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Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between AT&T IP Flexible Reach and IP Toll Free Services with Avaya Distributed Office (Release 1.2) – Issue 1.2

Abstract

These Application Notes describe the steps for configuring SIP trunking between the AT&T IP Flexible Reach and IP Toll Free services and an Avaya Distributed Office (Release 1.2) using various Avaya telephony endpoints.

AT&T IP Flexible Reach and IP Toll Free are managed Voice over IP communication solutions using SIP trunks to provide inbound and outbound local, long distance, international and toll-free services for U.S. sites.

AT&T 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 between the Avaya Solution and Interoperability Test Lab and AT&T's Virtual Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the AT&T IP Flexible Reach and IP Toll Free services and an Avaya Distributed Office (Release 1.2) using various Avaya telephony endpoints.

SIP (Session Initiation Protocol) is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [11] 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 Distributed Office and the network services offered by AT&T.

The AT&T IP Flexible Reach and IP Toll Free services are managed Voice over IP communication services using SIP trunks to provide inbound and outbound local, long distance, international and toll-free services for U.S. sites. In the remainder of this document, "AT&T services" will be used to generically refer to both AT&T IP Flexible Reach and AT&T IP Toll Free services.

AT&T 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 between the Avaya Solution and Interoperability Test Lab and AT&T's Virtual Interoperability Test Lab.

1.1. Typical Enterprise Customer Location

Figure 1 illustrates a typical customer location using an Avaya Distributed Office with SIP trunking to AT&T. This configuration includes:
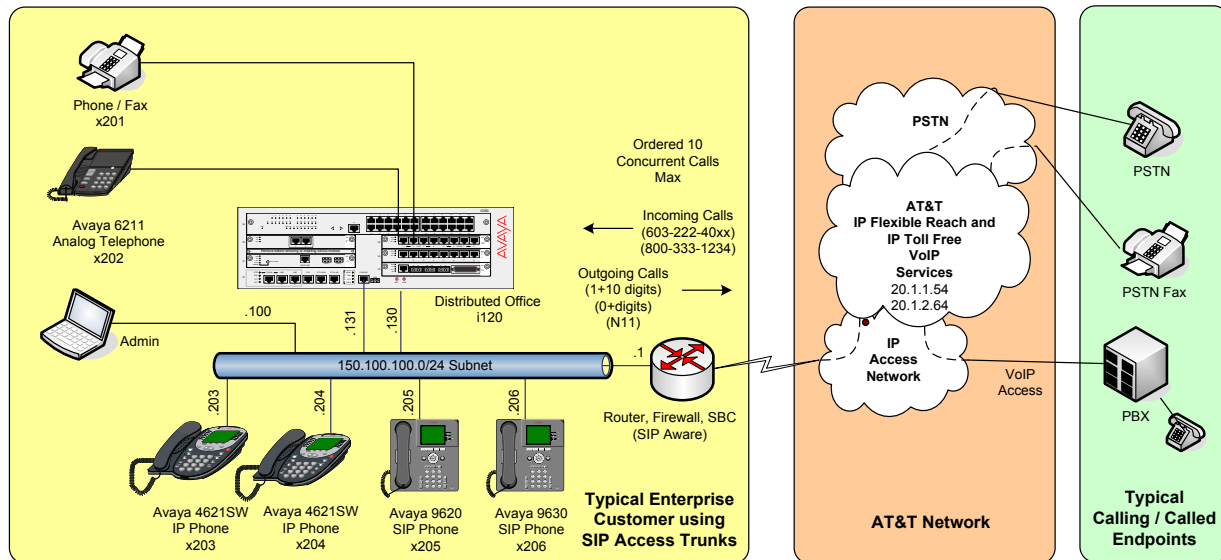- Avaya Distributed Office i120 providing the communication services for this customer location.
- Various Avaya telephones and other endpoints.
- IP routing and data network infrastructure to support IP connectivity between the enterprise location and the AT&T services.

For simplicity, aspects that may exist in customer configurations but are beyond the scope of these Application Notes are not addressed. Specifically,
- The initial installation and administration of the Avaya Distributed Office to provide basic telephony services is not addressed. The SIP trunking configuration described within assumes a previously configured system capable of extension to extension calling.
- The concepts presented in these Application Notes apply to both Avaya Distributed Office i120 and (the smaller) i40 configuration. However, the i40 is not specifically discussed.
- The use of analog or digital PSTN trunks in addition to the SIP trunking is not discussed.
- The configuration of Avaya 9600 and 4600 series telephones.
- IP Network Address Translation (NAT), firewalls, Application Layer Gateway (ALG), and/or Session Border Controller (SBC) devices may exist between the AT&T services

and the Avaya Distributed Office within a customer's communications infrastructure. These devices are neither shown nor addressed within these Application Notes. These devices generally must be SIP-aware and configured properly for SIP trunking to function properly. When configured correctly, they are transparent to the Avaya communications infrastructure.

SIP Trunking with AT&T IP Flexible Reach and IP Toll Free Services

Figure 1 – Typical SIP Trunking Configuration

Figure 2 illustrates the Network Connection information for the Avaya Distributed Office i120.
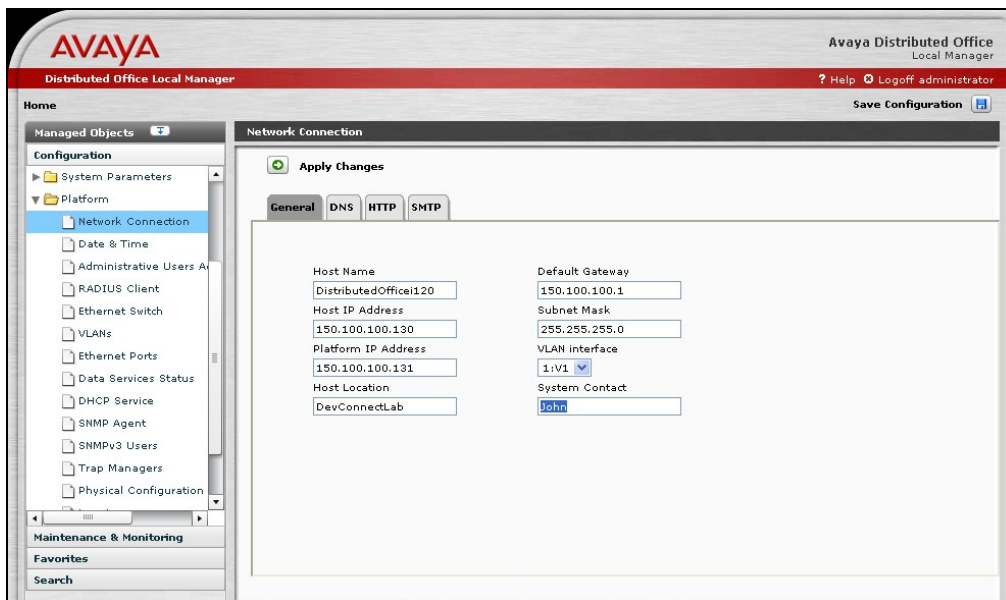
Figure 2 - Avaya Distributed Office Network Connection Assignments

Note the following information is provided to AT&T during the ordering process for the AT&T IP Flexible Reach and IP Toll Free services. The IP addresses mentioned below are examples only.

- Avaya Distributed Office **Host IP Address** of "150.100.100.130". This is the only address used for all SIP signaling between the Avaya Distributed Office and the AT&T Border Elements.

- The IP addresses of the Real Time Protocol (RTP) media endpoints will be from various addresses within the "150.100.100.0/24" subnet in these Application Notes.

  Specifically, the Avaya Distributed Office **Platform IP Address** of "150.100.100.131" is used by all analog telephones or fax machines, as well as during call establishment for all IP and SIP endpoints. After call establishment, direct media (a.k.a., shuffling) may occur leading to a transfer of the RTP path to other IP addresses assigned to the IP or SIP endpoints.

  It is a mandatory requirement that IP routing exist between any IP or SIP endpoints and the AT&T Border Elements whenever using direct media.

1.2. AT&T Services Configuration Information

These Application Notes provide **an illustrative example** of how the Avaya Distributed Office SIP trunking solution is configured with the AT&T IP Flexible Reach and IP Toll Free services.

The specific values provided below are illustrative only and must not be used for customer configurations. *Each customer must obtain the specific values for their configuration from AT&T during service provisioning of their AT&T IP Flexible Reach or IP Toll Free services.*

AT&T Provisioning Information	Illustrative Values in these Application Notes
AT&T Border Element IP Address(es)	20.1.1.54 20.1.2.64
G.729B, G.711MU Codecs 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 (DID) Numbers	1-603-222-40xx (where x is any digit)
DID Digits Passed in SIP Request URI	60322240xx
DID Digits Passed in SIP To Header	Same as SIP Request URI
Incoming Toll-Free Number	800-333-1234
Incoming Toll-Free Digits Passed in SIP Request URI	0000000034
Incoming Toll-Free Digits Passed in SIP To Header	0004152100034

Table 1 – Illustrative AT&T Network Provisioning Information

2. Equipment and Software Validated

The following equipment and software was used during the DevConnect compliance testing with the AT&T services.

