Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya Aura® Communication Manager 5.2.1 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 5.2.1 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.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager 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 5.2.1 is a telephony application server and is the point of connection between the enterprise endpoints and the Acme Packet Net-Net 3800. In the reference configuration, Avaya Aura® Communication Manager 5.2.1 is provisioned in an Access Element configuration (note that SIP endpoint is not supported in an Aura® Communication Manager 5.2.1 Access Element configuration). 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 5.2.1, Avaya IP and Digital 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.

[1] MIS/PNT does not support cRTP.

2.2. Test Results

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

- 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 5.2.1 in an Access Element 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 (Access Element configuration) provides the voice communications services for a particular enterprise site, including H.323 and Digital stations. Communication Manager Access Element configurations do not support SIP stations. In this reference configuration, Communication Manager runs on an Avaya S8720 Server. This solution is extensible to other Avaya S8xxx Servers.

  The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In this reference configuration, an Avaya G650 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 6400 Series Digital Telephone. An Avaya One-X® Agent, a PC based H323 softphone, was also used in the reference configuration. Note – SIP stations are not supported with the Communication Manager Access Element 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 the Communication Manager Control LAN (C-LAN).
- 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.

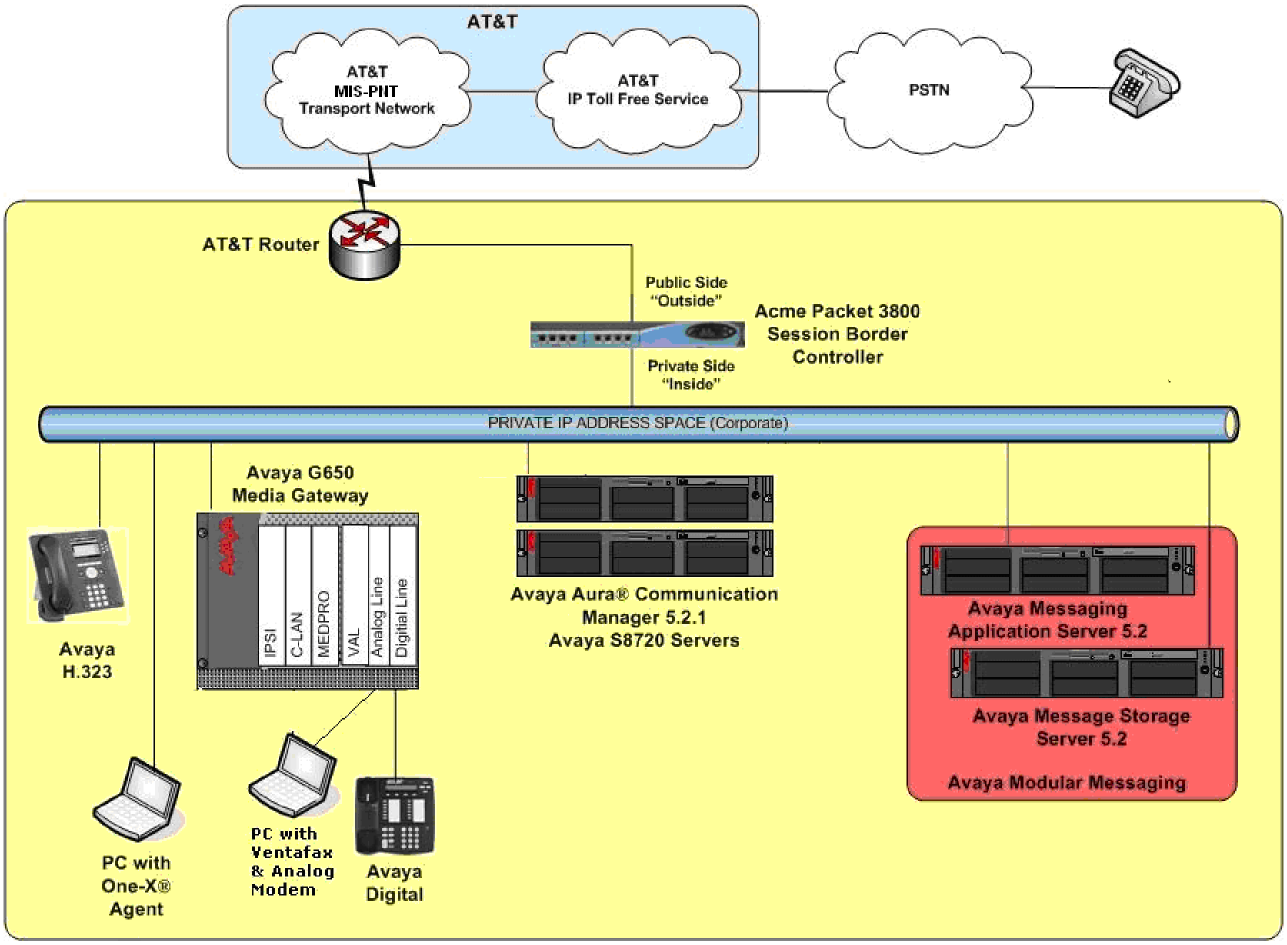

Figure 1: Reference configuration

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 specific 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	
C-LAN IP Address	192.168.67.14
Avaya Aura® Communication Manager extensions	26xxx
Voice Messaging Pilot Extension	26000
Avaya Modular Messaging	
Messaging Application Server (MAS) IP Address	192.168.67.141
Messaging Server (MSS) IP Address	192.168.67.140
Modular Messaging dial plan	17231126xxx
Pilot Number	17231126000
Acme Packet 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 telephone, or b) directly to an agent telephone or station.

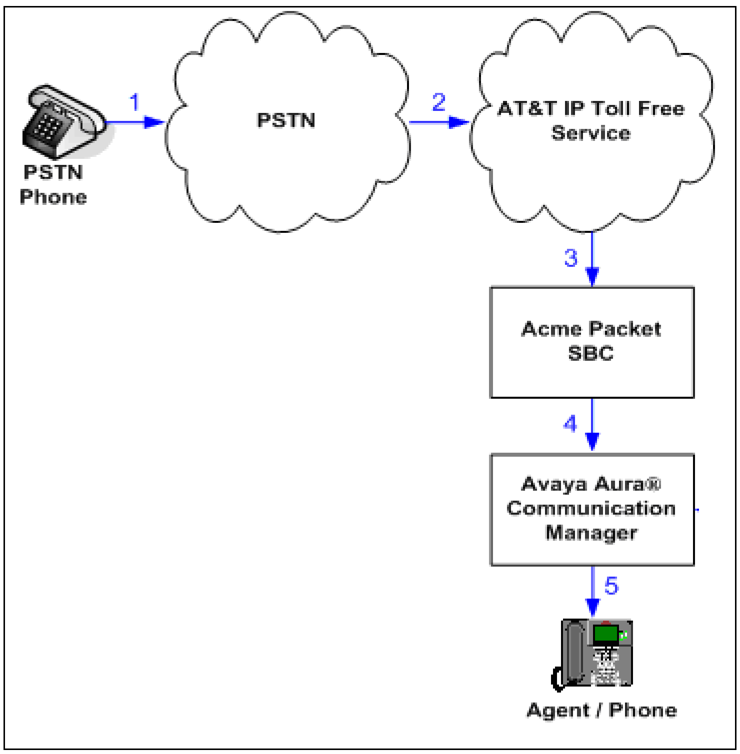

Figure 2: Inbound AT&T IP Toll Free Service Call to VDN / Agent / Phone

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. The Modular Messaging system is in MultiSite mode.

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.

