Avaya Solution & Interoperability Test Lab

Application Notes for Integrating Network Call Redirection passing User-to-User Information with Avaya Aura® 6.1 Infrastructure and AudioCodes Mediant™ 3000 Gateway for Look Ahead Interflow over SIP Trunks with Service-Provider User-to-User Treatment - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to successfully pass non-shared User-to-User Information (UUI) with Network Call Redirection (NCR) utilizing SIP REFER messages via Look-Ahead Interflow (LAI) over SIP Trunks with UUI Treatment as Service-Provider for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and AudioCodes Mediant™ 3000 Gateway (M3K). Network Call Redirection (NCR) provides an Avaya Aura® Communication Manager call routing method between sites on a public network or a Virtual Private Network (VPN).

Information in these Application Notes has been obtained through Full Stack Test conducted at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes focus on a sample configuration for passing non-shared User-to-User Information (UUI) with Network Call Redirection (NCR) utilizing Look-Ahead Interflow (LAI) over SIP trunks via SIP REFER messages with the UUI Treatment of the associated SIP trunks set to Service-Provider.

Network Call Redirection provides an Avaya Aura® Communication Manager call routing method between sites on a public network or a Virtual Private Network (VPN) that can reduce trunking costs. These cost reductions are particularly valuable in enterprises or multi-site call center environments where trunk costs are high.

Look-Ahead Interflow (LAI) enhances Call Vectoring for call centers with multiple ACD locations. LAI allows these centers to improve call-handling capability and agent productivity by intelligently routing calls among call centers to achieve an improved ACD load balance. The receiving switch is able to accept or deny interflowed calls sent by the sending switch.

When an incoming call arrives at an Avaya Aura® Communication Manager that has NCR feature enabled, call redirection is managed by the SIP Service Provider or VPN Switch instead of the local Avaya server. As a result, trunks that the server would otherwise retain to accomplish a trunk-to-trunk transfer are released after the call redirection takes place.

The SIP REFER message contains all information needed for Network Call Redirection. When the SIP call is answered by an agent or call vectoring event, a SIP REFER message is sent. The non-shared UUI is contained within the SIP REFER message. Non-shared UUI transports only ASAI UUI.

1.1 Solution Components

The following section describes high level solution components.

1.1.1 AudioCodes Mediant™ 3000 Gateway

AudioCodes Mediant™ 3000 Gateway provides consolidation of Public Switched Telephone System (PSTN) facilities into SIP. AudioCodes Mediant 3000 is a carrier class product that offers channel scalability of up to 2016 DS0s in a compact 19"-2U chassis. AudioCodes Mediant 3000 provides a web-based user interface that is used for operations, administration, management, and provisioning functions. Alternatively, there is also a Linux based or Solaris based Element Management System (EMS) server that can be used to provision the AudioCodes Mediant 3000. These additional servers will use an EMS client to communicate with the EMS server. The EMS client resides on a Microsoft Windows based PC, and provides the graphical user interface. Recommendation is to use only one of the available provisioning tools.

1.1.2 Avaya Aura® Session Manager

Avaya Aura® Session Manager is a SIP proxy/routing engine that is capable of routing SIP requests throughout a network. Avaya Aura® System Manager provides administration. Avaya Aura® Session Manager provides the following functionality:

SIP Routing Element
The SIP Routing Element provides site to site routing services including number/name resolution, richly manages network ingress and egress including carrier selection for least cost, time of day, load balancing, and media preferences. There may be multiple SIP Routing Elements – all share the same configuration data and some real time data.

Avaya Aura® Session Manager does the following:

- Routes SIP sessions across the network with centralized routing policies.
- Centralizes SIP registrations and location services.
- Provides the gateway for the enterprise for external SIP adjuncts.

1.1.3 Avaya Aura® System Manager

Avaya Aura® System Manager provides centralized administration for multiple instances of Avaya Aura® Session Manager.

Avaya Aura® System Manager does the following:

- Central administration of dial plans and network routing policy.
- Common user provisioning.
- Platform for centralized logs and alarms.
- Security Management.

1.1.4 Avaya Aura® Communication Manager

Avaya Aura® Communication Manager as an Evolution Server provides Avaya Aura® Communication Manager features for both SIP and non-SIP endpoints. It uses the full call model with Avaya Aura® Communication Manager as the only supported application. Additionally, Avaya Aura® Communication Manager provides Call Center Software functionality when a customer elects to talk with an agent. Calls are delivered to Avaya Aura® Communication Manager(s) via SIP trunks through Avaya Aura® Session Manager.

1.1.5 SIP Network Call Redirection and UUI Information Forwarding

Utilizing SIP trunks, SIP REFER or SIP 302 Moved Temporarily messages contain all the information needed to successfully complete an NCR transfer.

- A SIP REFER message is defined as a call that is answered whether by an agent, greeting, announcement, IVR, etc. Note in this sample configuration that vector steps using announcements or collection of digits will result in a SIP REFER message when using NCR.
- A SIP 302 Moved Temporarily message is sent when the call is not answered before performing an NCR request.

Non-shared UUI information forwarding is supported with NCR when the SIP Service Provider supports UUI transport in conjunction with the specific types used by the PSTN Central Office (CO). Non-shared UUI transported consists of only ASAI UUI.

1.2 Testing and Results

The Full Stack Testing performed demonstrates NCR over SIP Trunks with the LAI feature for Avaya Aura® Communication Manager. Testing included Avaya Aura® Communication Manager, Avaya Aura® Session Manager and AudioCodes Mediant™ 3000 Gateway elements. In addition, the associated tests conducted include non-shared UUI. The testing focused on verifying the SIP REFER message contains all the information necessary to complete the NCR transfer. No "true" service provider trunking was used in the testing.

The high-level call scenario tested consists of launching a call from a simulated PSTN. The call from the simulated PSTN would come into the AudioCodes Mediant™ 3000 Gateway (M3K) over an ISDN trunk. Via the routing tables on the M3K, the call routes to the Avaya Aura® Session Manager over a SIP trunk. Based on the Routing Policy defined within Avaya Aura® Session Manager the call will route to the primary Avaya Aura® Communication Manager. This Avaya Aura® Communication Manager will process the call and package UUI contents as applicable. With the NCR configuration and LAI call vectoring in place a SIP REFER message is generated and sent to the M3K in order for NCR to occur. The first leg of the call ends and the second leg of the call will be delivered to the secondary Avaya Aura® Communication Manager. The call then follows basic call processing for delivery to a call center agent.