Component	Version
Avaya	
Avaya Distributed Office i120	Release 1.2 (1.2.0_24.05)
Avaya 1608 IP (H.323) Telephone	Release 1.0491
Avaya 4621SW IP (H.323) Telephone	Release 2.9 SP1 (2.9.1)
Avaya 9620 one-X^{TM} Deskphone SIP Telephone	Release 2.0.4
Avaya 6211 Analog Telephone	n/a
MultiTech Fax Modem	Model MT5634ZBA
Venta Fax & Voice Fax Application	Release 2.8
AT&T	
IP Flexible Reach Service	Network Version VNI 11
IP Toll Free Service	Network Version VNI 11

Table 2 – Equipment and Version

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3. Configure Avaya Distributed Office

The Avaya Distributed Office i120 was installed and configured for basic station to station calling prior to the beginning of the configuration shown in these Application Notes. The installation and basic configuration details are outside of the scope of the SIP trunking application and not included here.

3.1. Login to Avaya Distributed Office

Using a web browser, access the Avaya Distributed Office Local Manager by entering "http://<ip-addr>/" where "<ip-addr>" is the **Host IP Address** of the Avaya Distributed Office. In these Application Notes, "http://150.100.100.130" is used.
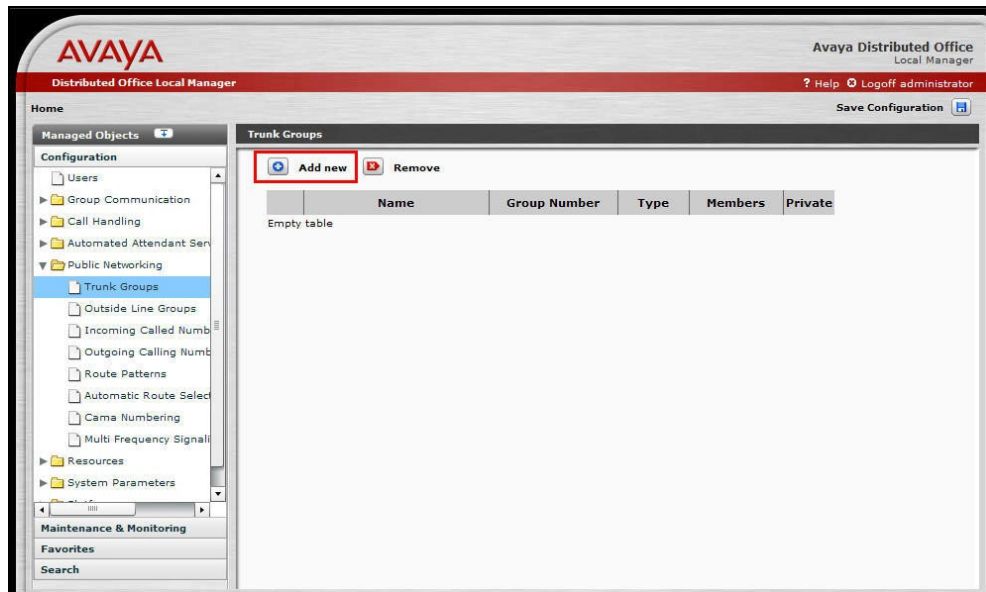
Log in with the appropriate credentials. The Home Local Manager screen is shown.
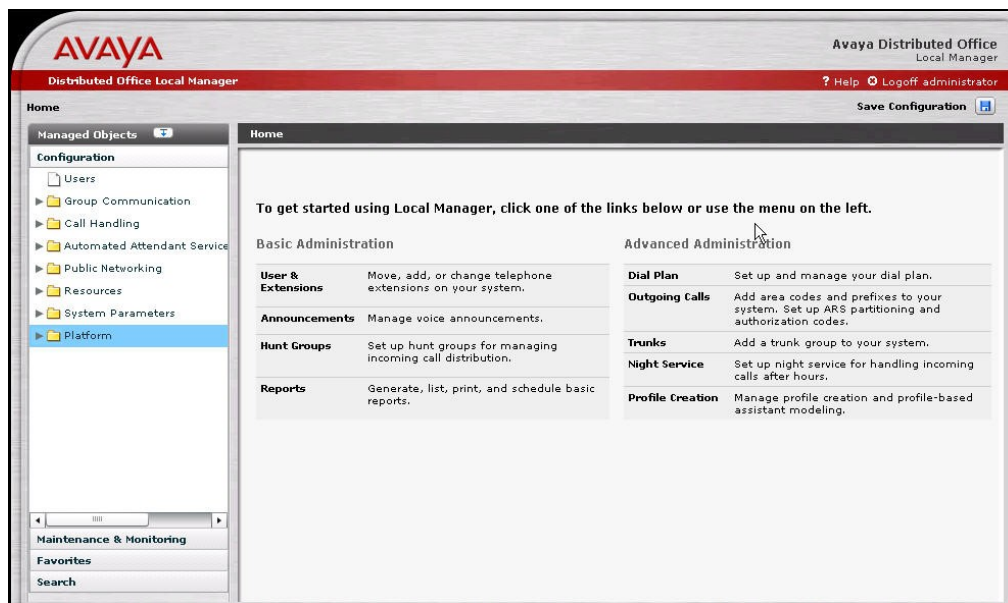
## 3.2. Add a SIP Trunk Group to the AT&T Services

From the left hand **Configuration** menu, expand the **Public Networking** option and select **Trunk Groups**. The **Trunk Groups** screen will be displayed.

Select **Add New** to display the **Add Trunk Group** screen.

