



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Notification Solution, 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, and conference. It can be applied to emergency broadcast and system alarming.

An Acme Packet Net-Net (Acme 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 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
- 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 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 1.2 does not support Answering Machine Detection leaving a voicemail
2. ANS 1.2 only supports a ptime of 20 msecs
3. ANS 1.2 does not support retrieving notifications from the ANS server
4. ANS 1.2 does not support transfer of calls to a help desk

The test objectives stated in **Section 2** with limitations as 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.

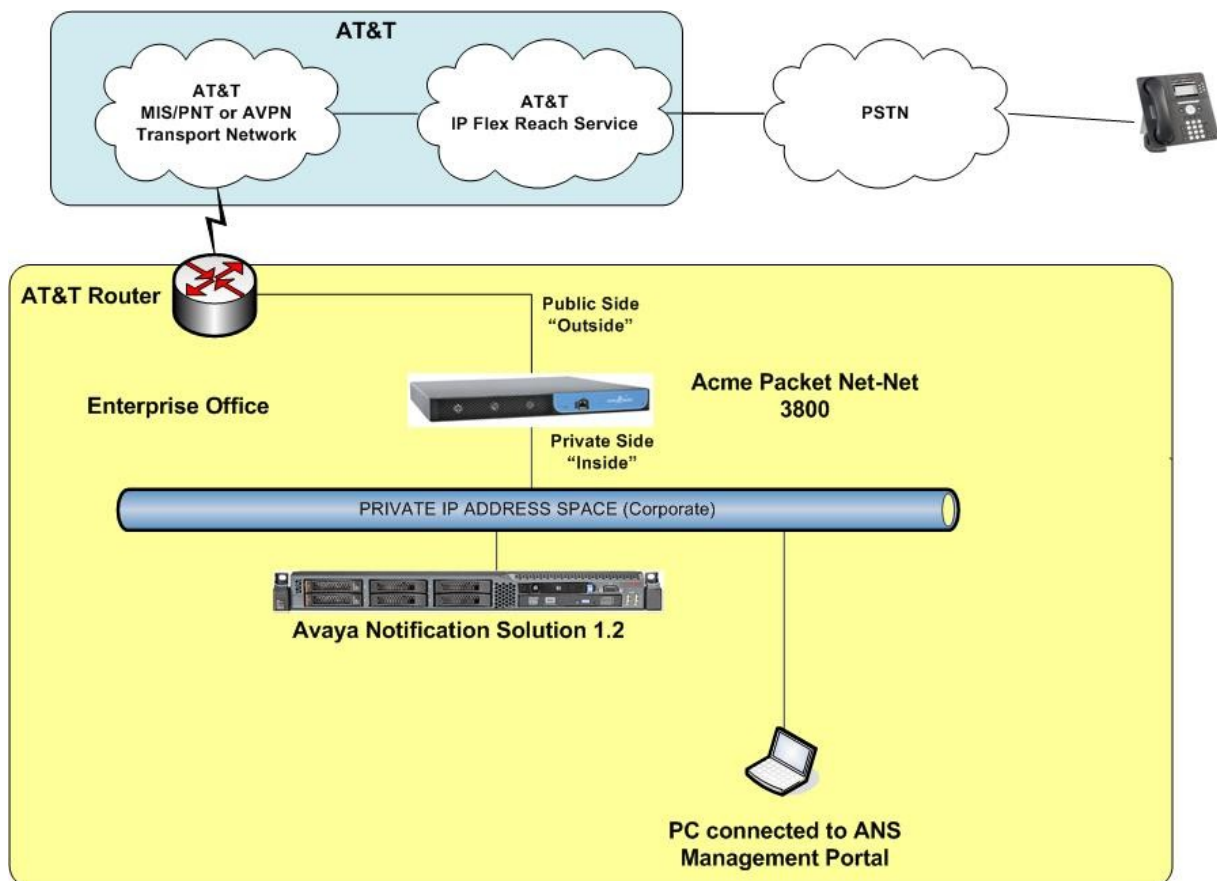


Figure 1: Reference Configuration

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	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

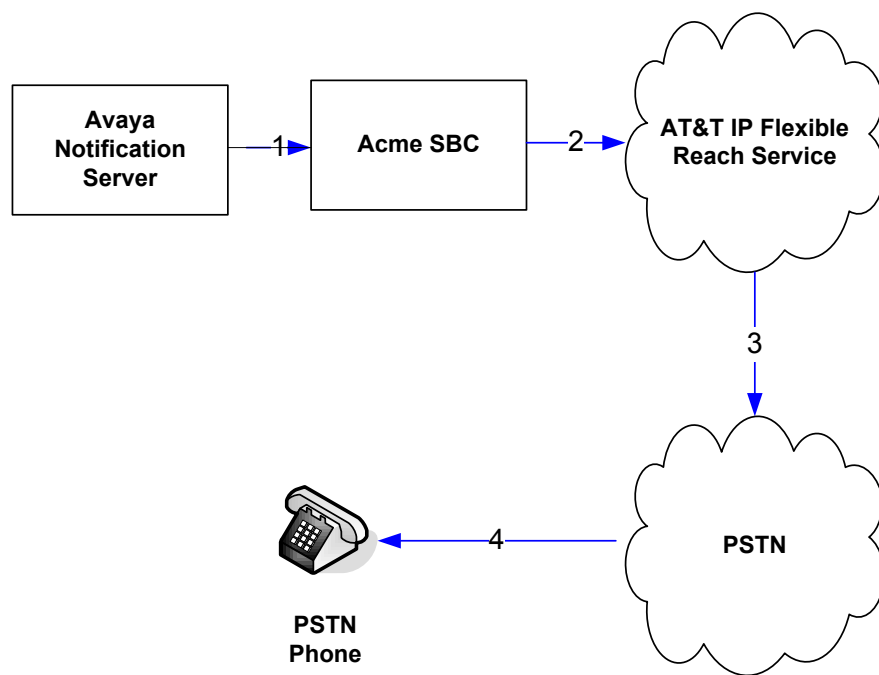
Table 1: Illustrative Values Used in this Reference Configuration

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.

