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) 3800[1] provides SIP Session Border Controller functionality between the AT&T IP Flexible Reach service and the enterprise internal network[2]. UDP transport protocol is used between the Acme Packet Net-Net SD and the AT&T IP Flexible Reach service.

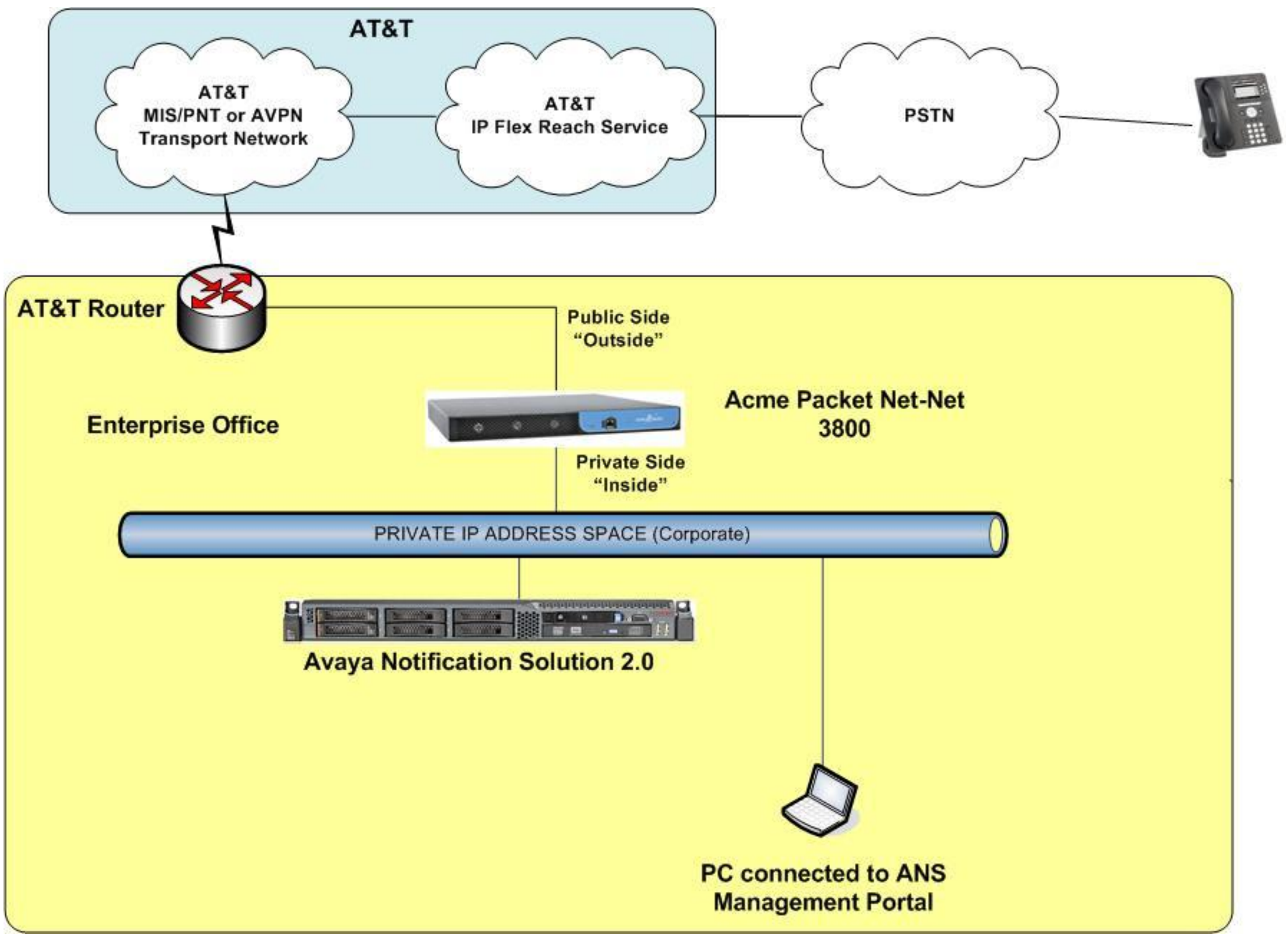


Figure 1: Reference Configuration

[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

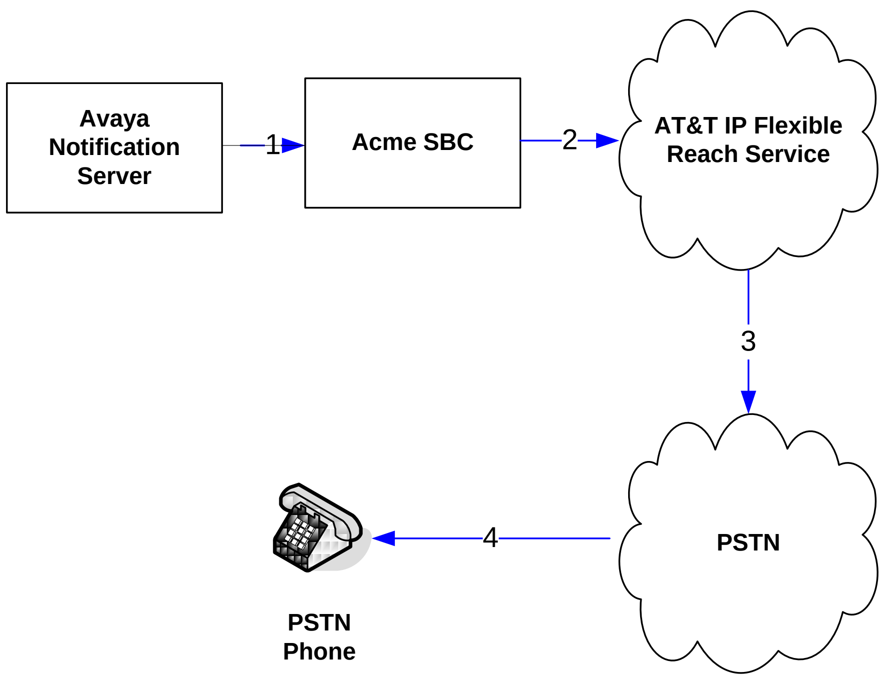
Table 1: Illustrative Values Used in this Reference Configuration

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.

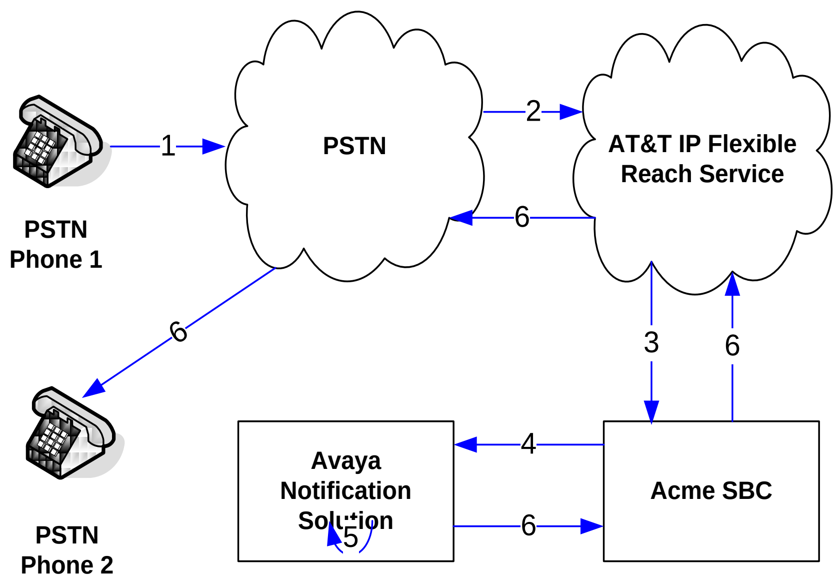
1. ANS originates a call to PSTN phone
2. Acme Packet SBC performs any necessary SIP header modifications, and routes the call to the AT&T IP Flexible Reach Service
3. The AT&T IP Flexible Reach service routes the call to PSTN
4. PSTN delivers the notification call to a PSTN phone subscriber on ANS.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

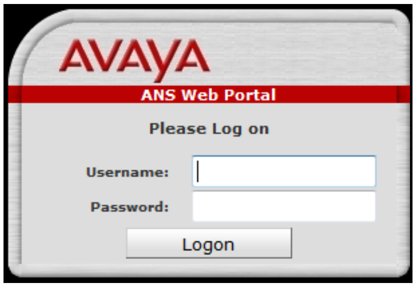
For this reference configuration, ANS was used to send notifications to its subscribers and trigger notifications for it subscribers by calling into the ANS system. Additionally, ad hoc conference feature for the ANS was tested by using inbound triggers or outbound notification for conference. Additionally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

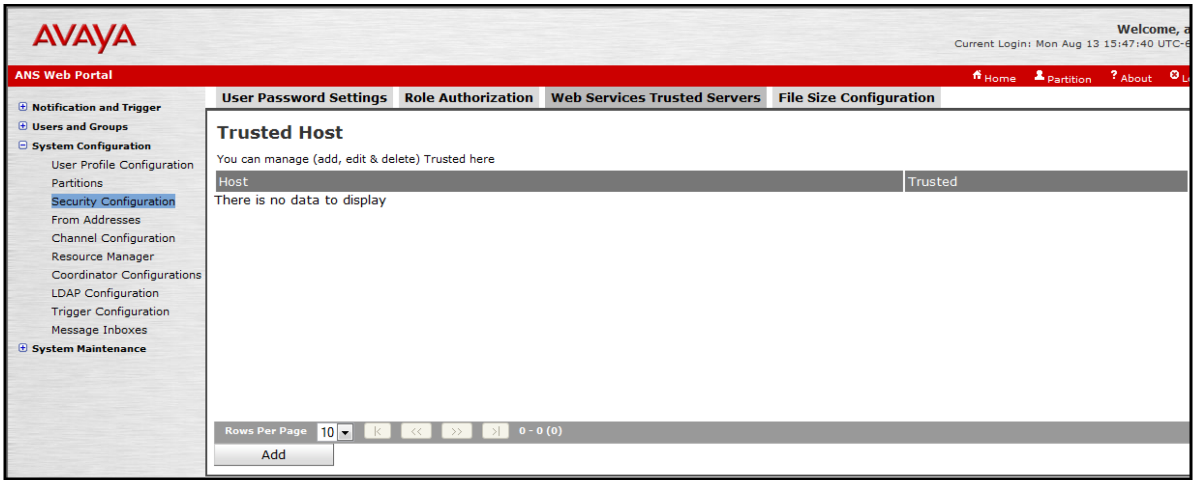


ANS Login Screen

5.3. Security Configuration

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

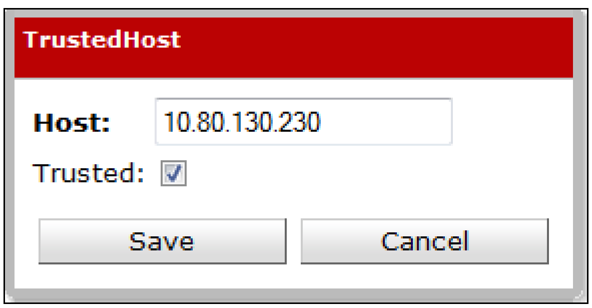
1. Navigate to **System Configuration→Security Configuration** and select **Web Services Trusted Servers** tab. Click **Add**.



