Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3 and Acme Packet Net-Net 6.2.0 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3, and Acme Packet Net-Net 3800 with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager R6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 6.2 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.3 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 SIP 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® Session Manager and 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 at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.3, and Acme Packet Net-Net 3800[1]with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

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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® Session Manager and 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:

1. A simulated enterprise with System Manager, Session Manager, Communication Manager, Avaya phones, fax machines (Ventafax application), Acme Packet Session Border Controller (SBC), and Avaya Aura® Communication Manager Messaging (CM Messaging).
2. A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via **AVPN** or **MIS-PNT** transport.

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

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for sample call flows) between Session Manager, Communication Manager, Acme Packet SBC and the AT&T IP Toll Free service.

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

- SIP trunking
- G729 and G711 codecs
- AT&T IP Toll Free calls to Communication Manager stations/agents using Vector Directory Numbers (VDNs), and vectors
- Two-way talk path establishment between PSTN and Communication Manager VDNs/vectors and stations/agents, via the AT&T Toll Free service
- Passing of RFC2833 DTMF events and their recognition by navigating automated voice menus
- T.38 fax calls between 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 Communication Manager that are directly routed to stations, and alternatively can be covered to CM Messaging
- Legacy Transfer Connect
- Alternate Destination Routing
- IP Re-routing
- Long Duration Calls

2.2. Test Results and Known Limitations

1. When the call is put on hold on Communication Manager, SDP with **a=sendonly** is sent to AT&T IP Flexible Reach service but it sends **a=inactive** in response which results in no Music-on-Hold being sent to PSTN. A Header Manipulation Rule was provided as shown in **Section 7** to send **a=sendrecv** to resolve this situation.
2. G.726 codec is not supported between AT&T IP Toll Free service and Communication Manager.
3. **Alternate Destination Routing/Ring No Answer** (ADR/RNA) only works if the call is destined for an agent. When AT&T IP Toll Free service receives a **181 Call Being Forwarded** message, it is able to process ADR/RNA properly which happens when the call is destined for an agent logged in at CPE. When call is destined for an endpoint on CPE, AT&T IP Toll Free service receives a **180 Ringing** and is unable to process ADR/RNA properly.
4. If Communication Manager receives an SDP offer with multiple codecs, where at least two of the codecs are supported in the codec set provisioned on Communication Manager, then Communication Manager selects a codec according to the priority order specified in the Communication Manager codec set, not the priority order specified in the SDP offer. For example, if the AT&T IP Toll Free service offers G.711, G.729A, and G.729B in that order, but the Communication Manager codec set contains G.729B, G729A, and G.711 in that order, then Communication Manager selects G.729B, not G.711. The practical resolution is to

provision the Communication Manager codec set to match the expected codec priority order in AT&T IP Toll Free SDP offers.
5. G.711 faxing is not supported between Communication Manager and the AT&T IP Toll Free service. Communication Manager does not support the protocol negotiation that AT&T requires for G.711 fax calls. 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 Communication Manager.

The test objectives stated in **Section 2.1** with limitations noted in this section were verified.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (888) 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:

- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager allows enterprises to implement centralized and policy-based routing, centralized yet flexible dial plans, consolidated trunking, and centralized access to adjuncts and applications.
- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Communication Manager provides the voice communication services for a particular enterprise site. In the reference configuration, Communication Manager 6.2 runs on an Avaya S8800 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya Servers.
- The Avaya Media Gateway provides the physical interfaces and resources for Communication Manager. In the reference configuration, an Avaya G650 Media Gateway is used. The G650 contains system boards such as the Control LAN (C-LAN) and Media Processor (MedPro). This solution is extensible to other Avaya Media Gateways.
- Avaya "desk" telephones are represented with Avaya 96x0 and 96x1 Series IP Telephones running H.323 and SIP, Avaya 6408D Series Digital Telephone, Avaya Analog phone and Avaya one-X® Communicator (SIP/H323), a PC based softphone.
- The Acme Packet SBC 3800[2] provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Toll Free service and

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

the enterprise internal network[3]. UDP transport protocol is used between the Acme Packet SBC and the AT&T Toll Free service.

- CM Messaging system provides the corporate voice messaging capabilities in the reference configuration. **The provisioning of CM Messaging is beyond the scope of this document**.
- Inbound calls were placed from PSTN to the Customer Premises Equipment (CPE) via the AT&T IP Toll Free service, through the Acme Packet SBC, Session Manager, and Communication Manager. Communication Manager terminated the calls using appropriate phone or fax stations. The H.323 phones at the CPE are registered to the Communication Manager C-LANs and the SIP phones are registered to Session Manager.

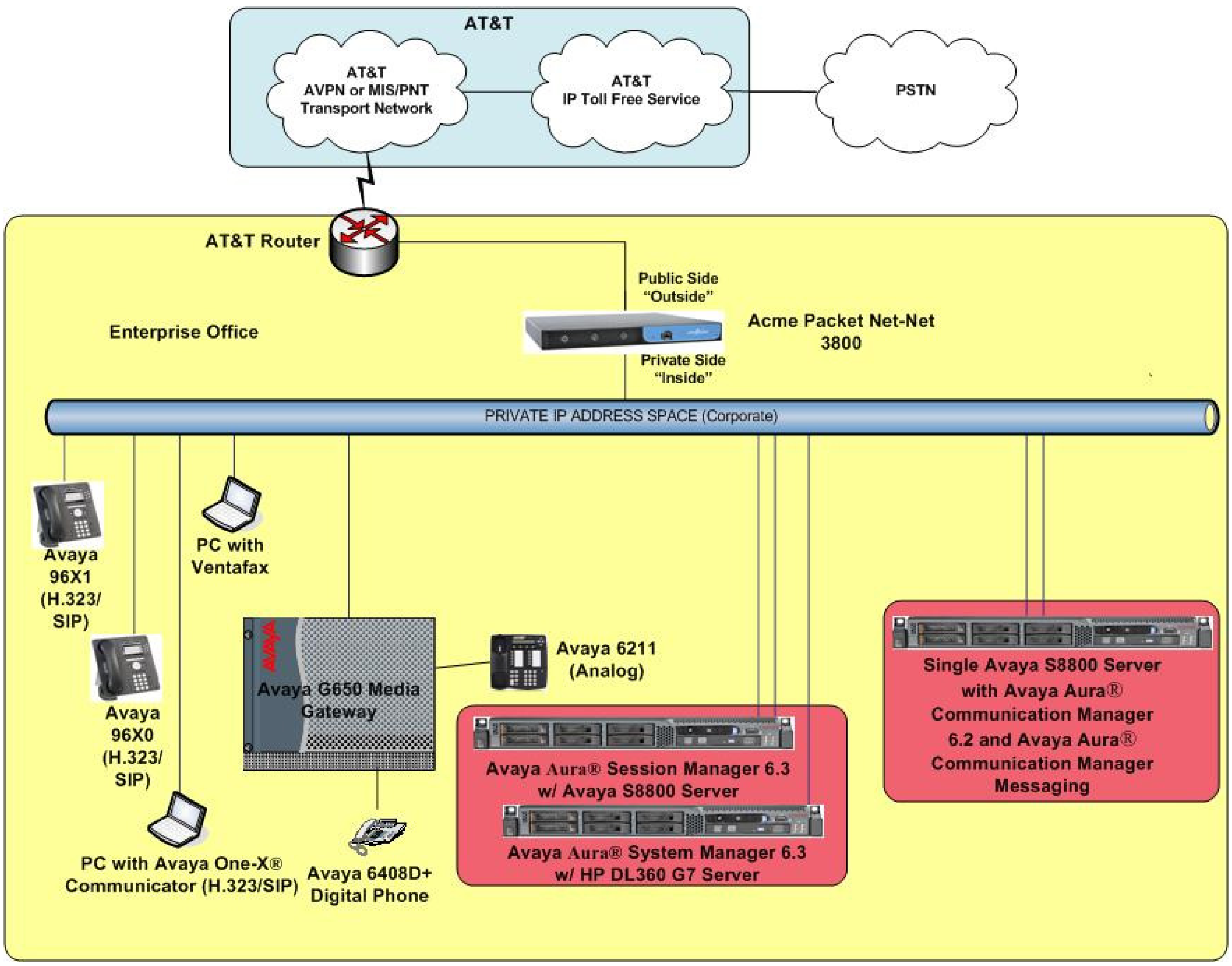


Figure 1: Reference Configuration

[3] The AT&T IP Toll Free service uses SIP over UDP to communicate with enterprise edge SIP devices (Acme Packet SBC) in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

3.1. Illustrative Configuration Information

The specific values used in this reference configuration listed in **Table 1** below are **for illustrative purposes only**. Customers must obtain and use the specific values for their configurations. For security purposes, real IP addresses and DNIS are not included.

Note - The AT&T IP Toll Free service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Toll Free service) are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DNIS digits as part of the IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
Avaya Aura® System Manager	
Management IP Address	10.80.130.120
Avaya Aura® Session Manager	
Management IP Address	10.80.130.121
Network IP Address	10.80.130.122
Avaya Aura® Communication Manager	
Control LAN (C-LAN) IP Address	10.80.130.102
Media Processor (MedPro) IP Address	10.80.130.103
Skill (Hunt Group) Extensions	53xxx
Agent Extensions	53xxx
Phone Extensions	50xxx
Announcement Extensions	33xxx
Vector Directory Numbers (VDN)	20xx
Acme Packet Session Border Controller	
IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Toll Free service)	192.168.62.50
IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager)	10.80.130.250
AT&T IP Toll Free service	
Border Element IP Address	192.242.225.210
DNIS Passed in Request URI used by Session Manager for routing	0000041535[0,1,7,8,9]10[5,6][0,1,7,8,9]

Table 1: Illustrative Values Used in this Reference Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya S8800 Server	Avaya Aura® System Manager 6.3 (6.3.0.8.923) System Platform 6.2.2.06002.0
Avaya S8800 Server	Avaya Aura® Session Manager 6.3 SP2 (6.3.0.0.630039)
Avaya S8800 Server	Avaya Aura® Communication Manager R6.2 SP5 (02.0.823.0-20396) System Platform 6.2.2.08001.0
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW06 FW057
TN799DP Control-LAN (C-LAN)	HW01 FW041
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW063
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	HW05 FW011
Avaya 9650 IP Telephone	H.323 R3.1.5
Avaya 9641G IP Telephone	H.323 R6.2.3.12
Avaya 9608 IP Telephone	SIP R6.2.1
Avaya one-X® Communicator (H323/SIP)	6.1.7.04-SP7-39506
Avaya Digital Telephone 6408D+	
Avaya Analog phone	-
Fax device	Ventafax Home Version 6.1.59.144
Acme Packet Net-Net 3800	SCX6.2.0 MR-6 Patch 5 (Build 916)
AT&T IP Toll Free service using AVPN/MIS-PNT transport service connection	VNI 26

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release 6.3

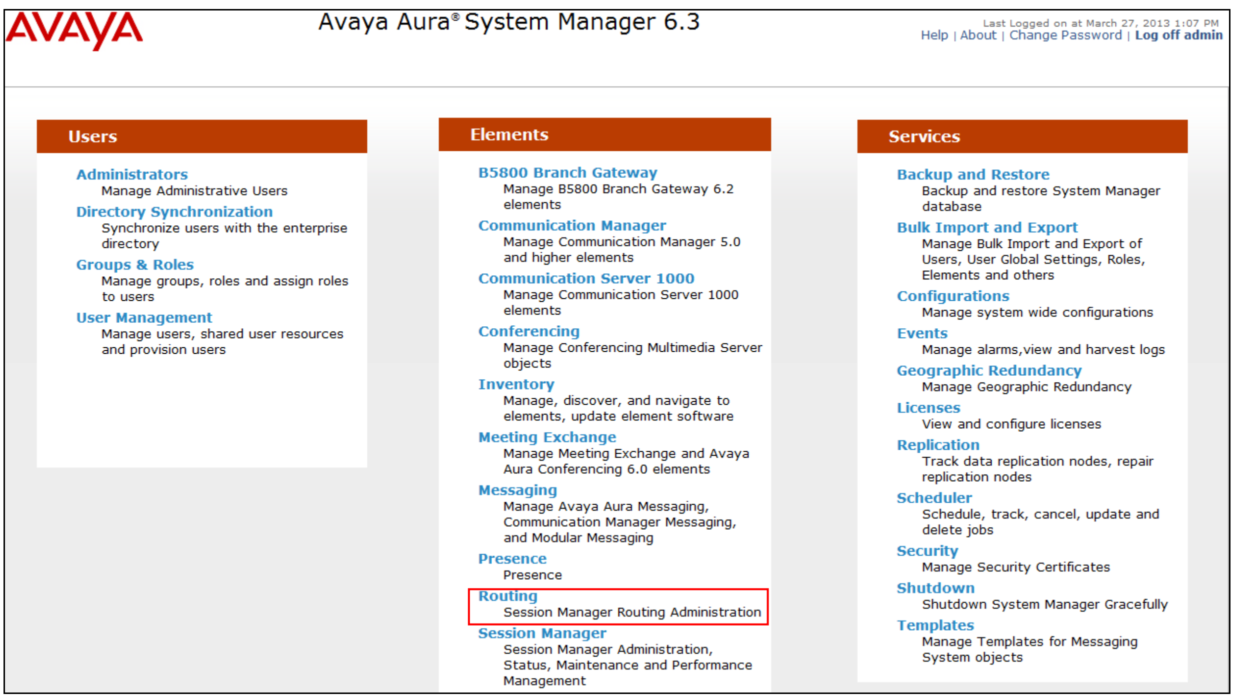
This section illustrates relevant aspects of the Session Manager configuration used in the verification of this compliance test solution for supporting AT&T IP Toll Free service.

Note – These Application Notes assume that basic System Manager and Session Manager administration has already been performed. Refer [**1**] to [**4**] for further details if necessary.

