Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Verizon Business VoIP Service with IP Trunking and Avaya Communication Manager Branch Edition – Issue 1.0

Abstract

These Application Notes describe the steps for configuring SIP trunking between the Verizon Business VoIP Service with IP Trunking and Avaya Communication Manager Branch Edition (formerly known as Distributed Office) using various Avaya telephony endpoints.

The Verizon Business VoIP Service with IP trunking used within these Application Notes is designed for business customers with an Avaya SIP telephony solution. The service provides local and/or long Distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Verizon 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 Verizon's Virtual Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring SIP trunking between the Verizon Business VoIP Service with IP trunking and Avaya Communication Manager Branch Edition (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 **[12]** 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 Communication Manager Branch Edition and the network services offered by Verizon.

The Verizon Business VoIP Service with IP trunking used within these Application Notes is designed for business customers using Avaya Communication Manager and Avaya SIP Enablement Services. The service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks. These trunks connect to the Avaya systems directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

The service is delivered to the enterprise customer location using either of two Verizon Business access options: Internet Dedicated Access (IDA) or Private IP (PIP). These access options differ with respect to whether public or private IP addressing is used for the Internet access. These access methods may include management of routers and firewalls by Verizon.

Single site users can choose combined local and long distance service packages that are available in 200+ metropolitan service areas or a long distance package only. Multi-site or campus configurations may only choose long distance service.

Outbound long distance voice and fax services include direct-dial calling within the 50 U.S. states. They may also include integrated dedicated toll-free service, enabling customers to use a single, cost-effective connection for incoming toll-free calls and outbound VoIP traffic.

***Note** - An Audio Codes MP-202 Gateway was used as the fax interface between the Avaya Communication Manager Branch Edition and the fax endpoint. **[10]**

Verizon 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 Verizon's Integration and Test Lab.

For more information on the Verizon Business VoIP Service with IP Trunking see reference **[9]**

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1.1. Typical Enterprise Customer Location

Figure 1 illustrates the reference configuration used for the testing Avaya Communication Manager Branch Edition with SIP trunking to Verizon IPT service. This configuration includes:

- Avaya Communication Manager Branch Edition 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 Verizon 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 Communication Manager Branch Edition 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 Communication Manager Branch Edition 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 concepts presented in these Application Notes apply to all supported Avaya IP and SIP telephones supported by Avaya Communication Manager Branch Edition. The configurations represented in this application note are specific to the Avaya 9600 and 1600 series telephones.
- IP Network Address Translation (NAT), firewalls, Application Layer Gateway (ALG), and/or Session Border Controller (SBC) devices may exist between the Verizon services and the Avaya Communication Manager Branch Edition 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.

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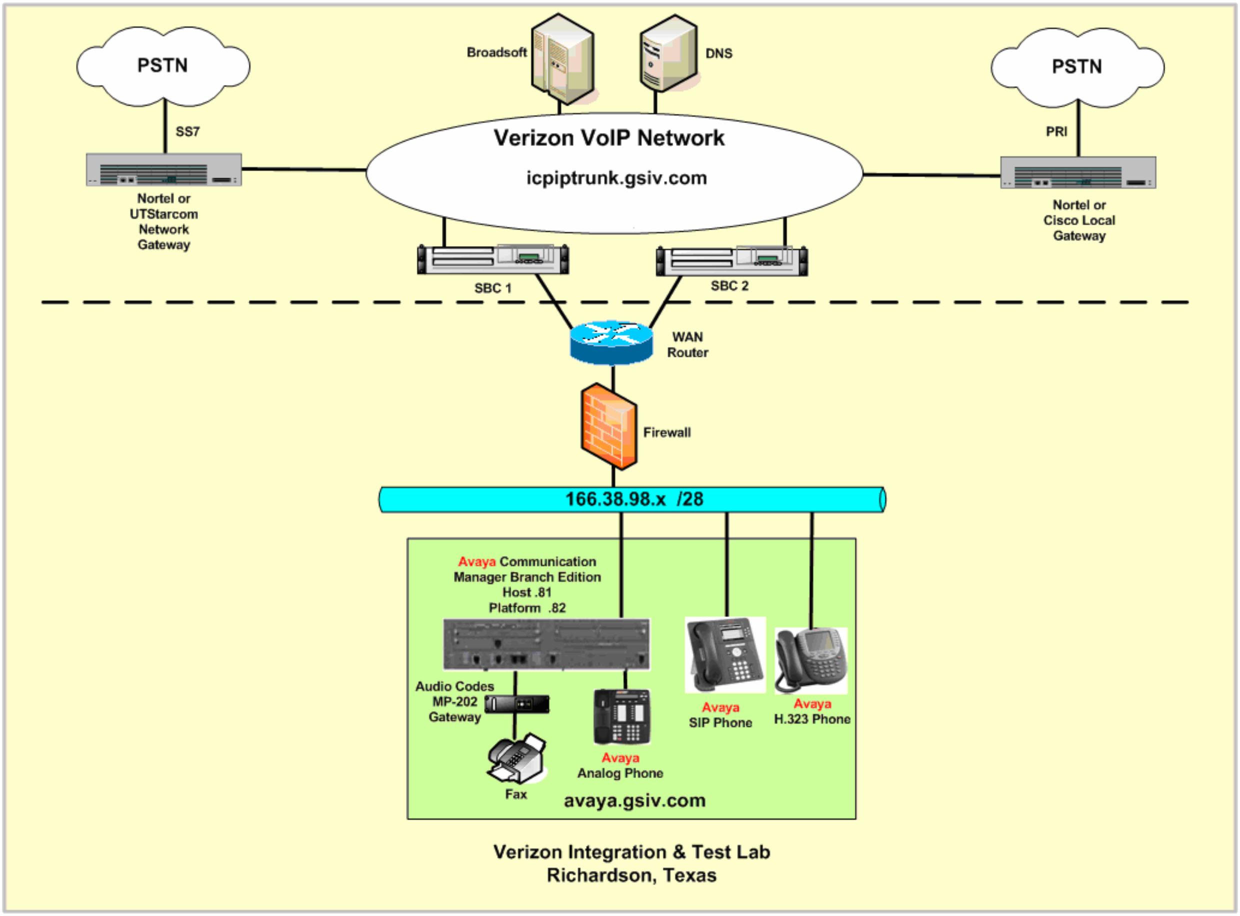

Figure 1 – Reference Configuration

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Figure 2 illustrates the Network Connection information for the Avaya Communication Manager Branch Edition i120.

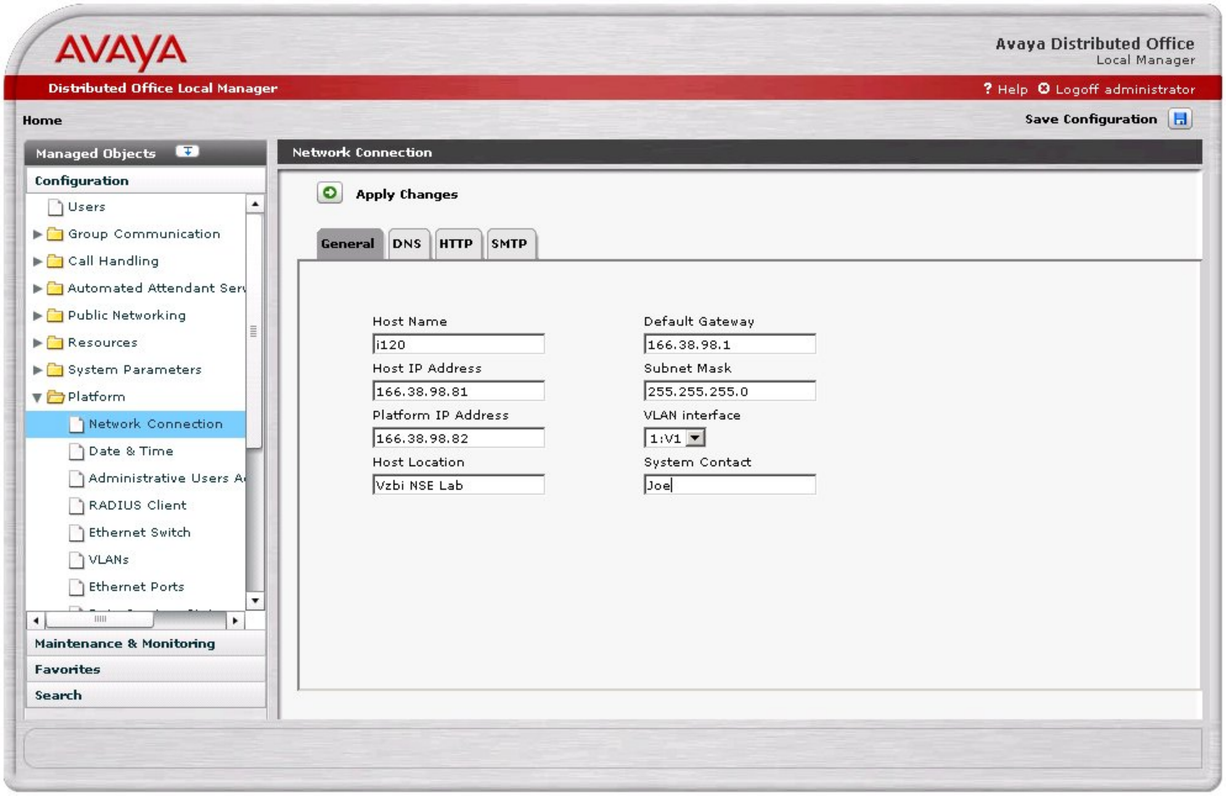

Figure 2 - Avaya Communication Manager Branch Edition Network Connection Assignments

Note the following information is provided to Verizon during the ordering process for the Verizon Business VoIP Service with IP trunking. The IP addresses mentioned below are examples only.