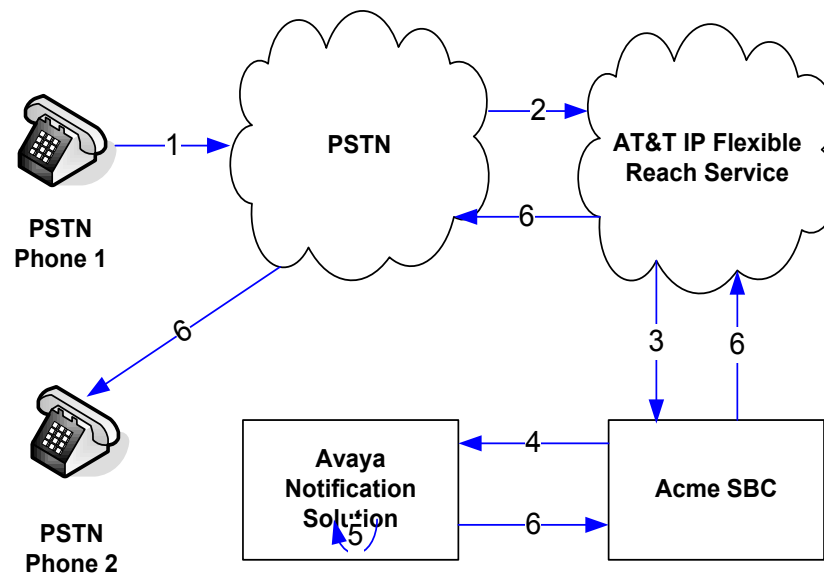
1. ANS originates a call to PSTN phone
2. Acme SBC performs any necessary SIP header modifications, and routes the call to the AT&T IP Flexible Reach Service
3. The AT&T IP Flexible Reach service routes the call to PSTN
4. PSTN delivers the notification call to a PSTN phone subscriber on ANS



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 SBC at CPE.
4. Acme SBC performs any necessary SIP header modifications, and routes the call to ANS
5. Based upon the option entered by PSTN Phone 1, ANS may send a notification to ANS subscriber/s on ANS.
6. Same as Steps 1 to 4 in the first scenario



Inbound Call Handled by ANS to trigger an Outbound Notification to the ANS subscribers

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 1.2 running on VMWare Virtual Machine	VMware vSphere ESX4.0 running Avaya Notification Solution 1.2 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 **Error! Reference source not found.** for further details if necessary.

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 also has a Text-to-Speech server for converting the Text entered while creating a notification, to equivalent speech.

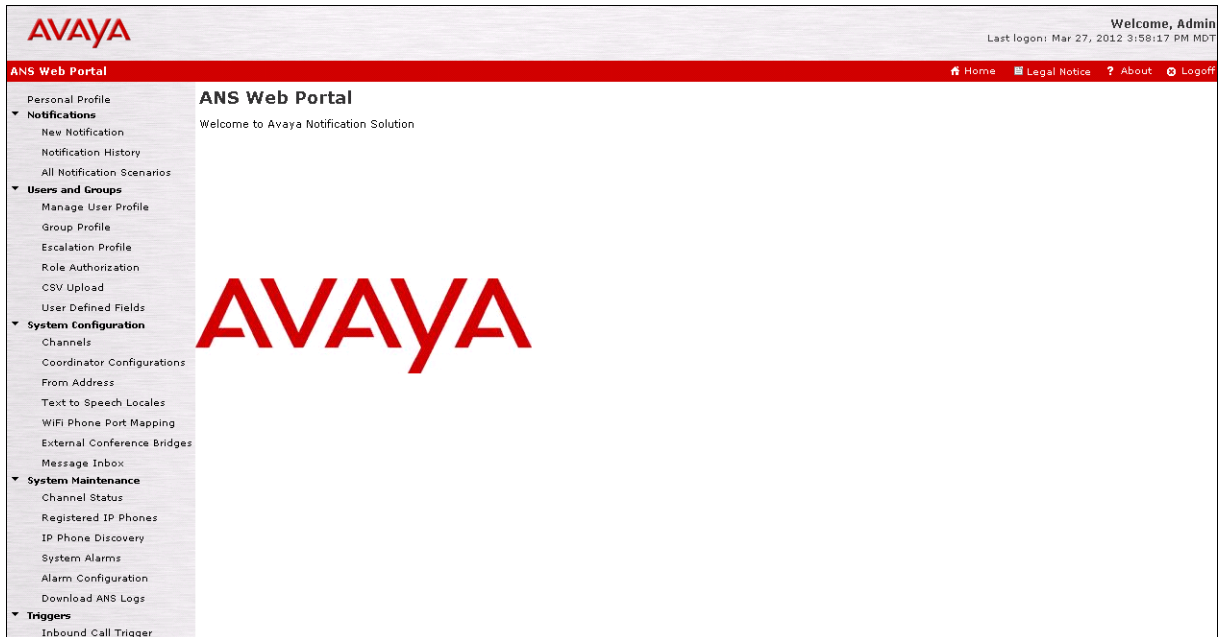
For this reference configuration, ANS was used to send notifications to its subscribers and trigger notifications for its subscribers by calling into the ANS system. Additionally, ad hoc conference feature for the ANS was tested by using inbound triggers or outbound notification for conference. When an inbound call arrives at ANS, it matches the user part of the **To** header in the INVITE to map to the appropriate notification. 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. VoIP Connection