The testing results validate two-way talk path, non-shared UUI data is present as applicable, displaying the ASAI UUI data on the agent's phone display when the UUI-Info button on the agent's phone is pressed, correct ASAI UUI data is displayed in the applicable message tracing and trunks will be seized or released as expected.

1.3 Assumptions

These Application Notes do not provide any configuration details for the following list of assumptions.

- Avaya Aura® Communication Manager has been installed and is operational
- Avaya Aura® System Manager has been installed and is operational
- Avaya Aura® Session Manager has been installed and is operational
- AudioCodes Mediant 3000 Gateway has been installed and is operational
- PSTN connectivity to the AudioCodes Mediant™ 3000 Gateway has already been established and is operational
- Adtran Patch Panel and 2800 MUX has been installed and is operational

1.4 Sample Configuration

The following diagram depicts the configuration used for testing.

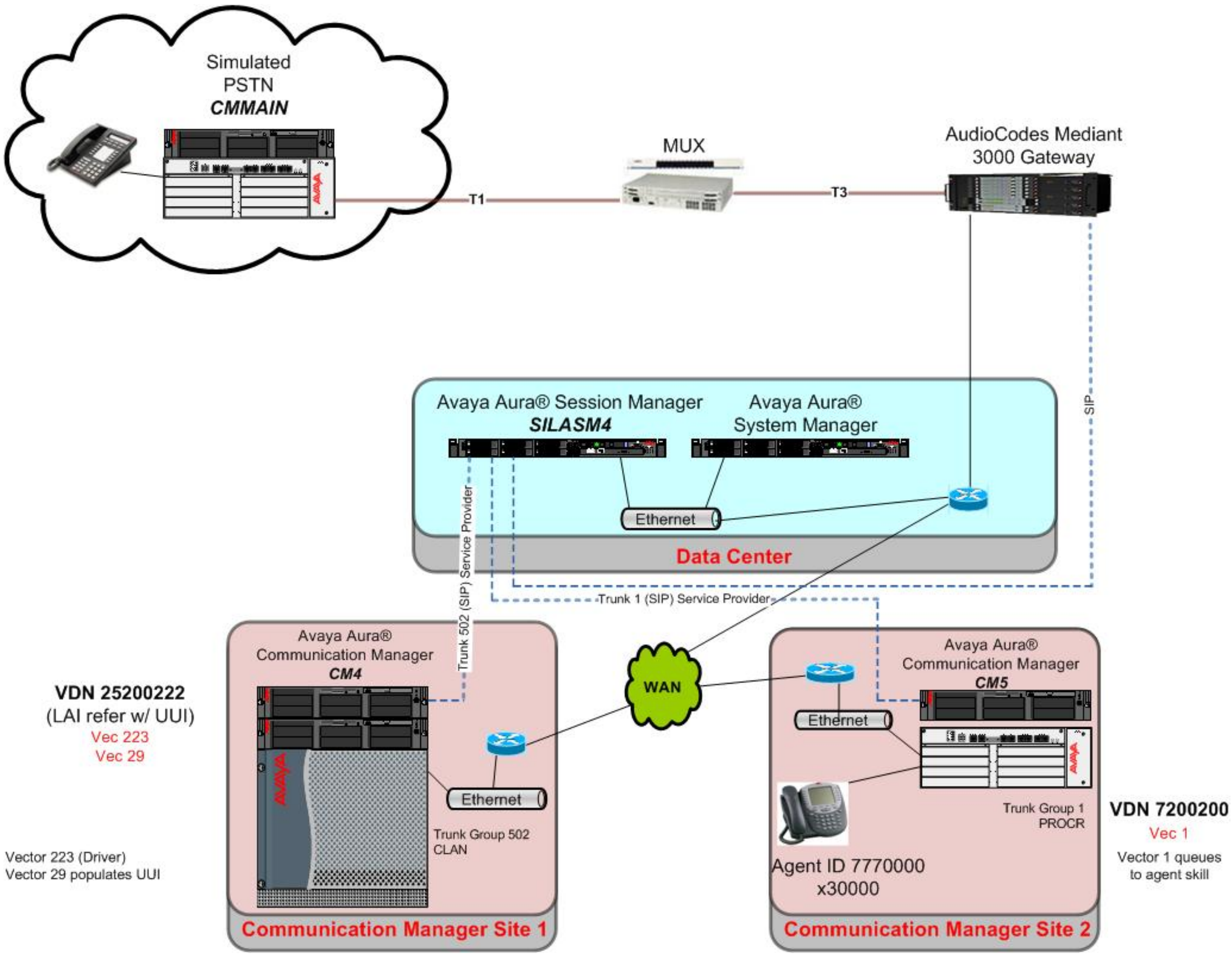


Figure 1

The lab configuration consists of three (3) Avaya Aura® Communication Managers, one (1) Avaya Aura® Session Manager, one (1) AudioCodes Mediant™ 3000 Gateway, one (1) Adtran patch panel and one (1) 2800 MUX. The LAI administration for NCR is set for a multi-switch configuration.

The first Communication Manager is strictly used to simulate PSTN calls (CMMain depicted in **Figure 1**). This Communication Manager is connected to an Adtran patch panel and 2800 MUX, which connects to the AudioCodes Mediant™ 3000 Gateway (M3K) via an ISDN trunk. This Communication Manager has one station administered for generating calls into the environment.

The second Communication Manager is used for receiving the inbound call from the PSTN and to initiate the NCR LAI SIP REFER (CM4 depicted in **Figure 1**). This Communication Manager appends non-shared UUI to the call as applicable. This Communication Manager is connected to

Session Manager (SILASM4 depicted in **Figure 1**) via a SIP Trunk.

The third Communication Manager will receive the redirected call and perform basic call processing in order to queue the call for delivery to an ACD Agent (CM5 depicted in **Figure 1**). This Communication Server has one station administered. This Communication Manager has one ACD Agent ID administered and assigned to a skill.

