Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between Cincinnati Bell eVantage and Avaya Aura™ Communication Manager Branch – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between Cincinnati Bell eVantage and Avaya Aura™ Communication Manager Branch (formerly known as Distributed Office) using various Avaya telephony endpoints.

The eVantage solution is a turn-key business trunking solution for customers. eVantage provides customers with a single IP connection that converges voice and data services to drive optimization, reduce costs, and offer enhanced features not typically available in the traditional PSTN network. Voice services, such as local, long distance, and toll free calling, as well a high speed data and Internet services, are the primary applications of the eVantage solution. The focus of these Application Notes is the configuration of SIP trunking to eVantage to provide voice access to the PSTN.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between Cincinnati Bell eVantage and Avaya Aura™ Communication Manager Branch (formerly known as Distributed Office) using various Avaya telephony endpoints.

The eVantage solution is a turn-key business trunking solution for customers. eVantage provides customers with a single IP connection that converges voice and data services to drive optimization, reduce costs, and offer enhanced features not typically available in the traditional PSTN network. Voice services, such as local, long distance, and toll free calling, as well a high speed data and Internet services, are the primary applications of the eVantage solution.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [7] 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 Communication Manager Branch and the network services offered by Cincinnati Bell.

1.1. Interoperability Compliance Testing

A simulated enterprise site with Communication Manager Branch supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP trunking solution provided by the eVantage solution. This allowed the enterprise site to use SIP trunking for calls to the PSTN.

The following features and functionality were covered during the SIP trunking interoperability compliance test:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Cincinnati Bell.
- Outgoing calls from the enterprise site were completed via eVantage to PSTN destinations.
- Calls using H.323, SIP, digital and analog endpoints supported by the Avaya IP Telephony network.
- Calls using Avaya IP Softphone.
- Various call types including: local, long distance, international, and toll free calls.
- Calls using the G.729a and G.711 µLAW codecs.
- DTMF tone transmission using RFC 2833 with successful call vectoring application.
- Telephone features such as hold, transfer, conference, and call forwarding.
- Interoperability with the Communication Manager Extension to Cellular (EC500) feature.

1.2. Support

For technical support on Cincinnati Bell eVantage, customers can call 1-866-914-9474.

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2. Reference Configuration

Figure 1 illustrates a typical customer location using Communication Manager Branch with SIP trunking to eVantage. This configuration includes:

- Communication Manager Branch 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 eVantage.
- A Cisco Unified Border Element (CUBE) that is located at the enterprise site but is owned and managed by Cincinnati Bell to provide troubleshooting capabilities up to the customer site. The configuration of the CUBE used for the compliance test is shown in Appendix A.

For security reasons, the middle two octets of all public IP addresses shown in these Application Notes have been replace with an "x". Any real routable PSTN numbers have also been changed to numbers that can not be routed by the PSTN.

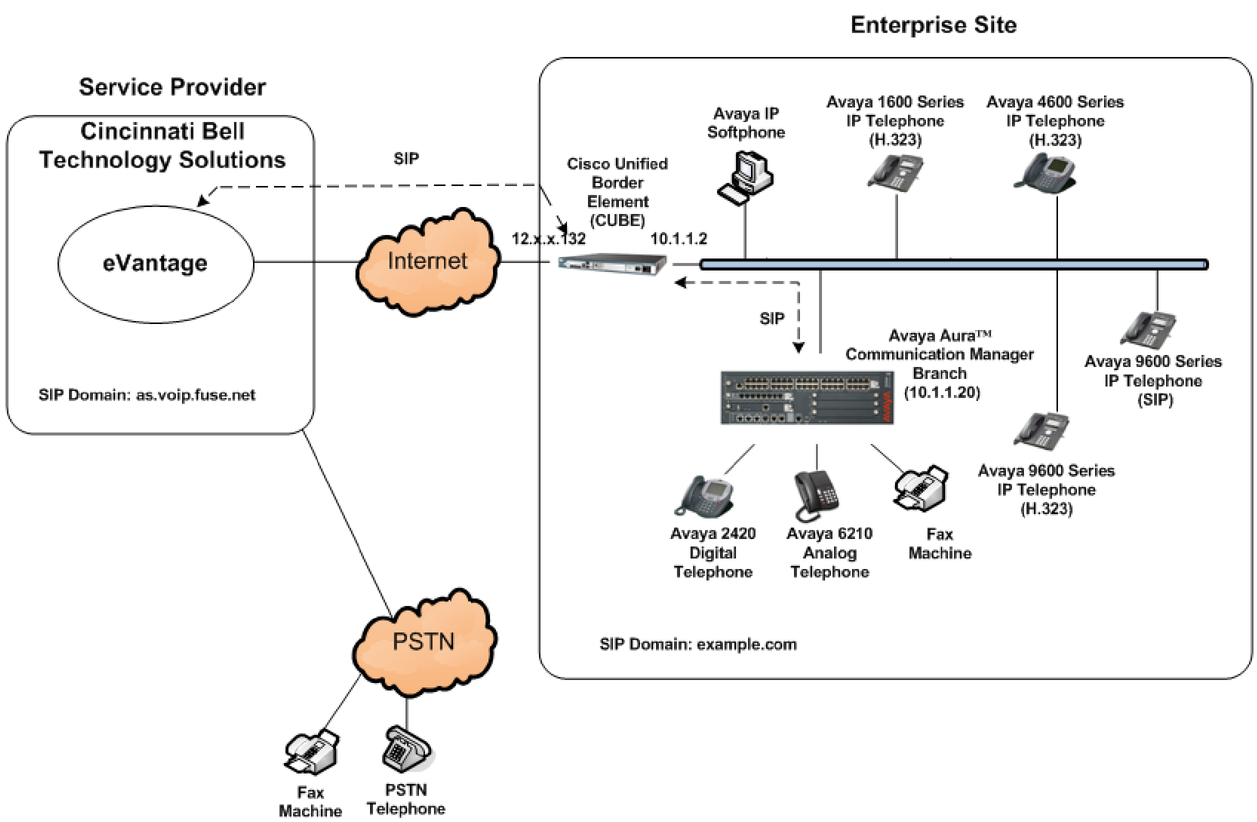

Figure 1 – Typical SIP Trunking Configuration

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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 Communication Manager Branch 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 Communication Manager Branch i120 and (the smaller) i40 configuration. However, the i40 is not specifically discussed.
- The use of analog or digital PSTN trunks in addition to SIP trunking is not discussed.
- The configuration of Avaya 9600, 4600, and 1600 Series IP telephones.

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

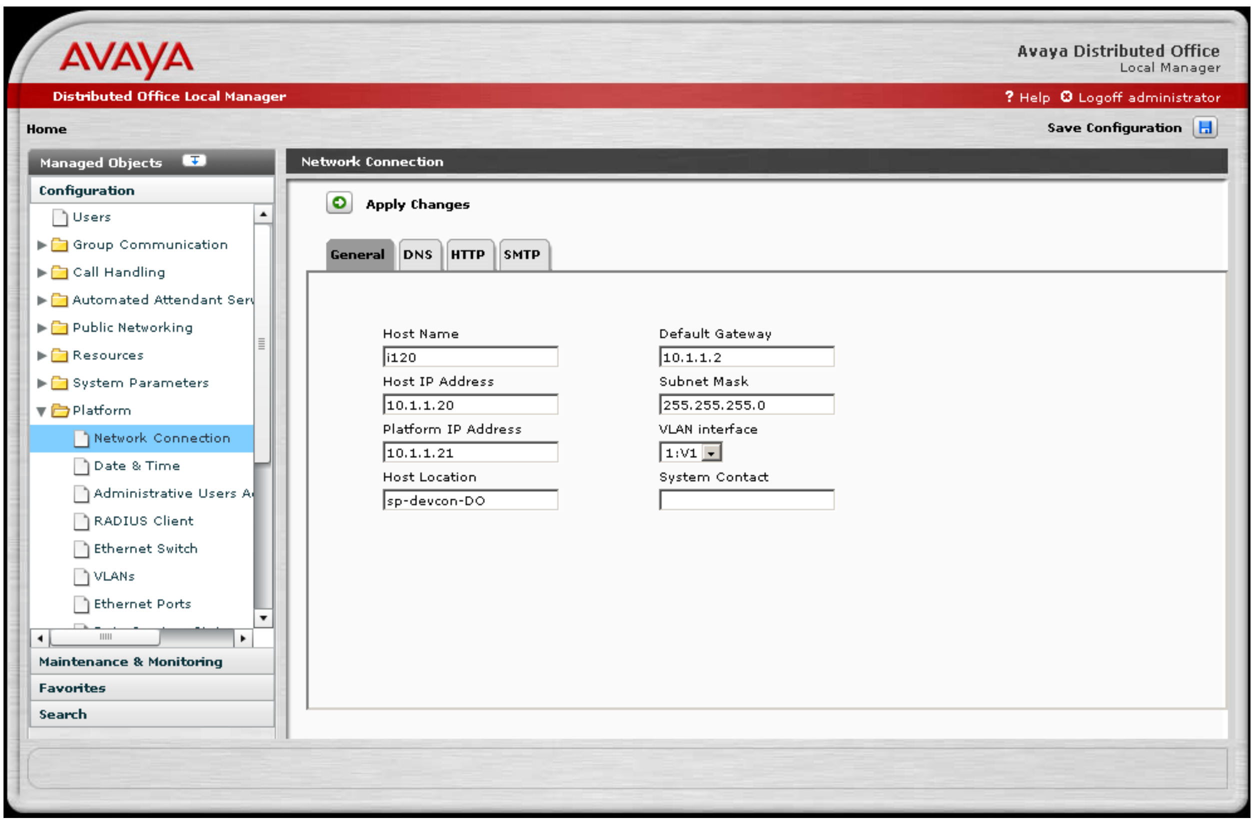

Figure 2 - Network Connection Assignments

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3. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya Aura™ Communication Manager Branch i120	Release 2.0 SP2 version 19 (2.0.0_28.01-SP-2.0.19)
Avaya 1600 Series Telephone	1.1 (H.323)
Avaya 4600 Series Telephone	2.9.1 (H.323)
Avaya 9600 Series Telephones	Avaya one-X™ Deskphone 3.0.2 (H.323) Avaya one-X™ Deskphone SIP 2.4.2 (SIP)
Avaya IP Softphone	R6.0 SP7 (H.323)
Avaya 6211 Analog Telephone	N/A
Fax Machine	N/A
Cincinnati Bell	
Cisco Unified Border Element (CUBE)	12.4 (24)T1

Table 1 – Equipment and Version

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4. Configure Communication Manager Branch

Communication Manager Branch 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.