On the Add Trunk Group screen:
- Set the **Trunk Type** to "SIP".
- Enter a short text description of the trunk group (e.g., "ATT-SIP") in the **Native Name** field.
- The **Name (ASCII)** field will default to the Native Name field. Modify the Name if necessary to provide a corresponding ASCII version.
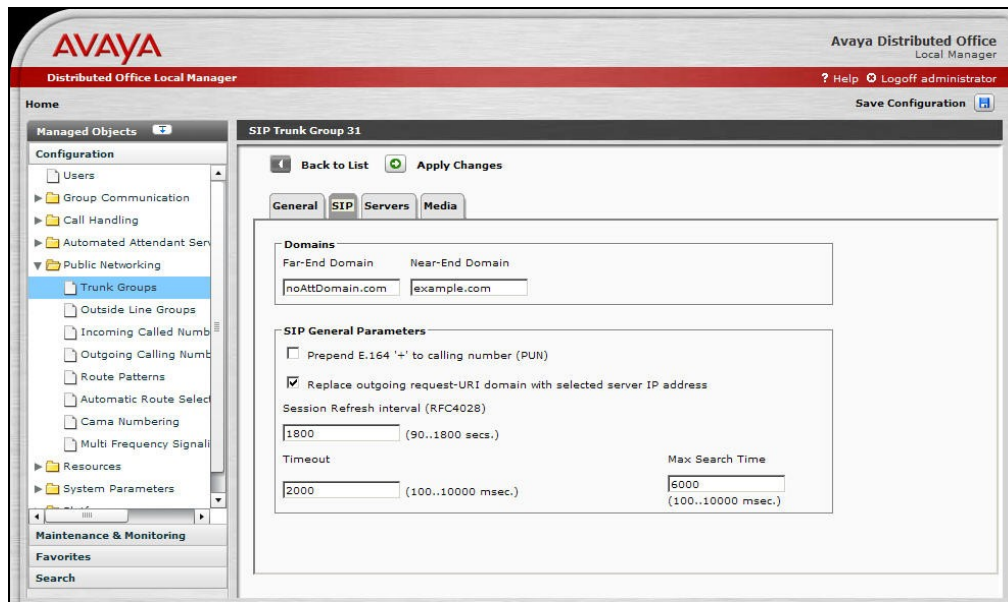- Press the **Continue** button.
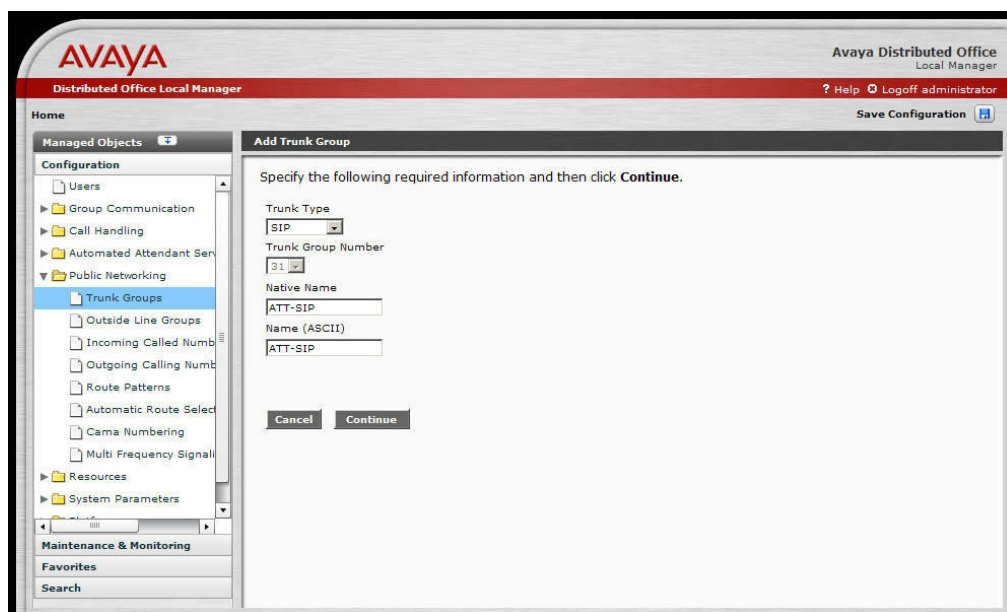
The Add SIP Trunk Group General Tab screen is shown.
- Select "two-way" as the **Direction** to support both incoming and outgoing calling on this trunk group.
- Press the **SIP** tab to advance to the next screen
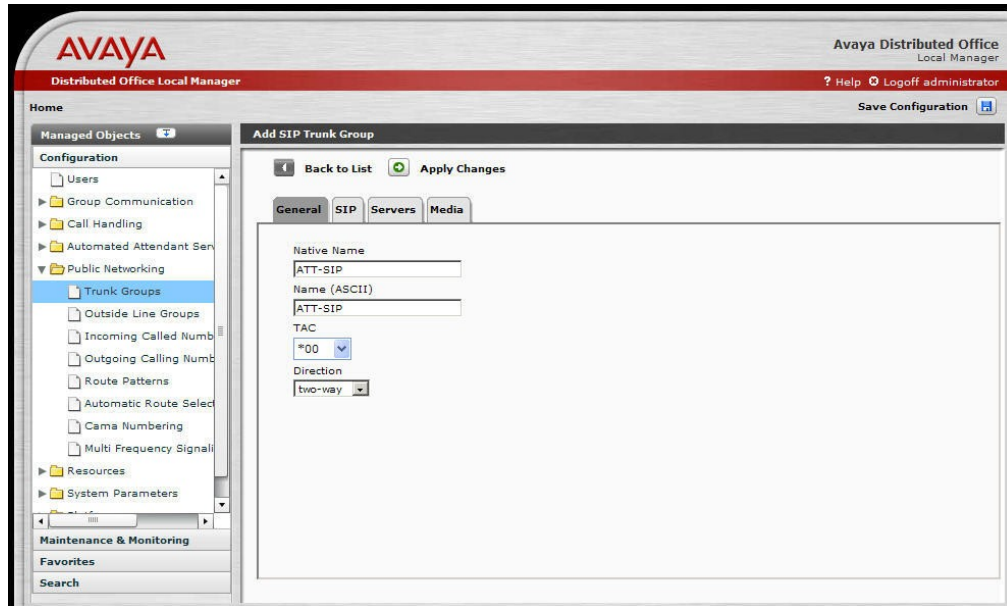
On the **SIP** tab:
- Enter a **Far-End Domain** value for the AT&T services. Since AT&T does not use domain addressing, the placeholder "noAttDomain.com" was entered in the form. It is not necessary that this domain be resolvable in DNS.
- Enter the customer's SIP domain for the Distributed Office in the **Near-End Domain** field. In these Application Notes, "example.com" was used. It is not necessary that this domain be resolvable for the AT&T SIP trunking.
- Check the **Replace outgoing request-URI domain with selected server IP address** box.
- Enter "1800" in the **Session Refresh Interval** field.
- The defaults shown for the **Timeout** and **Max Search Time** are used.
- Press the **Servers** tab to advance to the next screen.

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On the **Servers** tab:

- Enter the IP address of the primary AT&T Border Element <u>provided by AT&T</u> in the **Address** field. In this Application Note, "20.1.1.54" is used as noted in Section 1.2. It is not necessary to specify the port since the UDP default "5060" is used.
- Select "UDP" for the **Transport** field value.
- Repeat the **Address** entry and "UDP" **Transport** selection for the secondary AT&T Border Element Address provided by AT&T.
- The default **Priority** field settings shown are used.
- Press the **Media** tab to advance to the next screen.
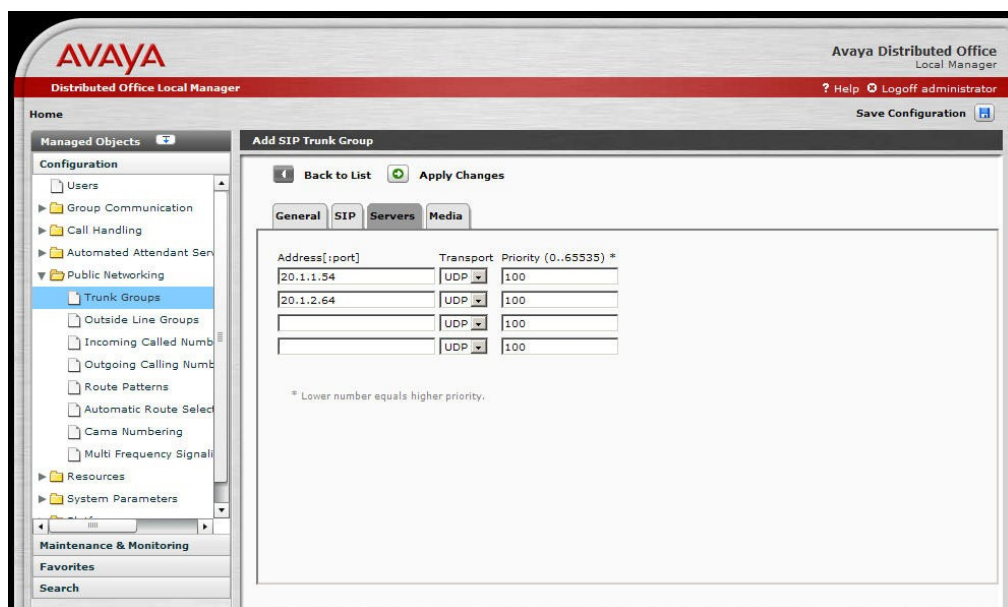
On the **Media** tab:

- Set the **Telephone Events RTP Payload Type** to match the value used by the Avaya 96xx series SIP phones. In these Application Notes "120" was used matching the Avaya 96xx series SIP phone default.[1]
  Note: For inbound trunk calls from an AT&T service, if the ability to send DTMF from 96xx series SIP phones back to the network is NOT required, then the above is sufficient. If that capability is required, then the RTP telephone events payload type configured on the Avaya Distributed Office and the Avaya 96xx series SIP phones must match the RTP telephone events payload type offered by the specific AT&T service. However, if the Avaya Distributed Office receives inbound trunk calls from multiple AT&T services, i.e., both IP Flexible Reach and IP Toll Free services, then it is not possible to simultaneously match the RTP telephone events payload types of those AT&T services. In that scenario, disabling the Direct Media option (see below) on Avaya Distributed Office will allow the 96xx series SIP phones to send DTMF back to the network on inbound calls from any of those AT&T services.
- Set the **Max Concurrent Calls** to the number of simultaneous calls supported. This value is specified by the customer when ordering the AT&T IP Flexible Reach and/or IP Toll Free services. It is a function of the bandwidth of the VoIP network access, codec choices and AT&T service limits.
- Check the **Direct Media** option (to allow media paths to be routed directly to IP and SIP endpoints).
- Select **Codec** row 1 to use "G.729B" or "G.729A" (see Note below) to use as the preferred codec choice.
- Select **Codec** row 2 to use "G.711MU" as the second code choice.
- Select the "2 (20ms)" Frames per packet choice for both codecs.
- Check the **Silence Suppression** option for both codecs.
- Select "t.38-standard" **Mode** with "0" **Redundancy** for fax support.

Note: Calls will use G.729 or G.711 depending on call origination and destination.
AT&T IP Flexible Reach supports two basic IP/PSTN platforms to provide a U.S.-based local footprint. On inbound SDP offers from AT&T to DO, platform 1 offers G.729B, whereas platform 2 offers G.729annexB=no. Both AT&T platforms also support G.711 in both sending and receiving offers. Since DO can be configured to use at most one variant of G.729, either with annexB or with no annexB, and due to the differences in the AT&T platforms, some call scenarios will use G.711 (assuming that G.711 has also been configured on DO). The following table shows the cases in which G.729 or G.711 will be negotiated.