The following administration activities are described:

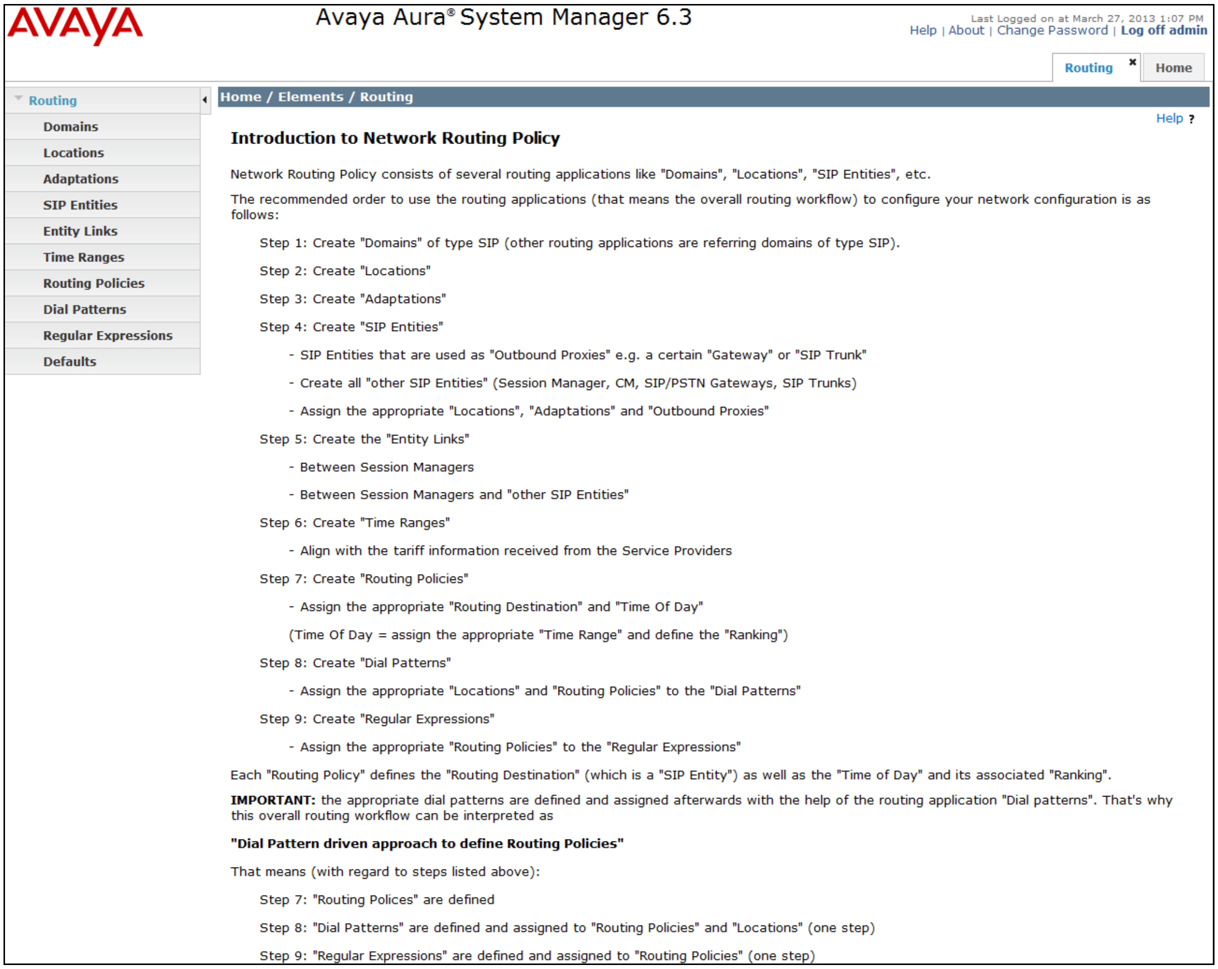
- Define SIP Domain
- Define Locations for routing purpose
- Configure the Adaptation Modules that are associated with various SIP Entities
- Define SIP Entities for Session Manager, Communication Manager, Acme Packet SBC, etc
- Define Entity Links between various SIP entities
- Define Routing Policies associated with Communication Manager, Acme Packet SBC, etc
- Define Dial Patterns which in conjunction with Routing Policies determine to which entity a call is routed to

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**http://<ip-address>**", where **<ip-address>** is the IP address of System Manager and logging in with the appropriate credentials. Once logged in, navigate to **Elements→Routing**.



System Manager Home Page

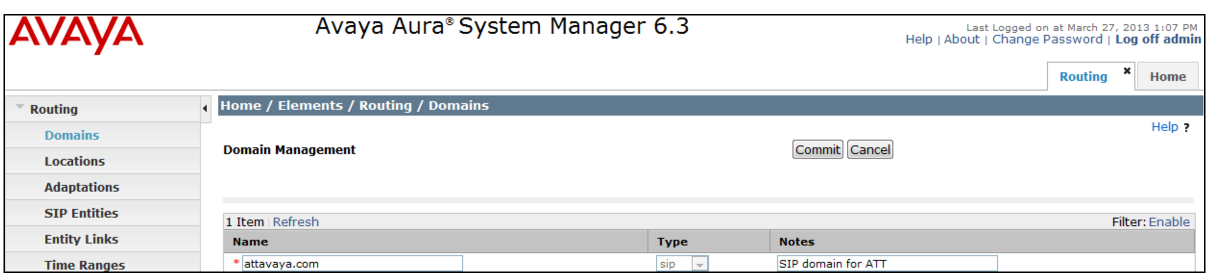
The screen below shows the various sub-headings with explanation of the left navigation menu that are referenced in this section.



Network Routing Policy Page

5.1. SIP Domain

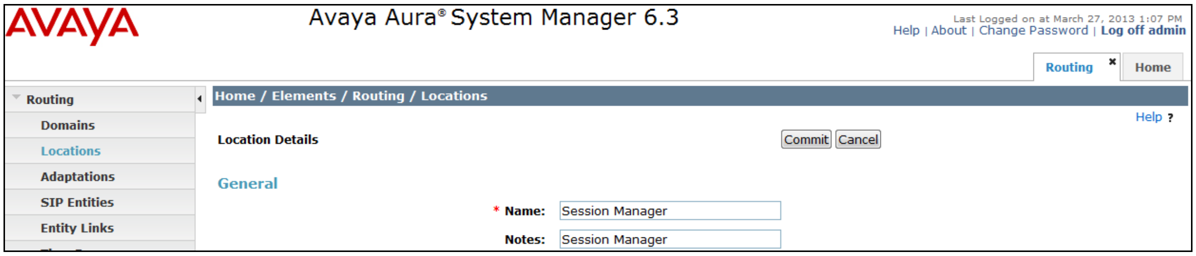
Navigate to **Routing→Domains** and click **New** (not shown). The following screen shows the domain used in this reference configuration.



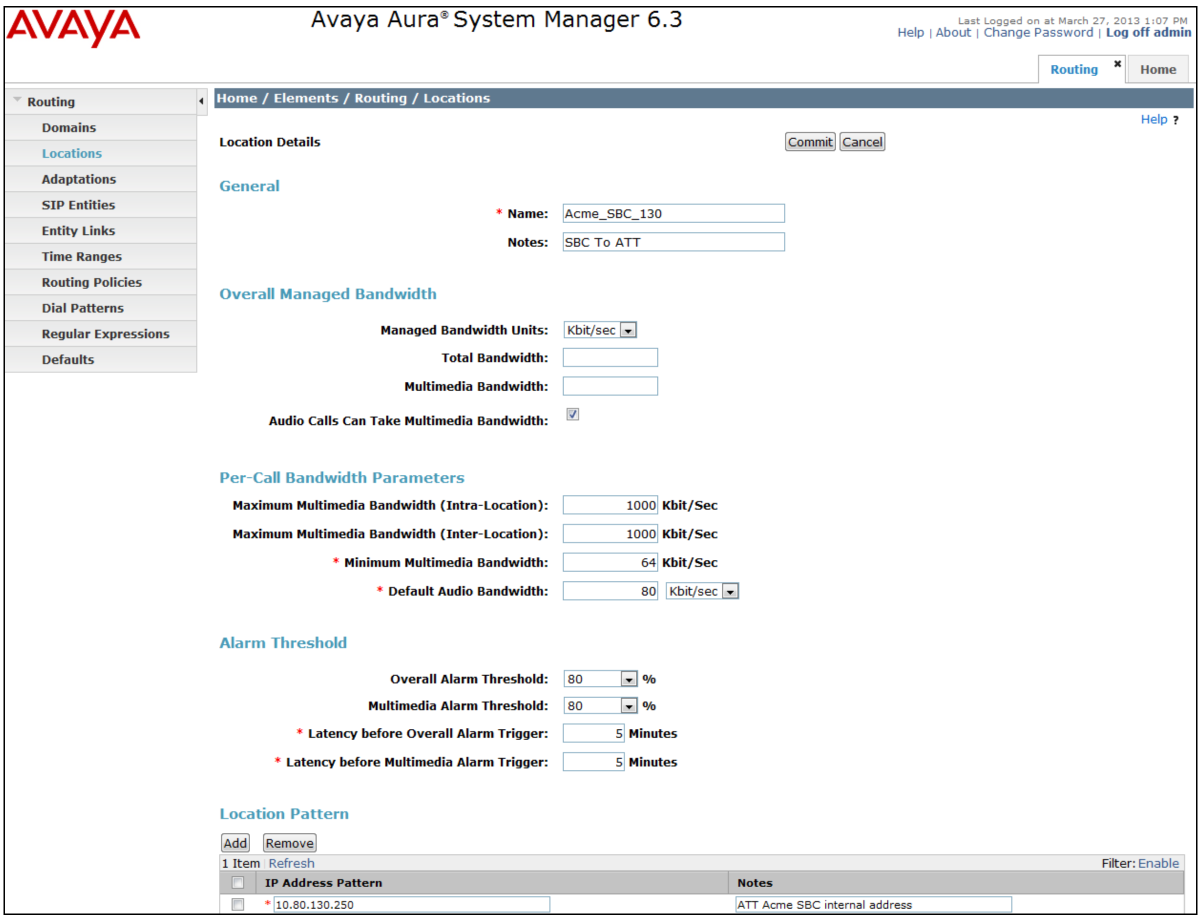
SIP Domains

5.2. Locations

Navigate to **Routing→Locations** and click **New** (not shown). The following screens show Location Details for various locations used in this AT&T IP Toll Free service testing.

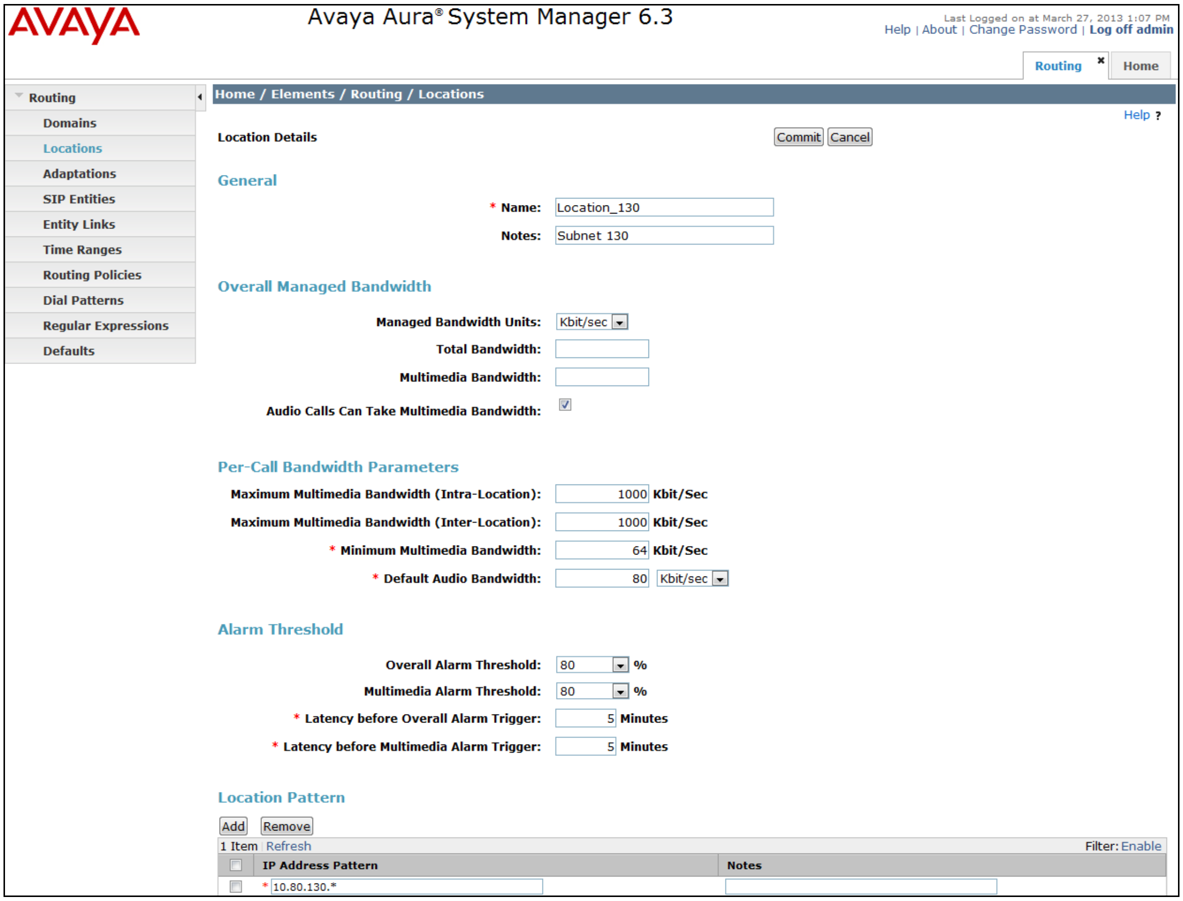


Session Manager Location Details



Acme Packet SBC Location Details

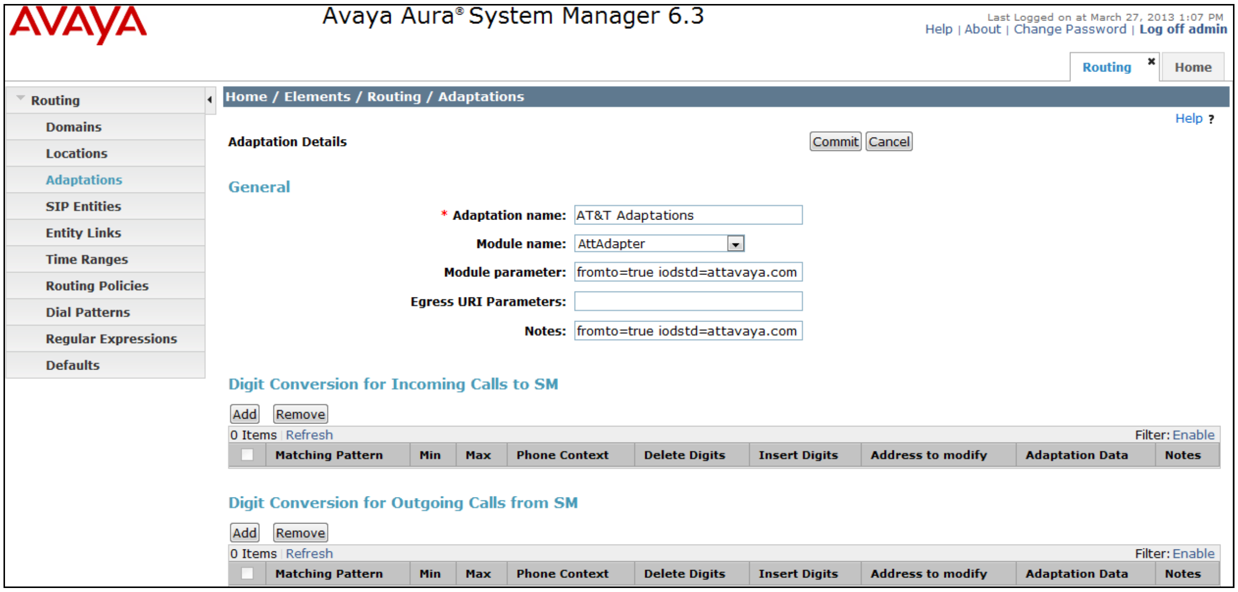
Location_130 shown below serves Communication Manager, CM Messaging and other elements on the CPE.



