Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya Aura® Communication Manager 6.0 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager and the Acme Packet Net-Net (models 3800, 4250, and 4500) with the AT&T IP Toll Free service using **MIS/PNT** transport service connections.

Avaya Aura® Communication Manager 6.0 is a telephony application server and is the point of connection between the enterprise endpoints and an Acme Packet Net-Net 3800. The Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Communication Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. Avaya Aura® Communication Manager interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

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.

TABLE OF CONTENTS

1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 6.0 and the Acme Packet Net-Net 3800 with the AT&T IP Toll Free service using **MIS-PNT** transport service connections.

Avaya Aura® Communication Manager 6.0 is a telephony application server and is the point of connection between the enterprise endpoints and the Acme Packet Net-Net 3800. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Communication Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing **MIS/PNT** transport. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. Avaya Aura® Communication Manager interaction with the AT&T IP Transfer Connect service option will be addressed in separate Application Notes.

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura® Communication Manager 6.0, Avaya IP and analog stations, fax machines (Ventafax application), Acme Packet Net-Net 3800, and Avaya Modular Messaging.
- A laboratory version of the AT&T IP Toll Free service network, to which the simulated enterprise was connected via **MIS/PNT** transport.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for examples) between Avaya Aura® Communication Manager, Acme Packet Net-Net 3800, and the AT&T IP Toll Free service using **MIS/PNT**[1] transport.

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 from the PSTN across the AT&T network. The following features were tested as part of this effort:

- SIP trunking.
- T.38 Fax.
- AT&T IP Toll Free calls to Avaya Aura® Communication Manager stations, Vector Directory Numbers (VDNs), and vectors.
- Navigating automated IP Toll Free features by passing DTMF signaling to activate IP Toll Free features such as hold, resume, conference and transfer.

2.2. Test Results

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

[1] MIS/PNT does not support cRTP.

- Inbound AT&T IP Toll Free service calls between Avaya Aura® Communication Manager VDNs/vectors and stations.
- Two-way talk path establishment between PSTN and Avaya Aura® Communication Manager VDNs/vectors and stations, via the AT&T Toll Free service.
- Navigating automated AT&T IP Toll Free menus by passing DTMF tone transmission using RFC 2833 to activate features such as hold, resume, conference and transfer between Avaya Aura® Communication Manager stations and the AT&T IP Toll Free service.
- G.729 and G.711 codecs.
- T.38 fax calls between Avaya Aura® Communication Manager and the AT&T IP Toll Free service/PSTN G3 and SG3 fax endpoints.
- Inbound AT&T IP Toll Free service calls to Avaya Aura® Communication Manager that are directly routed to stations, and alternatively can be covered to Avaya Modular Messaging.
- Long duration calls.

The test objectives stated in **Section 2.1**, with limitations as noted in **Section 2.2.1** below, were verified.

2.2.1. Known Limitations

1. SIP stations are not supported by Avaya Aura® Communication Manager 6.0 in the reference configuration.
2. G.726 codec is not supported between Avaya Aura® Communication Manager and the AT&T IP Toll Free service.
3. G.711 faxing is not supported between Avaya Aura® Communication Manager and the AT&T IP Toll Free service. Avaya Aura® Communication Manager does not support the protocol negotiation that AT&T requires to have G.711 fax calls work. T.38 faxing is supported, as is Group 3 and Super Group 3 fax. Fax speeds are limited to 9600 bps in the configuration tested. In addition, Fax Error Correction Mode (ECM) is not supported by Avaya Aura® Communication Manager.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

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

- Communication Manager provides the voice communications services for a particular enterprise site, including H.323 and analog stations (SIP stations are not supported in the reference configuration). In this reference configuration, Communication Manager runs on an Avaya S8800 Server. This solution is extensible to other Avaya S8xxx Servers.

  The Avaya Media Gateway provides the resources for Communication Manager. In this reference configuration, an Avaya G450 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.

- Avaya "desk" stations are represented in the reference configuration by Avaya 4610 and 9630 Series IP Telephones running H.323 software, as well as an Avaya 6211 Series Analog Telephone. An Avaya One-X® Agent, a PC based H323 softphone, was also used in the reference configuration.

- The Acme Packet Net-Net 3800[2] provides SIP Session Border Controller (SBC) functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network.

- An existing Avaya Modular Messaging system (in Multi-Site mode in the reference configuration) provides the corporate voice messaging capabilities in the reference configuration. However the provisioning of Modular Messaging is beyond the scope of this document.

- Inbound calls were sent from the AT&T IP Toll Free service, through the Acme Packet SBC, to Communication Manager. Communication Manager connects the call to the appropriate phone or fax extension. The H.323 stations on the enterprise side registered directly to Communication Manager

- A PC (via analog modem) running the Ventafax application, was used to test T.38 fax.

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

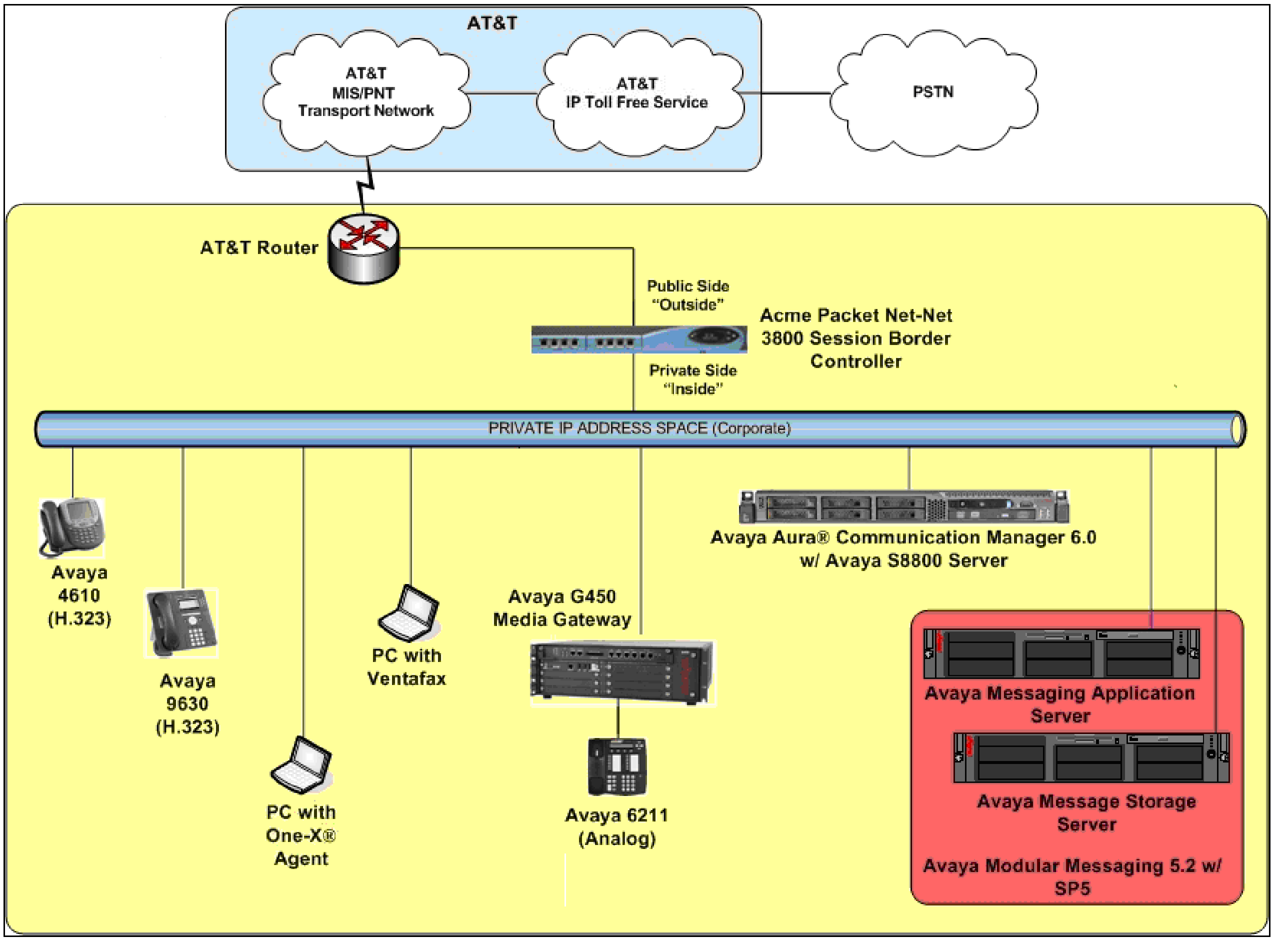


Figure 1: Reference configuration

3.1. Illustrative Configuration Information

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

Note - The AT&T IP Toll Free service border element IP address shown in this document is an example. AT&T Customer Care will provide the actual IP address as part of the AT&T IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® Communication Manager	
Processor Ethernet IP Address	192.168.67.202
Avaya Aura® Communication Manager extensions	40xxx = Stations 44xxx = VDNs 47xxx = Agents
Avaya CPE local dial plan	4xxxx
Voice Messaging Pilot Extension	46000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Messaging Server (MSS) IP Address	192.168.67.140
Modular Messaging mailbox dial plan	4xxxx
Acme Packet Net-Net SBC	
IP Address of "Outside" (Public) Interface (connected to AT&T Access Router/IP Toll Free Service)	192.168.64.130
IP Address of "Inside" (Private) Interface (connected to Avaya Aura® Communication Manager)	192.168.67.130
AT&T IP Toll Free Service	
Border Element IP Address	135.25.29.74
AT&T Access router interface (to Acme outside)	192.168.64.254
AT&T Access Router NAT address (Acme outside address)	135.16.170.55

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are handled by Communication Manager, two general call flows are described in this section. The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Toll Free service call to Communication Manager.