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**

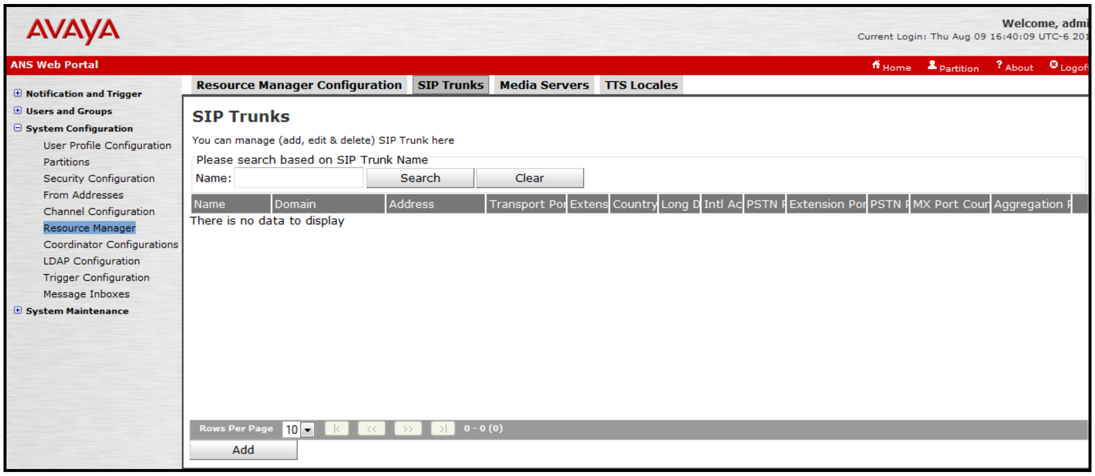


Security Configuration – TrustedHost

5.4. VoIP Connection

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

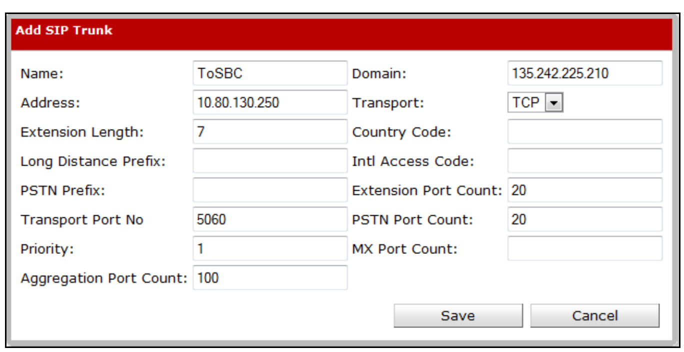
1. In the left pane, navigate to **System Configuration→Resource Manager** and select **SIP Trunks** tab. On the **SIP Trunks** page, click **Add**.



Resource Manager – SIP Trunks

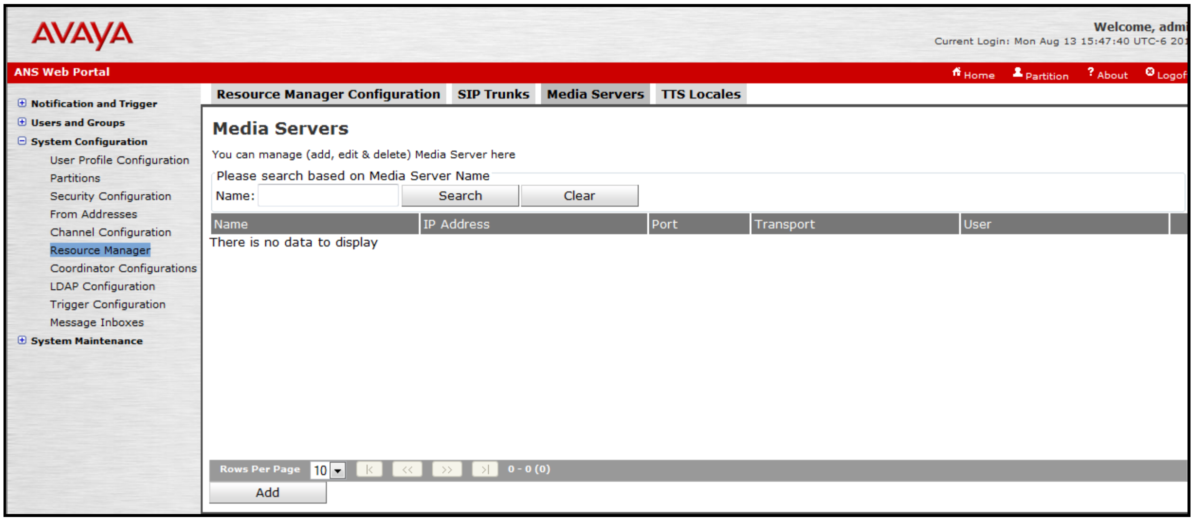
On the pop-up screen **Add SIP Trunk** shown below, configure as follows:

- **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**



Resource Manager – Add SIP Trunk

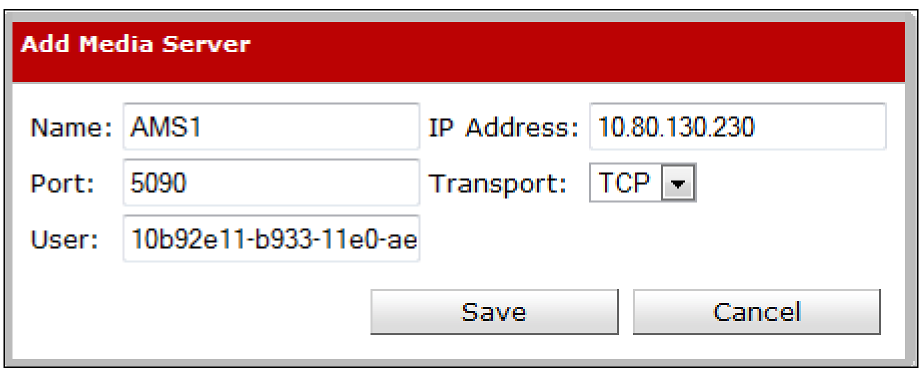
2. In the left pane, navigate to **System Configuration→Resource Manager** and select **Media Servers** tab. On the **Media Servers** page, click **Add**



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**

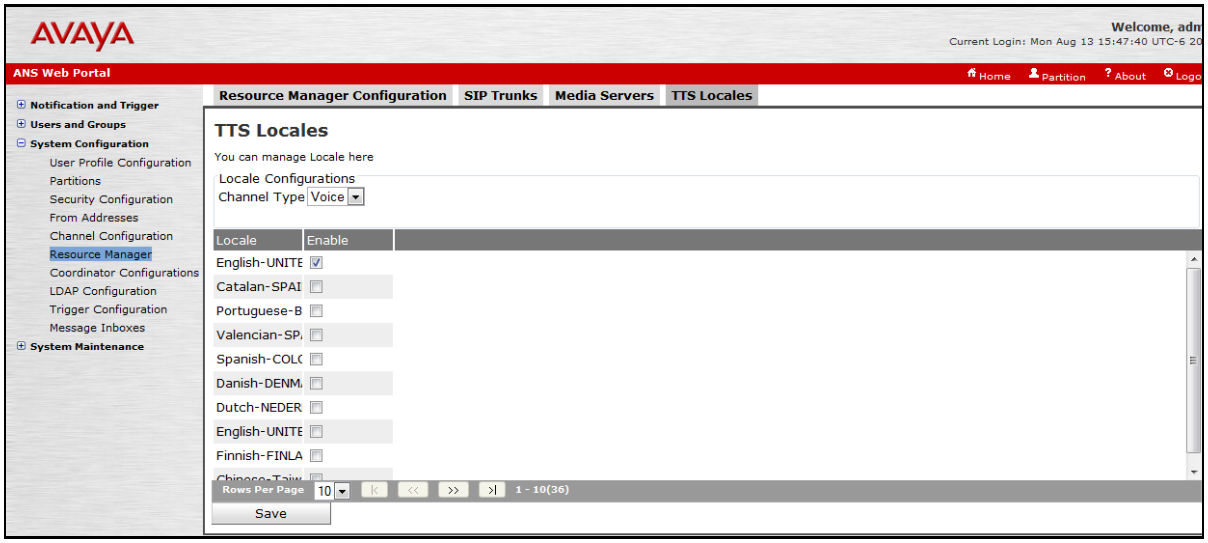


Resource Manager – Add Media Server

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**.

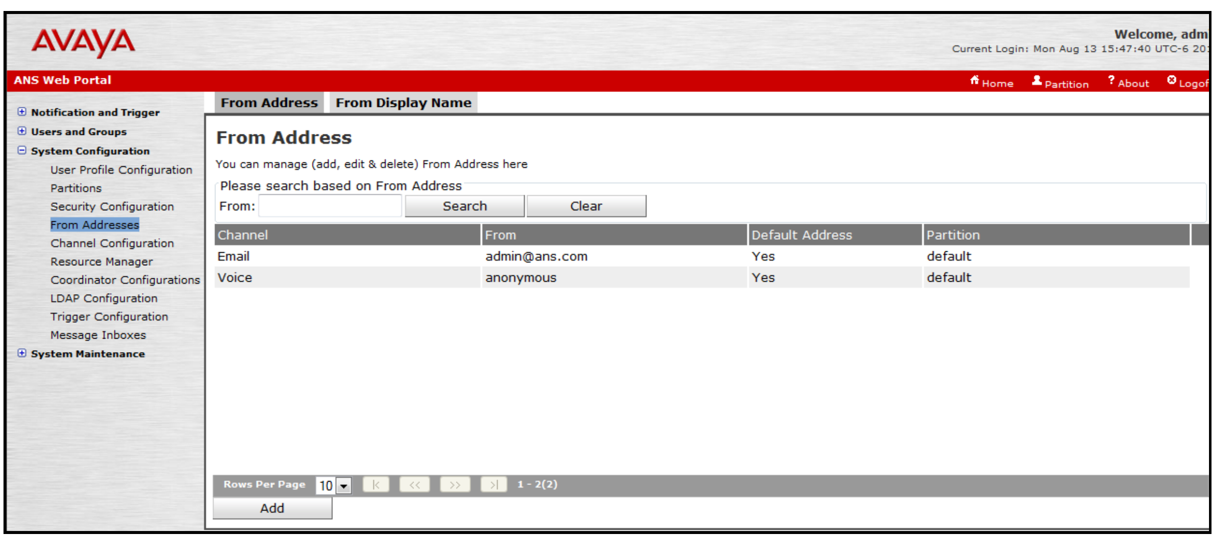


Resource Manager – TTS Locales

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**.



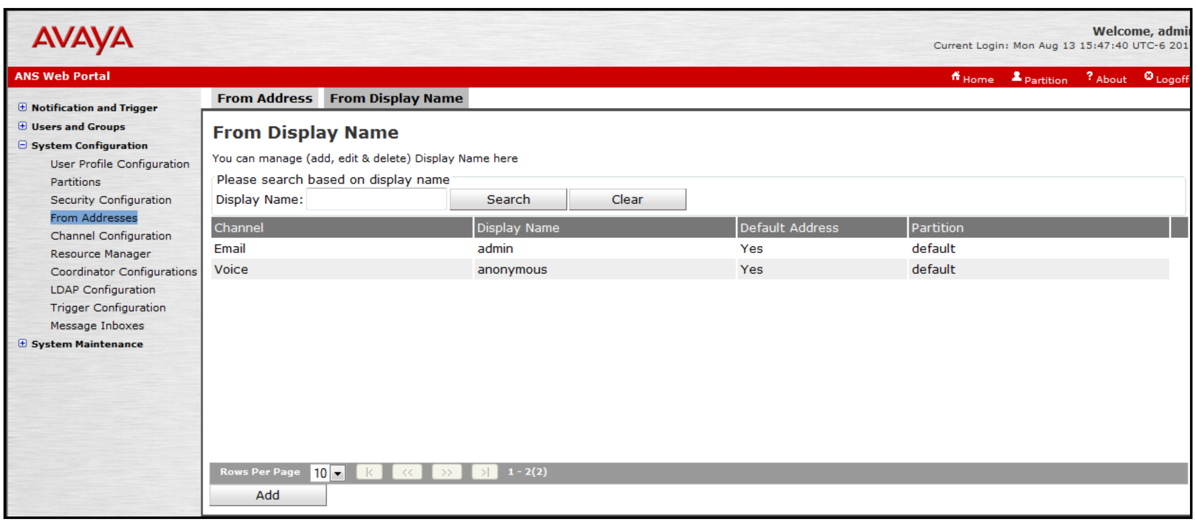
From Address - From Address

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

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**.



