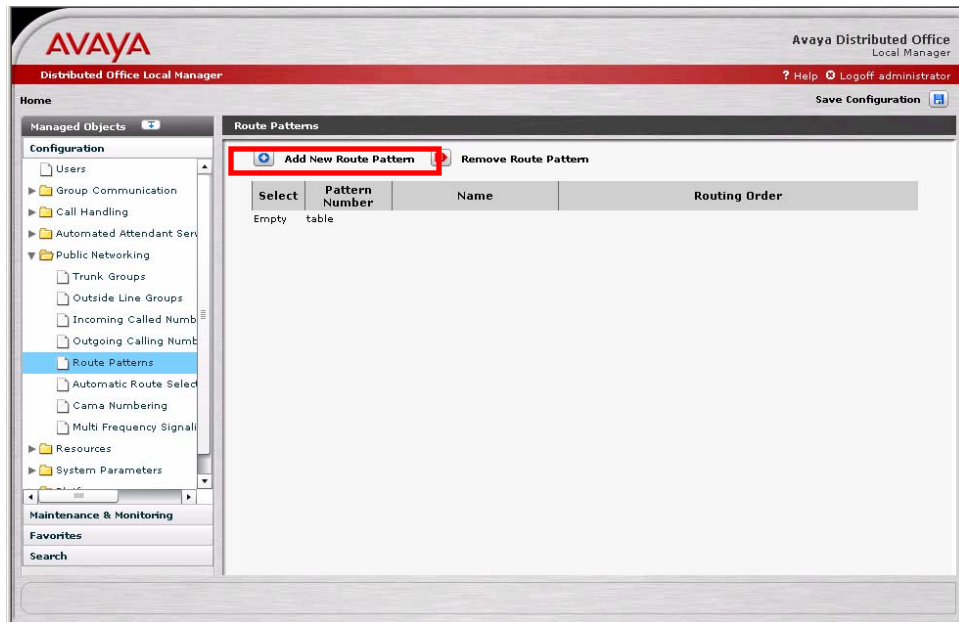
The Automatic Route Selection (ARS) feature is used to route the call out the desired trunk. In this compliance test the desired trunk is the Global Crossing SIP trunk.

First a route pattern must be administered to include the Global Crossing SIP trunk and any necessary digit manipulation to be applied to the outbound call (e.g. deleting digits or appending a “+” for international calls).

From the left hand **Configuration** menu, expand the **Public Networking** option and click **Route Patterns**. The **Route Patterns** summary screen will be displayed.



Click Add New Route Pattern



At the **Edit Route Pattern** screen:

1. Select an available **Pattern Number**.
2. Enter a description for the **Pattern Name**. “Route to GBLX” was used for this compliance test.
3. In the number 1 **Order** field row select “GBLX-VOIP (31)”. By selecting this as the first entry, this defines the Global Crossing SIP trunk as the first preference for a call choosing this route pattern. Subsequent trunk groups added to this route pattern in addition to the Global Crossing SIP trunk will have their trunk facilities utilized only if the number of available Global Crossing SIP trunk channels are exhausted or if the Global Crossing SIP trunks are out of service.
4. Do not enter anything in the # **Digits to Delete** or the **Digits to Insert** fields.<sup>3</sup> The digits dialed by the user, except the ARS feature access code, will be sent in the SIP Request URI to the Global Crossing service.
5. Click **Apply Changes** to save the entries and return to the **Route Patterns** summary screen.

Avaya Distributed Office Local Manager

Home

Managed Objects

Configuration

- Users
- Group Communication
- Call Handling
- Automated Attendant Services
- Public Networking
  - Trunk Groups
  - Outside Line Groups
  - Incoming Called Number
  - Outgoing Calling Number
  - Route Patterns
  - Automatic Route Selection
  - Call Numbering
  - Multi Frequency Signaling
- Resources
- System Parameters

Maintenance & Monitoring

Favorites

Search

Edit Route Pattern 6 (Route to GBLX)