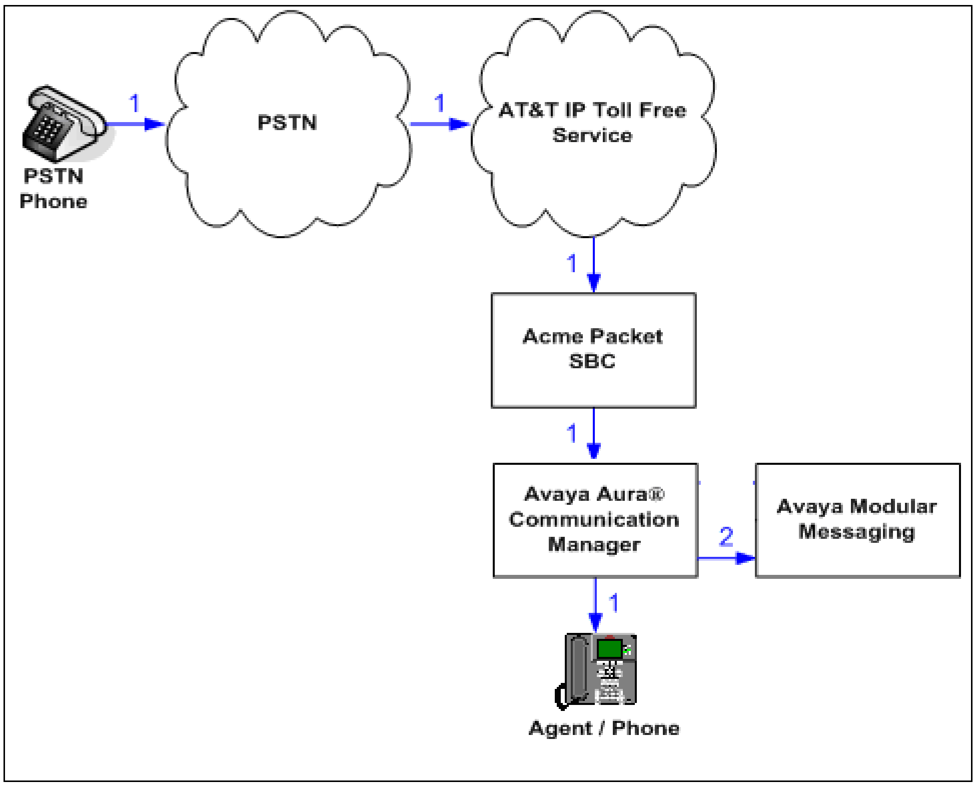

Figure 3: Inbound AT&T IP Toll Free Service Call to Agent / Phone Covered to Avaya Modular Messaging

4. Equipment and Software Validated

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

Component	Version
Avaya S8720 Server	Avaya Aura® Communication Manager 5.2.1 (R015x.02.1.016.4) with SP6 18576
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW15 FW053
TN799DP Control-LAN (C-LAN)	HW01 FW039
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW058
TN2501AP VAL-ANNOUNCEMENT	HW03 FW021
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW010
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release 3.110b
Avaya one-X® Agent	2.0.018.8
Avaya 6416D+ Digital Telephone	-
Avaya S3500 Servers for Avaya Modular Messaging (MAS and MSS)	Release 5.2 – SP5 with Patch 1 (9.0.350.5019)
Fax device	Ventafax Home Version 6.3.102
Acme Packet Net-Net 3800	SCX6.2.0 MR3 Patch 6 (Build 707)
AT&T IP Toll Free Service using MIS-PNT transport service connections.	VNI 18

Table 2: Equipment and Software Versions

5. Avaya Aura® Communication Manager 5.2.1

In the reference configuration Communication Manager 5.2.1 is provisioned in an Access Element configuration, supporting H.323 and Digital endpoints (SIP endpoints are not supported in this 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, C-LAN, Media Processor, 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 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. 5000).

```
display system-parameters customer-options                Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                        USED
                     Maximum Administered H.323 Trunks: 8000  0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 5000  250
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   2
   Maximum Number of Expanded Meet-me Conference Ports: 0     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**".

```
display 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? y
                  Enhanced EC500? y     ISDN/SIP Network Call Redirection? n
    Enterprise Survivable Server? n                         ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                ISDN-PRI? y
              ESS Administration? n              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                    Malicious Call Trace? n
     External Device Alarm Admin? n                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? n                Multifrequency Signaling? y
      Global Call Classification? n        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n              Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

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 "**26**" – local extensions for Communication Manager stations, agents, and Vector Directory Numbers (VDNs) in this reference configuration conform to this format.
- 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.