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[1] This default 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]).

|	DO is configured for G.729B	DO is configured for G.729annexB=no
DO to platform 1 call	G.729B is negotiated	G.711 is negotiated
DO to platform 2 call	G.711 is negotiated	G.729annexB=no is negotiated
Platform 1 to DO call	G.729B is negotiated	G.711 is negotiated
Platform 2 to DO call	G.711 is negotiated	G.729annexB=no is negotiated

Press **Apply Changes** before leaving the Add SIP Trunk Group screens.

3.2.1. Configure Outgoing Calling Number

The following entries determine the calling number that will be sent in the SIP From header for the corresponding extensions. In these Application Notes the extension numbers are in the range 200 through 299. The outgoing calling number will be the DID number prefix "160322240xx" where "xx" denotes the last two digits of the extension number.
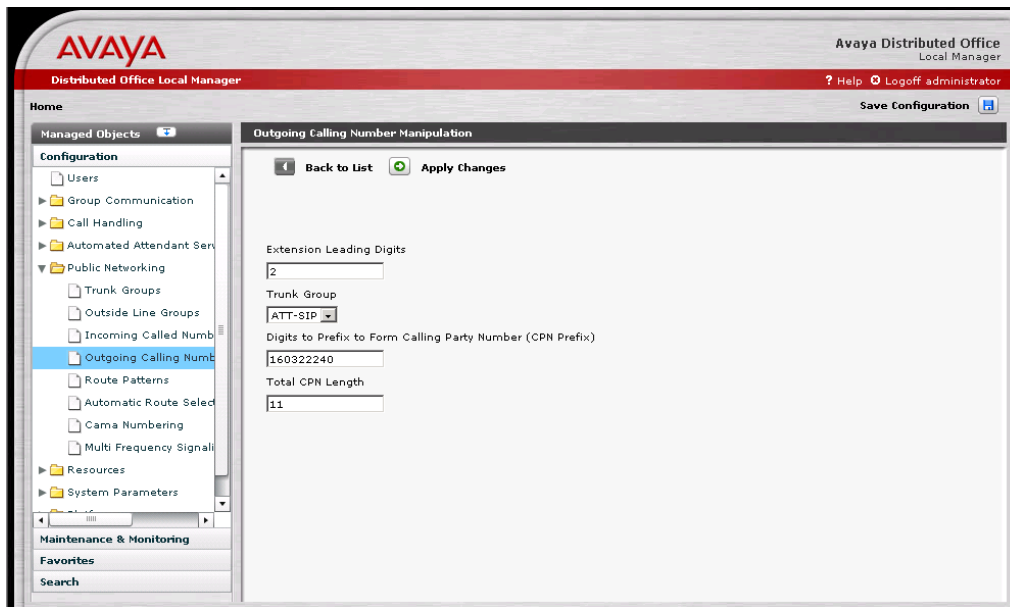
From the left hand **Configuration** menu, expand the **Public Networking** option and select **Outgoing Calling Number**. The **Outgoing Calling Number Manipulation** screen will be displayed.

- Select **Add** to display the next **Outgoing Calling Number Manipulation** listing screen.
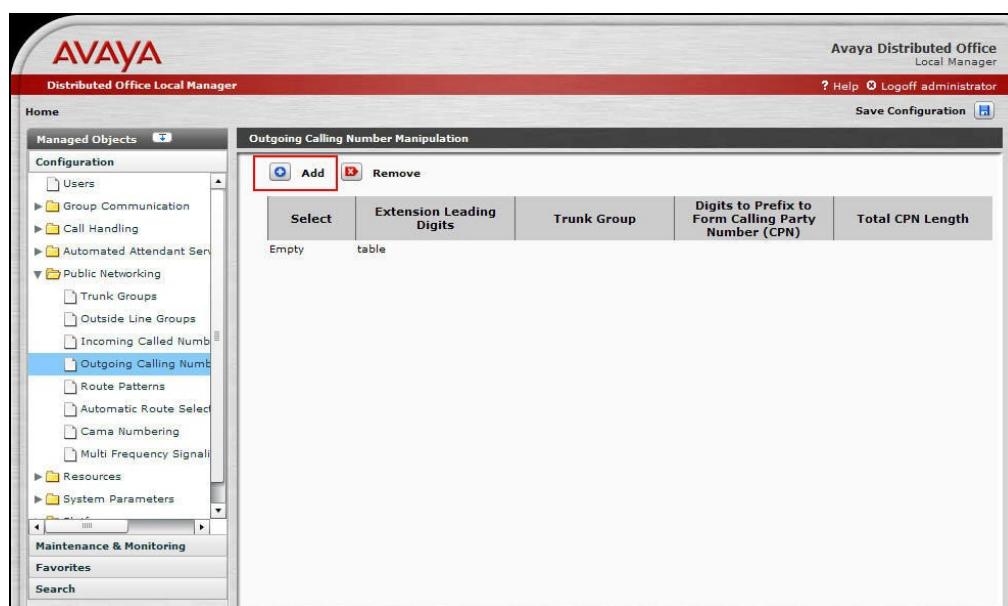

On the **Outgoing Calling Number Manipulation** entry screen,

- Enter the **Extension Leading Digits** necessary to match the applicable range of extension numbers. In these Application Notes, "2" is used to match all extensions in the range 200 through 299.
- Select the **Trunk Group** (e.g. "ATT-SIP") that this rule applies to.
- Enter the **Digits to Prefix to Form Calling Party Number**. In these Application Notes the 9 digit sequence "160322240" was used corresponding to the first 9 digits of the assigned DID number.
- Enter the length of the calling party number in the **Total CPN Length** field. In these Application Notes "11" was used. The last two digits of the extension number will be appended to the nine digit prefix to form the complete eleven digit calling party number.
- Press **Apply Changes** to record the entries and return to the **Outgoing Calling Number Manipulation** summary screen.

The **Outgoing Calling Number Manipulation** summary screen will be displayed.
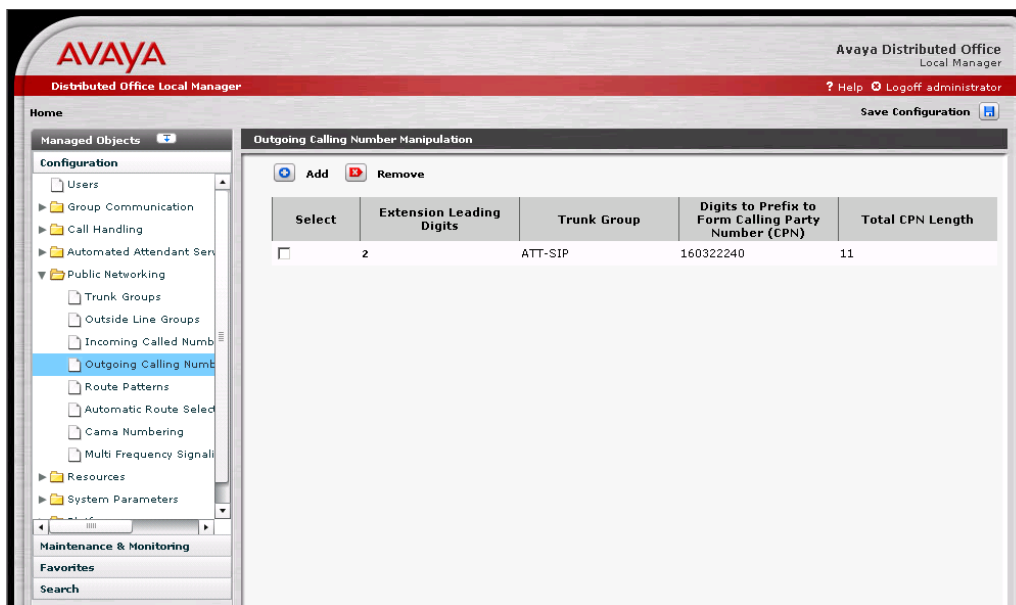

3.2.2. Configure Call Routing

3.2.2.1 Outbound Calls

The Automatic Route Selection (ARS) feature is used to choose the SIP trunk group to the AT&T IP Flexible Reach service for outgoing calls.