1. A PSTN phone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to the Acme Packet SBC.
4. The Acme Packet SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Communication Manager.
5. Depending on the called number, Communication Manager routes the call to a) a vector, which in turn, routes the call to an agent, or b) directly to an agent or phone.

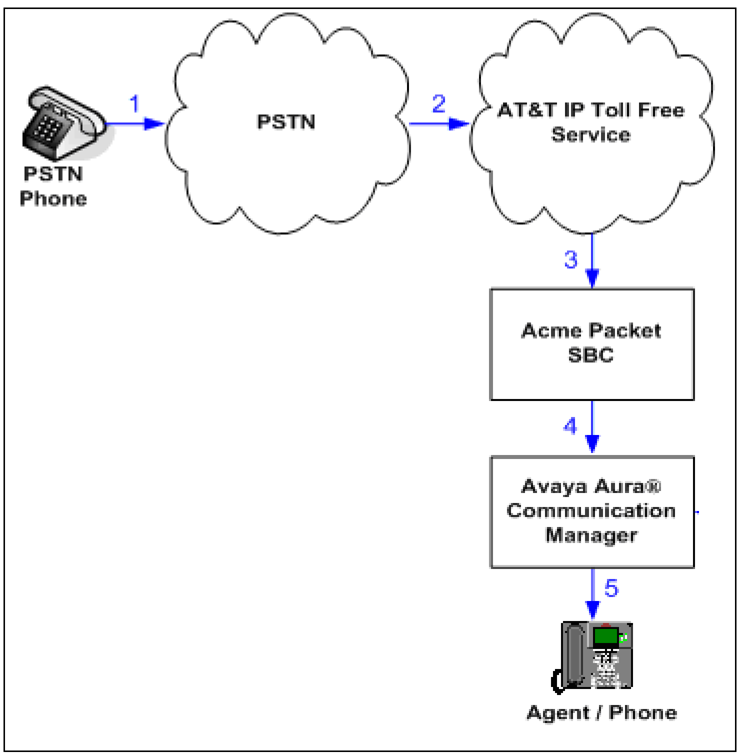


Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Phone

The second call scenario illustrated in **Figure 3** is an inbound call to Communication Manager that is covered to voicemail via an outbound call from Communication Manager. In this scenario, the voicemail system is a Modular Messaging system.

1. Same as the **Steps 1-5** from the first call scenario.
2. The called Communication Manager agent or phone does not answer the call, and the call covers to the agent's or phone's voicemail. Communication Manager forwards the call to Modular Messaging.
3. Modular Messaging answers the call and connects the caller to the called agent's or phone's voice mailbox. Note that the call continues to go through Communication Manager.

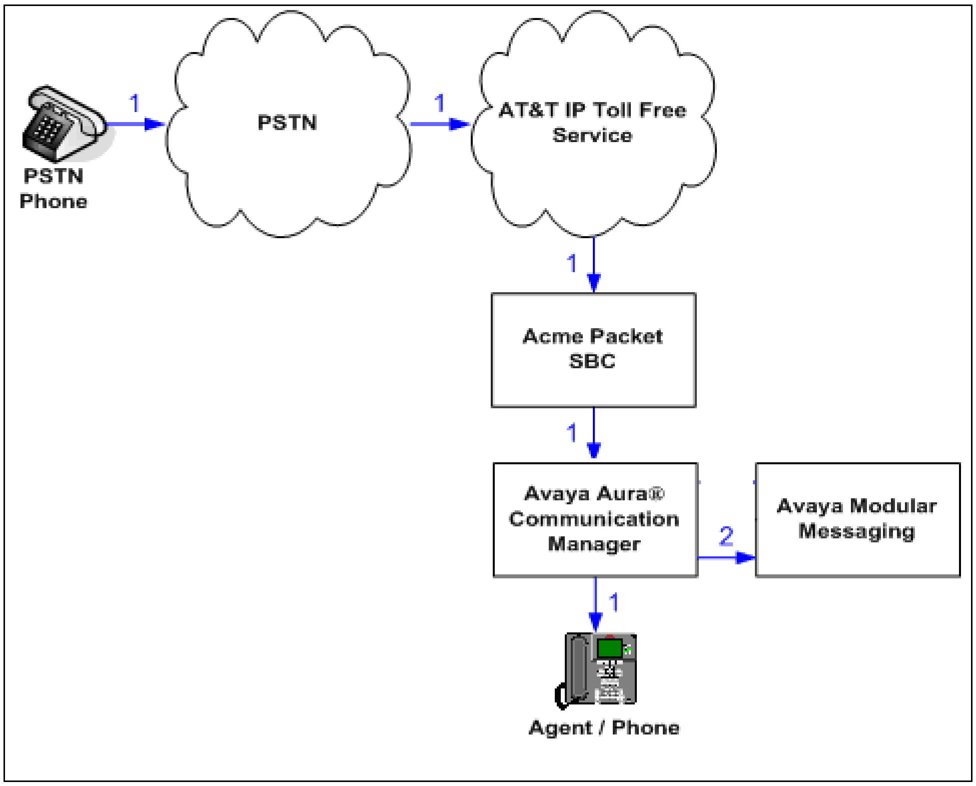


Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Phone Covered to Avaya Modular Messaging

4. Equipment and Software Validated

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

Component	Version
Avaya S8800 Server	Avaya Aura® Communication Manager 6.0 (R016x.00.0.345.0) with SP2 (18567)
Avaya G450 Media Gateway	31.18.1
MM711 Analog	HW31 FW95
Avaya 9630 IP Telephone – H323	Avaya one-X® Deskphone Edition H.323 Version S3.110b (ha96xxua3_11.bin)
Avaya one-X® Communicator	6.0.0.26
Avaya 4610SW IP Telephone – H323	a10d01b2_9_1.bin
Avaya one-X® Agent	2.0.09184.0
Avaya Modular Messaging (MAS and MSS) on Avaya S3500 Servers	Release 5.2 – SP5 with Patch 1 (9.0.350.5019)
Fax device	Ventafax Home Version 6.3.102.288
Acme Packet Net-Net 3800	SCX6.2.0 m5p1
AT&T IP Toll Free Service via AVPN or MIS/PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

Note - The solution integration validated in these Application Notes should be considered valid for deployment with Avaya Aura® Communication Manager release 6.0.1. Avaya agrees to provide service and support for the integration of Avaya Aura® Communication Manager release 6.0.1 with the AT&T IP Toll Free service offer, in compliance with existing support agreements for Avaya Aura® Communication Manager release 6.0, and in conformance with the integration guidelines as specified in the body of this document.

5. Avaya Aura® Communication Manager 6.0

In the reference configuration Communication Manager 6.0 is provisioned in an Evolution Server configuration, supporting H.323 and Analog endpoints (SIP endpoints are not supported in the reference configuration). This section describes the administration steps for Communication Manager in support of the reference configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration, including stations, Media Gateways, and announcement boards, etc., has already been performed. Consult **[1]** and **[2]** for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to these application notes. Other parameter values may or may not match based on local configurations and are shown for illustrative purposes.

5.1. System Parameters

This section reviews the Communication Manager licenses and features that are required for the reference configuration described in these Application Notes. For required licenses to access features that are not enabled in the steps that follow, contact an authorized Avaya account representative to obtain the licenses.

1. Enter the **display system-parameters customer-options** command. On Page 2 of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks (e.g. 24000).

```
change system-parameters customer-options                       Page   2 of  11
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 1
                       Maximum Administered SIP Trunks: 24000 24
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 4: System-Parameters Customer-Options Form – Page 2

2. On Page 4 of the **system-parameters customer-options** form:
    a. Verify that the **IP Trunks** field in the following screenshot is set to "**y**".

```
change system-parameters customer-options                       Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                                IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                          ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                            ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                   ISDN-PRI? y
              ESS Administration? y               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y                  Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                  Multifrequency Signaling? y
      Global Call Classification? y         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? y
                       IP Trunks? y
           IP Attendant Consoles? y
         (NOTE: You must logoff & login to effect the permission changes.)
```

Figure 5: System-Parameters Customer-Options Form – Page 4

5.2. Dial Plan

Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings administered in **Figure 6**:

- 3-digit dial access codes (indicated with a **Call Type** of "**dac**") beginning with the digit "**1**" – Trunk Access Codes (TACs) defined for trunk groups in this reference configuration conform to this format.
- 5-digit extensions with a **Call Type** of "**ext**" beginning with the digits "**4**" – local extensions for Communication Manager stations, agents, and Vector Directory Numbers (VDNs) in this reference configuration conform to this format (4xxxx).
- 1-digit feature access code (indicated with a **Call Type** of "**fac**") beginning with the digit "**8**" – access code for outbound AAR dialing
- 1-digit feature access code (indicated with a **Call Type** of "**fac**") beginning with the digit "**9**" – access code for outbound ARS dialing.
- 3 -digit feature access code (indicated with a **Call Type** of "**fac**") beginning with the character "*".

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   1           3   dac
   4           5   ext
   8           1   fac
   9           1   fac
   *           3   fac
```

Figure 6: Dialplan Analysis Form

5.3. IP Network Regions

Network Regions are used to manage various Communication Manager resources such as codecs, UDP port ranges, and inter-region communication. In the reference configuration, two network regions are used. One for Local/Modular Messaging calls, and one for AT&T IP Toll Free calls.