1.5 Detailed Call Flow

A caller on the simulated PSTN Communication Manager initiates a simulated PSTN call by dialing **925220220**. Based on the call routing administration of the simulated PSTN Communication Manager, **25220220** is passed over an ISDN trunk to the M3K.

When the call is received by the M3K, the called number prefix **25220** is matched against the M3K Routing Table. This entry will route the call to Session Manager.

The Network Routing Policy (NRP) administration on Session Manager will route the call to the SIP Entity named "silasm4_cm4 CLAN 01a11_5060_TCP". This is based on the NRP administration for the dial pattern entry of **2522**. This particular SIP Entity is a CLAN board within the Communication Manager (CM4 depicted in **Figure 1**). The trunk between the Session Manager and the SIP Entity for the Communication Manager is SIP Trunk group 502; Signaling Group 502, UUI Treatment Service-Provider.

As the call is received over trunk group 502 for Communication Manager (CM4 depicted in **Figure 1**), the incoming call is routed to VDN 25220220 with the associated vector 223.

The first step in vector 223 will wait 2 seconds hearing ringback. The next step will play an announcement. The announcement is necessary in the test configuration as the SIP REFER requires that the call is answered. The third vector step routes the call to vector 29. In vector 29 the vector variables are populated with ASAI UUI data. With the administration of the VDN 2520220 variables and vector 29 populates 96 bytes of ASAI UUI data.

After vector 29 processing completes, the call is sent back to vector 223 for further processing. The final step in vector 223 initiates the REFER to a VDN which resides on the receiving Communication Manager (CM5 depicted in **Figure 1**). The vector 223 step in reference for the NCR LAI SIP REFER is "route-to ~r7200200".

The sending Communication Manager (CM4 depicted in **Figure 1**) sends a SIP REFER back to the M3K via the Session Manager causing the first call to be dropped after a second call is issued and established with the receiving Communication Manager (CM5 depicted in **Figure 1**). The SIP REFER will route call back over SIP Trunk Group 502 to the Session Manager and onto the M3K. The M3K has a routing table entry for “7200200” that sends the call to Session Manager. The NRP administration on Session Manager will route the call to the SIP Entity Link named "silasm4 to silcm5-procr". This is based NRP administration for the dial pattern entry of 7200. This particular endpoint for the far-end SIP Entity is a PROCR interface on Communication

Manager (CM5 depicted in **Figure 1**). The trunk between the Session Manager & the SIP Entity for the particular Communication Manager is SIP Trunk Group 1; Signaling Group1, UUI Treatment Service-Provider.

As the call is received over trunk group 1 on Communication Manager (CM5 depicted in **Figure 1**) the incoming call is routed to VDN 7200200 with the associated vector 1.

Vector 1 will queue the call to Skill 1 for delivery to an ACD Agent. Actual call is delivered to Agent ID 7770000 at extension 30000.

If the receiving agent's phone has a **UUI-Info** button administered, the agent can press this button to see the ASAI UUI data passed. Depending on the type of phone, it may not display all ASAI UUI data.

The types of traces used to verify UUI information include MST traces on Avaya Aura® Communication Manager, SIP traces on Avaya Aura® Session Manager and traces on the M3K. Traces allow confirmation of ASAI UUI information.

1.6 Acronyms

ACD	Automatic Call Distributor
ARS	Automatic Route Selection
ASAI	Adjunct Switch Application Interface
CCS	Call Center Software
CM	Avaya Aura® Communication Manager
CO	Central Office
COS	Class of Service
DNIS	Dialed Number Identification Service
DNS	Domain Naming Service
DS0	Digital Signaling Level Zero
DS3	Digital Signal Level 3 (Digital Signal Level 3 T-Carrier)
EAS	Expert Agent Selection
EMS	Element Management System
FAC	Feature Access Code
IP	TCP/IP Address
ISDN	Integrated Digital Services Network
LAI	Look-Ahead Interflow
M3K	AudioCodes Mediant 3000 Gateway
MST	Message Sequence Trace
MUX	Multiplexer
NCR	Network Call Redirection
NRP	Network Routing Policy
PSTN	Public Switched Telephone Network
SAT	System Access Terminal (Avaya Aura® Communication Manager Administration Interface)
SIP	Session Initiation Protocol
SM	Avaya Aura® Session Manager
SMGR	Avaya Aura® System Manager
T3	Digital Carrier Facility (equivalent to DS3)
TCP/IP	Transmission Control Protocol/Internet Protocol
TLS	Transport Layer Security
UCID	Universal Caller Identification
UUI	User-to-User Information
VDN	Vector Directory Number

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Equipment	Software
Avaya Aura® Communication Manager	R016x.00.1.510.1 - Patch 18777
Avaya Aura® Session Manager	6.1.0.0.610023
Avaya Aura® System Manager	6.1.4.0/6.1.0.4.5072 - Patch 6.1.4.113
AudioCodes Mediant 3000 Gateway	5.80A.060.003
Avaya one-X® IP Deskphone • 9630 • 9650	H.323 R3.1 SP1

3. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Avaya Aura® Communication Manager. The sample environment is using a two-switch Look-Ahead Interflow configuration. The *sending* switch processes vector outflow, while the *receiving* switch processes vector inflow.

- Verify System Capabilities and Licensing †‡
- Configure IP Codec Set †‡
- Configure IP Network Region †‡
- Add Node Names and IP Addresses †‡
- Create Signaling Groups †‡
- Add/Configure Trunk Groups †‡
- Add/Configure Vector Directory Number †‡
- Add/Configure Vectors †‡
- Add/Configure Vector and VDN Variables †
- Add/Configure DialPlan Analysis †‡
- Add/Configure Skill Group ‡
- Add/Configure Station ‡
- Add/Configure Call Center Agent ‡

† Configuration/Capability on the *sending* switch in a two-switch Look-Ahead Interflow configuration.
‡ Configuration/Capability on the *receiving* switch in a two-switch Look-Ahead Interflow configuration.

Throughout this section the administration of Communication Manager is performed using a System Access Terminal (SAT). The following commands are entered on the system with the appropriate administrative permissions. Some administration screens have been abbreviated for clarity. These instructions assume that the Communication Manager has been installed, configured, licensed and provided with a functional dial plan.

