



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Notification Solution 2.0, and Acme Packet Net-Net 6.2.0 with AT&T IP Flexible Reach Service using MIS/PNT or AVPN Transport – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Notification Solution, and Acme Packet Net-Net (models 3800, 4250, or 4500) with the AT&T IP Flexible Reach service using **MIS/PNT** or **AVPN** transport connection.

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites. Avaya Notification Solution (ANS) provides real-time multimedia notification and response capabilities to many devices including IP Phones, Cellphones, and digital/analog phones. It can be applied to emergency broadcast and system alarming. An Acme Packet Net-Net is the point of connection between Avaya Notification Solution and the AT&T IP Flexible Reach service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

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 at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Notification Solution, and Acme Packet Net-Net (models 3800, 4250, or 4500) with the AT&T IP Flexible Reach service using MIS/PNT or AVPN transport connection. **Note that the configuration steps in these Application Notes are used for this reference configuration and are not meant to be prescriptive.**

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

Avaya Notification Solution (ANS) provides real-time multimedia notification and response capabilities to many devices including IP Phones, Cellphones, and digital/analog phones. It provides intelligent notification features such as notification cascading, acknowledgement gathering, Voice Mail Detection, Notification Message retrieval and conference. It can be applied to emergency broadcast and system alarming.

An Acme Packet Net-Net (Acme Packet SBC) is the point of connection between Avaya Notification Solution and the AT&T IP Flexible Reach service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise site with ANS server and Acme Packet Session Border Controller (Acme Packet SBC).
- A laboratory version of the AT&T IP Flexible Reach service, to which the simulated enterprise site was connected via MIS/PNT or AVPN transport connection.

The main test objectives were to verify the following features and functionality:

- Outbound calls from ANS server to notify the subscribers
- Outbound calls from ANS server to notify the subscribers to join a Conference Call
- Outbound calls from ANS server to detect Voice Mail service and leave a message
- Inbound call to ANS server to trigger an Outbound notification
- Inbound call to ANS server to trigger a Conference Call notification to the subscribers
- Inbound call to ANS server to retrieve Message notification left for the subscribers
- Inbound call to ANS server to retrieve Conference notification and join the conference
- Inbound and Outbound caller interaction with ANS, including prompting, and DTMF input
- Basic supplementary telephony features such as hold, resume, and conference
- G.729a and G.711 codec support
- Long duration calls using Conferencing

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying call flows (see **Section 3.2** for sample call flows) between ANS, Acme Packet Net-Net, and the AT&T IP Flexible Reach service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made to and from the PSTN across the AT&T network. The following features were tested as part of this effort:

- SIP trunking
- Passing of DTMF events and their recognition by navigating automated voice prompts
- ANS delivery of notifications to subscribers
- ANS and AT&T IP Flexible Reach service features such as hold, resume, and conference

2.2. Known Limitations/Test Results

1. ANS 2.0 does not support Compressed RTP (cRTP). So, cRTP was disabled on AT&T IP Flexible Reach service.
2. ANS 2.0 does not support G729 with annexb=yes as it is unable to detect Voice Mail service when silence suppression is enabled.
3. ANS 2.0 does not support transfer of calls to a help desk.

The test objectives stated in **Section 2** with limitations noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (888) 288-8362.

Avaya customers may obtain documentation and support for Avaya products by visiting <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The sample configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Avaya Notification Solution provides notification service to its subscribers. Avaya Notification Solution consists of Text to Speech server. A single server is used for Avaya Notification Server and Text to Speech server.
- The Acme Packet Net-Net Session Director (SD) 3800¹ provides SIP Session Border Controller functionality between the AT&T IP Flexible Reach service and the enterprise internal network². UDP transport protocol is used between the Acme Packet Net-Net SD and the AT&T IP Flexible Reach service.

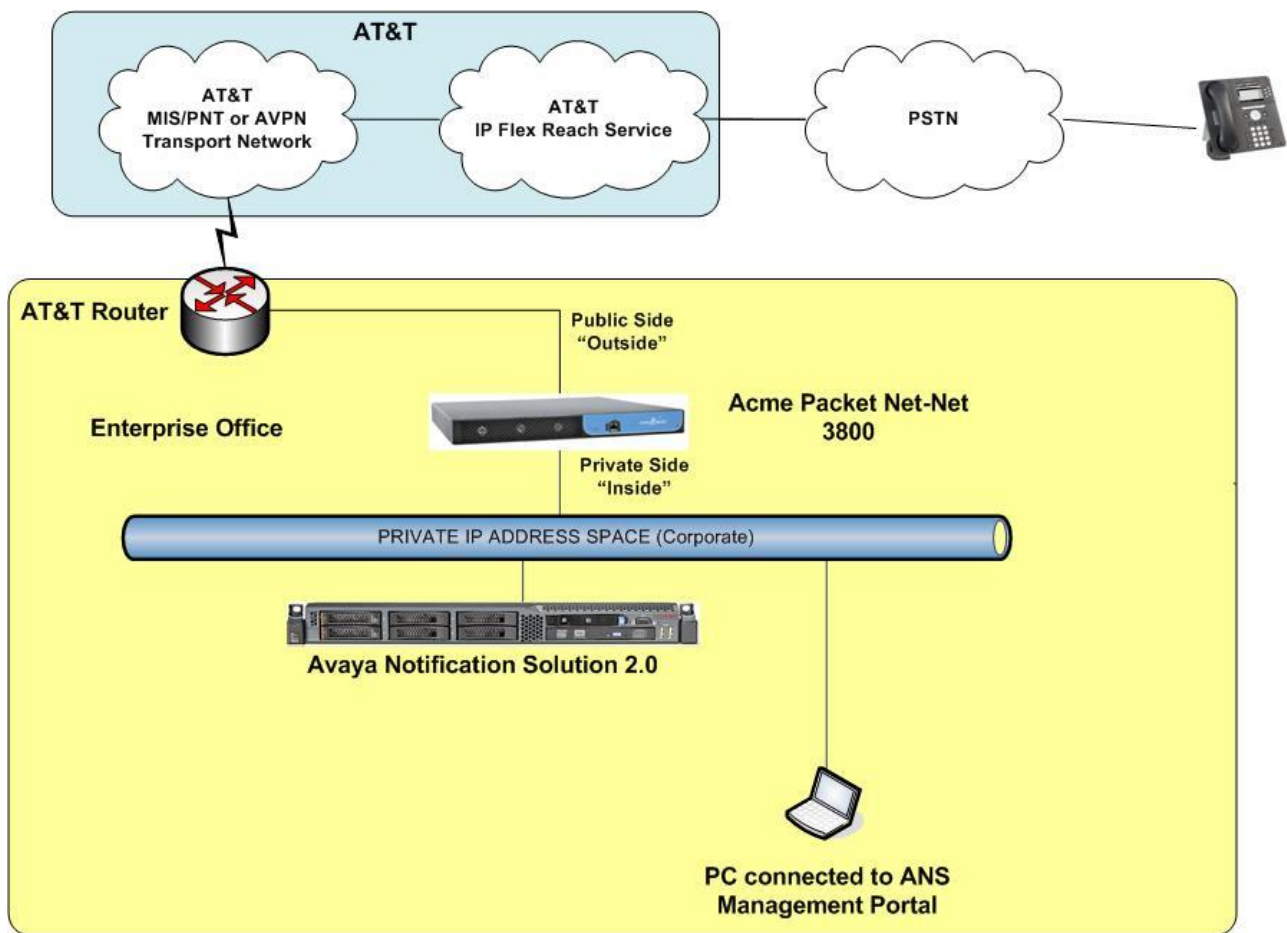


Figure 1: Reference Configuration

¹ Although an Acme Net-Net 3800 was used in the reference configuration, the 4250 and 4500 platforms are also supported.

² The AT&T IP Flexible Reach service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Acme Packet SBC in this reference configuration. In the reference configuration, Avaya Notification Solution uses SIP over TCP to communicate with the Acme Packet SBC.

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in this reference configuration, and are **for illustrative purposes only**. Customers must obtain and use the specific values for their specific configurations.

Component	Illustrative Value in these Application Notes
Avaya Notification Solution with Avaya Media Server	10.80.130.230
Acme Packet Session Border Controller	
IP Address of “Outside” Interface (connected to AT&T IP Flexible Reach Service)	192.168.62.51
IP Address of “Inside” Interface (connected to Avaya elements)	10.80.130.250
AT&T IP Flexible Reach Service	
Border Element IP Address	135.242.225.210

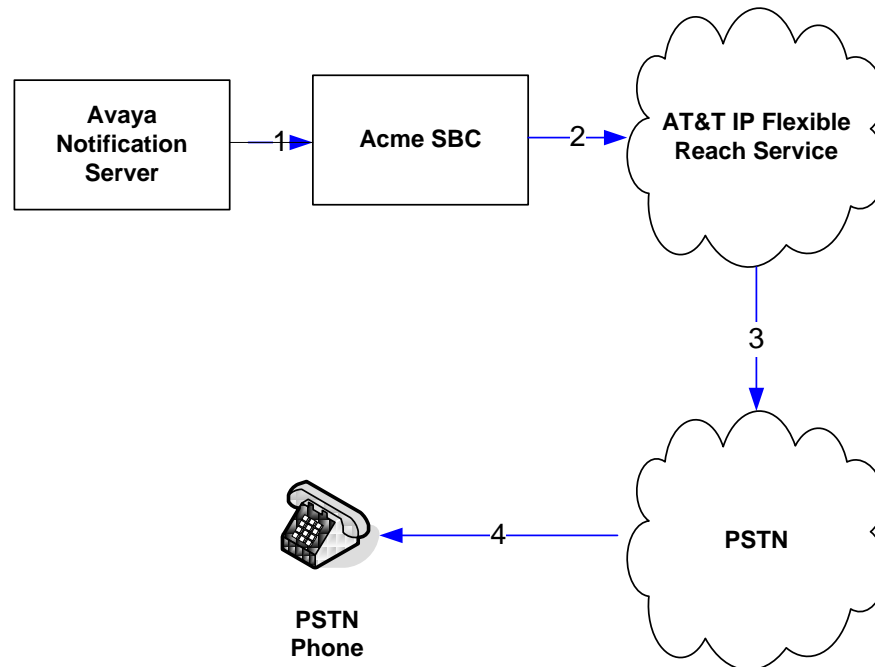
Table 1: Illustrative Values Used in this Reference Configuration

3.2. Call Flows

To understand how AT&T IP Flexible Reach calls are handled by ANS, several call flows are described in this section.