The "Local" region (region 1) is configured to use G.711 as the primary codec for optimal quality, but with G.729B and G.729Aas alternate codecs (codec set 1).

The "AT&T" region (region 2) is set to use G.729B and G.729A as the primary codecs to best utilize bandwidth, but G.711 is also specified so any G.711 calls originated from the network will be accepted (codec set 2). Note – See **Section 2.2.1** regarding support of G.726 codec.

Inter-region communication between Local and AT&T regions 1 and 2 is set to use codec set 2 as well.

Codec Set List	Region/Codec set	Inter-region Codec Set
Codec Set 1 – G.711Mu, G.729A, G.729B	1/1	Region 1 to 2 = Codec 2
Codec Set 2 – G.729B, G.729A, G.711Mu	2/2	Region 2 to 1 = Codec 2

Table 3: Network Regions and their related codecs

5.3.1. IP Network Region 1 – Local Region

In the reference configuration local Communication Manager IP elements (e.g. Processor Ethernet and Media Gateways) as well as other local Avaya devices (e.g. Modular Messaging) are assigned to ip-network-region 1. In the reference configuration H323 stations are assigned to region 1 as well.

1. Enter the **change ip-network-region x**, where **x** is the number of an unused IP network region (e.g. **region 1**). This IP network region will be used to represent the local CPE equipment. On page 1 of the form enter:
   - Enter **customerb.com** in the **Authoritative Domain** field.
   - Enter a descriptive name (e.g. **Local**).
   - Enter **1** for the **Codec Set** parameter.

- **Intra IP-IP Audio Connections** – Set to "**yes**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible within the same region.
- **Inter IP-IP Audio Connections** – Set to "**yes**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible between regions.
- **UDP Port Min**: - Set to **16384**
- **UDP Port Max**: - Set to **32767**

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: customerb.com
    Name: LOCAL
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
      Keep-Alive Count: 5
```

Figure 7: IP-Network-Region 1 Form – Page 1

2. On page 4 of the form:
   - Verify that region 1 is using codec 1 as specified on page 1 (this field is automatically populated in the **dst rgn** and **codec set** columns).
   - Next to the **dst rgn** row "**2**" enter **2** in the **codec set** column.
   - This results in codec set 2 being used for calls between region 1 (Local) and region 2 (AT&T).. Note that this relationship will be automatically populated on the region 2 form (see **Section 5.3.2**).

```
change ip-network-region 1                                     Page   4 of  20

Source Region: 1     Inter Network Region Connection Management     I       M
                                                                    G  A    t
dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
1    1                                                                all
2    2     y    NoLimit                                             n       t
3
```

Figure 8: IP-Network-Region 1 Form– Page 4

5.3.2. IP Network Region 2 – AT&T Region

In the reference configuration SIP trunk calls from AT&T are assigned to ip-network-region 2.

1. Enter the **change ip-network-region x**, where **x** is the number of an unused IP network region (e.g. **region 2**).
   - Enter **customerb.com** in the **Authoritative Domain** field.
   - Enter a descriptive name (e.g. **AT&T**).
   - Enter **2** for the **Codec Set** parameter.
   - **Intra IP-IP Audio Connections** – Set to "**yes**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible within the same region.
   - **Inter IP-IP Audio Connections** – Set to "**yes**", indicating that the RTP paths should be optimized to reduce the use of MedPro resources when possible between regions.
   - **UDP Port Min**: - Set to **16384**
   - **UDP Port Max**: - Set to **32767**

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location: 1       Authoritative Domain: customerb.com
    Name: AT&T
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 2              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 16384                     IP Audio Hairpinning? n
   UDP Port Max: 32767
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Figure 9: IP-Network-Region 2 Form for the AT&T IP Toll Free Service – Page 1

2. On Page 4 of the **ip-network-region** form:
   - Verify that region 2 is using codec 2 as specified on page 1 of the form (this field is automatically populated in the **dst rgn** and **codec set** columns).
   - Verify that region 1 is using codec 2 as specified in **Section 5.3.1** (this field was automatically populated in the **dst rgn** and **codec set** columns when the IP Network Region 1 form was submitted).
   - This results in codec set 2 being used for calls between AT&T and the Local regions.

```
change ip-network-region 2                                      Page   4 of  20

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                     G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    2     y    NoLimit                                             n       t
 2    2                                                                all
 3
 4
```

Figure 10: IP-Network-Region 2 Form for the AT&T IP Toll Free Service– Page 4

5.4. IP Codec Parameters

The “Local” IP Network Region 1 uses IP Codec set 1 (e.g. local station calls and calls to Modular Messaging). AT&T Toll Free calls access IP Network Region 2 and use IP Codec set 2.

5.4.1. IP Codec Set 1

1. Enter the **change ip-codec-set x** command, where **x** is the number of an IP codec set used only for internal calls. On Page 1 of the **ip-codec-set** form, ensure that “**G.711MU**”, “**G.729B**”, and “**G.729A**” are included in the codec list as shown in **Figure 11**.

```
change ip-codec-set 1                                         Page   1 of   2
                          IP Codec Set
    Codec Set: 1
    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2: G.729B            n          2        20
 3: G.729A            n          2        20
```

Figure 11: IP-Codec-Set Form for Internal Calls – Page 1

2. On Page 2 of the **ip-codec-set** form, set **FAX Mode** to “**t.38-standard**”.

```
change ip-codec-set 1                                         Page   2 of   2
                          IP Codec Set
                              Allow Direct-IP Multimedia? n
                    Mode                Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         off                   0
    Clear-channel   n                     0
```

Figure 12: IP-Codec-Set 1 Form for Internal Calls – Page 2

5.4.2. IP Codec Set 2

1. Enter the **change ip-codec-set x** command, where **x** is the number of an unused IP codec set (e.g. **2**). This IP codec set will be used for inbound AT&T IP Toll Free calls.
    a. On Page 1 of the **ip-codec-set** form, provision the codecs in the order shown in **Figure 13**. Note – See **Section 2.2.1** regarding support of G.726 codec.

```
change ip-codec-set 2                                          Page   1 of   2
                          IP Codec Set
    Codec Set: 2
    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729B           n          2        20
 2: G.729A           n          2        20
 3: G.711MU          n          2        20
```

Figure 13: IP-Codec-Set 2 Form for External Calls – Page 1

2. On Page 2 of the **ip-codec-set** form, set **FAX Mode** to "**t.38-standard**".

```
change ip-codec-set 2                                          Page   2 of   2
                          IP Codec Set
                              Allow Direct-IP Multimedia? n
                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         off                  0
    Clear-channel   n                    0
```

Figure 14: IP-Codec-Set 2 Form for External Calls – Page 2

5.5. IP Node Names Parameters

Node names define IP addresses to various Avaya components in the CPE.

1. Enter the **change node-names ip** command:
    a. Add a node name and the IP address for the Acme Packet 3800 (e.g. **Acme**).
    b. Add a node name and the IP address for Modular Messaging (e.g. **MM**).
2. Note the node name and IP address of the Processor Ethernet interface (**procr**) that was provisioned during installation.

```
change node-names ip                                             Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
Acme                  192.168.67.130
MM                    192.168.67.141
default               0.0.0.0
procr                 192.168.67.202
procr6                ::
```

Figure 15: Change Node-Names IP Form

5.6. IP Interfaces

1. In the reference configuration, the Processor Ethernet interface was assigned to region 1 during installation. Enter the **list ip-interface all** command to verify.