3.1 Verify System Capabilities and Licensing

3.1.1 SIP Trunk Capacity Check

Use the **display system-parameters customer-options** command to verify that an adequate number of SIP trunk members are licensed for the system. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections. Verify highlighted value, as shown below.

Note: this capability is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration.

```
display system-parameters customer-options              Page   2 of  11
                                OPTIONAL FEATURES
IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 0
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 18000 0
                  Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 128
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 50
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

3.1.2 AAR/ARS Routing Check

Verify that **ARS** is enabled on **Page 3** of **system-parameters customer-options** form.

```
display system-parameters customer-options                  Page   3 of  11
OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? y              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                      CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                        CAS Main? n
Answer Supervision by Call Classifier? y               Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? n                     DCS (Basic)? y
          ASAI Link Core Capabilities? y               DCS Call Coverage? y
          ASAI Link Plus Capabilities? y              DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                         DS1 MSP? y
```

3.1.3 ISDN/SIP Network Call Redirection Check

Verify that **ISDN/SIP Network Call Redirection** is enabled on **Page 4** of **system-parameters customer-options**.

```
display system-parameters customer-options                      Page   4 of  11
 Emergency Access to Attendant? y                                  IP Stations? y
         Enable 'dadmin' Login? y
         Enhanced Conferencing? y                            ISDN Feature Plus? n
                Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
  Enterprise Survivable Server? n                              ISDN-BRI Trunks? y
     Enterprise Wide Licensing? n                                     ISDN-PRI? y
            ESS Administration? y                   Local Survivable Processor? n
        Extended Cvg/Fwd Admin? y                         Malicious Call Trace? y
   External Device Alarm Admin? y                     Media Encryption Over IP? n
Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
              Flexible Billing? n
 Forced Entry of Account Codes? y                     Multifrequency Signaling? y
    Global Call Classification? y        Multimedia Call Handling (Basic)? y
         Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? Y
Hospitality (G3V3 Enhancements)? y                 Multimedia IP SIP Trunking? y
                     IP Trunks? y
         IP Attendant Consoles? y
```

3.1.4 Look-Ahead Interflow (LAI) Check

Verify that **Lookahead Interflow (LAI)** is enabled on **Page 6** of **system-parameters customer-options**.

```
display system-parameters customer-options                     Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 6.0
                              ACD? y                              Reason Codes? y
                     BCMS (Basic)? y                   Service Level Maximizer? n
       BCMS/VuStats Service Level? y                Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y      Service Observing (Remote/By FAC)? y
                Business Advocate? n               Service Observing (VDNs)? y
                  Call Work Codes? y                                Timed ACW? y
    DTMF Feedback Signals For VRU? y                      Vectoring (Basic)? y
                 Dynamic Advocate? n                  Vectoring (Prompting)? y
     Expert Agent Selection (EAS)? y              Vectoring (G3V4 Enhanced)? y
                          EAS-PHD? y               Vectoring (3.0 Enhanced)? y
                 Forced ACD Calls? n      Vectoring (ANI/II-Digits Routing)? y
              Least Occupied Agent? y      Vectoring (G3V4 Advanced Routing)? y
        Lookahead Interflow (LAI)? y                      Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y    Vectoring (Best Service Routing)? y
  Multiple Call Handling (Forced)? y                   Vectoring (Holidays)? y
PASTE (Display PBX Data on Phone)? y                  Vectoring (Variables)? y
```

3.1.5 Vector (Basic) Check

Verify that **Vectoring (Basic)** is enabled on **Page 6** of **system-parameters customer-options**.

Note: this capability is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration.

```
display system-parameters customer-options                  Page   6 of  11
                          CALL CENTER OPTIONAL FEATURES
                           Call Center Release: 6.0
                               ACD? y                             Reason Codes? y
                      BCMS (Basic)? y                Service Level Maximizer? n
       BCMS/VuStats Service Level? y               Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y      Service Observing (Remote/By FAC)? y
                 Business Advocate? n               Service Observing (VDNs)? y
                   Call Work Codes? y                             Timed ACW? y
     DTMF Feedback Signals For VRU? y                     Vectoring (Basic)? y
                  Dynamic Advocate? n                 Vectoring (Prompting)? y
      Expert Agent Selection (EAS)? y             Vectoring (G3V4 Enhanced)? y
                           EAS-PHD? y              Vectoring (3.0 Enhanced)? y
                  Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
              Least Occupied Agent? y     Vectoring (G3V4 Advanced Routing)? y
         Lookahead Interflow (LAI)? y                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y    Vectoring (Best Service Routing)? y
  Multiple Call Handling (Forced)? y                   Vectoring (Holidays)? y
PASTE (Display PBX Data on Phone)? y                  Vectoring (Variables)? y
```

3.1.6 Vectoring (Variables) Check

Verify that **Vectoring (Variables)** is enabled on **Page 6** of **system-parameters customer-options**.

Note: the Vectoring (Variables) is enabled in order to generate and populate simulated UUI contents for testing.

```
display system-parameters customer-options                  Page   6 of  11
                          CALL CENTER OPTIONAL FEATURES
                           Call Center Release: 6.0
                               ACD? y                             Reason Codes? y
                      BCMS (Basic)? y                Service Level Maximizer? n
       BCMS/VuStats Service Level? y               Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y      Service Observing (Remote/By FAC)? y
                 Business Advocate? n               Service Observing (VDNs)? y
                   Call Work Codes? y                             Timed ACW? y
     DTMF Feedback Signals For VRU? y                     Vectoring (Basic)? y
                  Dynamic Advocate? n                 Vectoring (Prompting)? y
      Expert Agent Selection (EAS)? y             Vectoring (G3V4 Enhanced)? y
                           EAS-PHD? y              Vectoring (3.0 Enhanced)? y
                  Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
              Least Occupied Agent? y     Vectoring (G3V4 Advanced Routing)? y
         Lookahead Interflow (LAI)? y                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y    Vectoring (Best Service Routing)? y
  Multiple Call Handling (Forced)? y                   Vectoring (Holidays)? Y
PASTE (Display PBX Data on Phone)? y                  Vectoring (Variables)? y
```

3.1.7 Expert Agent Selection (EAS) Check

Verify that **Expert Agent Selection (EAS)** is enabled on **Page 6** of **system-parameters customer-options** form.