The first call scenario illustrated below is an outbound call originating from ANS to the subscriber/s on PSTN.

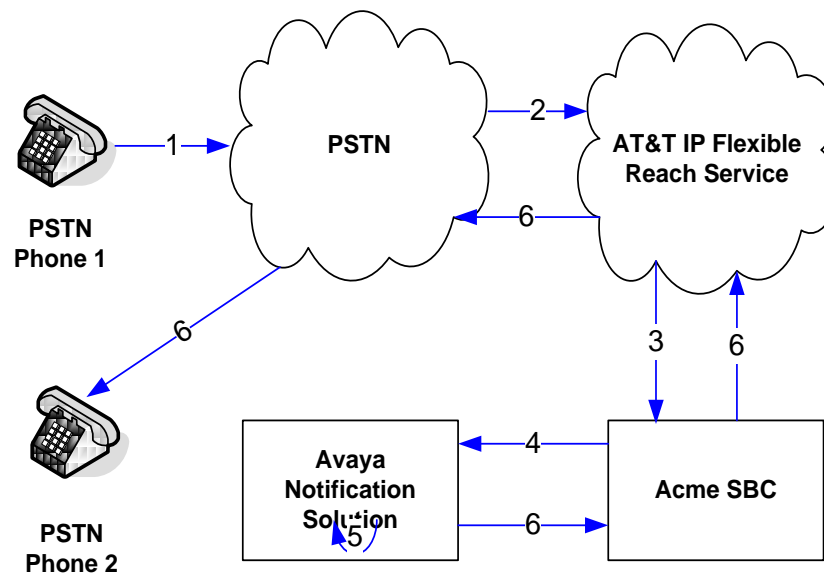
1. ANS originates a call to PSTN phone
2. Acme Packet SBC performs any necessary SIP header modifications, and routes the call to the AT&T IP Flexible Reach Service
3. The AT&T IP Flexible Reach service routes the call to PSTN
4. PSTN delivers the notification call to a PSTN phone subscriber on ANS. If the subscriber does not pick up the phone and a voicemail is detected, ANS leaves the notification message on subscriber's voicemail.



Outbound Call from ANS

The second call scenario illustrated below is an inbound call to ANS to trigger a notification by ANS to its subscriber/s. A notification could be an invite to join a conference or provide information to the ANS subscriber/s.

1. PSTN phone 1 calls ANS DID
2. PSTN delivers call to AT&T Flexible Reach Service
3. AT&T Flexible Reach Service routes the call to Acme Packet SBC at CPE
4. Acme Packet SBC performs any necessary SIP header modifications, and routes the call to ANS
5. Based upon the option entered by PSTN Phone 1, subscriber has an option to record a new message or retrieve a notification sent earlier or trigger an outbound notification
6. Same as Steps 1 to 4 in the first scenario and only performed if an outbound notification is triggered in **Step 5**



Inbound Call Handled by ANS to record/retrieve message or trigger an Outbound Notification

4. Equipment and Software Validated

The following equipment and software was used for the sample configuration described in these Application Notes.

Component	Version
Avaya Notification Solution 2.0	ANS2.0.3025 with InboundCallPatch3025
Avaya Media Server	Version 7.5.0.724
Avaya Notification Solution 2.0 running on VMWare Virtual Machine	VMware vSphere ESX4.0 running on a 32-bit Redhat Enterprise Linux 5.4
Acme Packet Net-Net Session Director 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Flexible Reach Service	VNI 23

Table 2: Equipment and Software Versions

5. Configure Avaya Notification Solution Server

These Application Notes assume that the necessary ANS licenses have been installed and basic ANS administration has already been performed. Consult [1] and [2] for further details if necessary. **Note** that in the following sections only the parameters used in this reference configuration are discussed. Default values are used for all other field configurations.

5.1. Background

ANS is a real-time multimodal notification system with a response gathering capability from the subscriber devices. These devices can be IP phones, cell phones, digital or analog phones. ANS broadcast text and audio messages to Avaya IP phones through IP without consuming any IP-PBX resources. ANS includes a Management Portal for creating notifications and checking their status. ANS stores user and group profiles locally and provides LDAP synchronization with external directory. ANS provides intelligent notification features such as escalation tree, and ad hoc conference. ANS can be used for mass notifications, emergency conferences and event notifications. ANS can detect voice mail system and leave a message for the subscriber on their voicemail. Subscribers can dial into ANS to retrieve the notification. ANS also has a Text-to-Speech server for converting the Text entered while creating a notification, to equivalent speech.

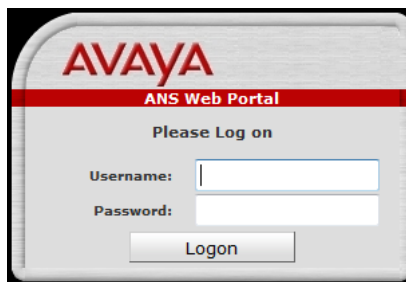
For this reference configuration, ANS was used to send notifications to its subscribers and trigger notifications for its subscribers by calling into the ANS system. Additionally, ad hoc conference feature for the ANS was tested by using inbound triggers or outbound notification for conference. Additionally Voice Mail Detection and Notification retrieval functionality was also tested.

When an inbound call arrives at ANS, it matches the user part of the **To** header in the INVITE to map to the appropriate trigger. For outbound calls from ANS, its Management Portal is used to invoke the notification.

Note: The following sections only show the configuration for the values which were changed in this Reference configuration. For all other fields, default values were used. Additionally, the screens shown below are abridged for clarity.

5.2. Logging in to ANS

Launch a web browser, enter **https://<IP address of the ANS server>:8443/ANSWebPortal** as the URL, and log in with the appropriate credentials to display the **ANS Web Portal** page.

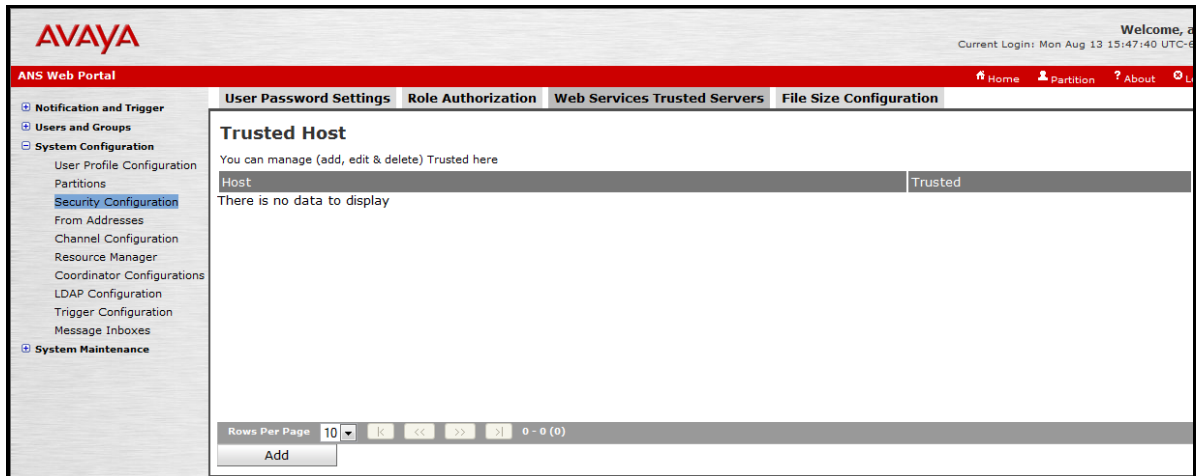


ANS Login Screen

5.3. Security Configuration

This section describes the steps to add ANS server as trusted host.

1. Navigate to **System Configuration**→**Security Configuration** and select **Web Services Trusted Servers** tab. Click **Add**.