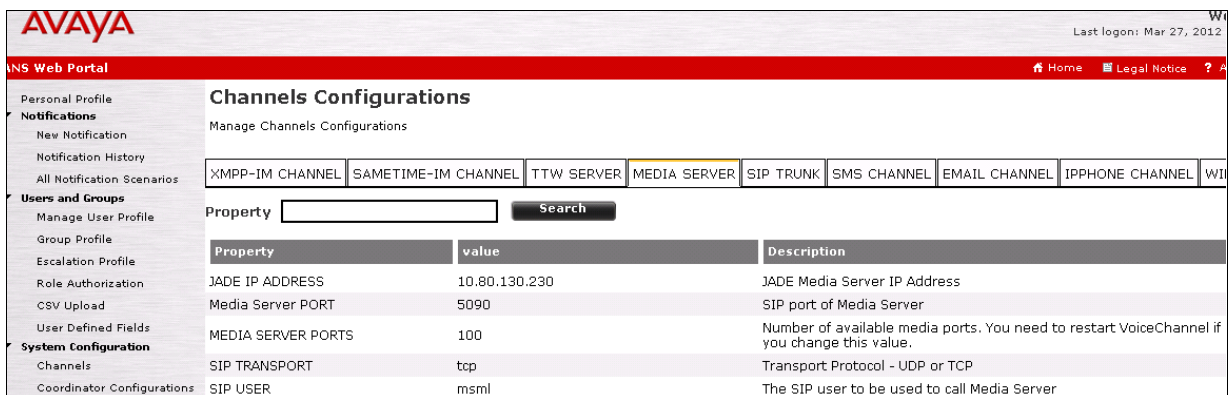
This section describes the steps to configure a SIP interface from ANS to the Acme SBC. **Note that only ONE SIP interface can be configured in ANS1.2.**

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




ANS Web Portal

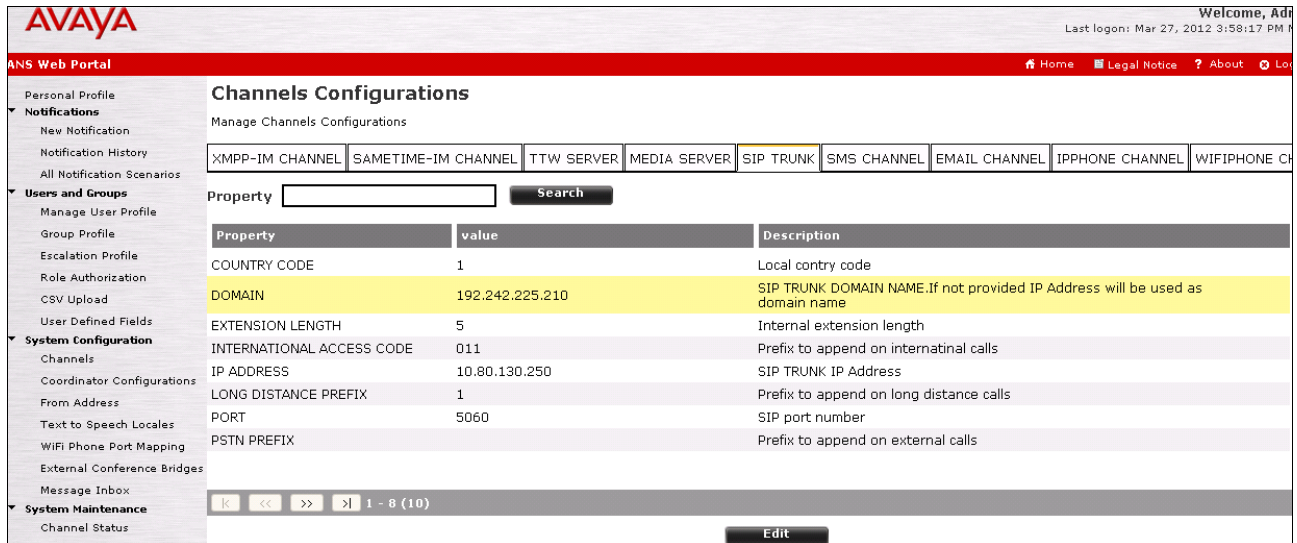
2. In the left pane, navigate to **System Configuration** → **Channels**. On the **Channels Configurations** page, select the **MEDIA SERVER** tab and verify **JADE IP ADDRESS** is set to the address of the ANS server, **Media Server PORT** field is set to **5090**, **MEDIA SERVER PORTS** is set to an appropriate value as per the number of licenses installed, **SIP TRANSPORT** is set to **tcp** and **SIP USER** is set to **msml**.



