Avaya Solution & Interoperability Test Lab

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

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Communication Manager R6.3, Avaya Aura® Session Manager R6.3, and the Acme Packet Net-Net 3800 with the AT&T IP Flexible Reach and IP Flexible Reach-Enhanced Features service using **AVPN** or **MIS/PNT** transport connections. The AT&T IP Flexible Reach is one of the many SIP-based Voice over IP services offered to enterprises for their voice communication needs. The AT&T IP Flexible Reach-Enhanced Features service is a SIP based service which includes additional network based features which are not part of IP Flexible Reach service.

Avaya Aura® Session Manager R6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. In the reference configuration, Avaya Aura® Communication Manager R6.3 is provisioned as a Telephony Application Server. Acme Packet Net-Net 3800 is the point of connection between Avaya Aura® Session Manager R6.3 and the AT&T IP Flexible Reach and IP Flexible Reach-Enhanced Features service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

2.1. Interoperability Compliance Testing

The interoperability compliance testing verified basic inbound and outbound call flows along with Enhanced Features with AT&T IP Flexible Reach service. **Section 3.2** provides call flows tested for AT&T IP Flexible Reach service.

The compliance testing was based on a test plan provided by AT&T. This test plan examines the functionality required by AT&T for solution certification as supported on the AT&T network. Calls were made to and from the PSTN across the AT&T network.

- AT&T IP Flexible Reach service
    - SIP trunking.
    - Inbound and outbound dialing including international calls.
    - Voicemail (leave and retrieve messages).
    - T.38 Fax.
    - Passing of DTMF events and their recognition by navigating automated menus.
    - Basic telephony features such as hold, resume, conference and transfer.
    - Call Forward with Diversion Header.

- AT&T Network IP Flexible Reach-Enhanced Features
    - Network based Simultaneous Ring
    - Network based Sequential Ring (Locate Me)
    - Network based Blind Call Transfer using SIP REFER on Communication Manager[1]
    - Network based Call Forwarding Always (CFA/CFU)
    - Network based Call Forwarding Ring No Answer (CF-RNA)
    - Network based Call Forwarding Busy (CF-Busy)
    - Network based Call Forwarding Not Reachable (CF-NR)

2.2. Test Results and Known Limitations

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

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. While using Meetme-Conference feature on Communication Manager, when the number of parties on PSTN connected to Communication Manager goes down to two, and if Network Call Redirection (NCR) is enabled, Communication Manager sends a REFER message back to AT&T IP Flexible Reach service which in turn acknowledges the REFER and a BYE is received by the remaining two parties on the conference. As a result, the two parties are directly connected to each other. This does not happen if one of the parties is on the Enterprise side and connected to Communication Manager. As a workaround, the DIDs used for this feature can use a separate trunk with NCR set to disabled as shown in **Section 6.6.1**.

[1] Network based Blind Call Transfer uses Vectors and VDNs on Communication Manager. Phone based transfers (attended or unattended) are not supported.

3. Unattended and Attended off-net transfer from Communication Manager phones is not supported. This may be supported when a two trunk solution is implemented and the call routes over NCR disabled trunk as shown in **Section 6.6.1**.
4. G.711 faxing is not supported between Communication Manager and the AT&T IP Flexible Reach service. 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 Communication Manager.
5. AT&T IP Flexible Reach service introduced a new **Resource-Priority** header in the initial INVITE for an inbound call. This header is not supposed to be present and AT&T is investigating this issue. This header creates a problem for calls being forwarded off-net as Communication Manager does not process this header properly and a defect **defsw130595** was entered against Communication Manager to investigate this issue. A Header Manipulation Rule shown in **Section 7** was provided to remove the **Resource-Priority** header from the initial **INVITE** sent by AT&T IP Flexible Reach service.
6. For outbound calls originating from Avaya, some additional headers such as **Endpoint-View**, **P-Location**, **Av-Global-Session-ID**, etc. are sent which may create problem because of packet size limitation implemented by AT&T IP Flexible Reach service. The presence of these additional headers makes AT&T IP Flexible Reach service return a **408 – Request** timeout error message. A Header Manipulation Rule shown in **Section 7** was provided to remove these headers.
7. For outbound calls origination from Avaya SIP telephones, two additional Bandwidth statements, **b= CT:64** and **b= AS:64** in the original **INVITE** are sent to AT&T IP Flexible Reach service. The bandwidth statements in the SDP of the original **INVITE** result in failure of calls to AT&T IP Teleconferencing service. A Header Manipulation Rule shown in **Section 7** was provided to remove these elements from the original INVITE to AT&T IP Flexible Reach service.
8. Calls From/to Customer Trunks via the same AT&T border element (e.g., “looped” calls), which result in Communications Manager sending a **491 Request Pending**, may experience a dropped call. This issue was observed during a **looped** call where Communication Manager phone dials an AT&T IP Flexible Reach number, and the network destination of that call is a second Communication Manager phone behind the same AT&T border element. Sometimes these calls result in Communication Manager issuing a **491 Request Pending** in response to the **looped Invite** from the network. When the network also **loops** the **491** back to Communication Manager, the network inserts a Contact header that contains the IP address of an internal AT&T network node. As a result, Communication Manager attempts to route subsequent **Invites** to this un-routable address. Eventually these **Invites** time out and the call may be dropped. This issue is under investigation by AT&T.
9. Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is the customer’s responsibility to ensure proper operation with its equipment/software vendor.

   While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

2.3. Support

AT&T customers may obtain support for the AT&T IP Flexible Reach 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:

- 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 R6.3 runs on an Avaya S8800 Server in a G650/Control LAN (C-LAN) configuration. This solution is extensible to other Avaya S8xxx 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 (H323/SIP) PC based softphone.
- The Acme Packet SBC provides SIP Session Border Controller functionality, including address translation and SIP header manipulation between the AT&T IP Flexible Reach service and the enterprise internal network[2]. UDP transport protocol is used between the Acme Packet SBC and the AT&T Flexible Reach 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 and outbound calls were placed between PSTN and the Customer Premises Equipment (CPE) via the AT&T IP Flexible Reach service, through the Acme Packet SBC, Session Manager, and Communication Manager. Communication Manager originated/terminated the calls using appropriate phone or fax stations. The H.323 phones at the CPE are registered to the Avaya Aura® Communication Manager C-LANs and the SIP phones are registered to Session Manager.

[2] The AT&T Enhanced IP Flexible Reach service uses SIP over UDP to communicate with enterprise edge SIP devices, e.g., the Acme Packet SBC in this sample configuration. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements, e.g., the Acme Packet SBC and Communication Manager. In the reference configuration, Session Manager uses SIP over TCP to communicate with the Acme Packet SBC and Communication Manager.

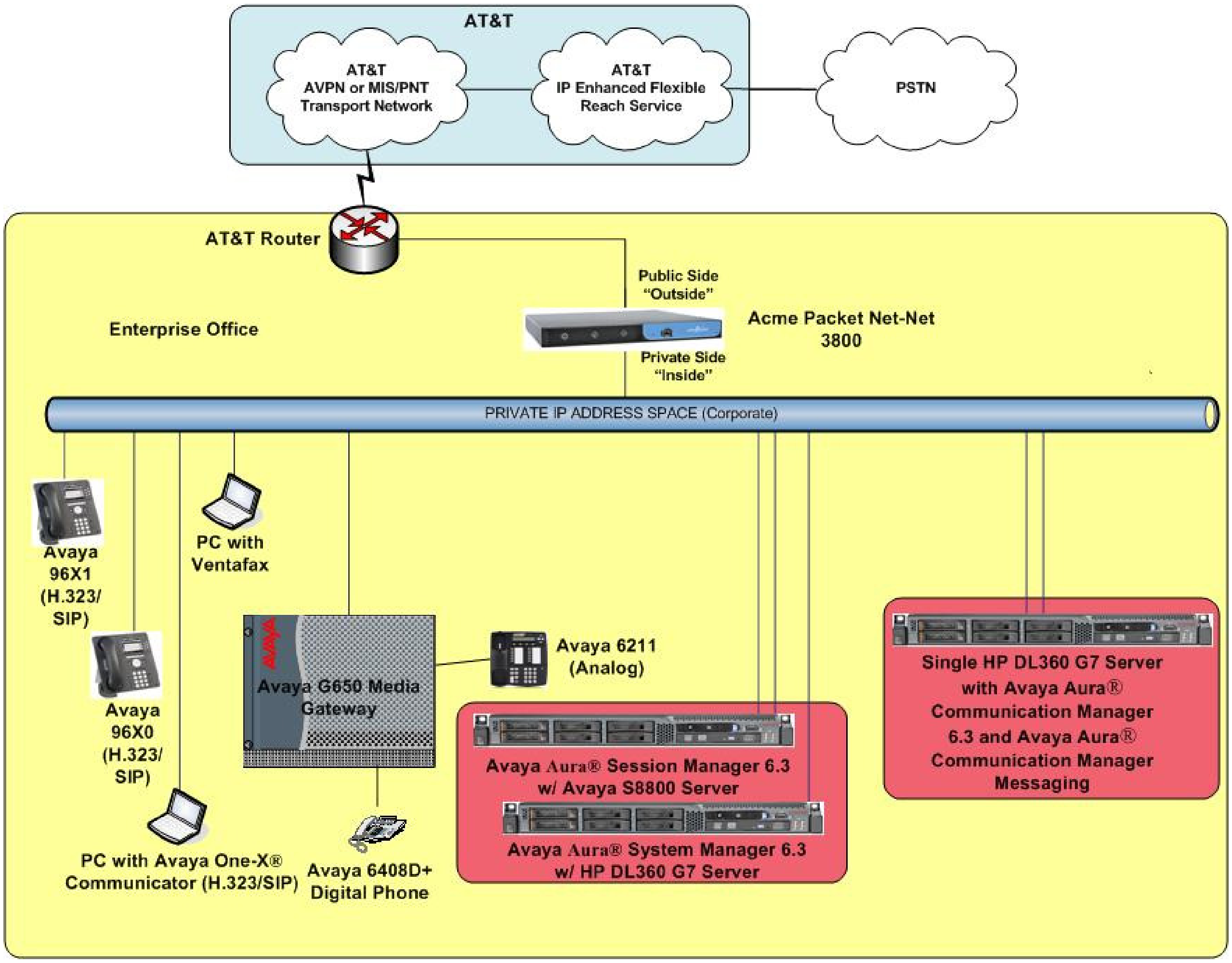


Figure 1: Reference Configuration

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 configurations. For security purposes, real IP addresses and DIDs were not included.

Note - The AT&T IP Flexible Reach-Enhanced Features service Border Element IP address and DNIS digits, (destination digits specified in the SIP Request URIs sent by the AT&T Flexible Reach-Enhanced Features 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 Flexible Reach-Enhanced Features 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	10.80.130.204
Media Processor (MedPro) IP Address	10.80.130.207
Avaya Aura® Communication Manager extensions	50xxx
Acme Packet Session Border Controller	
IP Address of “Outside” (Public) Interface (connected to AT&T Access Router/IP Flexible Reach-Enhanced Features service)	192.252.35.202
IP Address of “Inside” (Private) Interface (connected to Avaya Aura® Session Manager)	10.80.130.250
AT&T IP Flexible Reach-Enhanced Features service	
Border Element IP Address	135.194.131.41

Table 1: Illustrative Values Used in this Compliance Test

3.2. Call Flows

To understand how inbound AT&T IP Flexible Reach service calls are handled by Session Manager and Communication Manager, five basic call flows are described in this section, however for brevity not all possible call flows are described.

3.2.1. Inbound

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Flexible Reach service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a phone, fax, or in some cases, a vector.

1. A PSTN phone originates a call to an AT&T IP Flexible Reach service number.
2. The PSTN routes the call to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach 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 Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to Communication Manager.
6. Depending on the called number, Communication Manager routes the call to a phone, a fax or a vector.

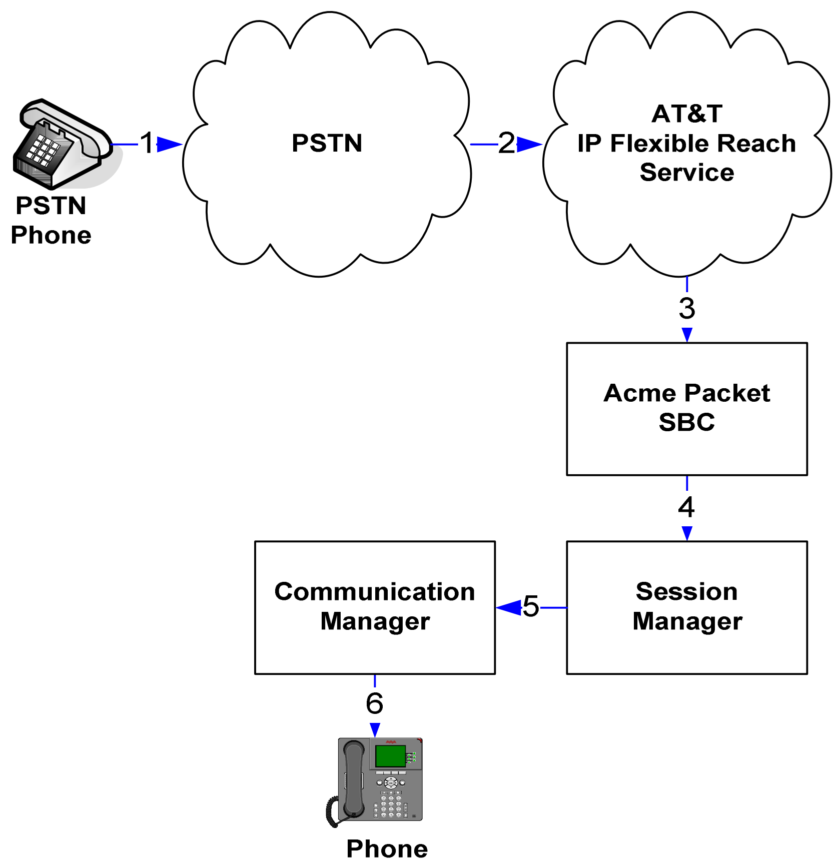


Figure 2: Inbound AT&T IP Flexible Reach Call

3.2.2. Outbound

The second call scenario illustrated in **Figure 3** is an outbound call initiated on Communication Manager, routed to Session Manager and is subsequently sent to the Acme SBC for delivery to AT&T IP Flexible Reach service.

1. Communication Manager phone or fax originates a call to an AT&T IP Flexible Reach service number for delivery to PSTN.
2. Communication Manager routes the call to Session Manager.
3. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager 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 the AT&T IP Flexible Reach service.
5. The AT&T IP Flexible Reach service delivers the call to PSTN.
6. PSTN delivers the call to PSTN Phone.

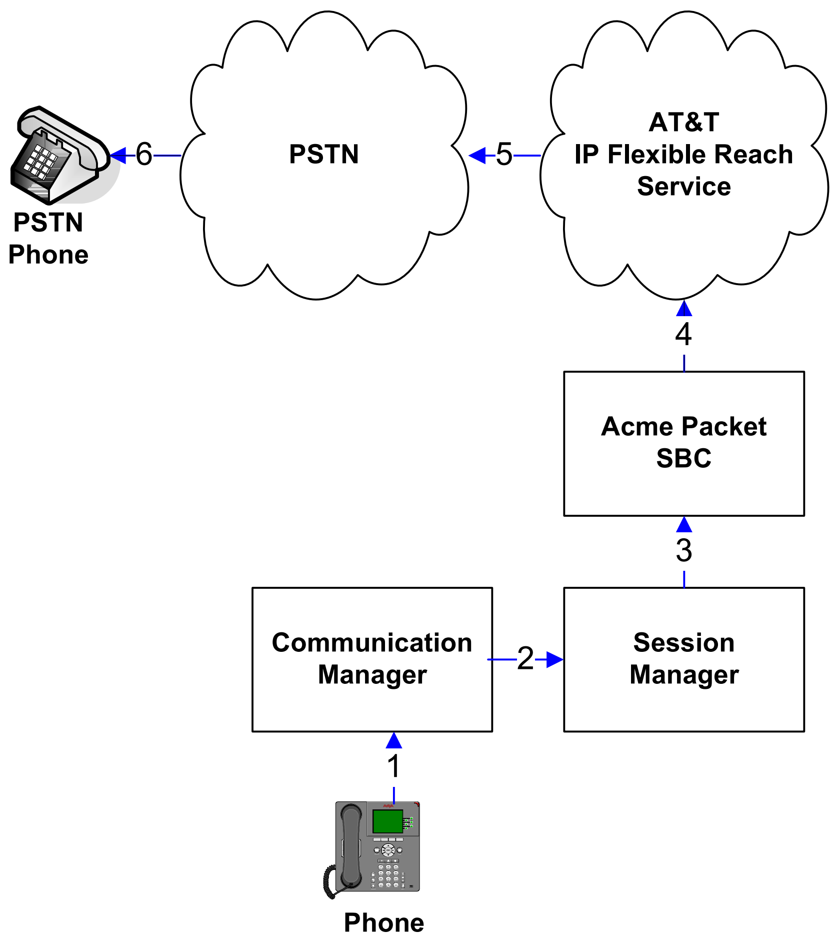


Figure 3: Outbound AT&T IP Flexible Reach Call

3.2.3. Call Forward Re-direction (Diversion Header)

The third call scenario illustrated in **Figure 4** is an inbound AT&T IP Flexible Reach service call that arrives on Session Manager and subsequently Communication Manager. Communication Manager routes the call to a destination station, however the station has set Call Forwarding to an alternate destination. Without answering the call, Communication Manager immediately redirects the call back to the AT&T IP Flexible Reach service for routing to the alternate destination.