Security Configuration – Trusted Host

On the pop-up screen **TrustedHost** shown below, configure as follows:

- **Host** – Enter IP address of ANS
- **Trusted** – Check the box
- Click **Save**

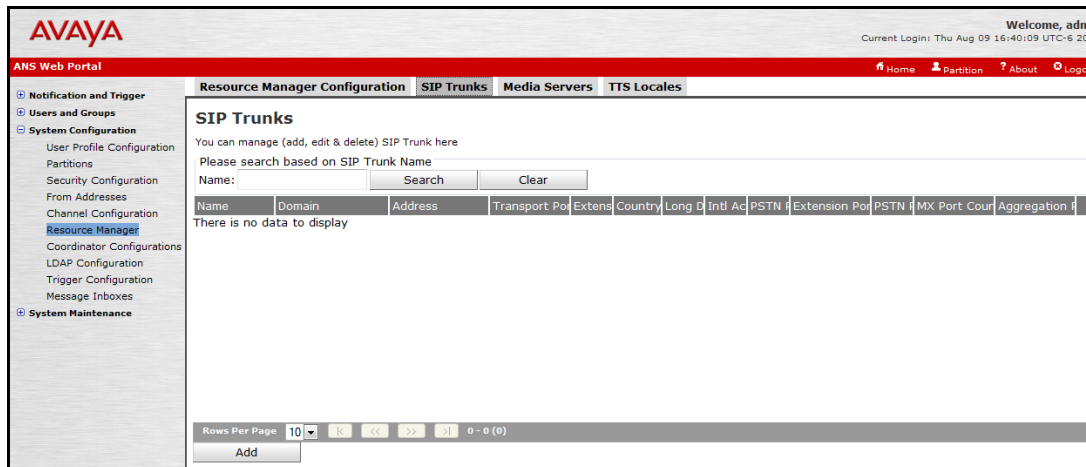
The screenshot shows a 'TrustedHost' pop-up form. It has a red header bar with the title 'TrustedHost'. Below the header, there are two fields: 'Host' with a text input field containing '10.80.130.230', and 'Trusted' with a checked checkbox. At the bottom of the form, there are two buttons: 'Save' and 'Cancel'.

Security Configuration – TrustedHost

5.4. VoIP Connection

This section describes the steps required to configure a SIP trunk between ANS and Acme Packet SBC.

1. In the left pane, navigate to **System Configuration**→**Resource Manager** and select **SIP Trunks** tab. On the **SIP Trunks** page, click **Add**.



Resource Manager – SIP Trunks

On the pop-up screen **Add SIP Trunk** shown below, configure as follows:

- **Name** – Enter any informative string
- **Domain** – Set to the AT&T Border Element IP Address. This domain is used in **From** and **To** headers. See **Section 6** for how the domain name is modified in the **From** header to the external interface of the Acme Packet SBC
- **Address** – Set to IP Address of the Acme Packet SBC internal interface
- **Transport** – Set to **TCP** (used in this reference configuration)
- **Extension Length** – Set to any valid length (default **7**)
- **Extension Port Count** – Set to a valid number based upon the number of licenses
- **Transport Port No** – Set to **5060** (default)
- **PSTN Port Count** – Set to a valid number based upon the number of licenses
- **Aggregation Port Count** – Set to a valid number based upon the number of licenses
- Click **Save**

The screenshot shows the 'Add SIP Trunk' pop-up form. It contains the following fields and values: Name: ToSBC, Domain: 135.242.225.210, Address: 10.80.130.250, Transport: TCP (selected from a dropdown), Extension Length: 7, Country Code: (empty), Long Distance Prefix: (empty), Intl Access Code: (empty), PSTN Prefix: (empty), Extension Port Count: 20, Transport Port No: 5060, PSTN Port Count: 20, Priority: 1, MX Port Count: (empty), and Aggregation Port Count: 100. At the bottom right are 'Save' and 'Cancel' buttons.

Resource Manager – Add SIP Trunk

2. In the left pane, navigate to **System Configuration**→**Resource Manager** and select **Media Servers** tab. On the **Media Servers** page, click **Add**

The screenshot shows the Avaya ANS Web Portal interface. The top navigation bar includes the Avaya logo, a welcome message for 'adm', and the current login time. The main navigation pane on the left lists various configuration categories, with 'Resource Manager' selected. The 'Media Servers' tab is active in the main content area. It displays a search bar and a table with columns for Name, IP Address, Port, Transport, and User. The table is currently empty, showing 'There is no data to display'. At the bottom, there is a pagination control showing 'Rows Per Page' set to 10 and an 'Add' button.

Resource Manager – Media Servers

On the pop-up screen **Add Media Server** shown below, configure as follows:

- **Name** – Enter any informative string
- **IP Address** – Set to IP Address of the of the Media Server which is **10.80.130.230** in this reference configuration
- Use default for all other fields and click **Save**

The screenshot shows the 'Add Media Server' pop-up screen. It has a red header with the title 'Add Media Server'. Below the header, there are five input fields: 'Name' (containing 'AMS1'), 'IP Address' (containing '10.80.130.230'), 'Port' (containing '5090'), 'Transport' (a dropdown menu set to 'TCP'), and 'User' (containing '10b92e11-b933-11e0-ae'). At the bottom right, there are two buttons: 'Save' and 'Cancel'.

Resource Manager – Add Media Server

5.5. Text To Speech Locales Configuration

ANS has a built-in Text to Speech service and configuration for this service is beyond the scope of these Application Notes. This section shows the step for configuring the right Locale for Text to Speech conversion on ANS.

- In the left pane, navigate to **System Configuration**→**Resource Manager** and select the **TTS Locales** tab. Check the appropriate **Enable** box and click **Save**.

Locale	Enable
English-UNITE	<input checked="" type="checkbox"/>
Catalan-SPAI	<input type="checkbox"/>
Portuguese-B	<input type="checkbox"/>
Valencian-SP	<input type="checkbox"/>
Spanish-COLX	<input type="checkbox"/>
Danish-DENM	<input type="checkbox"/>
Dutch-NEDER	<input type="checkbox"/>
English-UNITE	<input type="checkbox"/>
Finnish-FINLA	<input type="checkbox"/>
Chinese-Taiwan	<input type="checkbox"/>

Resource Manager – TTS Locales

5.6. From Address Configuration

This section describes steps to configure the user part for the **From** header to be sent in an outbound call. An address and display name are configured in the following steps.

1. Navigate to **System Configuration**→**From Address** and select **From Address** tab. On the **From Address** page, click **Add**.

Channel	From	Default Address	Partition
Email	admin@ans.com	Yes	default
Voice	anonymous	Yes	default

From Address - From Address

On the pop-up screen **Add From Address** shown below, configure as follows:

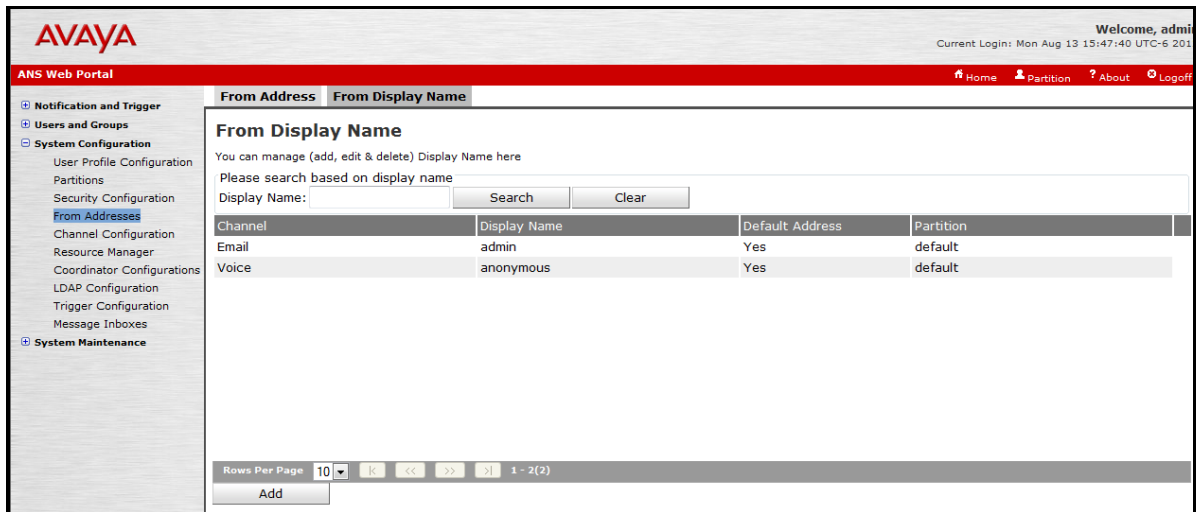
- **Channel** – Select **Voice** from a drop-down list
- **From** – Enter a valid telephone number
- **Default Address** – Check this box for at least one address
- Click **Save**
- Repeat this step to configure additional addresses



The 'Add From Address' dialog box has a red header. It contains the following fields: 'Channel' with a dropdown menu showing 'Voice', 'From' with a text input containing '7325551212', 'Default Address' with a checked checkbox, and 'Partition' with a dropdown menu showing 'default'. At the bottom are 'Save' and 'Cancel' buttons.

From Address – Add From Address

2. Navigate to **System Configuration**→**From Address** and select **From Display Name** tab. On the **From Display Name** page, click **Add**.



The screenshot shows the 'ANS Web Portal' with the 'From Display Name' tab selected. The page title is 'From Display Name'. Below the title is a search bar with the text 'Please search based on display name'. Below the search bar is a table with the following data:

Channel	Display Name	Default Address	Partition
Email	admin	Yes	default
Voice	anonymous	Yes	default

At the bottom of the page, there is a 'Rows Per Page' dropdown set to '10', a pagination bar showing '1 - 2(2)', and an 'Add' button.

From Address– From Display Name

On the pop-up screen **Add Display Name** shown below, configure as follows:

- **Channel** – Select **Voice** from drop-down list
- **Display Name** – Enter any informative string
- **Default Address** – Check this box for at least one address
- Click **Save**
- Repeat this step to configure additional display names

From Address– Add Display Name

5.7. TTW Server

This section describes steps to configure the TTW Server.