```
list ip-interface all
                                 IP INTERFACES
                                                                            Net
ON Type   Slot  Code/Sfx      Node Name/        Mask                        Rgn  VLAN
                              IP-Address/
                              Gateway Node
-- ------ ----- --------      ---------------   ----                        ---  ----
 y PROCR                      procr             /24                         1
                              192.168.67.202
                              192.168.67.1
 n PROCR                      procr6            /64                         1
```

Figure 16: List ip-interface all form

5.7. SIP Trunks

SIP trunks are defined on Communication Manager by provisioning a Signaling Group and a corresponding Trunk Group. Two SIP trunks are defined on Communication Manager in the reference configuration:

- For inbound AT&T IP Toll Free calls to Communication Manager – SIP Trunk **2**.
- For outbound Communication Manager calls to MM (coverage) and inbound Modular Messaging traffic (MWI) – SIP Trunk **3**.

Note – In the reference configuration TCP (port 5060) is used as the transport protocol between Communication Manager, Acme Packet, and Modular Messaging. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS (port 5061) to be used as the transport protocol where applicable.

5.7.1. Inbound AT&T Traffic

Communication Manager looks at the contents of the PAI header for admission control to the Signaling Groups via the *Far-End Domain* field. Note - This corresponds to the **sip-manipulation Mod_Inbound / Header-rule Inbound_PAI** setting (*customerb.com*) defined in the Acme SBC as shown in **Section 7**.

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g. **2**), and provision the following:
   - **Group Type** – Set to "**sip**".
   - **Transport Method** – Set to "**tcp**". The transport protocol used between Communication Manager and the Acme Packet SBC is TCP, and the transport protocol used between the Acme Packet SBC and the AT&T IP Toll Free service is UDP
   - **Peer Detection Enabled?** - Set to "**N**"
   - **Peer Server?** – Set to "**Others**"

- **Near-end Node Name** – Set to the node name of the Processor Ethernet interface noted in **Section 5.5** (e.g. **procr**).
- **Far-end Node Name** – Set to the node name of the Acme Packet SBC as administered in **Section 5.5** (e.g. **Acme**).
- **Near-end Listen Port** and **Far-end Listen Port** – set to "**5060**" (see Transport Method note above).
- **Far-end Network Region** – Set to the IP network region **2**, as defined in **Section 5.3.2** to represent the AT&T IP Toll Free service.
- **Far-end Domain** – Set to **customerb.com**..
- **DTMF over IP** – Set to "**rtp-payload**" to enable Communication Manager to use DTMF according to RFC 2833.
- **Direct IP-IP Audio Connections** – Set to "**y**", indicating that the RTP paths should be optimized to reduce the use of Media Processor resources when possible.
- **Enable Layer 3 Test** – Set to "y" to have Communication Manager send SIP OPTIONS "pings" to the Acme Packet SBC for link status.

```
add signaling-group 2
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? n  Peer Server: Others

 Near-end Node Name: procr                  Far-end Node Name: Acme
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 2

Far-end Domain: customerb.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

Figure 17: Signaling-Group 2 Form (inbound from AT&T)

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **2**). On Page 1 of the **trunk-group** form, provision the following:
   - **Group Type** – Set to "**sip**".
   - **Group Name** – Enter a descriptive name (e.g. **ATT_Inbound**).
   - **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g. **102**).
   - **Direction** – Set to "**incoming**".
   - **Service Type** – Set to "**public-ntwrk**".
   - **Signaling Group** – Set to the number of the signaling group administered in **Step 1** (e.g. **2**).

- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group (e.g. **20**).

```
add trunk-group 2                                             Page   1 of  21
                                TRUNK GROUP

Group Number: 2                    Group Type: sip          CDR Reports: y
Group Name: ATT_Inbound              COR: 1        TN: 1            TAC: 102
Direction: incoming          Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk           Auth Code? n
                                              Member Assignment Method: auto
                                                     Signaling Group: 2
                                                   Number of Members: 20
```

Figure 18: Trunk-Group 2 Form (inbound from AT&T) – Page 1

3. On page 2 of the form set **Preferred Minimum Session Refresh Interval(sec)** to **900**.

```
add trunk-group 2                                            Page   2 of  21
Group Type: sip
TRUNK PARAMETERS
     Unicode Name: auto
                                            Redirect On OPTIM Failure: 5000
            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 900
                                 Delay Call Setup When Accessed Via IGAR? n
```

Figure 19: Trunk-Group 2 Form (inbound from AT&T) – Page 2

4. On Page 3 of the form, set **Numbering Format** to **public**.

```
add trunk-group 2                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                   Maintenance Tests? y
                     Numbering Format: public
                                             UUI Treatment: service-provider
                                              Replace Restricted Numbers? n
                                             Replace Unavailable Numbers? n
                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
```

Figure 20: Trunk-Group 2 Form (inbound from AT&T) – Page 3

5. On Page 4 of the form:
   - Set **Support Request History?** to **n**.
   - Set **Telephone Event Payload Type:** to **100**.
6. Leave the remaining fields at their default values.

```
add trunk-group 2                                                   Page   4 of  21
                              PROTOCOL VARIATIONS
                        Mark Users as Phone? n
               Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                  Network Call Redirection? n
                     Send Diversion Header? n
                   Support Request History? n
              Telephone Event Payload Type: 100
        Convert 180 to 183 for Early Media? y
 Always Use re-INVITE for Display Updates? n
                              Enable Q-SIP? n
```

Figure 21: Trunk-Group 2 Form (inbound from AT&T) – Page 4

5.7.2. Modular Messaging Traffic (Coverage & MWI)

This trunk is used by Communication Manager to send call coverage to Modular Messaging, and for Modular Messaging to send MWI notifications to Communication Manager.

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group (e.g. **3**), and provision the form as shown in **Section 5.7.1** with the following changes:
   - **Far-end Node Name** – Set to the node name of Modular Messaging as administered in **Section 5.5** (e.g. **MM**).
   - **Far-end Network Region** – Set to the IP network region **1**, as defined in **Section 5.3.2** to represent the local CPE.

```
add signaling-group 3
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                           SIP Enabled LSP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
 Peer Detection Enabled? n  Peer Server: Others
 Near-end Node Name: procr                    Far-end Node Name: MM
 Near-end Listen Port: 5060                   Far-end Listen Port: 5060
                                           Far-end Network Region: 1
Far-end Domain: customerb.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Figure 22: Signaling-Group 3 Form (to/from Modular Messaging).

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group (e.g. **3**) and provision the form as shown in **Section 5.7.1** with the following changes:
   - **Group Name** – Enter a descriptive name (e.g. **MM**).

- **TAC** – Enter a trunk access code that is consistent with the dial plan (e.g. **103**).
- **Direction** – Set to "**two-way**".
- **Service Type** – Set to "**tie**".
- **Signaling Group** – Set to the number of the signaling group administered in **Step 1** (e.g. **3**).
- **Number of Members** – Enter the maximum number of simultaneous calls permitted on this trunk group (e.g. **20**).

```
add trunk-group 3                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip          CDR Reports: y
Group Name: MM                        COR: 1        TN: 1          TAC: 103
Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie          Auth Code? n
                                              Member Assignment Method: auto
                                                      Signaling Group: 3
                                                    Number of Members: 20
```

Figure 23: Trunk-Group 3 Form (to/from Modular Messaging) – Page 1

3. For page 2 of the form, use the same values as shown in **Section 5.7.1**.
4. On Page 3 of the form, set **Numbering Format** to **private**.

```
add trunk-group 3                                        Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y
                     Numbering Format: private
                                                UUI Treatment: service-provider
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
```

Figure 24: Trunk-Group 3 Form (MM) – Page 3

5. On Page 4 of the form use the same values as shown in **Section 5.7.1**.:

5.8. Public Unknown Numbering

For AT&T IP Toll Free service call admission control purposes, calling number origination SIP header contents (e.g. Contact and PAI) need to be converted to IP Toll Free service DIDs, instead of Communication Manager local extensions. Public-unknown-numbering is associated with the ***Numbering Format: public*** parameter specified on trunk 2 in **Section 5.7.1**.

These functions are accomplished using the Communication Manager *change public-unknown-numbering 0* command.

1. In the **public-unknown-numbering** form, for any local extension assigned to Communication Manager (stations, agents, skills, hunt groups, or VDNs), that may be called by the IP Toll Free service, provision an entry as follows:
   - **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
   - **Ext Code –** Enter the associated local extension (e.g. **40002** for Agent/Skill2).
   - **Trk Grp(s)** – Enter the number of the trunk group defined in **Section 5.7.1** (e.g. **2**).
   - **CPN Prefix** – Enter an associated IP Toll Free DID (e.g. **7323204300**).
   - **CPN Len** – Enter the total number of digits in the local extension range (e.g. **10**).
2. Add additional local extension to IP Toll Free DID entries as required.

```
change public-unknown-numbering 0                               Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN             CPN
Len Code            Grp(s)    Prefix          Len
                                                 Total Administered: 4
 5  40001           2         7323204300      10    Maximum Entries: 9999
 5  40002           2         7323204301      10
 5  40003           2         7323204302      10     Note: If an entry applies to
 5  40004           2         7323204303      10     a SIP connection to Avaya
                                                 Aura(tm) Session Manager,
                                                 the resulting number must
                                                 be a complete E.164 number.
```

Figure 25: Public-Unknown-Numbering Form

5.9. Private Unknown Numbering

Avaya Modular Messaging uses the History Info header for mail-box processing, so these must contain the Communication Manager extensions associated with the Modular Messaging mailboxes. Private-unknown-numbering is associated with the ***Numbering Format: private*** parameter specified on trunk 3 in **Section 5.7.2.**

These functions are accomplished using the Communication Manager *change private-unknown-numbering 0* command.

1. In the **private-unknown-numbering** form, for any local extension assigned to Communication Manager (stations, agents), that have mailboxes on Modular Messaging, enter a matching pattern. Note – In the reference configuration, the Modular Messaging mailboxes were defined using the corresponding Communication Manager extensions. If the Modular Messaging mailboxes are created with a different format, use the private-unknown-numbering form to convert the Communication Manager extensions to the mailbox format.

   - **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
   - **Ext Code –** Enter a matching pattern for Communication Manager extensions (e.g. **4**).
   - **Trk Grp(s)** – Enter the number of the Modular Messaging trunk group defined in **Section 5.7.2** (e.g. **3**).

- **CPN Prefix** – Leave blank if Modular Messaging also uses the Communication Manager extensions for the mailboxes. If Modular Messaging uses a different mailbox format, use this field to make that change.
- **CPN Len** – Enter the total number of digits for the Modular Messaging mailbox (e.g. **5**).

2. Add additional local extension/mailbox formats as required.