4.1. Log in to Communication Manager Branch

Using a web browser, access the Communication Manager Branch Device Manager by entering “http://<ip-addr>/” where “<ip-addr>” is the **Host IP Address** of Communication Manager Branch as shown in **Figure 2**. Log in with the appropriate credentials. The Device Manager Home screen is shown.

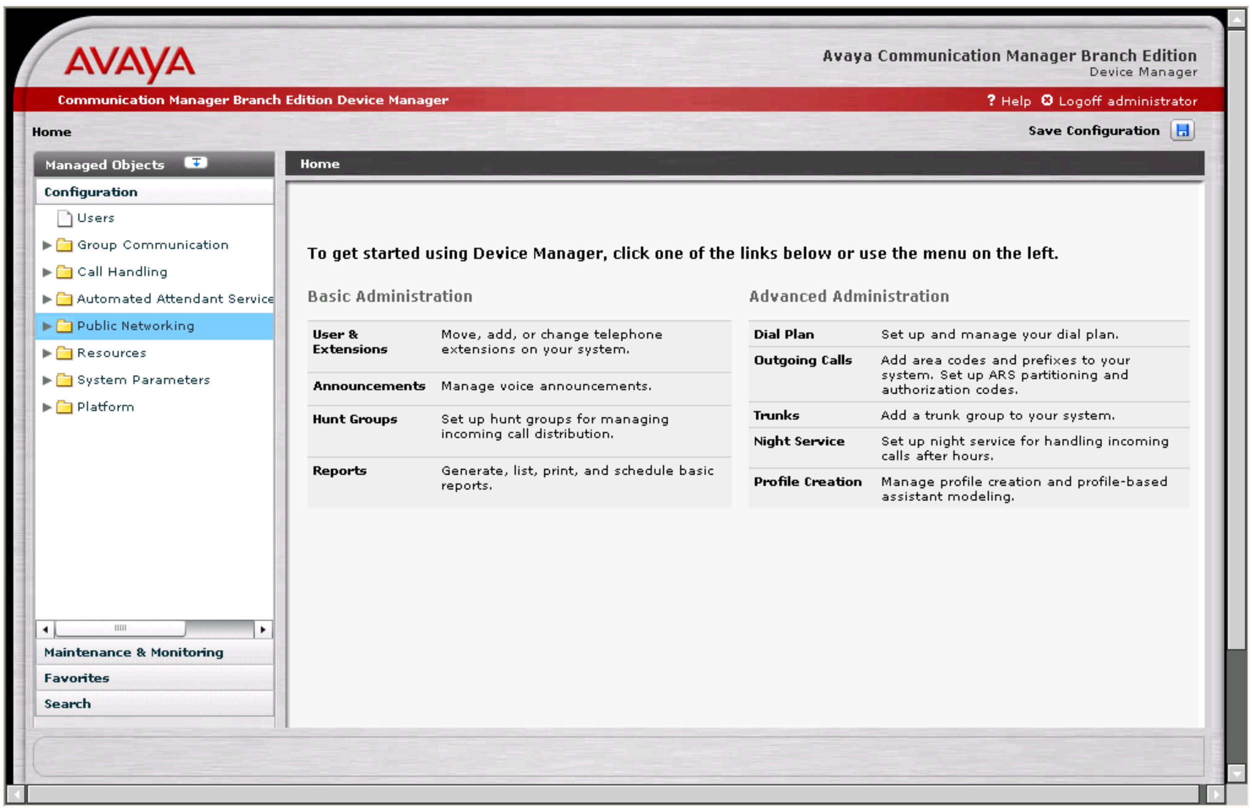

Figure 3 - Device Manager Home

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4.2. Configure a SIP Trunk Group

Configure a SIP Trunk that will connect to the Cincinnati Bell eVantage service via the on-site CUBE. 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.

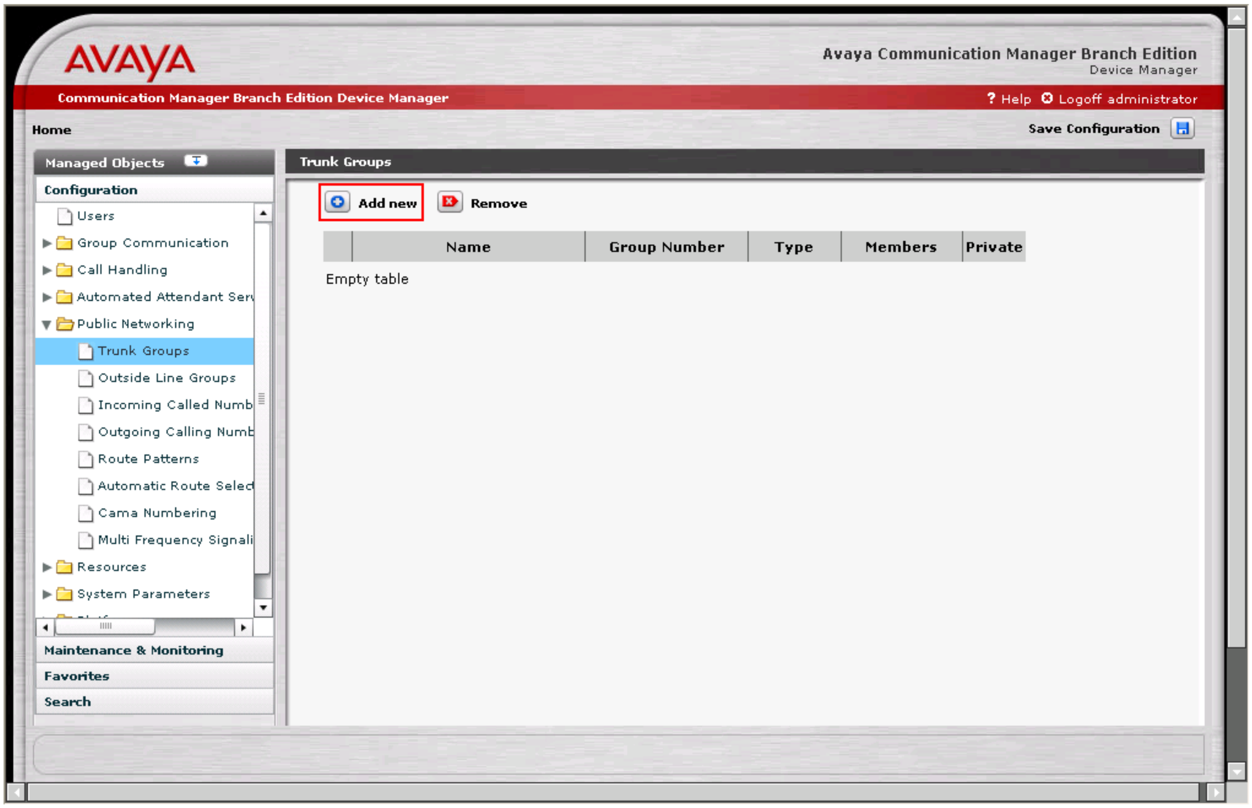

Figure 4 - Trunk Groups Screen

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On the **Add Trunk Group** screen:

- For the **Trunk Type**, select ***SIP*** from the pull-down menu.
- Enter a short text description of the trunk group 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.

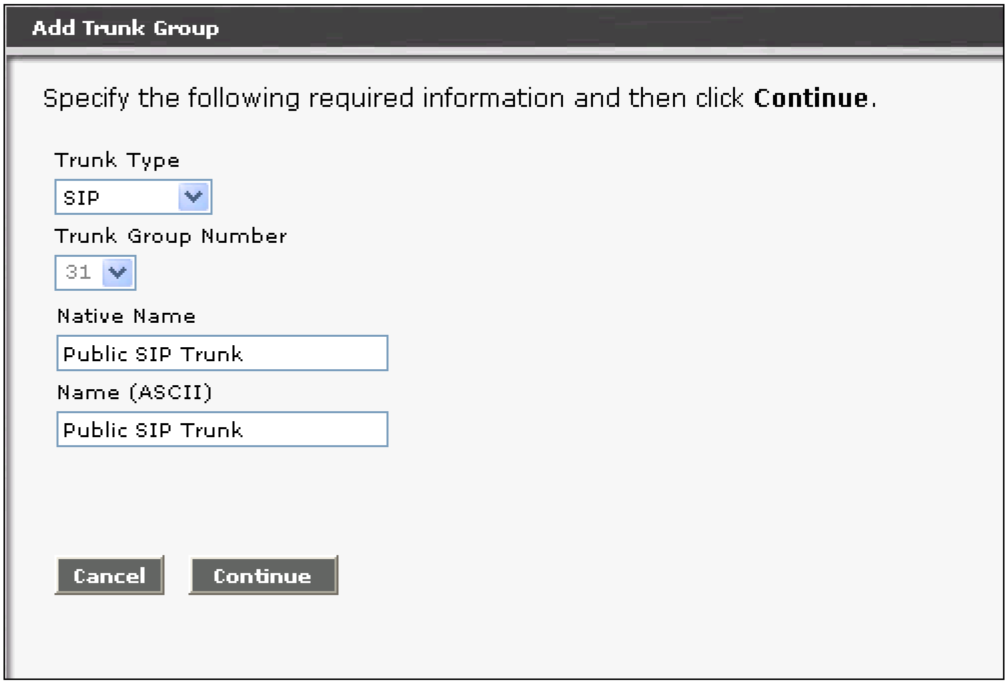

Figure 5 - Add Trunk Group Screen

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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.
- Default values may be used for all other fields.
- Press the **SIP** tab to advance to the next screen.