```
display system-parameters customer-options                  Page   6 of  11
                         CALL CENTER OPTIONAL FEATURES
                          Call Center Release: 6.0
                              ACD? y                          Reason Codes? y
                     BCMS (Basic)? y               Service Level Maximizer? n
       BCMS/VuStats Service Level? y             Service Observing (Basic)? y
BSR Local Treatment for IP & ISDN? y     Service Observing (Remote/By FAC)? y
                Business Advocate? n             Service Observing (VDNs)? y
                  Call Work Codes? y                             Timed ACW? y
    DTMF Feedback Signals For VRU? y                     Vectoring (Basic)? y
                 Dynamic Advocate? n                 Vectoring (Prompting)? y
     Expert Agent Selection (EAS)? y             Vectoring (G3V4 Enhanced)? y
                          EAS-PHD? y              Vectoring (3.0 Enhanced)? y
                 Forced ACD Calls? n     Vectoring (ANI/II-Digits Routing)? y
             Least Occupied Agent? y     Vectoring (G3V4 Advanced Routing)? y
        Lookahead Interflow (LAI)? y                     Vectoring (CINFO)? y
Multiple Call Handling (On Request)? y    Vectoring (Best Service Routing)? y
  Multiple Call Handling (Forced)? y                  Vectoring (Holidays)? y
PASTE (Display PBX Data on Phone)? y                 Vectoring (Variables)? y
```

3.1.8 Trunk-to-Trunk Check

Verify that **Trunk-to-Trunk Transfer** is enabled on **Page 1** of **system-parameters features** form.

Note: This feature can pose a significant security risk by increasing the risk of toll fraud and must be used with caution. To minimize the risk, a COS can be defined to allow trunk-to-trunk transfer for a specific trunk group(s). For more information regarding how to configure a Communication Manager to minimize toll fraud, see **Reference [4].**

Note: this capability is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration.

```
display system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
…
```

3.1.9 Expert Agent Selection (EAS) Enabled Feature Check

Verify that **Expert Agent Selection (EAS) Enabled** feature is enabled on **Page 11** of **system-parameters features** form.

```
display system-parameters features                              Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                    Delay:
    Message Waiting Lamp Indicates Status For: station
  VECTORING
                    Converse First Data Delay: 0      Second Data Delay: 2
               Converse Signaling Tone (msec): 100         Pause (msec): 70
                     Prompting Timeout (secs): 10
                 Interflow-qpos EWT Threshold: 2
    Reverse Star/Pound Digit For Collect Step? n
          Available Agent Adjustments for BSR? n
                             BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
              Service Observing: Warning Tone? y     or Conference Tone? n
     Service Observing Allowed with Exclusion? n
             Allow Two Observers in Same Call? n
PASTE (Display PBX Data on Phone)? y                 Vectoring (Variables)? y
```

3.2 Add Node Name of Avaya Aura® Session Manager

Using the **change node-names ip** command, add the node-name and IP address of the SIP signaling interface for the Session Manager, if not previously added. In this sample configuration, **silasm4** is the Session Manager. This same screen shot shows the node-name for the C-LAN interface which will be used in administering a SIP signaling-group in **Section 3.5**

Note: IP addresses have been partially hidden for security.

Note: this administration is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration. On the *receiving* switch the node-names ip form will use the PROCR interface as opposed to the C-LAN interface.

```
change node-names ip                                    Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
IP       silasm4           135.9.88.X
IP       sqa8730clan2A     135.9.88.X
```

3.3 Configure Codec Type

Using the **change ip-codec-set n** command where n is the next available number. In the sample configuration G.711MuLaw was the preferred codec.

Enter the following values:

- **Audio Codec - G.711** is entered as the first choice. Optionally enter in a secondary codec such as **G.729A**.

Note: this administration is also required on the receiving switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration. On the receiving switch the ip-codec will include the same values.

```
change ip-codec-set 4                                   Page   1 of   2
                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU            n         2        20
 2: G.729A             n         2        20
 3:

     Media Encryption
 1: none
```

3.4 Configure IP Network Regions

In the sample configuration calls to/from Session Manager for the sending Communication Manager (CM4 depicted in **Figure 1**) will be viewed as calls to/from ip-network-region **9**. The sending Communication Manager and the CLAN endpoint for the SIP trunk are in ip-network-region **1**. The sending Communication Manager also has VoIP resources in ip-network-region **1**. To enable communication between the two network regions requires additional administration of the **ip-network-region** and **signaling-group** forms as shown in the next few sections.

NOTE: these Application Notes do not cover the administration of the IP Network Region for VoIP resources. Assumption is the IP Network Region for VoIP resources already exists.

3.4.1 Configure IP Network Region for Session Manager

Using the command **change ip-network-region 9**, enter the following values

- **Authoritative Domain** — **dr.avaya.com** is entered. This is the authoritative domain for the environment.
- **Codec Set** — **4** is entered. IP Codec set created in **Section 3.3**
- **Intra-region IP-IP Direct Audio** — **yes**
- **Inter-region IP-IP Direct Audio** — **yes**

Note: this administration is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample configuration as both switches have similar network region design.

```
change ip-network-region 9      IP NETWORK REGION             Page   1 of  20
Region: 9
Location:          Authoritative Domain: dr.avaya.com
    Name: NCR SM
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 4              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Navigate to **Page 4** and verify that ip-network-region **1** and **9** are directly connected and use **ip codec set 4** as shown below.

```
change ip-network-region 9                                     Page   4 of  20
Source Region: 9     Inter Network Region Connection Management     I      S M
                                                                    G  A   y t
dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G   n c
rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L   c e
1    4     y    NoLimit                                             n      y t
2
3
4
5
6
7
8
9    4                                                                all
```

3.5 Add SIP Signaling Group

Using the command **add signaling-group x** where x is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. In the sample configuration, trunk group **502** and signaling group **502** were used to connect to Session Manager. Default values can be used for the remaining fields.