Channels Configurations – MEDIA SERVER

3. Select **SIP TRUNK** tab on the **Channels Configurations** page and configure as follows:
 - **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 SBC.
 - **IP ADDRESS** – Set to IP Address of the Acme SBC internal interface
 - **PORT** – Set to **5060**
 - Click  to display more entries on the **SIP TRUNK** form

Note: To change a particular property, highlight the field and click **Edit**



AVAYA Welcome, Admin Last login: Mar 27, 2012 3:58:17 PM

ANS Web Portal Home Legal Notice About

Channels Configurations
Manage Channels Configurations

XMP-IM CHANNEL SAMETIME-IM CHANNEL TTW SERVER MEDIA SERVER **SIP TRUNK** SMS CHANNEL EMAIL CHANNEL IPPHONE CHANNEL WIFIPHONE CHANNEL

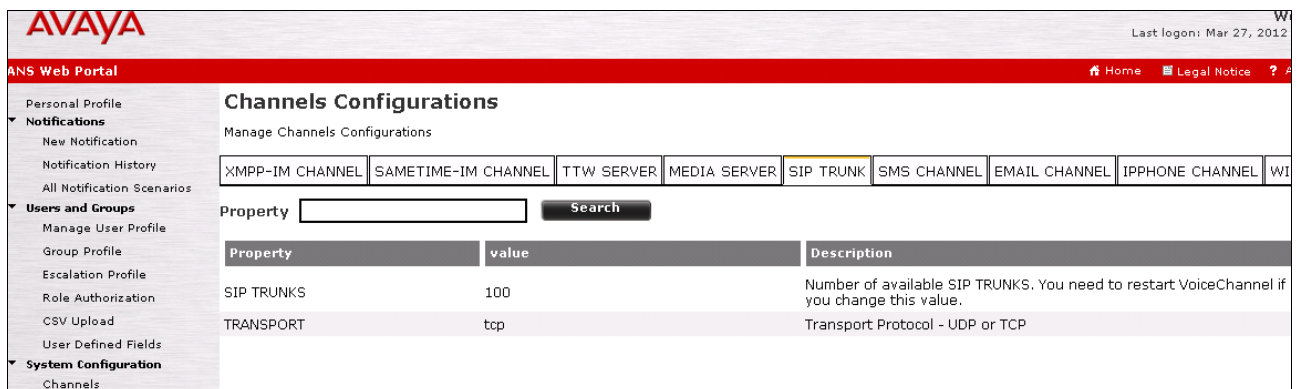
Property Search

Property	value	Description
COUNTRY CODE	1	Local country code
DOMAIN	192.242.225.210	SIP TRUNK DOMAIN NAME.If not provided IP Address will be used as domain name
EXTENSION LENGTH	5	Internal extension length
INTERNATIONAL ACCESS CODE	011	Prefix to append on international calls
IP ADDRESS	10.80.130.250	SIP TRUNK IP Address
LONG DISTANCE PREFIX	1	Prefix to append on long distance calls
PORT	5060	SIP port number
PTSTN PREFIX		Prefix to append on external calls

1 - 8 (10) Edit

Channels Configurations Page – SIP TRUNK

4. On the page displayed below, set the **SIP TRUNKS** field to the number of available licenses and set the **TRANSPORT** field to **tcp**



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ANS Web Portal Home Legal Notice About

Channels Configurations
Manage Channels Configurations

XMP-IM CHANNEL SAMETIME-IM CHANNEL TTW SERVER MEDIA SERVER **SIP TRUNK** SMS CHANNEL EMAIL CHANNEL IPPHONE CHANNEL WIFIPHONE CHANNEL

Property Search

Property	value	Description
SIP TRUNKS	100	Number of available SIP TRUNKS. You need to restart VoiceChannel if you change this value.
TRANSPORT	tcp	Transport Protocol - UDP or TCP

Channels Configurations Page – SIP TRUNK (cont.)

5.3. From Address Configuration

This section describes steps to configure the user part for the **From** header to be sent in an outbound call.

1. Navigate to **System Configuration**→**From Address** and select **VOICE** in **Channel Types** drop-down box.

AVAYA

ANS Web Portal

Personal Profile

▼ **Notifications**

- New Notification
- Notification History
- All Notification Scenarios

▼ **Users and Groups**

- Manage User Profile
- Group Profile
- Escalation Profile
- Role Authorization
- CSV Upload
- User Defined Fields

▼ **System Configuration**

- Channels
- Coordinator Configurations
- From Address

From Address Configurations

Manage From Address Configurations

Please select the Channel type

Channel Types

*** Address**

--	--

Add

Display Name

--	--

Add

From Address Configurations

2. Click **Add** in the **Address** section of the **From Address Configurations** page, and in the pop-up screen displayed, enter the phone number in the **From Address** field to be sent in the **From** header for an outbound call and click **Save**.

The screenshot shows the Avaya ANS Web Portal interface. The left sidebar contains navigation links under categories like 'Personal Profile', 'Notifications', 'Users and Groups', 'System Configuration', and 'System Maintenance'. The main content area is titled 'From Address Configurations' and includes a sub-header 'Manage From Address Configurations'. Below this, there's a prompt 'Please select the Channel type' with a dropdown menu set to 'VOICE'. There are two sections for adding configurations: one for '* Address' and another for 'Display Name', each with an 'Add' button. At the bottom, there's a table with one row showing 'From Address' as '7325551212' and 'Default' as checked. A red circle highlights this row. At the very bottom are 'Save' and 'Close' buttons.