- Avaya Communication Manager Branch Edition **Host IP Address** of "166.38.98.81". This is the only address used for all SIP signaling between the Avaya Communication Manager Branch Edition and the Verizon Border Elements.

- The IP addresses of the Real Time Protocol (RTP) media endpoints will be from various addresses within the "166.38.98.0.0/24" subnet in these Application Notes.

  Specifically, the Avaya Communication Manager Branch Edition **Platform IP Address** of "166.38.98.82" 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 Verizon Border Elements whenever using direct media.

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1.2. Verizon Services Configuration Information

These Application Notes provide **an illustrative example** of how the Avaya Communication Manager Branch Edition SIP trunking solution is configured for the Verizon Business VoIP Service with IP trunking.

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 Verizon during service provisioning.*

Verizon Provisioning Information	Illustrative Values in these Application Notes
Verizon Border Element IP Address(es)	166.34.120.87
G.729A, G.711MU Codecs Support	Yes
RFC 2833 (DTMF Event) Supported	Yes
Maximum Concurrent Calls with Direct Media disabled	15-30 (depends on codec)
Maximum Concurrent Calls with Direct Media enabled.(specified by customer during service ordering)	180
Assigned Direct Inward Dial (DID) Numbers	1-603-555-40xx (where x is any digit)
DID Digits Passed in SIP Request URI	55540xx
DID Digits Passed in SIP To Header	Same as SIP Request URI

Table 1 – Illustrative Verizon Network Provisioning Information

2. Equipment and Software Validated

The following equipment and software was used during the DevConnect compliance testing with the Verizon services.

Component	Version
Avaya	
Avaya Communication Manager Branch Edition i120	Release 1.2 (1.2.1_2.0.1)
Avaya 9620 one-X™ Deskphone SIP Telephone	Release 2.0.5
Avaya 1608 IP (H.323) Telephone	Release 1.0.3
Audio Codes MP-202	2.4.0_build_7
Verizon	
Verizon Business VoIP Service with IP Trunking	Integrated release 2008.4

Table 2 – Equipment and Version

3. Configure Avaya Communication Manager Branch Edition

The Avaya Communication Manager Branch Edition 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.

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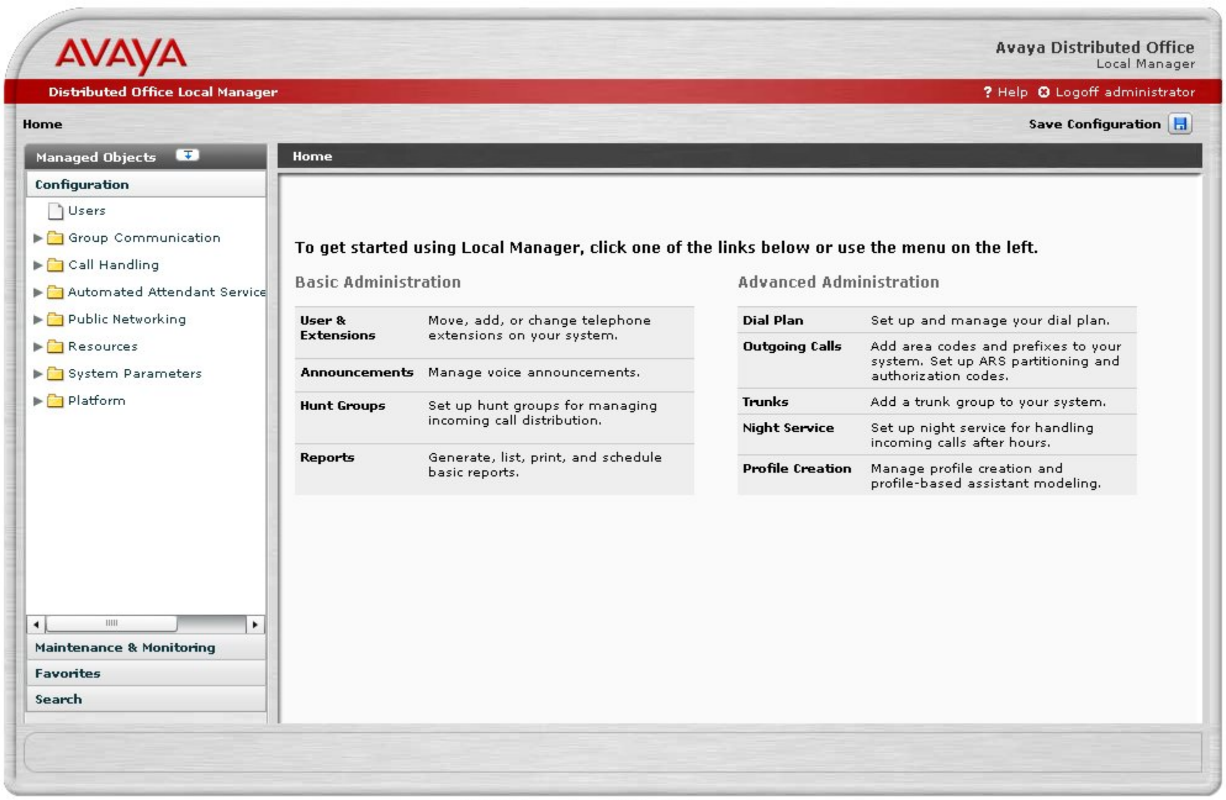
3.1. Login to Avaya Communication Manager Branch Edition

Using a web browser, access the Avaya Communication Manager Branch Edition Local Manager by entering “http://<ip-addr>/” where “<ip-addr>” is the **Host IP Address** of the Avaya Communication Manager Branch Edition. In these Application Notes, “http://166.38.98.81” is used.

Login with the appropriate credentials. The Home Local Manager screen is shown.



3.2. Add a SIP Trunk Group to the Verizon Services

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

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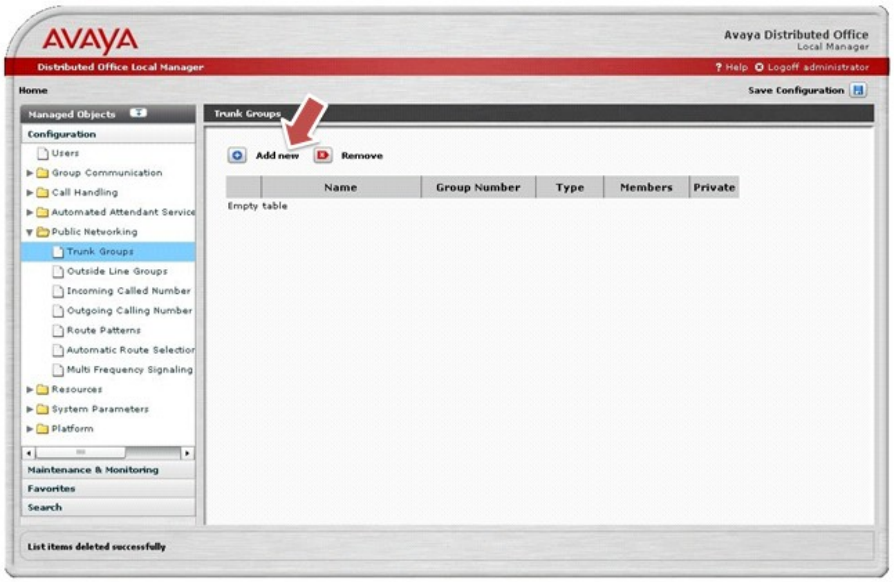
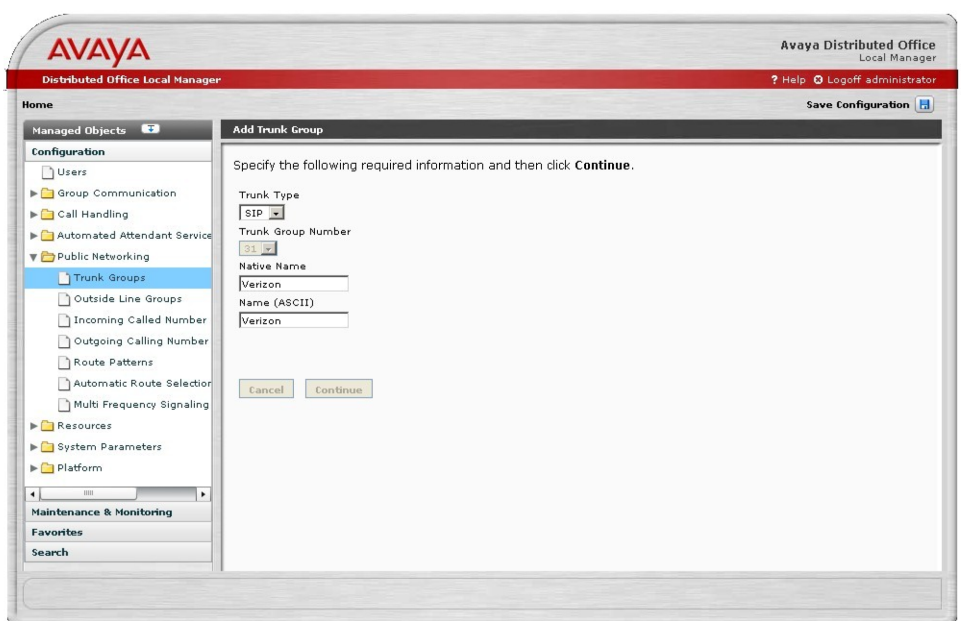
Select **Add New** to display the **Add Trunk Group** screen.