ARS administration begins with defining a route pattern which specifies the trunk group(s) and outbound digit manipulation rules to be used.
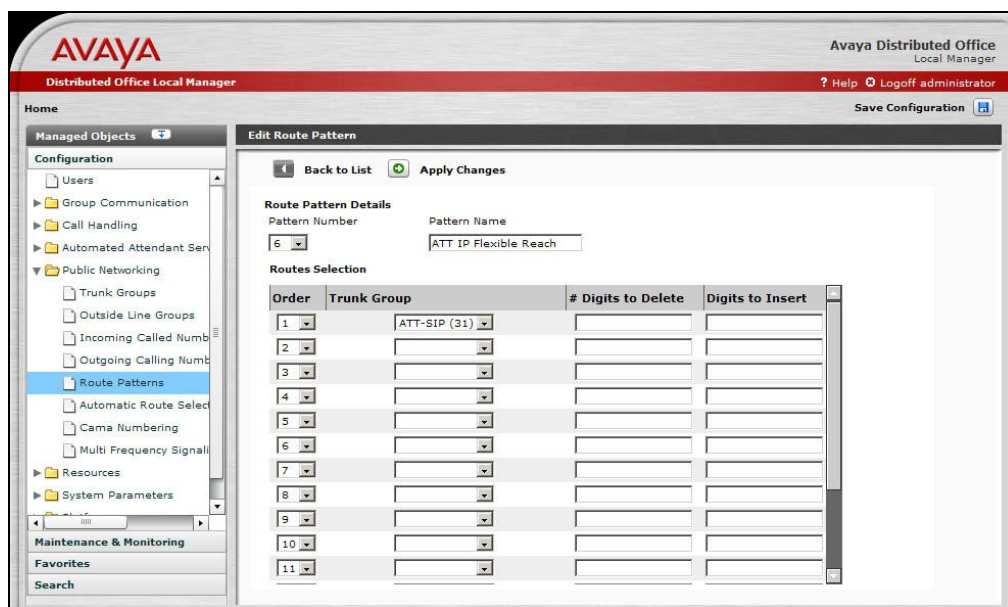
From the left hand **Configuration** menu, expand the **Public Networking** option and select **Route Patterns**. The **Route Patterns** summary screen will be displayed.

- Select **Add New Route Pattern** to display the **Edit Route Pattern** screen.
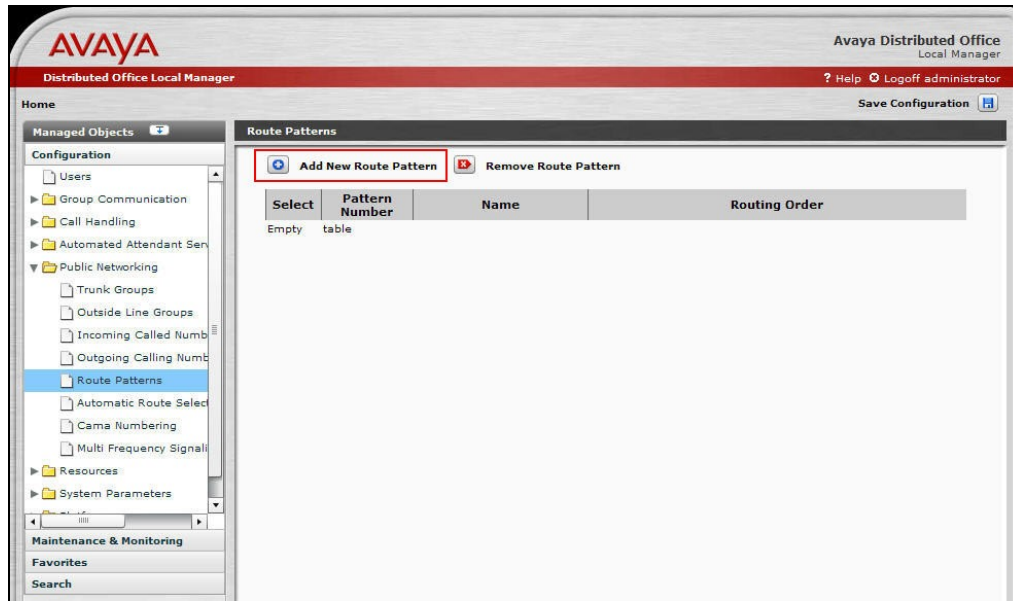
On the **Edit Route Pattern** screen,

- Select an available **Pattern Number**.
- Enter a short text description for the **Pattern Name**. In these Application Notes, "ATT IP Flexible Reach" was used.
- Select the "ATT-SIP (31)" **Trunk Group** in the number "1" **Order** row. This defines the ATT-SIP trunk group as the first (and only) choice trunk group within this route pattern.
- Leave the **# Digits to Delete** and **Digits to Insert** entries for row 1 blank. This means that the digits dialed at the telephone (without the digit "9" prefix used to denote an ARS routed call) will be sent in the SIP RequestURI to the AT&T IP Flexible Reach service.
- Press **Apply Changes** to record the route pattern entry and return to the **Route Patterns** screen.
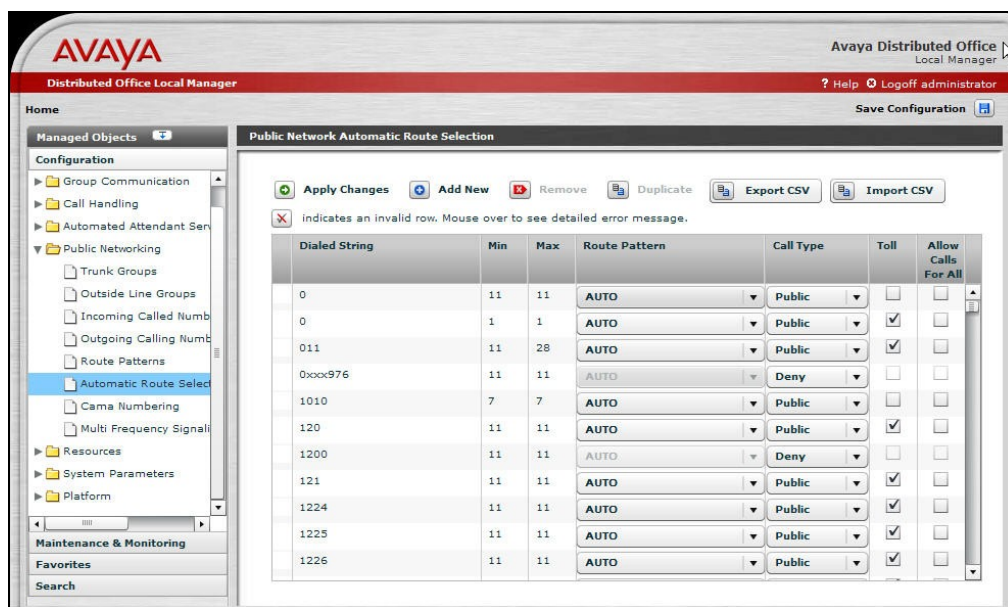
The **Route Patterns** screen is displayed.
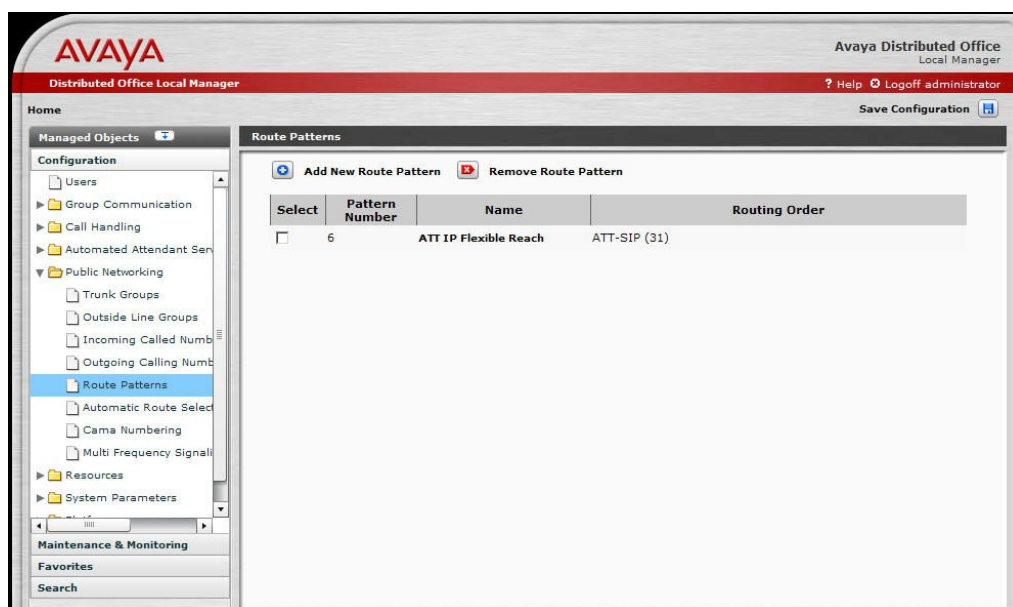
The next step in ARS administration is to define dialing patterns and the corresponding route patterns and call routing privileges.

From the left hand **Configuration** menu, expand the **Public Networking** option and select **Automatic Route Selection**. The **Public Network Automatic Route Selection** screen is displayed.