Subnet 130 Location Details

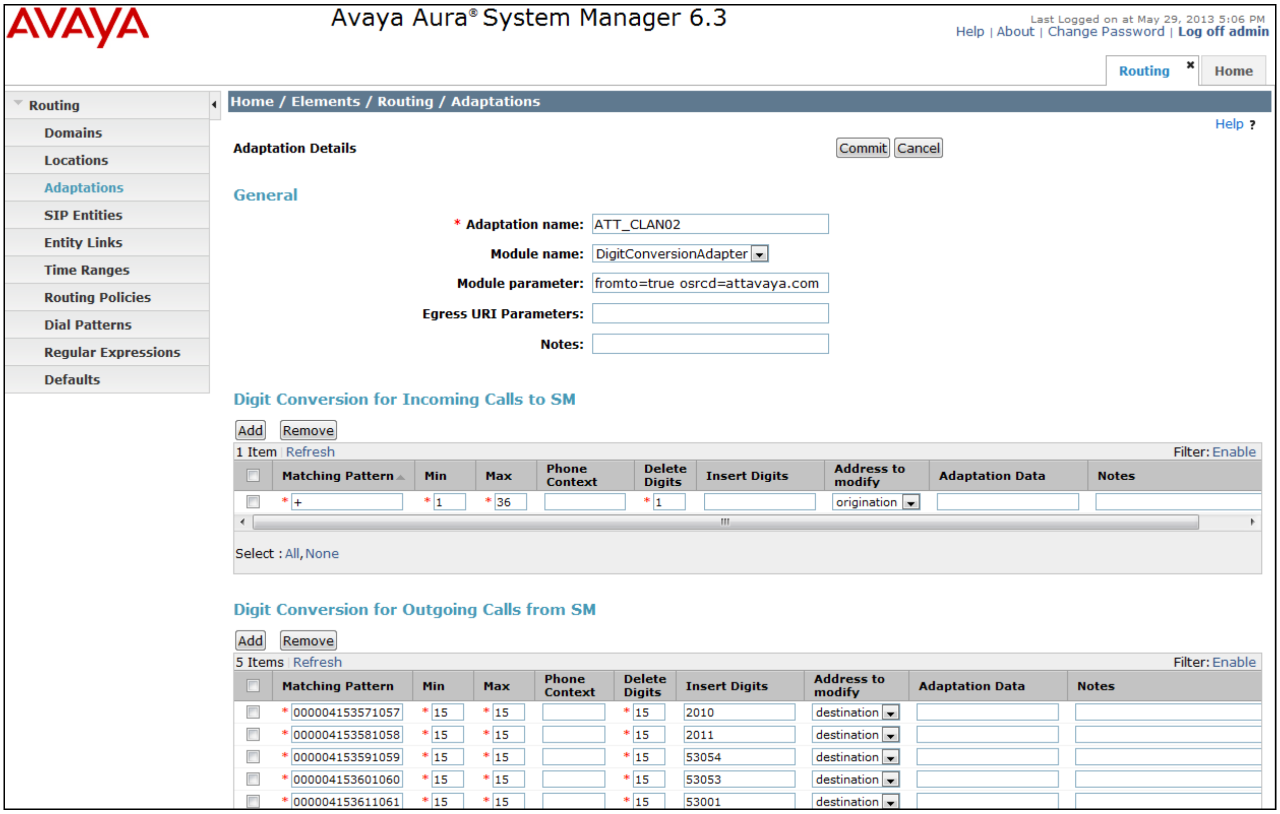
5.3. Configure Adaptations

The following screen displays the adaptation used for calls between AT&T IP Toll Free Service and Session Manager via Acme Packet SBC. The **Module parameter** field is set to **fromto=true iodstd=attavaya.com osrcd=192.168.62.50** (IP Address of the external interface of Acme Packet SBC) **odstd=135.242.225.210** (IP Address of AT&T IP Flexible Reach Border Element). This adaptation is applied to **AcmeSBCATT-5060** entity configured in **Section 5.4**.



Adaptation for Calls between AT&T IP Toll Free service and Session Manager

The following screen displays the adaptation used for calls between Session Manager and Communication Manager. The following screen shows the adaptation used for **CM62_CLAN1A02-5060** configured in **Section 5.4**. In this case, digit conversion is done before and after routing decision has been made based upon the user part of the SIP URI.

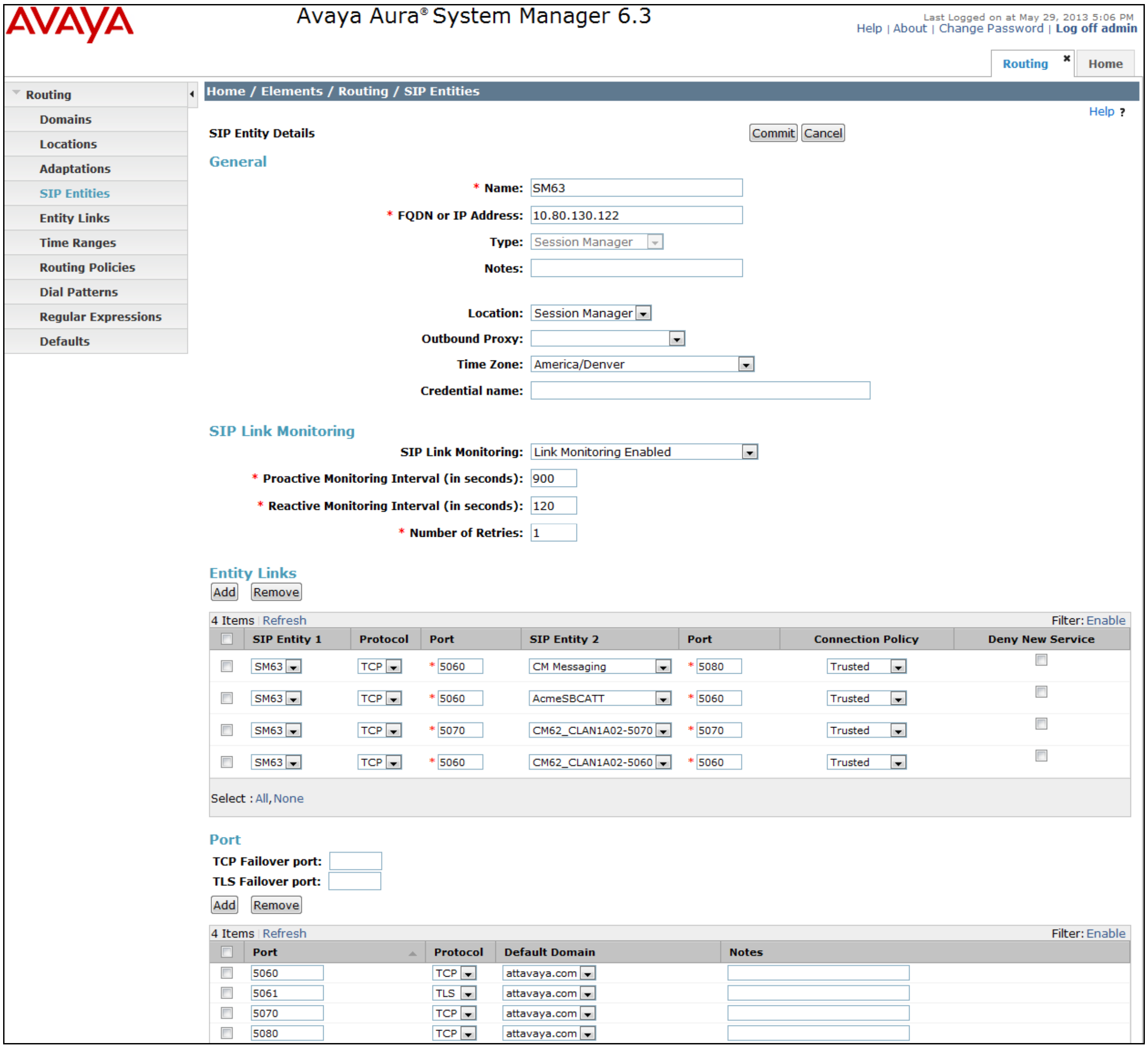


Adaptation for calls between Session Manager and Communication Manager

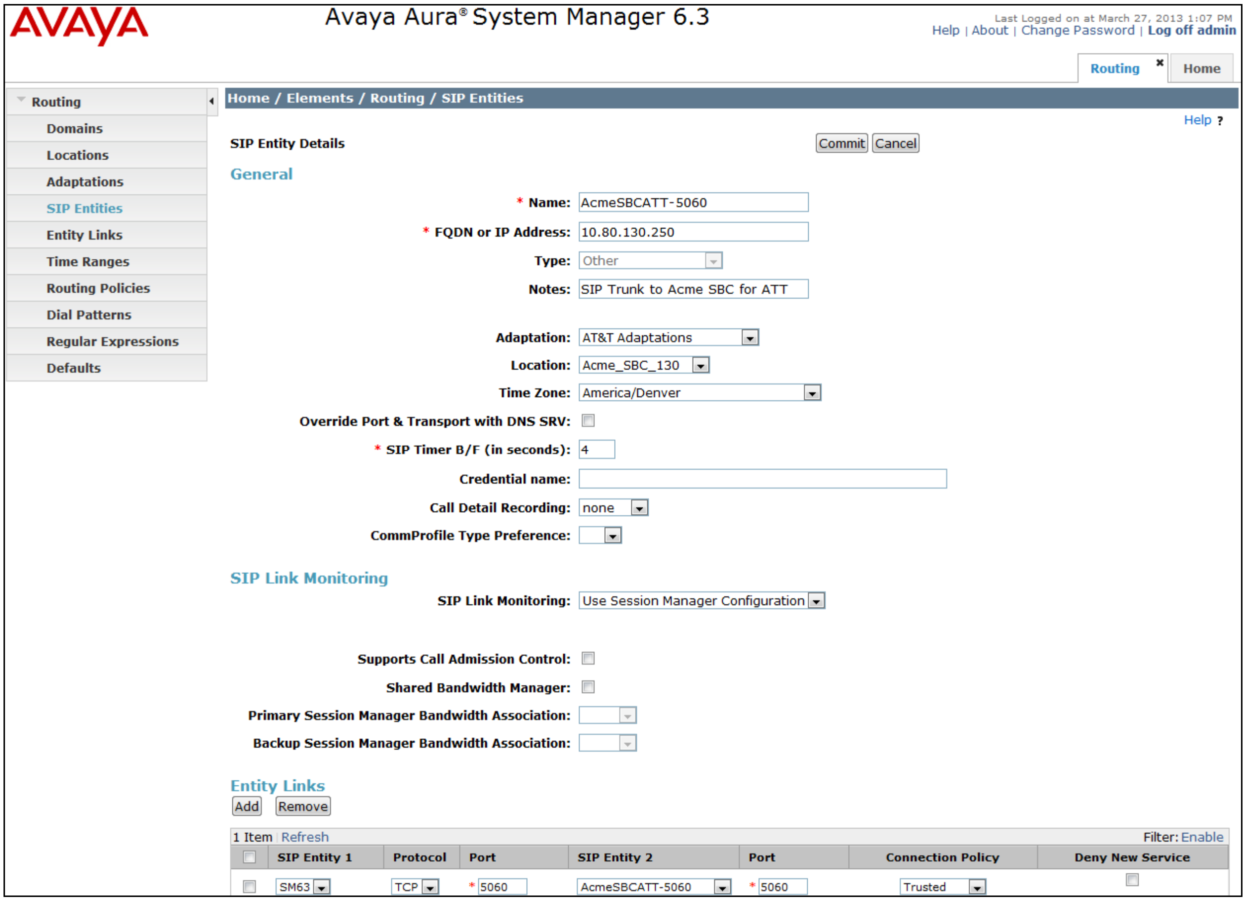
5.4. SIP Entities

The following screens show the entities along with Entity links configured for this reference configuration. See **Section 5.5** for Entity link configuration.