On the Add Trunk Group screen:

- Set the **Trunk Type** to “SIP”.
- Enter a short text description of the trunk group (e.g., “Verizon”) 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.



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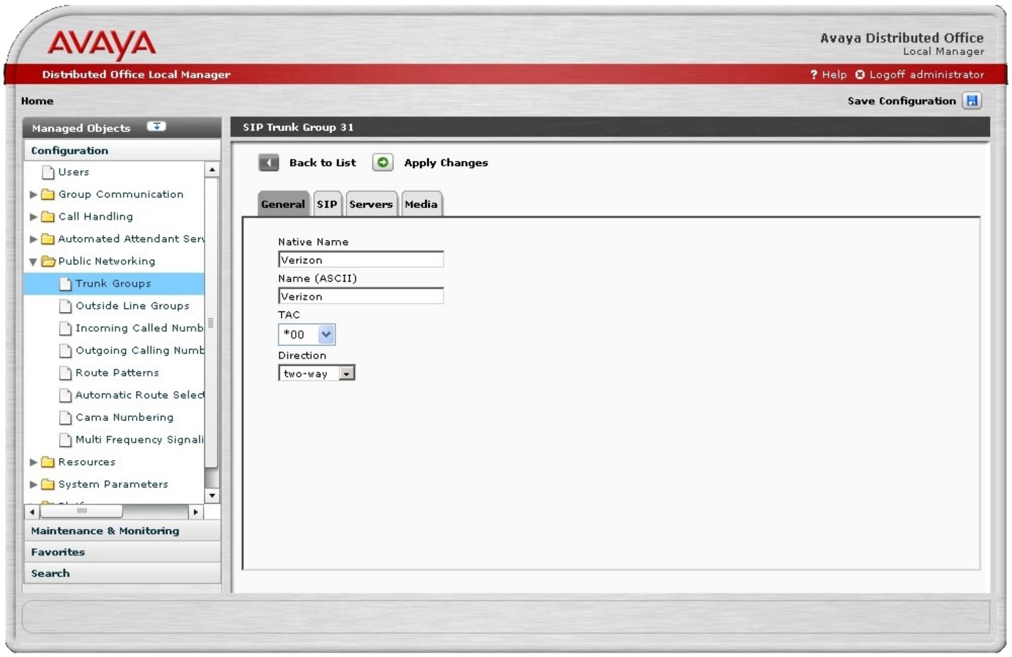
The Add SIP Trunk Group General Tab screen is shown.

- 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 Verizon services.
- Enter the customer’s SIP domain for the Communication Manager Branch Edition in the **Near-End Domain** field. In these Application Notes, “avaya.gsiv.com” was used. It is not necessary that this domain be resolvable for the Verizon SIP trunking.
- Enter “600” 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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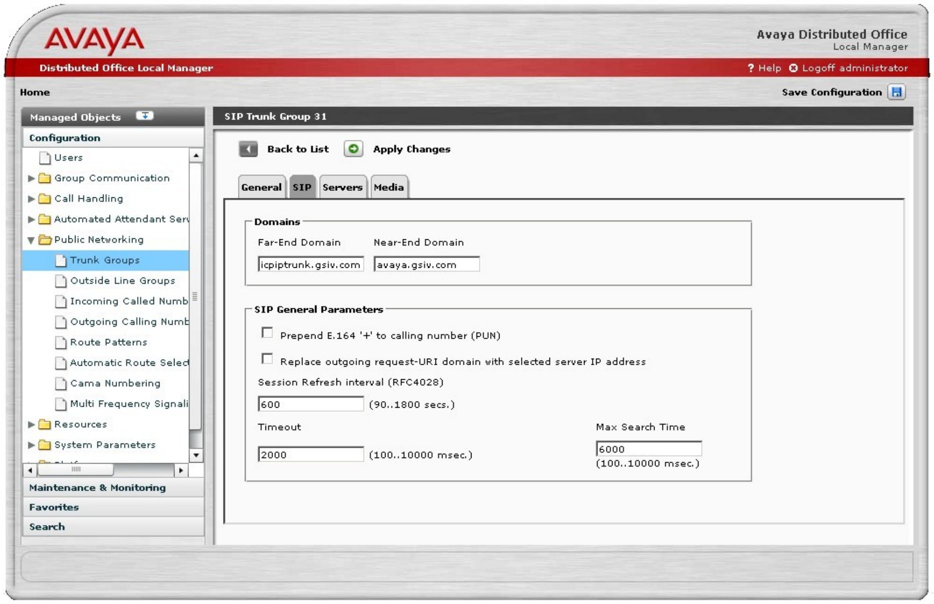
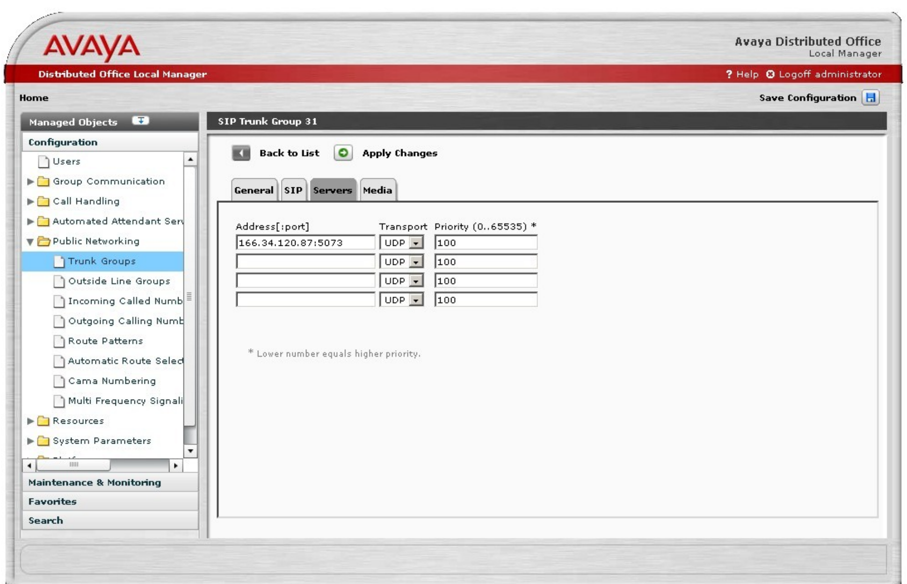


On the **Servers** tab:

- Enter the IP address of the primary Verizon Border Element provided by Verizon in the **Address** field. In this Application Note, “166.34.120.87:5073” is used as noted in Section 1.2. It is necessary to specify the port since the UDP default “5060” is not used.
- Select “UDP” for the **Transport** field value.
- The default **Priority** field settings shown are used.
- Press the **Media** tab to advance to the next screen.

***Note:** Avaya’s Best Practice recommendation is to use Port 5060 for all configurations



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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 "101" was used matching the Avaya 96xx series SIP phone default.[1]
  Note: For inbound trunk calls from an Verizon 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 Communication Manager Branch Edition and the Avaya 96xx series SIP phones must match the RTP telephone events payload type offered by the specific Verizon service. Disabling the Direct Media option (see below) on Avaya Communication Manager Branch Edition will allow the 96xx series SIP phones to send DTMF back to the network on inbound calls from any of those Verizon services.
- Set the **Max Concurrent Calls** to the number of simultaneous calls supported. This value is specified by the customer when ordering the Verizon services. It is a function of the bandwidth of the VoIP network access, codec choices and Verizon 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.729A" (see Note below) to use as the preferred codec choice.
- Select **Codec** row 2 to use "G.711MU" as the second code choice.

[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]).