From Address– From Display Name

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

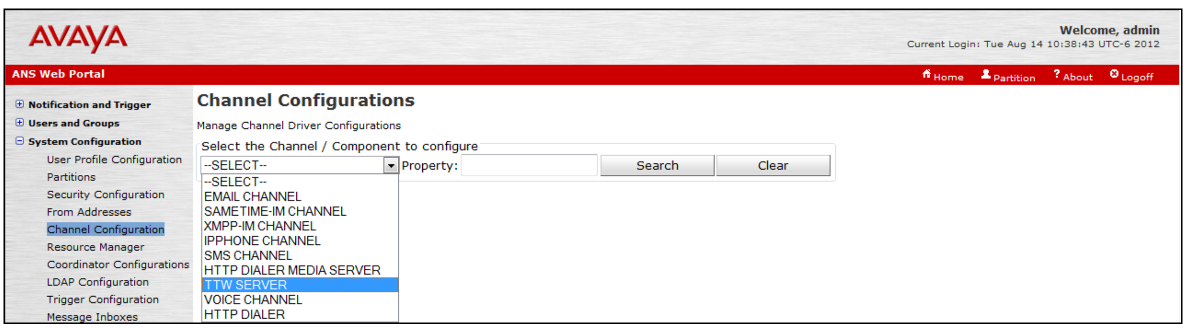
- **Channel** – Select **Voice** from drop-down list
- **Display Name** – Enter any informative string
- **Default Address** – Check this box for at least one address
- Click **Save**
- Repeat this step to configure additional display names

From Address– Add Display Name

5.7. TTW Server

This section describes steps to configure the TTW Server.

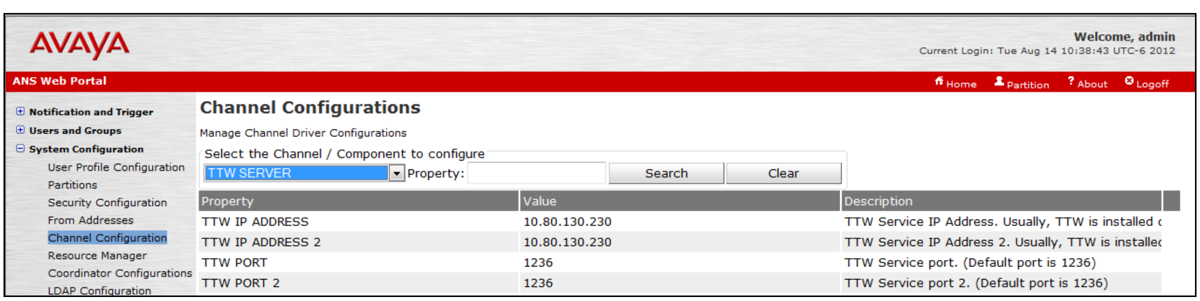
1. Navigate to **System Configuration→Channel Configuration** and select **TTW SERVER** from the drop-down list in the **Select the Channel/Component to configure** field.



Channel Configuration – Select Component

On the subsequent **Channel Configuration**s 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

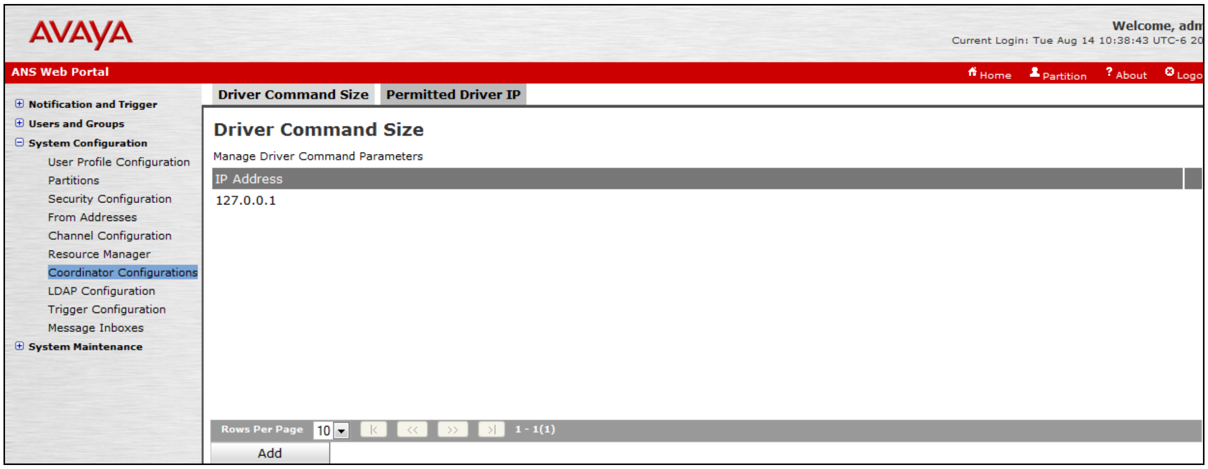


Channel Configuration – TTW Address

5.8. Permitted Driver IP Address

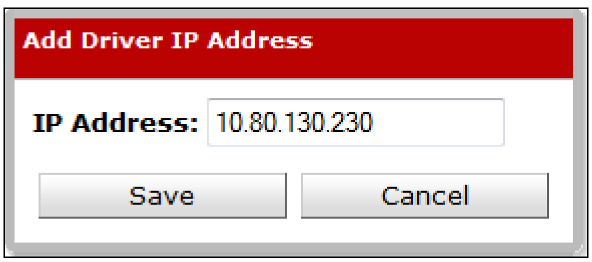
This section describes steps to configure the Permitted Driver IP Address.

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



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**.

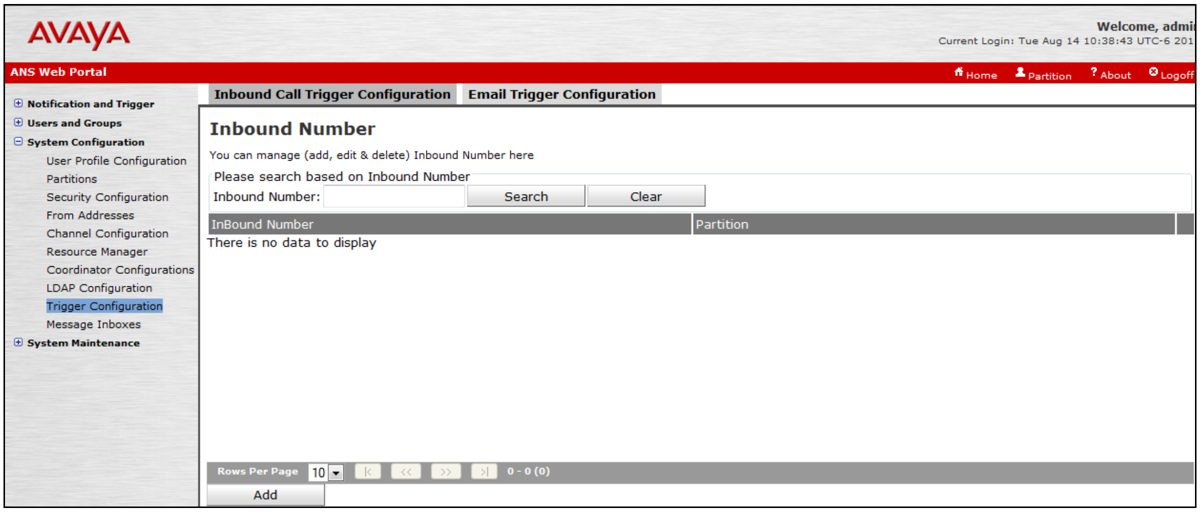


Coordinator Configuration – Add Driver IP Address

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.

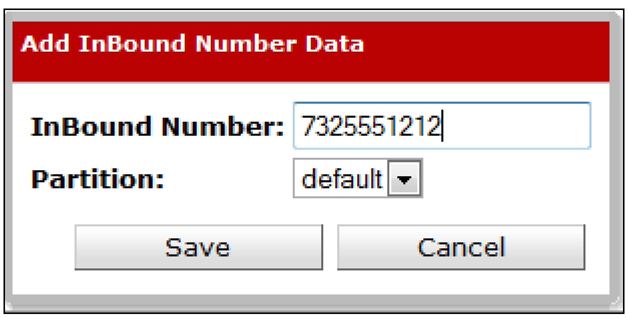
1. Navigate to **System Configuration→Trigger Configuration** and click **Add**.



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

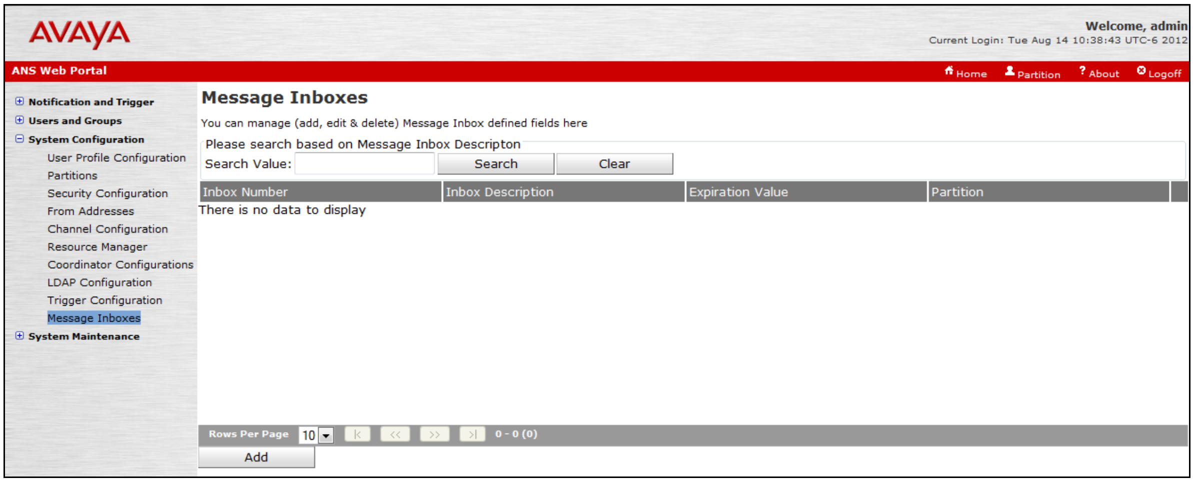


Trigger Configuration – Add Inbound Number Data

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.

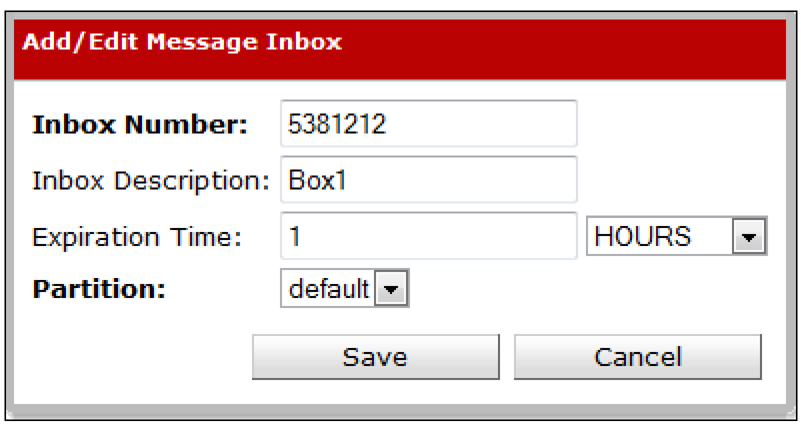
1. Navigate to **System Configuration→Message Inbox** and click **Add**.