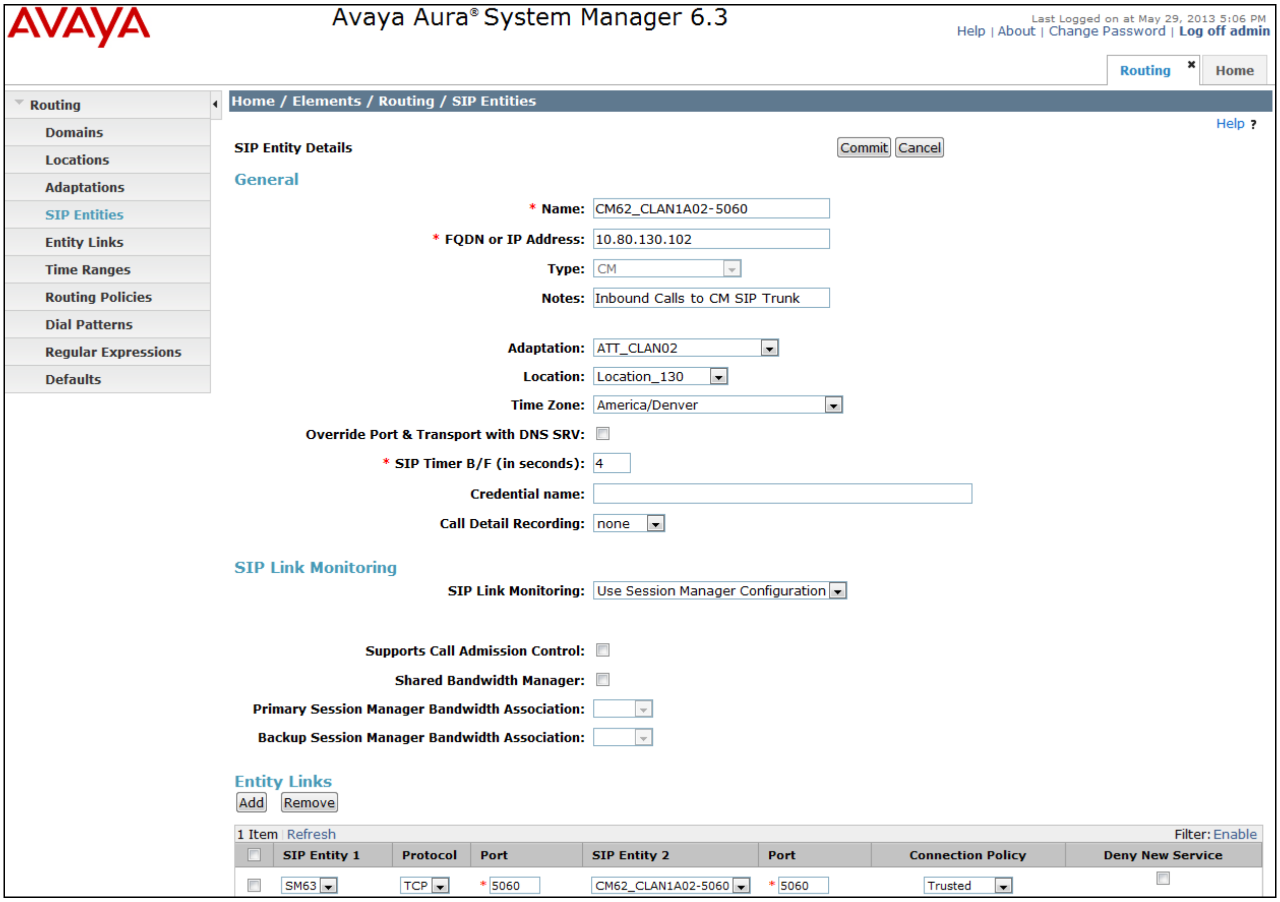
Note – In this reference configuration TCP is used as the transport protocol between Session Manager and all the SIP Entities including Communication Manager. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS to be used as transport protocol whenever possible.



Session Manager Entity

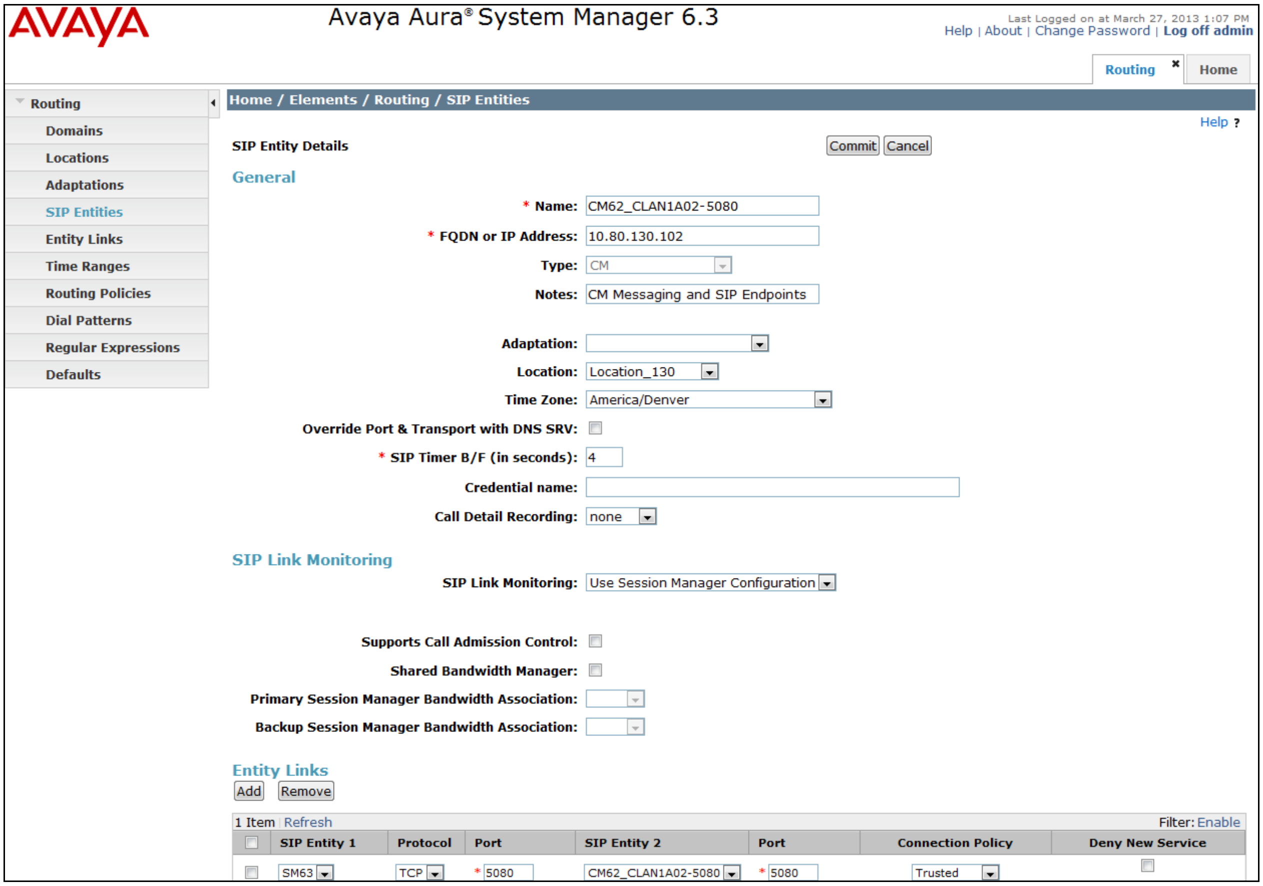


Acme Packet SBC Entity



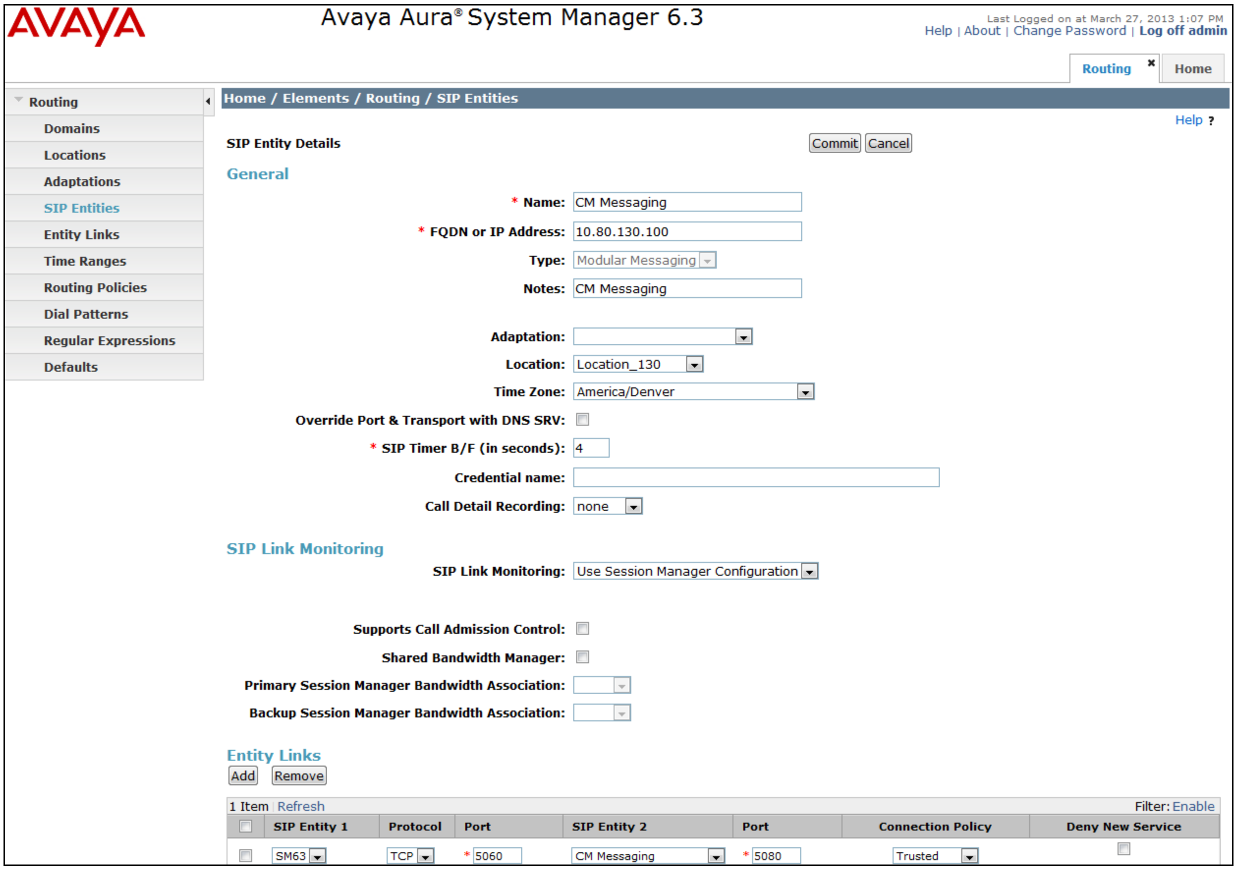
Communication Manager Entity for Inbound Calls (CM6.2CLAN1A02-5060)

The following screen shows SIP Entity configured for the Communication Manager trunk group used for CM Messaging and SIP endpoints. See **Section 6.6.2** for the trunk configuration from Communication Manager to Session Manager to support the CM Messaging and SIP endpoints.



Communication Manager Entity for SIP endpoints and coverage to CM Messaging (CM6.2CLAN1A02-5080)

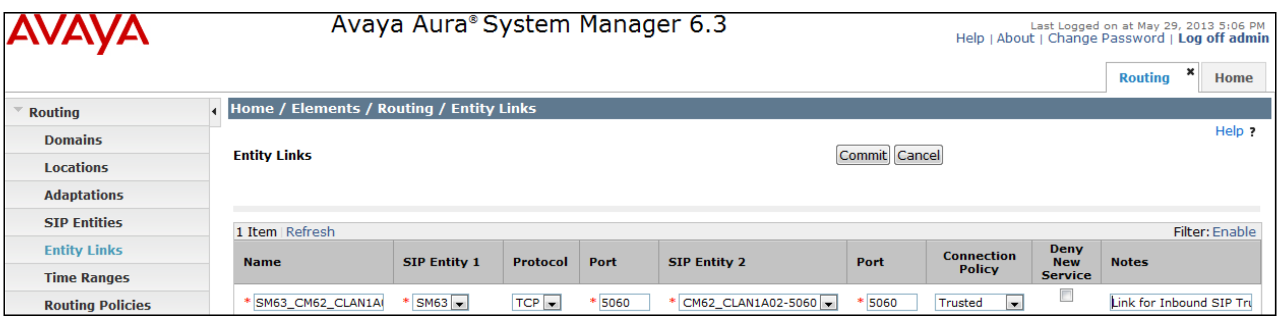
The following screen shows SIP Entity configured for the CM Messaging which is installed and configured on Communication Manager platform. Installation and configuration of CM Messaging is beyond the scope of this document. The **FQDN or IP Address** field is set to the Communication Manager server IP address (**10.80.130.100**).



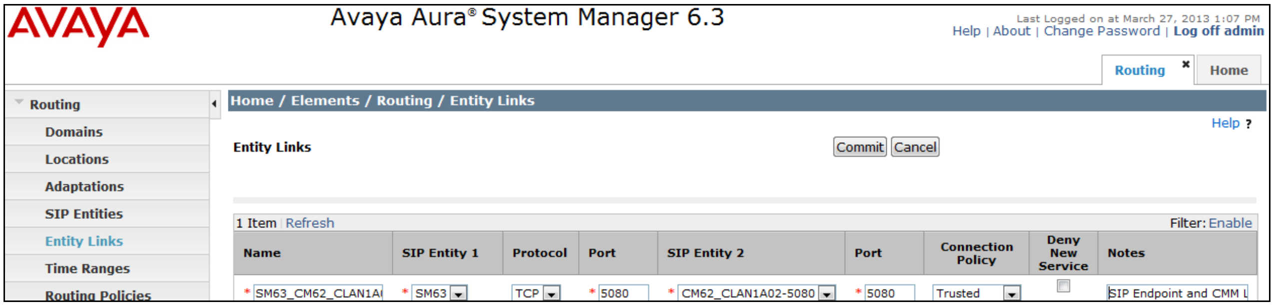
CM Messaging Entity

5.5. Entity Links

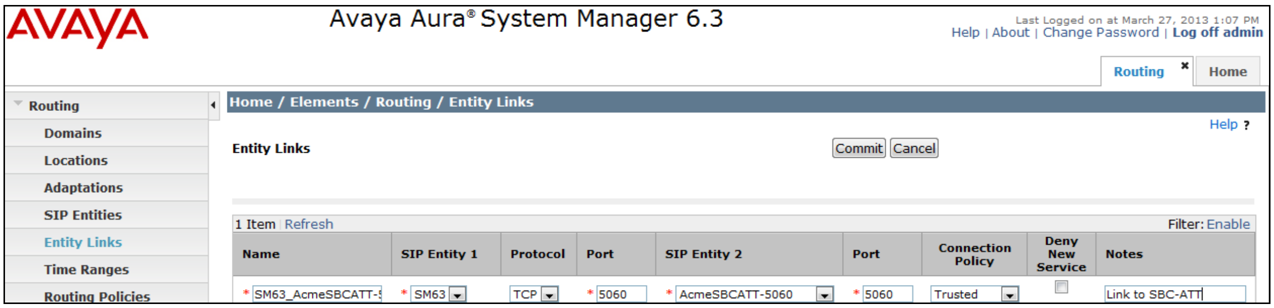
The following screens show the entity links configured for this reference configuration.