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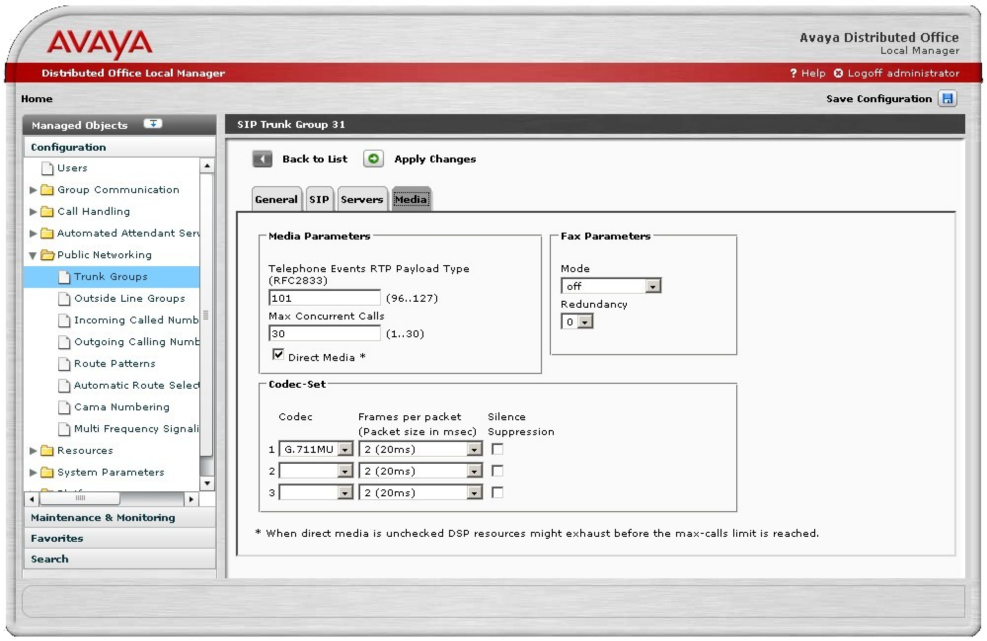
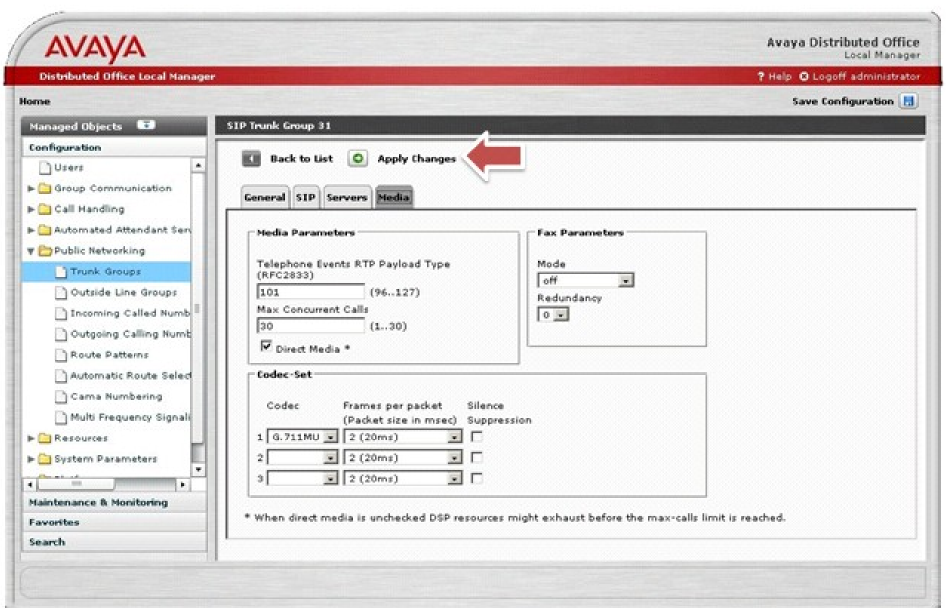


Press **Apply Changes** before leaving the Add SIP Trunk Group screens.



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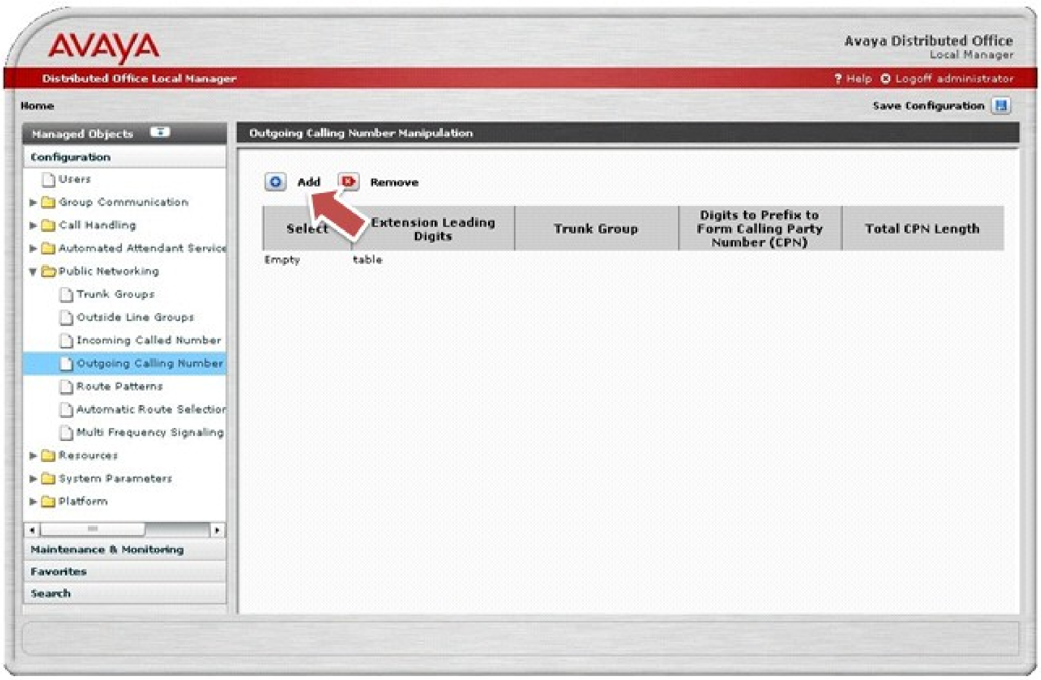
3.2.1. Configure Outgoing Calling Number

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 1000 through 4999.

From the left hand **Configuration** menu, expand the **Public Networking** option and select **Outgoing Calling Number Manipulation**. 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 originating extension numbers. In these Application Notes, “122” is used to match all extensions in the range 1220 through 1229.
- Select the **Trunk Group** (e.g. “Verizon”) that this rule applies to.
- Enter the length of the calling party number and CPN prefix digits in the **Total CPN Length** field. In these Application Notes “4” was used.
- Press **Apply Changes** to record the entries and return to the **Outgoing Calling Number Manipulation** summary screen.

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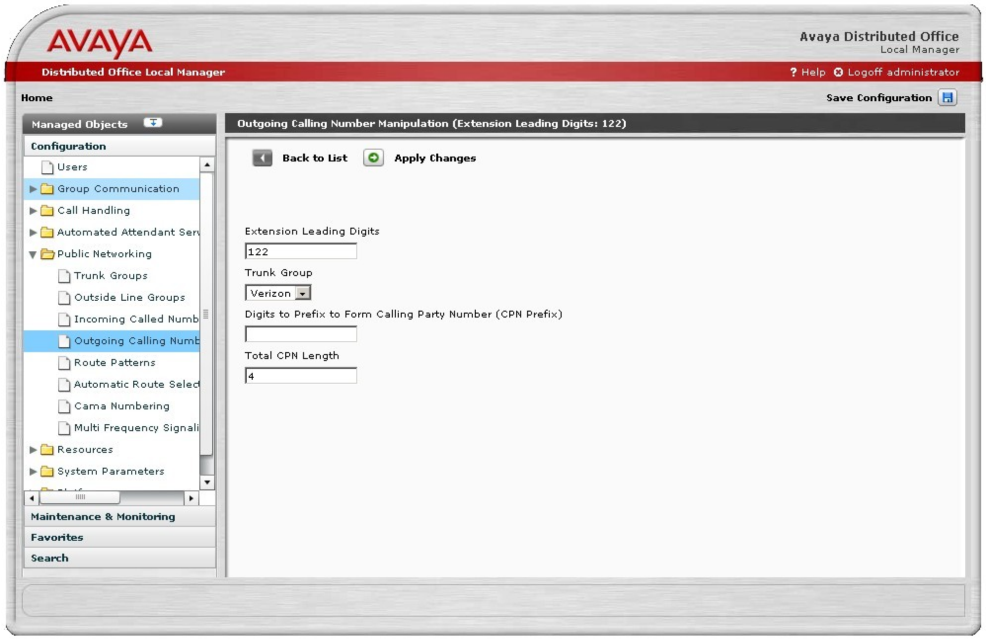
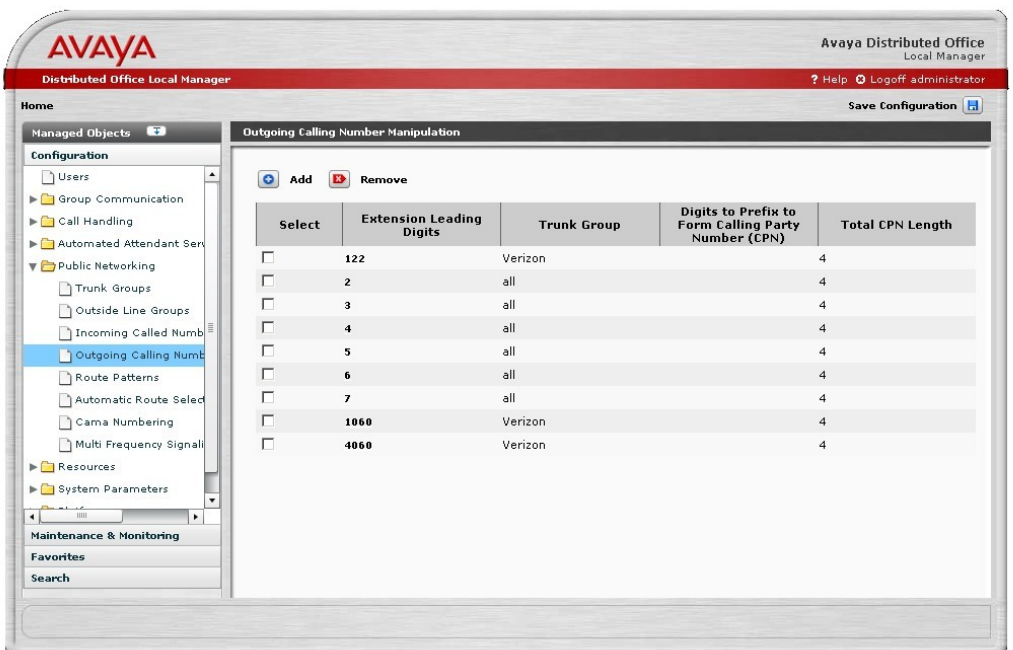


The **Outgoing Calling Number Manipulation** summary screen will be displayed.