1. Navigate to **System Configuration**→**Channel Configuration** and select **TTW SERVER** from the drop-down list in the **Select the Channel/Component to configure** field.

Channel Configuration – Select Component

On the subsequent **Channel Configurations** screen, configure as follows:

- **TTW IP ADDRESS** – Set to the IP address of the ANS Server
- **TTW IP ADDRESS 2** – Set to the IP address of the ANS Server

Property	Value	Description
TTW IP ADDRESS	10.80.130.230	TTW Service IP Address. Usually, TTW is installed c
TTW IP ADDRESS 2	10.80.130.230	TTW Service IP Address 2. Usually, TTW is instal
TTW PORT	1236	TTW Service port. (Default port is 1236)
TTW PORT 2	1236	TTW Service port 2. (Default port is 1236)

Channel Configuration – TTW Address

5.8. Permitted Driver IP Address

This section describes steps to configure the Permitted Driver IP Address.

1. Navigate to **System Configuration**→**Coordinator Configuration** and click **Add**.

The screenshot shows the Avaya ANS Web Portal interface. The top navigation bar includes the Avaya logo, the text 'ANS Web Portal', and a 'Welcome, admin' message with the current login time. The left sidebar contains a tree view with categories like 'Notification and Trigger', 'Users and Groups', 'System Configuration', and 'System Maintenance'. Under 'System Configuration', 'Coordinator Configurations' is selected. The main content area has two tabs: 'Driver Command Size' and 'Permitted Driver IP'. The 'Permitted Driver IP' tab is active, displaying a table with one row for 'IP Address' with the value '127.0.0.1'. At the bottom of the table, there is a 'Rows Per Page' dropdown set to 10, navigation buttons, and a page indicator '1 - 1(1)'. An 'Add' button is located at the bottom left of the table area.

Coordinator Configuration – Permitted Driver IP

On the pop-up screen **Add Driver IP Address** shown below set the **IP Address** field to ANS Server's IP address and click **Save**.

The screenshot shows a pop-up dialog titled 'Add Driver IP Address'. It has a red header bar with the title in white. Below the header, there is a label 'IP Address:' followed by a text input field containing the value '10.80.130.230'. At the bottom of the dialog, there are two buttons: 'Save' and 'Cancel'.

Coordinator Configuration – Add Driver IP Address

5.9. Inbound Number

ANS inspects the **To** header to determine the destination of the call. Inbound numbers are configured to determine which trigger is invoked when a call comes into ANS.

1. Navigate to **System Configuration→Trigger Configuration** and click **Add**.

The screenshot shows the AVAYA ANS Web Portal interface. The top navigation bar includes the AVAYA logo, a welcome message for 'admin', and links for Home, Partition, About, and Logout. The left sidebar lists various configuration categories: Notification and Trigger, Users and Groups, and System Configuration. Under System Configuration, 'Trigger Configuration' is highlighted. The main content area is titled 'Inbound Call Trigger Configuration' and 'Email Trigger Configuration'. The 'Inbound Number' section allows users to manage inbound numbers, with a search bar and a table. The table currently shows 'There is no data to display'. At the bottom, there is a pagination bar with 'Rows Per Page' set to 10 and a total of 0 records.

Trigger Configuration – Inbound Number

On the pop-up screen **Add Inbound Number Data** shown below, configure as follows:

- **Inbound Number** – Set to a valid DID
- Click **Save**
- Repeat this step to configure additional inbound numbers

The screenshot shows a pop-up window titled 'Add InBound Number Data'. It contains two input fields: 'InBound Number' with the value '7325551212' and 'Partition' with a dropdown menu set to 'default'. There are 'Save' and 'Cancel' buttons at the bottom.

Trigger Configuration – Add Inbound Number Data

5.10. Message Inbox

Message inboxes are configured to store message notifications on ANS. Subscribers can call ANS to record or retrieve a message notification.

1. Navigate to **System Configuration**→**Message Inbox** and click **Add**.

The screenshot shows the AVAYA ANS Web Portal interface. The top navigation bar includes the AVAYA logo, the text 'ANS Web Portal', and a 'Welcome, admin' message with the current login time. The left sidebar contains a tree view with categories like 'Notification and Trigger', 'Users and Groups', 'System Configuration', and 'System Maintenance'. The 'System Configuration' category is expanded, showing sub-items like 'User Profile Configuration', 'Partitions', 'Security Configuration', 'From Addresses', 'Channel Configuration', 'Resource Manager', 'Coordinator Configurations', 'LDAP Configuration', 'Trigger Configuration', and 'Message Inboxes'. The 'Message Inboxes' item is selected. The main content area is titled 'Message Inboxes' and contains a search bar and a table. The table has columns for 'Inbox Number', 'Inbox Description', 'Expiration Value', and 'Partition'. The table is currently empty, with the message 'There is no data to display' shown. At the bottom of the table, there is a 'Rows Per Page' dropdown set to '10' and a pagination bar showing '0 - 0 (0)'. An 'Add' button is located at the bottom left of the table area.

Message Inboxes

On the pop-up screen **Add Inbound Number Data** shown below, configure as follows:

- **Inbox Number** – Set to any valid number
- **Inbox Description** – Enter any informative description
- Click **Save**
- Repeat this step to configure additional Message inboxes

The screenshot shows the 'Add/Edit Message Inbox' pop-up screen. It has a red header bar with the title 'Add/Edit Message Inbox'. The form contains four fields: 'Inbox Number' with the value '5381212', 'Inbox Description' with the value 'Box1', 'Expiration Time' with the value '1' and a dropdown menu set to 'HOURS', and 'Partition' with a dropdown menu set to 'default'. At the bottom of the form are two buttons: 'Save' and 'Cancel'.

Trigger Configuration – Add Inbound Number Data

5.11. Manage User Profile

The following steps show the subscriber configuration on ANS.

1. In the left pane, navigate to **Users and Groups** → **Users** and click **Add**.

Manage User Profile

You can manage (add, edit & delete) user profile here

Please search based on User Id, First Name or Last Name.

Search Value:

Role: Active:

User Id	First Name	Middle Name	Last Name
AUSER	Anonymous	AnsAdmin	Ans
admin	Admin	AnsAdmin	Ans

Rows Per Page: 10 1 - 2(2)

Users – Manage User Profile

2. On the subsequent screen **Create User Profile** shown below, select the **User Details** tab and configure as follows:
 - **User Id** – Enter any valid id
 - **Time Zone** – Enter a valid timezone
 - **First Name** – Enter any informative string
 - **Last Name** – Enter any informative string
 - Click **Save**
 - Repeat this step to configure additional subscribers

Create User Profile

Create User Profile

User Details **Contact Information**

Users

User Id: **Title:**

First Name: **Last Name:** **Time Zone:**

Account Number: **Middle Name:**

Role:

Telephone Security PIN: **Web Login Password:**

Activate: ☒

Users – User Details

3. Select the **Contact Information** tab and enter valid contact information.

Note: Multiple contacts can be entered and Notification will be sent simultaneously to all the contacts.

AVAYA ANS Web Portal

Manage User Profile
You can manage (add, edit & delete) user profile here

User Details **Contact Information**

Point of Contact Information

Contact Information	Urgent Notifications Rule Time Profile	Delay(s)	Normal Notifications Rule Time Profile	Delay(s)
Work Phone	Anytime	0	Anytime	0
Mobile Phone	Anytime		Anytime	
Home Phone	Anytime		Anytime	
IP Phone Extension	Anytime		Anytime	
SMS Address	Anytime		Anytime	
Work Email	Anytime		Anytime	
Personal Email	Anytime		Anytime	
Sametime Address	Anytime		Anytime	
XMPP Address	Anytime		Anytime	

Save Cancel

Users – Contact Information

4. Repeat above step to configure additional subscribers.

5.12. Add Outbound Notifications

The following steps show the configuration of ANS for sending outbound notifications to its subscribers.

1. In the left pane, navigate to **Notifications and Trigger** → **Notifications Scenarios** and click **Add**.

ANS Web Portal

Notification Scenarios
You can manage (add, edit & delete) notification template here

Please search based on name, description
Scenario Name/Description: Search Clear

Scenario Name	Description	Owner
There is no data to display		

Rows Per Page 10 0 - 0 (0)
Add

Notification Scenarios

2. On the subsequent screen, select **Details** tab and configure as follows:
 - **Scenario Name** – Enter any informative string
 - **Scenario Description** – Enter a description (Optional)

The screenshot shows the 'ANS Web Portal' interface. On the left is a navigation menu with 'Notification and Trigger' expanded, showing options like 'My Notification Scenarios', 'Notification Scenarios' (highlighted), 'My Notification History', 'Notification History', 'My Escalations', 'Escalations', 'Usage Report', 'Inbound Call Triggers', 'Email Trigger Configuration', and 'Conference Bridges'. Below this are 'Users and Groups', 'System Configuration', and 'System Maintenance'. The main content area is titled 'Notification Scenarios' with a subtitle 'You can manage (add, edit & delete) notification template here'. It features several tabs: 'Details' (selected), 'Message', 'Users', 'Groups', 'Escalations', and 'Trigger Permissions'. The 'Details' tab contains the following fields: 'Scenario Name' (text box with 'OutboundMessage'), 'Scenario Description' (text box with 'essages to subscribers'), 'Owner' (text box with 'admin' and a 'Select' button), 'Expiration Time' (text box with '1' and a 'HOURS' dropdown), and 'Priority' (radio buttons for 'Normal' (selected), 'Urgent', and 'Crisis').