1. Same as the first call scenario in **Section 3.2.1**.
2. Because the Communication Manager phone has set Call Forward to another AT&T IP Flexible Reach service number, Communication Manager initiates a new call back out to Session Manager, the Acme Packet SBC, and to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service places a call to the alternate destination and upon answer, Communication Manager connects the calling party to the target party.

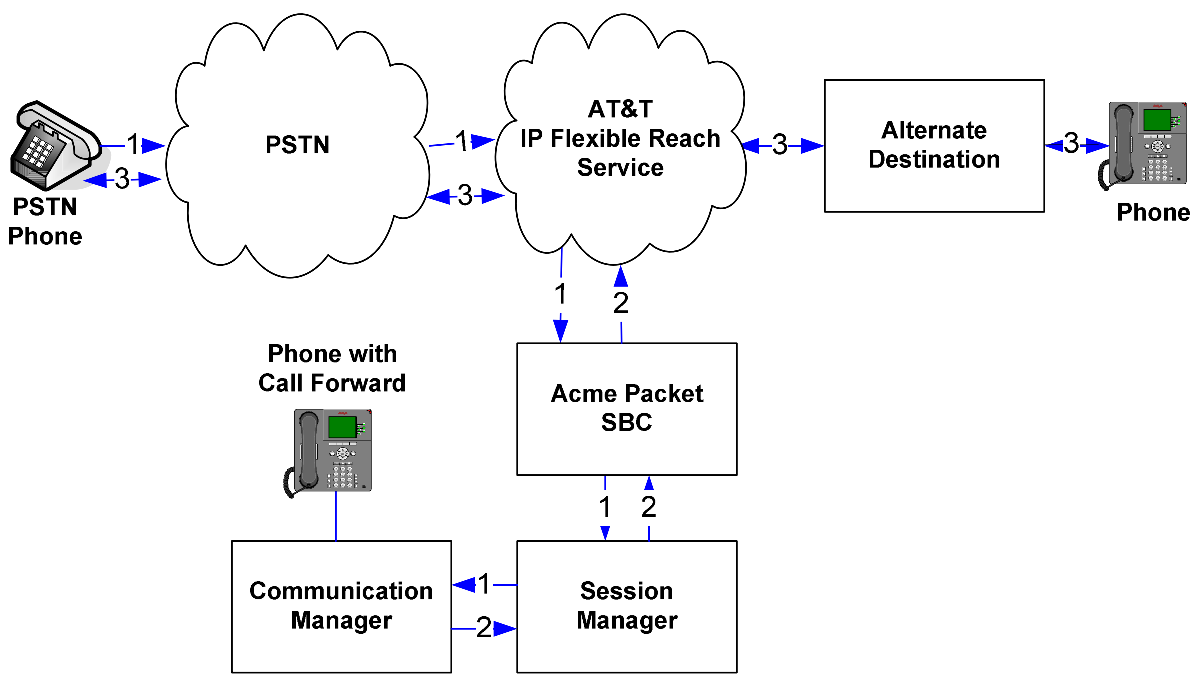


Figure 4: Re-directed (e.g., Call Forward) AT&T IP Flexible Reach Call

3.2.4. Coverage to Voicemail

The call scenario illustrated in **Figure 5** is an inbound call that is covered to voicemail. In this scenario, the voicemail system is a CM Messaging system connected to Session Manager. Note that this call scenario was not executed but is expected to work.

1. Same as the first call scenario in **Section 3.2.1**.
2. The called Communication Manager phone does not answer the call, and the call covers to the phone's voicemail. Communication Manager forwards[3] the call to Session Manager.
3. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Network Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to CM Messaging. CM Messaging answers the call and connects the caller to the called phone's voice mailbox. Note that the call[4] continues to go through Communication Manager.

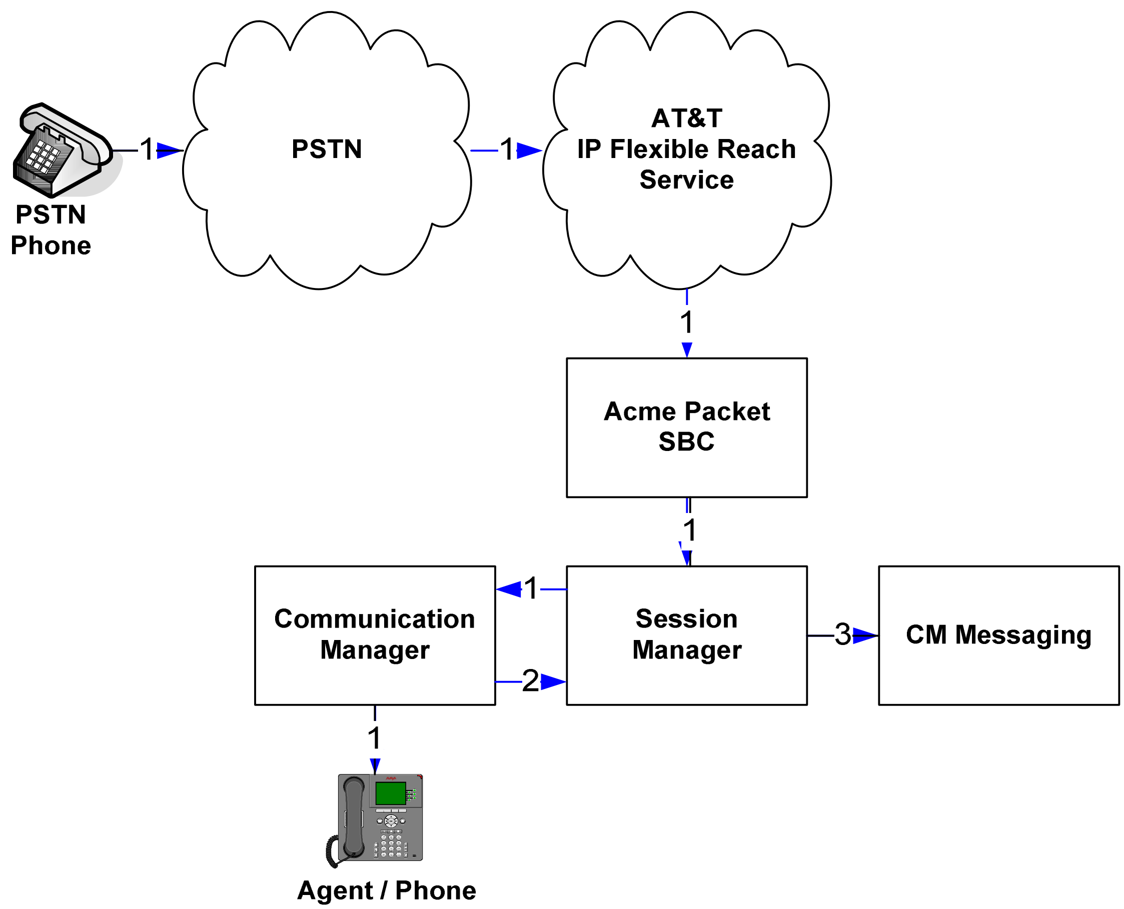


Figure 5: Coverage to Voicemail

[3] Communication Manager places a call to CM Messaging, and then connects the inbound caller to CM Messaging. SIP redirect methods, e.g., 302, are not used.

[4] The SIP signaling path still goes through Communication Manager. In addition, since the inbound call and CM Messaging use different codecs (G.729 and G.711, respectively), Communication Manager performs the transcoding, and thus the RTP media path also goes through Communication Manager.

3.2.5. AT&T IP Flexible Reach - Enhanced Features – Network Based Blind Transfer Using Refer (Communication Manager Vector) Call Flow

This section describes the call flow used for AT&T IP Flexible Reach-Enhanced Features service which uses SIP-Refer method for off-net blind transfers. The call scenario illustrated in figure below is an inbound AT&T IP Flexible Reach service call that arrives on Session Manager and is subsequently routed to Communication Manager, which in turn routes the call to a vector. The vector answers the call and then redirects the call back to the AT&T IP Flexible Reach service for routing to an alternate destination.

1. Same as the first call scenario in **Section 3.2.1**.
2. Communication Manager routes the call to a vector, which answers the call and plays an announcement, and attempts to redirect the call by sending a SIP REFER message back out on the SIP trunk on which the inbound call arrived. The SIP REFER message specifies the alternate destination, and is routed back through Session Manager and then the Acme Packet SBC to the AT&T IP Flexible Reach service.
3. The AT&T IP Flexible Reach service places a call to the target party (alternate destination) and upon answer, connects the calling party to the target party.
4. The AT&T IP Flexible Reach service clears the call on the referring party (Communication Manager).

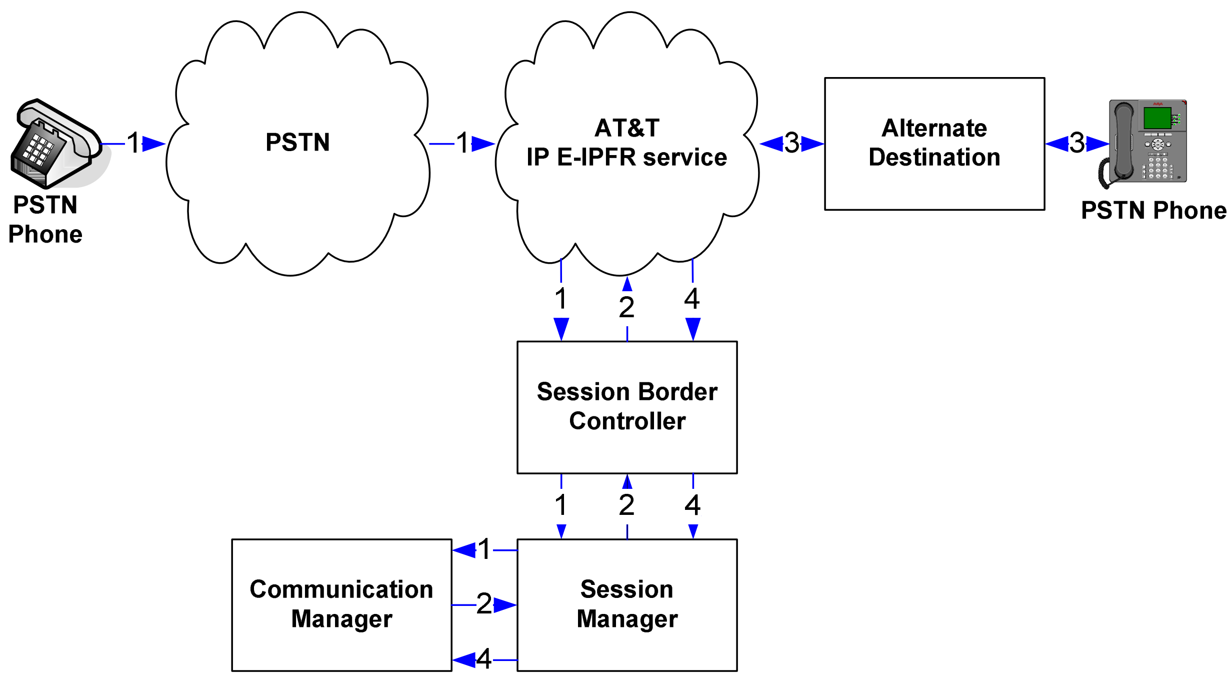


Figure 6: Inbound AT&T IP Flexible Reach – Post-Answer SIP REFER Redirection Call

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 (6.3.0.0.630039)
Avaya S8800 Server	Avaya Aura® Communication Manager 6.3 SP1 with CM Messaging (R016X.03.0.124.0 with patch 20850) System Platform 6.3.0.0.18002
Avaya G650 Media Gateway	
TN2312BP IP Server Interface (IPSI)	HW03 FW057
TN799DP Control-LAN (C-LAN)	HW01 FW041
TN2602AP IP Media Resource 320 (MedPro)	HW02 FW062
TN2501AP VAL-ANNOUNCEMENT	HW03 FW018
TN2224CP Digital Line	HW08 FW015
TN793B Analog Line	000005
Avaya 9650 IP Telephone	H.323 R3.2.0
Avaya 9641G IP Telephone	H.323 R6.2.3.13
Avaya 9641G IP Telephone	SIP R6.2.1.26
Avaya 9630 IP Telephone	SIP R2.6.9.1
Avaya one-X® Communicator (H323/SIP)	6.1.8.08-SP8-40314
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 Flexible Reach-Enhanced Features service using AVPN/MIS-PNT transport service connection	VNI 26

Table 2: Equipment and Software Versions

5. Configure Avaya Aura® Session Manager Release

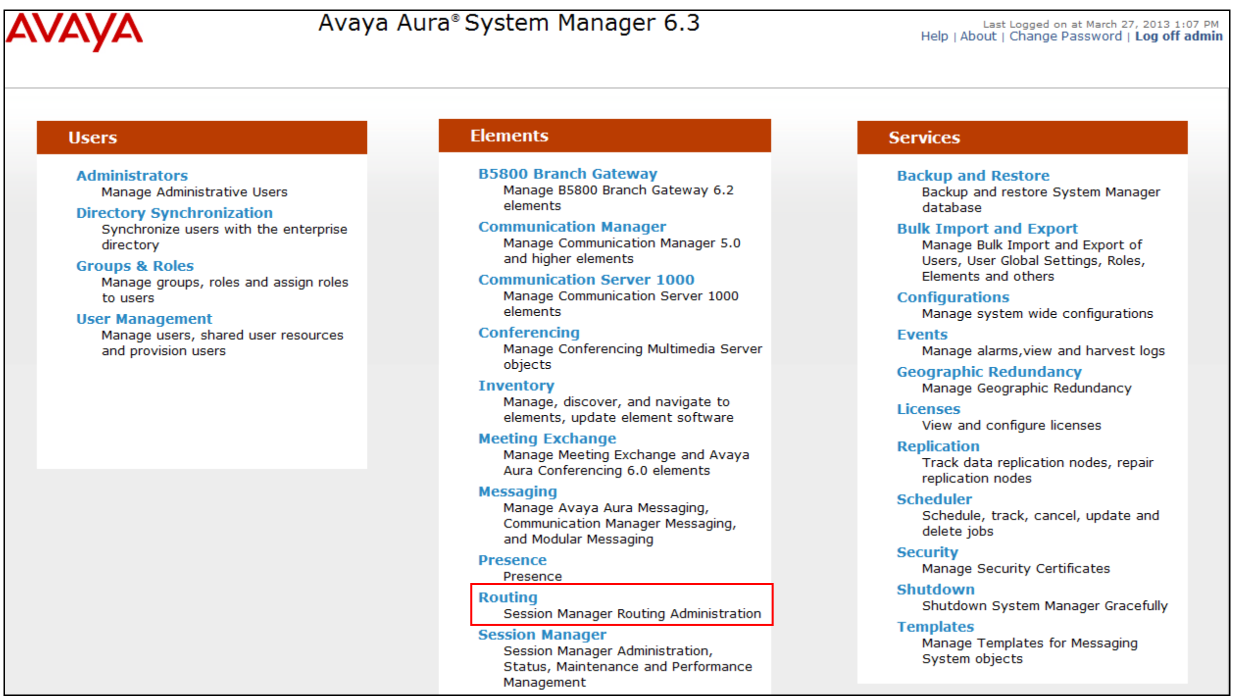
This section illustrates relevant aspects of the Session Manager configuration used in the verification of this compliance test solution for supporting AT&T IP Flexible Reach service. Some screens shown below may be abridged as only those parts of the screen were configured. For rest of the fields, the default values were used.