3.2.2. Configure Call Routing

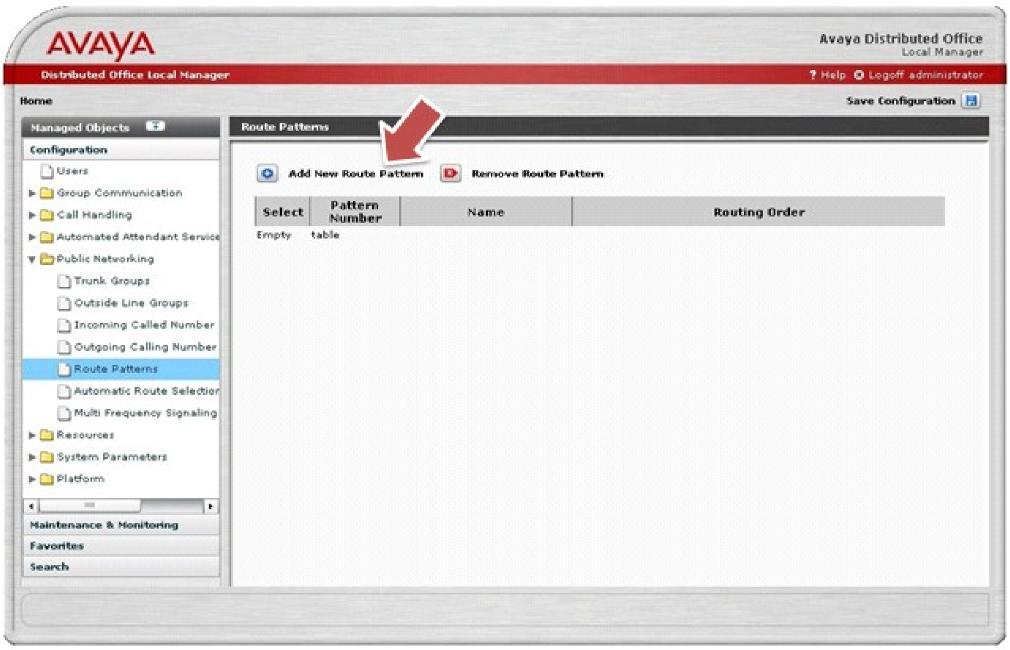
3.2.2.1 Outbound Calls

The Automatic Route Selection (ARS) feature is used to choose the SIP trunk group to the Verizon 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, "VerizonPSTN" was used.
- Select the "Verizon (31)" **Trunk Group** in the number "1" **Order** row. This defines the Verizon 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 Request URI to the Verizon service.
- Press **Apply Changes** to record the route pattern entry and return to the **Route Patterns** screen.

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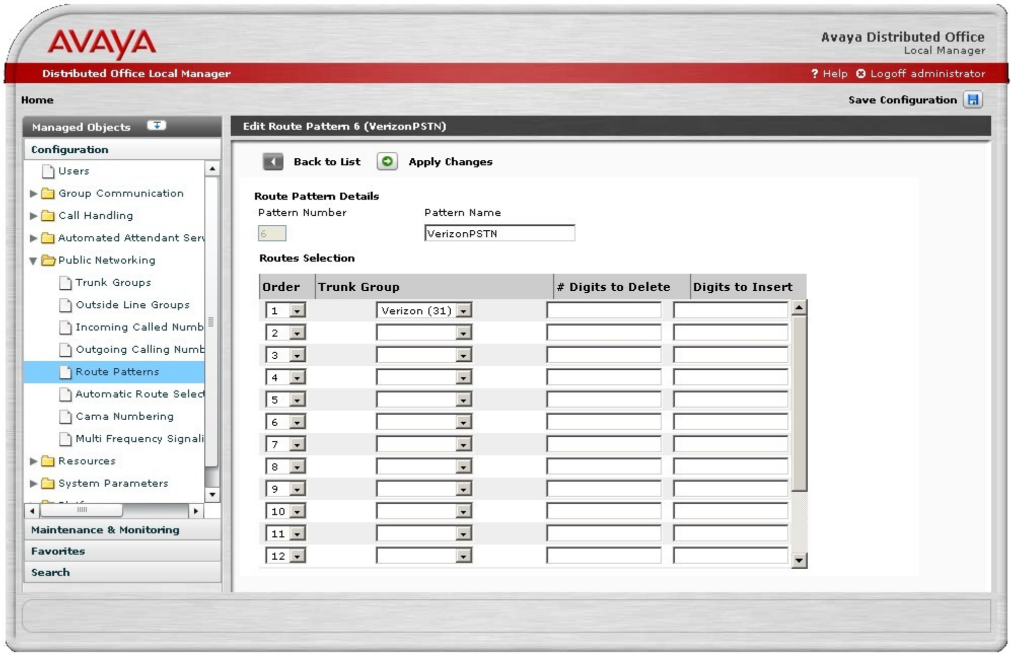
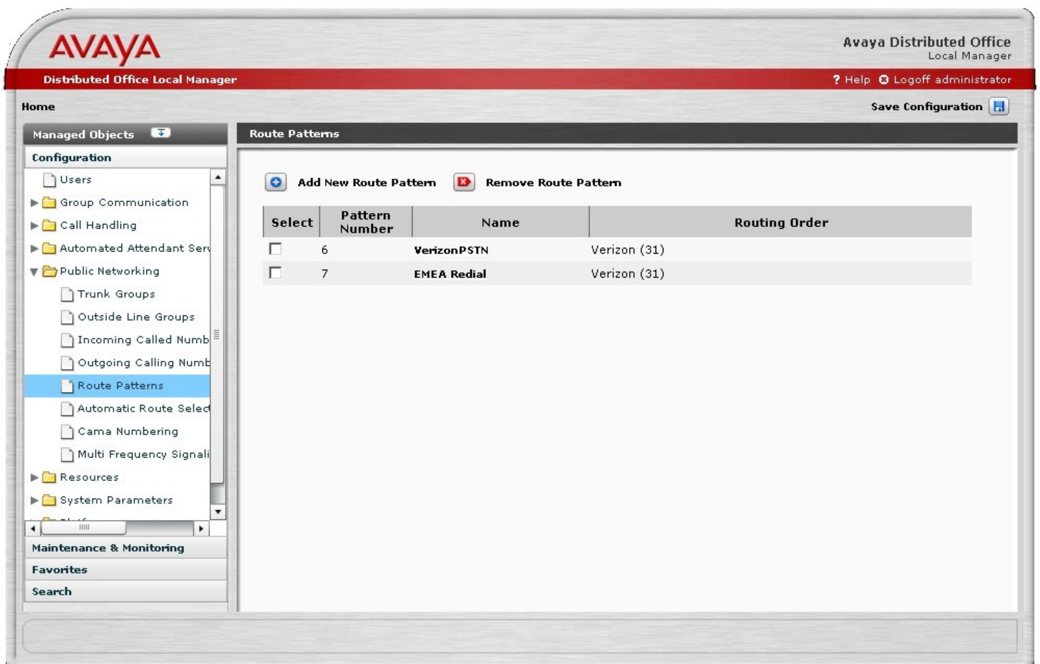


The **Route Patterns** screen is displayed.



The next step in ARS administration is to define dialing patterns and the corresponding route patterns and call routing privileges.