Notification Scenarios – Details

3. Select **Message** tab and click **Add Message**.

The screenshot shows the 'ANS Web Portal' interface, similar to the previous one, but with the 'Message' tab selected. The 'Common Messages' section contains three text boxes: 'Text Message Subject:', 'Text Message Body:', and 'Audio Message:'. The 'Text Message Body:' and 'Audio Message:' boxes have vertical scrollbars. To the right of the 'Audio Message' box is a 'Select W' button. Below this is a 'Messages' section with a grey header and the text 'There is no data to display'. At the bottom, there is a pagination bar with 'Rows Per Page' set to '5', navigation buttons (|<, <<, >>, >|), and '0 - 0 (0)'. An 'Add Message' button is located at the bottom left of the main content area.

Notification Scenarios – Message

On the pop-up screen **Add New Message** shown below, configure as follows:

- **Channel** – Select **VOICE** from the drop-down list
- **Caller ID** – Select one of the numbers configured in **Section 5.6, Step 1**
- **Display Name** - Select one of the values configured in **Section 5.6, Step 2**
- **Enable Inbound** – This field is enabled only if the notification needs to be saved in a Message Box configured in **Section 5.10** for subscribers to retrieve message later. Select one of the values configured in **Section 5.10, Step 1**
- **Greeting Prompt** – Enter any valid prompt which is played to the subscriber when the call is answered
- **Message Body** – Enter any valid message
- **Leave a message to voicemail** – Check this so that the message can be left on voicemail in case the subscriber is unavailable
- **Use same message body for voicemail** – Check this if the Voicemail Body is same as the Message Body. In this reference configuration, this field is checked
- Click **Save**

Add New Message

Messages **Choice**

Channel Messages
Required Fields in bold: 1.Message Body 2.Greeting Prompt must be provided

Channel: VOICE **Locale:** en-US
Caller ID: 7325551212 **Display Name:** ANS1212
Retry Count: 0 **Retry Delay(Seconds):** 0 **Ring Timeout(seconds):**
☐ Authenticate Recipient ☒ Enable OutBound ☒ Work Phone
☐ Bypass Human Detection ☒ Enable InBound 5381212 ☒ Mobile Phone
☐ Audio Conference ANS Ad-Hoc Conference ☒ Home Phone

Greeting Prompt: This is an important Message from Avaya **Select Wave File** **Record Through Telephone**
Message Body: emergency repairs. Please call 7325551212 for further details **Select Wave File** **Record Through Telephone**

☐ Bypass Answer Machine Detection? ☒ Leave a message to voicemail? ☒ Use same message body for voicemail?
Voicemail Body: The building is closed today due to emergency rep **Select Wave File** **Record Through Telephone**

Save **Cancel**

Notification Scenarios – Add New Message

4. In case an adhoc conference is configured (which is done in this reference configuration), then the screen in previous step will look similar to the one shown below. Note that **Audio Conference** box is checked.

Notification Scenarios – Add New Message (Audio Conference)

- Select the **Choice** tab and click **Add a question**

Notification Scenarios – Choice (Audio Conference)

- On the pop-up screen **Add Question** shown below, enter any informative string in **Question 1 Content** field and click **Save**

Add Question

Question 1 Content: Would you like to join the conference call?

Select Wave File Record Through Telephone

Save Cancel

Notification Scenarios – Add Question (Audio Conference)

- On the next screen where the question is added, highlight the question and following screen appears. Click **Add a choice**

Add New Message

Messages Choice

▼ Add Questions

Question ID	Question
Question 1	Would you like to join the conference call?

Rows Per Page: 5 | 1 - 1(1)

Add a question Edit Delete Clear Selection

Add Choices

Choice ID	Choice
There is no data to display	

Rows Per Page: 5 | 0 - 0 (0)

Add a choice

Notification Scenarios – Add a choice (Audio Conference)

On the pop-up screen **Add Question** shown below, configure as follows:

- Choice 1 Content** - Enter any informative text
- Acknowledgment after chose** – Enter any informative text
- Check the **Mark this choice as the “Affirmative” answer for reporting purposes** box
- Audio Conference** – Select this option
- Click **Save**

Add Question

Choice 1 Content: Press 1 for joining the conference

Acknowledgment after chosen: Thank you

☒ Mark this choice as the "Affirmative" answer for reporting purposes

☒ Audio Conference ☐ No Action

Select Wave File Record Through Telephone

Save Cancel

Notification Scenarios – Choice to join the conference (Audio Conference)

- The following screen displays another choice which gives the subscriber an opportunity to refuse to join the conference

Add Question			
Choice 2 Content	Press 2 for No	Select Wave File	Record Through Telephone
Acknowledgment after chosen	Good Bye	Select Wave File	Record Through Telephone
<input type="checkbox"/> Mark this choice as the "Affirmative" answer for reporting purposes			
<input type="radio"/> Audio Conference <input checked="" type="radio"/> No Action			

Notification Scenarios – Refusal to join the conference (Audio Conference)

- The screen below shows the choices configured in this step. Click **Save**

Add New Message	
Messages	Choice
▼ Add Questions	
Question ID	Question
Question 1	Would you like the join the conference call?
Rows Per Page 5 < << >> > 1 - 1(1)	
Add a question Edit Delete Clear Selection	
Add Choices	
Choice ID	Choice
Choice 1	Press 1 for joining the conference
Choice 2	Press 2 for No
Rows Per Page 5 < << >> > 1 - 2(2)	
Add a choice	
<div>Save</div> <div>Cancel</div>	

Notification Scenarios – Choices (Final)

5. Select the **Users** tab and highlight the subscriber/s to deliver the notification and click **Add**.

AVAYA Welcome, admin
Current Login: Wed Aug 15 10:45:29 UTC-6 2012

ANS Web Portal Home Partition ? About Logo

Notification and Trigger
My Notification Scenarios
Notification Scenarios
My Notification History
Notification History
My Escalations
Escalations
Usage Report
Inbound Call Triggers
Email Trigger Configuration
Conference Bridges

Notification Scenarios
You can manage (add, edit & delete) notification template here

Details Message Users Groups Escalations Trigger Permissions

Search Value: Search Clear

First Name	Last Name	User Id	Middle Name		First Name	Last Name	User Id	Middle Name	Conference
Anonymous	Ans	AUSER	AnsAdmin	Add					
Admin	Ans	admin	AnsAdmin	Remove					
1760	PSTN	PSTN1760							
1212	PSTN	PSTN1212							

There is no data to display

Rows Per Page 10 1 - 4(4) Rows Per Page 10 0 - 0 (0)

Save Cancel

Notification Scenarios – Add Users

- On the subsequent screen after the subscribers are added to receive notification, click **Save**

AVAYA Welcome, admin
Current Login: Wed Aug 15 10:45:29 UTC-6 2012

ANS Web Portal Home Partition ? About Logo

Notification and Trigger
My Notification Scenarios
Notification Scenarios
My Notification History
Notification History
My Escalations
Escalations
Usage Report
Inbound Call Triggers
Email Trigger Configuration
Conference Bridges

Notification Scenarios
You can manage (add, edit & delete) notification template here

Details Message Users Groups Escalations Trigger Permissions

Search Value: Search Clear

First Name	Last Name	User Id	Middle Name		First Name	Last Name	User Id	Middle Name	Conference
Anonymous	Ans	AUSER	AnsAdmin	Add	1760	PSTN	PSTN1760		Participant
Admin	Ans	admin	AnsAdmin	Remove	1212	PSTN	PSTN1212		Participant

Rows Per Page 10 1 - 2(2) Rows Per Page 10 1 - 2(2)

Save Cancel

Notification Scenarios – Users Added

6. The following screen shows all the notifications configured for this reference configuration

AVAYA Welcome, admin
Current Login: Wed Aug 15 10:45:29 UTC-6 2012

ANS Web Portal Home Partition ? About Logo

Notification and Trigger
My Notification Scenarios
Notification Scenarios
My Notification History
Notification History
My Escalations
Escalations
Usage Report
Inbound Call Triggers
Email Trigger Configuration
Conference Bridges

Notification Scenarios
You can manage (add, edit & delete) notification template here

Please search based on name, description

Scenario Name/Description: Search Clear

Scenario Name	Description	Owner
Conference	Notification to invite to join conference	admin
VoiceMailDetection	Notification to test voice mail detection	admin
OutboundMessage	Notification for subscribers on ANS	admin

Notification Scenarios configured

5.13. Add Inbound Triggers

The following steps show the configuration of ANS adding triggers on an inbound call to generate outbound notifications based upon the choices entered by the caller.

1. In the left pane, navigate to **Notification and Trigger**→**Inbound Call Trigger** and click **Add**.

The screenshot shows the ANS Web Portal interface. On the left is a navigation menu with categories: Notification and Trigger, Users and Groups, System Configuration, and System Maintenance. Under 'Notification and Trigger', 'Inbound Call Triggers' is selected. The main content area is titled 'Inbound Call Triggers' and contains a search bar with 'Search' and 'Clear' buttons. Below the search bar is a table with columns 'Trigger Name' and 'Trigger Description', which is currently empty with the message 'There is no data to display'. At the bottom, there is a 'Rows Per Page' dropdown set to 10 and an 'Add' button.