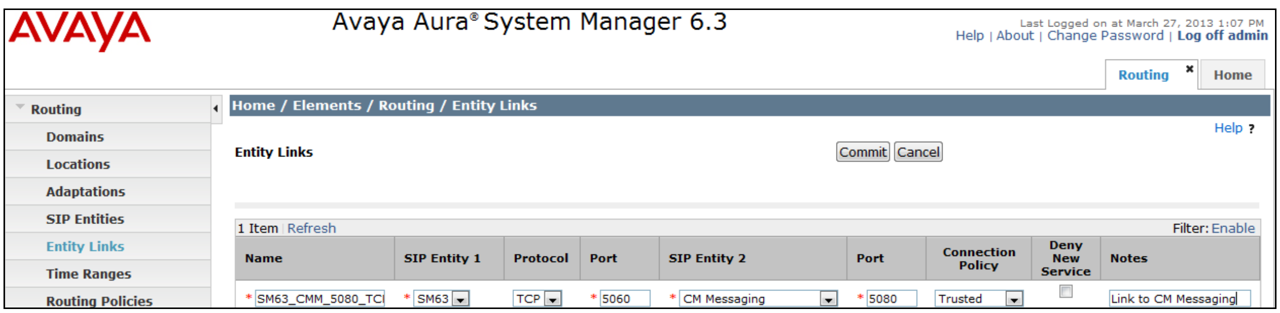
Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5060)



Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5080)



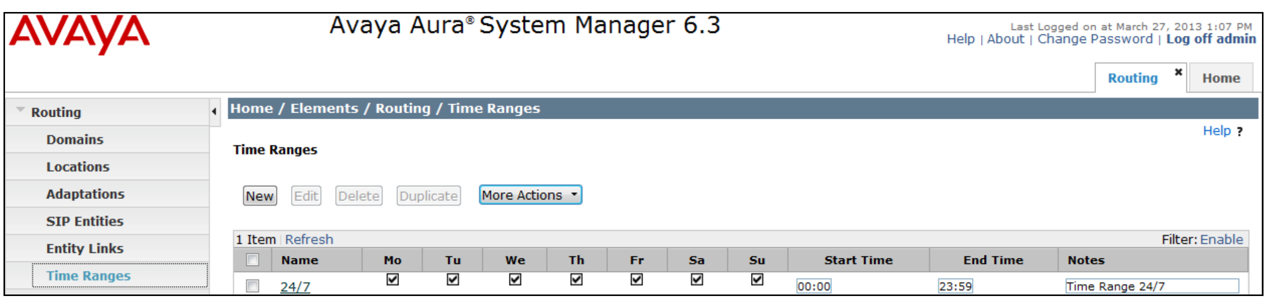
Entity link between Session Manager and Acme Packet SBC



Entity link between Session Manager and CM Messaging

5.6. Time Ranges

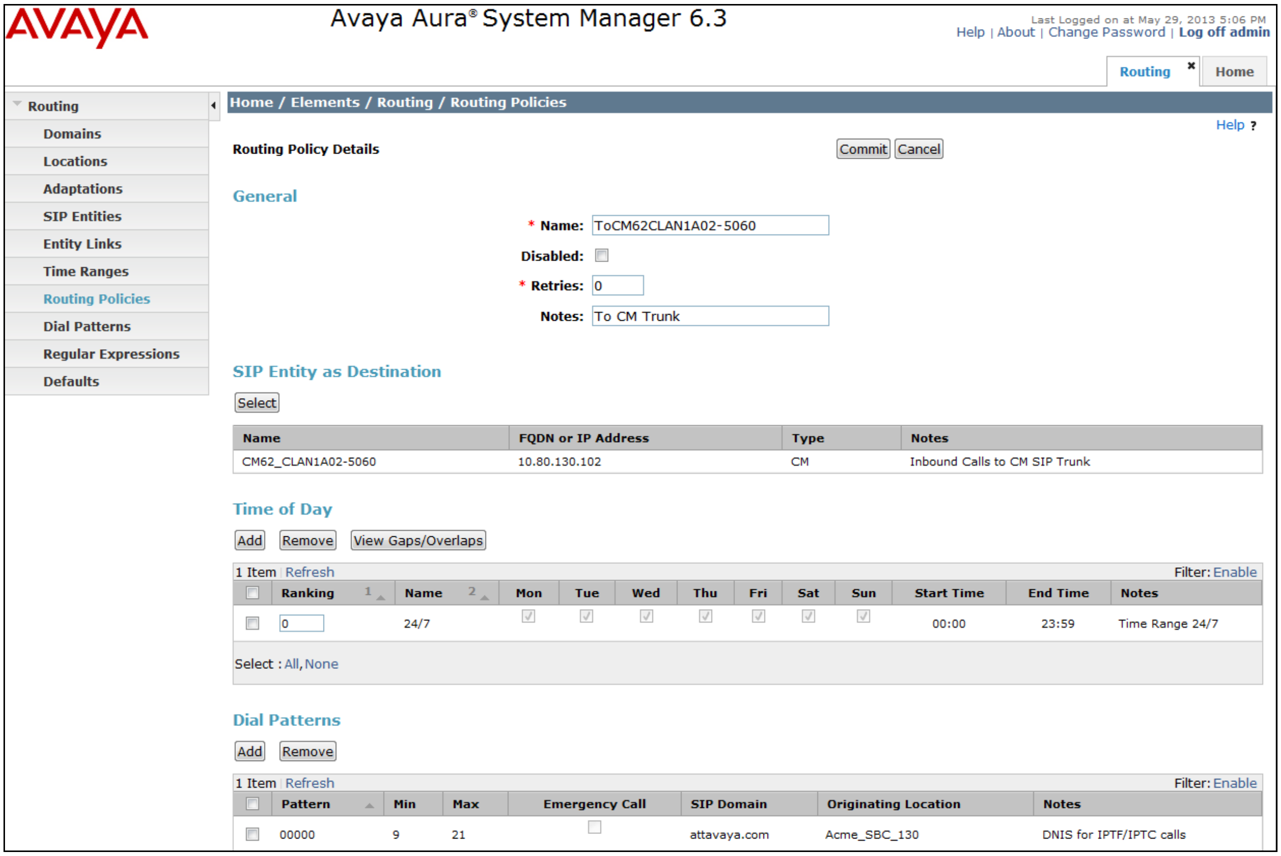
The following screen shows the time range used for AT&T IP Toll Free service testing.



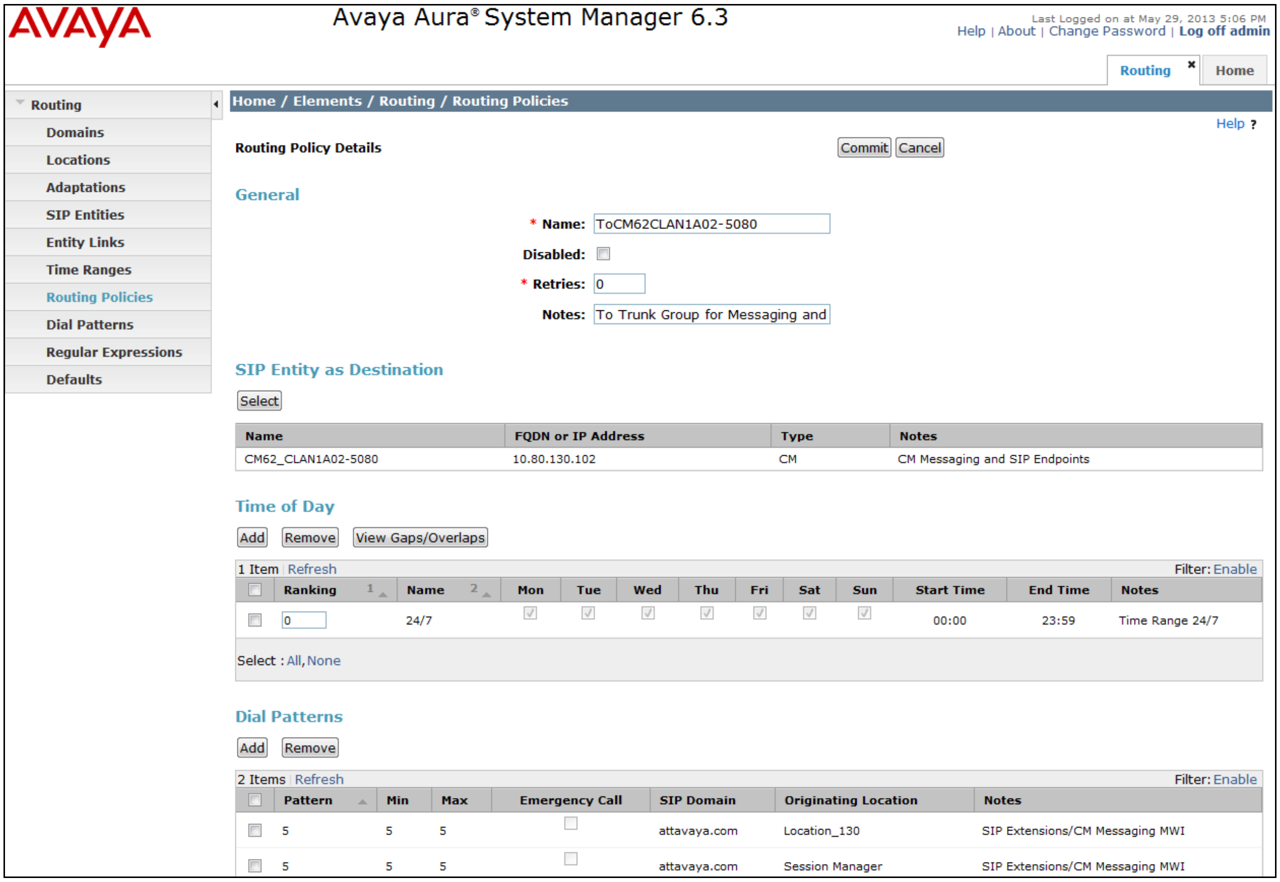
Time Range

5.7. Routing Policies

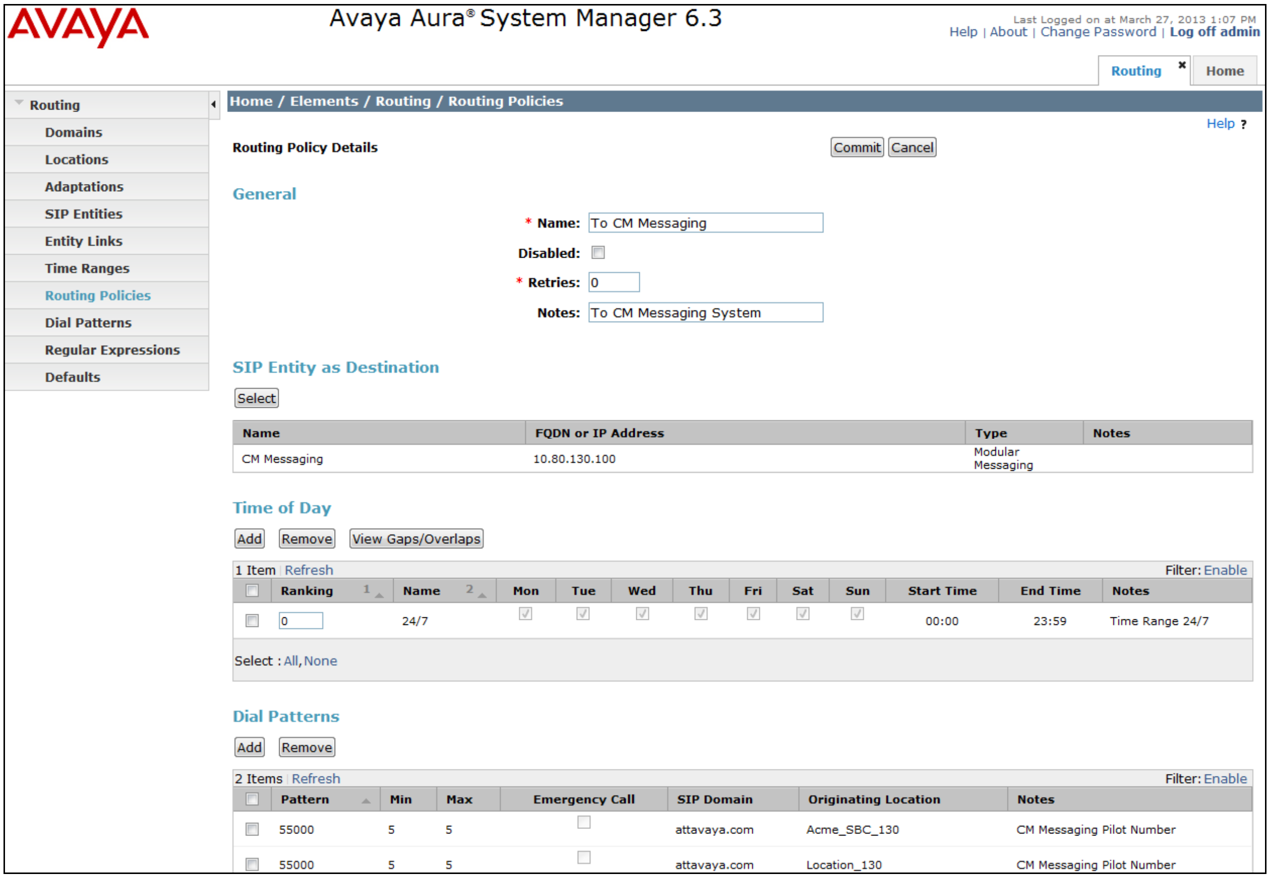
The following screens show routing policy along with dial patterns defined for AT&T IP Toll Free service. See **Section 5.8** for dial pattern configuration.