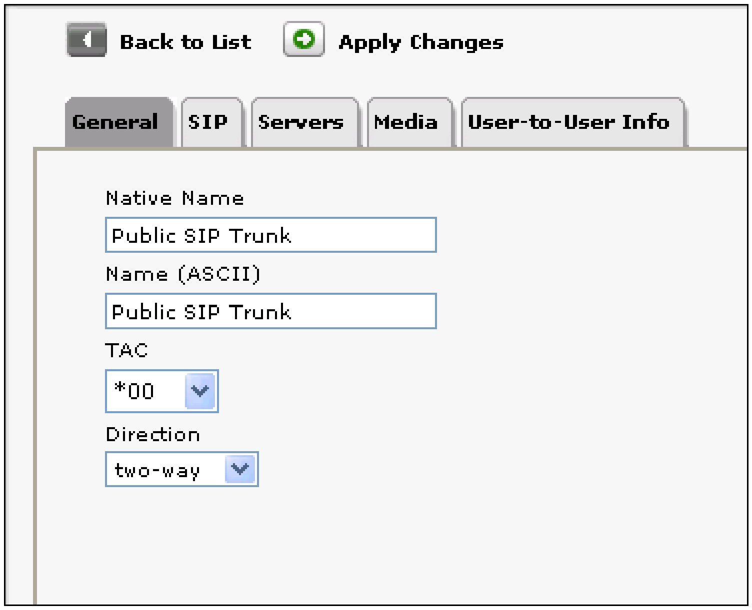

Figure 6 - Add SIP Trunk Group Screen – General tab

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On the **SIP** tab:

- Enter the **Far-End Domain** value supplied by the service provider.
- Enter the SIP domain of Communication Manager Branch in the **Near-End Domain** field. In these Application Notes, ***example.com*** was used.
- Check the box labeled **Replace outgoing request-URI domain with selected server IP address**.
- The **Session Refresh Interva**l default value of ***600*** is unacceptable to the eVantage service as being too small. A value of ***900*** was used for the compliance test.
- Default values can be used for all other fields.
- Press the **Servers** tab to advance to the next screen.

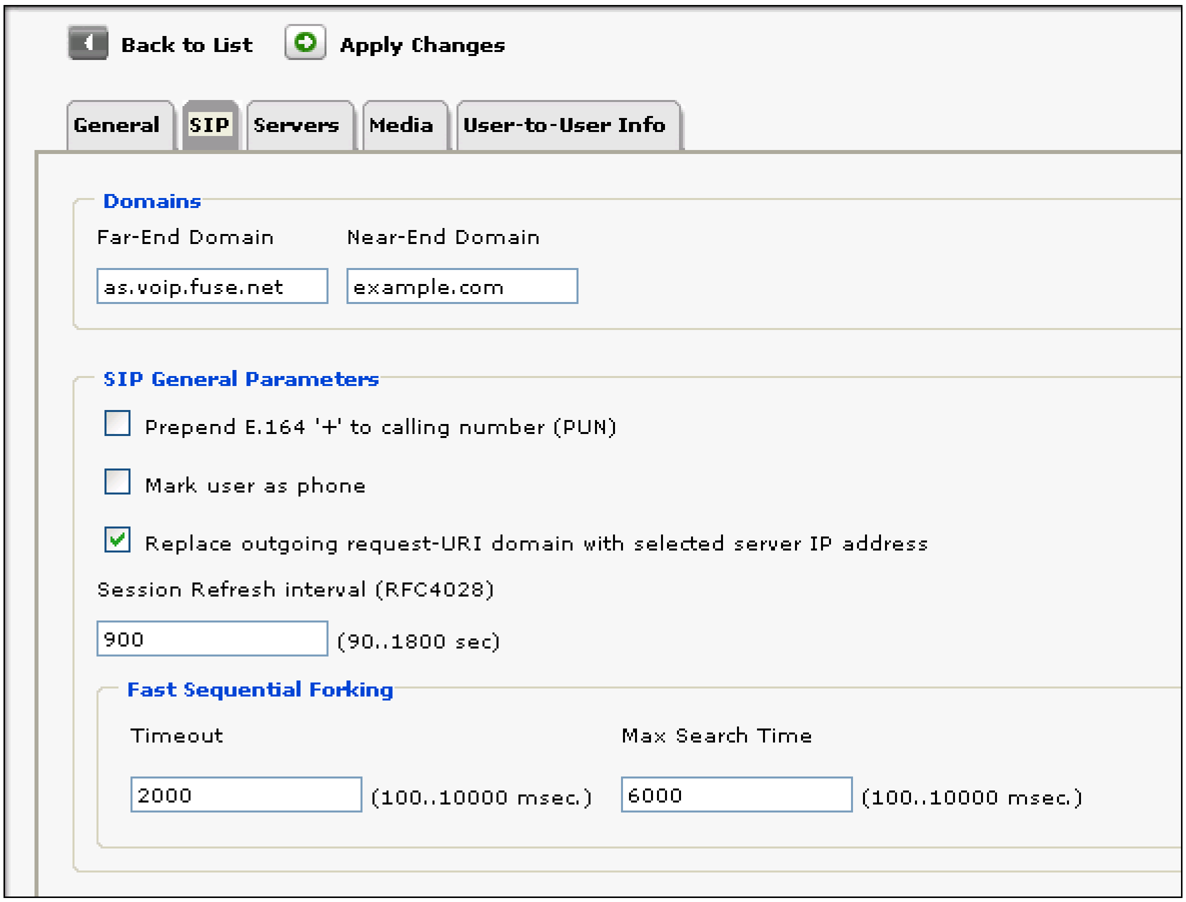

Figure 7 - Add SIP Trunk Group Screen – SIP Tab

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On the **Servers** tab:

- In the **Address** field, enter the private side IP address of the on-site CUBE which is owned and managed by Cincinnati Bell. In these Application Notes, ***10.1.1.2*** is used as noted in **Section 2**. It is not necessary to specify the port since the UDP default port of 5060 is used.
- Select ***UDP*** for the **Transport** field value.
- Default values may be used for all other fields.
- Press the **Media** tab to advance to the next screen.

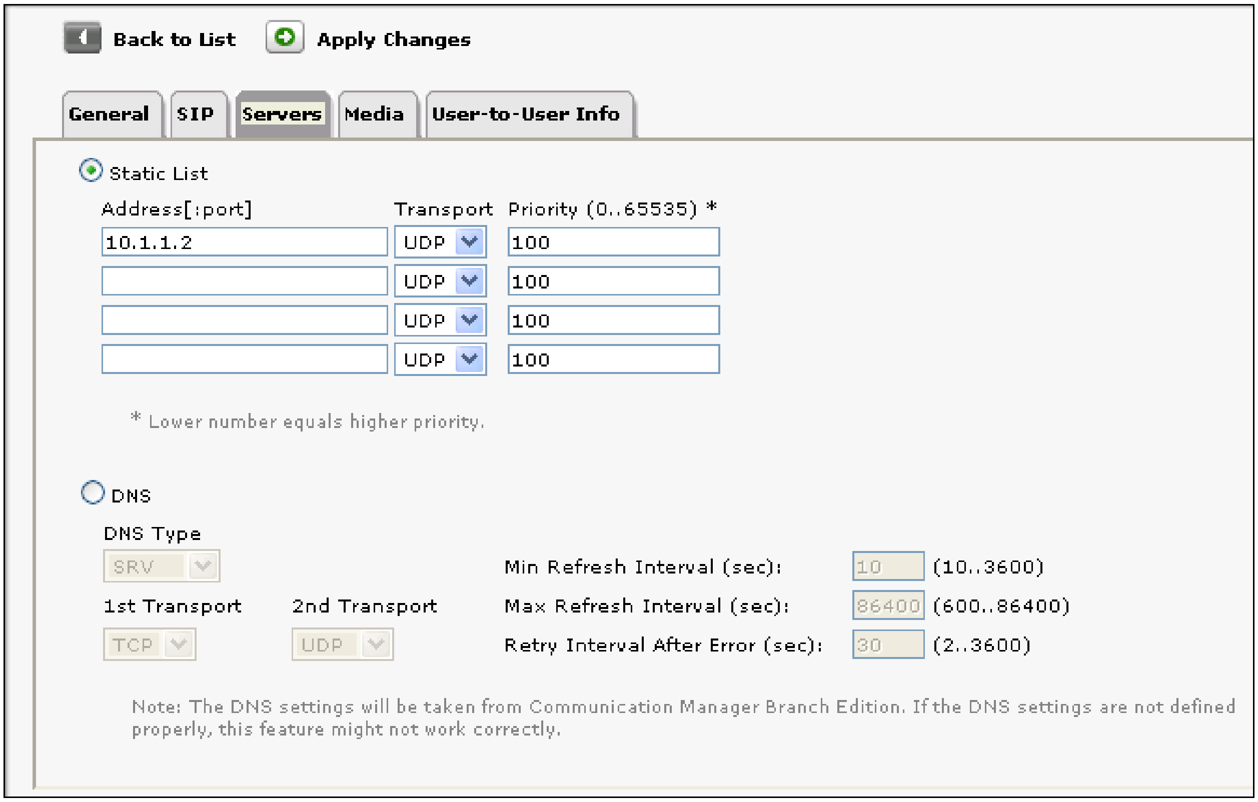

Figure 8 - Add SIP Trunk Group Screen – Servers Tab

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On the **Media** tab:

- Set the **Telephone Events RTP Payload Type** to ***101***. This is the recommended value to use with the Cincinnati Bell eVantage service. The default value used by the 9600 SIP Telephones should also be changed to 101[1] to minimize the possibility of using different DTMF values in each direction of the call as described in **Section 6**.
- Set the **Max Concurrent Calls** to the number of simultaneous calls supported. This value is specified by the customer when ordering the service. It is a function of the bandwidth of the VoIP network access, codec choices and service limits.
- Check the **Direct Media** option (to allow media paths to be routed directly to IP and SIP endpoints).
- Under the **Codec-Set**, enter the codecs in preferred order from 1 to 3. Select ***G.729a*** with ***2 (20ms)*** **Frames per packet** as the first **Codec** choice. Select ***G.711MU*** with ***2 (20ms)*** **Frames per packet** as the second **Codec** choice. Leave the third choice blank.
- Under the **Fax Parameters**, select ***off*** as the **Mode**.
- Default values may be used for all other fields.

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

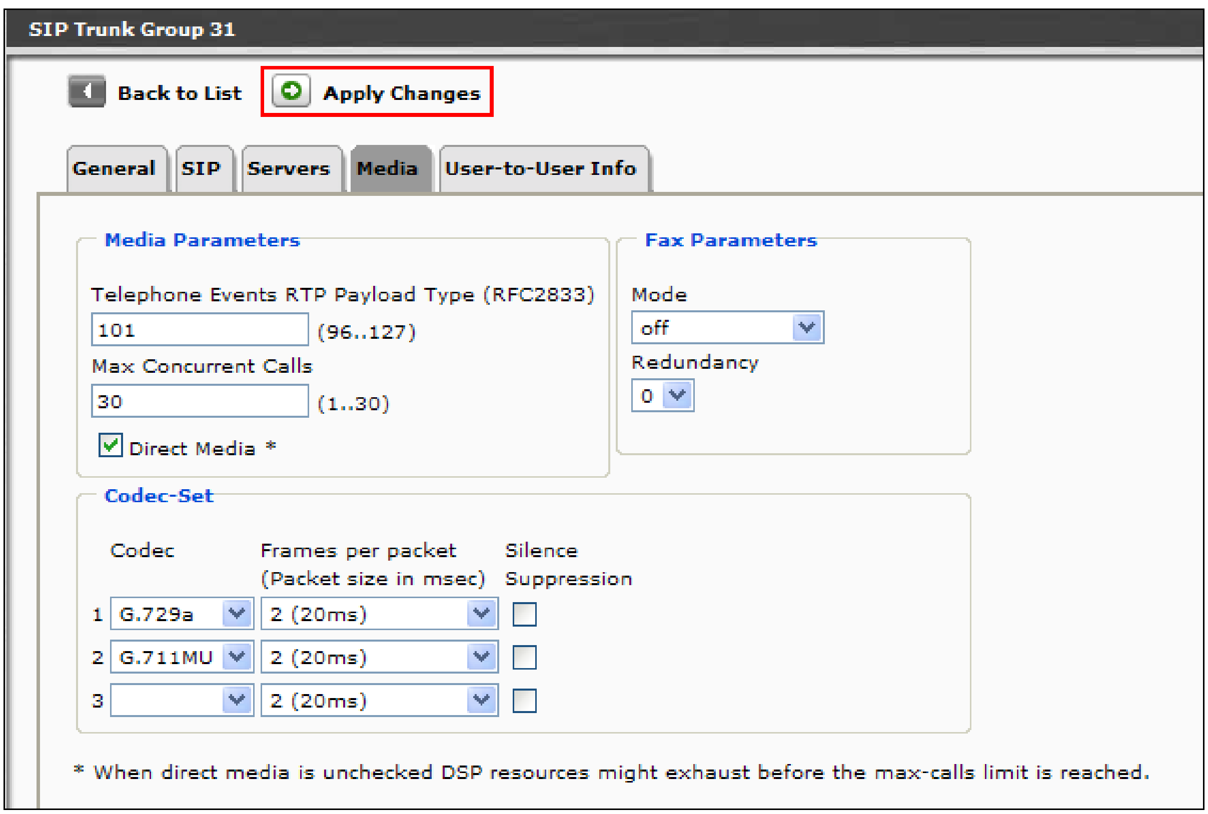

Figure 9 - Add SIP Trunk Group Screen – Media Tab

[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 how to set telephone options in the setting file are beyond the scope of these Application Notes but are found in [**6**].

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4.3. Configure Outgoing Calling Number

The following entries determine the calling number that will be sent in the SIP "From" header for the corresponding extensions.

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** screen.

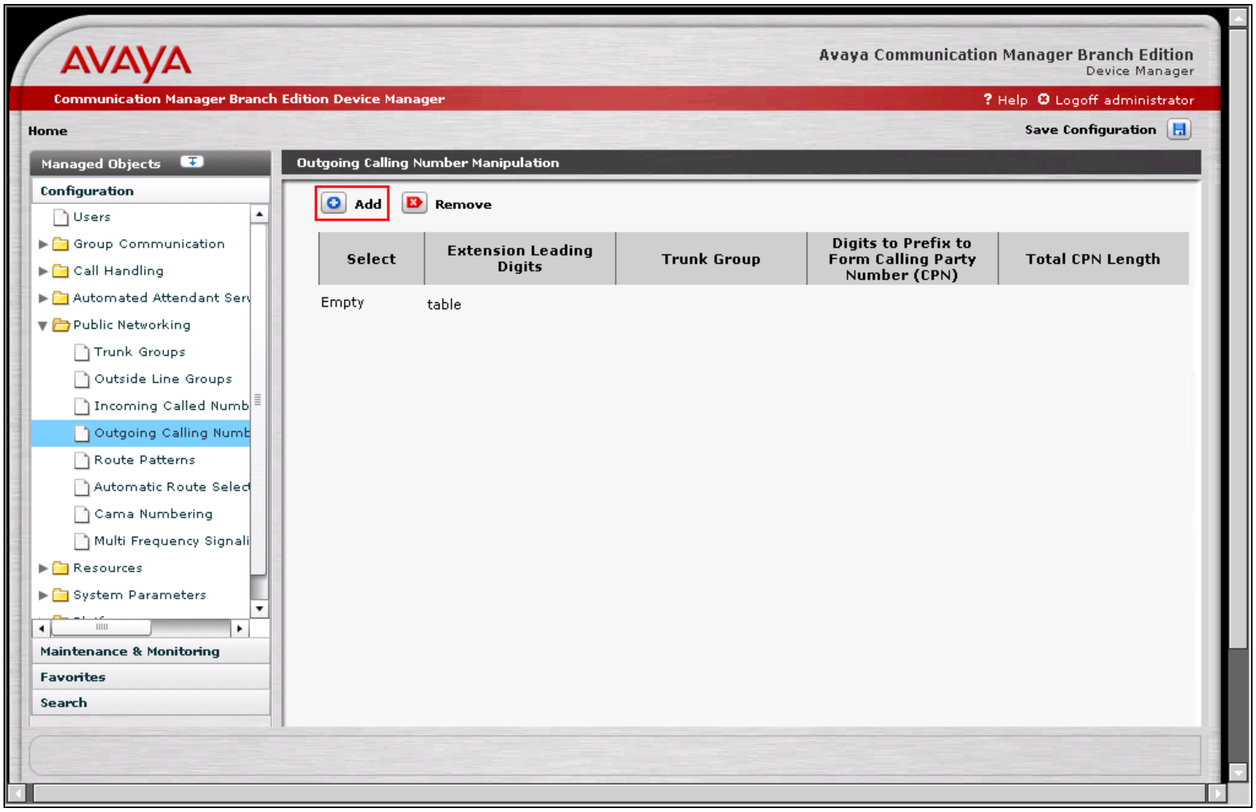

Figure 10 - Outgoing Calling Number Manipulation

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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, each extension number was configured to map to a unique DID number.
- Select the **Trunk Group** to which this rule applies.
- Enter the **Digits to Prefix to Form Calling Party Number**. In these Application Notes, the 10 digits of the DID number assigned to an extension was used as the calling party number for that extension. Thus, the complete 10 digit DID number is entered as the prefix.
- Enter the length of the calling party number in the **Total CPN Length** field. In these Application Notes, a length of ***10*** was used. Thus, the first 10 digits of the total number formed by the prefix + extension are sent as the calling party number.
- Press **Apply Changes** to record the entry and return to the **Outgoing Calling Number Manipulation** summary screen.