```
change private-numbering 0                                      Page   1 of   2
                          NUMBERING - PRIVATE FORMAT

Ext Ext             Trk         Private          Total
Len Code            Grp(s)      Prefix           Len
 5  4               3                            5     Total Administered: 1
                                                       Maximum Entries: 540
```

Figure 26: Private-Unknown-Numbering Form

5.10. Optional Features

The reference configuration uses hunt groups, vectors, and Vector Directory Numbers (VDNs), to provide additional functionality during testing:

- Hunt Group 1 – Modular Messaging coverage for Communication Manager extensions.
- VDN 47024/vector 24 – Auto-attendant.
- VDN 44000/vector 6 – Meet-me Conference
- VDN 44002/vector 2 – Skill2 (Agent2)

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and VDNs are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. Consult **[1], [2], [3], and [4]** for further details if necessary. The samples that follow are provided for reference purposes only.

5.10.1. Hunt Group for Station Coverage to Modular Messaging

Hunt group 1 is used in the reference configuration to verify the Send-All-Calls functionality. The hunt group (e.g. **1**) is defined with the 5 digit Modular Messaging pilot number (e.g. **46000** in **Figure 28**). The hunt group is associated with a coverage path (e.g.**H1** in **Figure 29**) and the coverage path is assigned to a station (e.g. **40002** in **Figure 30**).

```
display hunt-group 1                                            Page   1 of  60
                                HUNT GROUP

            Group Number: 1                               ACD? n
              Group Name: MM                            Queue? n
         Group Extension: 46000                        Vector? n
              Group Type: ucd-mia                Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

Figure 27: Hunt Group 1Form – Page 1

```
display hunt-group 1                                           Page   2 of  60
                                  HUNT GROUP
                      Message Center: sip-adjunct
     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                    (e.g., AAR/ARS Access Code)
     46000                    46000                     8
```

Figure 28: Hunt Group 1 Form – Page 2

```
 display coverage path 1
                                 COVERAGE PATH
                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage
COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
        Don't Answer?            y               y        Number of Rings: 3
                All?             n               n
 DND/SAC/Goto Cover?             y               y
COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1             Rng: 2  Point2:
  Point3:                        Point4:
```

Figure 29: Coverage Path 1 Form

```
display station 40002                                          Page   1 of   5
                                     STATION
Extension: 40002                         Lock Messages? n              BCC: M
     Type: 9630                          Security Code: 123456          TN: 1
     Port: S00000                      Coverage Path 1: 1              COR: 1
     Name: 9630_H323_One               Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 40002
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
                                                          IP Video? y
                              Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y
Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
                                              Customizable Labels? y
```

Figure 30: Station 40002 Form

5.10.2. Auto Attendant

A basic auto-attendant functionality is defined in the reference configuration for DTMF testing. The auto-attendant is defined by a VDN (e.g. **47024**) and a vector (e.g. **24**).

```
display vdn 47024                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER
                             Extension: 47024
                                 Name*: Auto-Attendant
                           Destination: Vector Number        24
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none
        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
* Follows VDN Override Rules
```

Figure 31: Auto Attendant VDN

```
display vector 24                                               Page   1 of   6
                                  CALL VECTOR
    Number: 24              Name: Auto-Attendant
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    4   secs hearing ringback
02 collect      5    digits after announcement 42012    for none
03 route-to     digits with coverage n
04 wait-time    5   secs hearing silence
05
```

Figure 32: Auto Attendant Vector

5.10.3. Meet-me Conference

A basic meet-me conference functionality is defined in the reference configuration for DTMF testing. The meet-me conference is defined by a VDN (e.g. **44000**) and a vector (e.g. **6**).

```
display vdn 44000                                               Page   1 of   3
                            VECTOR DIRECTORY NUMBER
                             Extension: 44000
                                  Name: MeetMeConf
                           Destination: Vector Number        6
                  Meet-me Conferencing? y
                                   COR: 1
                                    TN: 1
                                   COR: 1
                                    TN: 1
```

Figure 33: Meet-me Conference VDN – Page 1

```
display vdn 44000                                                Page   2 of   3
                            VECTOR DIRECTORY NUMBER
                  MEET-ME CONFERENCE PARAMETERS:
                    Conference Access Code: 123456
                     Conference Controller: 40000
                           Conference Type: 6-party
```

Figure 34: Meet-me Conference VDN – Page 2

```
display vector 6                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 6                Name: MeetMeConf
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? y           Lock? y
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    5   secs hearing ringback
02 collect      6    digits after announcement 42013
03 goto step    5             if digits           =      meet-me-access
04 goto step    2             if unconditionally
05 announcement 42001
06 route-to     meetme
07 stop
08
```

Figure 35: Meet-me Conference Vector

5.10.4. Skills

Skills are defined as a hunt groups and then are associated with VDNs/vectors.

```
display hunt-group 2                                             Page   1 of   4
                                  HUNT GROUP
            Group Number: 2                                 ACD? y
              Group Name: Skill2                          Queue? y
         Group Extension: 43002                          Vector? y
              Group Type: ead-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:
             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

Figure 36: Define skill hunt group

```
display vdn 44002                                            Page   1 of   3
                             VECTOR DIRECTORY NUMBER

                              Extension: 44002
                                  Name*: Skill2
                            Destination: Vector Number        2
                    Attendant Vectoring? n
                   Meet-me Conferencing? n
                     Allow VDN Override? n
                                    COR: 1
                                    TN*: 1
                               Measured: none
        VDN of Origin Annc. Extension*:
                             1st Skill*:
                             2nd Skill*:
```

Figure 37: Define skill VDN

```
display vector 2                                             Page   1 of   6
                                   CALL VECTOR

    Number: 2                   Name: Skill2
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 42002
03 queue-to     skill 2    pri m
04 wait-time    10  secs hearing music
05 announcement 42005
06 goto step    3             if unconditionally
07 stop
08
```

Figure 38: Define skill vector

6. Avaya Modular Messaging

In this reference configuration, Avaya Modular Messaging is used to verify DTMF, Message Waiting Indicator (MWI), as well as basic call coverage functionality. The Avaya Modular Messaging used in the reference configuration is provisioned for Multi-Site mode. Multi-Site mode allows Avaya Modular Messaging to serve subscribers in multiple locations. The administration for Modular Messaging is beyond the scope of these Application Notes, (consult **[5], [6], [7],** and **[8]** for further details).

7. Configure Acme Packet SBC

These Application Notes assume that basic Acme Packet SBC administration has already been performed. In the reference configuration two Acme Packet Net-Net 3800s[3] are implemented in a High Availability (HA) configuration. The Acme Packet SBC configuration used in the reference configuration is provided below as a reference.

Note - The AT&T IP Toll Free service border element IP addresses shown in this document are examples. AT&T Customer Care will provide the actual IP addresses as part of the IP Toll Free provisioning process.

Note – In the following sections, only the parameters that are highlighted in **bold** text are applicable to these application notes. Other parameter values may or may not match based on local configurations and are shown for illustrative purposes. Consult with Acme Packet Support **[9]** for further details and explanations on the configuration below.

ANNOTATION: The local policies below govern the routing of SIP messages between elements on the CPE, e.g. Communication Manager, and the AT&T IP Toll Free service. The Session Agent Groups (SAG) defined here, and further down, are provisioned under the session-groups "SP-PROXY" and "ENTERPRISE".

```
local-policy
      from-address
                               *
      to-address
                               *
      source-realm
                               INSIDE
      description
      activate-time            N/A
      deactivate-time          N/A
      state                    enabled
      policy-priority          none
      policy-attribute
            next-hop                 SAG:SP_PROXY
            realm                    OUTSIDE
            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
```

[3]Although an Acme Net-Net SD 3800 was used in the reference configuration, theses configurations also apply to the 4250, and 4500 platforms.

ANNOTATION: The local policy below governs the routing of SIP messages from the AT&T IP Toll Free service to Communication Manager.