Routing Policy for Communication Manager (CLAN1A02-5060)



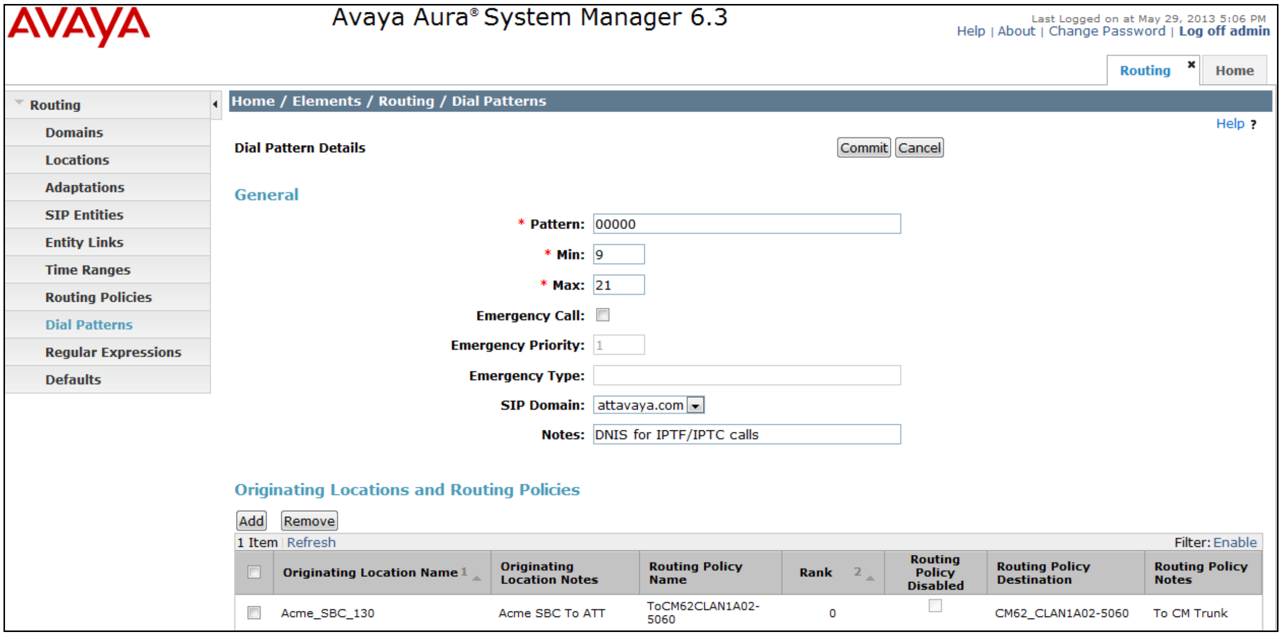
Routing Policy for Communication Manager (CLAN1A02-5080)



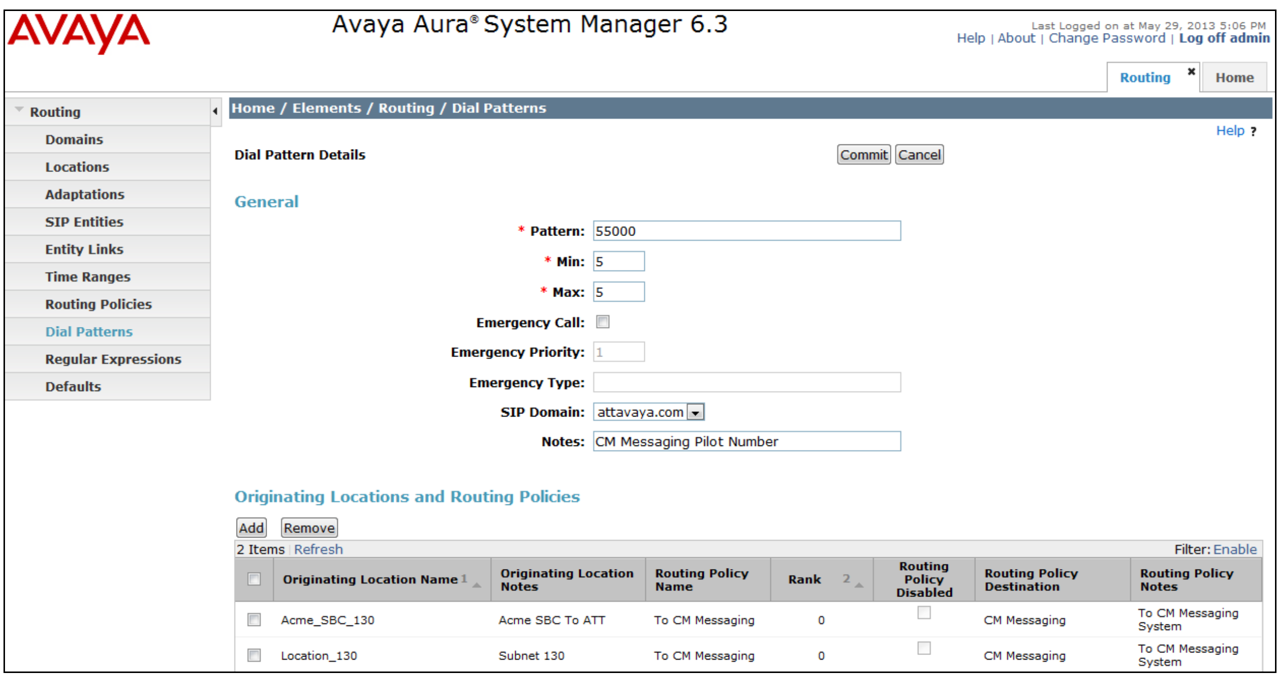
Routing Policy for CM Messaging Pilot Number

5.8. Dial Patterns

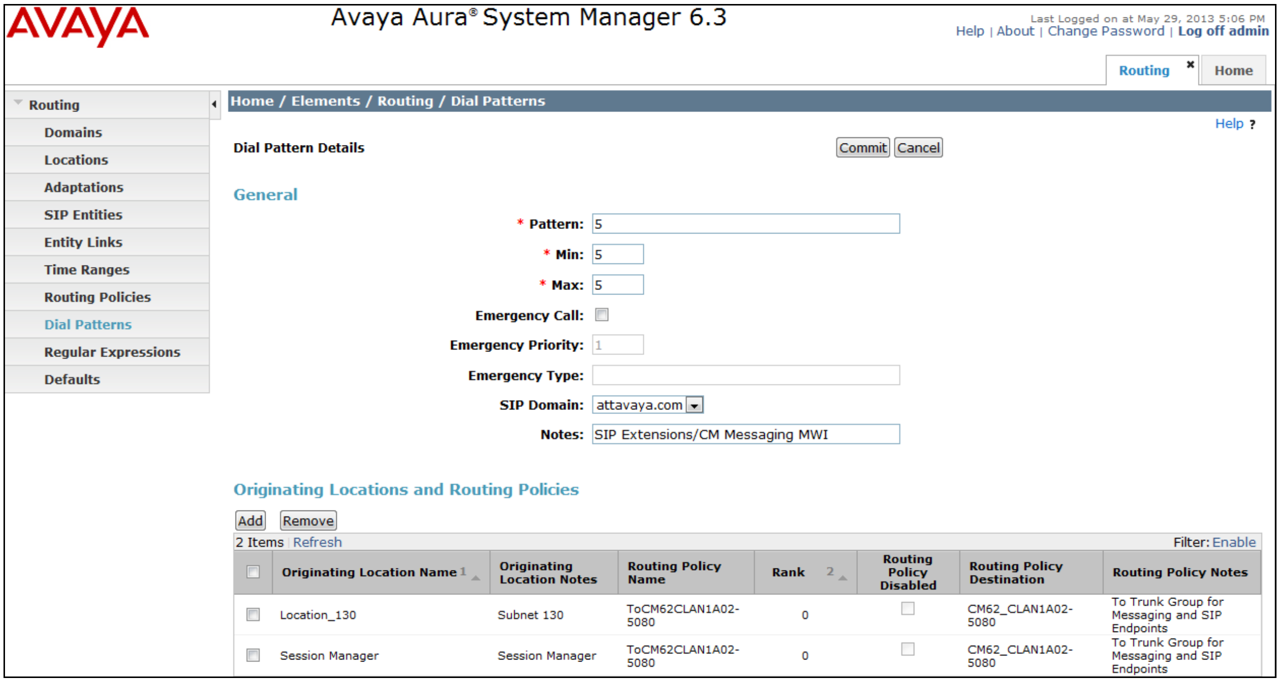
The following screens show dial patterns configured in this reference configuration.



Dial Pattern for Inbound Calls to Communication Manager (CLAN1A02-5060)



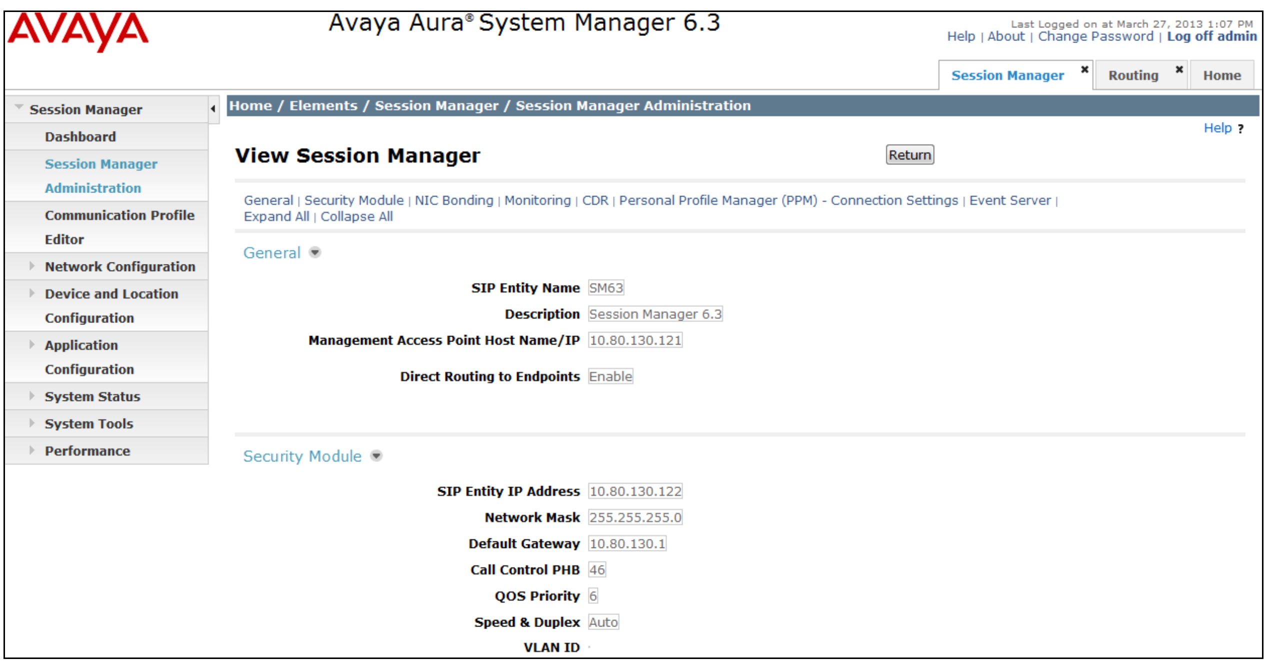
Dial Pattern for Covered Calls to CM Messaging



Dial Pattern for SIP Extension and CM Messaging MWI

5.9. Session Manager Administration

Navigate to **Home→Elements→Session Manager→Session Manager Administration** and in **Session Manager Instances** select the appropriate Session Manager already configured. The following screen shows the Session Manager instance **SM63** used in this reference configuration.



View Session Manager (SM63)

6. Configure Avaya Aura® Communication Manager 6.2

In this reference configuration Communication Manager 6.2 is provisioned in an Access Element configuration, supporting H.323, SIP and Digital endpoints. This section describes the administration steps for Communication Manager in support of the AT&T IP Toll Free service features listed in **Section 2**. These 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 [**5**] and [**6**] for further details if necessary.

Note – In the following sections, only the parameters that are highlighted in **bold** text are specifically applicable to this Application Notes. Other parameter values may or may not match based on local configurations.

6.1. 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.)
```

2. On **Page 4** of the **system-parameters customer-options** and verify that the **IP Trunks** field in 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
```

3. On **Page 11** of the **system-parameters features** and verify that the **Message Waiting Lamp Indicates Status For** field in is set to **loginID** to enable Message Waiting Indicator on the station where an agent is logged in.

```
display system-parameters features                            Page   11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
 EAS
        Expert Agent Selection (EAS) Enabled? y
       Minimum Agent-LoginID Password Length:
         Direct Agent Announcement Extension:
  Message Waiting Lamp Inidicates Status For: loginID
 VECTORING
                   Converse First Data Delay: 0    Second Data Delay: 2
              Converse Signaling Tone (msec): 100
                    Prompting Timeout (secs): 10
                Interflow-qpos EWT Threshold: 2
   Reverse Star/Pound Digit For Collect Step? n
         Available Agent Adjustments for BSR? n
                            BSR Tie Strategy: 1st-found
  Store VDN name in Station’s Local Call Log? n
 SERVICE OBSERVING
             Service Observing: Warning Tone? y    or Conference Tone? n
Service Observing/SSC Allowed with Exclusion? n
            Allow Two Observers in Same Call? n
```

6.2. Dial Plan

The dial plan defines how the digit string will be used locally by Communication Manager. Note that the values shown below are examples used in the reference configuration. Enter the **change dialplan analysis** command to provision the dial plan. Note the following dialed strings:

- 3-digit Dial Access Codes (indicated with a **Call Type** of **dac**) beginning with the digit **1** (e.g. Trunk Access Codes, TACs, defined for trunk groups in this reference configuration conform to this format).
- 4 or 5-digit Extensions with a **Call Type** of **ext** beginning with the digits **3xxxx**, **4xxx** or **5xxxx** (e.g. Announcements, Local extensions for Communication Manager stations, agents, Vector Directory Numbers, in this reference configuration conform to this format).
- 1-digit Facilities Access Code (indicated with a **Call Type** of **fac**) (e.g. **8/9** access code for AAR/ARS dialing).

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 1
    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   1           3   dac
   2           4   fac
   3           5   ext
   5           5   ext
   8           1   fac
   9           1   fac
```

6.3. IP Node Names

Following screen shows the node names used for AT&T IP Toll Free service provisioning.

```
change node-names ip                                         Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.80.130.1
CLAN-1A02           10.80.130.102
SM63                10.80.130.122
```

6.4. IP Codec Parameters

Following screen shows the codec set used in this reference configuration.

```
 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         3        30
 2: G.729A            n         3        30
 3: G.711MU           n         3        30
```

On **Page 2** of the **ip-codec-set** form, set **Mode** - **Fax** 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
```

6.5. IP Network Regions

Network Regions are used to group various Communication Manager Resources such as codecs, UDP port ranges, and inter-region communication. In this reference configuration only one network region was configured for all elements. Additional network regions can be defined if required. Enter **ip-network-region x**, where **x** is the number of an unused IP network region and configure as follows:

- **Name** - Enter any descriptive string.
- **Codec Set** – Set to Codec set configure in **Section 6.4**.
- **Intra and 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 within the same region.
- **UDP Port Min**: - Set to **16384** (Required for AT&T IP Toll Free service)
- **UDP Port Max**: - Set to **32767** (Required for AT&T IP Toll Free service)

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location:         Authoritative Domain: attavaya.com
    Name: ATT Calls
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? y
   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
```

On **Page 3** of the form, verify that region 2 is using codec set 2 as specified on **Page 1** (this field is automatically populated). If additional regions are configured, this form can dictate what codec set to be used for communication with elements belonging to different network regions.

```
change ip-network-region 2                                      Page   4 of  20
 Source Region: 2    Inter Network Region Connection Management    I       M
                                                                   G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening     Dyn A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions         CAC R  L    e
 1    2     y    NoLimit                                           n       t
 2    2                                                              all
```

6.6. SIP Trunks

Two trunks are configured for testing in this reference configuration.

- Trunk group to handle all inbound calls from AT&T IP Toll Free service
- Trunk group to handle CM Messaging and SIP extension registered with Session Manager

6.6.1. SIP Trunk for Inbound Calls with AT&T IP Toll Free service

This trunk is used to handle all the inbound calls from AT&T IP Toll Free service. Following steps are used to configure SIP Trunk on Communication Manager:

1. Enter the **add signaling-group x** command, where **x** is the number of an unused signaling group as shown in the following screen.