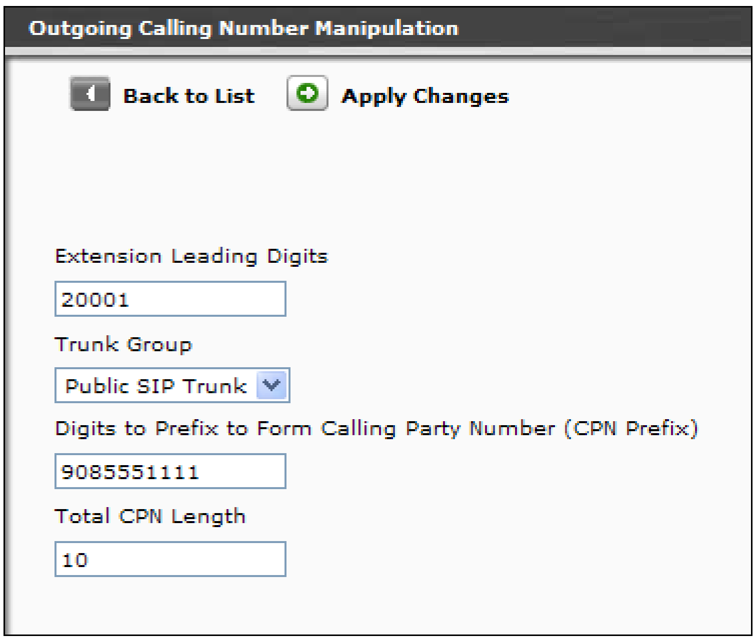

Figure 11 - Outgoing Calling Number Manipulation – New Entry

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Repeat this process to administer the calling party numbers for the remaining extensions. The **Outgoing Calling Number Manipulation** summary screen below shows entries created for extensions ***20000*** – ***20007***. The other entries, with **Extension Leading Digits** *2* – *7*, are default entries created by the system.

Outgoing Calling Number Manipulation

Add Remove

Select	Extension Leading Digits	Trunk Group	Digits to Prefix to Form Calling Party Number (CPN)	Total CPN Length
☐	**2**	all		5
☐	**3**	all		5
☐	**4**	all		5
☐	**5**	all		5
☐	**6**	all		5
☐	**7**	all		5
☐	**20000**	Public SIP Trunk	9085551110	10
☐	**20001**	Public SIP Trunk	9085551111	10
☐	**20002**	Public SIP Trunk	9085551112	10
☐	**20003**	Public SIP Trunk	9085551113	10
☐	**20004**	Public SIP Trunk	9085551114	10
☐	**20005**	Public SIP Trunk	9085551115	10
☐	**20006**	Public SIP Trunk	9085551116	10
☐	**20007**	Public SIP Trunk	9085551117	10

Figure 12 - Outgoing Calling Number Manipulation – Summary Screen

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4.4. Configure Call Routing - Outbound

The Automatic Route Selection (ARS) feature is used in these Application Notes to route outbound calls. ARS selects the proper route pattern to use to service a call based on the dialed digits of the destination. The route pattern defines the trunk to use and any digit manipulation required on the dialed string.

Begin the configuration of outbound call routing by creating a route pattern. 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.

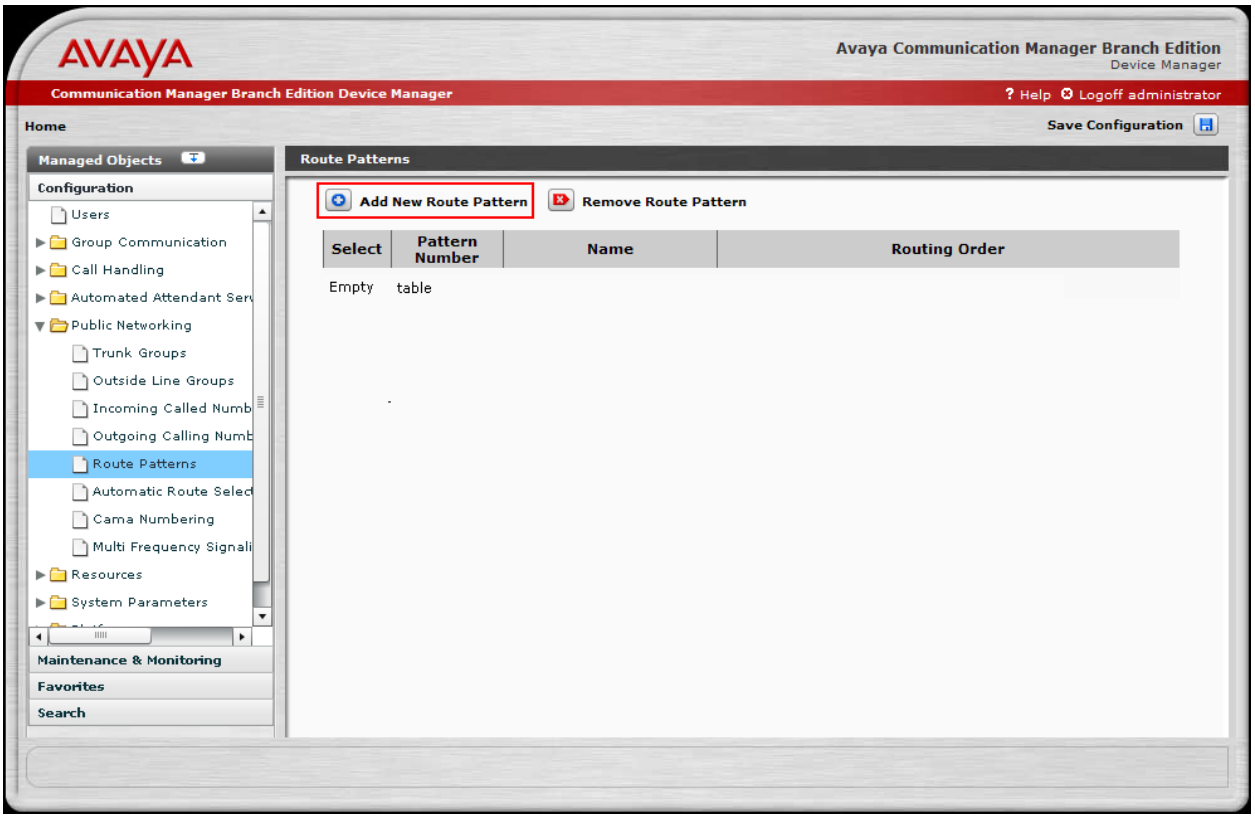

Figure 13 - Add New Route Patterns

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On the **Edit Route Pattern** screen,

- Select an available **Pattern Number**.
- Enter a short description for the **Pattern Name**.
- In the **Routes Selection** table, multiple trunks can be specified in order of priority. In these Application Notes, a single trunk was used. In the **Trunk Group** column of the first row, select the name of the SIP service provider trunk group defined in **Figure 5** from the pull-down menu. This defines the SIP service provider trunk group as the first (and only) choice in 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 service provider.
- Default values may be used for all other fields.
- Press **Apply Changes** to record the route pattern entry and return to the **Route Patterns** screen.

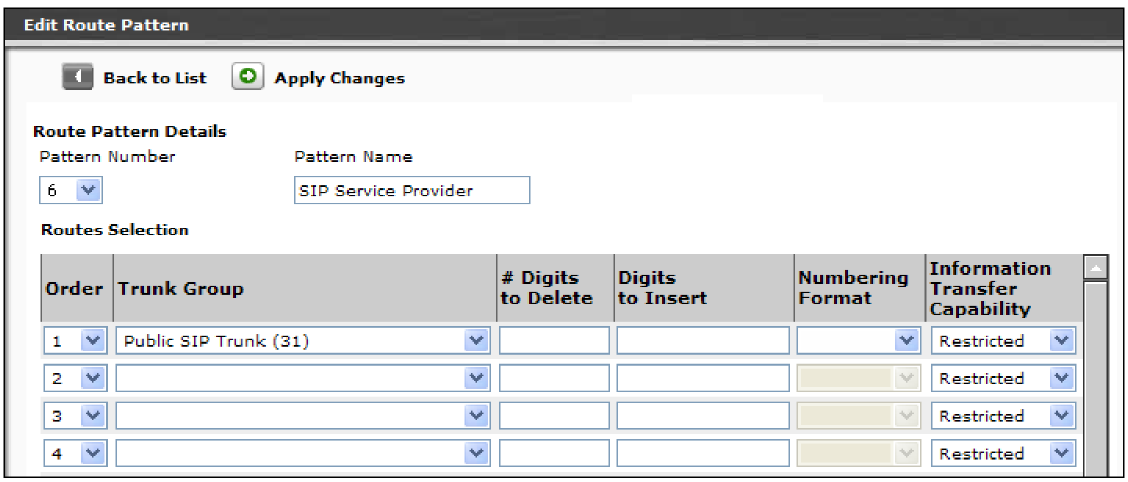

Figure 14 - New Route Pattern Screen

The **Route Patterns** screen is displayed.

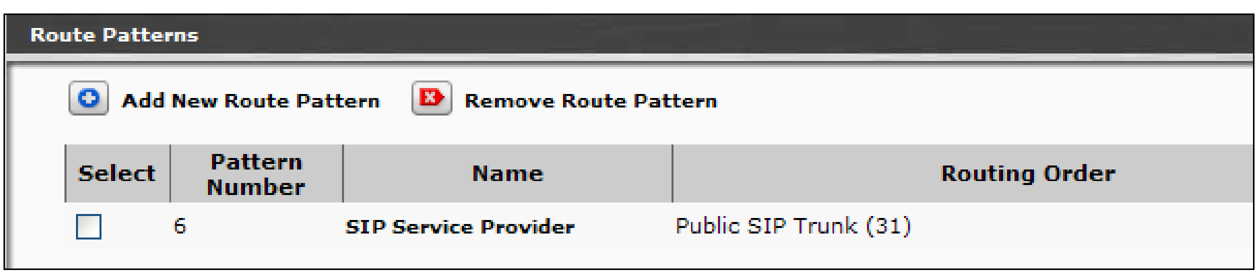

Figure 15 - Route Patterns – Summary Screen

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The next step in configuring outbound call routing is to map dialing patterns to the corresponding route patterns and call routing privileges using ARS.