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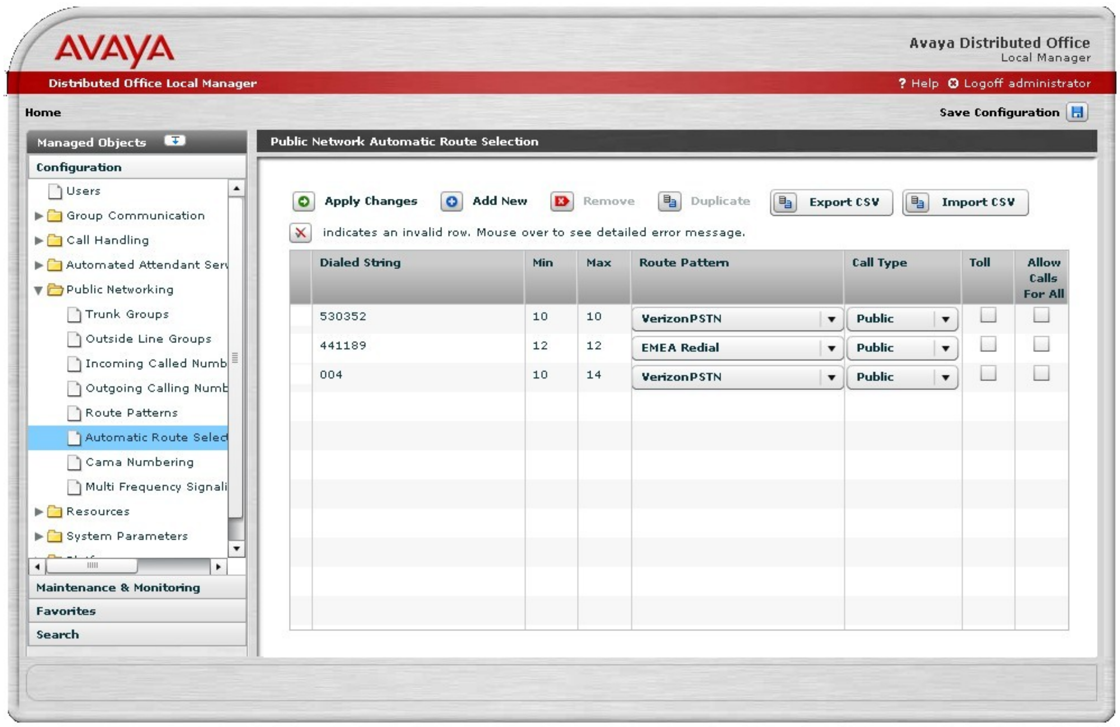
From the left hand **Configuration** menu, expand the **Public Networking** option and select **Automatic Route Selection**. The **Public Network Automatic Route Selection** screen is displayed.



The figure above shows three Public Automatic Route Selection entries. The purposes of the routes are as follows:

- 530352 – This route handles all outbound traffic to North America (NAR).
- 441189 – This route handles all outbound traffic to Europe and Middle East and Africa (EMEA).
- 004 – This route is used to capture and insert CPN digits for extensions dialed using the re-dial option from a call log.

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”.

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- **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 Communication Manager Branch Edition 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 530-352-xxxx (where "x" is any digit) are to be routed via the Verizon service without requiring toll calling privileges.

- Enter "530352" for the **Dialed String**.
- Enter "10" for **Min**.
- Enter "10" for **Max**.
- Select "VerizonPSTN" 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 530-352-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 530-352-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.

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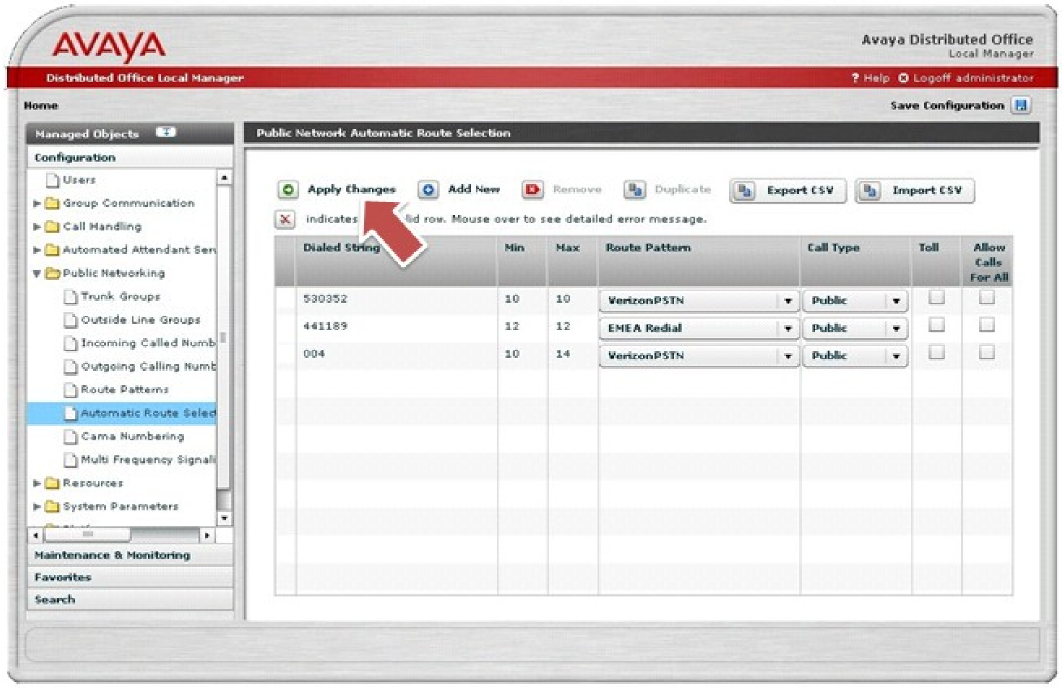

3.2.2.2 Inbound Calls

This step configures the routing of incoming DID calls to the associated Avaya Communication Manager Branch Edition extensions.

In these Application Notes, the incoming PSTN DID numbers 1-603-555-4000 through 4099 are assigned by Verizon. They are assigned to the extensions as shown in **Table 3**.

Dialed PSTN Number	Digits Received (within SIP INVITE message)	Extension Assigned
1-603-555-40xx	55540xx	40xx

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.

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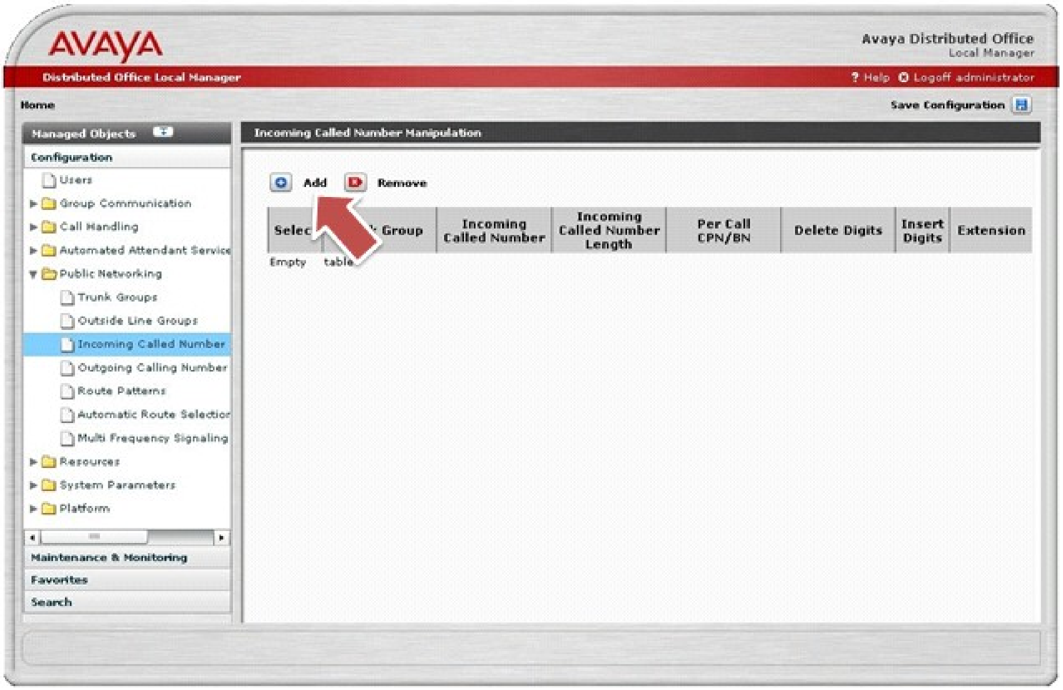

From the **Add Incoming Called Number Manipulation** screen, enter the following to administer the assignments for the 1-603-555-40xx numbers:

- Select "Verizon" as the **Trunk Group**.
- Enter "555" as the **Called Number** digit pattern to be matched. Note that Verizon does not send the dialed "1" in the incoming digits.
- Enter "7" as the **Called Number Length**. This is the total number of digits sent by Verizon.
- Select **Digits** as the means that will be used to map the incoming DID number to the corresponding extension.
- Enter "3" as the **# of Digits to Delete** (from the beginning of the 7 incoming digits received).
- Press **Apply Changes** to record the information entered and redisplay the **Incoming Called Number Manipulation** screen.