Back to List Apply Changes

Route Pattern Details

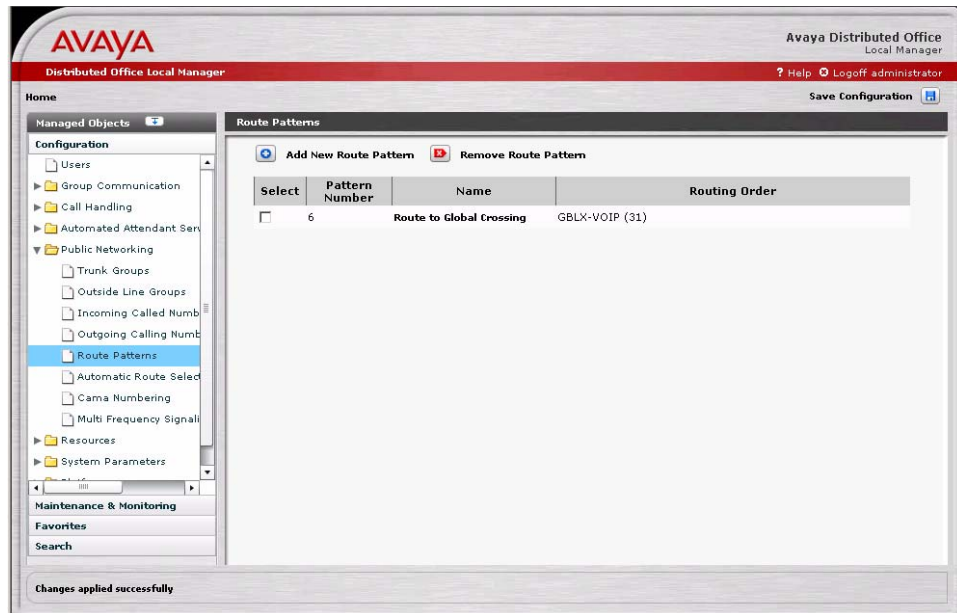
Pattern Number: 6 Pattern Name: Route to GBLX

Routes Selection

Order	Trunk Group	# Digits to Delete	Digits to Insert
1	GBLX-VOIP (31)		
2			
3			
4			
5			
6			
7			
8			
9			
10			
11			

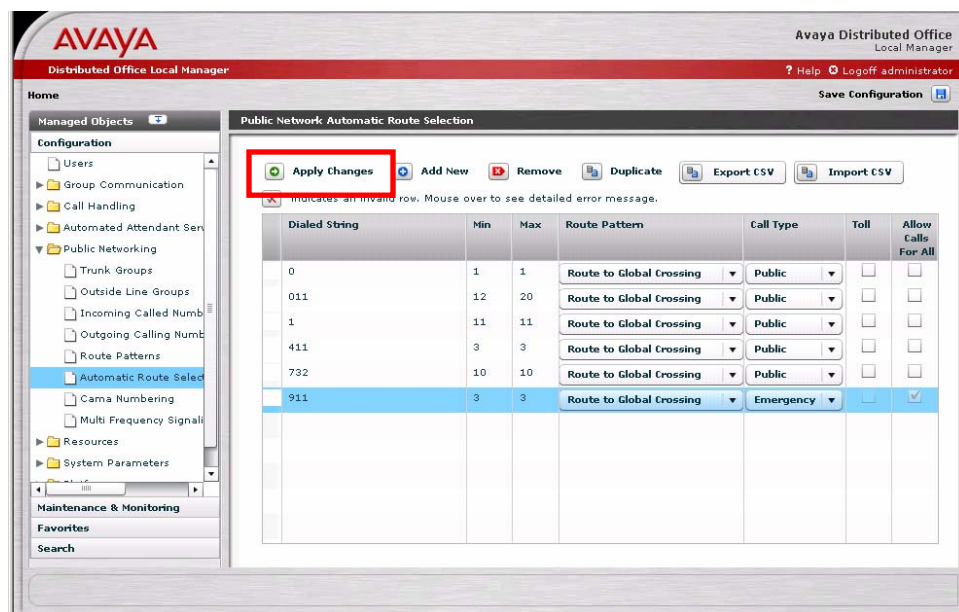
<sup>3</sup> A “+” character can be entered in the **Digits to Insert** field to support E.164 international dialing.

The **Route Pattern** summary screen is displayed.



In the compliance test, calls to 732-xxx-xxxx (where “x” refers to any digit) are routed via the Global Crossing SIP trunk service without requiring toll calling privileges. From the **Configuration** menu in the left hand pane, click **Automatic Route Selection** under the **Public Networking** tab and configure as follows:

1. In the **Dialed String** field enter “732”.
2. In the **Min** field enter “10”.
3. In the **Max** field enter “10”.
4. In the **Route Pattern** field enter “Route to Global Crossing”.
5. In the **Call Type** field select “Public”.
6. To allow extensions with any user privilege to place calls, uncheck **Toll**.<sup>4</sup>
7. Uncheck **Allow Calls for All** to prevent extensions with no privileges from being able to place 732-xxx-xxxx calls if desired.
8. Click **Apply Changes** to save the entries and make them take affect.



<sup>4</sup> The user privilege level is assigned to a station on the station administration form. For further information refer to the Avaya Distributed Office administration guide or online help.