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

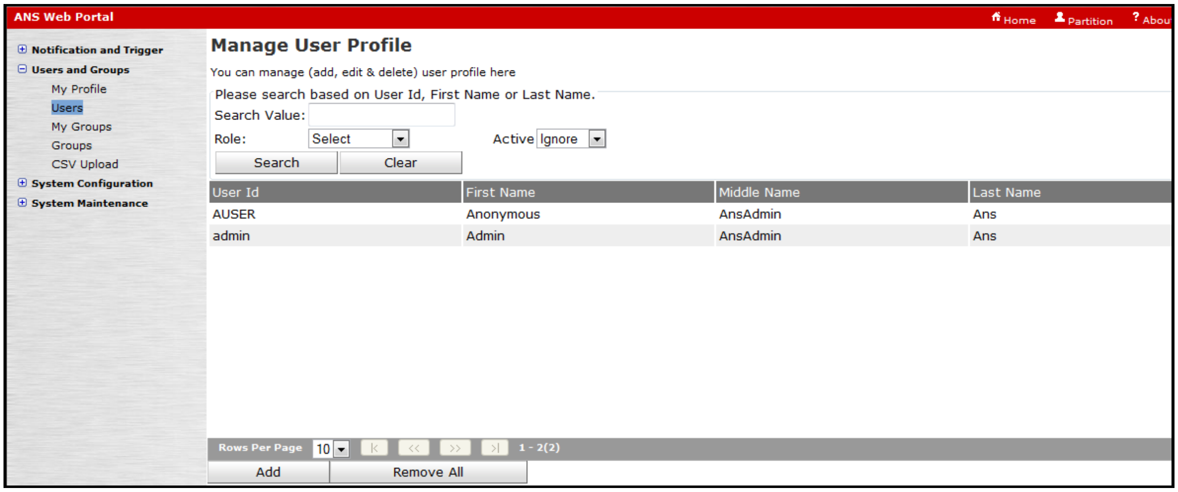


Trigger Configuration – Add Inbound Number Data

5.11. Manage User Profile

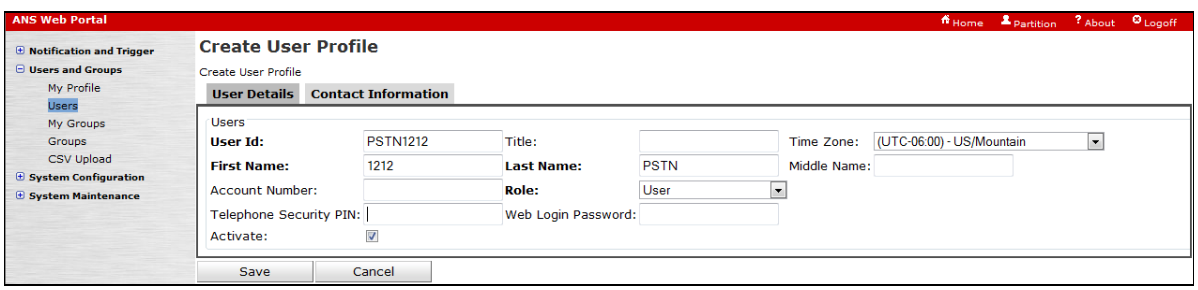
The following steps show the subscriber configuration on ANS.

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



Users – Manage User Profile

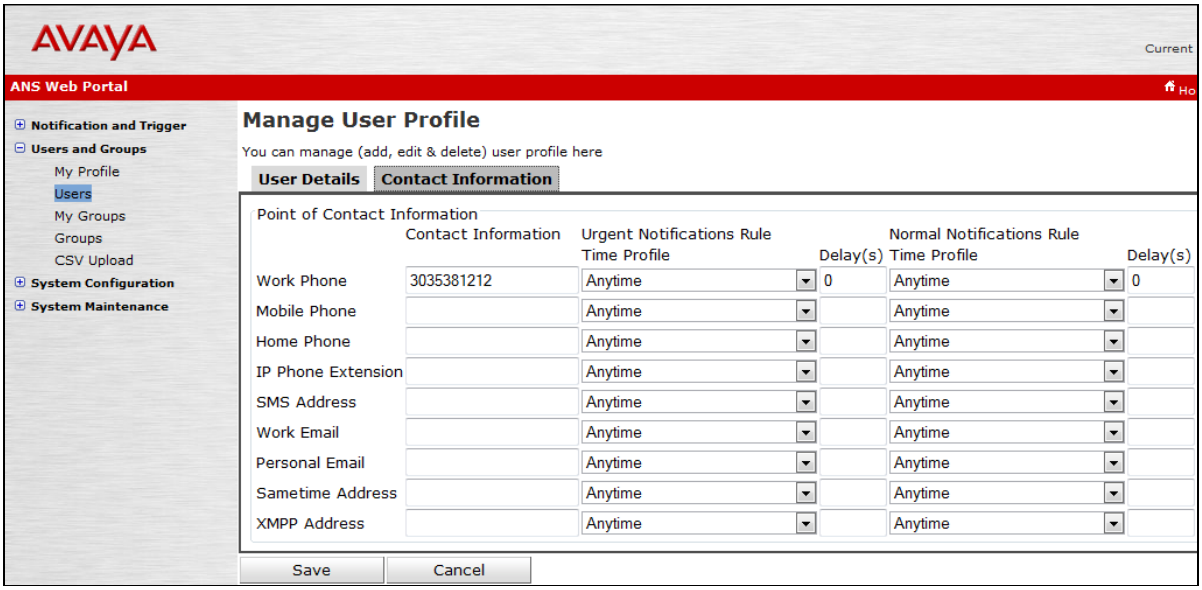
2. On the subsequent screen **Create User Profile** shown below, select the **User Details** tab and configure as follows:
   - **User Id** – Enter any valid id
   - **Time Zone** – Enter a valid timezone
   - **First Name** – Enter any informative string
   - **Last Name** – Enter any informative string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



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.



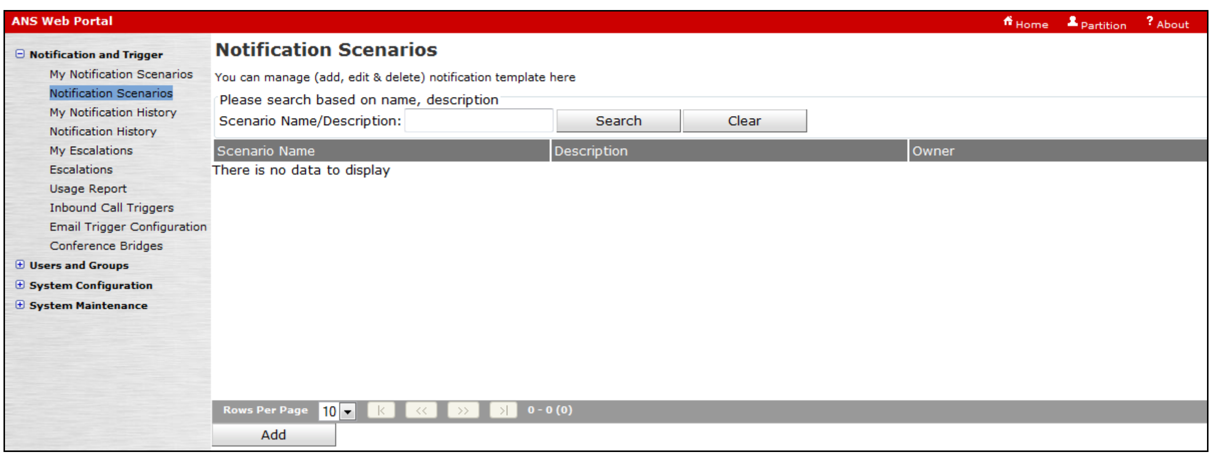
Users – Contact Information

4. Repeat above step to configure additional subscribers.

5.12. Add Outbound Notifications

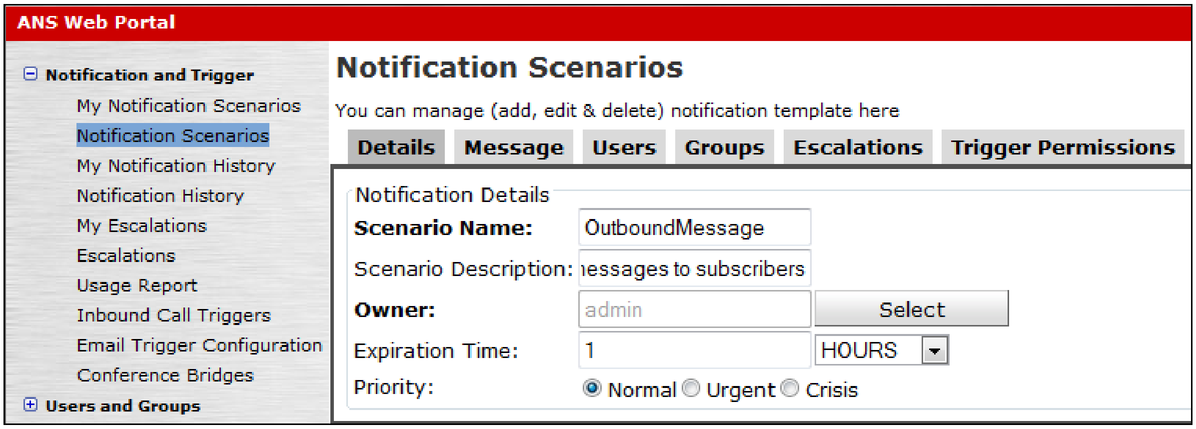
The following steps show the configuration of ANS for sending outbound notifications to its subscribers.