```
add signaling-group  1                                          Page 1 of 1
                             SIGNALING GROUP

Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                 Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

 Near-end Node Name: CLAN_1A02               Far-end Node Name: SM63
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 2
Far-end Domain: attavaya.com
                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? n
         Enable Layer 3 Test? y                 Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

2. Enter the **add trunk-group x** command, where **x** is the number of an unused trunk group.

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: ATT                         COR: 1        TN: 1        TAC: 101
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

3. On **Page 2** of the **trunk-group** form set the **Preferred Minimum Session Refresh Interval (sec)** field to **900.** This entry will actually cause a value of 1800 to be generated in the SIP header.

```
add trunk-group 1                                              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
 Disconnect Supervision - In? y  Out? y
           XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

4. On **Page 3** of the **trunk-group** form set **Numbering Format** field to **public**

```
add trunk-group 1                                             Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y
                     Numbering Format: public
                                         UUI Treatment: service-provider
                                           Replace Restricted Numbers? n
                                          Replace Unavailable Numbers? n
                               Modify Tandem Calling Number: no
 Show ANSWERED BY on Display? y
```

5. On **Page 4** of the **trunk-group** form:

- Set **Support Request History?** field to **n**.
- Set **Telephone Event Payload Type** field to the RTP payload type required by the AT&T IP Toll Free service (e.g. **100**).

```
add trunk-group 1                                             Page   4 of  21
                              PROTOCOL VARIATIONS
                           Mark Users as Phone? n
                 Prepend '+' to Calling Number? n
           Send Transferring Party Information? n
                      Network Call Redirection? n
                         Send Diversion Header? n
                       Support Request History? n
                  Telephone Event Payload Type: 100

            Convert 180 to 183 For Early Media? n
      Always Use re-INIVIT for Display Updates? n
            Identity for Calling Party Display? P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? n
```

6.6.2. SIP Trunk for CM Messaging and SIP Endpoints

This SIP trunk is used for coverage to CM Messaging and SIP Endpoints. Configuration for this trunk is similar to the trunk group configured in **Section 6.6.1** with the differences shown in the screens below:

```
add signaling-group  3                                       Page   1 of   1
                             SIGNALING GROUP

Group Number: 3              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

 Near-end Node Name: CLAN_1A02              Far-end Node Name: SM63
 Near-end Listen Port: 5080               Far-end Listen Port: 5080
                                       Far-end Network Region: 2
       Far-end Domain: attavaya.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? n          Alternate Route Timer(sec): 6
```

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

Group Number: 3                    Group Type: sip          CDR Reports: y
  Group Name: CM Messaging/SIP Endpoints  COR: 1       TN: 1          TAC: 103
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 3
                                                     Number of Members: 10
```

On **Page 3** of the trunk-group form set **Numbering Format** field to **private**

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

On **Page 4** of the **trunk-group** form, make sure that **Support Request History?** field is set to **y** [default].

```
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

            Convert 180 to 183 For Early Media? n
      Always Use re-INIVIT for Display Updates? n
            Identity for Calling Party Display? P-Asserted-Identity
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? n
```

6.7. Public Unknown Numbering

This form is used to populate the **History-info** and user part of the **To** header with the Communication Manager extension. In this reference configuration, all extension of length **5** and prefixed by **5** entering Communication Manager on trunk group **3** will populate the required SIP headers with the correct telephone extension. Additionally, this form is used for inbound calls to populate the user part in **Contact** and **PAI** headers.

```
change public-unknown-numbering 0                            Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext              Trk       CPN          CPN
Len Code             Grp(s)    Prefix       Len
 5                   3                      5        Total Administered: 2
 5  50001            3         73232        10         Maximum Entries: 9999
```

6.8. Alternate Automated Routing (AAR) Table

Automatic Alternate Routing (AAR) is used to direct calls to CM Messaging and SIP Endpoints registered with Session Manager via the route pattern defined in **Section 6.9**. In the following screen, **5005** string used for calls to SIP endpoints and **55000** is the pilot number used for coverage to CM Messaging via Session Manager using trunk configured in **Section 6.6.2**.

```
 change ars analysis 1                                       Page   1 of   2
                            ARS DIGIT ANALYSIS TABLE
                                 Location:  all        Percent Full:    15

          Dialed            Total     Route     Call    Node   ANI
          String           Min  Max  Pattern    Type    Num    Reqd
    5005                   5    5    3          aar            n
    55000                  5    5    3          aar            n
```

6.9. Route Pattern

This form defines the SIP trunk to be used based on the route pattern selected by the AAR table for calls destined to CM Messaging and SIP endpoints registered with Session Manager (see **Section 6.6.2**).

- **Grp No** – Set to **3** i.e. the trunk group configured in **Section 6.6.2**.
- **FRL** – Set to **0** (zero).

```
change route-pattern 3                                            Page 1 of   3
                    Pattern Number: 2   Pattern Name: ToCMM
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 3    0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

  BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
  0 1 2 M 4 W     Request                                Dgts Format
                                                       Subaddress
 1: y y y y y n  n             rest                                     none
 2: y y y y y n  n             rest                                     none
 3: y y y y y n  n             rest                                     none
 4: y y y y y n  n             rest                                     none
 5: y y y y y n  n             rest                                     none
 6: y y y y y n  n             rest                                     none
```


6.10. Optional Features

6.10.1. Call Center Provisioning

For provisioning the call center functionality, verify that the call center parameters are enabled as shown below. Verify that an agent login id is created with an appropriate skill. Verify the skill (hunt group) for that agent is in place. Make sure that a VDN as per the dial plan is in place along with the vector which lists the steps to be executed when an inbound call is received from AT&T IP Toll Free service.

Note - The administration of Communication Manager Call Center elements – hunt groups, vectors, and Vector Directory Numbers (VDNs) are beyond the scope of these Application Notes. Additional licensing may be required for some of these features. If necessary, consult [**5**] and [**6**] for further details. The configuration steps that follow are provided for reference purposes only.

```
display system-parameters customer-options                    Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES

                          Call Center Release: 5.0

                                    ACD? y                        Reason Codes? n
                            CMS (Basic)? y           Service Level Maximizer? n
              BCMS/VuStats Service Level? y         Service Observing (Basic)? n
  BSR Local Treatment for IP & ISDN? n  Service Observing (Remote/By FAC)? n
                      Business Advocate? n           ervice Observing (VDNs)? n
                        Call Work Codes? n                         Timed ACW? n
        DTMF Feedback Signals For VRU? n                Vectoring (Basic)? y
                       Dynamic Advocate? n            Vectoring (Prompting)? y
           Expert Agent Selection (EAS)? y         Vectoring (G3V4 Enhanced)? y
                                EAS-PHD? y          Vectoring (3.0 Enhanced)? y
                       Forced ACD Calls? n  Vectoring (ANI/II-Digits Routing)? y
                   Least Occupied Agent? n  Vectoring (G3V4 Advanced Routing)? y
             Lookahead Interflow (LAI)? n                 Vectoring (CINFO)? n
Multiple Call Handling (On Request)? n   Vectoring (Best Service Routing)? n
    Multiple Call Handling (Forced)? n              Vectoring (Holidays)? n
  PASTE (Display PBX Data on Phone)? n             Vectoring (Variables)? n

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

Call Center Optional Features Form

In the reference configuration below, an inbound call from AT&I IP Toll Free service is handled using the **VDN 2010** which routes the call to **Vector 10** and based upon the digits specified by the caller, the call is directed to an appropriate skill. **Skill 11** is shown for reference purposes. Additional skills can be similarly added.

```
display agent-loginID 53001                                 Page   1 of   3
                                 AGENT LOGINID

                Login ID: 53001                                     AAS? n
                    Name: 53001-9650-Agent                        AUDIX? n
                      TN: 1                               LWC Reception: spe
                     COR: 1                      LWC Log External Calls? n
           Coverage Path: 1                    AUDIX Name for Messaging:
           Security Code:
                                           LoginID for ISDN/SIP Display? n
                                                               Password:
                                                 Password (enter again):
                                                            Auto Answer: station
                                                      MIA Across Skills: system
                                              CW Agent Considered Idle: system
                                             Aux Work Reason Code Type: system
                                               Logout Reason Code Type: system
                      Maximum time agent in ACW before logout (sec): system
                                              Forced Agent Logout Time:   :

     WARNING:  Agent must log in again before changes take effect
```

Agent Form – Page 1

```
display agent-loginID 53001                                   Page   2 of   3
                                 AGENT LOGINID
      Direct Agent Skill:                                   Service Objective? n
Call Handling Preference: skill-level                 Local Call Preference? n

    SN   RL SL          SN   RL SL          SN   RL SL          SN   RL SL
 1: 11      1       16:                 31:                 46:
 2:                 17:                 32:                 47:
 3:                 18:                 33:                 48:
```

Agent Form – Page 2

```
display hunt-group 11                                          Page   1 of   4
                                  HUNT GROUP

            Group Number: 11                                    ACD? y
              Group Name: Skill-11                            Queue? y
         Group Extension: 53011                              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:
```

Skill (Hunt Group) Form – Page 1

```
display hunt-group 11                                         Page   2 of   3
                                  HUNT GROUP

                      Skill? y      Expected Call Handling Time (sec): 180
                        AAS? n
                   Measured: none
       Supervisor Extension:

        Controlling Adjunct: none

     Multiple Call Handling: none
Time ACW Interval (sec):     After Xfer or Held Call Drops? n
```

Skill (Hunt Group) Form – Page 2

```
display vdn 2010                                          Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                                 Extension: 2010
                                      Name: To SelectSkill
                               Destination: Vector Number        10
                       Attendant Vectoring? n
                      Meet-me Conferencing? n
                        Allow VDN Override? n
                                       COR: 1
                                       TN#: 1
                                  Measured: none

        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
* Follows VDN override rules
```

VDN (Vector Directory Number) Form

```
display vector 10                                           Page   1 of   6
                                  CALL VECTOR
    Number: 10                Name: RouteToSkill
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 collect      1    digits after announcement 33002    for none
03 goto vector  11   @step 2 if digits          =      1
04 goto vector  12   @step 2 if digits          =      2
05 goto vector  13   @step 2 if digits          =      3
06 goto step    2            if unconditionally
```

Vector (RouteToSkill) Form

```
display vector 11                                          Page   1 of   6
                                  CALL VECTOR
    Number: 11                Name: Skill 11
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock?
n
     Basic? y   EAS? n   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing?
y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 33003
03 queue-to     skill 11   pri m
04 announcement 33006
05 goto step    3             if unconditionally
06
```

Vector (Skill 11) Form

6.10.2. CM Messaging Coverage Path and Hunt Group

Hunt group 1 is used in the reference configuration to verify CM Messaging coverage functionality. This hunt group is defined with the 5 digit CM Messaging pilot number **55000**. The hunt group is associated with call **coverage path 1** in form below and the coverage path is assigned to a station (e.g., **50001**). Communication Manager will use the AAR access code **8** (defined in **Section 6.2**) to dial CM Messaging (e.g. **855000**) as shown on **hunt-group** form below.

```
display coverage path 1                                      Page   1 of   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: 4
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n
COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1             Rng: 4  Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

Coverage Path Form

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

            Group Number: 1                                ACD? n
              Group Name: CM Messaging                   Queue? n
         Group Extension: 55000                         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: grp-name
```

Hunt Group Form – 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)
     55000                    55000                    8
```

Hunt Group Form – Page 2

```
display station 50001                                           Page   1 of   5
                                     STATION
Extension: 50001                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: 123456          TN: 1
     Port: S00000                      Coverage Path 1: 1              COR: 1
     Name: H323-96XX-50001             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: 50001
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                   Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
                                                          IP Video? n
                               Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y
```

Station Form

6.11. Saving Translations

To save all Communication Manager provisioning changes, enter the command **save translations.**

7. Configure Acme Packet Session Border Controller

These Application Notes assume that basic Acme Packet SBC administration has already been performed. The Acme Packet SBC configuration used in the reference configuration is provided below as a reference. The notable settings are highlighted in bold and brief annotations are provided on the pertinent settings. Use **putty** or similar tool to access Acme Packet SBC for configuration. Consult with Acme Packet Support [**7**] for further details and explanations on the configuration below.

ANNOTATION: The local policies below governs the routing of SIP messages from elements on the network on which the Avaya elements, e.g., Session Manager, Communication Manager, etc., reside to the AT&T IP services. The Session Agent Groups (**SAG**) defined here, and further down, provisioned under the session-groups **SP-PROXY** and **ENTERPRISE.** Note: Although **Enterprise** policy is not used for AT&T IP Toll Free service but is left in there in case the customer is using AT&T IP Flexible Reach service.

```
local-policy
        from-address
                                *
        to-address
                                *
        source-realm
                                Enterprise
        description
        activate-time           N/A
        deactivate-time         N/A
        state                   enabled
        policy-priority         none
        policy-attribute
                next-hop                sag:SP_PROXY
                realm                   ATT
                action                  none
                terminate-recursion     disabled
                carrier
                start-time              0000
                end-time                2400
                days-of-week            U-S
                cost                    0
                app-protocol
                state                   enabled
                methods
                media-profiles
                lookup                  single
                next-key
                eloc-str-lkup           disabled
                eloc-str-match
```

ANNOTATION: The local policy below governs the routing of SIP messages from the AT&T IP services to Session Manager.