Inbound Call Triggers

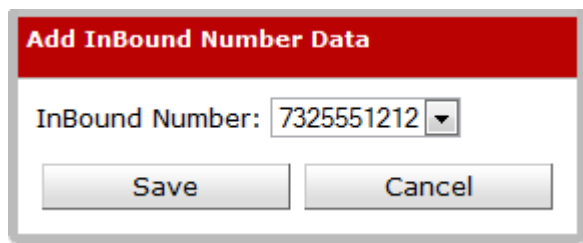
On the pop-up screen **Add Trigger** shown below, configure as follows:

- **Inbound Call Trigger Name** – Enter any informative string
- **Inbound Call Trigger Description** – Optional
- **Greeting Prompt** – Enter any informative string
- Scroll down and click **Add** (not shown) at the bottom of **Inbound Numbers** section

The screenshot shows the 'Add Trigger' pop-up window. It has two tabs: 'General' and 'Choice'. The 'General' tab is active. It contains fields for 'Inbound Call Trigger Name' (filled with 'InboundCallTrigger'), 'Inbound Call Trigger Description' (filled with 'essages, invoke triggers'), 'Locale' (a dropdown menu showing 'en-US'), 'Greeting Prompt' (filled with 'Welcome to Avaya ANS'), and 'Trigger Access Pin' (empty). To the right of the 'Greeting Prompt' field are two buttons: 'Select Wave File' and 'Record Through Telephone'. Below these fields is a section titled 'InBound Numbers' which contains a table with columns 'InBound Number' and an empty row below it with the message 'There is no data to display'. At the bottom of this section is a 'Rows Per Page' dropdown set to 5. At the very bottom of the window are 'Save' and 'Cancel' buttons.

Inbound Call Triggers – Add Trigger

- On the pop-up screen **Add Inbound Number Data** shown below, select one of the numbers configured in **Section 5.9** and click **Save**

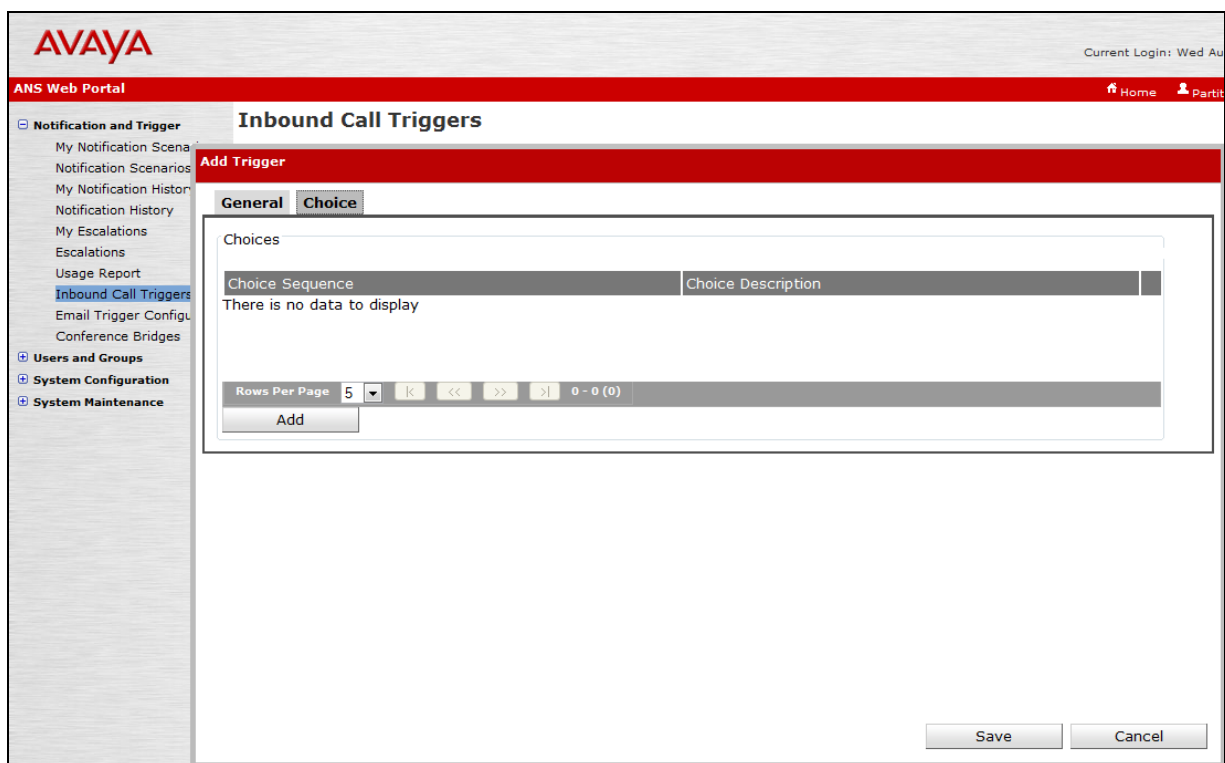


Add Inbound Number Data

InBound Number:

Inbound Call Triggers – Add Inbound Number Data

- Select the **Choice** tab and click **Add**



AVAYA Current Login: Wed Au

ANS Web Portal Home Partit

Inbound Call Triggers

Add Trigger

General **Choice**

Choices

Choice Sequence	Choice Description
There is no data to display	

Rows Per Page: 5 0 - 0 (0)

Inbound Call Triggers – Choice

- On the pop-up screen **Add Choice Data**, check the **Single Scenario** box and click **Select**

Inbound Call Triggers – Add Choice Data

- On the pop-up screen **Add Scenario**, select one of the notifications created in **Section 5.12**

Inbound Call Triggers – Add Scenario

- Following screen shows all the choices configured for the Inbound Trigger in this reference configuration. Click **Save**

Add Trigger

General

Choice

Choices

Choice Sequence	Choice Description
1	Press 1 for sending notification
2	Press 2 for sending conference notification
3	Press 3 for recording notification message
4	Press 4 for retrieving notification message

Rows Per Page

5

k

<<

>>

>|

1 - 4(4)

Add

Save

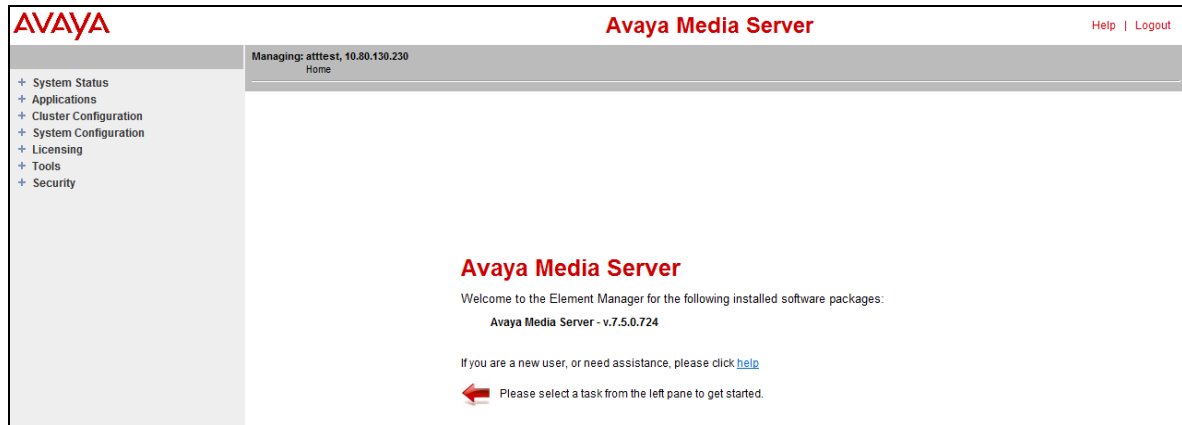
Cancel

Inbound Call Triggers – Add Trigger (Final)

5.14. Avaya Media Server

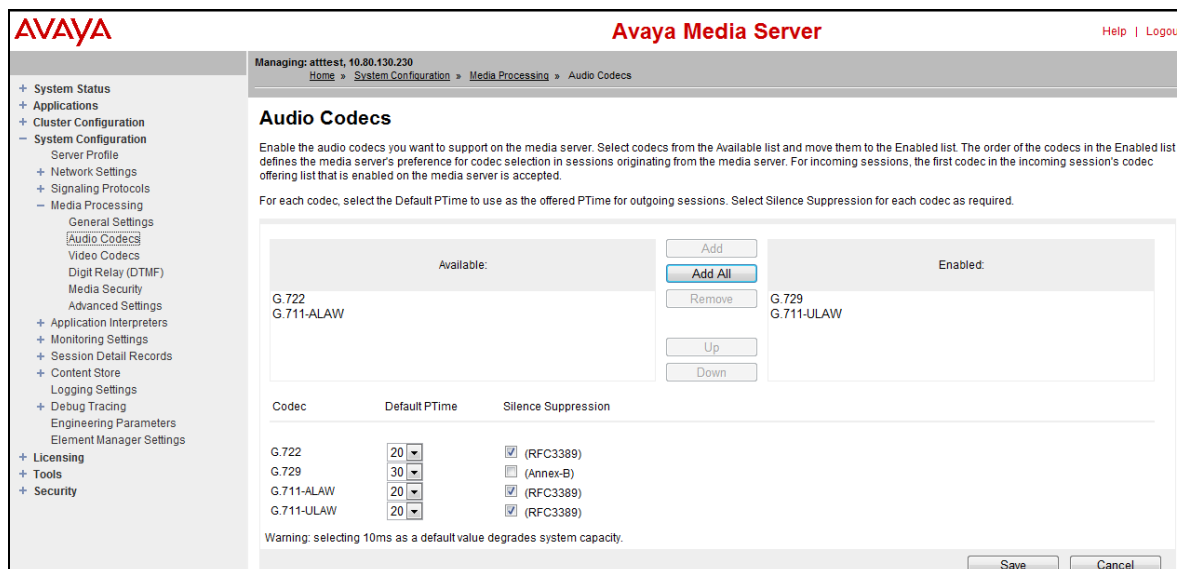
The installation of Avaya Media Server is beyond the scope of this document. For installation and basic system administration steps refer to [1, 2]. This section covers configuration related to IP Codec Set, Ptime, Dynamic Payload and RTP Port Range.