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



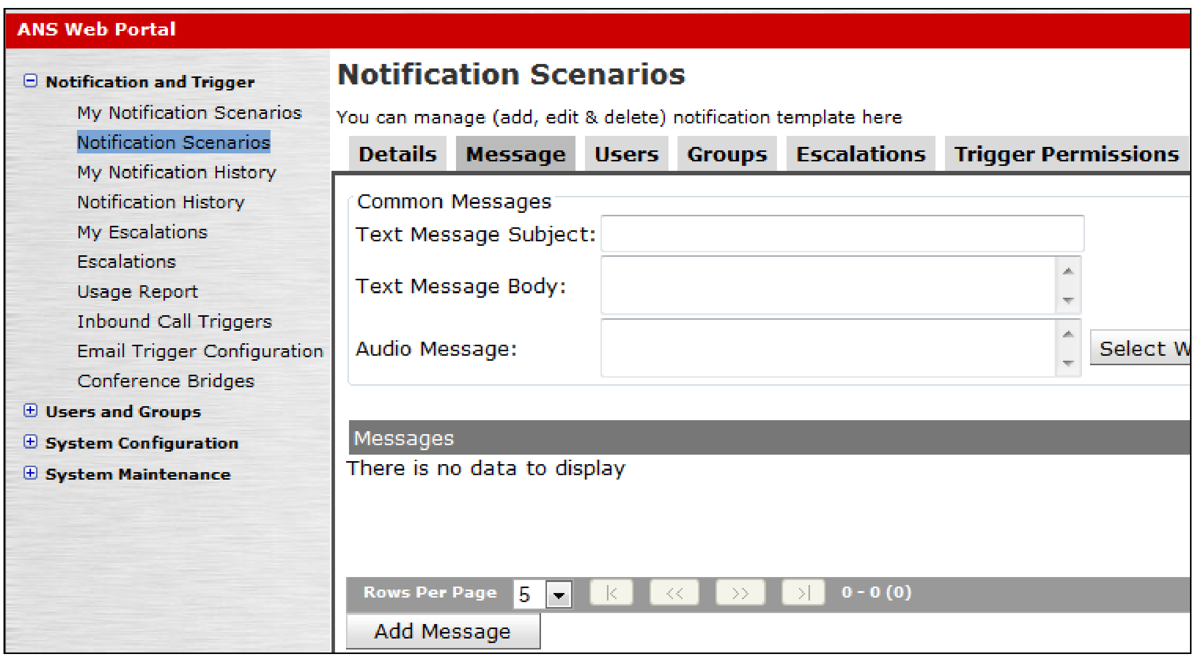
Notification Scenarios

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



Notification Scenarios – Details

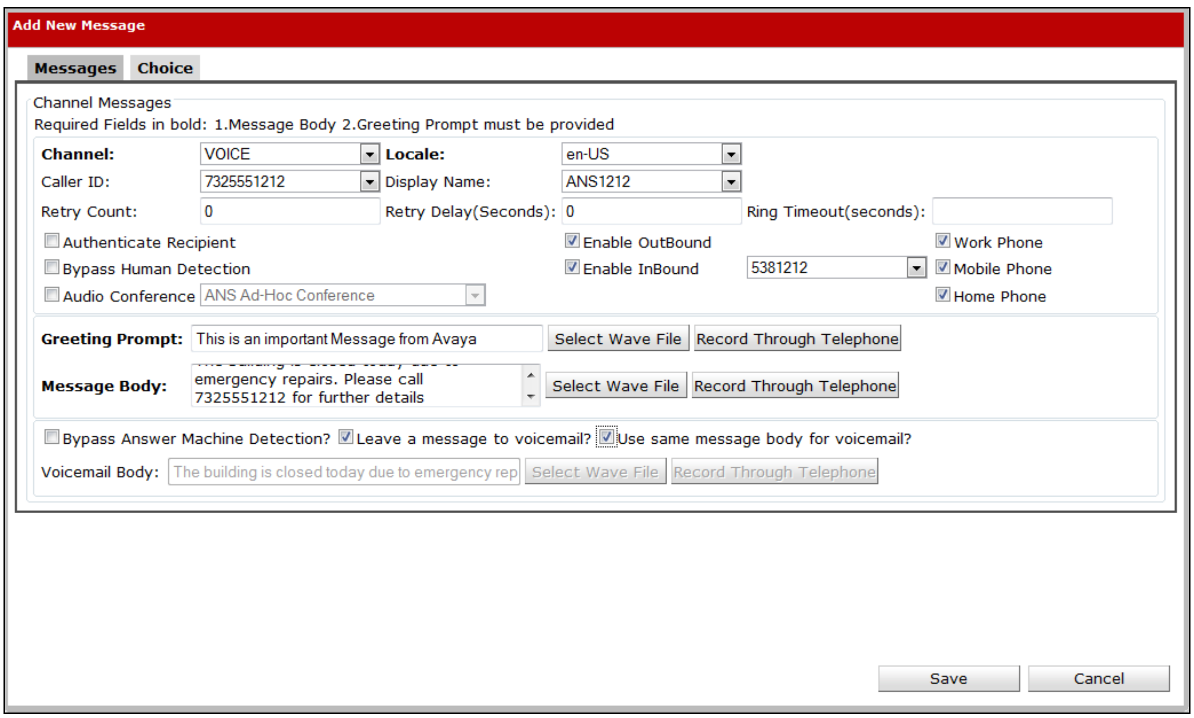
3. Select **Message** tab and click **Add Message**.



Notification Scenarios – Message

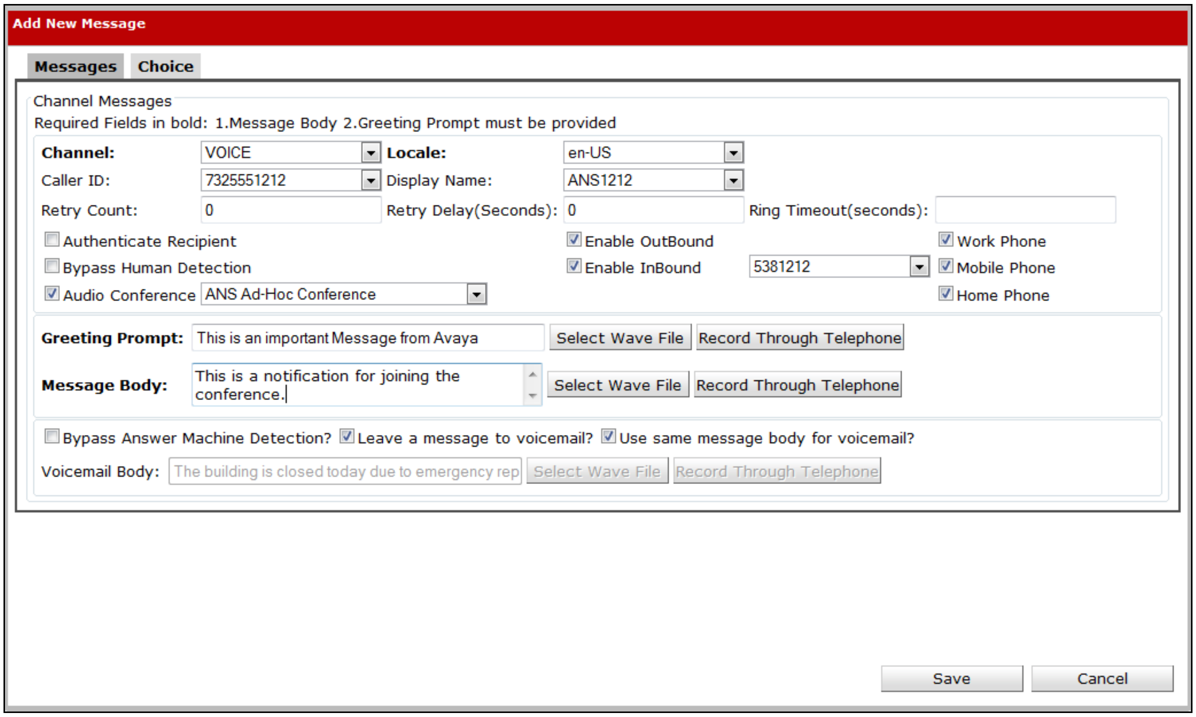
On the pop-up screen **Add New Message** shown below, configure as follows:

- **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**



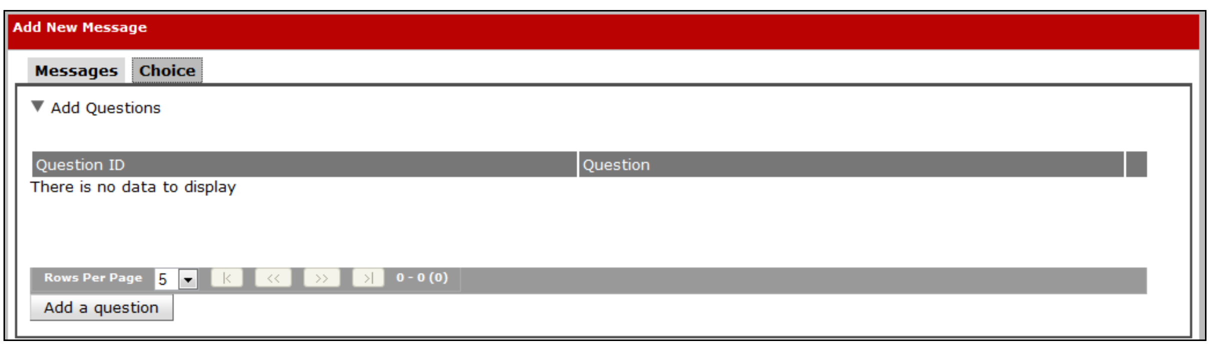
Notification Scenarios – Add New Message

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**

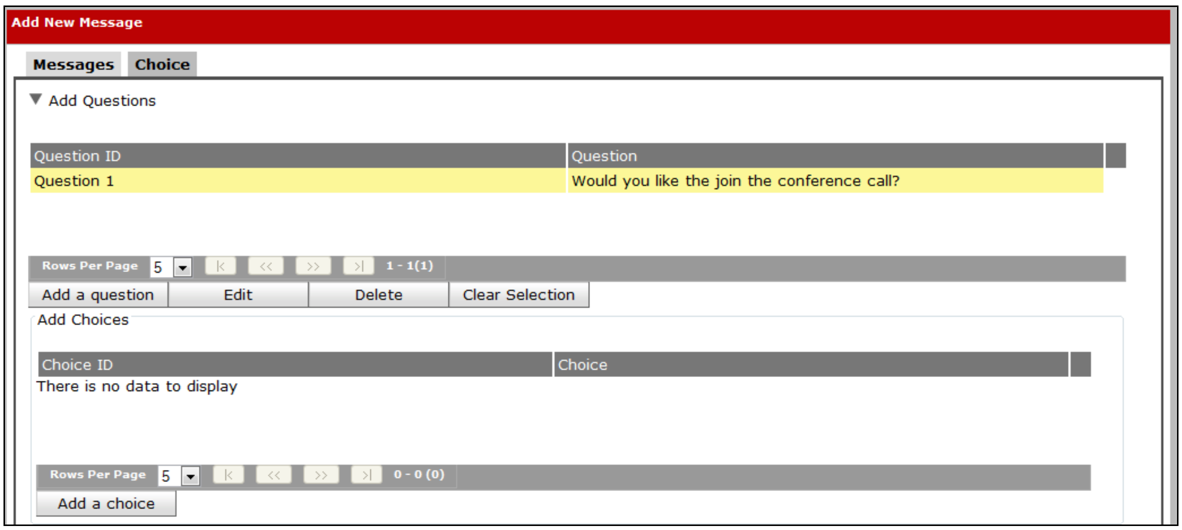


Notification Scenarios – Choice (Audio Conference)

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

Notification Scenarios – Add Question (Audio Conference)

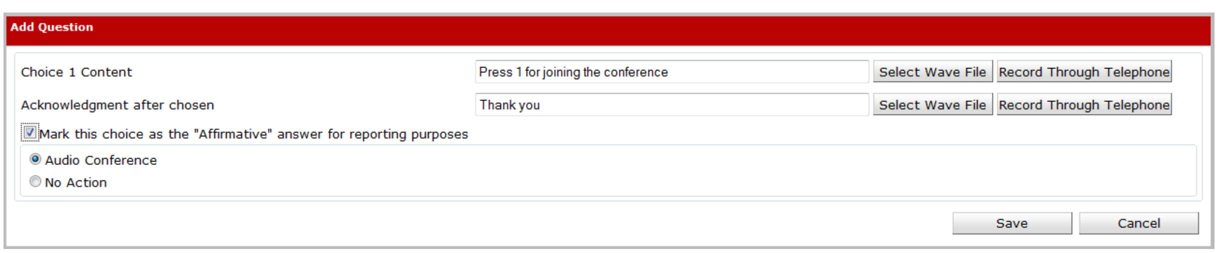
- On the next screen where the question is added, highlight the question and following screen appears. Click **Add a choice**