- **Group Type:** **sip**
- **Transport Method:** **tcp**
- **IMS Enabled?:** **n**
- **Peer Detection Enabled?**: **y**
- **Peer Server:** Use default value. **Note:** default value is replaced with "**SM**" after SIP trunk to Session Manager is established
- **Near-end Node Name:** CLAN board node name from **Section 3.2**
- **Far-end Node Name:** Session Manager node name from **Section 3.2**
- **Near-end Listen Port:** **5060**
- **Far-end Listen Port:** **5060**
- **Far-end Network Region:** IP-network-region from **Section 3.4.1**
- **Far-end Domain:** Authoritative Domain from **Section 3.4.1**
- **Enable Layer 3 Test:** **y**

Note: this administration is also required for the signaling group used on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample. On the *receiving* switch the PROCR node-name is used as opposed to the C-LAN node-name.

```
add signaling-group 502
                               SIGNALING GROUP
Group Number: 502            Group Type: sip
IMS Enabled? n        Transport Method: tcp
      Q-SIP? n                                            SIP Enabled LSP? n
   IP Video? n                                  Enforce SIPS URI for SRTP? y
 Peer Detection Enabled? y  Peer Server: SM

Near-end Node Name: sqa8730clan2A         Far-end Node Name: silasm4
Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 9
                                   Far-end Secondary Node Name:
Far-end Domain: dr.avaya.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
```

(1) TCP was used for the sample configuration for testing and debugging purposes. However, TLS would typically be used in production environments.
(2) If any call originating from the SIP phone is not expected to be answered within 3 minutes such would happen if the is made to a VDN and agents are not available within 3 minutes, this value may need to be increased.

3.6 Add SIP Trunk Group

Add the corresponding trunk group controlled by this signaling group using the command **add trunk-group x** where x is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name.
- **Outgoing Display**: **y**
- **TAC:** An available trunk access code.
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group added in **Section 3.5**
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager

Note: this administration is also required for the trunk group used on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample.

```
add trunk-group 502                                          Page   1 of  22
                                TRUNK GROUP

Group Number: 502                  Group Type: sip          CDR Reports: y
Group Name: SIP TG to silasm4 - Shared  COR: 1        TN: 1        TAC: #502
Direction: two-way        Outgoing Display? y
Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 502
                                                     Number of Members: 10
```

Once the add command is completed, trunk members will be automatically generated based on the value in the **Number of Members** field.

On **Page 2**, set the **Preferred Minimum Session Refresh Interval** to **1200**.
Note: To avoid extra SIP messages, all SIP trunks connected to Session Manager should be configured with a minimum value of 1200.

```
add trunk-group 502                                            Page   2 of  22
      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): 1200
 Disconnect Supervision - In? y  Out? y
           XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

On **Page 3**, set the following values:

- **Numbering Format** **public**
- **UUI Treatment** **service provider**

```
add trunk-group 502                                            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

 DSN Term? N
```

On **Page 5**, set the following values:

- **Network Call Redirection** **y**

```
add trunk-group 502                                           Page   5 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? y
                 Network Call Redirection? y
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 127


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                             Enable Q-SIP? n
 DSN Term? N
```

3.7 Administering Public/Unknown Numbering Format

Administer a Calling Party Number (CPN) Prefix for each Vector Directory Number (VDN) that maps to a vector used to place LAI calls.

To enable endpoints to dial extensions defined in Communication Manager, use the command **change public-numbering x** where x is the number used to identify the public number plan. The trunk group used in the example below is the Trunk Group added in **Section 3.6**.

- **Ext Len:** Enter the extension length allowed by the dial plan
- **Ext Code:** Enter leading digit (s) from extension number
- **Trunk Grp:** Enter the SIP Trunk Group number for the SIP trunk between Communication Manager and Session Manager
- **CPN Prefix:** Leave blank unless an enterprise canonical numbering scheme is defined in Session Manager. If so, enter the appropriate prefix.

```
change public-unknown-numbering 2                                   Page 1 of 2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN           CPN
Len Code            Grp(s)    Prefix        Len
                                                  Total Administered: 7
                                                  Maximum Entries: 9999
7  25220220         502                     7
```

3.8 Administer Incoming Vector to Initiate the LAI SIP REFER on Sending Communication Manager

Administer the incoming vector that will initiate the LAI SIP REFER. This vector is associated to the initial incoming VDN in **Section 3.10**. The first primary step of vector processing will play an announcement in order to force a NCR LAI SIP REFER. Vector processing continues onto vector 29 in order to populate non-shared UUI data. Once vector 29 completes, the call is returned for further vector processing. The NCR LAI SIP REFER is performed once the **route-to number ~r** step is executed. In this sample configuration the call redirects to 7200200, a VDN on the receiving Communication Manager (CM5 as depicted in **Figure 1**).

To create the vector use the command **change vector x** where x is the number of the vector. The sample vector below plays an announcement, then utilizes vector 29 to populate ASAI UUI data, followed by performing the SIP REFER.

```
change vector 223                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 223              Name: Driver
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
    Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 announcement 3010
03 goto vector  29   @step 1 if unconditionally
04 route-to     number ~r7200200
05
```

3.9 Administering the Vector to Populate ASAI UUI on Sending Communication Manager

Administer the vector to populate simulated ASAI UUI data. This vector is associated with the vector created in **Section 3.8.**

To create the vector use the command **change vector x** where x is the number of the vector. The sample vector below uses the VDN Variables in **Section 3.10** and Vector Variables in **Section 3.14** to create the 96 bytes of simulated ASAI UUI data. As vector steps 1 thru 5 build the first 80 bytes of ASAI UUI from the VDN Variables, step 6 will add the remaining 16 bytes of ASAI UUI data for a total of 96 bytes from the Vector Variables table.

```
change vector 29                                                  Page   1 of   6
                                  CALL VECTOR
    Number: 29               Name: Populate UUI
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 set          ZA     = V1     CATR  none
02 set          ZB     = V2     CATR  none
03 set          ZD     = V3     CATR  none
04 set          ZE     = V4     CATR  none
05 set          ZF     = V5     CATR  none
06 set          ZG     = none   CATR  1234567890123456
07 return
```

3.10 Administering the Vector Directory Number (VDN) for NCR LAI SIP REFER with non-Shared UUI Data on Sending Communication Manager

Administer the VDN for populating non-shared UUI. This VDN uses the vector from **Section 3.8.**

Create the VDN using the command **add vdn x** where x is the extension of the VDN.

- **Extension:** Enter the extension allowed by the dial plan
- **Name:** Enter a descriptive name
- **Destination:** Enter the initial incoming vector that was created in **Section 3.8**
- **Allow VDN Override:** Disable VDN Override

```
add vdn 25200222                                                  Page   1 of  3
VECTOR DIRECTORY NUMBER
                             Extension: 25200222
                                 Name*: NCR LAI UUI
                           Destination: Vector Number        223
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                  COR: 1
                                  TN*: 1
                             Measured: both
        Acceptable Service Level (sec): 20

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

On **Page 3** of the VDN form, assign digits to the VDN Variables in order to simulate ASAI UUI data. Based on the protocol up to 96 bytes of ASAI UUI can be passed. The first 80 bytes of ASAI UUI are created using the VDN Variables. The remaining bytes will be appended based on vector programming.