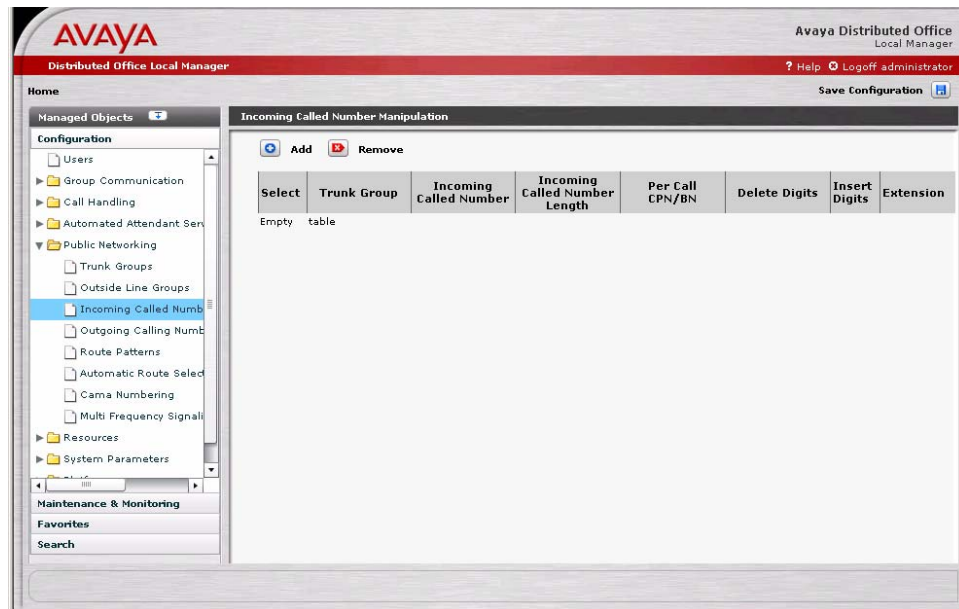
## 4.4.2. Inbound Calls

Below are steps to configure the routing of incoming DID calls to the associated Avaya Communication Manager Branch Edition extensions.

In this compliance test, the incoming PSTN DID numbers used are 602-466-7270, 602-466-7298, 602-466-7299, and 888-246-9143. All numbers are assigned by Global Crossing.

To administer the **Incoming Called Number**:

From the **Configuration** menu in the left hand pane, click **Incoming Called Number**. The **Incoming Called Number Configuration** summary screen will be displayed. Click **Add** to enter the **Incoming Called Number Manipulation** screen.



At the **Add Incoming Called Number Manipulation** screen, enter the following information:

1. In the **Trunk Group** field select “GBLX-VOIP”.
2. In the **Called Number** field enter the DID number provided by Global Crossing which the user desires to ring the associated station. Note Global Crossing does not send the dialed “1” in the incoming digit string.
3. In the **Called Number Length** field enter “10”.
4. Select the **Extensions** radio button.
5. Select the station desired to associate with the DID.<sup>5</sup>
6. Click **Apply Changes** to save the administration and redisplay the **Add Incoming Called Number Manipulation** screen.

The screenshot shows the Avaya Distributed Office Local Manager interface. The left sidebar contains a tree view with categories like Configuration, Maintenance & Monitoring, Favorites, and Search. The 'Incoming Called Number' option is selected. The main panel is titled 'Add Incoming Called Number Manipulation'. It contains several fields: 'Trunk Group' (set to GBLX-VOIP), 'Called Number' (set to 6024667298), 'Called Number Length' (set to 10), and 'Per Call CPN\BN'. There are also radio buttons for 'Digits' and 'Extension' (selected), and a dropdown for '# of Digits to Delete' (set to 20001). A 'Digits to Insert' field is also present. The 'Apply Changes' button is highlighted with a red rectangle.

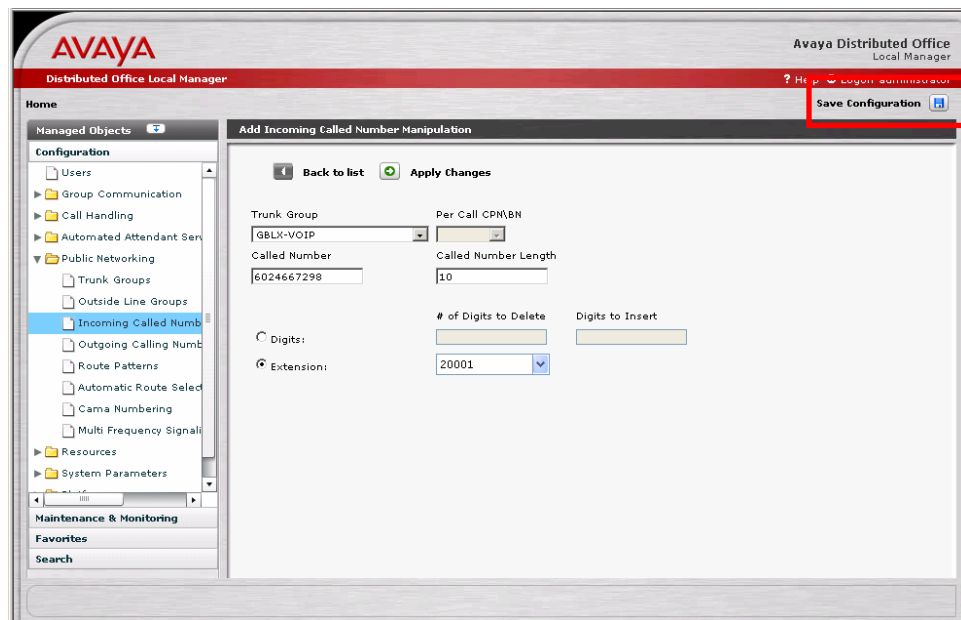
Repeat the steps above in the **Incoming Called Number Manipulation** to associate other DID numbers with the desired station.