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**

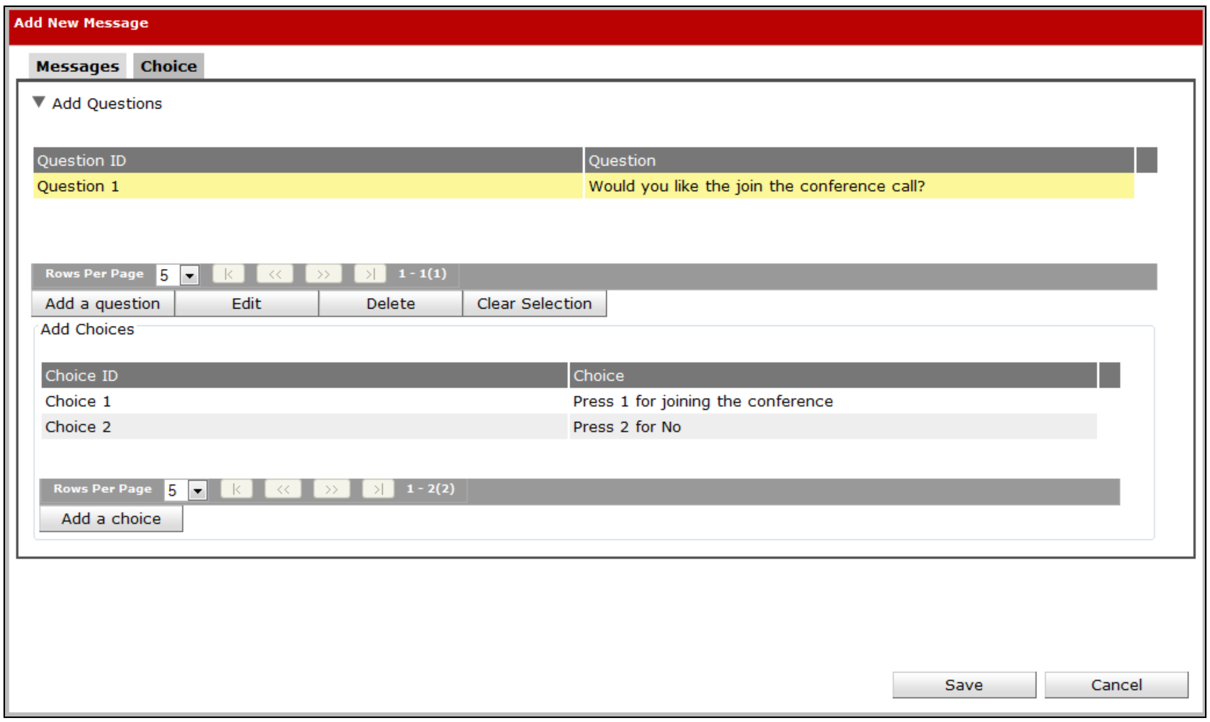


Notification Scenarios – Choice to join the conference (Audio Conference)

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

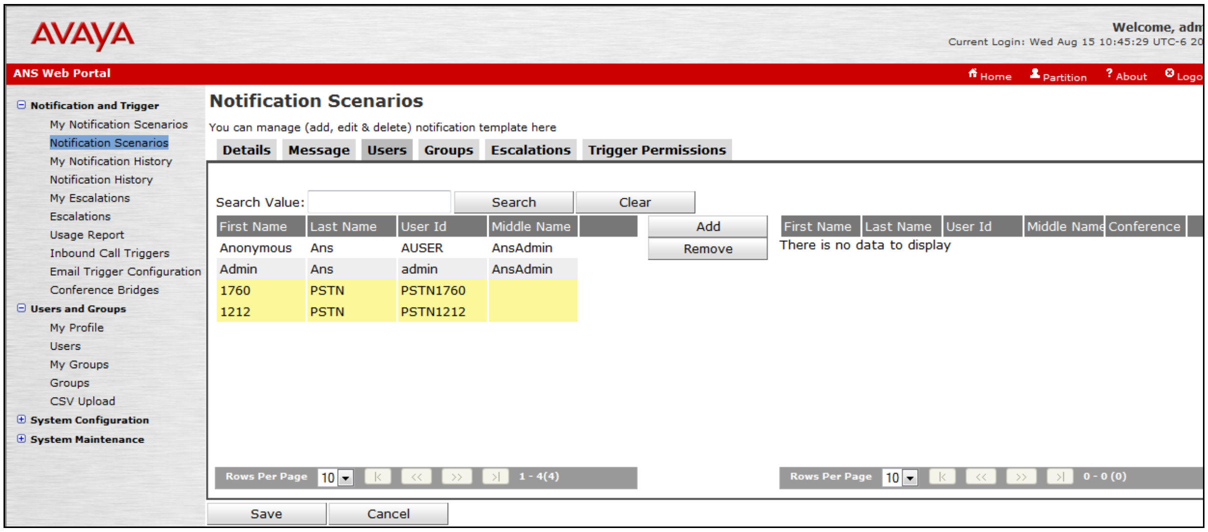
Notification Scenarios – Refusal to join the conference (Audio Conference)

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



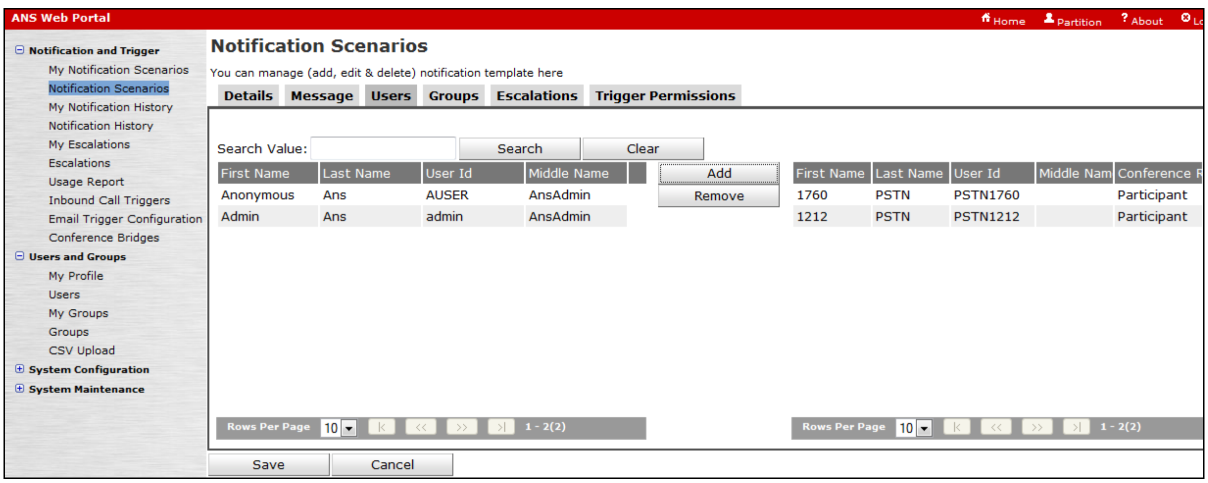
Notification Scenarios – Choices (Final)

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



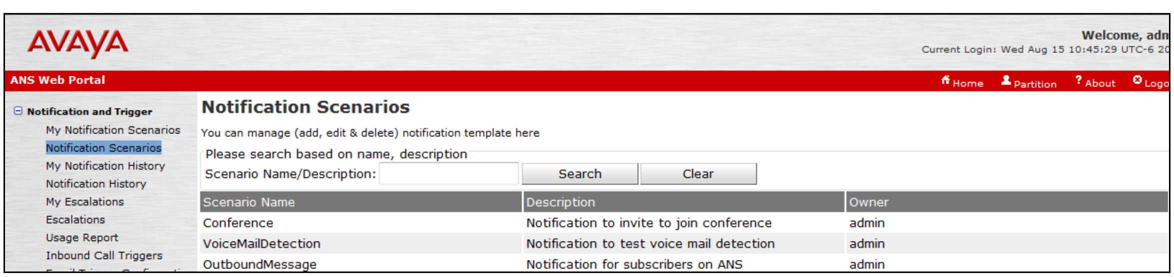
Notification Scenarios – Add Users

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



Notification Scenarios – Users Added

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

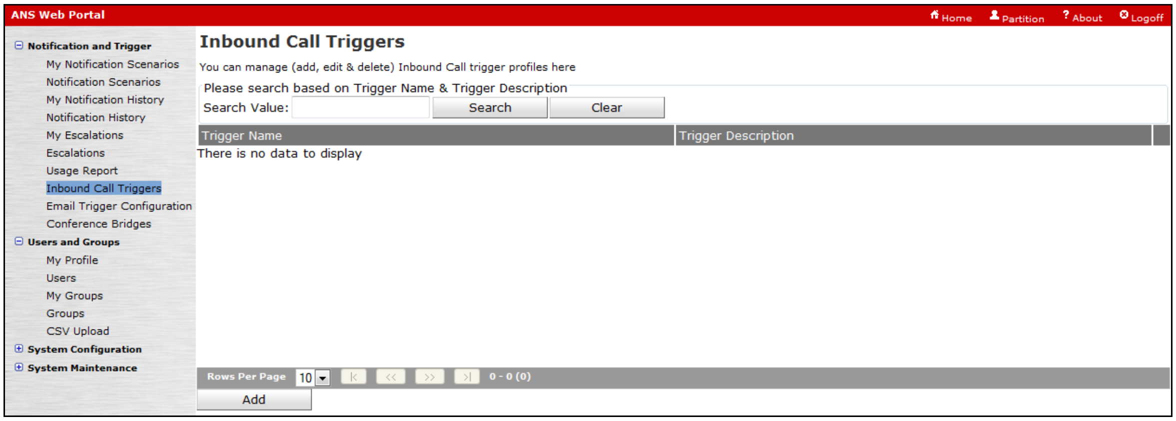


Notification Scenarios configured

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.

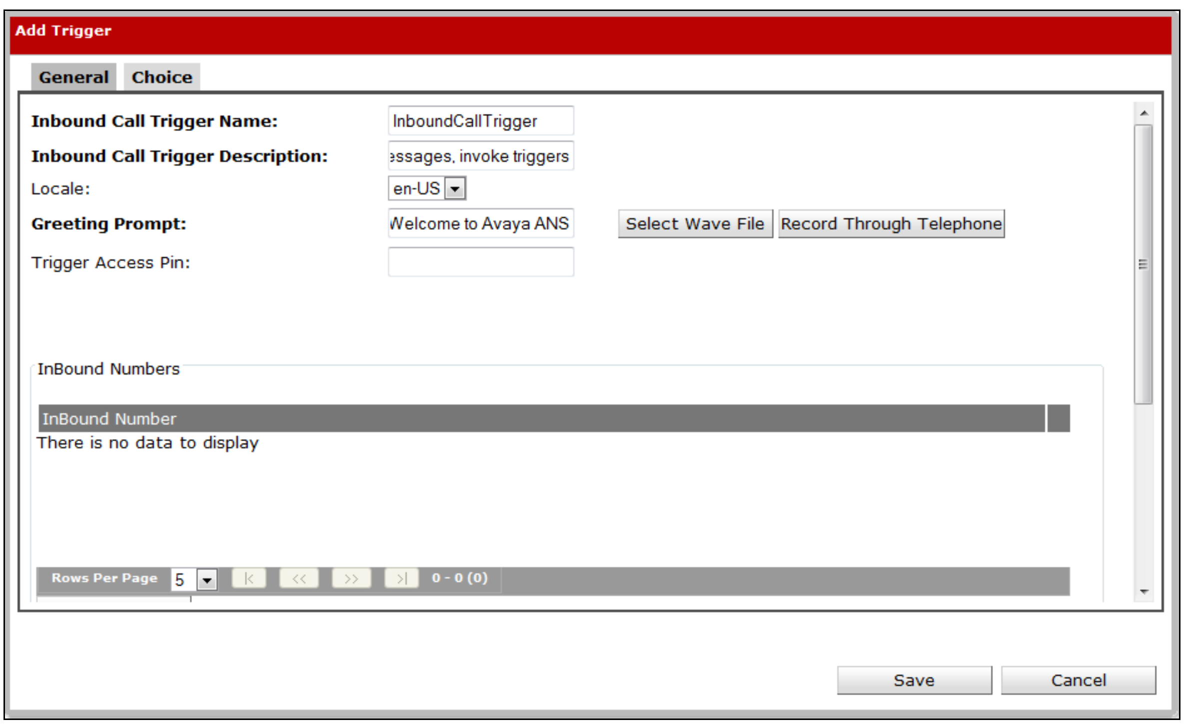
1. In the left pane, navigate to **Notification and Trigger→Inbound Call Trigger** and click **Add**.