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

The following administration activities are described:

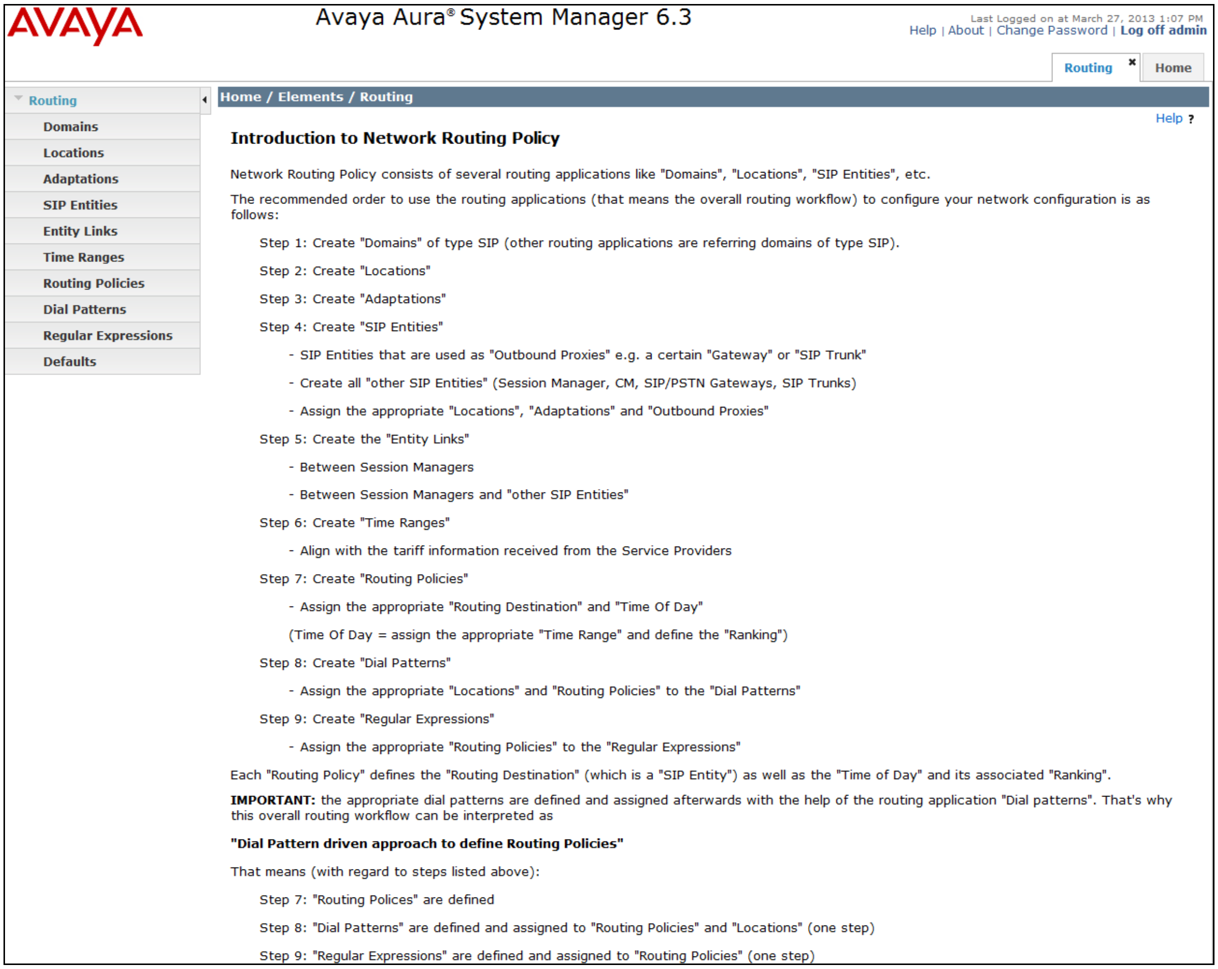
- Define SIP Domain
- Define Locations for routing purposes
- Configure the Adaptation Modules that are associated with various SIP Entities
- Define SIP Entities for Session Manager, Communication Manager, Acme Packet SBC, etc
- Define Entity Links between various SIP entities
- Define Routing Policies 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

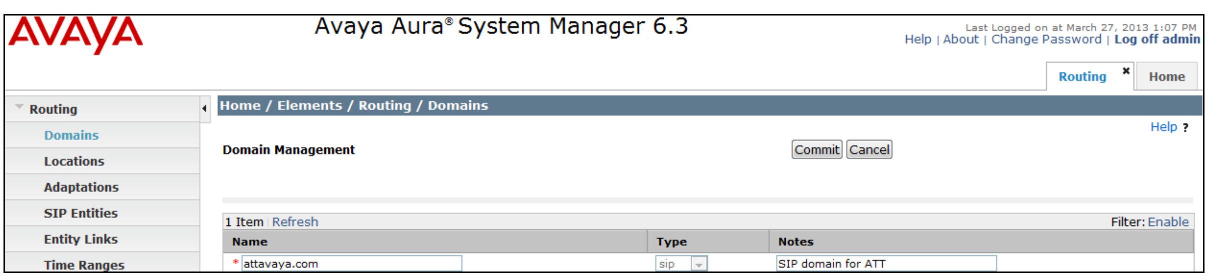
The screen below shows the various sub-headings with explanation of the left navigation menu that are referenced in this section.



Network Routing Policy Page

5.1. SIP Domain

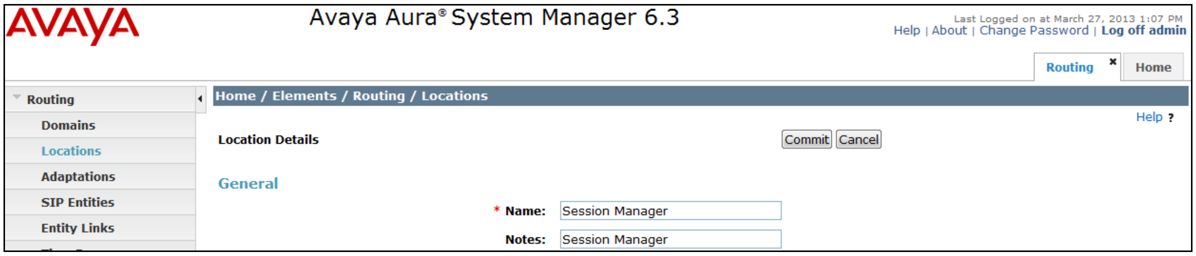
Navigate to **Routing→Domains** and click **New** (not shown). The following screen shows the domain used in this reference configuration.



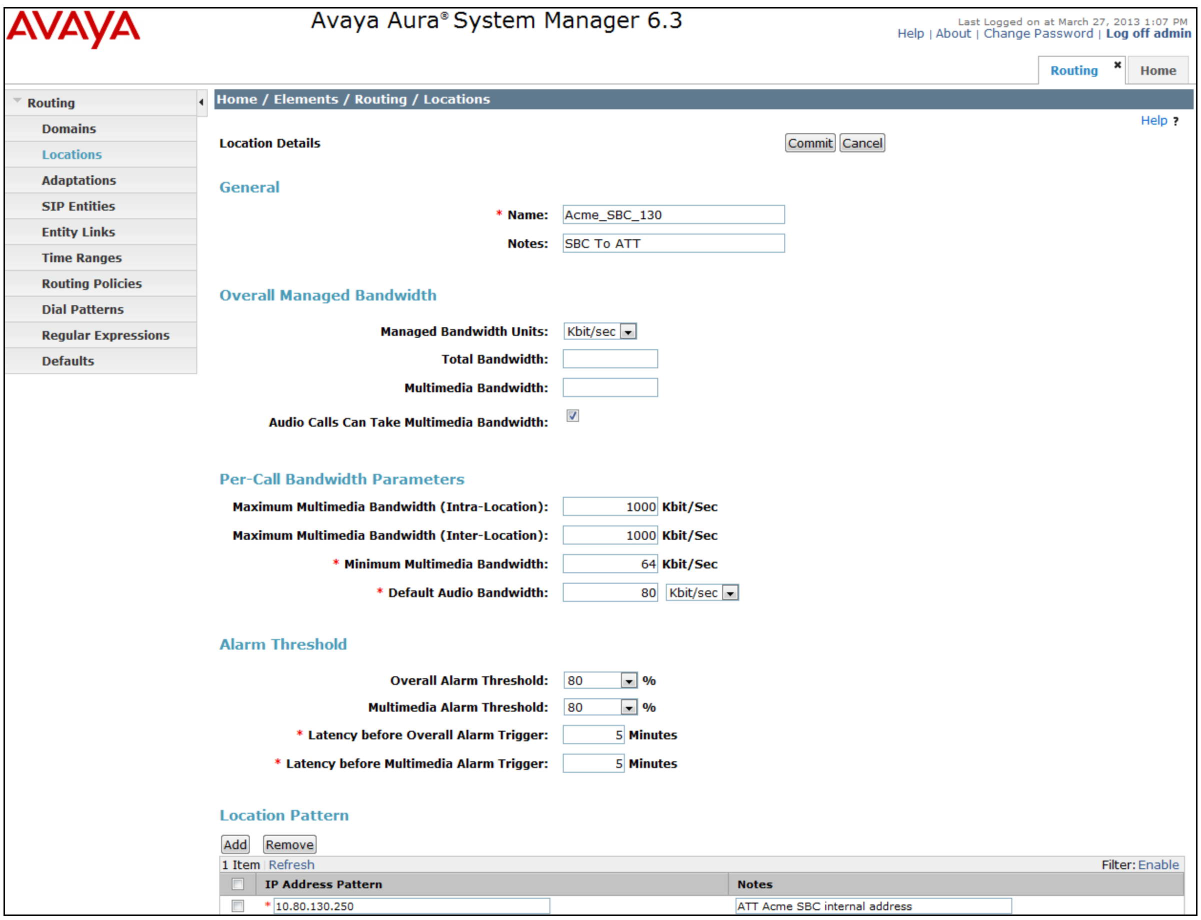
SIP Domains

5.2. Locations

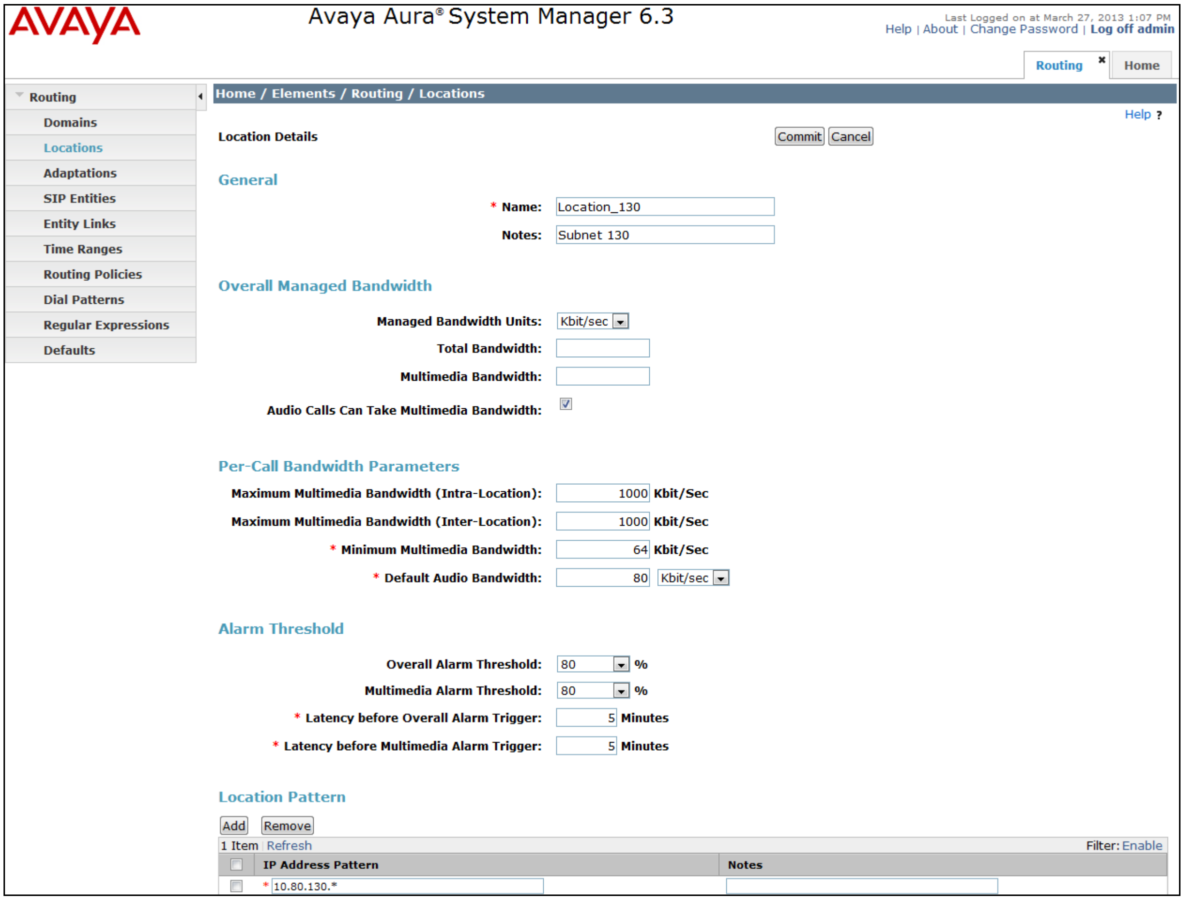
Navigate to **Routing→Locations** and click **New** (not shown). The following screens show Location Details for various locations used in this AT&T IP Flexible Reach service testing.



Session Manager Location Details



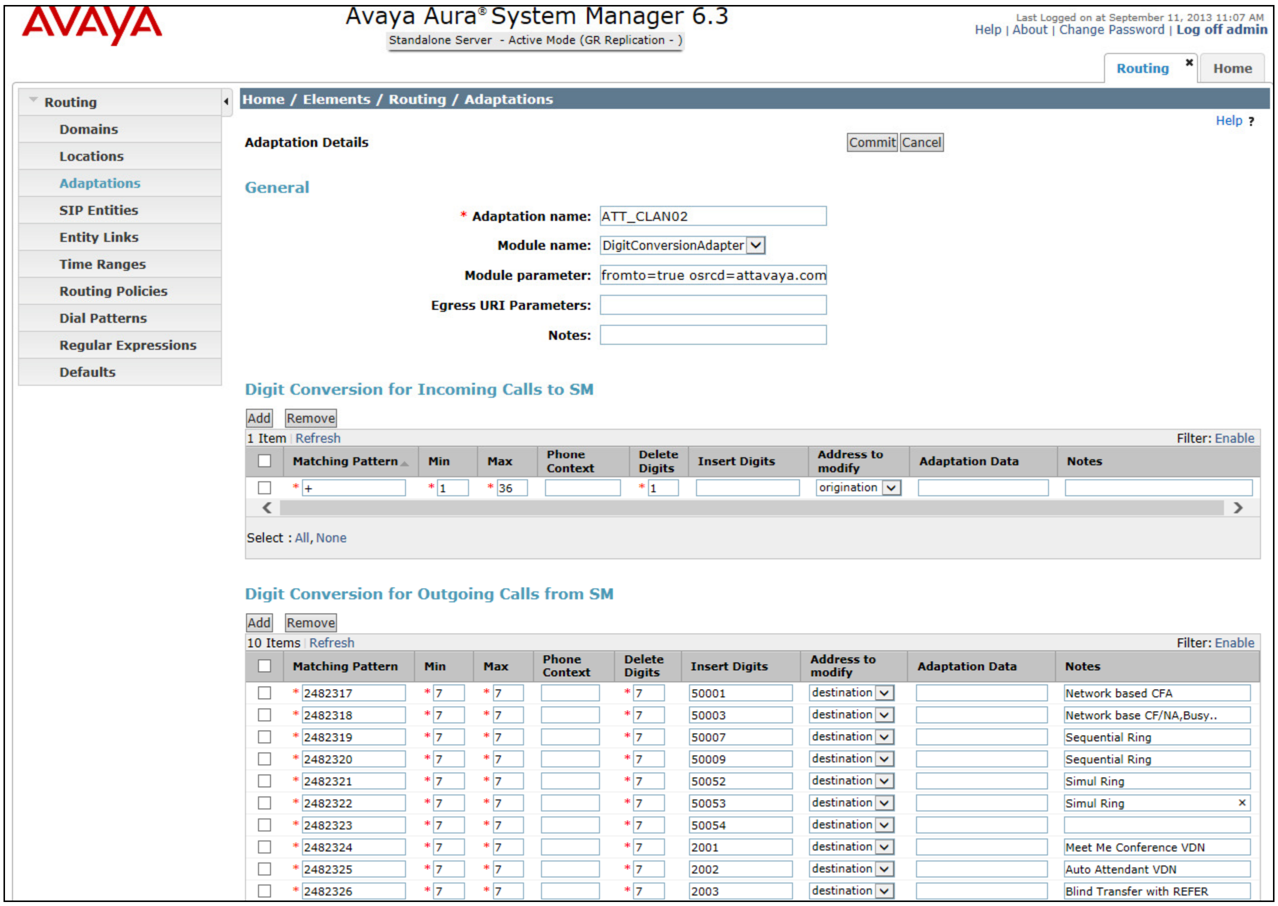
Acme Packet SBC Location Details



Subnet 130 Location Details

5.3. Configure Adaptations

The following screen displays the adaptations used for inbound calls to support AT&T IP Flexible Reach service along with Enhanced Features like Simultaneous and Sequential ring. In this reference configuration, DID **2482321** was used for simultaneous ring feature where an INVITE is sent to both extensions **50052** and **50053** and DID **2482319** was used for sequential ring feature where extension **50007** rings first and if not answered extension **50009** will ring. Additionally, DID **2482317** and **2842318** were used for both Network based Call Forwarding features. DID **2482326** was used to adapt to invoke Refer method on Communication Manager as described in **Section 6.6.2** to transfer calls off-net. All the DIDs were used for basic inbound calls.