<sup>5</sup> An alternative method is to select **Digits**. In the **# of Digits to Delete** field enter the number of leading digits to delete. In the **Digits to Insert** field enter the digits desired to be inserted. For further information refer to the Avaya Distributed Office administration guide or online help.

### 4.4.3. Save Avaya Communication Manager Branch Edition Configuration

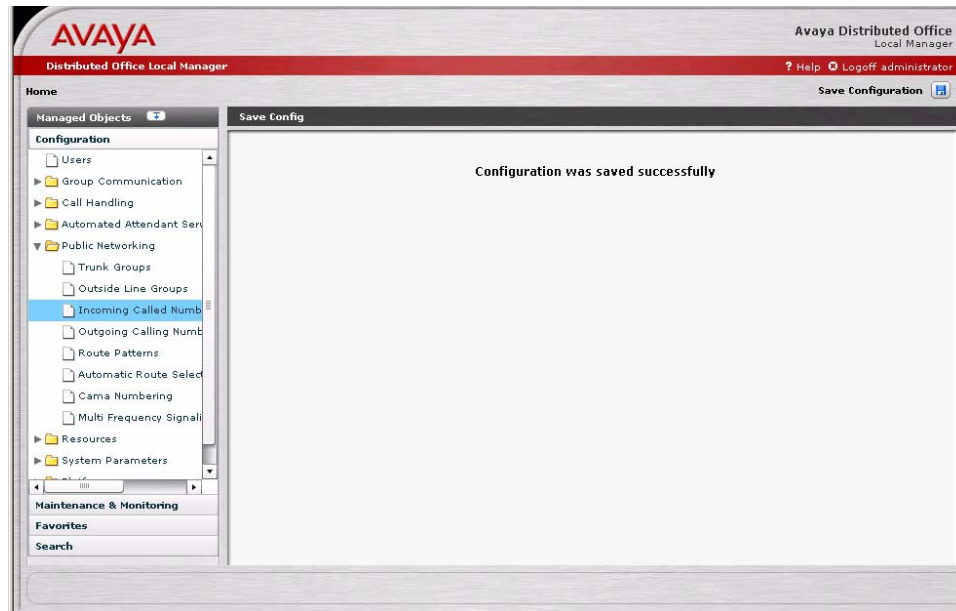
The necessary administration to configure Avaya Communication Manager Branch Edition SIP trunking with Global Crossing SIP trunk service is now complete.

To save the current configuration, click **Save Configuration** which is located in the upper right hand corner of the **Add Incoming Called Number Manipulation** screen.





The configuration data is now saved in non-volatile memory. These configuration changes will now be applied in case Avaya Communication Manager Branch Edition is rebooted or a power failure occurs.<sup>6</sup>



## 5. Interoperability and Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Global Crossing Service and Avaya Communication Manager Branch Edition. This section covers the general test approach and the test results.

### 5.1. General Test Approach

The Avaya Communication Manager Branch Edition i120 was connected using SIP trunking via the internet to the Global Crossing Service in Global Crossing's test lab. Avaya Communication Manager Branch Edition was configured for the generally available service offered by Global Crossing.

Below is a list of features and functionality verified during the SIP trunk interoperability and compliance testing. Unless mentioned to the contrary, all tests were successfully completed.

- Outbound calls to the PSTN
- Inbound calls to Avaya Communication Manager Branch Edition from the PSTN using DID and toll-free service

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<sup>6</sup> The configuration data can be saved on an external storage device such as a PC or USB drive. For further information refer to the Avaya Distributed Office administration guide or online help.

- Calls originating from and terminating at the Avaya 1608 IP H.323 telephone
- Calls originating from and terminating at the Avaya 4610 IP H.323 telephone
- Calls originating from and terminating at the Avaya 9620 IP SIP telephone
- Calls originating from and terminating at the Avaya 6211 analog telephone
- Calls originating from and terminating at the Avaya IP Softphone
- G.711U, G.711A, G.729A, G.729AB codecs for voice calls
- DTMF tone transmission using RFC2833 with successful Voice Mail navigation
- Telephone features such as hold, transfer, conference, and voice mail
- Trunk to trunk call forwarding, transfers, and EC500 feature operation
- Direct Media (also known as “shuffling”) with IP and SIP telephones
- T.38 FAX support with G.711U and G.711A codecs