Inbound Call Triggers

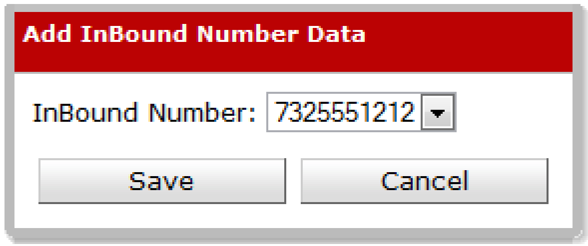
On the pop-up screen **Add Trigger** shown below, configure as follows:

- **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



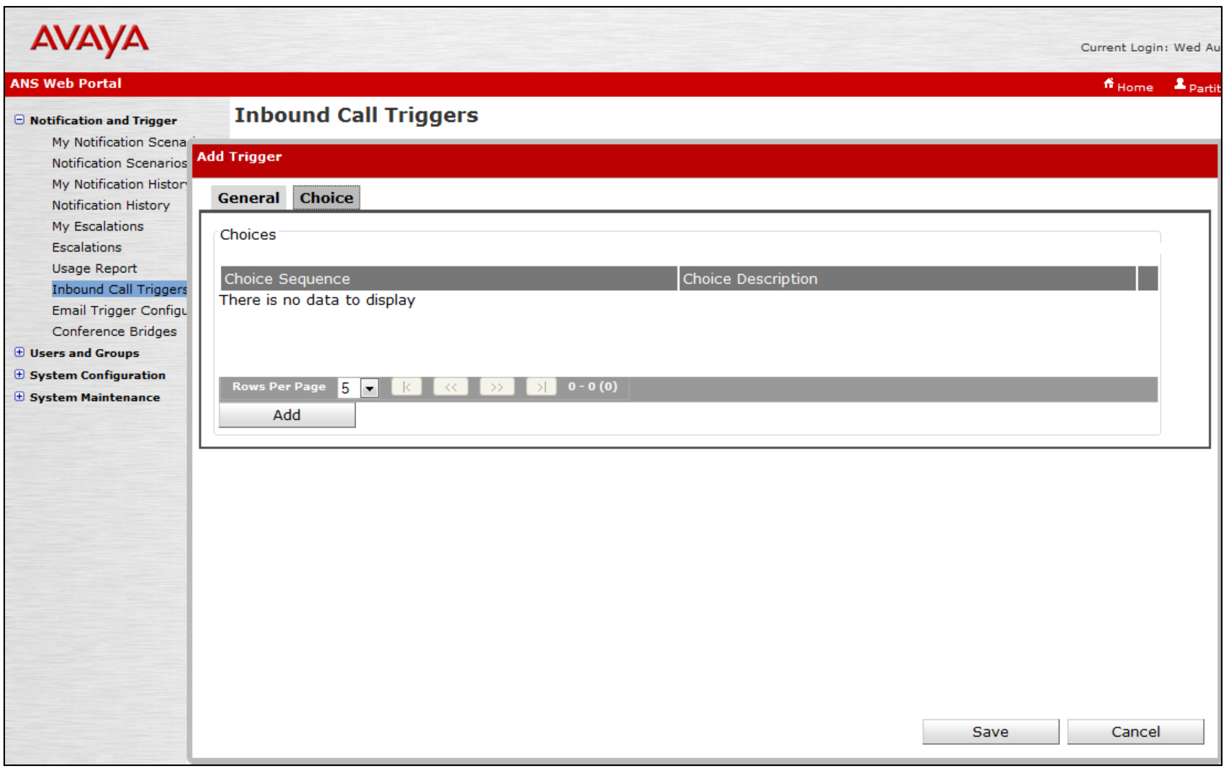
Inbound Call Triggers – Add Trigger

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



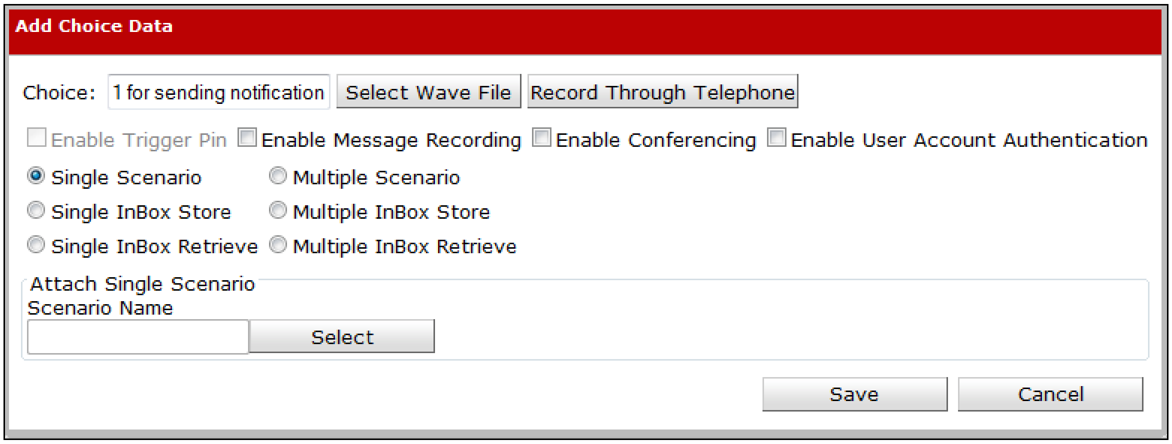
Inbound Call Triggers – Add Inbound Number Data

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



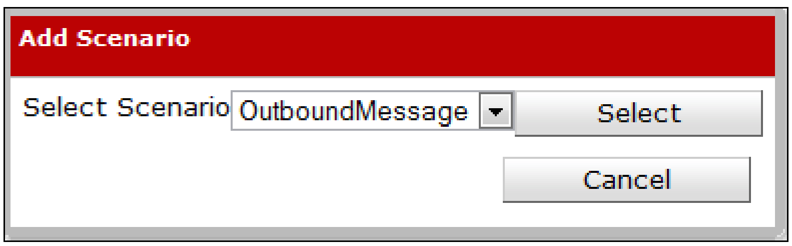
Inbound Call Triggers – Choice

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



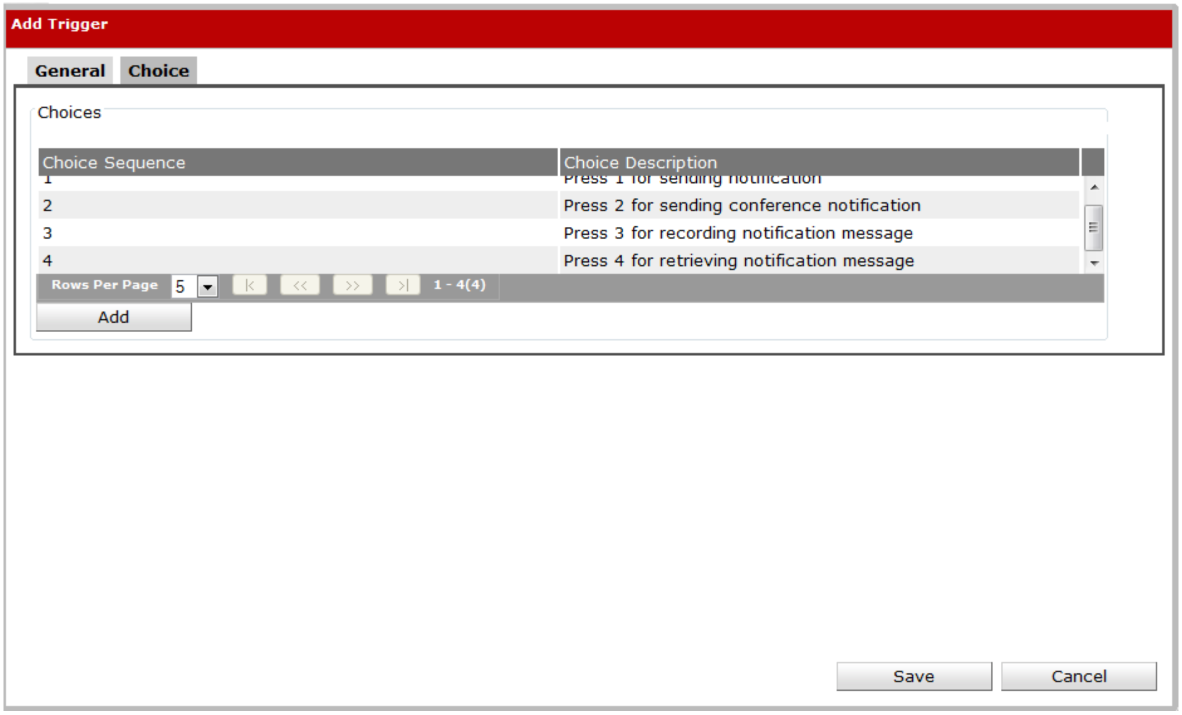
Inbound Call Triggers – Add Choice Data

- 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**

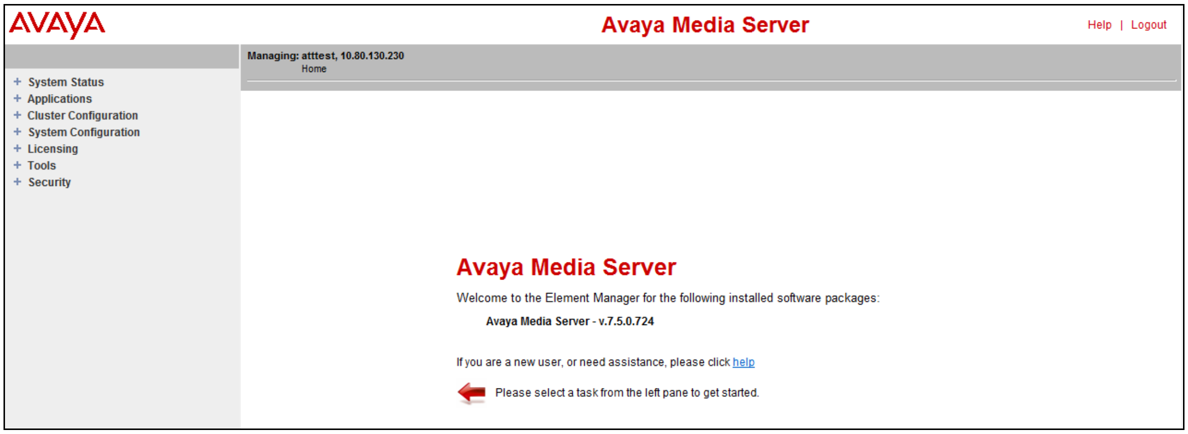


Inbound Call Triggers – Add Trigger (Final)

5.14. Avaya Media Server

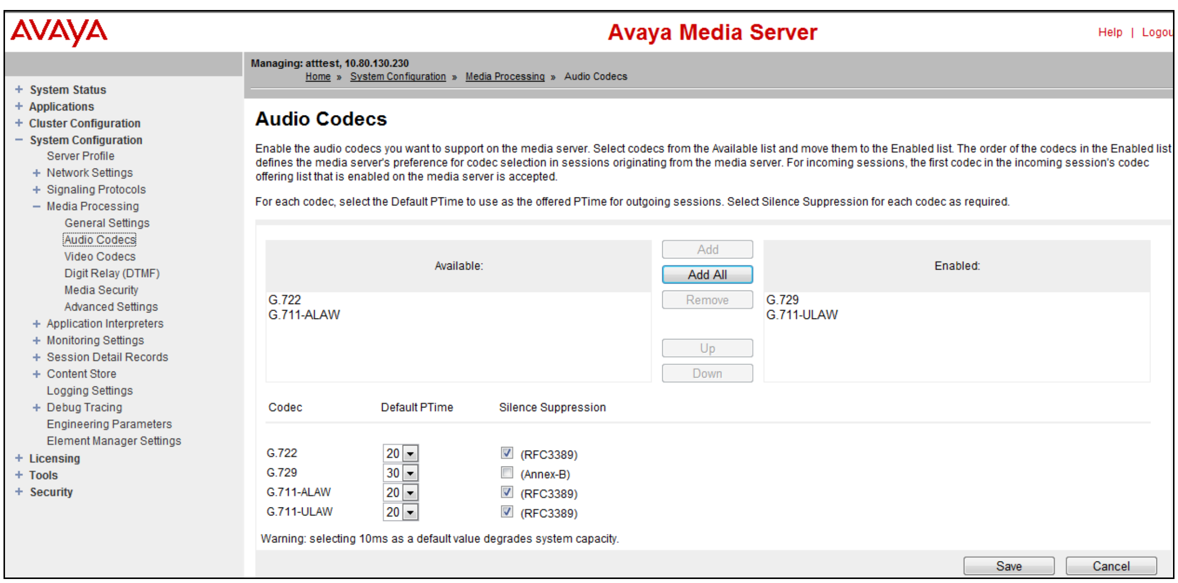
The installation of Avaya Media Server is beyond the scope of this document. For installation and basic system administration steps refer to [**1**, **2**]. This section covers configuration related to IP Codec Set, Ptime, Dynamic Payload and RTP Port Range.