To access the ARS table, use the **Configuration** menu on the left side of the window to expand the **Public Networking** option and select **Automatic Route Selection**. The **Public Network Automatic Route Selection** screen is displayed. The entries shown below are the entries used for the compliance test. Entries are added to the table by clicking the **Add New** link at the top of the table. This will allow data to be added at the end of the table for a new entry.

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**). As an example, the third entry in the table will route any 11 digit number beginning with a 1 to the **Route Pattern** called ***SIP Service Provider***.

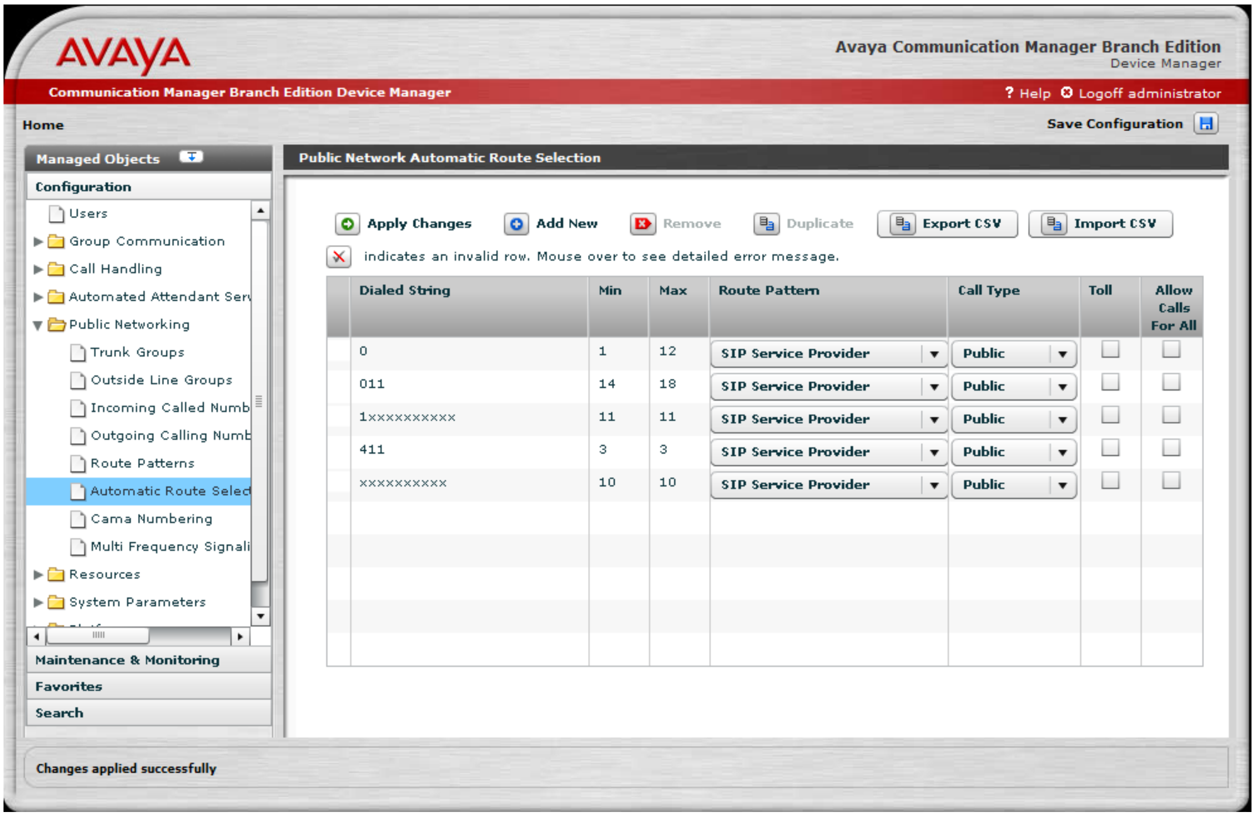

Figure 16 - Public Network Automatic Route Selection

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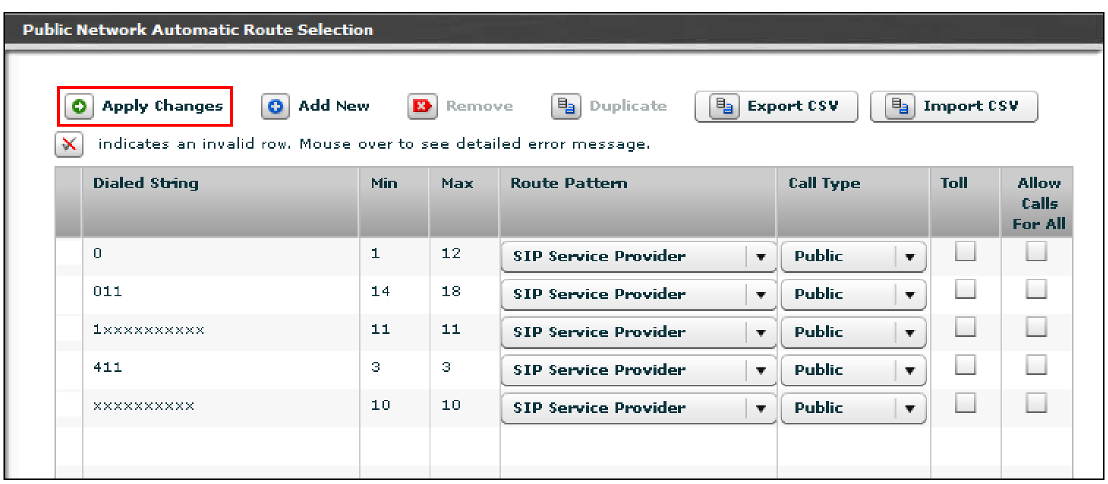
Each of the columns in the table is described below:

- **Dialed String**: A predefined string to be matched by user-dialed numbers. An ***x*** character can be used as a wild card to represent any digit. In the example above, ***1xxxxxxxxxx*** will match a dialed string beginning with a 1 followed by any 10 digits.
- **Min**: The minimum number of user-dialed digits to collect in order to match the dialed string. In our example, this field is set to ***11***.
- **Max**: The maximum number of user-dialed digits to collect in order to match the dialed string. In our example, this field is set to ***11***. This combined with the previous **Min** value means the **Dialed String** matching pattern is limited to only 11 digit numbers for our example.
- **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. In the compliance test, all matching patterns were mapped to the route pattern defined for the SIP service provider.
- **Call Type**: The type of call that will be placed. Choices include ***Deny***, ***Local***, ***Public***, ***Emergency*** and ***Crisis-Alert***. In our example, the call type is set to ***Public***.
- **Toll**: Specifies the extension's privilege level necessary to place the call. Only extensions having "admin" and "high" privileges are able to place toll calls. In our example, the **Toll** box is unchecked to allow extensions with low, medium, high and administrative user privilege levels to place these calls. (Note: the user privilege level is assigned to an extension during user administration and beyond the scope of these Application Notes.)
- **Allow Calls for All**: Specifies that any phone may place a call for this dialed pattern. In our example, not checking the **Allow Calls for All** will prevent extensions with no privileges from being able to place these calls.

After completing any ARS entries/changes, press **Apply Changes** to save the data.



Dialed String	Min	Max	Route Pattern	Call Type	Toll	Allow Calls For All
0	1	12	SIP Service Provider	Public		
011	14	18	SIP Service Provider	Public		
1xxxxxxxxxx	11	11	SIP Service Provider	Public		
411	3	3	SIP Service Provider	Public		
xxxxxxxxxx	10	10	SIP Service Provider	Public		

Figure 17 - Public Network Automatic Route Selection – Summary Screen

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4.5. Configure Call Routing - Inbound

This step configures the routing of incoming DID calls to the associated Communication Manager Branch extensions. In these Application Notes, the incoming PSTN DID numbers listed in **Figure 12** are assigned to the extensions as shown in
Table 2.

Dialed PSTN Number	Digits Received (within SIP INVITE message)	Extension Assigned
1-908-555-1110	9085551110	20000
1-908-555-1111	9085551111	20001
1-908-555-1112	9085551112	20002
…	…	…
1-908-555-1117	9085551117	20007

Table 2 - Incoming DID Number Assignments

Begin the incoming DID assignments 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.