## 5.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following compatibility issues were observed:

Item	Issue Observed	Discussion/Workaround
Phone on Avaya Communication Manager Branch Edition calling a busy phone in the PSTN.	A call is originated from a phone on Avaya Communication Manager Branch Edition to a phone in the PSTN that is busy. The network may send either a “486 Busy Here” or a “183 Session Progress” to Avaya Communication Manager Branch Edition.	This behavior is transparent to the caller.
Placing a call on hold at the PSTN phone for a call between the PSTN and an Avaya Communication Manager Branch Edition phone.	When the call is placed on hold at the PSTN phone, the network continues to send RTP with a payload of silence (FFFFFFF....) to the Avaya Communication Manager Branch Edition station.	This is how Global Crossing performs hold. This behavior is transparent to the callers.
G.729AB set as the first choice in the codec set list at Global Crossing services and G.729A as the only choice in the codec list of Avaya Communication Manager Branch Edition.	If Avaya Communication Manager Branch Edition has only the G.729A codec set listed in the codec set list, calls will not be able to terminate to an Avaya Communication Manager Branch Edition station.	Coordinate codec set list with the Global Crossing administrator to match the codec set list in Avaya Communication Manager Branch Edition.
Office ID is not displayed when calling a station on Avaya Communication Manager Branch Edition from an EC500 endpoint.	Origination mapping is not working. When a call is made from the EC500 endpoint into the enterprise, the EC500 endpoint’s number is displayed on the destination. Also, the service numbers can not be utilized from the EC500 endpoint.	This will be addressed in a future release of Avaya Communication Manager Branch Edition.
Dialing the operator “0”	Unable to test – Not supported by the Global Crossing Interop Lab	Global Crossing’s production network supports this through a 3 <sup>rd</sup> party vendor.
Dialing “411” for information	Unable to test – Not supported	Global Crossing’s production

	by the Global Crossing Interop Lab	network supports this through a 3 <sup>rd</sup> party vendor.
Emergency Service (Dialing 911)	Unable to test – Not supported by the Global Crossing Interop Lab	Global Crossing's production network supports this through a 3 <sup>rd</sup> party vendor.

### 5.3. Verifying the Configuration

This section provides steps for the user to perform to verify proper operation of the SIP trunking configuration as described in the Application Notes.

- **Incoming Calls** – Verify that calls placed from a PSTN telephone to the assigned DID numbers are properly routed via the SIP trunk group to the expected extension. Verify the voice path exists in both directions, display information is as expected, calls remain active and stable for several minutes, and the calls disconnect properly.
- **Outbound Calls** - Verify that calls placed to a PSTN telephone from the Avaya Communication Manager Branch Edition phone are properly routed via the SIP trunk group administered in the route pattern. Verify the voice path exists in both directions, display information is as expected, calls remain active and stable for several minutes, and the calls disconnect properly.
- **Inbound DTMF Digit Navigation** – Verify inbound DID calls can properly navigate the Avaya Communication Manager Branch Edition automated attendant feature.
- **Outbound DTMF Digit Navigation** – Verify outbound calls can properly navigate a voice mail system reached via a PSTN number.

### 5.4. Troubleshooting Tools

Avaya Communication Manager Branch Edition provides several troubleshooting tools that can be helpful in diagnosing SIP trunking issues.

The **Maintenance & Monitoring/Network Diagnostics** menu provides the administrator the ability to perform the following operations:

- **Ping** – used to test whether a particular host is reachable over the IP network.
- **Traceroute** – maps the path the data packets take between Avaya Communication Manager Branch Edition and other endpoints outside of the local network.
- **ARP** – shows the content of the ARP cache.

The **Maintenance & Monitoring/Telephony/Trunk Groups** menu provides:

- **Test Selected** – runs diagnostic tests verifying the operation of the SIP signaling channel for the selected SIP trunk group.

- **Trace Selected** – Provides a diagnostic trace of the call processing activities using the selected SIP trunk group.
- **Get Hourly Statistics** – Shows the hourly traffic statistics for the selected trunk group.

The **Maintenance & Monitoring/Telephony/SIP Traces** menu permits real time tracing of the SIP signaling to be displayed, captured, and download.

The **Configuration/Platform/Ethernet Switch** menu provides access to the **Ethernet Switch System Parameters** screen. The **Mirror Port** tab on this screen provides the ability to designate a specific Ethernet switch port to monitor (such as the connection used to reach the Global Crossing network). This mirror port may be used with a SIP protocol analyzer such as Wireshark or Ethereal to monitor the SIP and RTP communication between the Global Crossing network and Avaya Communication Manager Branch Edition. This can be invaluable in supporting advanced troubleshooting.

## 6. Conclusions

These Application Notes describe the steps for configuring SIP trunking between an Avaya Communication Manager Branch Edition and Global Crossing SIP services as described in section 1.

The configuration shown in these Application Notes is representative of a typical customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

## 7. References

Avaya Communication Manager Branch Edition product documentation is available at

<http://support.avaya.com>.