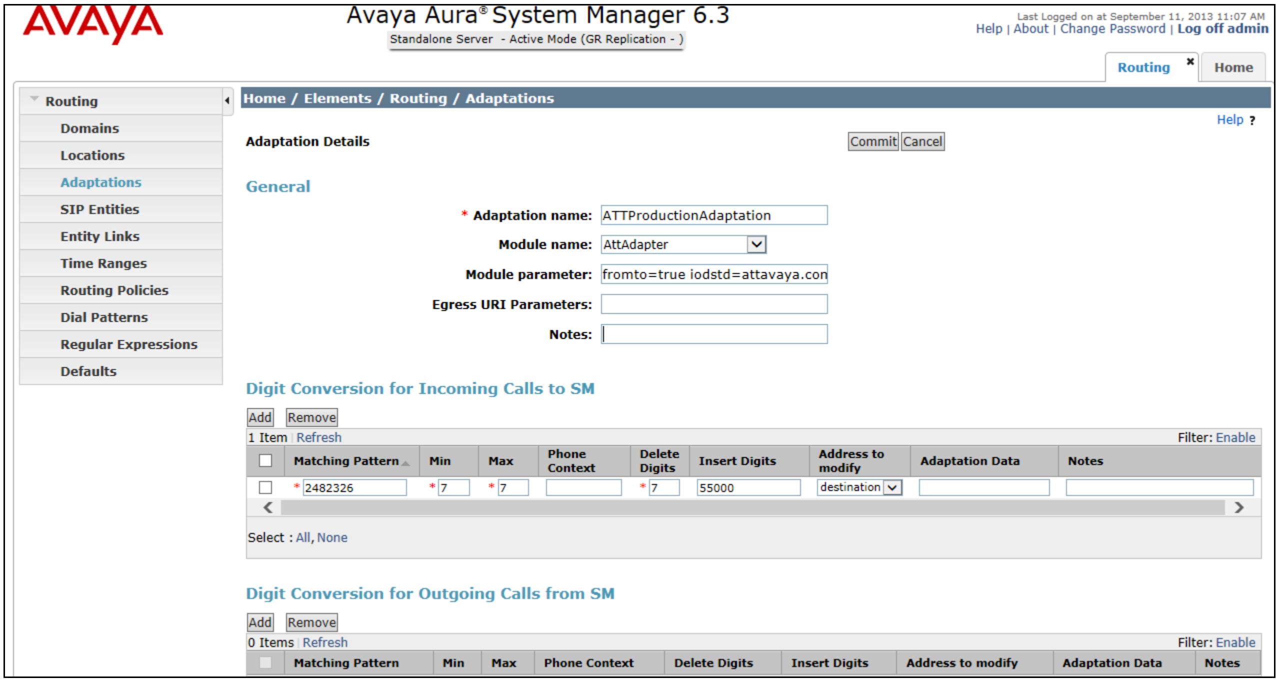


Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
+	1	36		1		origination		

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
2482317	7	7		7	50001	destination		Network based CFA
2482318	7	7		7	50003	destination		Network base CF/NA,Busy..
2482319	7	7		7	50007	destination		Sequential Ring
2482320	7	7		7	50009	destination		Sequential Ring
2482321	7	7		7	50052	destination		Simul Ring
2482322	7	7		7	50053	destination		Simul Ring
2482323	7	7		7	50054	destination		
2482324	7	7		7	2001	destination		Meet Me Conference VDN
2482325	7	7		7	2002	destination		Auto Attendant VDN
2482326	7	7		7	2003	destination		Blind Transfer with REFER

Communication Manager Adaptations

The following screen shows the adaptation used for outbound calls to AT&T IP Flexible Reach service. The **Module parameter** field is set to **fromto=true iodstd=attavaya.com osrcd=192.252.35.202** (IP Address of the external interface of Acme Packet SBC) **odstd=135.194.131.41** (IP Address of AT&T IP Flexible Reach Border Element). Additionally, the digit conversion is done on one of the DIDs to map to the Pilot Number for the CM Messaging system to retrieve messages.

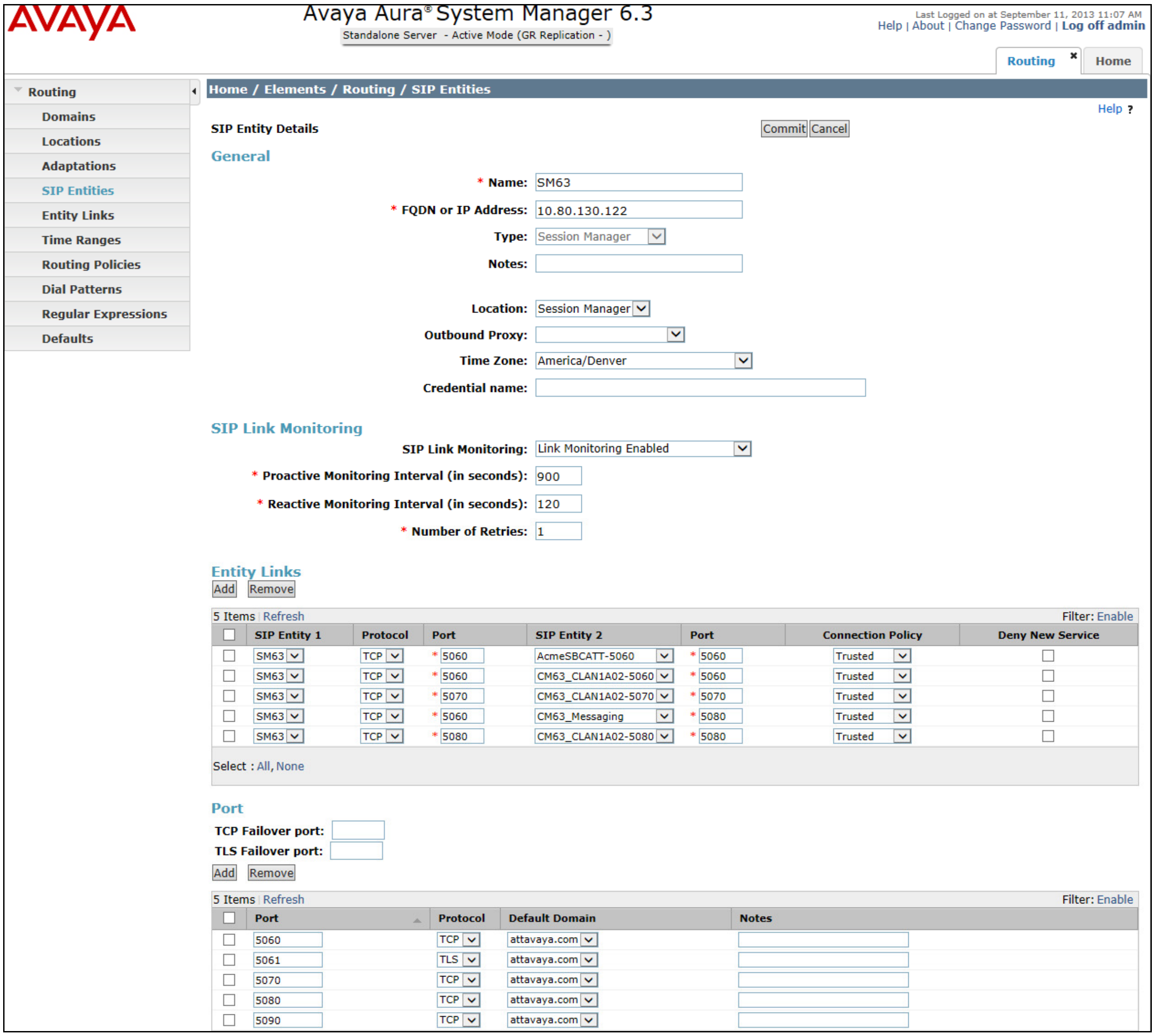


Acme Packet SBC Adaptation

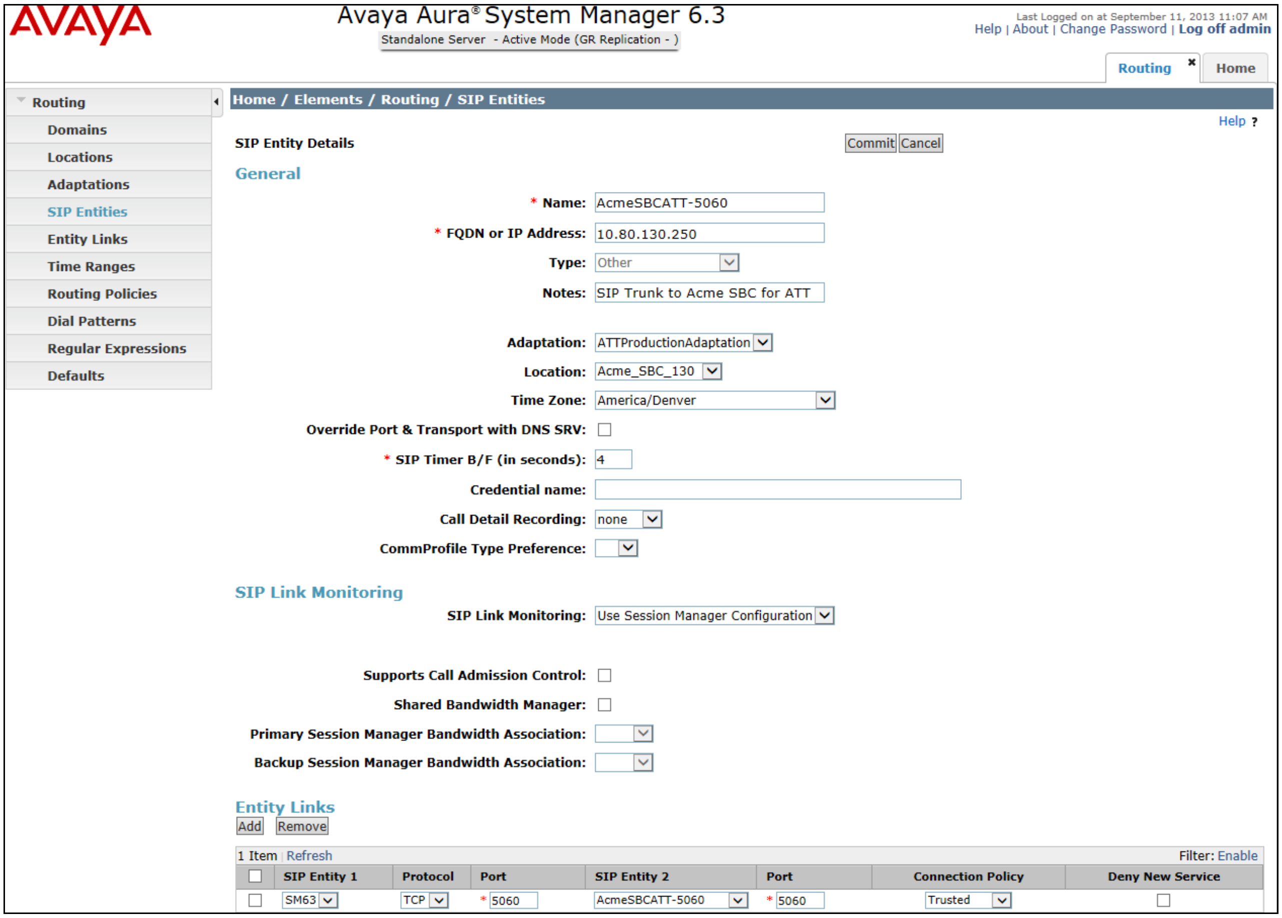
5.4. SIP Entities

The following screens show the entities along with Entity links configured for AT&T IP Flexible Reach service. 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 when possible.