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all         Percent Full:    2
       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     26           5    ext
     8            1    fac
     9            1    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).

Inter-region communication between Local and AT&T regions 1 and 2 are 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 elements (e.g. C-LAN and Media Processor boards) 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 **customera.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  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: customera.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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

Figure 7: IP-Network-Region Form for the Communication Manager elements – Page 1

2. On page 3 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** column "**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   3 of  19
 Source Region: 1     Inter Network Region Connection Management      I        M
                                                                      G  A     e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G     a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L     s
 1    1                                                                all
 2    2     y    NoLimit                                              n
 3
```

Figure 8: IP-Network-Region Form for the Communication Manager elements – Page 3

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 **customera.com** in the **Authoritative Domain** field.
   - Enter a descriptive name (e.g. **AT&T_IPTF**).
   - 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  19
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: customera.com
    Name: AT&T IPTF
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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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 Form for the AT&T IP Toll Free Service – Page 1

2. On Page 3 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   3 of  19

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

Figure 10: IP-Network-Region Form for the AT&T IP Toll Free Service– Page 3

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**.

```
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

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 (**Acme3800**).
    b. Add a node name and the IP address for Modular Messaging (**MM**).
2. Note the node name and IP address of a C-LAN board (**MainCLAN1a03**), and the Media Processor board (**MainMP1A04**) that were provisioned during installation.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
Acme3800            192.168.67.130
Gateway001          192.168.67.1
MainCLAN1A03        192.168.67.14
MainMP1A04          192.168.67.15
MainVAL1A06         192.168.67.17
MM                  192.168.67.141
default             0.0.0.0
procr               0.0.0.0
```

Figure 15: Change Node-Names IP Form

5.6. IP Interfaces

1. Enter the **list ip-interface all** command and verify that the C-LAN and Media Processor were assigned to region 1 during installation

```
list ip-interface all
                                IP INTERFACES
                                                                    Net
ON Type   Slot  Code/Sfx      Node Name/       Mask  Gateway Node      Rgn  VLAN
                              IP-Address
-- ------ ----- --------      ---------------  ----  ---------------   ---  ----
 y C-LAN  01A03 TN799  D      MainCLAN1A03     /24   Gateway001        1    n
                              192.168.67.14
 y MEDPRO 01A04 TN2602        MainMP1A04       /24   Gateway001        1    n
                              192.168.67.15
```

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

5.7.1. Inbound AT&T Traffic

Communication Manager looks at the contents of the PAI for admission control to the Signaling Groups via the *Far-End Domain* field..

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**". **Note** – In the reference configuration TCP was used to simplify protocol tracing, however TLS/port 5061 is the Avaya best practices recommendation. 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.
   - **Near-end Node Name** – Set to the node name of the C-LAN board noted in **Section 5.5** (e.g. **MainCLAN1A03**）.
   - **Far-end Node Name** – Set to the node name of the Acme Packet SBC as administered in **Section 5.5** (e.g. **Acme3800**).
   - **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 **customera.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
                        Transport Method: tcp
  IMS Enabled? n
 Near-end Node Name: MainCLAN1A03         Far-end Node Name: Acme3800
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 2
Far-end Domain: customera.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                       Direct IP-IP Early 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.
   - **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:
                                     Auth Code? n
Service Type: public-ntwrk                              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
```

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
```

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
```