- [1] Avaya Distributed Office Documentation Map, 03-602021
- [2] Overview of Avaya Distributed Office, 03-602024
- [3] Avaya Distributed Office i120 Installation Quick Start, 03-602289
- [4] Avaya Distributed Office i40 Installation Quick Start, 03-602288
- [5] Feature Description for Avaya Distributed Office, 03-602027
- [6] Avaya Application Solutions: IP Telephony Deployment Guide, 555-245-600
- [7] 4600 Series IP Telephone LAN Administrator Guide, 555-233-507
- [8] Avaya one-X™ Deskphone SIP for 9600 Series IP Telephones Administrator Guide, 16-601944

Global Crossing SIP service Descriptions

- [9] *Global Crossing VoIP On-Net Plus™*  
[http://www.globalcrossing.com/enterprise/voip/voip\\_onnet\\_plus.aspx](http://www.globalcrossing.com/enterprise/voip/voip_onnet_plus.aspx)
- [10] *Global Crossing VoIP Outbound™*  
[http://www.globalcrossing.com/enterprise/voip/voip\\_outbound.aspx](http://www.globalcrossing.com/enterprise/voip/voip_outbound.aspx)
- [11] *Global Crossing VoIP Local Service™*  
[http://www.globalcrossing.com/enterprise/voip/voip\\_local.aspx](http://www.globalcrossing.com/enterprise/voip/voip_local.aspx)
- [12] *Global Crossing VoIP Toll Free Service™*

[http://www.globalcrossing.com/enterprise/voip/voip\\_tollfree.aspx](http://www.globalcrossing.com/enterprise/voip/voip_tollfree.aspx)

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: <http://www.rfc-editor.org/rfcsearch.html>

[13] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard

[14] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

## 8. Appendix A: Sample SIP Invite Messages

This section displays the format of typical SIP INVITE messages sent between Global Crossing and Avaya Communication Manager Branch Edition. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

### 8.1. Sample SIP INVITE Message from Global Crossing services to Avaya Communication Manager Branch Edition:

No.	Time	Source	Destination	Protocol Info
2	2.421616	192.168.20.35	10.1.1.20	SIP/SDP

Request: INVITE sip:6024667298;npdi=yes@10.1.1.20:5060, with session description

Frame 2 (1086 bytes on wire, 1086 bytes captured)  
Ethernet II, Src: Netscree\_d6:fe:80 (00:10:db:d6:fe:80), Dst: Avaya\_f2:33:27 (00:04:0d:f2:33:27)  
Internet Protocol, Src: 192.168.20.35 (192.168.20.35), Dst: 10.1.1.20 (10.1.1.20)  
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)  
Session Initiation Protocol  
Request-Line: INVITE sip:6024667298;npdi=yes@10.1.1.20:5060 SIP/2.0  
Message Header  
Via: SIP/2.0/UDP 192.168.20.35:5060;branch=z9hG4bK02Bcea70b3f8d29e08b  
From: <sip:7328521640@192.168.20.35>;tag=gK024937ab  
To: <sip:6024667298@10.1.1.20:5060>  
Call-ID: 503508461\_8978@192.168.20.35  
CSeq: 27464 INVITE  
Max-Forwards: 70  
Allow: INVITE, ACK, CANCEL, BYE, REGISTER, INFO, SUBSCRIBE, NOTIFY, PRACK, UPDATE, OPTIONS  
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed  
Contact: <sip:7328521640@192.168.20.35:5060>  
Remote-Party-ID: <sip:7328521640@192.168.20.35:5060>;privacy=off  
Supported: timer, 100rel  
Session-Expires: 64800  
Min-SE: 90  
Content-Disposition: session; handling=required  
Content-Type: application/sdp  
Content-Length: 285

#### Message Body

##### Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): Sonus\_UAC 3507 10043 IN IP4 192.168.20.35  
Session Name (s): SIP Media Capabilities  
Connection Information (c): IN IP4 192.168.20.36  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 24108 RTP/AVP 18 0 8 100  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): rtpmap:0 PCMU/8000  
Media Attribute (a): rtpmap:8 PCMA/8000  
Media Attribute (a): rtpmap:100 telephone-event/8000  
Media Attribute (a): fmp:100 0-15  
Media Attribute (a): sendrecv  
Media Attribute (a): maxptime:20

## 8.2. Sample 200 OK Message from Avaya Communication Manager Branch Edition to Global Crossing services:

No.	Time	Source	Destination	Protocol	Info
11	2.684129	10.1.1.20	192.168.20.35	SIP	

Status: 200 OK

Frame 11 (396 bytes on wire, 396 bytes captured)

Ethernet II, Src: Avaya\_f2:33:27 (00:04:0d:f2:33:27), Dst: Netscree\_d6:fe:80 (00:10:db:d6:fe:80)

Internet Protocol, Src: 10.1.1.20 (10.1.1.20), Dst: 192.168.20.35 (192.168.20.35)

User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)