- Launch a web browser, enter **https://<IP address of the ANS server>:9443/** as the URL, and log in with the appropriate credentials to display Avaya Media Server (AMS) home page as shown below.



AMS Home Page

- Navigate to **System Configuration→Media Processing→Audio Codecs** and verify G729 codec is Enabled and on top of the list and the **Silence Suppression** box is unchecked for G729. See **Section 2.2, item 2** for further details.



AMS Audio Codecs

- Navigate to **System Configuration→Media Processing→Digital Relay (DTMF)** and set Specify Type field to **100** and click **Save**

AMS DTMF

- Navigate to **System Configuration→Media Processing→Advanced Settings** and set **Starting Port for Conferencing** field to **16384** and **Last Port for Conferencing** field to **32767** as required by AT&T IP Flexible Reach service and click **Save**

AMS Advanced Setting for RTP ports

6. Configure Acme Packet Session Border Controller

These Application Notes assume that basic Acme Packet SBC administration has already been performed. The Acme Packet SBC configuration used in the sample configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. . Use **putty** or similar tool to access Acme Packet SBC for configuration. Consult with Acme Packet Support [3] for further details and explanations on the configuration below.

ANNOTATION: The local policy below governs the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Notification Solution, Communication Manager, etc., reside to the AT&T IP Flexible Reach service.

```
local-policy
  from-address
                                *
  to-address
                                *
  source-realm
                                Enterprise
  description
  activate-time
                                N/A
  deactivate-time
                                N/A
  state
                                enabled
  policy-priority
                                none
  last-modified-by
                                admin@console
  last-modified-date
                                2011-08-12 10:25:23
  policy-attribute
    next-hop
                                sag:SP_PROXY
    realm
                                ATT
    action
                                none
    terminate-recursion
                                disabled
    carrier
    start-time
                                0000
    end-time
                                2400
    days-of-week
                                U-S
    cost
                                0
    app-protocol
                                SIP
    state
                                enabled
    methods
    media-profiles
```

ANNOTATION: The local policy below governs the routing of SIP messages from the AT&T IP Flexible Reach service to Notification Solution.

```
local-policy
  from-address
                                *
  to-address
                                *
  source-realm
                                ATT
```

description	
activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-08-12 10:25:23
policy-attribute	
next-hop	10.80.130.230
realm	Enterprise
action	none
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
app-protocol	SIP
state	enabled
methods	
media-profiles	

media-manager	
state	enabled
latching	enabled
flow-time-limit	86400
initial-guard-timer	300
subsq-guard-timer	300
tcp-flow-time-limit	86400
tcp-initial-guard-timer	300
tcp-subsq-guard-timer	300
tcp-number-of-ports-per-flow	2
hnt-rtcp	disabled
algd-log-level	NOTICE
mbcd-log-level	NOTICE
red-flow-port	1985
red-mgcp-port	1986
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
media-policing	enabled
max-signaling-bandwidth	7752190
max-untrusted-signaling	80
min-untrusted-signaling	20
app-signaling-bandwidth	0
tolerance-window	30
rtcp-rate-limit	0
min-media-allocation	32000
min-trusted-allocation	60000
deny-allocation	32000
anonymous-sdp	disabled
arp-msg-bandwidth	32000
fragment-msg-bandwidth	0
rfc2833-timestamp	disabled
default-2833-duration	100

```

rfc2833-end-pkts-only-for-non-sig enabled
translate-non-rfc2833-event disabled
dnssalg-server-failover disabled
last-modified-by admin@console
last-modified-date 2010-09-08 10:22:03

network-interface
  name wancom0
  sub-port-id 0
  description
  hostname
  ip-address 192.9.230.221
  pri-utility-addr
  sec-utility-addr
  netmask 255.255.255.0
  gateway 192.9.230.254
  sec-gateway
  gw-heartbeat
    state disabled
    heartbeat 0
    retry-count 0
    retry-timeout 1
    health-score 0
  dns-ip-primary
  dns-ip-backup1
  dns-ip-backup2
  dns-domain
  dns-timeout 11
  hip-ip-list
  ftp-address
  icmp-address
  snmp-address
  telnet-address
  last-modified-by admin@console
  last-modified-date 2011-08-12 10:21:39

```

ANNOTATION: The network interface below defines the IP addresses on the interface connected to the network on which the Avaya elements reside.

```

network-interface
  name s0p0
  sub-port-id 0
  description
  hostname
  ip-address 10.80.130.250
  pri-utility-addr
  sec-utility-addr
  netmask 255.255.255.0
  gateway 10.80.130.1
  sec-gateway
  gw-heartbeat
    state disabled
    heartbeat 0
    retry-count 0
    retry-timeout 1

```

health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	attavaya.com
dns-timeout	11
hip-ip-list	10.80.130.250
ftp-address	
icmp-address	10.80.130.250
snmp-address	
telnet-address	
last-modified-by	admin@console
last-modified-date	2011-08-12 14:58:25

ANNOTATION: The network interface below defines the IP addresses on the interface connected to the network on which the AT&T IP Flexible Reach service resides.

network-interface	
name	slp0
sub-port-id	0
description	
hostname	
ip-address	192.168.62.51
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.128
gateway	192.168.62.1
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	192.168.62.50
ftp-address	
icmp-address	192.168.62.50
snmp-address	
telnet-address	
last-modified-by	admin@console
last-modified-date	2011-08-12 10:24:07
ntp-config	
server	192.9.1.2
last-modified-by	admin@console
last-modified-date	2009-03-12 10:20:46
phy-interface	
name	wancom0
operation-type	Control

port	2
slot	0
virtual-mac	
wancom-health-score	9
last-modified-by	admin@console
last-modified-date	2011-08-12 10:21:30

phy-interface	
name	s0p0
operation-type	Media
port	0
slot	0
virtual-mac	00:08:25:a0:f3:68
admin-state	enabled
auto-negotiation	enabled
duplex-mode	FULL
speed	100
last-modified-by	admin@console
last-modified-date	2011-08-13 15:29:00

phy-interface	
name	s1p0
operation-type	Media
port	0
slot	1
virtual-mac	00:08:25:a0:f3:6e
admin-state	enabled
auto-negotiation	enabled
duplex-mode	FULL
speed	100
last-modified-by	admin@console
last-modified-date	2011-08-13 15:29:23

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP Flexible Reach service resides, and applies SIP manipulations **ModifyMaxptime**.

realm-config	
identifier	ATT
description	
addr-prefix	0.0.0.0
network-interfaces	
	s1p0:0
mm-in-realm	enabled
mm-in-network	enabled
mm-same-ip	enabled
mm-in-system	enabled
bw-cac-non-mm	disabled
msm-release	disabled
generate-UDP-checksum	disabled
max-bandwidth	0
fallback-bandwidth	0
max-priority-bandwidth	0
max-latency	0
max-jitter	0
max-packet-loss	0

observ-window-size	0
parent-realm	
dns-realm	
media-policy	
in-translationid	
out-translationid	
in-manipulationid	modifyMaxptime
out-manipulationid	NAT_IP
manipulation-string	
class-profile	
average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	4
maximum-signal-threshold	3000
untrusted-signal-threshold	10
nat-trust-threshold	0
deny-period	60
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2009-04-22 19:26:23

ANNOTATION: The realm configuration **Enterprise** below represents the internal network on which the Avaya elements reside.

realm-config

identifier	Enterprise
description	
addr-prefix	0.0.0.0
network-interfaces	s0p0:0
mm-in-realm	enabled
mm-in-network	enabled
mm-same-ip	enabled
mm-in-system	enabled
bw-cac-non-mm	disabled
msm-release	disabled
generate-UDP-checksum	disabled
max-bandwidth	0
fallback-bandwidth	0
max-priority-bandwidth	0
max-latency	0
max-jitter	0
max-packet-loss	0
observ-window-size	0
parent-realm	
dns-realm	
media-policy	
in-translationid	
out-translationid	
in-manipulationid	
out-manipulationid	
manipulation-string	
class-profile	
average-rate-limit	0
access-control-trust-level	high
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
accounting-enable	enabled
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
codec-policy	

codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
match-media-profiles	
qos-constraint	
last-modified-by	admin@console
last-modified-date	2011-08-12 19:50:37

ANNOTATION: The session agent below represents Notification Solution used in this reference configuration.

session-agent	
hostname	ANS
ip-address	10.80.130.230
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	UDP+TCP
realm-id	Enterprise
egress-realm-id	
description	Avaya Notification Solution
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS ;hops=0
ping-interval	180
ping-send-mode	keep-alive

ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
reuse-connections	TCP
tcp-keepalive	enabled
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
last-modified-by	admin@console
last-modified-date	2011-08-17 17:36:26

ANNOTATION: The session agent below represents the AT&T IP Flexible Reach service border element.