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 following:
   - **Group Type** – Set to "**sip**".
   - **Transport Method** – Set to "**tcp**".
   - **Near-end Node Name** – Set to the node name of the C-LAN board noted in **Section 5.5** (e.g. **MainCLAN1A03**).
   - **Far-end Node Name** – Set to the node name of Modular Messaging as administered in **Section 5.5** (e.g. **MM**).
   - **Near-end Listen Port** and **Far-end Listen Port** – set to "**5060**"
   - **Far-end Network Region** – Set to the IP network region to **1**, as defined in **Section 5.3.1**.
   - **Far-end Domain** – Set to **customera.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 "**n**".

```
add signaling-group 3
                                SIGNALING GROUP
 Group Number: 3            Group Type: sip
                       Transport Method: tcp
  IMS Enabled? n
   Near-end Node Name: MainCLAN1A03          Far-end Node Name: MM
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
Far-end Domain: customera.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? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                       Direct IP-IP Early 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**). On Page 1 of the **trunk-group** form, provision the following:
   - **Group Type** – Set to "**sip**".
   - **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**.
   - **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                     Auth Code? n          Night Service:
Service Type: tie                                       Signaling Group: 3
                                                      Number of Members: 20
```

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

3. For pages 2, and 3 of the form, use the same values as shown in **Section 5.7.1**.
4. On Page 4 of the form:
   a. Set **Support Request History?** to **y**.
   b. Set **Telephone Event Payload Type:** to **100**.
5. Leave the remaining fields at their default values.

```
add trunk-group 3                                               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? y
            Telephone Event Payload Type: 100
```

Figure 24: Trunk-Group 3 Form (to/from Modular Messaging) – Page 4

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.

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.

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. VDN **26112** 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. **7323204301**).
   - **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.
3. In the **public-unknown-numbering** form, enter sufficient matching digits for the local extension dial plan (e.g. **26xxx**). This will be used to populate the History Info headers for coverage calls to Modular Messaging.
   - **Ext Len** – Enter the total number of digits in the local extension range (e.g. **5**).
   - **Ext Code –** Enter sufficient digits to match the local extension dial plan (e.g. **26**).
   - **Trk Grp(s)** – Enter the number of the trunk group defined in **Section 6.7.2** (e.g. **3**).
   - **CPN Prefix** – Leave this field blank (digits remain unchanged)
   - **CPN Len** – Enter the total number of digits in the local extension range (e.g. **5**).

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                             Total
Ext Ext            Trk       CPN              CPN      Total Administered: 2
Len Code           Grp(s)    Prefix           Len      Maximum Entries: 9999
 5  26             3                          5
 5  26112          2        7323204301        10
```

Figure 25: Public-Unknown-Numbering Form

5.9. 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 26298/vector 8 – Auto-attendant.
- VDN 26299/vector 5 – Meet-me Conference
- VDN 26112/vector 1002 – 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.

5.9.1. 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. **26000** in **Figure 27**). The hunt group is associated with a coverage path (e.g.**H1** in **Figure 28**) and the coverage path is assigned to a station (e.g. **26102** in **Figure 29**).

```
display hunt-group 1                                           Page   1 of  60
                                  HUNT GROUP
            Group Number: 1                                    ACD? n
              Group Name: MM                                 Queue? n
         Group Extension: 26000                             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 26: 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)
     26000                    26000                     8
```

Figure 27: 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 28: Coverage Path 1 Form

```
display station 26102                                         Page   1 of   5
                                     STATION
Extension: 26102                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: 123456          TN: 1
     Port: S00000                      Coverage Path 1: 1              COR: 1
     Name: H323-9630                   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: 26102
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n
                                             Customizable Labels? y
```

Figure 29: Station 26102 Form

5.9.2. 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. **26298**) and a vector (e.g. **8**).

```
display vdn 26298                                           Page   1 of   2
                            VECTOR DIRECTORY NUMBER
                             Extension: 26298
                                 Name*: auto attend
                           Destination: Vector Number        8
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
```

Figure 30: Auto Attendant VDN

```
display vector 8                                            Page   1 of   6
                                  CALL VECTOR
    Number: 8                Name: auto attend
                                              Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    4   secs hearing ringback
02 collect      5    digits after announcement 26504
03 route-to     digits with coverage n
04 wait-time    5   secs hearing silence
05 stop
```

Figure 31: Auto Attendant Vector

5.9.3. Meet-me Conference

A basic meet-me conference functionality is defined in the reference configuration for DTMF testing. The meet-me conference functionality is defined by a VDN (e.g. **26299**) and a vector (e.g. **5**).