local-policy

```
        from-address
                            *
        to-address
                            *
        source-realm
                            ATT
        description
        activate-time       N/A
        deactivate-time     N/A
        state               enabled
        policy-priority     none
        policy-attribute
                next-hop                10.80.130.122
                realm                   Enterprise
                action                  none
                terminate-recursion     disabled
                carrier
                start-time              0000
                end-time                2400
                days-of-week            U-S
                cost                    0
                app-protocol            SIP
                state                   enabled
                methods
                media-profiles
                lookup                  single
                next-key
                eloc-str-lkup           disabled
                eloc-str-match
network-interface
        name                wancom0
        sub-port-id         0
        description
        hostname
        ip-address          192.9.230.221
        pri-utility-addr
        sec-utility-addr
        netmask             255.255.255.0
        gateway             192.9.230.254
        sec-gateway
        gw-heartbeat
                state                   disabled
                heartbeat               0
                retry-count             0
                retry-timeout           1
                health-score            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout         11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
```

ssh-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	**s0p0**	
sub-port-id	0	
description		
hostname		
ip-address	**10.80.130.250**	
pri-utility-addr		
sec-utility-addr		
netmask	**255.255.255.0**	
gateway	**10.80.130.1**	
sec-gateway		
gw-heartbeat		
	state	disabled
	heartbeat	0
	retry-count	0
	retry-timeout	1
	health-score	0
dns-ip-primary		
dns-ip-backup1		
dns-ip-backup2		
dns-domain	attavaya.com	
dns-timeout	11	
hip-ip-list	**10.80.130.250**	
ftp-address		
icmp-address	**10.80.130.250**	
snmp-address		
telnet-address		
ssh-address		

```
ANNOTATION: The network interface below defines the IP addresses on the
interface connected to the network on which the AT&T IP services reside.
```

network-interface

name	**s1p0**	
sub-port-id	0	
description		
hostname		
ip-address	**192.168.62.50**	
pri-utility-addr		
sec-utility-addr		
netmask	**255.255.255.128**	
gateway	**192.168.62.1**	
sec-gateway		
gw-heartbeat		
	state	disabled
	heartbeat	0
	retry-count	0
	retry-timeout	1
	health-score	0
dns-ip-primary		
dns-ip-backup1		

dns-ip-backup2
dns-domain
dns-timeout 11
hip-ip-list **192.168.62.50**
ftp-address
icmp-address **192.168.62.50**
snmp-address
telnet-address
ssh-address

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP services reside, and applies the SIP manipulation **NAT_IP.** Note that this manipulation is not used for AT&T IP Toll Free service but is kept in here for customers also subscribing to AT&T IP Flexible Reach service.

realm-config

identifier	**ATT**
description	
addr-prefix	0.0.0.0
network-interface	**s1p0:0**
mm-in-realm	enabled
mm-in-network	enabled
mm-same-ip	enabled
mm-in-system	enabled
bw-cac-non-mm	disabled
msm-release	disabled
generate-UDP-checksum	disabled
max-bandwidth	0
fallback-bandwidth	0
max-priority-bandwidth	0
max-latency	0
max-jitter	0
max-packet-loss	0
observ-window-size	0
parent-realm	
dns-realm	
media-policy	
media-sec-policy	
in-translationid	
out-translationid	
in-manipulationid	
out-manipulationid	**NAT_IP**
manipulation-string	
manipulation-pattern	
class-profile	
average-rate-limit	0
access-control-trust-level	none
invalid-signal-threshold	0
maximum-signal-threshold	0
untrusted-signal-threshold	0
nat-trust-threshold	0
deny-period	30
ext-policy-svr	
diam-e2-address-realm	
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
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
constraint-name	
call-recording-server-id	
xnq-state	xnq-unknown
hairpin-id	0
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	
sip-profile	
sip-isup-profile	
block-rtcp	disabled
hide-egress-media-update	disabled

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

realm-config

identifier	**Enterprise**
description	
addr-prefix	0.0.0.0
network-interfaces	**s0p0:0**
mm-in-realm	enabled
mm-in-network	enabled
mm-same-ip	enabled
mm-in-system	enabled
bw-cac-non-mm	disabled
msm-release	disabled
generate-UDP-checksum	disabled
max-bandwidth	0
fallback-bandwidth	0
max-priority-bandwidth	0

```
max-latency                    0
max-jitter                     0
max-packet-loss                0
observ-window-size             0
parent-realm
dns-realm
media-policy
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit             0
access-control-trust-level     none
invalid-signal-threshold       0
maximum-signal-threshold 0
untrusted-signal-threshold  0
nat-trust-threshold            0
deny-period                    30
ext-policy-svr
diam-e2-address-realm
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            enabled
dyn-refer-term                 disabled
codec-policy
codec-manip-in-realm           disabled
constraint-name
call-recording-server-id
xnq-state                      xnq-unknown
hairpin-id                     0
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	
sip-profile	
sip-isup-profile	
block-rtcp	disabled
hide-egress-media-update	disabled

ANNOTATION: The session agent below represents the Session Manager used in this reference configuration.

session-agent

hostname	**SM63**
ip-address	**10.80.130.122**
port	**5060**
state	**enabled**
app-protocol	**SIP**
app-type	
transport-method	**UDP+TCP**
realm-id	**Enterprise**
egress-realm-id	
description	
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=1**
ping-interval	**180**
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
media-profiles	
in-translationid	
out-translationid	
trust-me	enabled

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 enabled
tcp-reconn-interval 10
max-register-burst-rate 0
register-burst-window 0
sip-profile
sip-isup-profile

ANNOTATION: The session agent below represents the AT&T IP services border element. The Acme Packet SBC 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.

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

max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	**OPTIONS;hops=70**
ping-interval	**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	enabled
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	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0
sip-profile	
sip-isup-profile	

ANNOTATION: The session agent below is used for failover testing to ATT IP services. The state is changed to **enabled** when the testing is performed.

session-agent

hostname	**1.1.1.1**
ip-address	**1.1.1.1**
port	**5060**
state	**disabled**
app-protocol	**SIP**
app-type	
transport-method	**UDP**
realm-id	**ATT**
egress-realm-id	
description	**ATT-Failover**
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	
loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	**OPTIONS;hops=70**
ping-interval	**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	NONE
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 services border element.

Note - Multiple session-agents may be specified in a session-group. The *strategy* parameter may be used to select how these multiple session-agents are used (e.g. *Hunt* and *RoundRobin*).

session-group

group-name	**SP_PROXY**
description	
state	**enabled**
app-protocol	**SIP**
strategy	**RoundRobin**
dest	
	1.1.1.1
	135.242.225.210
trunk-group	
sag-recursion	enabled
stop-sag-recurse	401,407

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

sip-interface

state	enabled	
realm-id	**ATT**	
description		
sip-port		
	address	**192.168.62.50**
	port	**5060**
	transport-protocol	**UDP**
	tls-profile	
	allow-anonymous	all
	ims-aka-profile	
carriers		
trans-expire	0	

```
invite-expire                   0
max-redirect-contacts           0
proxy-mode
redirect-action
contact-mode                    none
nat-traversal                   none
nat-interval                    30
tcp-nat-interval                90
registration-caching            disabled
min-reg-expire                  300
registration-interval           3600
route-to-registrar              disabled
secured-network                 disabled
teluri-scheme                   disabled
uri-fqdn-domain
trust-mode                      all
max-nat-interval                3600
nat-int-increment               10
nat-test-increment              30
sip-dynamic-hnt                 disabled
stop-recurse                    401,407
port-map-start                  0
port-map-end                    0
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
sip-ims-feature                 disabled
operator-identifier
anonymous-priority              none
max-incoming-conns              0
per-src-ip-max-incoming-conns 0
inactive-conn-timeout           0
untrusted-conn-timeout          0
network-id
ext-policy-server
default-location-string
charging-vector-mode            pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode                  none
implicit-service-route          disabled
rfc2833-payload                 101
rfc2833-mode                    transparent
constraint-name
response-map
local-response-map
ims-aka-feature                 disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive                   none
add-sdp-invite                  disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```

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

sip-interface

state		enabled
realm-id		**Enterprise**
description		
sip-port		
	address	**10.80.130.250**
	port	**5060**
	transport-protocol	**TCP**
	tls-profile	
	allow-anonymous	all
	ims-aka-profile	
carriers		
trans-expire		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		**rejectOptions**
manipulation-string		
manipulation-pattern		
sip-ims-feature		disabled
operator-identifier		
anonymous-priority		none
max-incoming-conns		0
per-src-ip-max-incoming-conns		0
inactive-conn-timeout		0
untrusted-conn-timeout		0
network-id		
ext-policy-server		
default-location-string		
charging-vector-mode		pass
charging-function-address-mode		pass


```
        ccf-address
        ecf-address
        term-tgrp-mode                none
        implicit-service-route        disabled
        rfc2833-payload               101
        rfc2833-mode                  transparent
        constraint-name
        response-map
        local-response-map
        ims-aka-feature               disabled
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                 none
        add-sdp-invite                disabled
        add-sdp-profiles
        sip-profile
        sip-isup-profile
```

ANNOTATION: The SIP manipulations shown below are used for modifying several headers (To, From and Contact) to hide the CPE topology.

sip-manipulation

```
        name                          NAT_IP
        description                   Topology hiding for To, From headers
        split-headers
        join-headers
        header-rule
                name                          manipFrom
                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                          manipTo
                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
new-value $REMOTE_IP
header-rule
name modContactPlus
header-name Contact
action manipulate
comparison-type pattern-rule
msg-type any
methods INVITE
match-value
new-value
element-rule
name modVal
parameter-name
type uri-user
action find-replace-all
match-val-type any
comparison-type case-sensitive
match-value \+(.*)
new-value $modContactPlus.$modVal.$1

ANNOTATION: The SIP header manipulation shown below modifies the **sendonly** value in SDP to **sendrecv** using header rule **modsendonly**. See **Section 2.2, Item 1** for further details.

header-rule
name modsendonly
header-name Content-type
action manipulate
comparison-type case-sensitive
msg-type any
methods INVITE
match-value
new-value
element-rule
name modmline
parameter-name application/sdp
type mime
action find-replace-all
match-val-type any
comparison-type case-sensitive
match-value sendonly
new-value sendrecv

ANNOTATION: The SIP manipulation shown below intercepts the SIP OPTIONS message from AT&T Border Element and responds with Acme Packet alive message.

sip-manipulation
name rejectOptions
description
split-headers
join-headers
header-rule

name **RejectOpts**
header-name **From**
action **reject**
comparison-type **case-sensitive**
msg-type **request**
methods **OPTIONS**
match-value
new-value **405:"Acme Packet is alive, check back later"**

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

```
steering-pool
        ip-address                192.168.62.50
        start-port                16384
        end-port                  32767
        realm-id                  ATT
steering-pool
        ip-address                10.80.130.250
        start-port                16384
        end-port                  32767
        realm-id                  Enterprise
system-config
        hostname                  Enterprise-Acme
        description
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled              enabled
        enable-snmp-auth-traps    disabled
        enable-snmp-syslog-notify disabled
        enable-snmp-monitor-traps disabled
        enable-env-monitor-traps  disabled
        snmp-syslog-his-table-length 1
        snmp-syslog-level         WARNING
        system-log-level          WARNING
        process-log-level         NOTICE
        process-log-ip-address    0.0.0.0
        process-log-port          0
        collect
                sample-interval           5
                push-interval             15
                boot-state                disabled
                start-time                now
                end-time                  never
                red-collect-state         disabled
                red-max-trans             1000
                red-sync-start-time       5000
                red-sync-comp-time        1000
```

push-success-trap-state	disabled
call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	**192.168.62.1**
restart	enabled
exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	disabled
cli-more	disabled
terminal-height	24
debug-timeout	0
trap-event-lifetime	0
default-v6-gateway	::
ipv6-support	disabled
cleanup-time-of-day	00:00

8. Verification Steps

The following steps may be used to verify this reference configuration:

8.1. Avaya Aura® Communication Manager

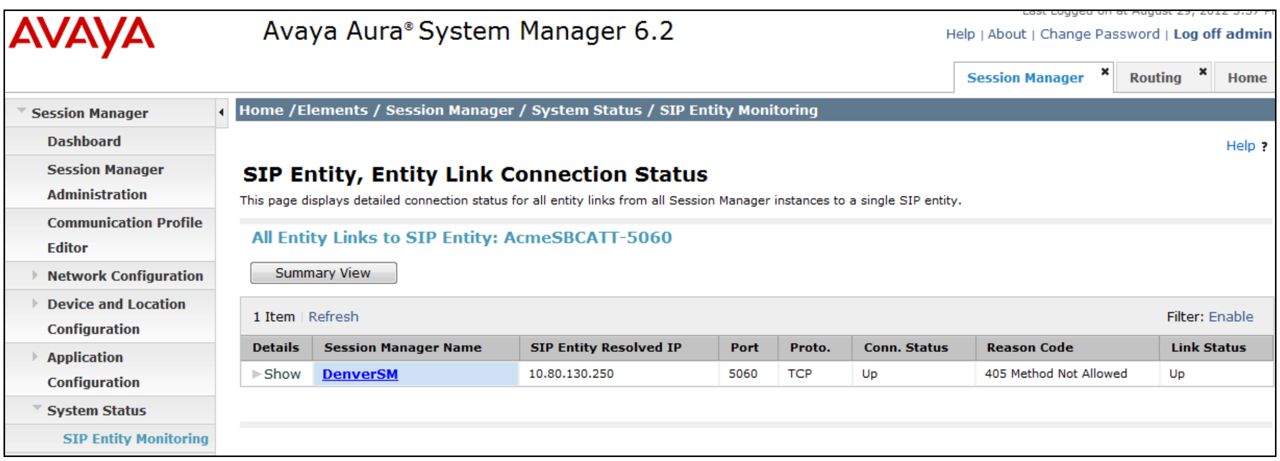
The following examples are only a few of the monitoring commands available on Communication Manager. See [**5**] and [**6**] for more information.

- From the Communication Manager console connection enter the command ***list trace tac xxx***, where ***xxx*** is a trunk access code to verify that the inbound or outbound calls are using the right trunk groups. Similarly, ***list trace station***, ***list trace vdn***, and ***list trace vector***, ***status trunk*** and ***status station*** commands can be used on Communication Manager.

8.2. Avaya Aura® Session Manager

Navigate to **Home→ Elements→ Session Manager→ System Status → SIP Entity Monitoring** and click on the SIP Entity for which the status is required. Following screen shows status for the entity link between Session Manager and Acme Packet SBC.

Note: The Reason Code column indicates that Session Manager has received a **SIP 405 Method Not Allowed** response (normal for this reference configuration) to the **SIP OPTIONS** it generated. This response is sufficient for SIP Link Monitoring to consider the link up.



8.3. AT&T IP Toll Free

1. Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.
2. Verify basic call functions such as hold, transfer and conference.
3. Verify the use of DTMF signaling.

9. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and the Acme Packet SBC can be configured to interoperate successfully with the AT&T IP Toll Free service using either AVPN or MIS-PNT transport. This solution provides users of Avaya Aura® Communication Manager the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection. These Application Notes further demonstrated that the Avaya Aura® Session Manager Adaptation Module could be utilized to do digit manipulation for inbound calls.

Note: These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

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.

Avaya Aura® Session Manager/System Manager

[1] Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.3, December 2012

[2] Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473 Issue 2, November 2010

[3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, Release 6.3, December 2012

[4] Administering Avaya Aura® System Manager, Release 6.3, Issue 1.0, December 2012

Avaya Aura® Communication Manager

[5] Administering Avaya Aura® Communication Manager, Issue 7.0, Release 6.2, December 2012, Document Number 03-300509

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

Acme Packet Support (login required):

[7] http://www.acmepacket.com/support.htm

AT&T IP Toll Free Service Descriptions:

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

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