From Address Configuration page – Add Address

3. Click **Add** in the **Display Name** section of the **From Address Configurations** page and enter any value in the **Display Name** field and click **Save**. Multiple Display names can be configured in this step.

ANS Web Portal

From Address Configurations

Manage From Address Configurations

Please select the Channel type

Channel Types

*** Address**

7325550167

7325551212

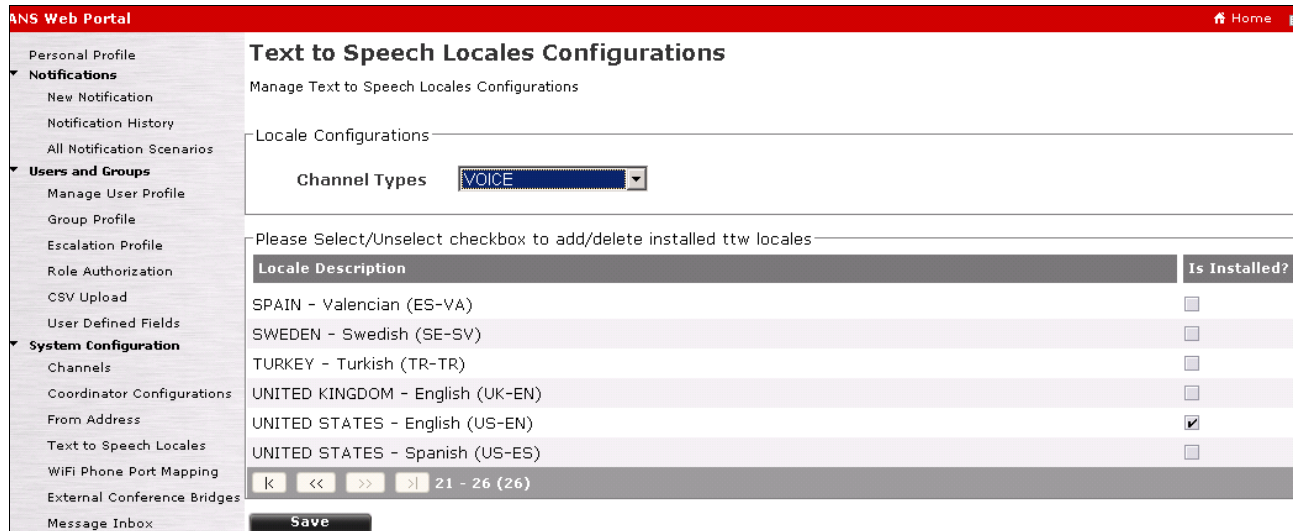
Display Name
<input type="text" value="Sonus1212"/>
Default <input checked="" type="checkbox"/>

From Address Configuration – Add Display Name

5.4. 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** → **Text to Speech Locales** and check the appropriate value in the **Is Installed** field. Click **Save**.



ANS Web Portal Home

Text to Speech Locales Configurations
Manage Text to Speech Locales Configurations

Locale Configurations

Channel Types: **VOICE**

Please Select/Unselect checkbox to add/delete installed ttw locales

Locale Description	Is Installed?
SPAIN - Valencian (ES-VA)	<input type="checkbox"/>
SWEDEN - Swedish (SE-SV)	<input type="checkbox"/>
TURKEY - Turkish (TR-TR)	<input type="checkbox"/>
UNITED KINGDOM - English (UK-EN)	<input type="checkbox"/>
UNITED STATES - English (US-EN)	<input checked="" type="checkbox"/>
UNITED STATES - Spanish (US-ES)	<input type="checkbox"/>

k << >> >| 21 - 26 (26)

Save

Text to Speech Locales Configurations Page

5.5. Manage User Profile

In the left pane, navigate to **Users and Groups**→**Manage User Profile** and click on **Add** on the **Manage User Profile** page (not shown). On the **Create User Profile** page shown below, configure as follows:

- **User ID** – Set to any descriptive value
- **First Name** – Set to any unique value
- **Last Name** – Set to any name
- **Work Phone** – Set to the phone number of a subscriber
- Click **Save** (not shown)

Note: If other phone numbers are entered then the notification will be sent simultaneously to the other phone numbers too.

AVAYA

Welcome,
Last logon: Mar 27, 2012 3:58:17 PM

ANS Web Portal

HomeLegal NoticeAbout

Personal Profile

Notifications

Users and Groups

System Configuration

System Maintenance

Triggers

Create User Profile

Create User Profile

* Indicates required field. In phone numbers fields please enter numeric or +(numeric)

User Profile

* User ID: Sonus1212

Title:

* First Name: 1212

Middle Name:

Last Name: Sonus

Telephone Security PIN:

Role: User

Web Login Password:

Activate ☒

Contact Information

	Contact Information	Urgent Notifications Rule(seconds)	Normal Notifications Rule(seconds)
Work Phone	3035551212	0	0
Mobile Phone		0	0
Home Phone		0	0
IP Phone Extension		0	0
Work Email		0	0
Personal Email		0	0
SMS Address		0	0
Sametime Address		0	0
XMPP Address		0	0
WIFI PHONE		0	0

Create User Profile Page

5.6. 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**→**All Notification Scenarios** and click **Add** on the **All Notification Scenarios** page (not shown). On the **Details** step shown below, configure as follows:
 - **Scenario Name** – Enter any descriptive name
 - **Description** – Enter description of the notification
 - **Priority** – Set a valid priority
 - **Expiration Time** – Set the time for the notification to expire
 - Click **Next**