The following fields are present:
- **Dialed String**: A predefined string to be matched by user-dialed numbers.
- **Min**: The minimum number of user-dialed digits to collect in order to match the dialed string.
- **Max**: The maximum number of user-dialed digits to collect in order to match the dialed string.
- **Route Pattern**: The name of the route pattern (with associated trunk groups and digit manipulation rules) to use when the **Dialed String**, **Min** and **Max** patterns are matched.
- **Call Type**: The type of call that will be placed. Choices include "deny", "local", "public", "emergency" and "crisis-alert".
- **Toll**: Specifies the extensions' privilege level necessary to place the call. Only extensions having "admin" and "high" privileges are able to place toll calls.
- **Allow Calls for All**: Specifies that any phone may place a call for this dialed pattern.

Further information can be found within the Distributed Office online-help function located on each screen.

ARS administration involves configuring the **Route Pattern**, **Call Type** and calling privileges (e.g., **Toll** and **Allow Calls for All** options) for a specific dialing pattern (e.g. the combination of **Dialed String**, **Min** and **Max**).

In these Application Notes, calls to 1-732-xxx-xxxx (where "x" is any digit) are to be routed via the AT&T IP Flexible Reach service without requiring toll calling privileges.
- Enter "1732" for the **Dialed String**.
- Enter "11" for **Min**.
- Enter "11" for **Max**.
- Select "ATT IP Flexible Reach" as the **Route Pattern**.

- Select "Public" as the **Call Type**.
- Uncheck **Toll** to allow extensions with low, medium, high and administrative user privilege levels to place 1-732-xxx-xxxx calls. (Note: the user privilege level is assigned to an extension during User administration, which is beyond the scope of these Application Notes.)
- Uncheck **Allow Calls for All** to prevent extensions with no privileges from being able to place 1-732-xxx-xxxx calls.
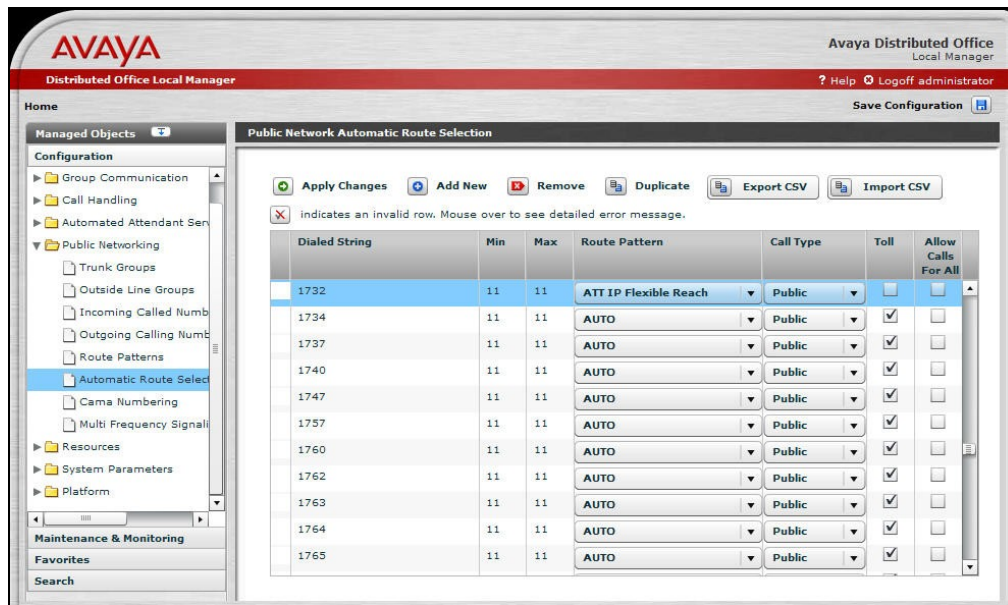
The figure below illustrates configuration information for a number of other dialing patterns.

After completion of the ARS entries:
- Press **Apply Changes** to record the ARS entries.
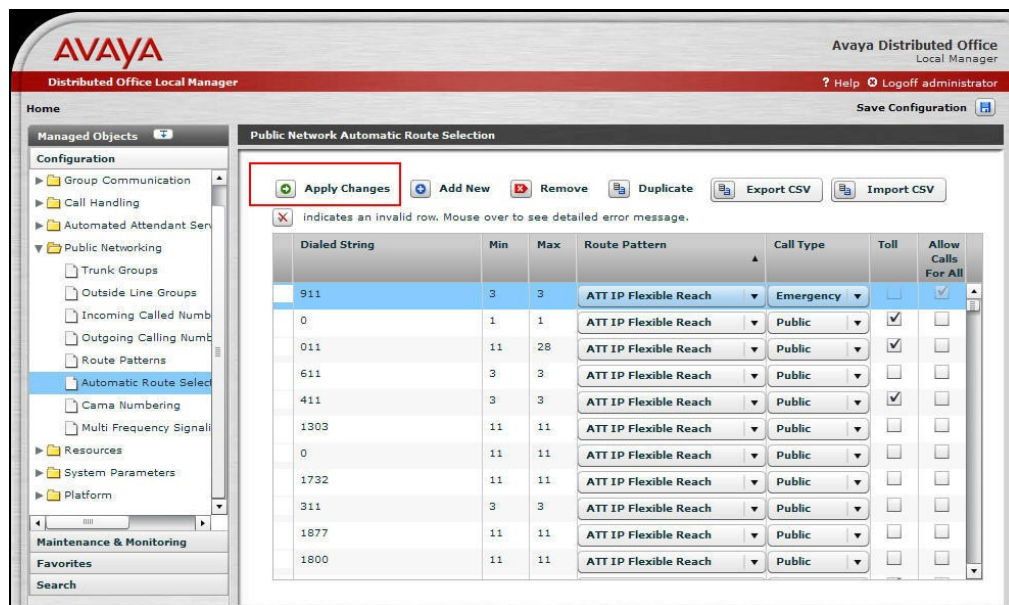
3.2.2.2 Inbound Calls

This step configures the routing of incoming DID calls to the associated Avaya Distributed Office extensions.

In these Application Notes, the incoming PSTN DID numbers 1-603-222-4000 through 4099 and the toll-free number 1-800-333-1234 are assigned by AT&T. They are assigned to the extensions as shown in **Table 3**.

Dialed PSTN Number	Digits Received (within SIP INVITE message)	Extension Assigned
1-603-222-40xx	60322240xx	2xx
1-800-333-1234	0000000034 (Note that the number of digits sent by the AT&T IP Toll Free service be up to 15 digits)	207

**Table 3 - Incoming DID Number Assignments**

Begin configuring the incoming DID assignments as follows:.
- From the left hand **Configuration** menu, expand the **Public Networking** option and select **Incoming Called Number Manipulation**. The **Incoming Called Number Manipulation** screen will be displayed.
- Select **Add** to display the **Add Incoming Called Number Manipulation** screen.
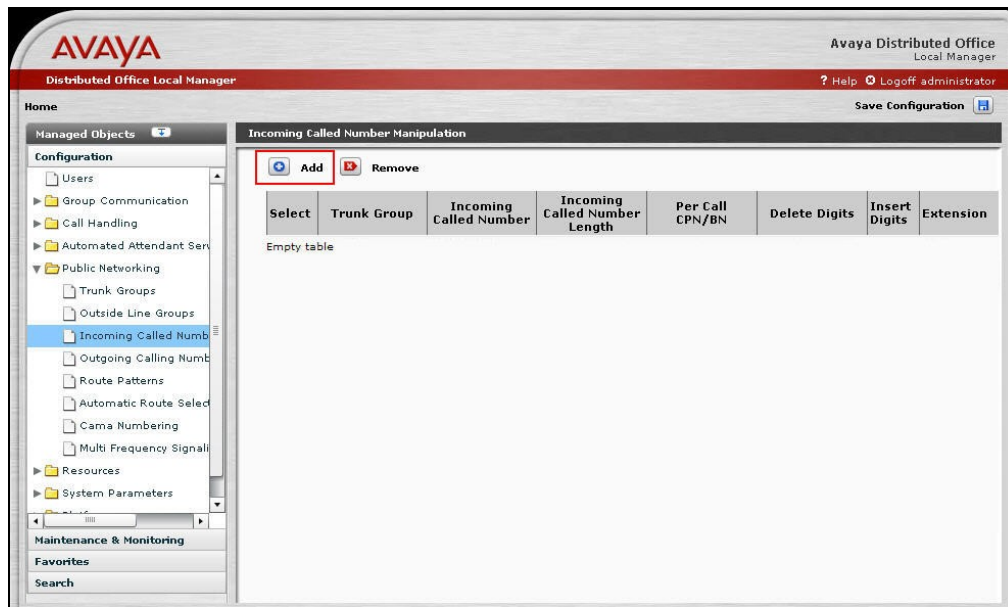
From the **Add Incoming Called Number Manipulation** screen, enter the following to administer the assignments for the 1-603-222-40xx numbers:

- Select "ATT-SIP" as the **Trunk Group**.
- Enter "60322240" as the **Called Number** digit pattern to be matched. Note that AT&T does not send the dialed "1" in the incoming digits.
- Enter "10" as the **Called Number Length**. This is the total number of digits sent by AT&T.
- Select **Digits** as the means that will be used to map the incoming DID number to the corresponding extension.
- Enter "8" as the **# of Digits to Delete** (from the beginning of the 10 incoming digits received).
- Enter "2" as the **Digits to Insert** (to the front of the digits remaining after the digit deletion above).
- Press **Apply Changes** to record the information entered and redisplay the **Incoming Called Number Manipulation** screen.
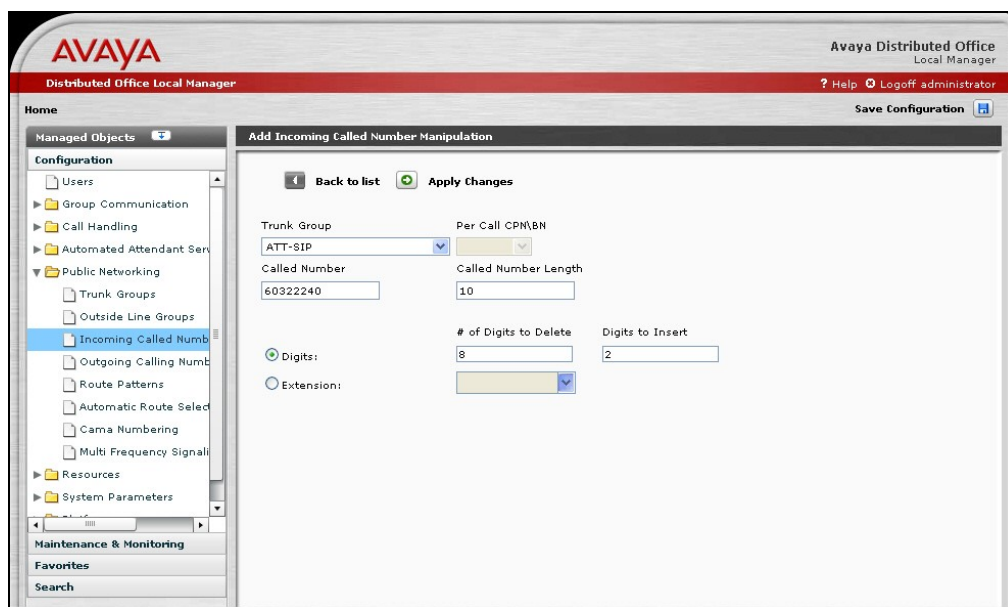
Repeat the **Add Incoming Called Number Manipulation** process to administer the mapping of the incoming IP Toll Free number.

Note that the **Called Number** digits in this case are "0000000034" (not the 1-800-333-1234 dialed PSTN number) and that the **Extension** choice is used to directly assign the incoming number to extension "207".
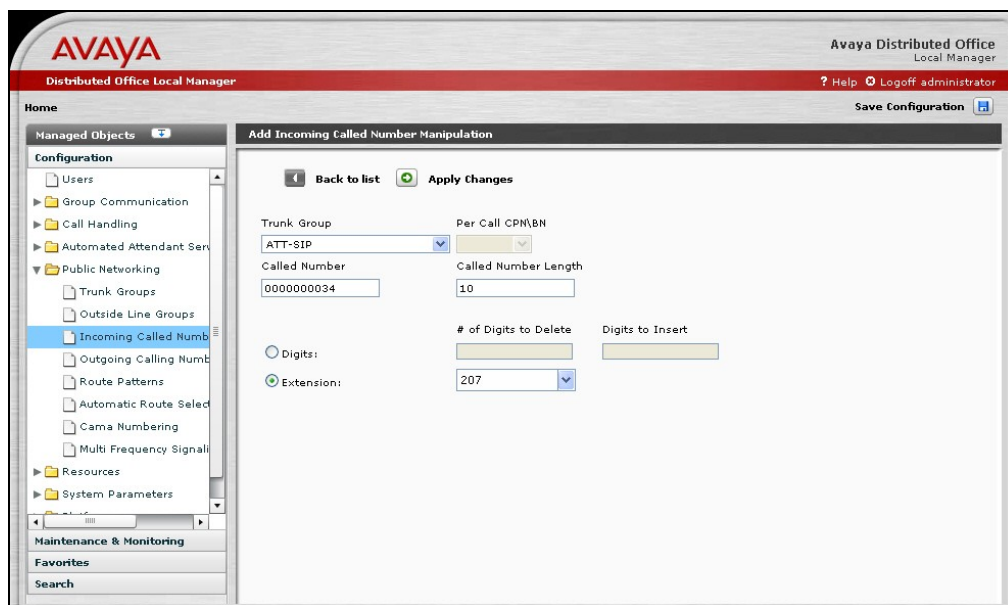
After the **Apply Changes** was performed, the resulting **Incoming Called Number Manipulation** screen is shown.
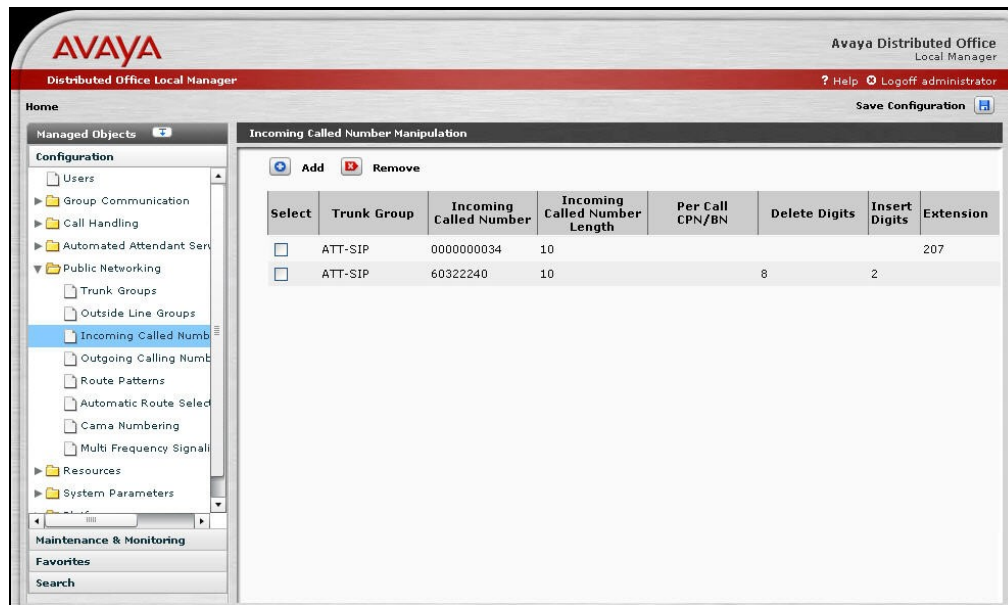
3.2.3. Save Avaya Distributed Office Configuration

The configuration of the Avaya Distributed Office SIP trunking with the AT&T service is now complete.

Save the Avaya Distributed Office configuration (in non-volatile memory) by pressing the **Save Configuration** link found in the upper right hand corner. This prevents the administration changes from being lost upon a reboot or power failure.

4. Interoperability Compliance Testing

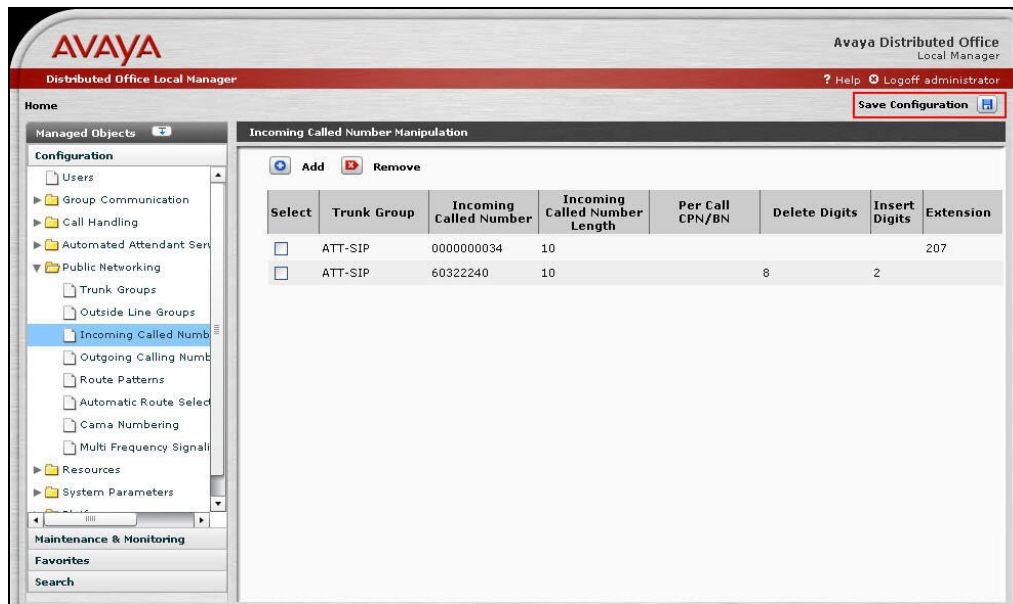This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the AT&T IP Flexible Reach and IP Toll Free services and the Avaya Distributed Office. This section covers the general test approach and the test results.