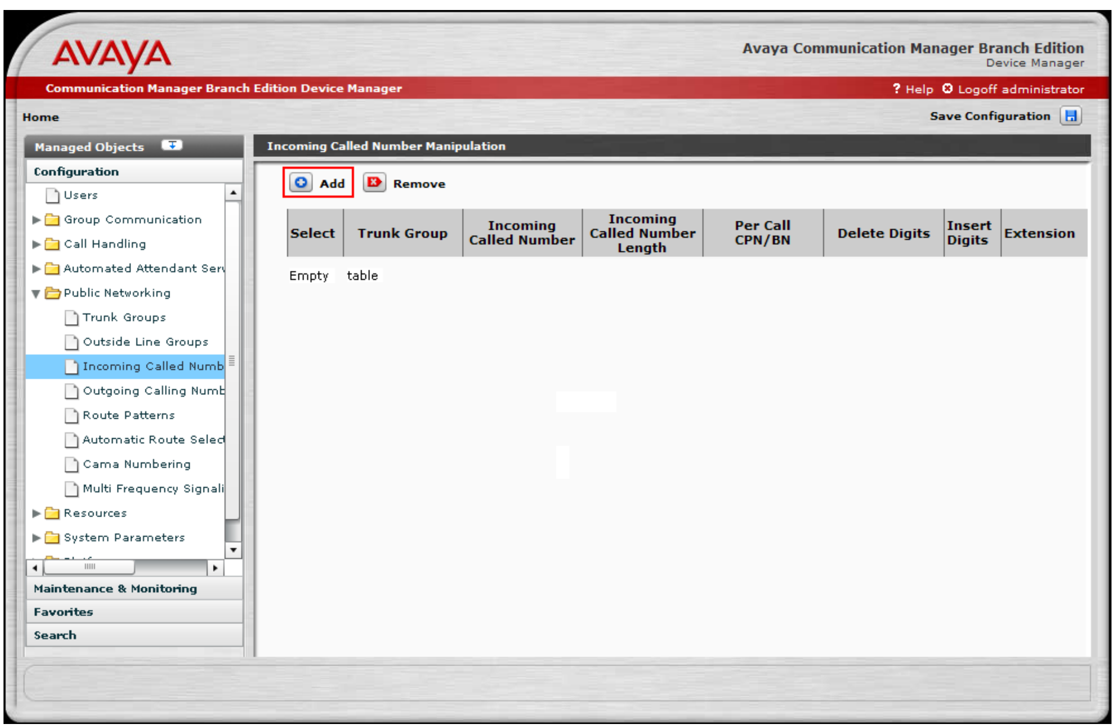

Figure 18 - Incoming Called Number Manipulation

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From the **Add Incoming Called Number Manipulation** screen, enter the following to administer the assignments for the DID numbers:

- For the T**runk Group**, select the SIP service provider trunk group defined in **Section 4.2**.
- For the **Called Number,** enter the digit pattern to be matched.
- Enter ***10*** as the **Called Number Length**. This is the total number of digits sent by the service provider.
- Click the **Extension** button and from the pull-down menu select the extension to map to the called number.
- Press **Apply Changes** to record the information entered and redisplay the **Incoming Called Number Manipulation** screen.

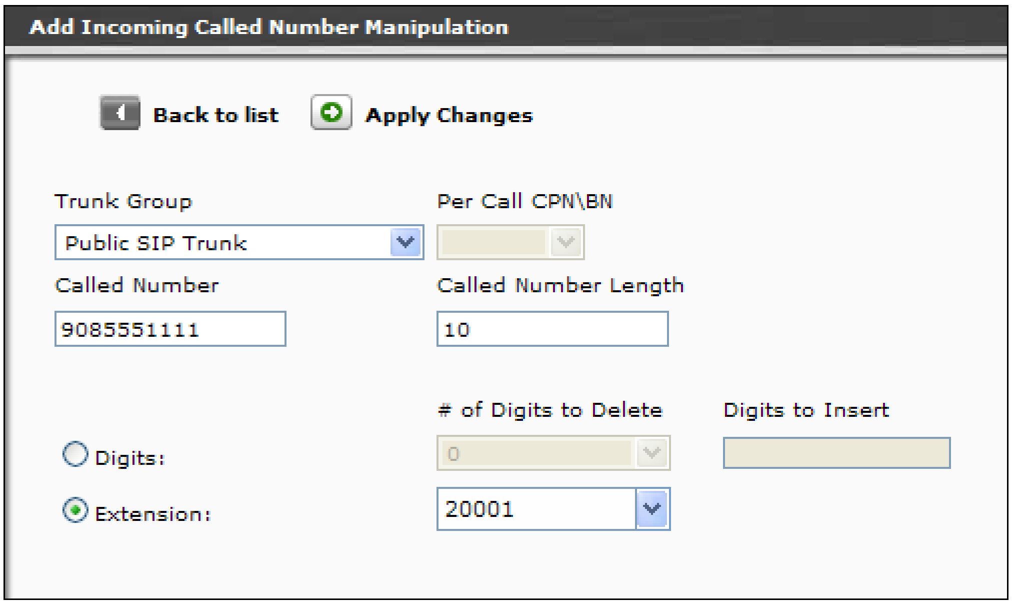

Figure 19 - Add Incoming Called Number Manipulation

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Repeat the **Add Incoming Called Number Manipulation** process to administer the mapping for the other DID numbers. After the **Apply Changes** is performed, the resulting **Incoming Called Number Manipulation** screen is shown.

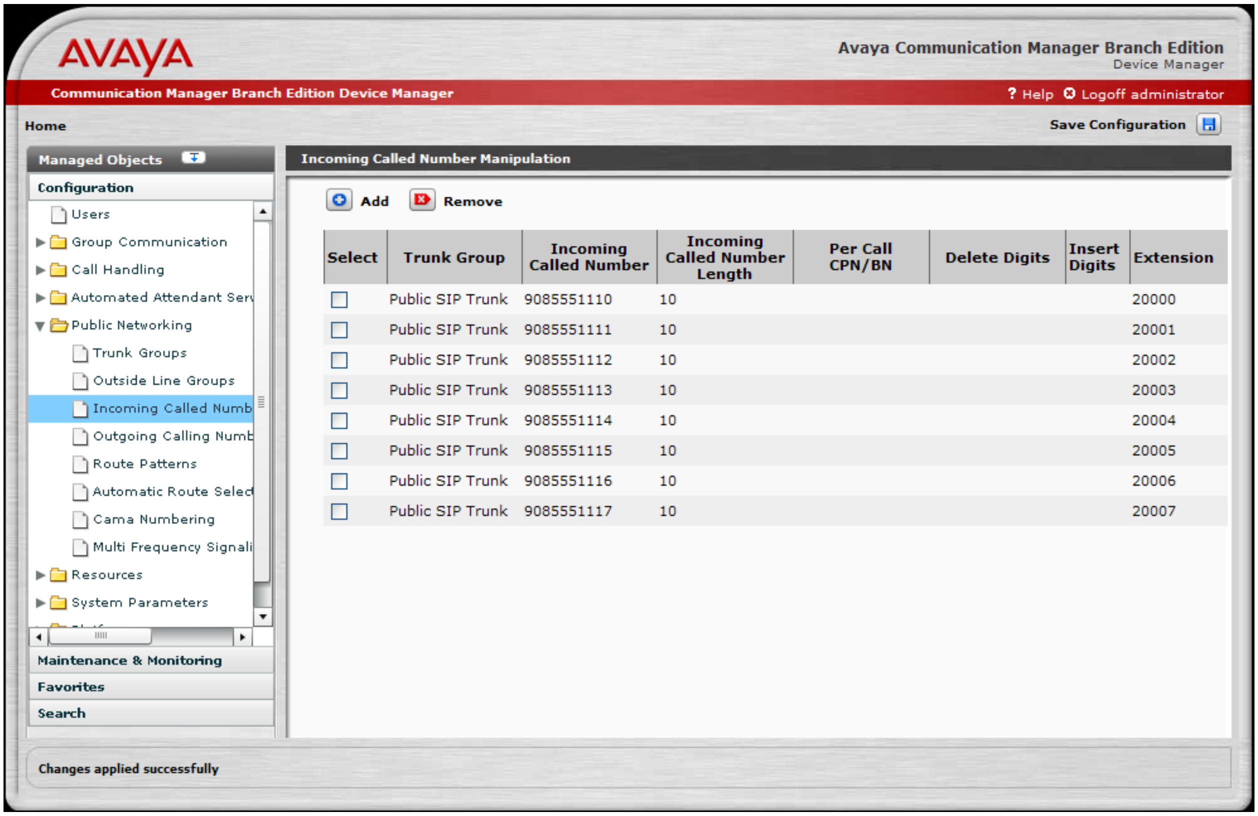

Figure 20 - Incoming Called Number Manipulation – Summary Screen

4.6. Save Communication Manager Branch Configuration

The configuration of the Communication Manager Branch SIP trunking is now complete. Save the Communication Manager Branch 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.

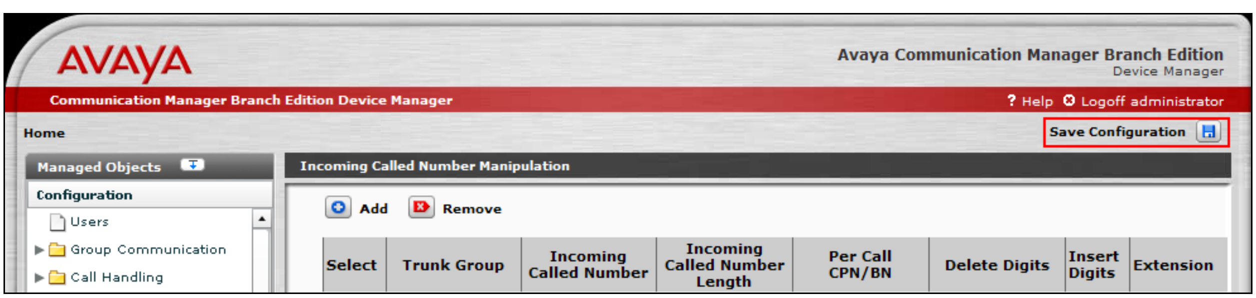

Figure 21 - Save Configuration

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5. Configure the Cincinnati Bell eVantage Service

To use Cincinnati Bell eVantage Service, a customer must request service from Cincinnati Bell using their sales processes. The process can be started by contacting Cincinnati Bell via the corporate web site and requesting information via the online sales links or telephone numbers or contacting a Cincinnati Bell sales representative.