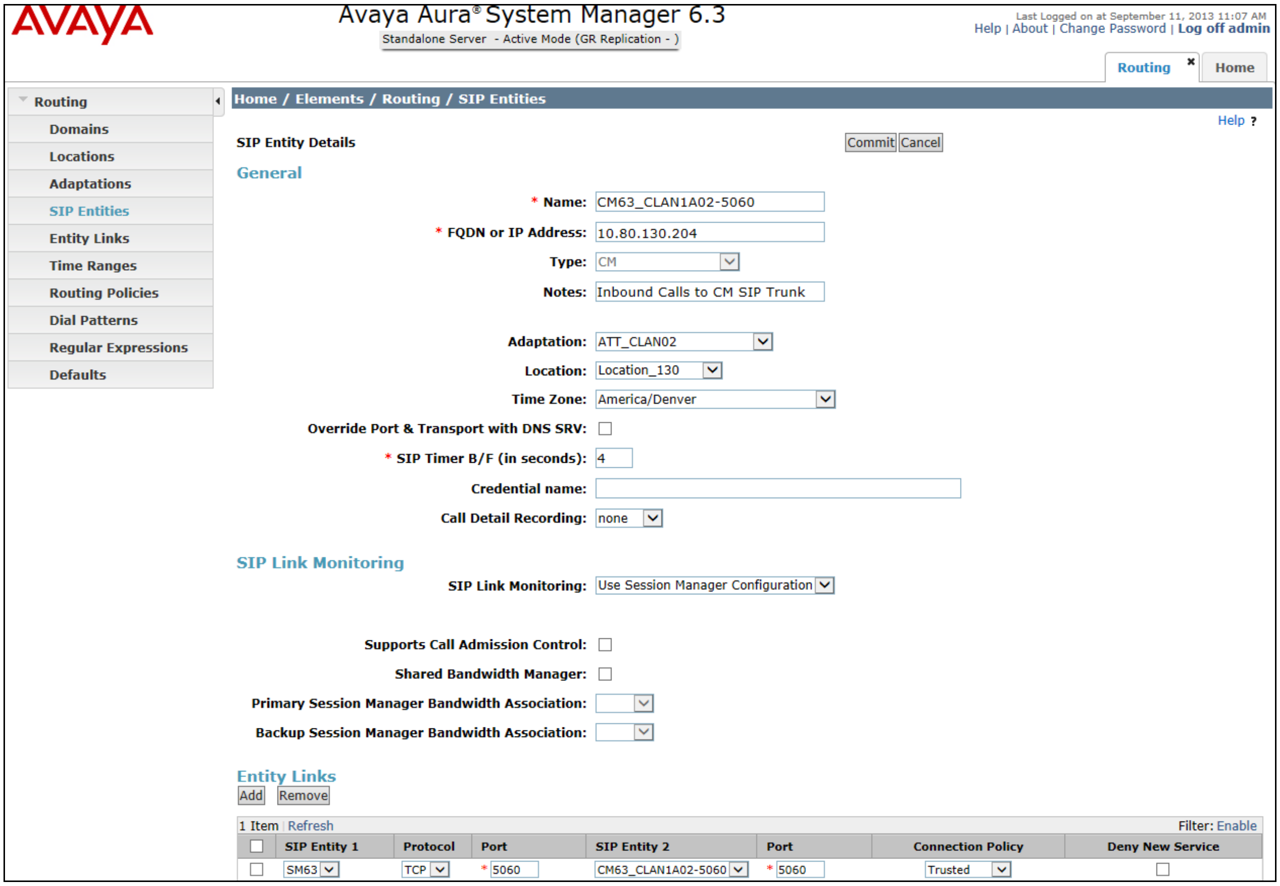


Session Manager Entity



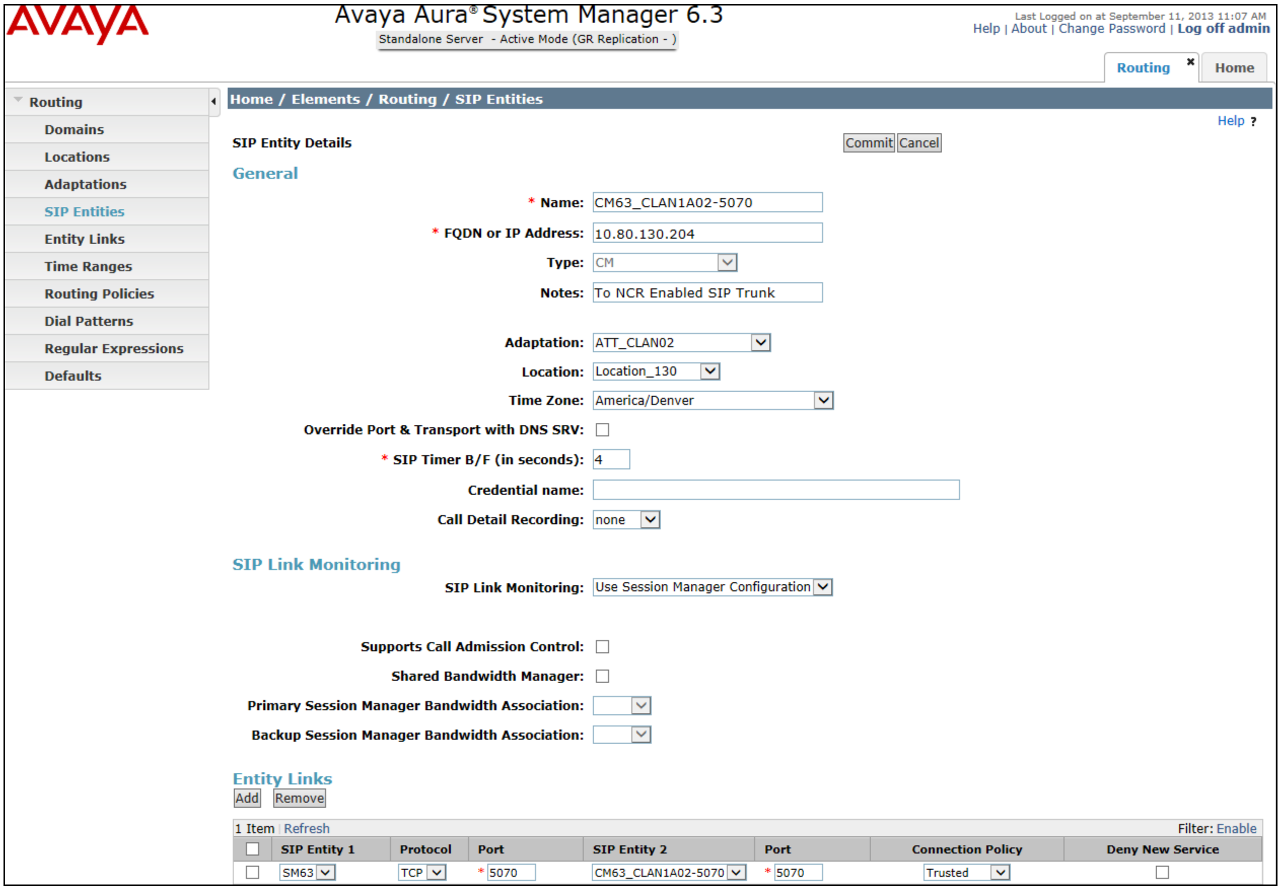
Acme Packet SBC Entity

The following screen shows SIP Entity configured for the Communication Manager trunk group with NCR disabled. See **Section 2.2** (**Items 2**, **4**) and **Section 6.6.1** for further details.



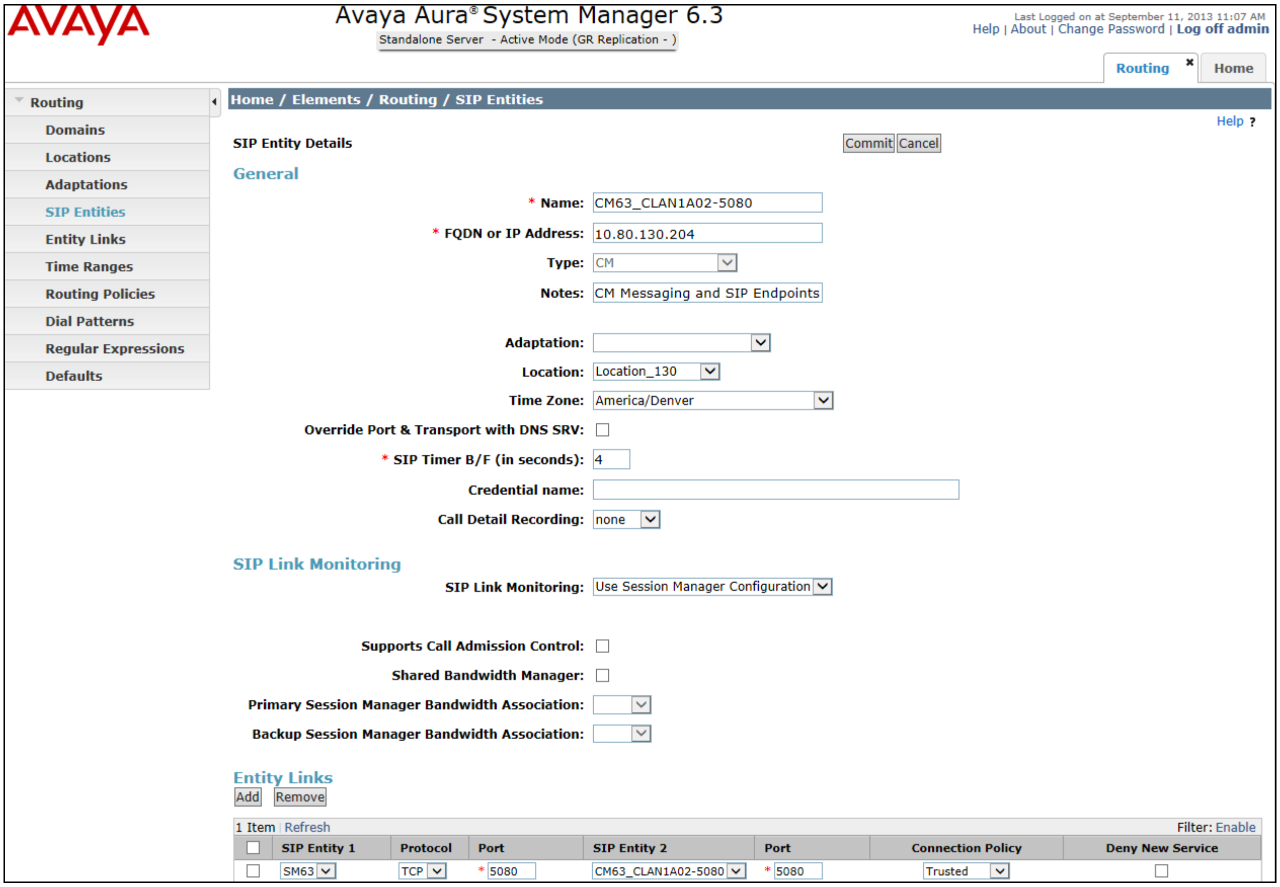
Communication Manager Entity (CM63_CLAN1A02-5060)

The following screen shows SIP Entity configured for the Communication Manager trunk group with NCR enabled. See **Section 2.2** (**Items 2**, **4**) and **Section 6.6.2** for further details.



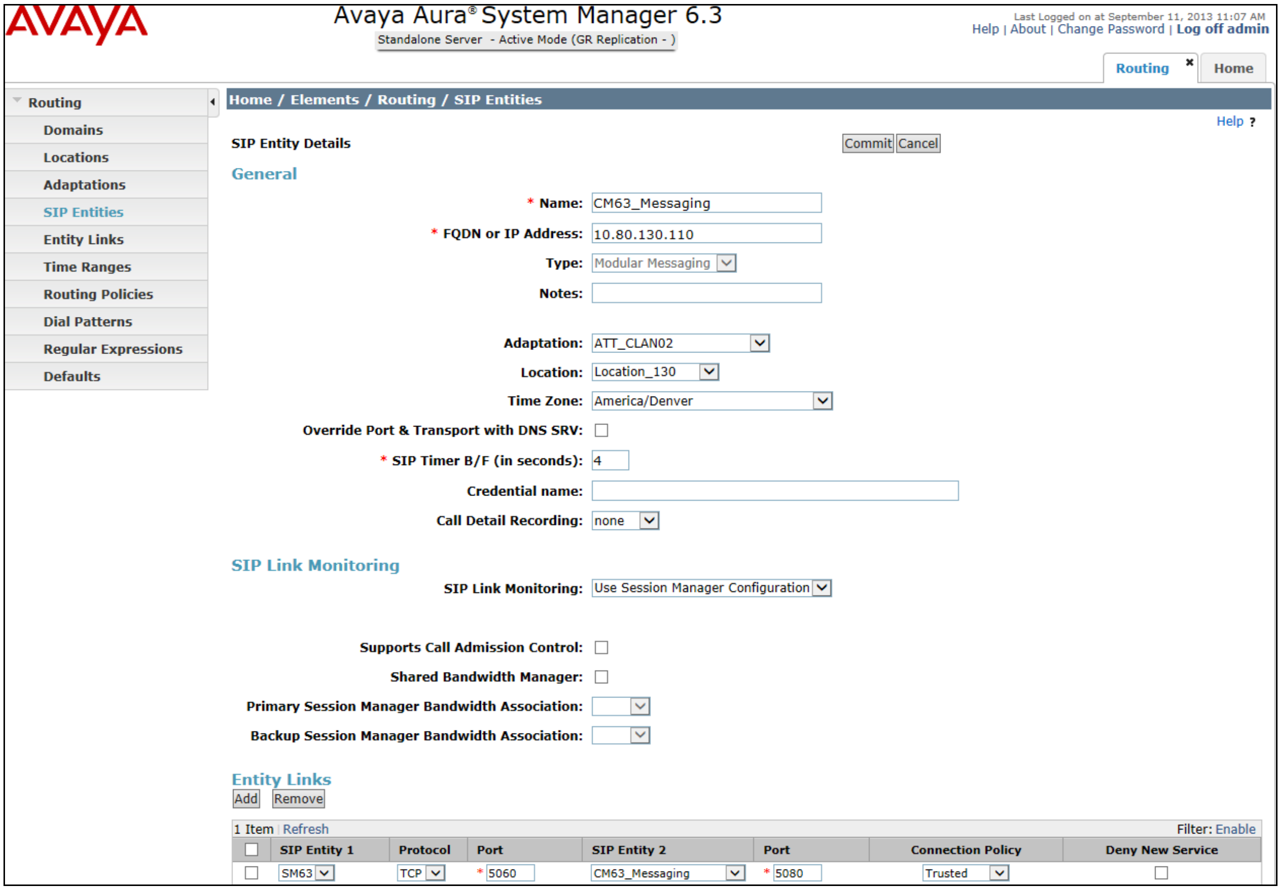
Communication Manager Entity (CM63_CLAN1A02-5070)

The following screen shows SIP Entity configured for the Communication Manager trunk group used for CM Messaging and SIP endpoints. See **Section 6.6.3** for the trunk configuration from Communication Manager to Session Manager to support the messaging functionality.



Communication Manager Entity (CM6.3CLAN1A02-5080)

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.

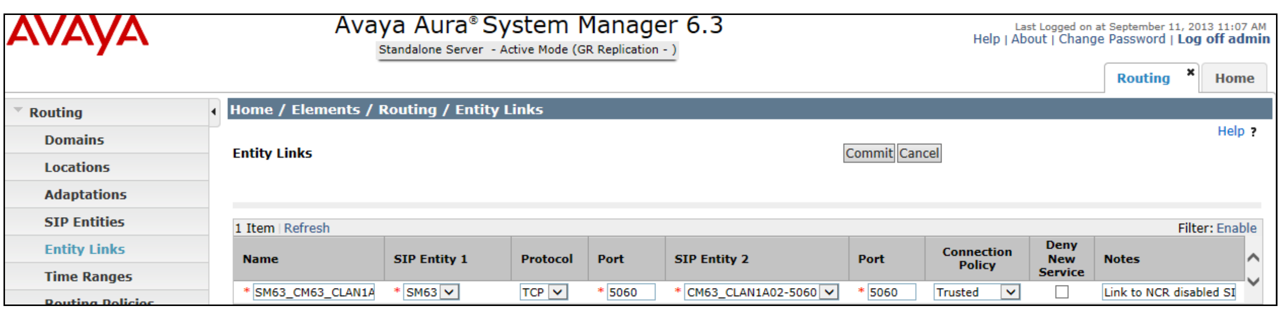


CM Messaging Entity

5.5. Entity Links

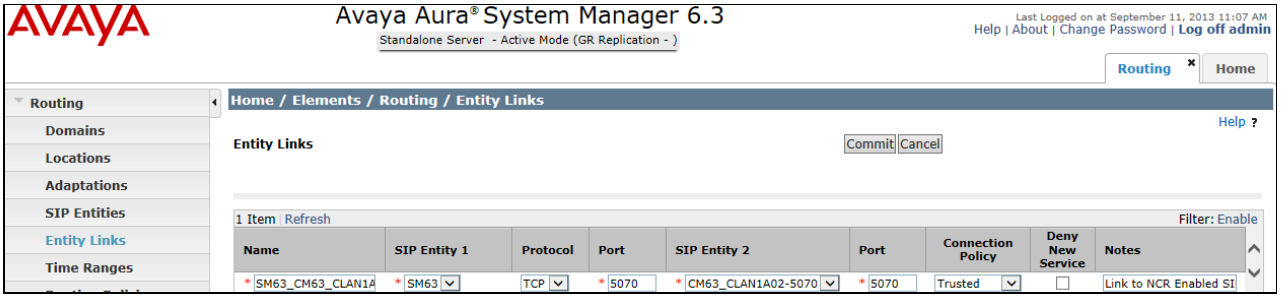
The following screens show the entity links configured for this reference configuration.

The screen below shows an Entity link configured for the Communication Manager trunk group with NCR disabled.