The screenshot shows the AVAYA ANS Web Portal interface. The left sidebar contains a navigation menu with categories: Personal Profile, Notifications (expanded), Users and Groups, and System Configuration. Under Notifications, 'All Notification Scenarios' is selected. The main content area is titled 'All Notification Scenarios' and includes a sub-header 'You can manage (add, edit & delete) notification template here'. A progress bar at the top indicates three steps: 1 Details (active), 2 Messages, and 3 Recipients. The 'Details' step contains the following fields: 'Scenario Name' (text box with 'SonusVoiceOutbound'), 'Description' (text box with 'Outbound Notification for participants on Sonus Network'), 'Priority' (radio buttons for Normal, Urgent, and Crisis, with 'Normal' selected), and 'Expiration Time' (text box with '5' and a dropdown menu set to 'MINUTES'). At the bottom right are 'Cancel' and 'Next>>' buttons.

All Notification Scenarios – Details

2. Click **Add Message**

The screenshot shows the AVAYA ANS Web Portal interface at the 'Messages' step. The left sidebar is the same as in the previous screenshot. The main content area is titled 'All Notification Scenarios' and includes the same sub-header. The progress bar now shows '2 Messages' as the active step. Below the progress bar, the 'Scenario Name' and 'Description' fields are visible. A new section titled 'Messages' with an envelope icon is shown. It contains a table with a single row: 'Messages' | '0 - 0 (0)'. Below the table are navigation buttons: '<', '<<', '>>', and '>'. At the bottom right is an 'Add Message' button.

All Notification Scenarios – Message

3. Configure notification as follows:
 - **Channel** – Select **VOICE** from the drop-down box
 - **Caller Id** – Set to one of the values configured in **Section 5.3, Step 2**
 - **Display Name** – Set to one of the values configured in **Section 5.3, Step 3**
 - **User Custom Prompt** - Enter any text which is played by Text to Speech server when the subscriber picks up the phone
 - **Message Body** – Enter the message/notification to be delivered
 - Click **Save** at the bottom of the screen (not shown)

General		Questions	
<i>* Required Fields: 1.Message Body 2.Use Custom Prompt must be provided</i>			
* Channel	VOICE	* Locale	UNITED STATES - English
Display Name	Sonus1212	Caller Id:	7325551212
Ring Timeout (seconds)		Retry Count	0
Authenticate Recipient?	<input type="checkbox"/>	Retry Delay (seconds)	0
Bypass Human detection?	<input type="checkbox"/>		
Audio Conference	<input type="checkbox"/> ANS Ad-Hoc Conference		
* Use Custom Prompt:	You have an important message This message is for subscribers on Sonus Network.		
	Select Wave File Record Through Telephone		
* Message Body	This message is for subscribers on Sonus Network.		
	Select Wave File Record Through Telephone		

All Notification Scenarios – General

4. In case an adhoc conference is configured (which was 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. Click **Questions**.

General		Questions	
<i>* Required Fields: 1.Message Body 2.Use Custom Prompt must be provided</i>			
* Channel	VOICE	* Locale	UNITED STATES - English
Display Name	ANS	Caller Id:	7325551212
Ring Timeout (seconds)	20	Retry Count	0
Authenticate Recipient?	<input type="checkbox"/>	Retry Delay (seconds)	0
Bypass Human detection?	<input type="checkbox"/>		
Audio Conference	<input checked="" type="checkbox"/> ANS Ad-Hoc Conference		
* Use Custom Prompt:	This is a notification from Avaya. This is a test call for joining the conference.		
	Select Wave File Record Through Telephone		
* Message Body	This is a test call for joining the conference.		
	Select Wave File Record Through Telephone		

All Notification Scenarios – General (Audio Conference)

- Click **Add a question**. Note that questions can be added for any notification scenario but in this reference configuration, it was only added for conferencing.

All Notification Scenarios – Question

- Enter any relevant text in **Question 1 Content** field and click **Save** (not shown).

All Notification Scenarios – Add a Question

- Highlight the question and click **Add a Choice**.

All Notification Scenarios – Add a Choice

- In the choice details, text is entered for choice fields. In this reference configuration, only text was entered which is converted to speech using the TTS server on ANS. The following choice relates to a response from subscriber to join the conference.

- Choice 1 Content** – Enter any informative text
- Acknowledgement after chosen** – Enter any informative text
- Mark this choice as the “Affirmative” answer for reporting purposes** – Enabled
- Audio Conference** – Enabled
- Click **Save** (not shown)

All Notification Scenarios – Add a Choice (Details)

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

* Choice 2	Press 2 for No	Select Wave File	Record Through Telephone
Content			
Acknowledgment after chosen	Goodbye!	Select Wave File	Record Through Telephone
<input type="radio"/> Mark this choice as the "Affirmative" answer for reporting purposes			
Transfer			
<input type="radio"/> Phone Number <input type="text"/>			
<input type="radio"/> Audio Conference			
<input checked="" type="radio"/> No Action			

All Notification Scenarios – Add a Choice (Details)

10. The screen below shows the choice configured in previous steps. Click **Save** (not shown).

General	Questions
Add Questions	
Question Id	Questions
Question 1	Are you available for the conference call now?
<input type="button" value="K"/> <input type="button" value="Left"/> <input type="button" value="Right"/> <input type="button" value="End"/> 1 - 1 (1)	
<input type="button" value="Add a question"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/> <input type="button" value="Clear Selection"/>	
Add Choices	
Choice Id	Choices
Choice 1	Press 1 for yes
* Choice 2	Press 2 for No