During the signup process, Cincinnati Bell will require that the customer provide the IP address used to reach the Communication Manager Branch server as well as IP addresses to assign to the public and private sides of the on-site CUBE. Cincinnati Bell will perform all configurations on the CUBE (including the IP address of the Cincinnati Bell SIP proxy/SBC) and will provide the eVantage service SIP domain and Direct Inward Dialed (DID) numbers to the customer. This information is used to complete the Communication Manager Branch configuration discussed in the previous sections.

6. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the Cincinnati Bell eVantage Service and Communication Manager Branch. This section covers the general test approach and the test results.

Communication Manager Branch i120 was connected using SIP trunking (via the Internet) to the Cincinnati Bell eVantage Service. The general test approach included the following:

- 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 Communication Manager Branch voice mail menus.

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

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

The following compatibility issues were observed during testing:

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- **0+ Dialing**: eVantage does not support 0+ dialing (0 + PSTN number). Users will hear a recorded message that the number dialed is out of service. Users may dial 0 to reach an automated attendant or operator to place the call.
- **T.38 Fax**: eVantage does not support T.38 Fax capabilities across the SIP trunk to the enterprise.
- **Unable to remotely access voicemail using G711mu codec**: When a user attempts to place a an inbound call from the PSTN via the SIP trunk to the Communication Manager Branch voicemail number to retrieve messages, the user is unable to login to his mailbox. The direct cause for this behavior is that the voicemail application is not detecting DTMF events sent to it from the eVantage service. This problem occurs when using the G.711 codec. When using this codec, the call gets shuffled directly to the voicemail application to collect the digits. The voicemail application attempts to renegotiate the DTMF payload values used for the call to be different values in each direction of the call. The eVantage service does not support using different DTMF payload values in each direction. When using the G.729 codec, the call is not shuffled and the digits are collected by the media gateway and the same DTMF payload value is used in both directions of the call. Possible workarounds for this problem would be to use G.729a on the SIP trunk or disable shuffling. However, either one of these changes would apply to all calls to/from the eVantage service and not just inbound calls to voicemail.
- **Extension to Cellular (EC500):** The EC500 feature applies to a user who can be reached at their Avaya desk phone or a cellular phone over the PSTN by dialing a single DID number. When a call is made to this DID number from the PSTN, the desk phone and cellular phone should ring simultaneously allowing the user to answer the call on either phone depending on their location. However, in this configuration, when an incoming PSTN call arrives to an Avaya desk phone with EC500 enabled, the outgoing EC500 call to the user's cellular phone over the PSTN is denied by the eVantage service. The outgoing call is denied because Avaya sends out the calling number of the PSTN user, which is unknown to the eVantage service and can't be authenticated. In this case, only the Avaya desk phone will ring since the outgoing EC500 call was denied. If the call originates from a local Avaya telephone (which has a DID number assigned to it), this issue does not occur because the eVantage service can authenticate the local Avaya user via the DID number. A possible workaround for this issue is for Cincinnati Bell to modify the CUBE configuration to overwrite the originating caller ID of all outbound calls to a number that can be authenticated by the eVantage service, such as the main enterprise telephone number. The downside of this approach is that the actual DID number of the originating caller is not sent to the far-end for any outbound call. The caller ID would always show the main enterprise telephone number. This workaround was not tested as part of the compliance test.
- **Call Forwarding Off-Net:** This issue is similar to the EC500 issue described above in that an incoming PSTN call delivered to an Avaya station with Call Forwarding enabled to an off-net PSTN phone will be denied by the eVantage service because it won't be able to authenticate the calling number of the PSTN user sent by Avaya. In this case, the call will not be forwarded and the PSTN caller will hear "busy" tone. If the call originates from a local Avaya telephone (which has a DID number assigned to it), this issue does not occur because the eVantage service can authenticate the local Avaya user. A possible workaround for this issue is the same workaround discussed above for EC500, by

modifying the CUBE configuration to overwrite the originating caller ID of all outbound calls to a number that can be authenticated by the eVantage service.

- **Outbound Calling Party Number (CPN) Block:** Outbound calls with CPN block enabled will not complete. With CPN block enabled, the calling number in the outbound INVITE is not present in the SIP "From" header but instead is placed in the P-Asserted-Identity (PAI) header. eVantage requires the number in the From header for authentication and does not support using PAI for this purpose.
- **IP Softphone Telecommuter Mode**. If using IP Softphone in telecommuter mode, PSTN calls can not be transferred or conferenced. These calls will fail. Communication Manager Branch does not send an ACK to the eVantage service in response to a 200 OK message on the transferred/conferenced leg of the call.

7. Verification Steps

7.1. Verification Tests

Configuration verification was performed with use of the **ping** command to confirm network connectivity between Communication Manager Branch and the Cincinnati Bell VoIP network. Once verified, an initial incoming and outgoing call were completed prior to testing and reviewed with the use of a SIP protocol analyzer.

7.2. Troubleshooting Tools

Communication Manager Branch has several troubleshooting tools that can be helpful to diagnosis 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** – 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 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 service provider VoIP 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 the SIP service provider and Communication Manager Branch. This can be extremely valuable to support advanced troubleshooting.

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8. Conclusion

These Application Notes describe the steps for configuring SIP trunking between an Avaya Aura™ Communication Manager Branch and Cincinnati Bell eVantage Service.

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.

9. Additional References

Communication Manager Branch product documentation is available at http://support.avaya.com.

[1] Overview of Avaya Aura™ Communication Manager Branch, December 2009, 03-602024.
[2] Avaya Aura™ Communication Manager Branch i120 Installation Quick Start, December 2009, 03-602289.
[3] Avaya Aura™ Communication Manager Branch i40 Installation Quick Start, December 2009, 03-602288.
[4] Feature Description for Avaya Aura™ Communication Manager Branch, December 2009, 03-602027.
[5] Avaya Application Solutions: IP Telephony Deployment Guide, Issue 6, January 2008, 555-245-600.
[6] Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide, November 2009, 16-300698.

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.

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

Documentation on the eVantage service is available from Cincinnati Bell.

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APPENDIX A: CUBE Configuration

Below is a sample configuration used in **Figure 1**. It is included here only as a reference without explanation.

For security reasons, the middle two octets of all public IP addresses shown below have been replace with an "x". Any real routable PSTN numbers have also been changed to numbers that can not be routed by the PSTN. In addition, any router login usernames and passwords have been deleted from the configuration shown below.

Contents of the CUBE configuration file:

```
Current configuration : 3363 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname DevConnectCUBE
!
boot-start-marker
boot system flash:c2801-adventerprisek9_ivs-mz.124-24.T1.bin
boot-end-marker
!
logging message-counter syslog
logging queue-limit 150000
logging buffered 15000000
no logging console
enable password XXXXXXXX
!
no aaa new-model
clock timezone EST -5
clock summer-time EST recurring
ip source-route
!
!
ip cef
ip name-server 12.x.x.67
no ipv6 cef
!
multilink bundle-name authenticated
!
!
no voice call carrier capacity active
!
voice service voip
 address-hiding
 allow-connections sip to sip
 fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback pass-through
g711ulaw
 sip
```


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```
  localhost dns:as.voip.fuse.net
   outbound-proxy dns:edge.voip.fuse.net
!
!
voice class codec 99
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
!
voice-card 0
 dsp services dspfarm
!
!
archive
 log config
  hidekeys
!
!
ip tcp synwait-time 5
!
class-map match-all class11
 description --- match VoIP RTP ---
 match access-group 111
 class-map match-all class10
 description --- match VoIP signaling ---
 match access-group 110
!
!
policy-map udp-policy
 class class10
    priority percent 9
 class class11
    priority percent 65
!
!
translation-rule 40
 Rule 1 null null
!
!
interface FastEthernet0/0
 description --- WAN ---
 ip address 12.x.x.132 255.255.255.128
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 10.1.1.2 255.255.255.0
 speed 100
 full-duplex
!
ip default-gateway 12.x.x.129
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 12.x.x.129
ip route 192.x.x.0 255.255.255.0 192.x.x.1
no ip http server
no ip http secure-server
```


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```
!
!
control-plane
!
!
mgcp fax t38 ecm
mgcp behavior g729-variants static-pt
!
!
dial-peer voice 100 voip
 description OutToAcme
 destination-pattern .T
 voice-class codec 99
 voice-class sip asymmetric payload full
 session protocol sipv2
 session target dns:as.voip.fuse.net
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description OutToAvaya
 destination-pattern 908555111.
 voice-class codec 99
 voice-class sip asymmetric payload full
 voice-class sip outbound-proxy ipv4:10.1.1.20
 session protocol sipv2
 session target ipv4:10.1.1.20
 session transport tcp
 incoming called-number .
 dtmf-relay rtp-nte
 no vad
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 credentials username 9085551110 password 7
014355560E5F225F001D68593C5245432E5D22 realm as.voip.fuse.net
 credentials username 9085551111 password 7
1542595E537B7971096267724452355452077D realm as.voip.fuse.net
 credentials username 9085551112 password 7
101F5D4A5347465B2D557F7B717F6A627B4452 realm as.voip.fuse.net
 no remote-party-id
 retry invite 2
 retry response 3
 retry bye 3
 retry prack 6
 timers expires 300000
 registrar dns:as.voip.fuse.net expires 1800
 sip-server dns:as.voip.fuse.net
 connection-reuse
!
!
gatekeeper
 shutdown
```


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```
!
!
line con 0
line aux 0
line vty 0 4
 session-timeout 60
 login local
 transport input telnet ssh
line vty 5 15
 session-timeout 60
 privilege level 15
 login local
 transport input telnet ssh
!
scheduler allocate 20000 1000
end
```


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