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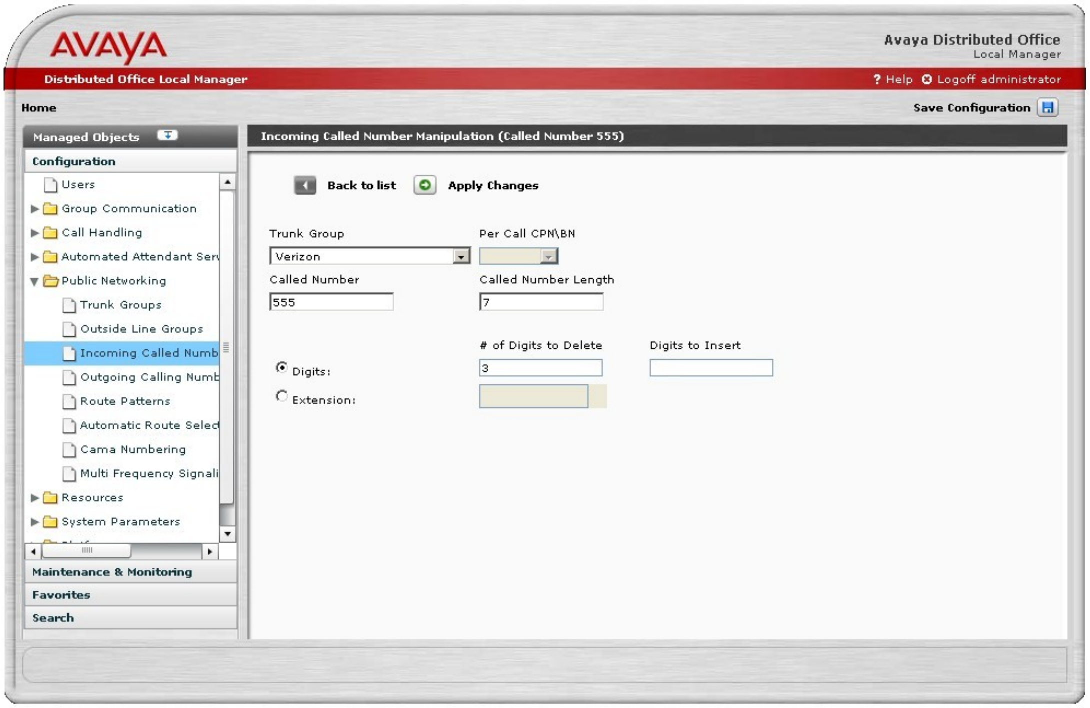
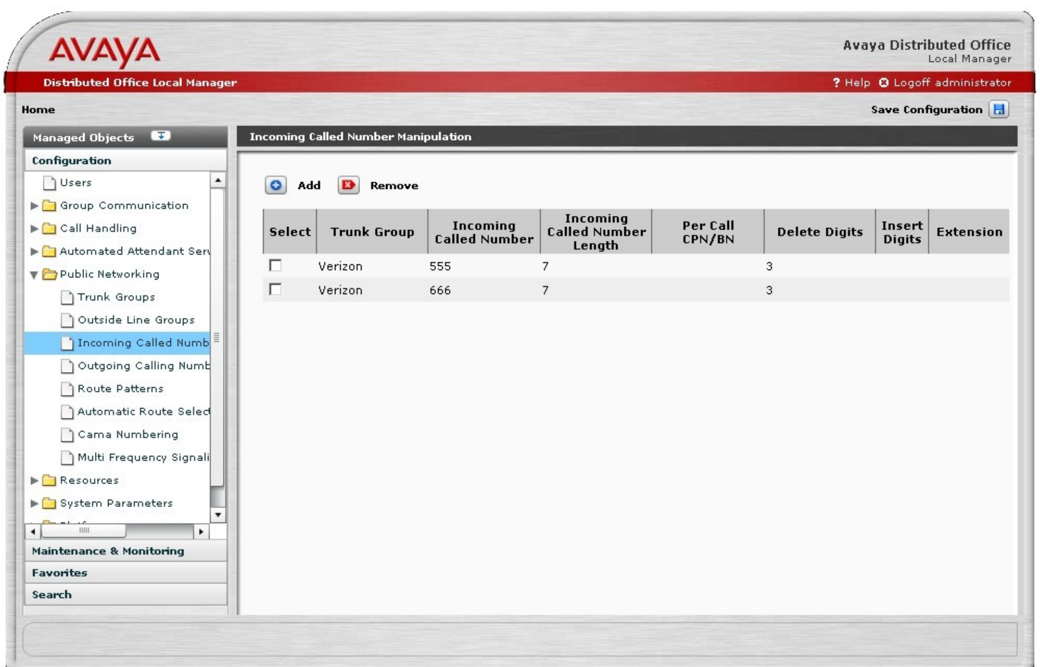


Repeat the **Add Incoming Called Number Manipulation** process to administer the mapping of the incoming IP Toll Free number.

After the **Apply Changes** was performed, the resulting **Incoming Called Number Manipulation** screen is shown.



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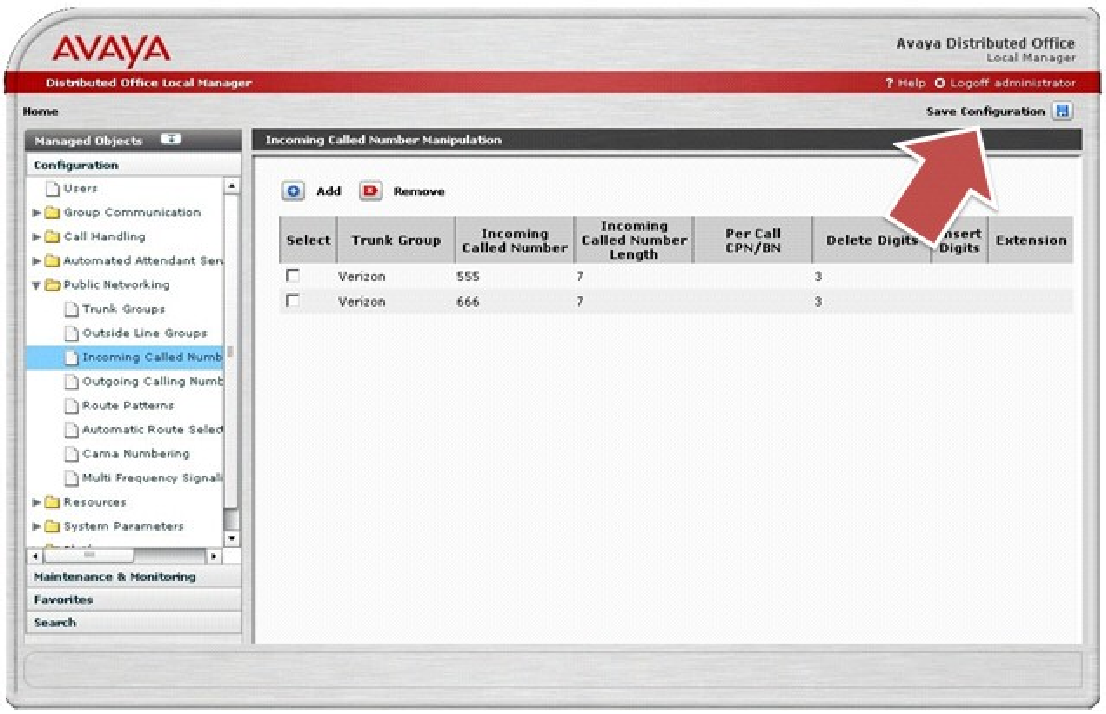
3.2.3. Save Avaya Communication Manager Branch Edition Configuration

The configuration of the Avaya Communication Manager Branch Edition SIP trunking with the Verizon service is now complete.

Save the Avaya Communication Manager Branch Edition 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.



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4. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Verizon Business VoIP Service with IP Trunking and the Avaya Communication Manager Branch Edition. This section covers the general test approach and the test results.

4.1. General Test Approach

Avaya Communication Manager Branch Edition i120 (Release 1.2) was connected using SIP trunking (via general purpose Internet services) to the Verizon CPE Interoperability Testing laboratory. The Avaya Communication Manager Branch Edition was configured as if using the generally available service provided by Verizon.

The Verizon Business compliance test criteria are specified in reference [11].

4.2. Test Results

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

Item	Issue Observed	Discussion / Workaround
Signaling CLI: Privacy	When presentation restriction is requested by the caller, the customer must still send the P-Asserted-Identity header, along with the Privacy header (Privacy: id) defined in RFC 3323 to Verizon. When presentation restriction is requested from the PSTN-caller, the retail customer will not receive the CLI and the From header will contain 'anonymous'.	The retail customer must set the From header to anonymous if privacy is requested. Correction for this item is available in a future release.
Diversion Header: Transmission.	The CMBE R 1.2.1 does not support sending a Diversion Header. This prevents the retail customer from transferring or forwarding a call off-net that also originated off-net.	Correction for this item is available in a future release.
SIP Servers: DNS SRV and DNS A Records.	The CMBE R 1.2.1 does not resolve DNS A record queries to resolve hostnames.	Correction for this item is available in a future release.
Call Forward: REFER	The CMBE R 1.2.1 does not support forwarding calls via the REFER methodology. All signaling and media will be handled by CMBE rather than the forwarded endpoint.	CMBE supports forwarding using the RE-INVITE method.

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

- Inbound DTMF Digit Navigation – Verify inbound DID calls can properly navigate the Avaya Communication Manager Branch Edition 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 Communication Manager Branch Edition 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 be 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

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Verizon 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 Verizon and the Avaya Communication Manager Branch Edition. This can be extremely valuable to support advanced troubleshooting.

6. Support

For technical support on Verizon Business VoIP Service, contact Verizon Business customer Service at 1-800-265-2316 or via their online support at:
http://www.verizonbusiness.com/us/customer/

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 Communication Manager Branch Edition (Release 1.2) and Verizon Business VoIP Service with IP Trunking. Based on the findings documented with this document, the integration of Avaya Communication Manager Branch Edition with Verizon Business IP Trunk service offer successfully validated for interoperability.

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.

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8. References

The Avaya Distributed Office product documentation is available at http://support.avaya.com.