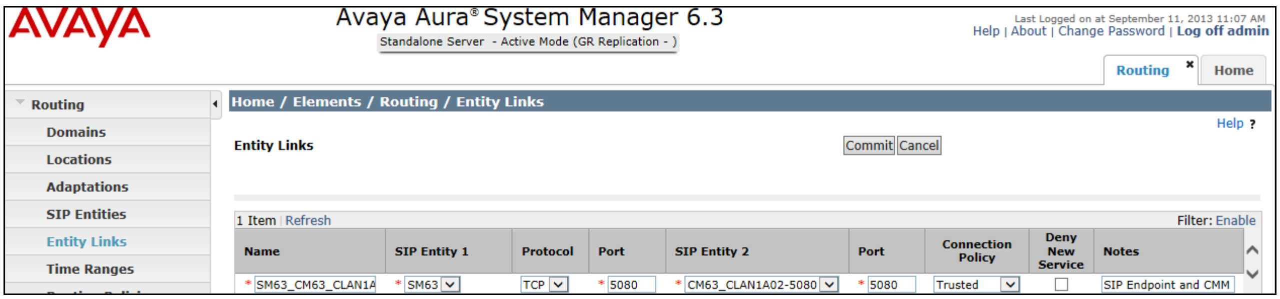


Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5060)

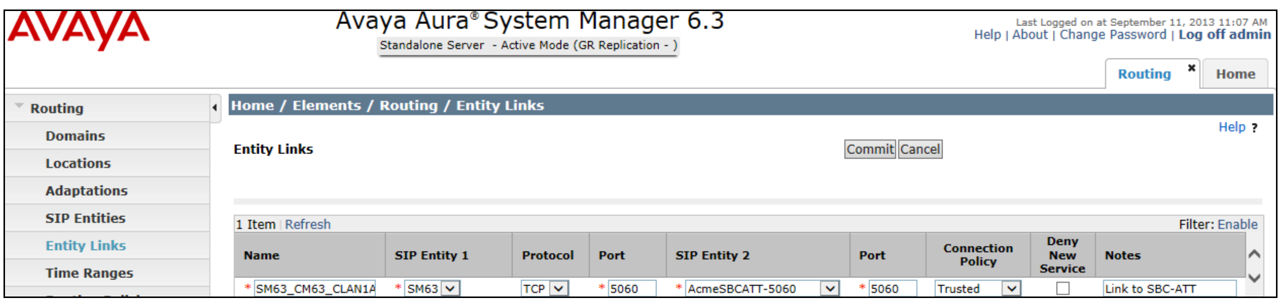
The screen below shows an Entity link configured for the Communication Manager trunk group with NCR enabled.



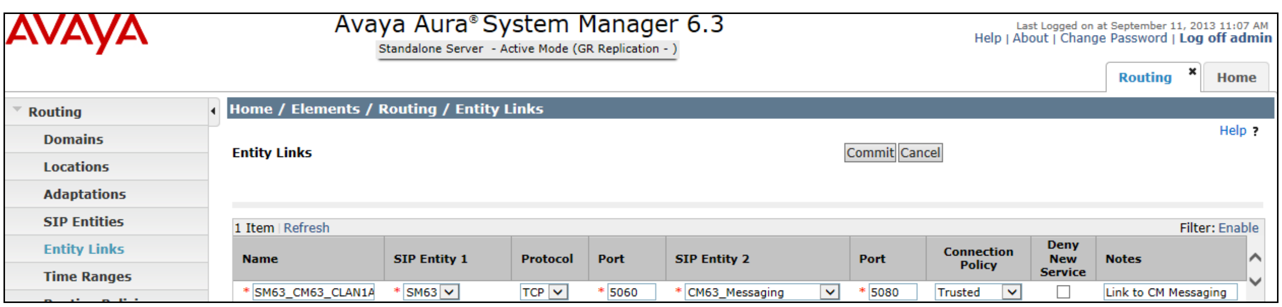
Entity link between Session Manager and Communication Manager (CLAN1A02, Port 5070)



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



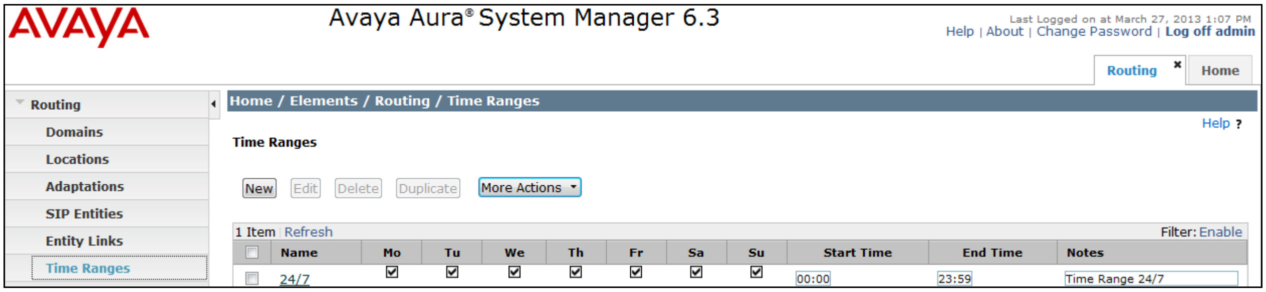
Entity link between Session Manager and Acme Packet SBC



Entity link between Session Manager and CM Messaging

5.6. Time Ranges

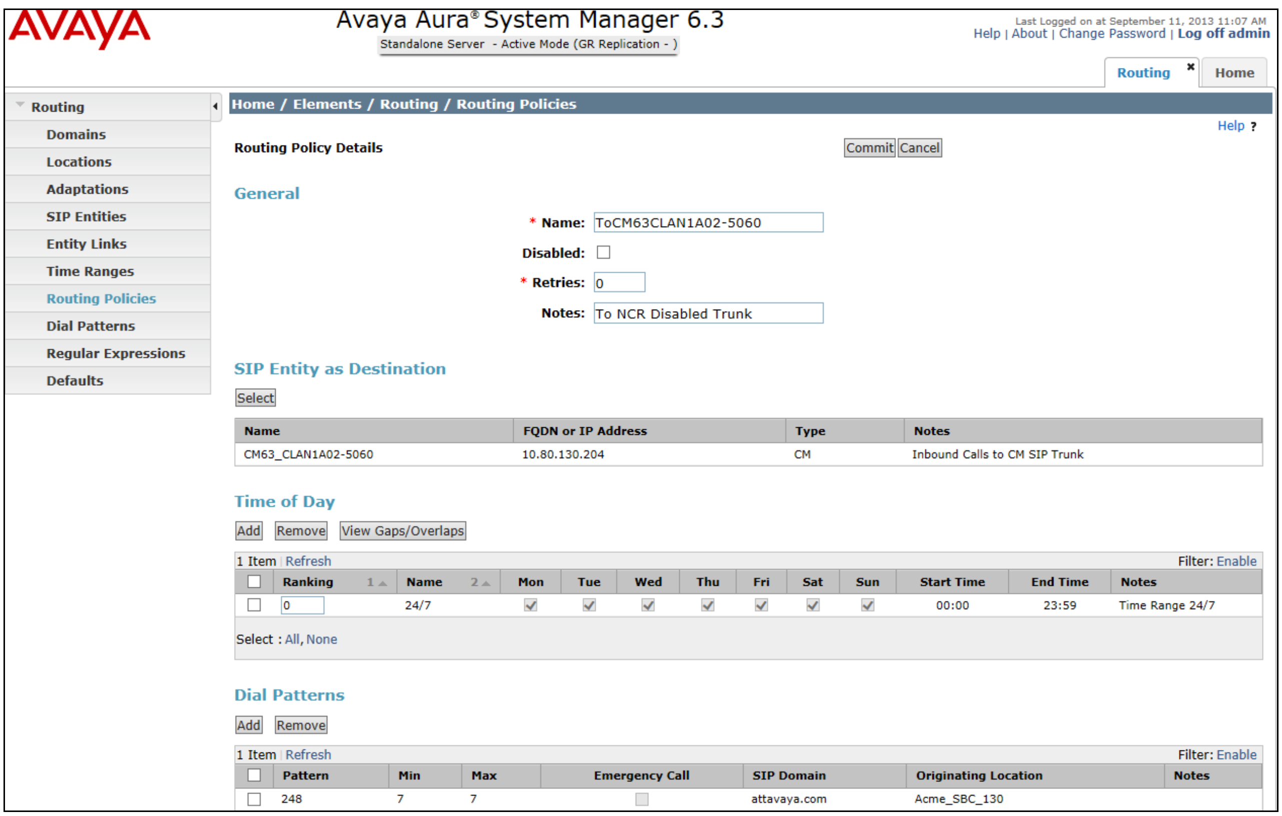
The following screen shows the time range used for AT&T IP Flexible Reach service testing.



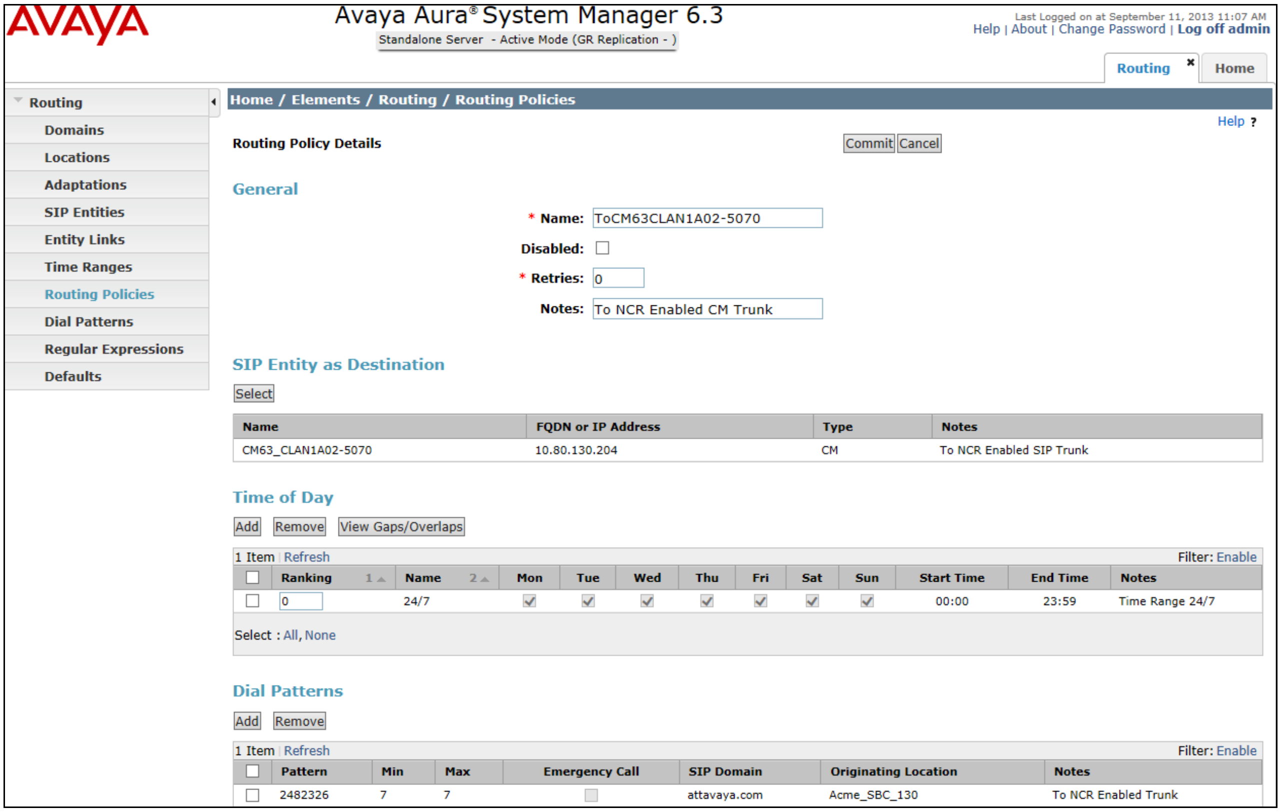
Time Range

5.7. Routing Policies

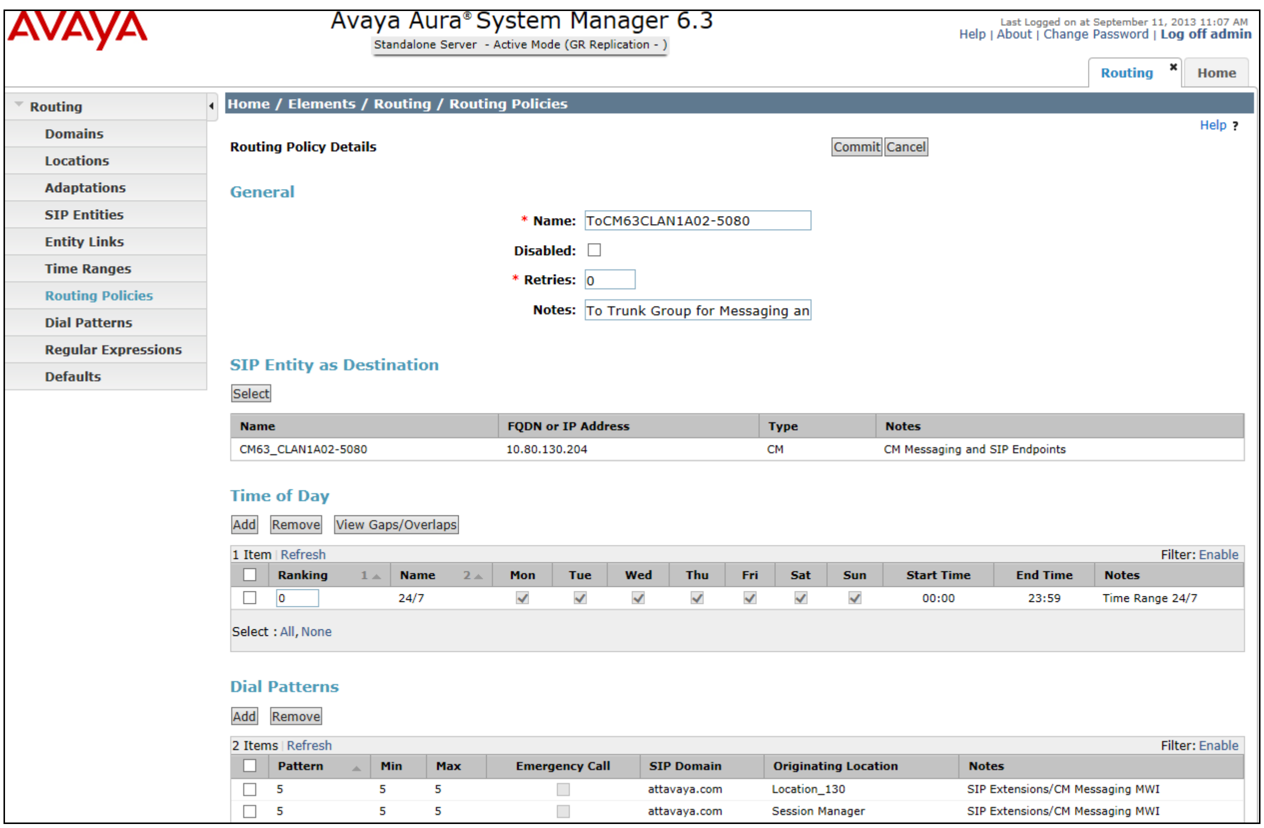
The following screens show routing policies along with dial patterns defined for AT&T IP Flexible Reach service. See **Section 5.8** for dial pattern configuration.