All Notification Scenarios – Choices (Final)

11. The screen below displays the configured message. Click **Next**.

ANS Web Portal	
<ul style="list-style-type: none"> Personal Profile Notifications <ul style="list-style-type: none"> New Notification Notification History All Notification Scenarios Users and Groups <ul style="list-style-type: none"> Manage User Profile Group Profile Escalation Profile Role Authorization CSV Upload User Defined Fields System Configuration <ul style="list-style-type: none"> Channels Coordinator Configurations From Address Text to Speech Locales WiFi Phone Port Mapping External Conference Bridges Message Inbox 	<h3>All Notification Scenarios</h3> <p>You can manage (add, edit & delete) notification template here</p> <p>1 Details → 2 Messages → 3 Recipients</p> <p>Scenario Name: <input type="text" value="SonusVoiceOutbound"/> Description: <input type="text" value="Outbound Notification to participants on Sonus Network"/></p> <p>Messages</p> <p>VOICE</p> <p><input type="button" value="K"/> <input type="button" value="Left"/> <input type="button" value="Right"/> <input type="button" value="End"/> 1 - 1 (1)</p> <p><input type="button" value="Add Message"/></p> <p><input type="button" value="Previous"/> <input type="button" value="Cancel"/> <input type="button" value="Next"/></p>

All Notification Scenarios – Message (Final)

12. Click on **Users** to add the recipients of this notification

AVAYA
ANS Web Portal

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

1 Details → 2 Messages → 3 Recipients

Scenario Name: SonusVoiceOutbound Description: Outbound Notification to participants on Sonus Network

Users
Groups
Escalations

<<Previous Cancel Save

All Notification Scenarios – Recipients

13. Highlight the recipients in the **Users** area to send the notification and then click **Add** to move them to the **Selected Users** area

Users

* Please click on Conf Role column to change the conference role of the user/group/escalation.

Search value: Search

User Id	First Name	Middle Name	Last Name
NSN3908	3908		NSN
Sonus1212	1212		Sonus
Sonus1760	1760		Sonus
Sonus1761	1761		Sonus
Sonus1932	1932		Sonus

Add >>> <<< Remove

User Id	First Name	Middle Name	Last Name	Conf Role
---------	------------	-------------	-----------	-----------

6 - 10 (13) 0 - 0 (0)

All Notification Scenarios – Recipients (Cont.)

14. The screen below shows the final screen with recipients. Click **Save** (not shown)

Users

* Please click on Conf Role column to change the conference role of the user/group/escalation.

Search value: Search

User Id	First Name	Middle Name	Last Name
NSN3908	3908		NSN
Sonus1212	1212		Sonus
Sonus1760	1760		Sonus
Sonus1761	1761		Sonus
Sonus1932	1932		Sonus

Add >>> <<< Remove

User Id	First Name	Middle Name	Last Name	Conf Role
Sonus1212	1212		Sonus	Participant
Sonus1760	1760		Sonus	Participant

6 - 10 (13) 1 - 2 (2)

All Notification Scenarios – Recipients (Final)

15. The following screen shows all the notifications configured in this reference configuration

The screenshot displays the Avaya ANS Web Portal interface. The left sidebar contains navigation links: Personal Profile, Notifications (expanded), Users and Groups, and System Configuration. The main content area is titled 'All Notification Scenarios' and includes a search bar and a table of configured scenarios.

Scenario Name	Scenario Creator	Scenario Description
NSNVoiceOutbound	admin	Outbound Notification for participants on NSN network
SonusVoiceOutbound	admin	Outbound Notification to participants on Sonus Network
TDMVoiceOutbound	admin	Outbound Notification for participants on TDM Gateway
test_conf_call	admin	

All Notification Scenarios configured

5.7. 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 **Triggers→Inbound Call Triggers** (not shown) and click on **Add** (not shown) on the **Manage Inbound Call Trigger** page (not shown). In the General section shown below, configure as follows:
 - **Inbound Call Trigger Name** – Enter any descriptive value
 - **Inbound Call Trigger Description** – Enter any descriptive string
 - **Locale** – Select **UNITED STATES – English(US-EN)** from the drop-down list
 - **Greeting Prompt** – Enter any text which is converted to speech and played to the caller when inbound call is received by ANS
 - Click **Add** to add an inbound number

AVAYA
ANS Web Portal
Manage Inbound Call Trigger
You can manage (add, edit & delete) Inbound Call trigger profiles here
Please search based on Trigger Name & Trigger Description
General Choice
Inbound Call Trigger Name* InboundCallTrigger
Inbound Call Trigger Description* InboundCallTrigger
Locale UNITED STATES - English (US-EN)
Greeting Prompt* Welcome to Avaya Select Wave File Record Through Telephone
Trigger Access Pin
Inbound Numbers
InBound Number
0 - 0 (0)
Add
Allow Inbound Numbers
Add inbound number

Manage Inbound Call – General

2. Enter a valid phone number in the **InBound Number** field and click **Save**.

InBound Number 7323331111
Save Cancel
Save

Manage Inbound Call – Inbound Number

3. Select **Choice** and click **Add**

Choice Sequence	Choice Description
< << >> >	0 - 0 (0)

Add

Manage Inbound Call – Choice

4. On the Choice pop-up screen, configure as follows:
- Choice – Enter any descriptive string which is converted into speech when an inbound call comes into ANS
 - Single Scenario – Select this option
 - Click on **Select** next to the **Select Scenario** (not shown) and select the corresponding Notification to be triggered from the drop-down list. These notifications were configured in **Section 5.6**.
 - Click **Add Scenario**
 - Click **Save** (not shown)

Choice: tification call, to subscribers on, Sonus Network. [Select Wave File] [Record Th...]