- **V1:** Enter 16 characters to simulate ASAIUUI
- **V2:** Enter 16 characters to simulate ASAIUUI
- **V3:** Enter 16 characters to simulate ASAIUUI
- **V4:** Enter 16 characters to simulate ASAIUUI
- **V5:** Enter 16 characters to simulate ASAIUUI

```
add vdn 5200222                                            Page   3 of  3
                            VECTOR DIRECTORY NUMBER
                                VDN VARIABLES*
                    Var  Description       Assignment
                    V1    ASAI 1-16        5200221520022152
                    V2    ASAI 17-32       0022152002215200
                    V3    ASAI 33-48       2215200221520022
                    V4    ASAI 49-64       1520022152002215
                    V5    ASAI 65-80       2002215200221520
                    V6
                    V7
                    V8
                    V9
                   VDN Time-Zone Offset*: + 00:00
                  Daylight Savings Rule*: system
* Follows VDN Override Rules
```

3.11 Administer Initial Incoming Vector on Receiving Communication Manager

Administer the incoming vector that will queue the call to a call center agent. This vector is associated to the initial incoming Vector Directory Number (VDN) in **Section 3.12**. The sample vector will queue the call to a call center agent.

Using the command **change vector x** where x is the number of the vector to create. Add the basic steps below. For **step 2**, use the skill group number from **Section 3.15.**

```
change vector 1                    CALL VECTOR                  Page   1 of   6
    Number: 1                 Name: queue-to NCRUUI
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n          Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y  ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    2   secs hearing ringback
02 queue-to     skill 1    pri l
03 busy
```

3.12 Administer the initial incoming Vector Directory Number (VDN) on Receiving Communication Manager

Administer the initial incoming VDN. This VDN uses the vector from **Section 3.11**

Using the command **add vdn x** where x is the extension of the VDN to create and enter the following values:

- **Extension:** Enter the extension allowed by the dial plan
- **Name:** Enter a descriptive name
- **Destination:** Enter Vector Number created in **Section 3.11**

```
add vdn 720-0200                                               Page   1 of  3
                              VECTOR DIRECTORY NUMBER
                             Extension: 720-0200
                                 Name*: CM5 VDN Manual Test 7200200
                           Destination: Vector Number        1
                   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*:
```

3.13 Administer Variables for Vectors on Sending Communication Manager

Using the command **change variables** add the applicable variables.

- **Variable ZA:** bytes 1-16 of ASAIUUI as referenced in **Section 3.9**
- **Variable ZB**: bytes 17-31 of ASAIUUI as referenced in **Section 3.9**
- **Variable ZD:** bytes 32-47 of ASAIUUI as referenced in **Section 3.9**
- **Variable ZE:** bytes 48-63 of ASAIUUI as referenced in **Section 3.9**
- **Variable ZF:** bytes 64-80 of ASAIUUI as referenced in **Section 3.9**
- **Variable ZG:** bytes 81-96 of ASAIUUI as referenced in **Section 3.9**

Note: the actual variables for vectors form has a total 39 pages. The form depicted below was consolidated to single screen shot for reference.

```
change variables
                              VARIABLES FOR VECTORS
Var Description               Type    Scope Length Start Assignment       VAC
ZA  ASAI 1-16                 asaiuui L     16     1
ZB  ASAI 17-32                asaiuui L     16     17
ZD  ASAI 33-48                asaiuui L     16     33
ZE  ASAI 49-64                asaiuui L     16     49
ZF  ASAI 65-80                asaiuui L     16     65
ZG  ASAI 81-96                asaiuui L     16     81
```

3.14 Administer Dial Plan Analysis

In the screenshot below, the following entries are added by using the command **change dialplan analysis**:

- **Dialed String:** **252** is for extensions local to CM. Call type is **ext**
  8 is for the total length of the digit string
  ***** is for feature access codes local to CM. Call Type is **fac**
  # is for dialed access codes local to CM. Call Type is **dac**

Note: this administration is also required on the *receiving* switch that processes vector inflow for the two-switch Look-Ahead Interflow sample. The sample configuration uses parameters that are applicable for the *receiving* switch. The difference is the dialed string is **7** with a total length of **7** on the *receiving* switch. All other values remain the same.

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 4

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   252          8     ext
   *            3     fac
   #            4     dac
```

3.15 Administer Agent Skill Group on Receiving Communication Manager

Administer the agent skill group. Using the command **add hunt x** where x is the skill group number and enter the following values:

- **Group Name:** Enter descriptive name for the agent skill
- **Group Extension:** Enter the lead hunt extension for the skill
- **ACD:** **y**
- **Queue:** **y**
- **Vector:** **y**

```
add hunt-group 1                                            Page   1 of   4
                                  HUNT GROUP
            Group Number: 1                               ACD? y
              Group Name: Hunt Group One NCR-UUI        Queue? y
         Group Extension: 777-7777                     Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                    MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:
             Queue Limit: unlimited
 Calls Warning Threshold: 2    Port:
  Time Warning Threshold: 60   Port:
```

On **Page 2**, set the following values:

- **Skill:** **y**

```
add hunt-group 1                                            Page   2 of   4
                                  HUNT GROUP
                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n
                 Measured: none
     Supervisor Extension:
      Controlling Adjunct: none
   Multiple Call Handling: none
 Timed ACW Interval (sec):          After Xfer or Held Call Drops? n
```

3.16 Configure Answering Station on Receiving Communication Manager

Add a station to the receiving Communication Manager for the call center agent.

Using the command **add station x** where x is a valid extension number defined in the system. The sample configuration uses a type 9650 h.323 phone. Enter the following values for **Page 1** of the change station form:

- **Phone Type:** Set to 9650
- **Name:** Display name for user
- **Security Code:** Numeric password used when the station is registered.