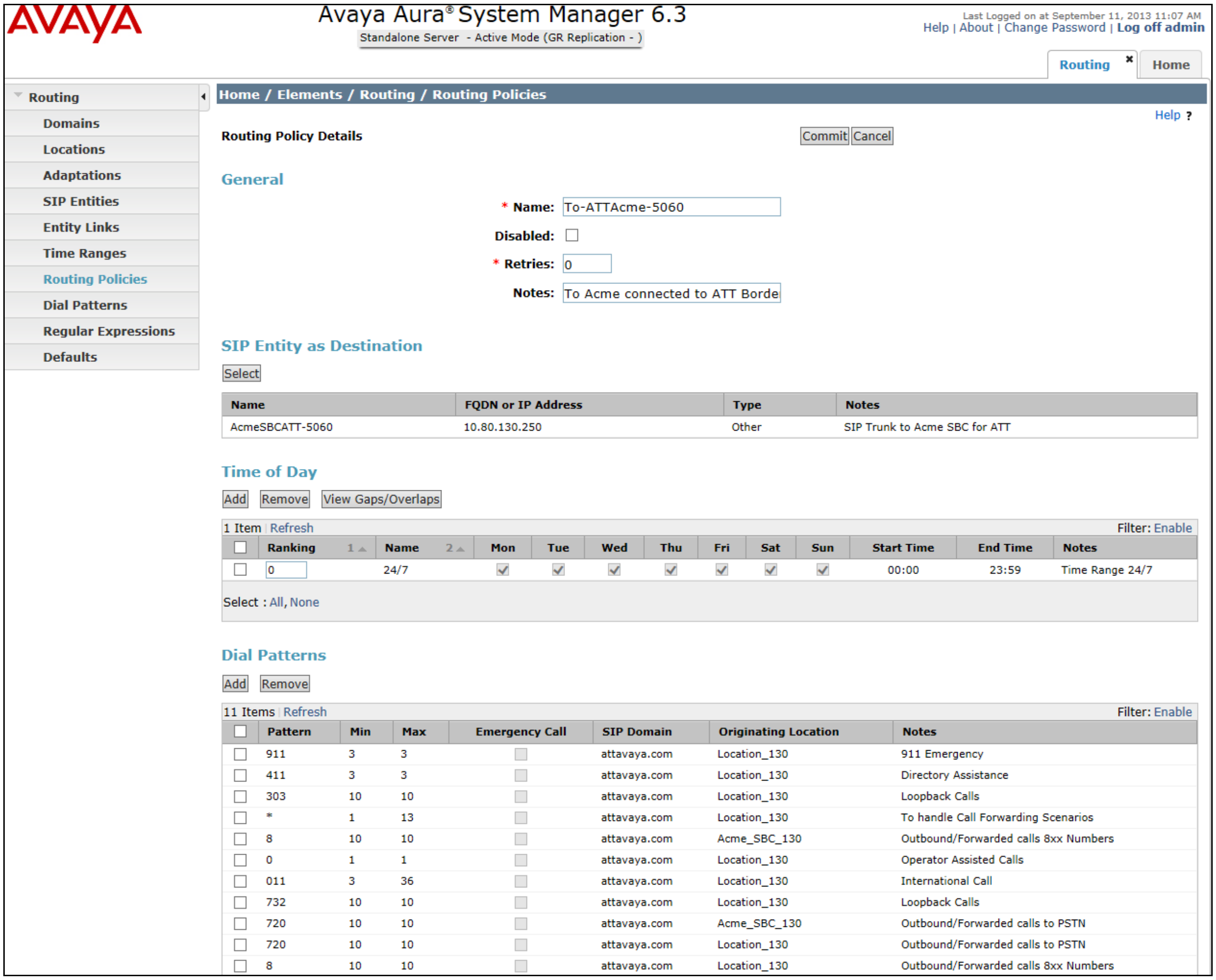
Routing Policy for Communication Manager (CLAN1A02-5060)



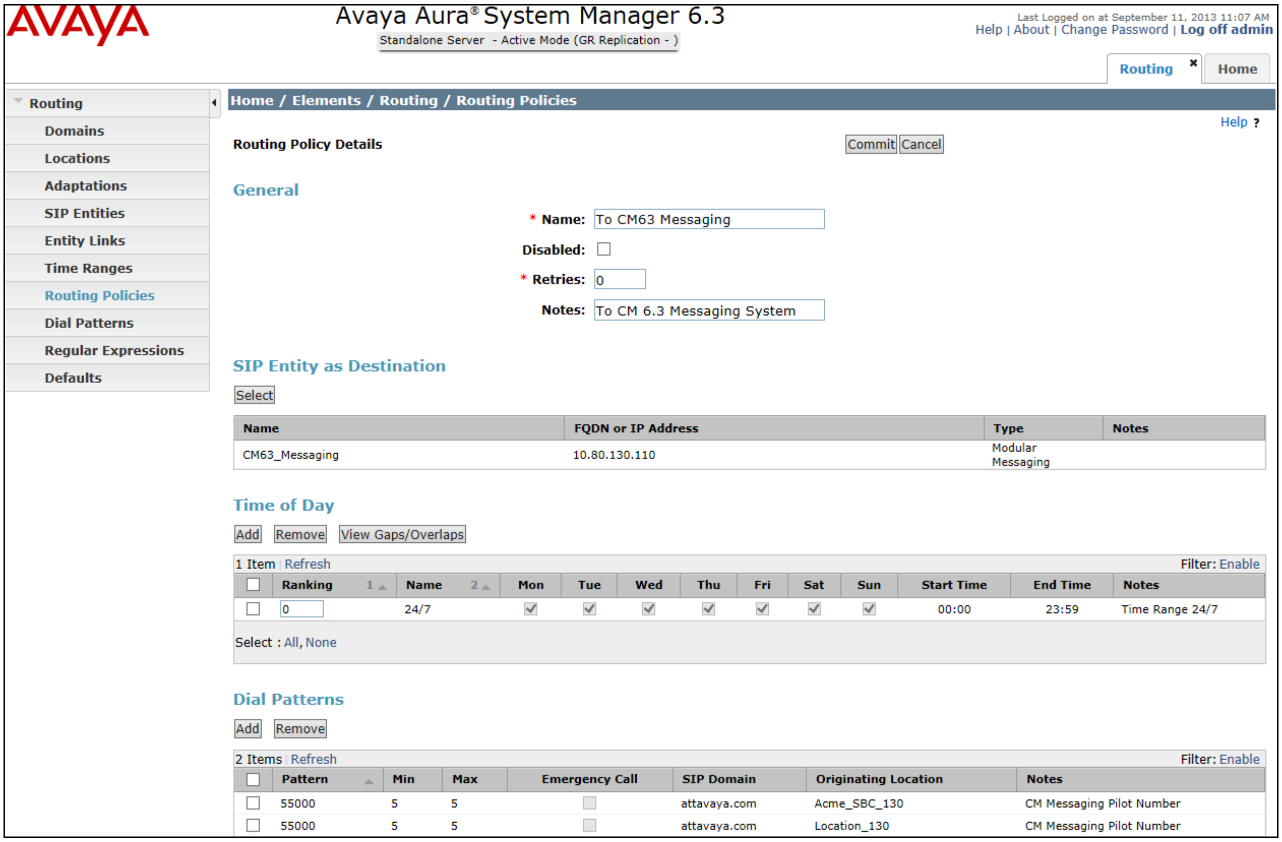
Routing Policy for Communication Manager (CLAN1A02-5070)



Routing Policy for Communication Manager (CLAN1A02-5080)



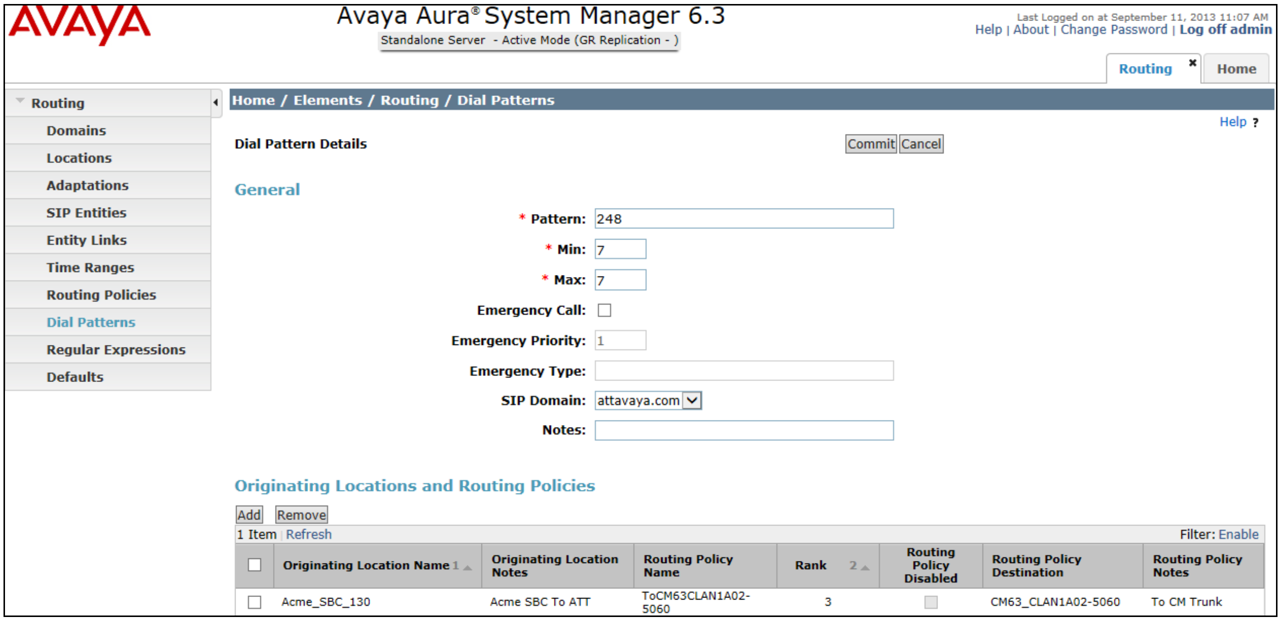
Routing Policy for Acme Packet SBC



Routing Policy for CM Messaging Pilot Number

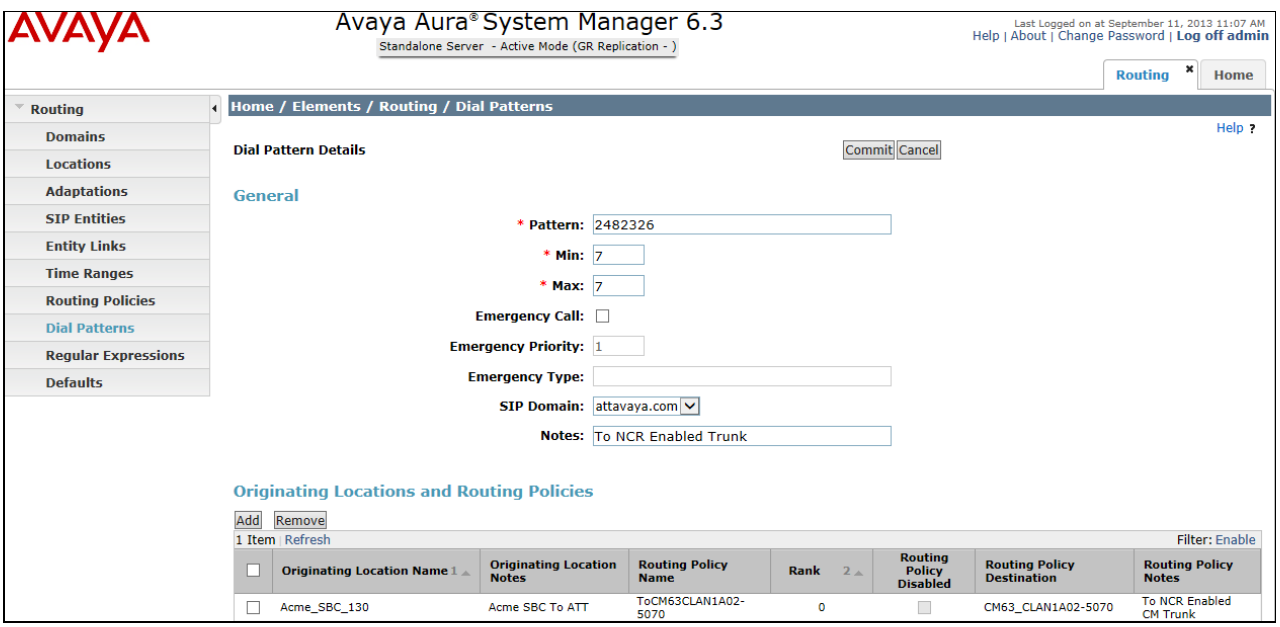
5.8. Dial Patterns

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

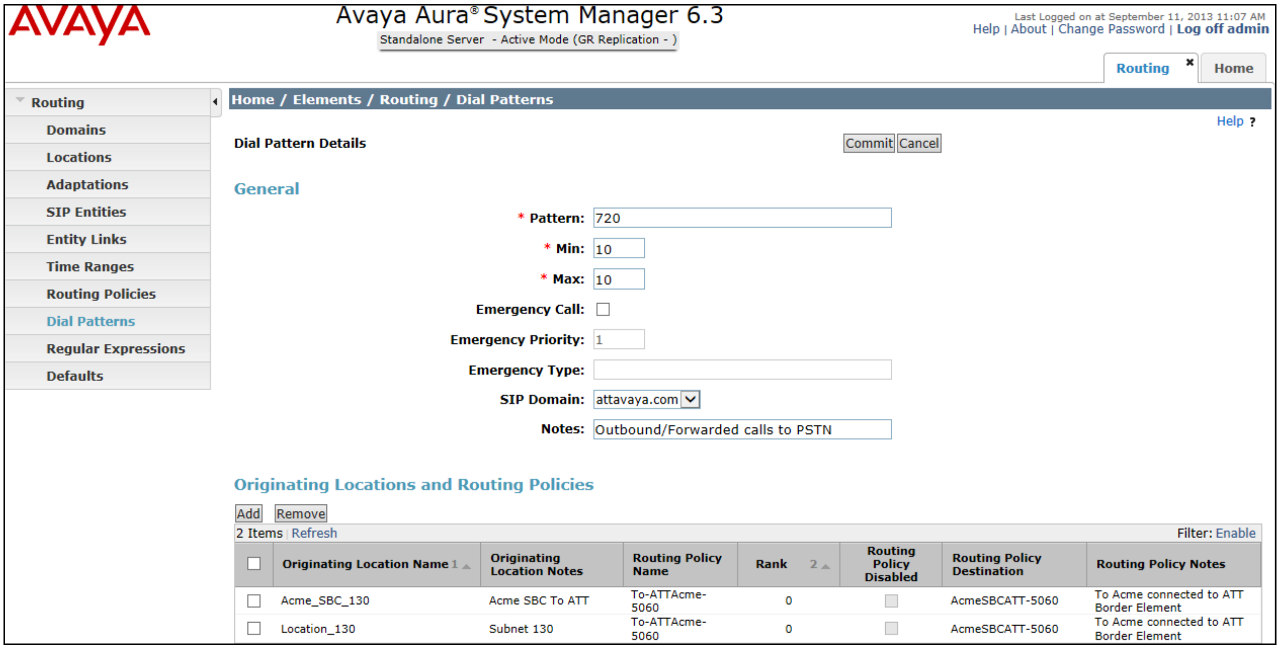


Dial Pattern for Inbound Calls to Communication Manager (CLAN1A02-5060)

The following screen show the dial pattern configured to support network based Blind Transfer feature listed in **Section 2.1** under AT&T IP Flexible Reach-Enhanced Features. See corresponding trunk configuration for Communication Manager in **Section 6.6.2**.

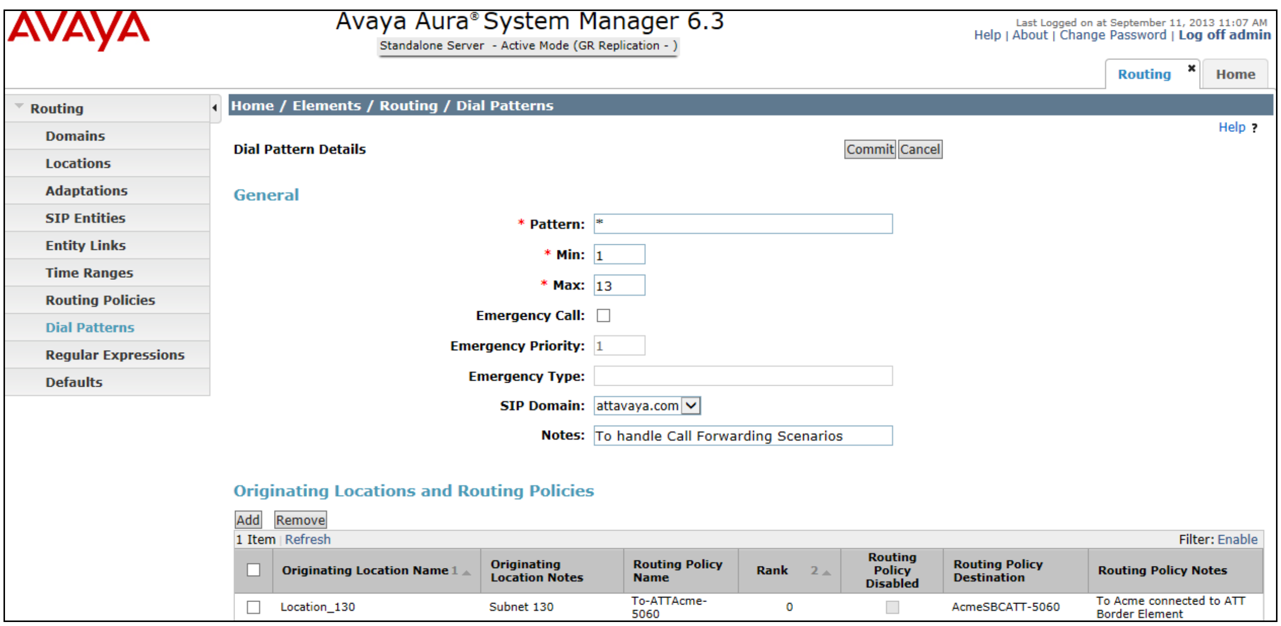


Dial Pattern for Network based Blind Transfer (CLAN1A02-5070)



Dial Pattern for Outbound/Forwarded Calls

The following screen show the dial pattern configured to support network based Call Forwarding features setup listed in **Section 2.1** under AT&T IP Flexible Reach-Enhanced Features. See corresponding configuration for Communication Manager in **Section 6.8.3**.