```
local-policy
      from-address
                                     *
      to-address
                                     *
      source-realm
                                     OUTSIDE
      description
      activate-time                  N/A
      deactivate-time                N/A
      state                          enabled
      policy-priority                none
      policy-attribute
            next-hop                       SAG:ENTERPRISE
            realm                          INSIDE
            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        775880
      max-untrusted-signaling        80
      min-untrusted-signaling        20
```

```
      app-signaling-bandwidth        0
      tolerance-window               30
      rtcp-rate-limit                0
      min-media-allocation           2000
      min-trusted-allocation         4000
      deny-allocation                64000
      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

network-interface
      name                           wancom1
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.1.1
      sec-utility-addr               169.254.1.2
      netmask                        255.255.255.252
      gateway
      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


network-interface
      name                           wancom2
      sub-port-id                    0
      description
      hostname
      ip-address
      pri-utility-addr               169.254.2.1
      sec-utility-addr               169.254.2.2
      netmask                        255.255.255.252
      gateway
      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
```

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

```
network-interface
      name                           s0p0
      sub-port-id                    0
      description
      hostname
      ip-address                     192.168.64.130
      pri-utility-addr               192.168.64.131
      sec-utility-addr               192.168.64.132
      netmask                        255.255.255.0
      gateway                        192.168.64.1
      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                    192.168.64.130
      ftp-address
        icmp-address                   192.168.64.130
      snmp-address
      telnet-address
```

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                          s0p1
      sub-port-id                   0
      description
      hostname
      ip-address                    192.168.67.130
      pri-utility-addr              192.168.67.131
      sec-utility-addr              192.168.67.132
      netmask                       255.255.255.0
      gateway                       192.168.67.1
      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                   192.168.67.130
      ftp-address                   192.168.67.130
        icmp-address                  192.168.67.130
      snmp-address
      telnet-address

ntp-config
      server                        135.8.139.1
      last-modified-by              admin@console
      last-modified-date            2009-11-04 00:27:53

phy-interface
      name                          s0p1
      operation-type                Media
      port                          1
      slot                          0
      virtual-mac                   00:08:25:a0:f3:69
      admin-state                   enabled
      auto-negotiation              enabled
      duplex-mode                   FULL
      speed                         100


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

phy-interface
      name                           s1p0
      operation-type                 Media
      port                           0
      slot                           1
      virtual-mac                    00:08:25:a0:f3:6e
      admin-state                    disabled
      auto-negotiation               enabled
      duplex-mode                    FULL
      speed                          100

phy-interface
      name                           s1p1
      operation-type                 Media
      port                           1
      slot                           1
      virtual-mac                    00:08:25:a0:f3:6f
      admin-state                    disabled
      auto-negotiation               enabled
      duplex-mode                    FULL
      speed                          100

phy-interface
      name                           wancom1
      operation-type                 Control
      port                           1
      slot                           0
      virtual-mac
      wancom-health-score            8

phy-interface
      name                           wancom2
      operation-type                 Control
      port                           2
      slot                           0
      virtual-mac
      wancom-health-score            9
```

ANNOTATION: The realm configuration "OUTSIDE" below represents the external network on which the AT&T IP Toll Free service resides, and applies the sip-manipulation NAT_IP.

```
realm-config
      identifier                     OUTSIDE
      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              NAT_IP
manipulation-string
class-profile
average-rate-limit              0
access-control-trust-level      medium
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
```

ANNOTATION: The realm configuration "INSIDE" below represents the internal network on which the Avaya elements reside, and applies the sip-manipulation **Mod_Inbound_to_From.**

```
realm-config
        identifier                     INSIDE
        description
        addr-prefix                    0.0.0.0
        network-interfaces
                                       s0p1:0
        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       Mod_Inbound
        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

redundancy-config
      state                          enabled
      log-level                      INFO
      health-threshold               75
      emergency-threshold            50
      port                           9090
      advertisement-time             500
      percent-drift                  210
      initial-time                   1250
      becoming-standby-time          180000
      becoming-active-time           100
      cfg-port                       1987
      cfg-max-trans                  10000
      cfg-sync-start-time            5000
      cfg-sync-comp-time             1000
      gateway-heartbeat-interval     0
      gateway-heartbeat-retry        0
      gateway-heartbeat-timeout      1
      gateway-heartbeat-health       0
      media-if-peercheck-time        0
      peer
            name                           acmesbc-pri
            state                          enabled
            type                           Primary
            destination
                  address                        169.254.1.1:9090
                  network-interface              wancom1:0
            destination
                  address                        169.254.2.1:9090
```

```
            network-interface             wancom2:0
      peer
            name                          acmesbc-sec
            state                         enabled
            type                          Secondary
            destination
                  address                       169.254.1.2:9090
                  network-interface             wancom1:0
            destination
                  address                       169.254.2.2:9090
                  network-interface             wancom2:0
```

> **ANNOTATION**: The **session agent** below represents the AT&T IP Toll Free service network border element. The Acme will attempt to send calls to the border element based on successful responses to the OPTIONS "ping-method". The AT&T IP Toll Free service border element is also specified in the **session-group** section below. Redundant network session-agents may be defined (see **Addendum 1**).
>
> **NOTE** – The ***ping-method OPTIONS;hops=20*** parameter shown below was a setting used in the reference test environment. Acme Packet best practices recommends a setting of ***OPTIONS;hops=0*** in customer deployments.

```
session-agent
      hostname                      135.25.29.74
      ip-address                    135.25.29.74
      port                          5060
      state                         enabled
      app-protocol                  SIP
      app-type
      transport-method              UDP
      realm-id                      OUTSIDE
      egress-realm-id
      description                   AT&T_BE
      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=20
ping-interval                  60
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              NONE
tcp-keepalive                  none
tcp-reconn-interval            0
max-register-burst-rate        0
register-burst-window          0
```

ANNOTATION: The session agent below represents the Communication Manager Processor Ethernet interface used in the reference configuration(see **Section 5.5**).

```
session-agent
      hostname                 192.168.67.202
      ip-address               192.168.67.202
      port                     5060
      state                    disabled
      app-protocol             SIP
      app-type
      transport-method         StaticTCP
      realm-id                 INSIDE
```

```
egress-realm-id
description                     ACM60
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                 Proxy
loose-routing                   enabled
send-media-session              enabled
response-map
ping-method                     OPTIONS;hops=0
ping-interval                   60
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
manipulation-pattern
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                  none
        tcp-reconn-interval            0
        max-register-burst-rate        0
        register-burst-window          0
        sip-profile
        sip-isup-profile
```

ANNOTATION: The **session group** below specifies the AT&T IP Toll Free service border element (see **session-agent 135.25.29.74** above). **Note** - Multiple session-agents may be specified in a session-group for network redundancy(see **Addendum 1**).

```
session-group
        group-name                     SP_PROXY
        description
        state                          enabled
        app-protocol                   SIP
        strategy
        dest
                                       135.25.29.74
        trunk-group
        sag-recursion                  enabled
        stop-sag-recurse               401,407
```

ANNOTATION: The session group below represents the Communication Manager session-agent defined above. This session-group is specified in the local-policy source-realm "OUTSIDE". Please note that multiple destinations can be added if more than one Communication Manager exists.

```
session-group
        group-name                     ENTERPRISE
        description
        state                          enabled
        app-protocol                   SIP
        strategy                       Hunt
        dest
                                       192.168.67.202
        trunk-group
        sag-recursion                  disabled
        stop-sag-recurse               401,407
```

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                  INSIDE
      egress-realm-id                INSIDE
      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


sip-feature
      name                           Replaces
      realm
      support-mode-inbound           Pass
      require-mode-inbound           Pass
      proxy-require-mode-inbound     Pass
      support-mode-outbound          Pass
      require-mode-outbound          Pass
      proxy-require-mode-outbound    Pass
```

```
ANNOTATION: The SIP interface below is used to communicate with the AT&T IP Toll
Free service.
```

```
sip-interface
      state                          enabled
      realm-id                       OUTSIDE
      description
      sip-port
            address                        192.168.64.130
            port                           5060
            transport-protocol             UDP
            tls-profile
            allow-anonymous                agents-only
            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
refer-call-transfer            disabled
route-unauthorized-calls
tcp-keepalive                  none
add-sdp-invite                 disabled
add-sdp-profiles
```

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

```
sip-interface
      state                          enabled
      realm-id                       INSIDE
      description
      sip-port
            address                        192.168.67.130
            port                           5060
            transport-protocol             TCP
            tls-profile
            allow-anonymous                agents-only
            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
refer-call-transfer               disabled
route-unauthorized-calls
tcp-keepalive                     none
add-sdp-invite                    disabled
add-sdp-profiles
```

ANNOTATION: The **NAT_IP** sip-manipulation below performs address translation and topology hiding for SIP messages between the AT&T IP Toll Free services and the Avaya elements. The NAT function is comprised of the header rules **manipFrom** and **manipTo.**
In the header-rule **manipFrom** the Acme will convert this value to the "outside" IP address of the Acme (**$Local_IP**).
In the header-rule **manipTo**, the Acme will convert this value to the IP address of the AT&T IP Toll Free border element (**$Remote_IP**).