#### Session Initiation Protocol

Status-Line: SIP/2.0 200 OK

#### Message Header

From: <sip:7328521640@192.168.20.35>;tag=gK024937ab

To: <sip:6024667298@10.1.1.20:5060>;tag=80e8234fc78fdd160948e0a2ee00

Call-ID: 503508461\_8978@192.168.20.35

CSeq: 27465 PRACK

Via: SIP/2.0/UDP

192.168.20.35:5060;psrrposn=1;received=192.168.20.35;branch=z9hG4bK02Bcea8102ea7e4b757

Server: Avaya CM/R013w.01.2.023.0

Content-Length: 0

## 8.3. Sample 200 OK with SDP Message from Avaya Communication Manager Branch Edition to Global Crossing services:

No.	Time	Source	Destination	Protocol	Info
82	4.222846	10.1.1.20	192.168.20.35	SIP/SDP	

Status: 200 OK, with session description

Frame 82 (956 bytes on wire, 956 bytes captured)

```

Ethernet II, Src: Avaya_f2:33:27 (00:04:0d:f2:33:27), Dst: Netscree_d6:fe:80
(00:10:db:d6:fe:80)
Internet Protocol, Src: 10.1.1.20 (10.1.1.20), Dst: 192.168.20.35
(192.168.20.35)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
  Status-Line: SIP/2.0 200 OK
  Message Header
    From: <sip:7328521640@192.168.20.35>;tag=gK024937ab
    To: <sip:6024667298@10.1.1.20:5060>;tag=80e8234fc78fdd160948e0a2ee00
    Call-ID: 503508461_8978@192.168.20.35
    CSeq: 27464 INVITE
    Via: SIP/2.0/UDP
192.168.20.35:5060;received=192.168.20.35;branch=z9hG4bK02Bcea70b3f8d29e08b
  Record-Route: <sip:10.1.1.20:6002;lr;transport=tls>
  Record-Route: <sip:10.1.1.20:5060;lr>
  Contact: "Tony" <sip:7322020001@10.1.1.20:6002;transport=tls>
  P-Asserted-Identity: "Tony" <sip:7322020001@example.com:6002>
  Supported: 100rel,timer,replaces,join,histinfo
  Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
  Server: Avaya CM/R013w.01.2.023.0
  Session-Expires: 64800;refresher=uas
  Require: timer
  Content-Type: application/sdp
  Content-Length: 150
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0
      Owner/Creator, Session Id (o): - 1 2 IN IP4 10.1.1.20
      Session Name (s): -
      Connection Information (c): IN IP4 10.1.1.21
      Time Description, active time (t): 0 0
      Media Description, name and address (m): audio 2414 RTP/AVP 18 100
      Media Attribute (a): rtpmap:18 G729/8000
      Media Attribute (a): rtpmap:100 telephone-event/8000

```

## 8.4. Sample SIP INVITE Message from Avaya Communication Manager Branch Edition to Global Crossing services:

No.	Time	Source	Destination	Protocol Info
2	2.618352	10.1.1.20	208.49.73.35	SIP/SDP

Request: INVITE sip:7328521640@208.49.73.35;transport=udp, with session description

```

Frame 2 (1239 bytes on wire, 1239 bytes captured)
Ethernet II, Src: Avaya_f2:33:27 (00:04:0d:f2:33:27), Dst: Netscree_d6:fe:80
(00:10:db:d6:fe:80)
Internet Protocol, Src: 10.1.1.20 (10.1.1.20), Dst: 208.49.73.35
(208.49.73.35)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
  Request-Line: INVITE sip:7328521640@208.49.73.35;transport=udp SIP/2.0

```



Message Header  
 Call-ID: 80922ccc68fdd159948e0a2ee00  
 CSeq: 1 INVITE  
 From: "Tony"  
 <sip:7322020001@example.com:6002>;tag=80922ccc68fdd158948e0a2ee00  
 Record-Route:  
 <sip:10.1.1.20:5060;lr>,<sip:10.1.1.20:6002;lr;transport=tls>  
 To: "7328521640" <sip:7328521640@noGBLXDomain.com>  
 Via: SIP/2.0/UDP  
 10.1.1.20:5060;branch=z9hG4bK838383030303939393694.0,SIP/2.0/TLS  
 10.1.1.20:6002;psrrposn=2;received=10.1.1.20;branch=z9hG4bK80922ccc68fdd15a948e0a2ee00  
 Content-Length: 174  
 Content-Type: application/sdp  
 Contact: "Tony" <sip:7322020001@10.1.1.20:6002;transport=tls>  
 Max-Forwards: 69  
 User-Agent: Avaya CM/R013w.01.2.023.0  
 Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS  
 History-Info: <sip:7328521640@noGBLXDomain.com>;index=1  
 History-Info: "7328521640" <sip:7328521640@noGBLXDomain.com>;index=1.1  
 Supported: 100rel,timer,replaces,join,histinfo  
 Min-SE: 3600  
 Session-Expires: 3600;refresher=uac  
 P-Asserted-Identity: "Tony" <sip:7322020001@example.com:6002>  
 Message Body  
 Session Description Protocol  
 Session Description Protocol Version (v): 0  
 Owner/Creator, Session Id (o): - 1 1 IN IP4 10.1.1.20  
 Session Name (s): -  
 Connection Information (c): IN IP4 10.1.1.21  
 Time Description, active time (t): 0 0  
 Media Description, name and address (m): audio 2412 RTP/AVP 0 18 120  
 Media Attribute (a): rtpmap:0 PCMU/8000  
 Media Attribute (a): rtpmap:18 G729/8000  
 Media Attribute (a): rtpmap:120 telephone-event/8000