- 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.



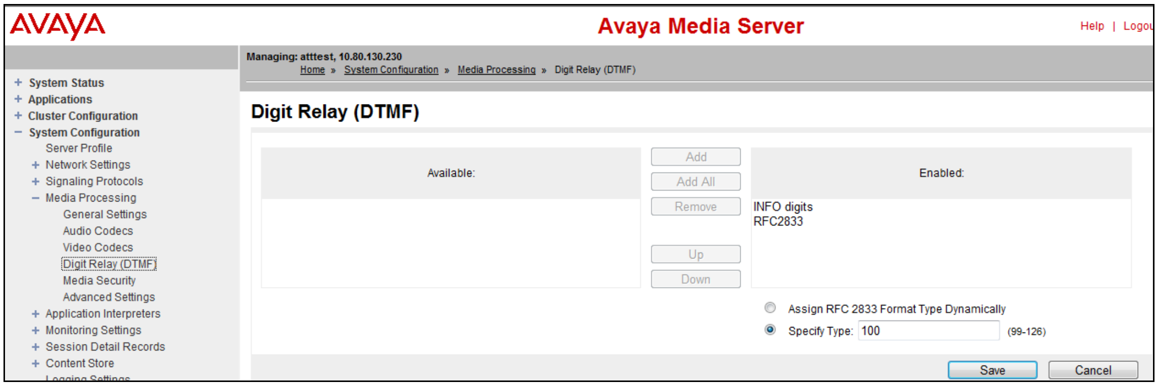
AMS Home Page

- 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 in unchecked for G729. See **Section 2.2**, **item 2** for further details.



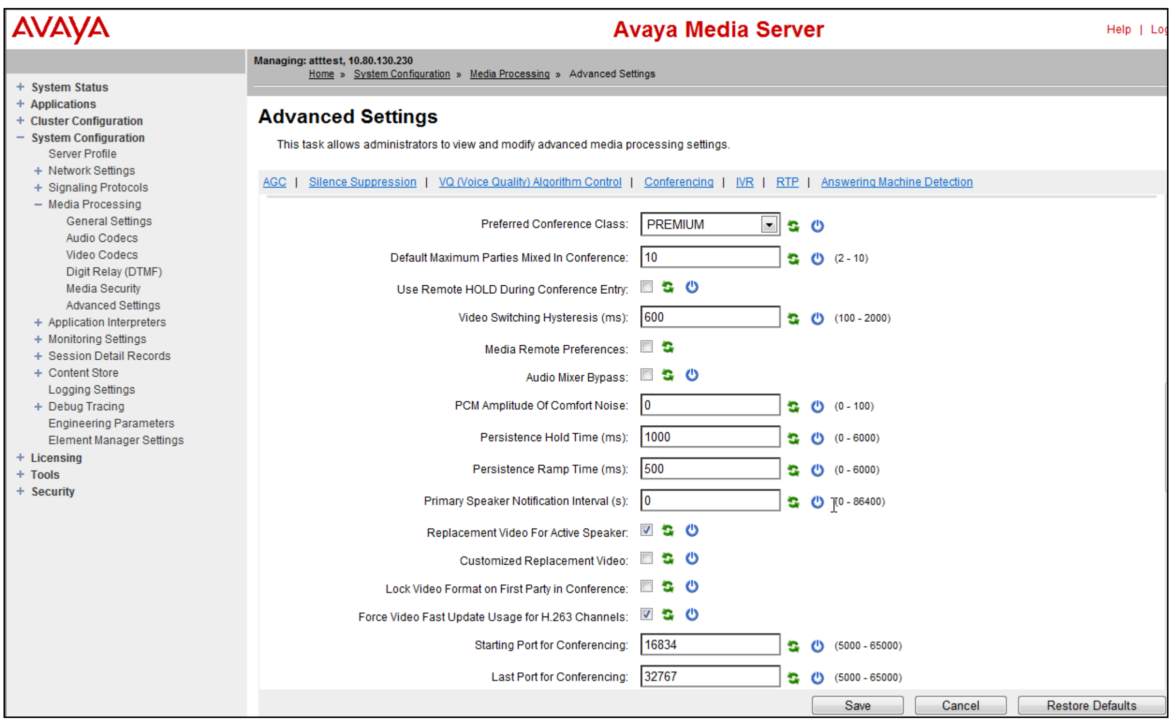
AMS Audio Codecs

- 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
      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                        s1p0
    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

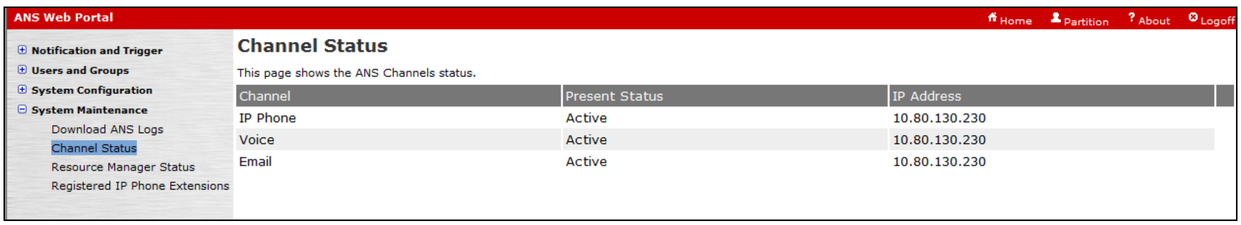
The following steps may be used to verify the configuration:

- 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.

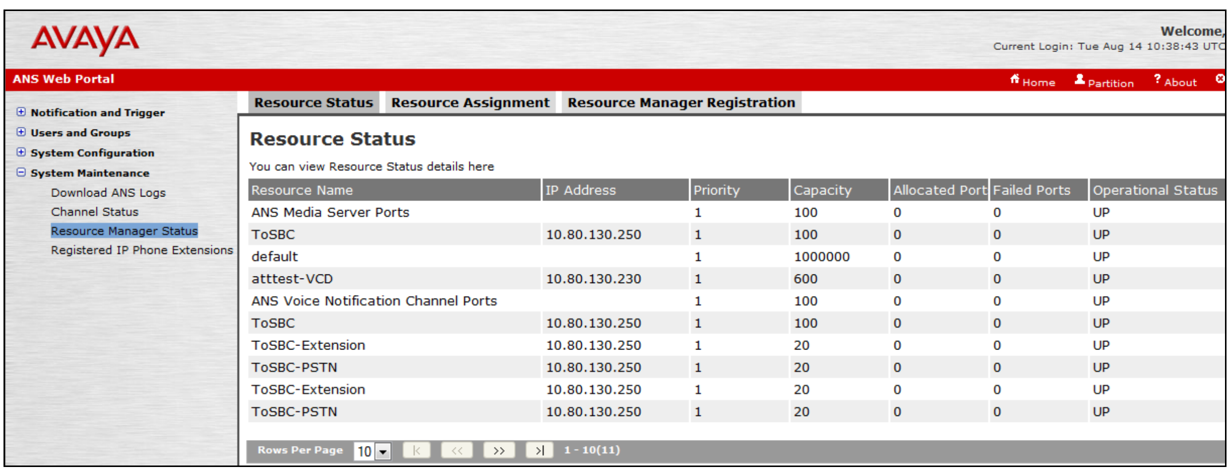
1. Navigate to **System Maintenance→Channel Status** to verify the SIP trunk is properly configured. If **VOICE** shows **ACTIVE** status, then the configuration is good.



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

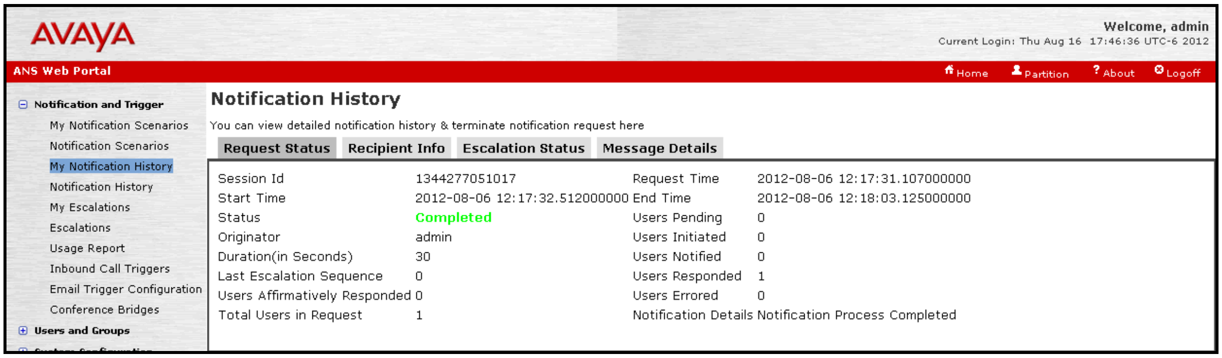
2. Navigate to **System Maintenance→Resource Manger Status** and make sure **Operational Status** of all the resource is **UP**.



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

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

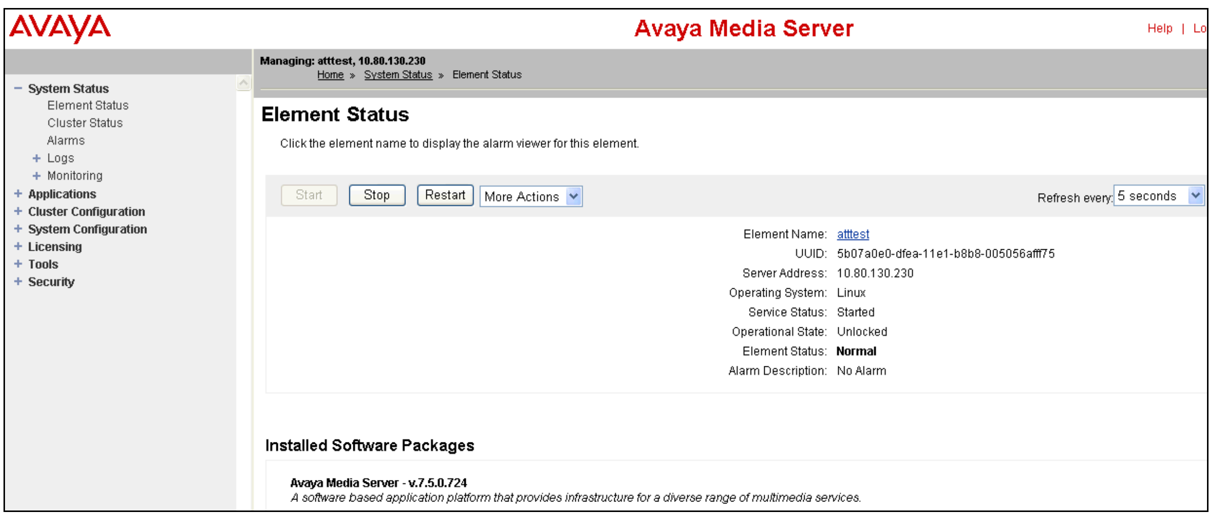


Notification History

7.3. Avaya Media Server

The following commands are issued from the ANS Web Portal.

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



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 /

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.