Dial Pattern for Additional Network Features

5.9. Avaya Aura® 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

In this reference configuration Communication Manager 6.3 is provisioned as the Telephony Application Server, supporting H.323, SIP, Analog and Digital. This section describes the administration steps for Communication Manager in support of the AT&T IP Flexible Reach 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 these Application Notes. Other parameter values may or may not match based on local configurations. Also **NCR** feature may require additional licensing.

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, verify that the **IP Trunks** field 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
```

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 and 5-digit Extensions with a **Call Type** of **ext** beginning with the digits **5xxxx** (e.g., Local extensions for Communication Manager stations, agents, and Vector Directory Numbers, VDNs, in this reference configuration conform to this format).
- 1-digit Facilities Access Code (indicated with a **Call Type** of **fac**) (e.g., **9** access code for outbound ARS dialing). Note – ARS is typically used for public trunk calls. In the reference configuration ARS is used for calls to PSTN via the AT&T IP Flexible Reach service (see **Section 6.8**).
- 3-digit Facilities Access Codes (indicated with a **Call Type** of **fac**) beginning with the character ***** used for Call Forwarding features of AT&T IPFR-EF.

```
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   ext
   3            5   ext
   5            5   ext
   9            1   fac
   *            3   fac
```

6.3. IP Node Names

Following screen shows the node names used for AT&T IP Flexible Reach service provisioning.

```
change node-names ip                                                Page   1 of   2
                                IP NODE NAMES
    Name                 IP Address
Gateway001             10.80.130.1
CLAN-1A02              10.80.130.204
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                   ECM:y
    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:

- **Authoritative Domain** – Set to **attavaya.com** to match the domain configured in **Section 5.1**.
- **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 Flexible Reach service)
- **UDP Port Max**: - Set to **32767** (Required for AT&T IP Flexible Reach service)

```
change ip-network-region 2                                     Page   1 of  20
                             IP NETWORK REGION
  Region: 1
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 4** 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    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                                                              n  all
 3
```

6.6. SIP Trunks

Three trunks are configured for testing in this reference configuration.

- Trunk group for NCR disabled to handle all inbound and outbound calls
- Trunk group for NCR enabled to handle the blind transfer call using SIP Refer
- Trunk group to handle CM Messaging and SIP extension registered with Session Manager

6.6.1. NCR Disabled SIP Trunk for Inbound and Outbound Calls with AT&T IP Flexible Reach

This SIP trunk is used in the reference configuration for all features listed in **Section 2** except for Network-based Blind Transfer.

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
 Prepend '-' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
   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 (e.g., **1**).

```
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: 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: 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 **Network Call Redirection?** to **n**.
- Set **Send Diversion Header?** field to **y**.
- Set **Support Request History?** field to **n**.
- Set **Telephone Event Payload Type** field to the RTP payload type required by the AT&T IPFR-EF service (e.g., **100**).
- Set **Identity for Calling Party Display** field to **From** to display the Calling Party Number using the **From** header in the SIP INVITE.

```
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? y
                       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? From
Block Sending Calling Party Location in INVITE? n
                                  Enable Q-SIP? n
```

6.6.2. NCR Enabled SIP Trunk for Network Based Blind Transfer call with AT&T IP Flexible Reach – Enhanced Features service

This SIP trunk is used for network based blind transfer using vectors and only for inbound calls. See **Section 6.9** for vector configuration. 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  2                                      Page   1 of   1
                              SIGNALING GROUP

Group Number: 2             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
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? N
   Near-end Node Name: CLAN_1A02             Far-end Node Name: SM63
 Near-end Listen Port: 5070                Far-end Listen Port: 5070
                                        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
```

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

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: ATT                        COR: 1       TN: 1       TAC: 102
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                      Signaling Group: 2
                                                    Number of Members: 10
```

On **Page 4** of the trunk-group form, set **Network Call Redirection?** to **y**.

Note: **NCR** feature may require additional licensing

```
add trunk-group 2                                              Page   4 of  21
                              PROTOCOL VARIATIONS
                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? 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
```

6.6.3. 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
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
     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                  Initial IP-IP Direct 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/Alerting/Diverting/Connected 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? From
            Block Sending Calling Party Location in INVITE? n
                                              Enable Q-SIP? n
```

6.7. Public Unknown Numbering

In the public unknown numbering form, Communication Manager local extensions are converted to AT&T Flexible Reach numbers (previously assigned by AT&T) and directed to the “public” trunks defined in **Section 6.6**. Use the **change public-unknown-numbering 0** command to add entries for AT&T IP Flexible Reach service DIDs. 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
 4  2               1                           4
 5  5                                           5
 5  50001           1         30324            10        Total Administered: 4
 5  50003           1         2482317          7            Maximum Entries: 9999
```

6.8. Outbound Call Routing From Avaya Aura® Communication Manager

Route pattern and ARS analysis table forms are configured for outbound calls to PSTN using AT&T IP Flexible Reach service.

6.8.1. Route Pattern

Route patterns are used to direct calls to the appropriate SIP trunk using either the Automatic Route Selection (ARS) or Automatic Alternate Routing (AAR) dialing tables. Use the **change route-pattern x** command, where **x** is an available route to define new route pattern. The following screen shows the route pattern (**1**) used to support AT&T IP Flexible Reach features.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: To_ATT
                             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: 1   0                                                               n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     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
```

Similarly, another **Route Pattern 3** was configured to handle CM Messaging and SIP endpoint calls.

6.8.2. AAR Dialing for CM Messaging and SIP Endpoints

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 6.8.1**. In the following screen **5005** string used for calls to SIP endpoints and **55000** is the pilot number used for coverage to CM Messaging using trunk configured in **Section 6.6.3** via Session Manager.

```
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
```

6.8.3. ARS Dialing for AT&T IP Flexible Reach service

Automatic Route Selection (ARS) is used to direct calls to AT&T Flexible Reach service via the route pattern defined in **Section 6.8.1**. Following screen shows the entries made for ARS dialing to support outbound AT&T IP Flexible Reach service calls.

```
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
    303                  10   10   1          natl          n
    720                  10   10   1          natl          n
```

6.8.4. ARS Dialing for AT&T IP Flexible Reach-Enhanced Features

Following screen shows the entries made for ARS dialing to support additional AT&T IP Flexible Reach-Enhanced Features service calls.

- *72 – To enable Call Forwarding Unconditional
- *73 – To disable Call Forwarding Unconditional
- *90 – To enable Call Forwarding Busy
- *91 – To disable Call Forwarding Busy
- *92 – To enable Call Forwarding – Ring No Answer
- *93 – To disable Call Forwarding – Ring No Answer
- *94 – To enable Call Forwarding – Not Reachable
- *95 – To disable Call Forwarding – Not Reachable

Note: All these features are enabled on a particular line and multiple features can be enabled at the same time. Refer to AT&T feature documentation for priority order for these features.

```
change ars analysis *                                           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
    *72                   13   13     1        natl         n
    *73                    3    3     1        natl         n
    *90                   13   13     1        natl         n
    *91                    3    3     1        natl         n
    *92                   13   13     1        natl         n
    *93                    3    3     1        natl         n
    *94                   13   13     1        natl         n
    *95                    3    3     1        natl         n
```

6.9. Post-Answer Redirection

This section provides an example of Post-Answer Redirection. In this example, the inbound call is routed to the VDN shown in screen below, which invokes the vector shown in the next screen.

```
display vdn 2018                                              Page   1 of   3
                           VECTOR DIRECTORY NUMBER
                             Extension: 2018
                                 Name*: NCR Ringback REFER
                           Destination: Vector Number        18
                   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
```

Sample VDN for Post-Answer Redirection

```
display vector 18                                             Page   1 of   6
                              CALL VECTOR

    Number: 18                Name: NCRRefer
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock?
n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing?
y
  Prompting? y   LAI? n  G3V4 Adv Route? y   CINFO? n   BSR? y  Holidays? n
  Variables? y   3.0 Enhanced? Y
01 #    NCR Refer with ringback
02 wait-time    2   secs hearing ringback
03 # Answer call with announcement
04 announcement 33007
05 # Refer
06 route-to     number ~r7209772643      with cov n if unconditionally
10 #    Play this announcement only on redirect failure
11 disconnect   after announcement 33008
12
```

Sample Vector for Post-Answer Redirection

6.10. Saving Translations

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

7. Configure Acme Packet Session Border Controller (SBC)

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 govern 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 Flexible Reach service. The Session Agent Groups (**SAG**) defined here, and further down, is provisioned under the session-groups **SP-PROXY.**

```
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            SIP
                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 IPFR-EF service 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 Flexible Reach service resides.

network-interface

name	**s1p0**
sub-port-id	0
description	
hostname	
ip-address	**192.252.35.202**
pri-utility-addr	
sec-utility-addr	
netmask	**255.255.255.128**
gateway	**192.252.35.1**
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	**192.252.35.202**
ftp-address	
icmp-address	**192.252.35.202**
snmp-address	
telnet-address	
ssh-address	

ANNOTATION: The realm configuration **ATT** below represents the external network on which the AT&T IP Flexible Reach service resides, and applies the SIP manipulation **modSendRecv**.

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	**removeHeader**
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 IPFR-EF service 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 Flexible Reach service border element is also specified in the **session-group** section below.

session-agent

hostname	**135.194.131.41**
ip-address	**135.194.131.41**
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 IPFR-EF service. 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 IPFR-EF service 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
	135.194.131.41
trunk-group	
sag-recursion	enabled
stop-sag-recurse	401,407

ANNOTATION: The SIP interface below is used to communicate with the AT&T IPFR-EF service.

sip-interface

state	enabled
realm-id	**ATT**
description	
sip-port	
address	**192.252.35.202**

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 manipulation shown below is used for deleting a header **Resource-Priority** from an INVITE request. See **Section 2.2, Item 5** for further details.

sip-manipulation
    name **removeHeader**
    description **Remove Incoming Header**
    split-headers
    join-headers
    header-rule
        name **deleteResourcePriority**
        header-name **Resource-Priority**
        action **delete**
        comparison-type **pattern-rule**
        msg-type **request**
        methods **INVITE**
        match-value
        new-value

ANNOTATION: The SIP manipulations shown below are used for modifying several headers (To, From, Contact and Diversion) 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
header-rule
        name                    modDiversion
        header-name                 Diversion
        action                  manipulate
        comparison-type             pattern-rule
        msg-type                  any
        methods                   INVITE
        match-value
        new-value
        element-rule
                name                        modDiversionVal
                parameter-name
                type                      uri-user
```

action find-replace-all
match-val-type any
comparison-type case-sensitive
match-value \+(.*)
new-value $modDiversion.$modDiversionVal.$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 Header manipulations shown below are used to remove **Endpoint-View** header and Bandwidth statement from SDP. See **Section 2.2, Item 6,7** for further explanation. Note that some additional Avaya specific headers were removed to reduce the packet size.

header-rule
name deleteEndpointView
header-name Endpoint-View
action delete
comparison-type pattern-rule
msg-type any
methods
match-value
new-value
header-rule
name deletePlocation
header-name P-Location
action delete
comparison-type pattern-rule

```
            msg-type                    any
            methods
            match-value
            new-value
        header-rule
            name                        deleteAvGlobalSessionID
            header-name                 Av-Global-Session-ID
            action                      delete
            comparison-type             pattern-rule
            msg-type                    any
            methods
            match-value
            new-value
        header-rule
            name                        deletePAvMessageId
            header-name                 P-AV-Message-Id
            action                      delete
            comparison-type             pattern-rule
            msg-type                    any
            methods
            match-value
            new-value
        header-rule
            name                        deleteElement
            header-name                 Content-Type
            action                      manipulate
            comparison-type             case-sensitive
            msg-type                    any
            methods                     INVITE
            match-value
            new-value
            element-rule
                name                        deleteBandwidth
                parameter-name              application/sdp
                type                        mime
                action                      find-replace-all
                match-val-type              any
                comparison-type             pattern-rule
                match-value                 \Rb=[AC][ST]:64
                new-value
```

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 Flexible Reach service 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	**192.252.35.202**
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	**192.252.35.1**
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. AT&T IP Flexible Reach

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 disconnects properly. Repeat the above step for an outbound call.
2. Verify basic call functions such as hold, transfer, and conference.
3. Verify the use of DTMF signaling.

8.2. AT&T IP Flexible Reach-Enhanced Features

1. Based upon the DIDs provided for Network based Simultaneous Ring, verify that the primary and secondary endpoints ring at the same time and calls can be answered on either phone.
2. Based upon the DIDs provided for Network based Sequential Ring (Locate Me), verify that the primary endpoint rings for a designated time determined by the network and if not answered the secondary endpoint rings and call with talk path can be verified at each endpoint.
3. Based upon the DIDs provided for Network based Blind Transfer (using Communication Manager vector generated REFER), the call can be referred/transferred off-net to another PSTN endpoint using AT&T IP Flexible reach network.
4. Verify that all network based call forwarding features listed in **Section 2.1** can be enabled and calls can be successfully re-directed and answered at the forwarded PSTN number.

8.3. Avaya Aura® Communication Manager

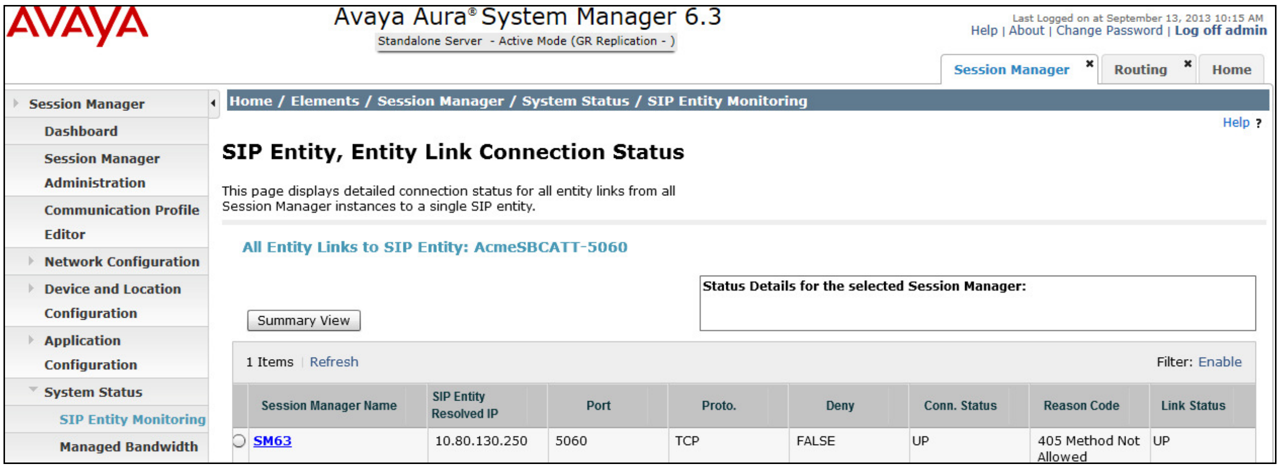
The following examples are only a few of the monitoring commands available on Communication Manager. See [**5**] and [**6**] for more information.

- From the Communication Manager console connection, enter the command ***list trace tac xxx***, (not shown) 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.

8.4. 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.



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 Flexible Reach service using either AVPN or MIS-PNT transport. This solution provides users of Avaya Aura® Communication Manager the ability to support inbound and outbound calls and additional network features over an AT&T IP Flexible Reach SIP trunk service connection.

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, Release 6.3, Issue 3, October 2013
[2] Installing and Configuring Avaya Aura® Session Manager, Doc ID 03-603473 Issue 2, November 2010
[3] Maintaining and Troubleshooting Avaya Aura® Session Manager, Release 6.3, Issue 3, October 2013
[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 8, Release 6.3, May 2013, 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 Flexible Reach-Enhanced Features Service Descriptions:

[8] AT&T Enhanced IP Flexible Reach Service description - http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-toll-free-enterprise/