[1] Avaya Distributed Office Documentation Map Release 1.2, 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 Release 1.2, 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
[9] The Verizon Business VoIP IP Trunking service description is available at: http://www.verizonbusiness.com/us/products/voip/trunking/
[10] The following documentation is available on request from www.audiocodes.com
MP-202 Telephone Adapter User's Manual: Version 2.4.0: Document: LTRT-50604
MP-202 Telephone Adapter Quick Installation Guide: Document LTRT-50404
[11] *Verizon Business Product Integration requirement Avaya IP-PBX 5.1 SIP TRUNK Interoperability Testing, Date:10/10/08, Rev 1.1*

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.

[12] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[13] RFC 3262 – *Reliability of Provisional Responses in SIP*, June 2002, Proposed Standard
[14] RFC 3264 – *An Offer/Answer Model with SDP*, June 2002, Proposed Standard
[15] RFC 4028 – *Session Timers in SIP,* April 2005, Proposed Standard
[16] RFC 4244 – *An Extension to SIP for REQUEST HISTORY Information*, November 2005, Proposed Standard
[17] RFC 4566 – *Session Description Protocol – SDP*, July 2006, Proposed Standard
[18] RFC 3325 – *Private Extension to SIP – Network Asserted Identity within Trusted Networks*, November 2002, Proposed Standard
[19] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard
[20] RFC 3581 – *An Extension to Session Initiation Protocol (SIP) for Symmetric Response Routing*, August 2003, Proposed Standard

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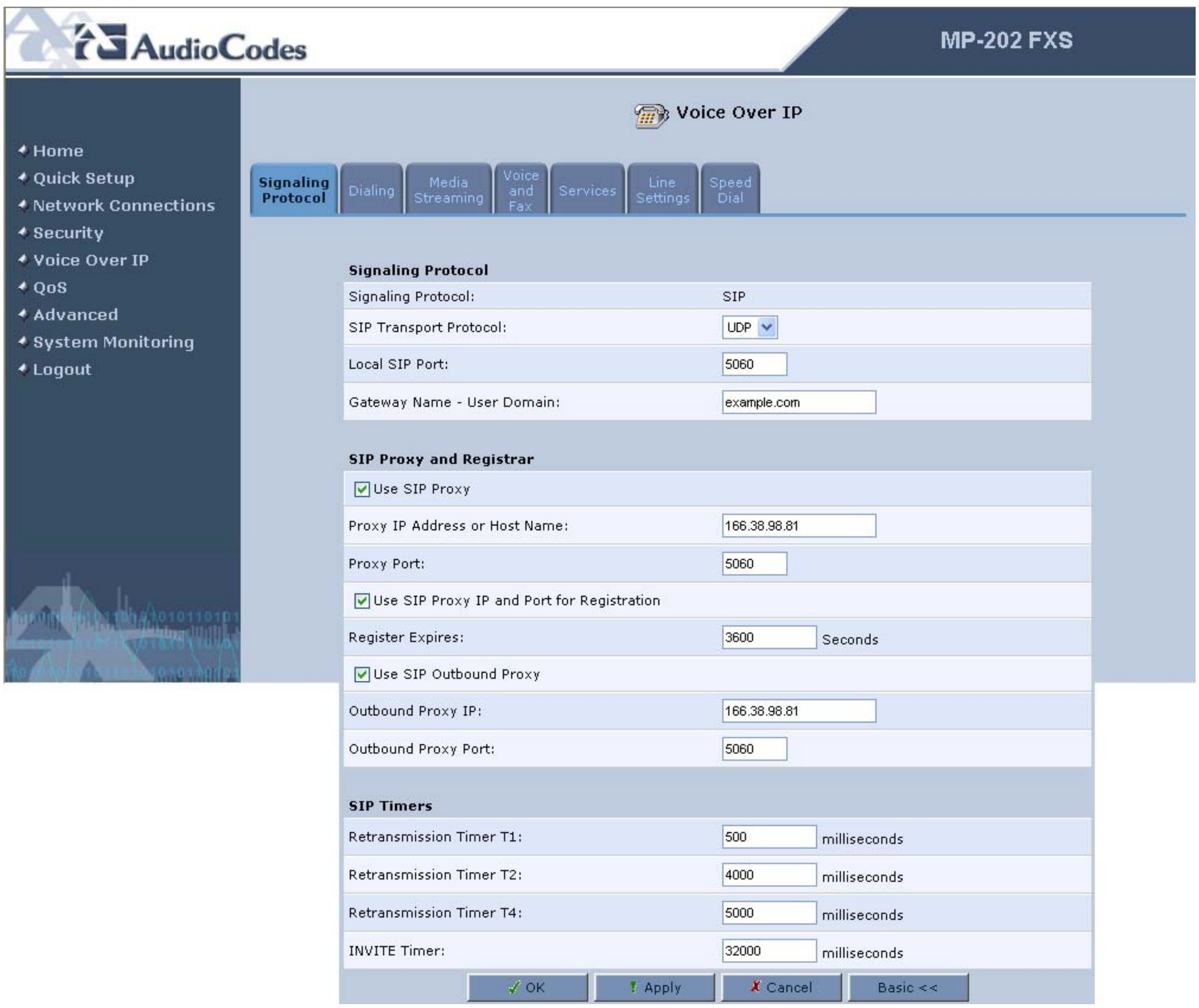
Appendix A. Fax and MP-202 Administration

- Make sure that the **Codec** G.711MU is present as the second or third choice.
- Make sure that **Direct Media** option is checked.
- The MP-202 has two FXS ports – connect the fax line to port #1.
- Connect the WAN Ethernet port of the MP-202 to one of the LAN ports on CMBE.
- Connect to the web-interface of the MP-202 (see the MP-202 manual [10])
    - Configure the CMBE Host IP address as the SIP Proxy
    - Configure the CMBE Host IP address as the Outbound Proxy on the '*use SIP outbound Proxy*' tab.



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- Open '*VoIP Parameters > Media Streaming*' tab. Configure only G.711MU.
    - *If you select the advanced button, you will also be able to configure the RFC2833 payload type used by Verizon.*

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- Open '*VoIP Parameters > Voice and Fax*' tab. Select '*transparent*' in the '*fax transport mode*'. Check the CNG detection checkbox.

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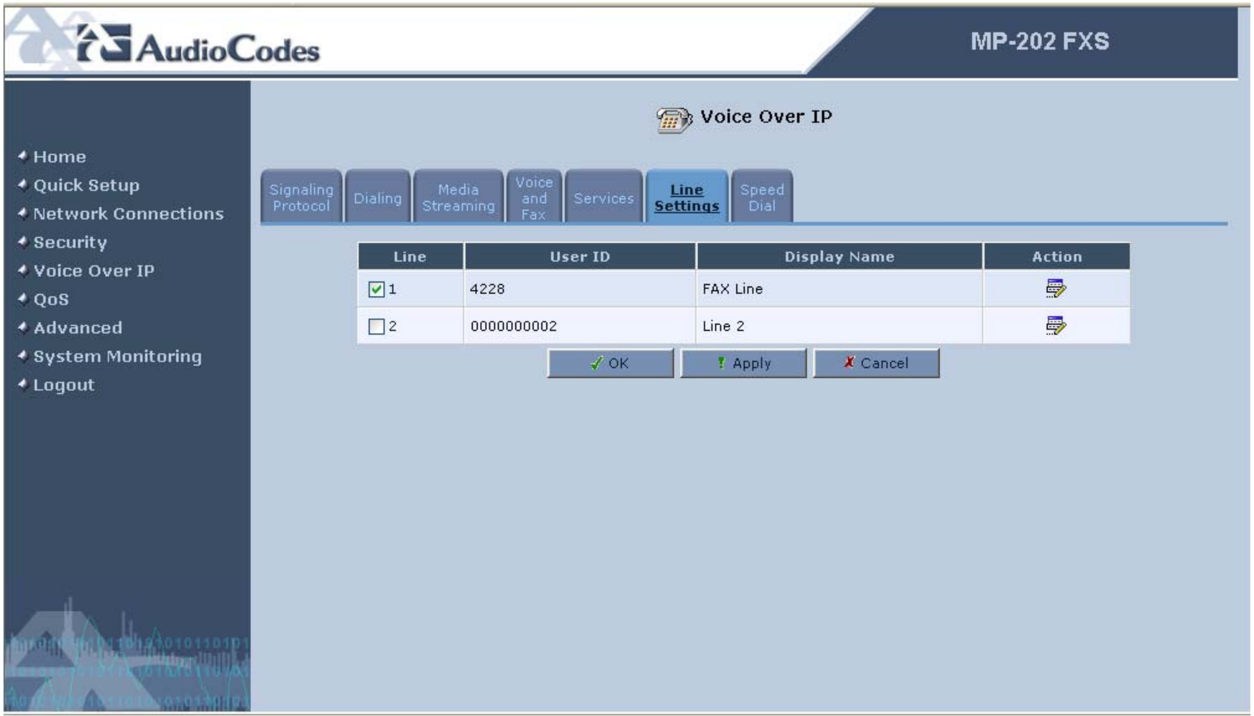
- Open ‘VoIP Parameters > Line Settings’ tab. Configure the SIP user parameters to match CMBE. Use user-id=authentication-name / CMBE extension. Configure the password the match the user extension.



- By default the WAN port is configured to work with DHCP. If you do not have a DHCP server on your LAN; you will need to administer an IP address for the MP-202.

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