```
sip-manipulation
      name                              NAT_IP
      description
      header-rule
            name                              manipFrom
            header-name                       From
            action                            manipulate
            comparison-type                   case-sensitive
```

```
        match-value
        msg-type                        request
        new-value
        methods
        element-rule
              name                            FROM
              parameter-name
              type                            uri-host
              action                          replace
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value                       $LOCAL_IP
  header-rule
        name                            manipTo
        header-name                     To
        action                          manipulate
        comparison-type                 case-sensitive
        match-value
        msg-type                        request
        new-value
        methods
        element-rule
              name                            TO
              parameter-name
              type                            uri-host
              action                          replace
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value
              new-value                       $REMOTE_IP
```

ANNOTATION: The **Mod_Inbound** sip-manipulation below modifies To, From, PAI, and Request URI headers leaving the Acme inside interface to Communication Manager. The To headers are modified to *customerb.com* instead of Acme outside address (192.168.64.130), and the From header are modified from the AT&T BE address (135.25.29.74) to the Acme inside address 192.168.67.130. The inbound PAI is modified from the AT&T BE IP address to *customerb.com*. The inbound Request URI is modified from the Communication Manager Processor Ethernet (procr) IP address to *customerb.com.*

```
sip-manipulation
  name                            Mod_Inbound
  description
  split-headers
  join-headers
  header-rule
        name                            Inbound_To
        header-name                     To
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        request
        methods
```

```
        match-value
        new-value
        element-rule
              name                              To
              parameter-name
              type                              uri-host
              action                            replace
              match-val-type                    any
              comparison-type                   case-sensitive
              match-value                       192.168.64.130
              new-value                         customerb.com
  header-rule
        name                              Inbound_From
        header-name                       From
        action                            manipulate
        comparison-type                   case-sensitive
        msg-type                          request
        methods
        match-value
        new-value
        element-rule
              name                              From
              parameter-name
              type                              uri-host
              action                            replace
              match-val-type                    any
              comparison-type                   case-sensitive
              match-value
              new-value                         $LOCAL_IP
  header-rule
        name                              Inbound_RURI
        header-name                       request-uri
        action                            manipulate
        comparison-type                   case-sensitive
        msg-type                          request
        methods                           INVITE
        match-value
        new-value
        element-rule
              name                              modRURI
              parameter-name
              type                              uri-host
              action                            replace
              match-val-type                    any
              comparison-type                   case-sensitive
              match-value                       192.168.67.202
              new-value                         customerb.com
   header-rule
        name                              Inbound_PAI
        header-name                       P-Asserted-Identity
        action                            manipulate
        comparison-type                   case-sensitive
        msg-type                          request
        methods
        match-value
```

```
        new-value
        element-rule
              name                            modPAI
              parameter-name
              type                            uri-host
              action                          replace
              match-val-type                  any
              comparison-type                 case-sensitive
              match-value                     135.25.29.74
              new-value                       customerb.com
```

> **ANNOTATION**: The steering pools below define the IP Addresses and RTP port ranges on the respective realms. The "OUTSIDE" realm IP Address will be used as the CPE media traffic IP Address to communicate with AT&T. **The "OUTSIDE" realm RTP port range is an AT&T IP Toll Free service requirement.** Likewise, the IP Address and RTP port range defined for the "INSIDE" realm steering pool will be used to communicate with the Avaya elements. Please note that the "INSIDE" realm port range does not have to be within the range specified below.

```
steering-pool
      ip-address                      192.168.64.130
      start-port                      16384
      end-port                        32767
      realm-id                        OUTSIDE
      network-interface


steering-pool
      ip-address                      192.168.67.130
      start-port                      16384
      end-port                        32767
      realm-id                        INSIDE
      network-interface


system-config
      hostname                        acmesbc
      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               135.8.139.1
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
```

8. Verification Steps

The following steps may be used to verify the configuration:

8.1. General

1. Place an inbound call to a VDN/vector, agent or phone, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Verify that the AT&T IP Toll Free features for hold, resume, conference and transfer can be executed via RFC 2833 DTMF signaling.
3. Place an inbound call to an agent or phone, but do not answer the call. Verify that the call covers to Modular Messaging voicemail. Retrieve the message from Modular Messaging.

8.2. Avaya Aura® Communication Manager 6.0

The following examples are only a few of the monitoring commands available on Communication Manager. See **[1] and [2]** for more information.

1. From the Communication Manager console connection enter the command ***list trace tac xxx***, where **xxx** is a trunk access code defined for the SIP trunk to AT&T (e.g. **102**)

```
list trace tac 102                                                    Page   1
                                LIST TRACE
time            data
14:09:48 TRACE STARTED 03/30/2011 CM Release String cold-00.0.345.0-18567
14:10:10 SIP<INVITE sip:000001012@customerb.com:5060;transport=t
14:10:10 SIP<cp SIP/2.0
14:10:10     active trunk-group 2 member 1  cid 0x22d
14:10:10    0  0 ENTERING TRACE cid 557
14:10:10    2  1 vdn e44002 bsr appl   0 strategy 1st-found override n
14:10:10    2  1 wait 2 secs hearing ringback
14:10:10 SIP>SIP/2.0 183 Session Progress
14:10:10     dial 44002
14:10:10     ring vector 2    cid 0x22d
14:10:10     G729 ss:off ps:20
             rgn:2 [192.168.67.130]:16908
             rgn:1 [192.168.67.203]:16390
14:10:10     xoip options: fax:T38 modem:off tty:US  uid:0x5000d
             xoip ip: [192.168.67.203]:16390
14:10:12    2  2 announcement 42002
14:10:12 SIP>SIP/2.0 183 Session Progress
14:10:12    2  2      announcement: board 001V9 ann ext: 42002
14:10:12 SIP>SIP/2.0 200 OK
14:10:12     active announcement    42002 cid 0x22d
14:10:12     hear annc board 001V9 ext 42002 cid 0x22d
14:10:12 SIP<ACK sip:192.168.67.202;transport=tcp SIP/2.0
14:10:14     idle announcement     cid 0x22d
14:10:14    2  3 queue-to
14:10:14    2  3      queuing to skill   2 pri m
14:10:14    2  3 Local Agent Preference=n
14:10:14    2  3 Agent Login ID: 47002  Logged in at station: 40002
14:10:14 SIP>INVITE sip:8181084000@192.168.67.130:5060;transport
```