4.1. General Test Approach

Avaya Distributed Office i120 (Release 1.2) was connected using SIP trunking (via general purpose Internet services) to the AT&T Virtual Interoperability Testing laboratory. The Avaya Distributed Office was configured as if using the generally available service provided by AT&T.

The following features and functionality were covered during the SIP trunking interoperability compliance testing. All testing was successfully completed unless noted otherwise.

- Outgoing calls to PSTN telephones.
- Incoming calls to AT&T provided DID and Toll Free numbers from PSTN telephones.
- Calls using Avaya 4600 IP Telephones with the H.323 firmware configurations.
- Calls using Avaya 9600 Series Telephones with the SIP firmware configurations.
- Calls using Avaya 6211 Analog telephone.
- G.729B and G.711MU codecs for voice calls.
- T.38 codec for fax calling.
- DTMF tone transmission using RFC 2833 with successful Voice Mail / IVR navigation.
- Telephone features such as hold, transfer, conference, and voice mail.
- Trunk to trunk call forwarding, transfers and EC-500 feature operation.
- Direct Media (also known as "shuffling") with IP and SIP telephones.

The following feature is not supported by Avaya Distributed Office with the AT&T services and was not tested.

- Fax detection during G.729 voice with codec change to G.711mu. (T.38 fax is supported instead).

4.2. Test Results

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

The following compatibility issues described in Table 4 were observed.

Item	Issue Observed	Discussion / Workaround
Failure of T.38 fax calls to Super Group 3 fax machines connected to the AT&T Hosted IP Communication Service (HIPCS)	Fax calls (using T.38) to higher speed "super group 3 (SG3) fax machines connected to the AT&T HIPCS service failed to connect and transmit. This failure is due to a known AT&T issue. Operating at normal group 3 speeds was satisfactory.	Avoid using SG3 fax mode when using AT&T SIP trunks to call a fax machine connected to the AT&T HIPCS service. Use alternate PSTN trunking or operate fax machine at lower group 3 speeds.
Emergency 911/E911 Services Limitations	Discussion item added per AT&T request. Contact AT&T for additional information. Note: Explicit 911/E911 testing was not performed during this Avaya DevConnect interoperability testing.	While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the AT&T IP Flexible Reach Service Guide[2] in detail to understand these limitations and restrictions.

Table 4: Summary of Issues Identified During Interoperability Testing

5. Verification Steps

5.1. Verification Tests

This section provides steps that may be performed to verify the operation of the SIP trunking configuration described in the Application Notes.

- Incoming Calls – Verify that calls placed from a PSTN telephone to the DID number assigned are properly routed via the SIP trunk group(s) to the expected extension. Verify the talk-path exists in both directions, that calls remain stable for 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 ARS route patterns. Verify that the talk-path exists

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[2] Access the AT&T IP Flexible Reach Service Guide at
http://new.serviceguide.att.com/portals/sgportal.portal?_nfpb=true&_pageLabel=voip_page

in both directions and that calls remain stable and disconnect properly.

- Inbound DTMF Digit Navigation – Verify inbound DID calls can properly navigate the Avaya Distributed Office automated attendant function.

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

5.2. Troubleshooting Tools

The Avaya Distributed Office has several troubleshooting tools that can be helpful to diagnose SIP trunking issues.

The **Maintenance & Monitoring / Network Diagnostics** menu permits IP pings and traceroutes to performed.

The **Maintenance & Monitoring / Telephony / Trunk Groups** menu provides:
- **Test Selected** – runs tests to verify the operation of the SIP signaling channel for the selected SIP trunk group.
- **Trace Selected** – provided 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 SIP trunk group.

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

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 AT&T network). This mirror port may be used with a SIP protocol analyzer such as WireShark (a.k.a., Ethereal) to monitor the SIP and RTP communications between AT&T and the Avaya Distributed Office. This can be extremely valuable to support advanced troubleshooting.

6. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach Service by calling 1-877-288-8362. Support for the AT&T IP Toll Free Service should be directed to 1-800-325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. The "Connect with Avaya" section provides the worldwide support directory. In the United States, 1-866-GO-AVAYA (1-866-462-8292) provides access to the overall sales and service support menu. Customers may also use specific numbers (provided on

support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

7. Conclusion

These Application Notes describe the steps for configuring SIP trunking between an Avaya Distributed Office (Release 1.2) and AT&T IP Flexible Reach and / or IP Toll Free services.

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.

8. References

The Avaya Distributed Office 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

AT&T IP Flexible Reach and IP Toll Free Service Descriptions
[9] *AT&T IP Flexible Reach Overview,*
http://www.business.att.com/service_fam_overview.jsp?repoid=ProductSub-Category&repoitem=eb_ip_flexreach&serv_port=eb_voip&serv_fam=eb_ip_flexreach&segment=ent_biz
[10] *AT&T IP Toll Free Overview,*
http://www.business.att.com/service_fam_overview.jsp?repoid=ProductSub-Category&repoitem=ip_toll-free&serv_port=eb_voip&serv_fam=ip_toll-free&segment=ent_biz

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.

[11] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[12] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

RL; Reviewed;
SPOC 3/13/2009

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APPENDIX A: Sample SIP INVITE Messages

This section displays the format of typical SIP INVITE messages sent between AT&T and the Avaya Distributed 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 Message from the AT&T services to the Avaya Distributed Office:

```
INVITE sip:6032224007@150.100.100.130:5060 SIP/2.0
Via: SIP/2.0/UDP 20.1.1.54:5060;branch=z9hG4bK34g23u20bgg0tcs313g0.1
From: "OUT_OF_AREA" <sip:+17328251462@20.1.1.54:5060;user=phone>;tag=dsc2c89fc1
To: <sip:6032224007@150.100.100.130;user=phone>
Call-ID: ASE_1211557972012_13726_null_135.25.217.80
CSeq: 1 INVITE
Max-Forwards: 68
Contact: <sip:+17328251462@20.1.1.54:5060;transport=udp>
P-Asserted-Identity: "OUT_OF_AREA" <sip:7328251462@20.1.1.54:5060>
P-DCS-Billing-Info: CBE166B7003D6BD400000000312D303530303030000026CD/0@135.25.31.10
Allow: INVITE,ACK,CANCEL,BYE,INFO,PRACK
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay,
multipart/mixed
Accept-Language: en; q=0.0
Content-Length: 257
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 18095 30118 IN IP4 20.1.1.54
s=SIP Media Capabilities
c=IN IP4 20.1.1.54
t=0 0
m=audio 16806 RTP/AVP 2 18 0 96
a=rtpmap:2 G726-32/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendrecv
```

Sample SIP INVITE Message from Avaya Distributed Office to the AT&T services:

```
INVITE sip:17328251462@20.1.1.54;transport=udp SIP/2.0
Call-ID: 80cef5d33c2edd1aa8483eb2e400
CSeq: 1 INVITE
From: "Ima Sip9640" <sip:6032224007@example.com:6002>;tag=80cef5d33c2edd1a98483eb2e400
Record-Route: <sip:150.100.100.130:5060;lr>,<sip:150.100.100.130:6002;lr;transport=tls>
To: "17328251462" <sip:17328251462@noAttDomain.com>
Via: SIP/2.0/UDP 150.100.100.130:5060;branch=z9hG4bK83838303030333636361fa8.0,SIP/2.0/TLS
150.100.100.130:6002;psrrposn=2;received=150.100.100.130;branch=z9hG4bK80cef5d33c2edd1ab8483eb2e4
00,SIP/2.0/UDP 141.150.155.80;psrrposn=1;received=141.150.155.80;branch=z9hG4bK13_29d54-
6f3e5d18483949f2_I
Content-Length: 186
Content-Type: application/sdp
Contact: "Ima Sip9640" <sip:6032224007@150.100.100.130:6002;transport=tls>
Max-Forwards: 67
User-Agent: Avaya CM/R013w.01.2.024.0
Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
History-Info: <sip:17328251462@noAttDomain.com>;index=1
History-Info: "17328251462" <sip:17328251462@noAttDomain.com>;index=1.1
Supported: 100rel,timer,replaces,join,histinfo
Min-SE: 3600
Session-Expires: 3600;refresher=uac
P-Asserted-Identity: "Ima Sip9640" <sip:6032224007@example.com:6002>

v=0
o=- 1 1 IN IP4 150.100.100.130
s=-
c=IN IP4 150.100.100.131
t=0 0
m=audio 2350 RTP/AVP 18 0 101
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:120 telephone-event/8000
```

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