## 8.5. Sample 200 OK Message from Global Crossing services to Avaya Communication Manager Branch Edition:

No.	Time	Source	Destination	Protocol Info
23	5.819527	192.168.20.35	10.1.1.20	SIP

Status: 200 OK

Frame 23 (494 bytes on wire, 494 bytes captured)  
 Ethernet II, Src: Netscree\_d6:fe:80 (00:10:db:d6:fe:80), Dst: Avaya\_f2:33:27 (00:04:0d:f2:33:27)  
 Internet Protocol, Src: 192.168.20.35 (192.168.20.35), Dst: 10.1.1.20 (10.1.1.20)  
 User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)  
 Session Initiation Protocol  
 Status-Line: SIP/2.0 200 OK  
 Message Header  
 Via: SIP/2.0/UDP 10.1.1.20:5060;branch=z9hG4bK038383560303369393888.0

```

Via: SIP/2.0/TLS
10.1.1.20:6002;received=10.1.1.20;branch=z9hG4bK0ec64cec68fdd15b948e0a2ee00;p
srrposn=1
  From: "Tony"
<sip:7322020001@example.com:6002>;tag=80922ccc68fdd158948e0a2ee00
  To: "7328521640" <sip:7328521640@noGBLXDomain.com>;tag=gK02c935f6
  Call-ID: 80922ccc68fdd159948e0a2ee00
  CSeq: 2 PRACK
  Record-Route: <sip:10.1.1.20:5060;lr>
  Content-Length: 0

```

## 8.6. Sample 200 OK with SDP Message from Global Crossing services to Avaya Communication Manager Branch Edition:

No.	Time	Source	Destination	Protocol Info
192	7.926146	192.168.20.35	10.1.1.20	SIP/SDP

Status: 200 OK, with session description

```

Frame 192 (1178 bytes on wire, 1178 bytes captured)
Ethernet II, Src: Netscree_d6:fe:80 (00:10:db:d6:fe:80), Dst: Avaya_f2:33:27
(00:04:0d:f2:33:27)
Internet Protocol, Src: 192.168.20.35 (192.168.20.35), Dst: 10.1.1.20
(10.1.1.20)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
  Status-Line: SIP/2.0 200 OK
  Message Header
    Via: SIP/2.0/UDP 10.1.1.20:5060;branch=z9hG4bK838383030303939393694.0
    Via: SIP/2.0/TLS
10.1.1.20:6002;received=10.1.1.20;branch=z9hG4bK80922ccc68fdd15a948e0a2ee00;p
srrposn=2
  From: "Tony"
<sip:7322020001@example.com:6002>;tag=80922ccc68fdd158948e0a2ee00
  To: "7328521640" <sip:7328521640@noGBLXDomain.com>;tag=gK02c935f6
  Call-ID: 80922ccc68fdd159948e0a2ee00
  CSeq: 1 INVITE
  Record-Route: <sip:10.1.1.20:5060;lr>
  Record-Route: <sip:10.1.1.20:6002;transport=tls;lr>
  Accept: application/sdp,application/isup,application/dtmf,
application/dtmf-relay, multipart/mixed
  Contact: <sip:7328521640@192.168.20.35:5060>
  Allow: INVITE, ACK, CANCEL, BYE, REGISTER, INFO, SUBSCRIBE, NOTIFY,
PRACK, UPDATE, OPTIONS
  Require: timer
  Supported: timer
  Session-Expires: 3600;refresher=uac
  Content-Disposition: session; handling=required
  Content-Type: application/sdp
  Content-Length: 238
  Message Body
    Session Description Protocol
      Session Description Protocol Version (v): 0

```

Owner/Creator, Session Id (o): Sonus\_UAC 24069 26839 IN IP4  
192.168.20.35  
Session Name (s): SIP Media Capabilities  
Connection Information (c): IN IP4 192.168.20.36  
Time Description, active time (t): 0 0  
Media Description, name and address (m): audio 24104 RTP/AVP 18 120  
Media Attribute (a): rtpmap:18 G729/8000  
Media Attribute (a): rtpmap:120 telephone-event/8000  
Media Attribute (a): fmp:120 0-15  
Media Attribute (a): sendrecv  
Media Attribute (a): maxptime:20

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