```
14:10:14 SIP>=tcp SIP/2.0
14:10:14     2  3 LEAVING VECTOR PROCESSING cid 557
14:10:15      active station    40002 cid 0x22d
14:10:15 SIP<SIP/2.0 200 OK
14:10:15 SIP>ACK sip:8181084000@192.168.67.130:5060;transport=tcp
```

Figure 39: Communication Manager *list trace tac 102* – Inbound call to Skill/Agent.

2. Similar Communication Manager commands are, ***list trace station***, ***list trace vdn***, and ***list trace vector***. Other useful commands are ***status trunk*** and ***status station***.

8.3. Protocol Traces

Using a SIP protocol analyzer (e.g. Wireshark), monitor the SIP traffic at the Acme Packet SBC public "outside" interface connection to the AT&T IP Toll Free service.

1. The following is an example of an inbound call filtering on the SIP protocol.

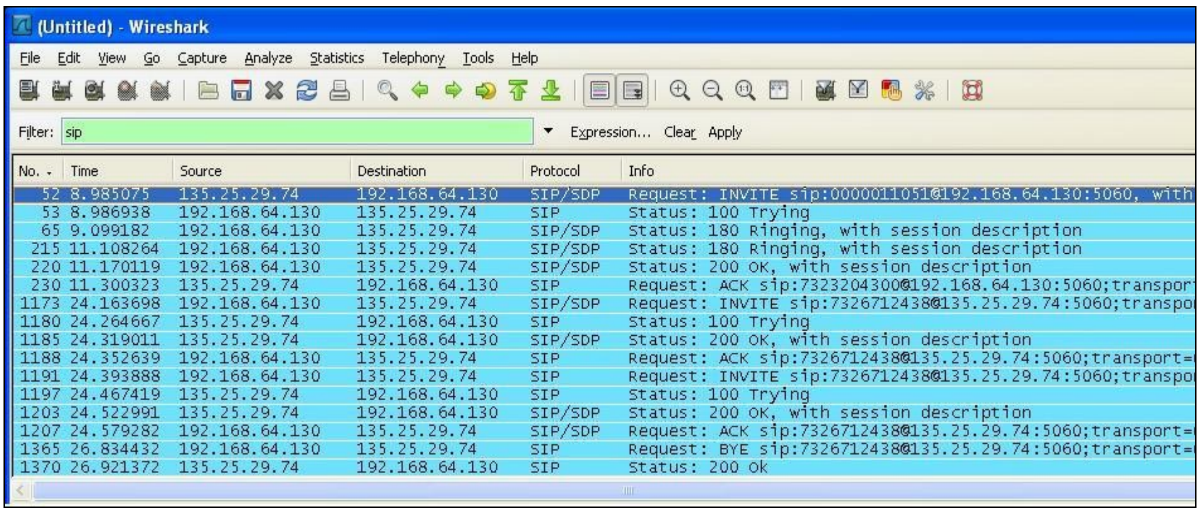


Figure 40: –SIP Protocol trace – Inbound call from AT&T

2. The following is an example of an inbound call filtering on outbound DTMF events.

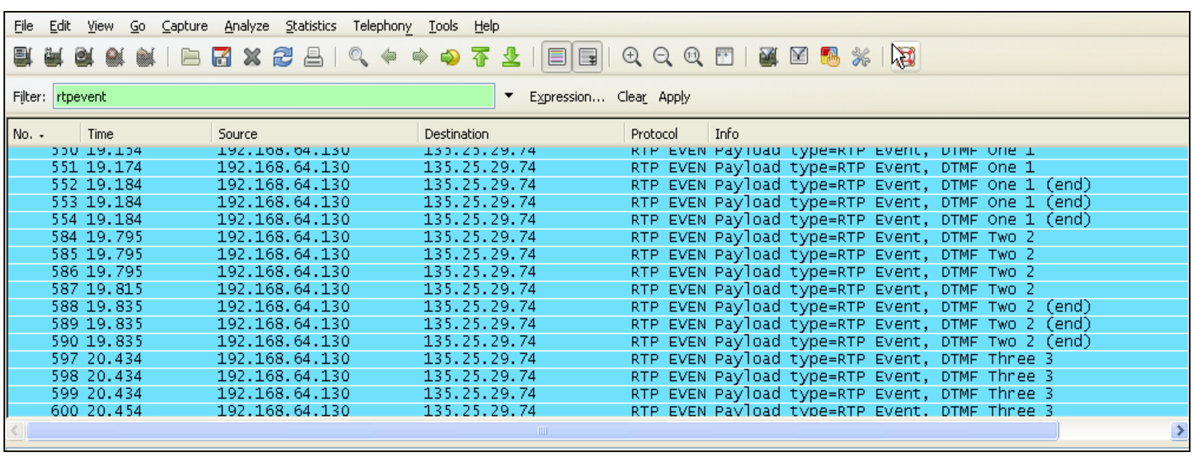
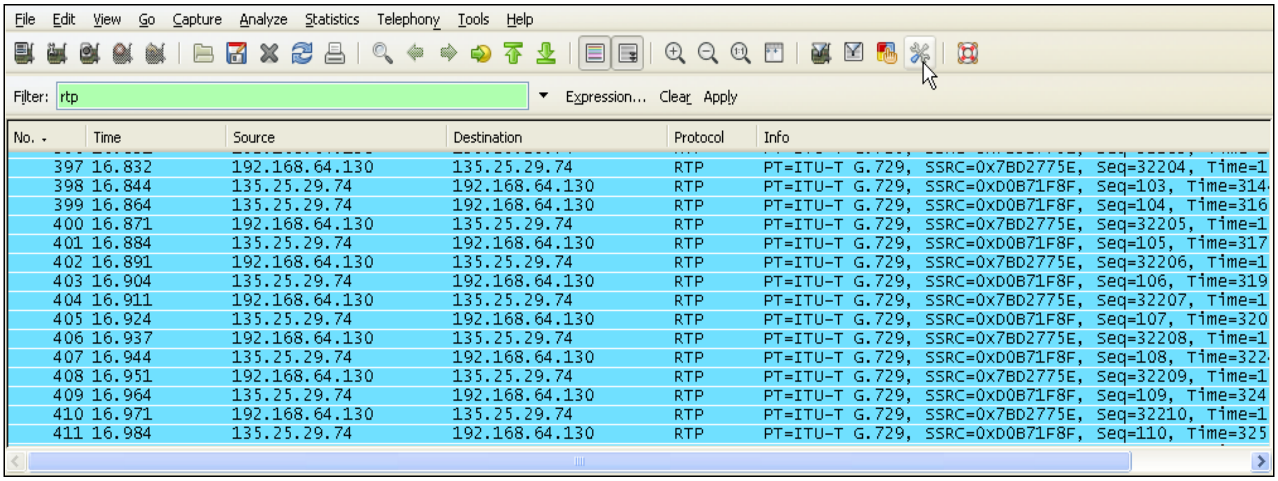


Figure 41: – RTPEvent (DTMF) trace – Outbound DTMF events to AT&T

3. The following is an example of an inbound call filtering on RTP.



File Edit View Go Capture Analyze Statistics Telephony Tools Help

Filter: rtp Expression... Clear Apply

No.	Time	Source	Destination	Protocol	Info
397	16.832	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32204, Time=1
398	16.844	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=103, Time=314
399	16.864	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=104, Time=316
400	16.871	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32205, Time=1
401	16.884	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=105, Time=317
402	16.891	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32206, Time=1
403	16.904	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=106, Time=319
404	16.911	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32207, Time=1
405	16.924	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=107, Time=320
406	16.937	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32208, Time=1
407	16.944	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=108, Time=322
408	16.951	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32209, Time=1
409	16.964	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=109, Time=324
410	16.971	192.168.64.130	135.25.29.74	RTP	PT=ITU-T G.729, SSRC=0x7BD2775E, Seq=32210, Time=1
411	16.984	135.25.29.74	192.168.64.130	RTP	PT=ITU-T G.729, SSRC=0xD0B71F8F, Seq=110, Time=325

Figure 42: – RTP trace (showing codec used) – inbound call to AT&T

8.4. Acme Packet SBC

The Acme Packet SBC provisioning can be checked by entering the command **verify-config**. Acme maintenance manuals may be found at **[9]** for additional maintenance commands**.**

9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Communication Manager 6.0 and the Acme Packet Net-Net 3800 can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of Avaya Aura® Communication Manager the ability to support inbound calls over an AT&T IP Toll Free SIP trunk service connection via **MIS/PNT** transport. These Application Notes further demonstrated that the Acme Packet Net-Net is utilized to provide SIP header manipulation for inbound calls. The reference 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.

10. References

The Avaya product documentation is available at http://support.avaya.com unless otherwise noted.

[1] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010
[2] *Avaya Aura® Communication Manager Feature Description and Implementation,* Release 6.0, 555-245-205, Issue 8.0, June 2010
[3] *Administering Avaya Aura® Call Center Features*, Release 6.0, June 2010
[4] *Programming Call Vectors in Avaya Aura® Call Center*, 6.0, June 2010
[5] *Modular Messaging Multi-Site Guide Release 5.1*, June 2009
[6] *Modular Messaging for Microsoft Exchange Release 5.1 Installation and Upgrades*, June 2009
[7] *Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release 5.1 Installation and Upgrades*, June 2009
[8] *Modular Messaging for IBM Lotus Domino 5.1 Installation and Upgrades*, June 2009

Acme Packet Support (login required):
[9] http://www.acmepacket.com/support.htm

AT&T IP Toll Free Service Descriptions:
[10] *AT&T IP Toll Free*
http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/

11. Addendum 1 - Acme Packet Net-Net Redundancy to Multiple AT&T Border Elements

AT&T may provide multiple network border elements for redundancy purposes. The Acme Packet Net-Net SBC can be provisioned to support this redundant configuration.

Given two AT&T border elements **135.25.29.74** (Primary) and **135.25.29.75 (Secondary)**, and building on the configuration shown in **Section 8**, the Acme Packet Net-Net SBC is provisioned as follows.

ANNOTATION: The **session agents** below represent the AT&T IP Toll Free service border elements. The Acme will attempt to send calls to the Primary or Secondary border elements based on successful responses to the OPTIONS "ping-method". Both AT&T IP Toll Free service border elements are also specified in the **session-group** section below.

NOTE – The ***ping-method OPTIONS;hops=20*** parameter shown below was a setting used in the reference test environment. Acme Packet best practices recommends a setting of ***OPTIONS;hops=0*** in customer deployments.

```
session-agent
      hostname                      135.25.29.74
      ip-address                    135.25.29.74
      port                          5060
      state                         enabled
      app-protocol                  SIP
      app-type
      transport-method              UDP
      realm-id                      OUTSIDE
      egress-realm-id
      description                   AT&T_BE_Primary
      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=20
        ping-interval                  60
        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              NONE
        tcp-keepalive                  none
        tcp-reconn-interval            0
        max-register-burst-rate        0
        register-burst-window          0


session-agent
        hostname                       135.25.29.75
        ip-address                     135.25.29.75
        port                           5060
        state                          enabled
        app-protocol                   SIP
        app-type
        transport-method               UDP
        realm-id                       OUTSIDE
        egress-realm-id
        description                    AT&T_BE_Secondary
        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=20
ping-interval                  60
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              NONE
tcp-keepalive                  none
tcp-reconn-interval            0
max-register-burst-rate        0
register-burst-window          0
```

ANNOTATION: The **session group** below specifies the AT&T IP Toll Free service border elements (see **session-agents** above). Also a **strategy** of "RoundRobin" is defined. This means the Acme will alternatively select between the two session-agents. An alternative is to use a strategy of "Hunt" (the secondary BE will only be used if access to the Primary fails). This session-group is also specified in the local-policy source-realm "INSIDE".

```
session-group
      group-name                      SP_PROXY
      description
      state                           enabled
      app-protocol                    SIP
      strategy                        RoundRobin
      dest
                                           135.25.29.74
                                           135.25.29.75
      trunk-group
      sag-recursion                   enabled
      stop-sag-recurse                401,407
```

ANNOTATION: - The following header-rule is added to the "**NAT_IP**" sip-manipulation shown in **Section 8**. This header-rule inserts the IP address of the AT&T BE being used for the call (determined by the session-group above) into the SIP Request-URI header.

```
header-rule
                name                            manipRURI
                header-name                     request-uri
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods                         INVITE
                match-value
                new-value
                element-rule
                        name                            modRURI
                        parameter-name
                        type                            uri-host
                        action                          replace
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       $REMOTE_IP
```

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.