session-agent	
hostname	135.242.225.210
ip-address	135.242.225.210
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	UDP
realm-id	ATT
egress-realm-id	
description	AT&T Border Element
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0

max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;hops=70
ping-interval	180
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
last-modified-by	admin@console
last-modified-date	2011-08-17 17:36:20

session-agent

hostname	1.1.1.1
ip-address	1.1.1.1
port	5060
state	disabled(Only enabled for failover testing)
app-protocol	SIP
app-type	
transport-method	UDP
realm-id	ATT
egress-realm-id	
description	AT&T Failover
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS ;hops=70
ping-interval	180
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
li-trust-me	disabled
in-manipulationid	
out-manipulationid	
manipulation-string	
p-asserted-id	

trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
last-modified-by	admin@console
last-modified-date	2011-08-17 17:36:20

ANNOTATION: The session-group is used for testing the failover scenario. In this case, an OPTION is sent to 1.1.1.1 and when no response is received, the 135.242.225.210 is tried

```

session-group
  group-name          SP_PROXY
  description
  state              enabled
  app-protocol        SIP
  strategy            RoundRobin
  dest
                    1.1.1.1
                    135.242.225.210

  trunk-group
  sag-recursion       enabled
  stop-sag-recurse    401,407
  last-modified-by    admin@135.9.62.155
  last-modified-date  2012-03-19 17:11:46

```

ANNOTATION: The sip-config defines global sip-parameters, including SIP timers, SIP options, which realm to send requests to if not specified elsewhere, and enabling the SD to collect statistics on requests other than REGISTERs and INVITEs.

```

sip-config
  state              enabled
  operation-mode      dialog
  dialog-transparency enabled
  home-realm-id       Enterprise
  egress-realm-id     Enterprise
  nat-mode            None
  registrar-domain
  registrar-host
  registrar-port      0
  register-service-route always
  init-timer          500
  max-timer           4000
  trans-expire        32
  invite-expire       180
  inactive-dynamic-conn 32
  enforcement-profile
  pac-method
  pac-interval        10
  pac-strategy         PropDist
  pac-load-weight      1
  pac-session-weight   1
  pac-route-weight     1
  pac-callid-lifetime  600
  pac-user-lifetime    3600
  red-sip-port         1988
  red-max-trans        10000
  red-sync-start-time  5000
  red-sync-comp-time   1000

```

add-reason-header	disabled
sip-message-len	4096
enum-sag-match	disabled
extra-method-stats	enabled
registration-cache-limit	0
register-use-to-for-lp	disabled
options	max-udp-length=0 set-inv-exp-at-100-resp
add-ucid-header	disabled
last-modified-by	admin@console
last-modified-date	2011-08-12 10:22:04

ANNOTATION: The SIP interface below is used to communicate with the AT&T IP Flexible Reach service.

sip-interface	
state	enabled
realm-id	ATT
description	
sip-port	
address	192.168.62.51
port	5060
transport-protocol	UDP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	
manipulation-string	
sip-ims-feature	disabled

operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
last-modified-by	admin@console
last-modified-date	2009-04-22 18:14:23

ANNOTATION: The SIP interface below is used to communicate with the Avaya elements.

sip-interface	
state	enabled
realm-id	Enterprise
description	
sip-port	
address	10.80.130.250
port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	30
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300

registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	
manipulation-string	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
last-modified-by	admin@console
last-modified-date	2009-04-16 18:07:58

ANNOTATION: The sip-manipulation modifies the **maxptime** attribute to **ptime**.

sip-manipulation

name	modifyMaxptime
description	Modify maxptime attribute
header-rule	
name	ReplaceMaxptime
header-name	Content-Type

action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	
element-rule	
name	modmline
parameter-name	application/sdp
type	mime
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	maxptime
new-value	ptime
last-modified-by	admin@console
last-modified-date	2011-10-22 19:25:08

ANNOTATION: The steering pools listed below define the RTP port range on the respective realms.

steering-pool	
ip-address	192.168.62.51
start-port	16384
end-port	32767
realm-id	ATT
network-interface	
last-modified-by	admin@console
last-modified-date	2011-08-25 19:11:47
steering-pool	
ip-address	10.80.130.250
start-port	16384
end-port	32767
realm-id	Enterprise
network-interface	
last-modified-by	admin@console
last-modified-date	2011-08-12 10:25:12
system-config	
hostname	Enterprise-Acme
description	
location	
mib-system-contact	
mib-system-name	
mib-system-location	
snmp-enabled	enabled
enable-snmp-auth-traps	disabled
enable-snmp-syslog-notify	disabled
enable-snmp-monitor-traps	disabled
enable-env-monitor-traps	disabled
snmp-syslog-his-table-length	1
snmp-syslog-level	WARNING
system-log-level	WARNING
process-log-level	NOTICE
process-log-ip-address	0.0.0.0

process-log-port	0
collect	
sample-interval	5
push-interval	15
boot-state	disabled
start-time	now
end-time	never
red-collect-state	disabled
red-max-trans	1000
red-sync-start-time	5000
red-sync-comp-time	1000
push-success-trap-state	disabled
call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	192.168.62.1
restart	enabled
exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	enabled
cli-more	disabled
terminal-height	24
debug-timeout	0
trap-event-lifetime	0
last-modified-by	admin@console
last-modified-date	2011-08-12 10:20:46

7. Verification Steps

7.1. General

The following steps may be used to verify the configuration:

- Place an inbound call from PSTN to ANS, and verify that an announcement is played. Interact with the ANS prompts and verify that ANS responds properly to the DTMF digits entered.
- From the ANS web interface, trigger a notification and verify the notification is delivered to the subscriber and the participant is able to confirm the receipt of notification.

7.2. Avaya Notification Solution

The following commands are issued from the ANS Web Portal.

1. Navigate to **System Maintenance**→**Channel Status** to verify the SIP trunk is properly configured. If **VOICE** shows **ACTIVE** status, then the configuration is good.

Channel	Present Status	IP Address
IP Phone	Active	10.80.130.230
Voice	Active	10.80.130.230
Email	Active	10.80.130.230

Channel Status

2. Navigate to **System Maintenance**→**Resource Manger Status** and make sure **Operational Status** of all the resource is **UP**.

Resource Name	IP Address	Priority	Capacity	Allocated Port	Failed Ports	Operational Status
ANS Media Server Ports		1	100	0	0	UP
ToSBC	10.80.130.250	1	100	0	0	UP
default		1	1000000	0	0	UP
atttest-VCD	10.80.130.230	1	600	0	0	UP
ANS Voice Notification Channel Ports		1	100	0	0	UP
ToSBC	10.80.130.250	1	100	0	0	UP
ToSBC-Extension	10.80.130.250	1	20	0	0	UP
ToSBC-PSTN	10.80.130.250	1	20	0	0	UP
ToSBC-Extension	10.80.130.250	1	20	0	0	UP
ToSBC-PSTN	10.80.130.250	1	20	0	0	UP

Resource Status

- The following screen shows the notification sent by ANS was successfully delivered to the subscriber.

AVAYA

Current Login: Thu Aug 16 17:46:36 UTC-6 2012

ANS Web Portal

Notification and Trigger

My Notification Scenarios

Notification Scenarios

My Notification History

Notification History

My Escalations

Escalations

Usage Report

Inbound Call Triggers

Email Trigger Configuration

Conference Bridges

Users and Groups

Notification History

You can view detailed notification history & terminate notification request here

Request Status

Recipient Info

Escalation Status

Message Details

Session Id

1344277051017

Start Time

2012-08-06 12:17:32.512000000

Status

Completed

Originator

admin

Duration(in Seconds)

30

Last Escalation Sequence

0

Users Affirmatively Responded

0

Total Users in Request

1

Request Time

2012-08-06 12:17:31.107000000

End Time

2012-08-06 12:18:03.125000000

Users Pending

0

Users Initiated

0

Users Notified

0

Users Responded

1

Users Errored

0

Notification Details

Notification Process Completed

Notification History

7.3. Avaya Media Server

The following commands are issued from the ANS Web Portal.

- Navigate to **System Status**→**Element Status** and verify the **Element Status** is **Normal** for proper operation of Avaya Media Server.

AVAYA

Avaya Media Server

Help | Log Out

System Status

Element Status

Cluster Status

Alarms

Logs

Monitoring

Applications

Cluster Configuration

System Configuration

Licensing

Tools

Security

Managing: attest, 10.80.130.230

[Home](#) » [System Status](#) » Element Status

Element Status

Click the element name to display the alarm viewer for this element.

StartStopRestartMore Actions

Refresh every: 5 seconds

Element Name: [attest](#)

UUID: 5b07a0e0-dfea-11e1-b8b8-005056aff75

Server Address: 10.80.130.230

Operating System: Linux

Service Status: Started

Operational State: Unlocked

Element Status: **Normal**

Alarm Description: No Alarm

Installed Software Packages

Avaya Media Server - v.7.5.0.724

A software based application platform that provides infrastructure for a diverse range of multimedia services.

AMS Status

8. Conclusion

As illustrated in these Application Notes, ANS and the Acme Packet SBC can be configured to interoperate successfully with the AT&T IP Flexible Reach service. This solution provides users of ANS the ability to send outbound notifications from the web interface; trigger outbound notification by dialing into ANS and initiate and hold adhoc conference calls.

The sample configuration shown in these Application Notes is representative of a basic enterprise 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. References

The Avaya product documentation is available at <http://support.avaya.com>.

[1] *ANS R2.0 Installation and Administration Guide*

[2] *ANS 2.0 Operation Guide*

Acme Packet Support (login required):

[3] <http://support.acmepacket.com>

AT&T IP Flexible Reach Service Descriptions:

[4] *AT&T IP Flexible Reach*

<http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-enterprise/>

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