```
display vdn 26299                                           Page   1 of   2
                            VECTOR DIRECTORY NUMBER

                             Extension: 26299
                                  Name: meet-me vdn 1
                           Destination: Vector Number        5

                  Meet-me Conferencing? y
                                   COR: 1
                                    TN: 1
```

Figure 32: Meet-me Conference VDN – Page 1

```
display vdn 26299                                               Page   2 of   2
                                VECTOR DIRECTORY NUMBER

                  MEET-ME CONFERENCE PARAMETERS:

                    Conference Access Code: 123456
                     Conference Controller: 26201
                           Conference Type: 6-party
```

Figure 33: Meet-me Conference VDN – Page 2

```
display vector 5                                                Page   1 of   6
                                      CALL VECTOR

    Number: 5                Name: meet-me vec
                                              Meet-me Conf? y           Lock? y
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? n   CINFO? n   BSR? n   Holidays? n
 Variables? n   3.0 Enhanced? n
01 wait-time    5   secs hearing ringback
02 collect      6    digits after announcement 26501
03 goto step    5             if digits           =      meet-me-access
04 goto step    2             if unconditionally
05 announcement 26503
06 route-to     meetme
07 stop
08
```

Figure 34: Meet-me Conference Vector

5.9.4. Skills

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

```
change hunt-group 2                                             Page   1 of   3
                                      HUNT GROUP
            Group Number: 2                                    ACD? y
              Group Name: Skill2                             Queue? y
         Group Extension: 26002                             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:
```

Figure 35: Define skill hunt group

```
change vdn 26112                                                Page   1 of   3
                             VECTOR DIRECTORY NUMBER
                              Extension: 26112
                                  Name*: Skill2
                            Destination: Vector Number        1002
                   Meet-me Conferencing? n
                     Allow VDN Override? n
                                    COR: 1
                                    TN*: 1
                               Measured: none
                             1st Skill*:
                             2nd Skill*:
                             3rd Skill*:
* Follows VDN Override Rules
```

Figure 36: Define skill VDN

```
change vector 1002                                              Page   1 of   6
                                   CALL VECTOR
    Number: 1002             Name: Skill2
                                                Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? n
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 26012
03 queue-to     skill 2    pri m
04 wait-time    10  secs hearing music
05 announcement 26015
06 goto step    3             if unconditionally
07 stop
```

Figure 37: 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 – 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.

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.

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 4500platforms.

```
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 C-LAN board used in the reference configuration(see **Section 5.5**).

```
session-agent
      hostname                      192.168.67.14
      ip-address                    192.168.67.14
      port                          5060
      state                         disabled
      app-protocol                  SIP
      app-type
      transport-method              StaticTCP
      realm-id                      INSIDE
      egress-realm-id
      description                   521_clan
      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
                              192.168.67.14
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 *customera.com* instead of the Acme outside address (192.168.64.130), and the From headers 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 *customera.com*. The inbound Request URI is modified from the Communication Manager C-LAN IP address to *customera.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                         customera.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                      192.168.67.14
              new-value                        customera.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                        customera.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 disconnects 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 5.2.1

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 **xx** 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
13:53:42 SIP<INVITE sip:0000091049@customera.com:5060;transport=
13:53:42 SIP<tcp  SIP/2.0
13:53:42     active trunk-group 20 member 1  cid 0x19f
13:53:42    0  0 ENTERING TRACE cid 415
13:53:42 1002  1 vdn e26112 bsr appl   0 strategy 1st-found override n
13:53:42 1002  1 wait 2 secs hearing ringback
13:53:42 SIP>SIP/2.0 180 Ringing
13:53:42     dial 26112
13:53:42     ring vector 1002    cid 0x19f
13:53:42     G729 ss:off ps:30
             rgn:2 [192.168.67.130]:18248
             rgn:1 [192.168.67.15]:30668
13:53:42     xoip options: fax:T38 modem:off tty:US  uid:0x50010
             xoip ip: [192.168.67.15]:30668
13:53:44 1002  2 announcement 26012
13:53:48     active station    26102 cid 0x19f
13:53:48 SIP>INVITE sip:7326712438@192.168.67.130:5060;transport
13:53:48 SIP>=tcp SIP/2.0
13:53:48 SIP<SIP/2.0 200 OK
13:53:48 SIP>ACK sip:7326712438@192.168.67.130:5060;transport=tc
13:53:48 SIP>p SIP/2.0
13:53:48     G729A ss:off ps:30
             rgn:2 [192.168.67.130]:18248
             rgn:1 [192.168.67.80]:18862
```

Figure 38: 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.