☐ Enable Trigger Pin ☐ Enable Message Recording ☐ Enable Conferencing

☒ Single Scenario ☐ Multiple Scenario

☐ Single InBox Store ☐ Multiple InBox Store

☐ Single InBox Retrieve ☐ Multiple InBox Retrieve

Attach Single : **Select Scenario**

Scenario Nam

SonusVoiceOutbound
NSNVoiceOutbound
SonusVoiceOutbound
TDMVoiceOutbound
test conf call
test voice outbound

[Add Scenario] [Cancel]

Manage Inbound Call – Choice to trigger Outbound Notification

5. Following screen shows the Choice configured for conferencing and sending a conference notification to the subscriber.

Manage Inbound Call – Choice to trigger Outbound Conference Notification

6. Following screen shows all the choices configured for the Inbound Trigger in this reference configuration

Manage Inbound Call – Choice (Final)

5.8. Configuring Codec and RTP ports Offered by Avaya Notification Solution

This configuration is required to change the codecs and payload type offered and their order by ANS. Additionally, AT&T requires the RTP port range of 16384 to 32767.

When ANS sends an outbound notification, AT&T IP Flexible Reach service prefers a G729 codec.

- Access the ANS via the command line interface and navigate to the **/opt/rhel48/usr/Avaya/MediaServer/config** directory
- Edit the **audioPreferences.cfg** file
- Change the order of the codec and their corresponding payload types

Configure the RTP ports in the range of 16384 to 32767 as follows:

- Access the ANS via the command line interface and navigate to the **/opt/rhel48/usr/Avaya/MediaServer/config** directory
- Edit the **softMediaServer.cfg** file
- Search for **baseRtpPort** and change the value to **16384**
- Search for **maxChannels** and change the value to **5000** (Maximum allowed in ANS 1.2)
- Change the order of the codec and their corresponding payload types

After saving the file with the changes, enter the following commands:

- **/etc/init.d/acms stop**
- **/etc/init.d/acms start**

6. Configure Acme Packet Session Border Controller

The Acme 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. 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
dnsalg-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                       slp0
  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
realm-id
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 below defines 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	205.168.62.1
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 System Manager console.

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.

The screenshot displays the Avaya ANS Web Portal interface. The top header features the Avaya logo and the text "ANS Web Portal". A left-hand navigation menu is visible, containing sections for "Personal Profile", "Notifications", "Users and Groups", "System Configuration", and "System Maintenance". The "Channel Status" page is selected, showing a table of channel statuses. The table has two columns: "Channel" and "Status". The rows listed are EMAIL (ACTIVE), IPPHONE (ACTIVE), and VOICE (ACTIVE). A "k" icon is visible in the bottom right corner of the table area.

Channel	Status
EMAIL	ACTIVE
IPPHONE	ACTIVE
VOICE	ACTIVE

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

AVAYA Welcome, Last login: Mar 27, 2012 5:25:00 PM

ANS Web Portal Home Legal Notice About

Personal Profile

Notifications

New Notification

Notification History

All Notification Scenarios

Users and Groups

Manage User Profile

Group Profile

Escalation Profile

Role Authorization

CSV Upload

User Defined Fields

System Configuration

Channels

Coordinator Configurations

From Address

Text to Speech Locales

WiFi Phone Port Mapping

External Conference Bridges

Message Inbox

System Maintenance

Channel Status

Registered IP Phones

IP Phone Discovery

System Alarms

Alarm Configuration

Download ANS Logs

Triggers

Inbound Call Trigger

Notification History

You can view detailed notification history & terminate notification request here

Request Status

Session Id:	1330547159123	Request Time:	2012-02-29 13:25:59.223
Start Time:	2012-02-29 13:25:59.354	End Time:	2012-02-29 13:26:19.849
Status:	Completed	Users Notified:	0
Originator:	ivrtrigger	Users Contacted:	0
Duration(in Seconds):	20	Users Responded:	1
Last Escalation Sequence	0	Total Users in Request 1	
Affirmative Response	0		

Notification Details: Notification Process Completed

Conference Status

Conference status is not available for this request

Recipient Status

Recipient Name Search

Recipient Name	First Name	Last Name	Status
test1	test1		Responded

1 - 1 (1)

Point Of Contact Status

Point Of Contact	Address	Status	Timestamp	Details
WorkPhone	3143463907	Pending	2012-02-29 13:26:03.236	
WorkPhone	3143463907	Initiated	2012-02-29 13:26:13.123	
WorkPhone	3143463907	Notified	2012-02-29 13:26:20.022	
WorkPhone	3143463907	Responded	2012-02-29 13:26:32.622	Notification Delivered.

Escalation Status

Status	Level	Description	Trace Time Stamp
ESCALATION_PENDING	0		2012-02-29 13:25:59.408

Message Details

Messages

VOICE

8. Conclusion

As illustrated in these Application Notes, ANS and the Acme 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 R1.2 Administration Guide*, April 2011

[2] *Avaya Notification Solution 1.2 SPI Administration Guide*, August 2011

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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