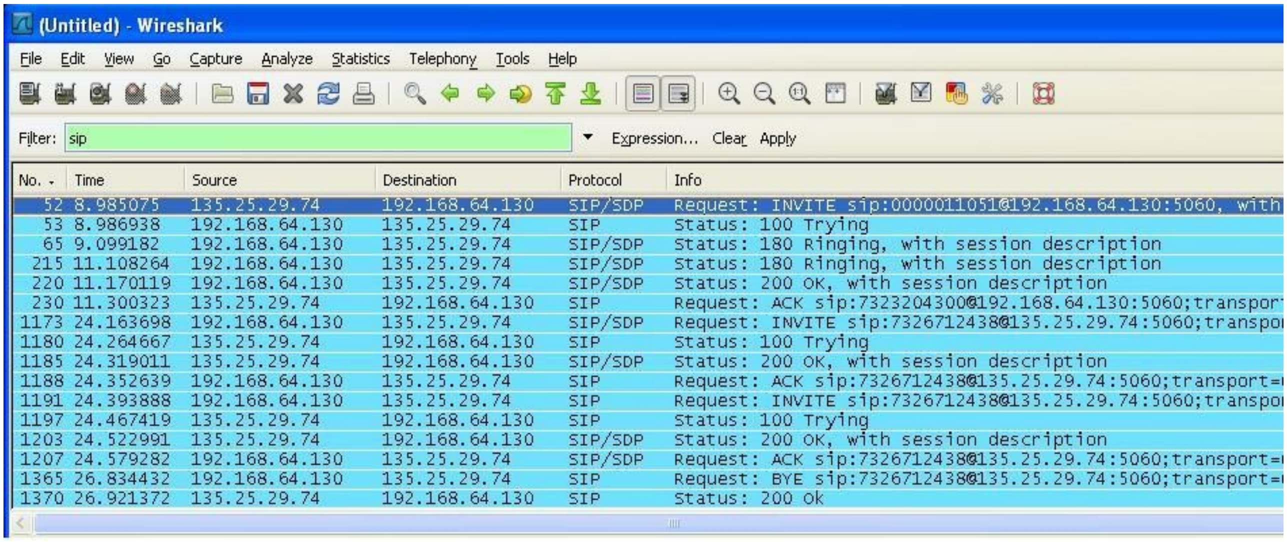

Figure 39: –SIP Protocol trace – Inbound call from AT&T

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

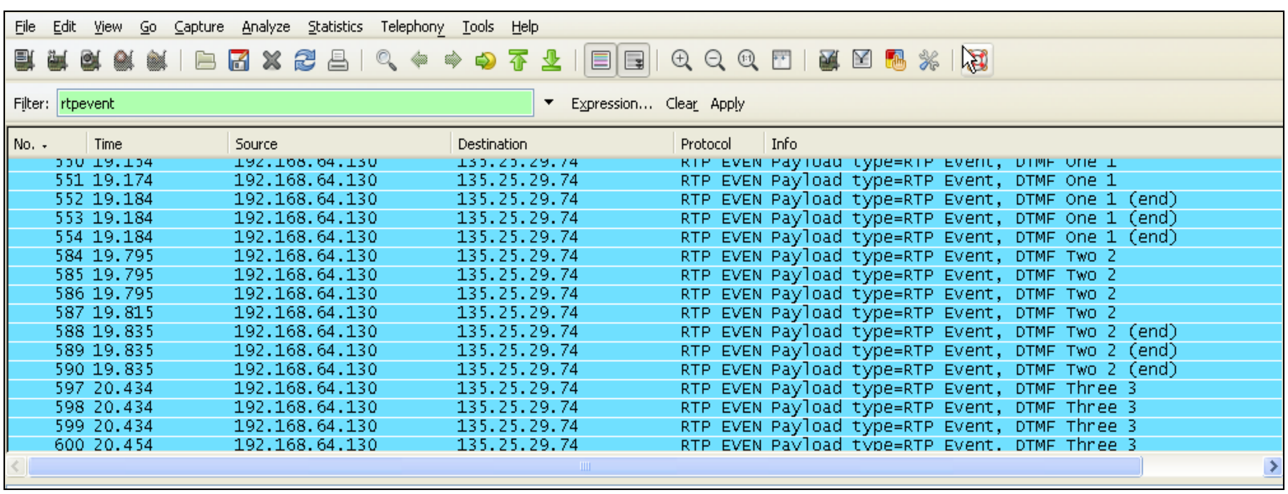

Figure 40: – RTPEvent (DTMF) trace – Outbound DTMF events to AT&T

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

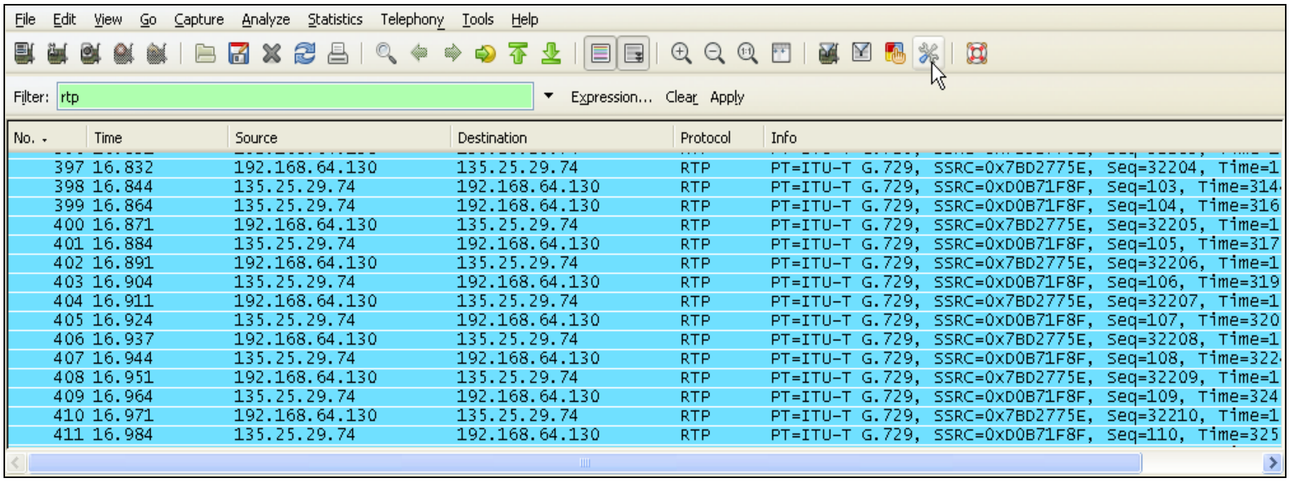

Figure 41: – RTP trace (showing codec used) – inbound call to AT&T

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 5.2.1 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] *Administering Avaya Aura® Communication Manager*, Issue 5.0, Release 5.2, May 2009, Document Number 03-300509

[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Issue 7, Release 5.2, May 2009, Document Number 555-245-205

[3] *Avaya Aura® Call Center 5.2 Call Vectoring and Expert Agent Selection (EAS) Reference*, Release 5.2, April 2009, Document Number 07-600780

[4] *Avaya Aura® Call Center 5.2 Automatic Call Distribution Reference*, Release 5.2, April 2009, Document Number 07-602568

[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 Flexible Reach service border elements are also specified in the **session-group** section below.

```
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 Flexible Reach 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.