```
add station 30000                                            Page   1 of   5
                                     STATION
Extension: 30000                        Lock Messages? n               BCC: 0
     Type: 9650                         Security Code: 123456           TN: 1
     Port: S00022                     Coverage Path 1:                 COR: 1
     Name: Station 30000 Agent 7770000 Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 30000
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y
                                               IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y*
```

On Page 5 add a **uui-info** button:

Note: this button is used to show the ASAI UUI data received on the display of the Agent's phone when pressed.

```
add station 30000                                            Page   5 of   5
                                     STATION
AUXILIARY BUTTON ASSIGNMENTS
Main View                               Shifted View
 4: auto-in            Grp:             12:
 5: manual-in          Grp:             13:
 6: after-call         Grp:             14:
 7: aux-work    RC:    Grp:             15:
 8: uui-info                            16:
```

3.17 Configure Agent ID on Receiving Communication Manager

For the sample configuration add an agent id that is associated to the skill group on Communication Manager in order to receive ACD calls.

Using the command **add agent x** where x is a valid extension number for an agent id defined in the system. Fill in the indicated fields. Enter the following values on **Page 1** of the change station form:

- **Name:** Display name for agent
- **Password:** Numeric password used when the agent logs into a station.

```
add agent-loginID 7770000                                   Page   1 of   3
                                 AGENT LOGINID
                Login ID: 777-0000                              AAS? n
                    Name: Agent 7770000                       AUDIX? n
                      TN: 1                            LWC Reception: spe
                     COR: 1                   LWC Log External Calls? n
           Coverage Path:                 n  AUDIX Name for Messaging:
           Security Code:
                                        LoginID for ISDN/SIP Display? n
                                                            Password: 1234
                                              Password (enter again): 1234
                                                         Auto Answer: station
                                                   MIA Across Skills: system
                                           ACW Agent Considered Idle: system
                                           Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :
     WARNING:  Agent must log in again before changes take effect
```

On **Page 2** associate the agent to the skill created in **Section 3.15**:

Note: SN is the skill number and SL is the skill level.

```
add agent-loginID 7770000                                   Page   2 of   3
                                 AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
Call Handling Preference: skill-level                Local Call Preference? n

    SN   RL SL          SN   RL SL          SN   RL SL          SN   RL SL
 1: 1       1
```

3.18 Configure Feature Access Codes on Receiving Communication Manager

For the sample configuration administer feature access codes for the call center agents.

Using the command **change feature-access-codes**, administer valid values for the system.
On **Page 5**, enter following values on the **change feature-access-codes** form:

- **Auto-In Access Code:** FAC to staff in the agent ID to their respective skill.
- **Login Access Code:** FAC to login the agent ID to their respective skill.
- **Logout Access Code:** FAC to logout the agent ID from their skill.

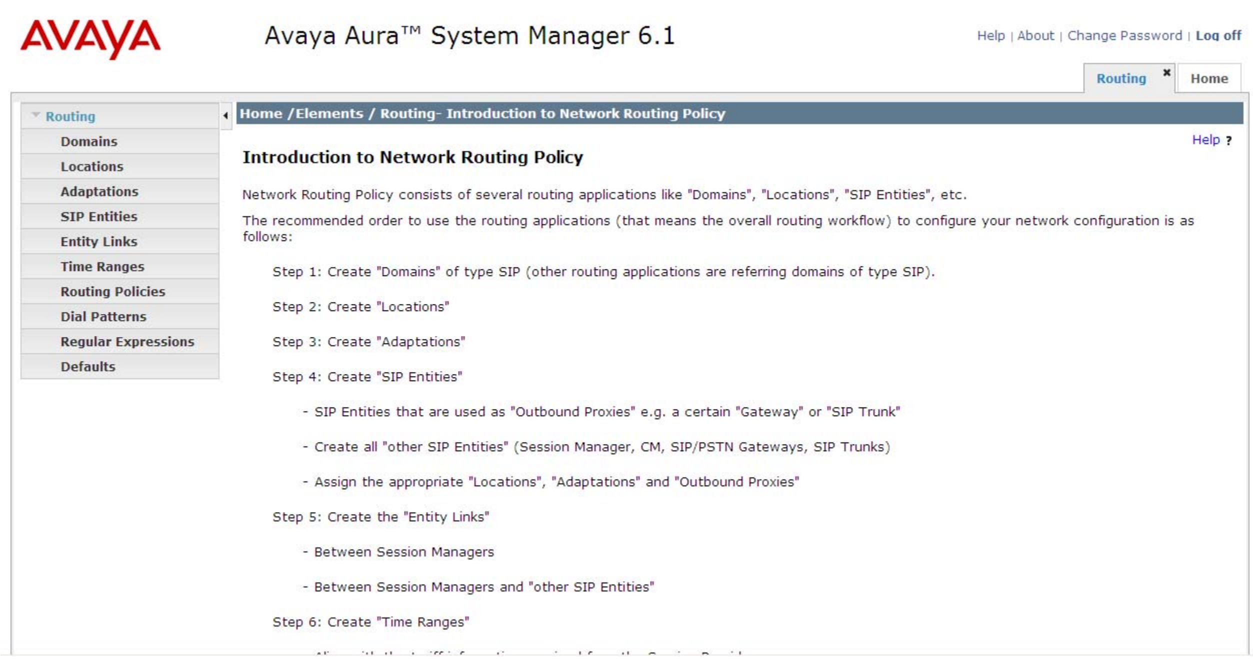
```
change feature-access-codes                                  Page   5 of  11
                               FEATURE ACCESS CODE (FAC)

                                 Call Center Features
  AGENT WORK MODES
                    After Call Work Access Code: *75
                             Assist Access Code: *76
                            Auto-In Access Code: *71
                           Aux Work Access Code: *72
                              Login Access Code: *88
                             Logout Access Code: *89
                          Manual-in Access Code: *73
  SERVICE OBSERVING
      Service Observing Listen Only Access Code: *77
      Service Observing Listen/Talk Access Code: *78
          Service Observing No Talk Access Code: *79
```

4. Configure Avaya Aura® Session Manager

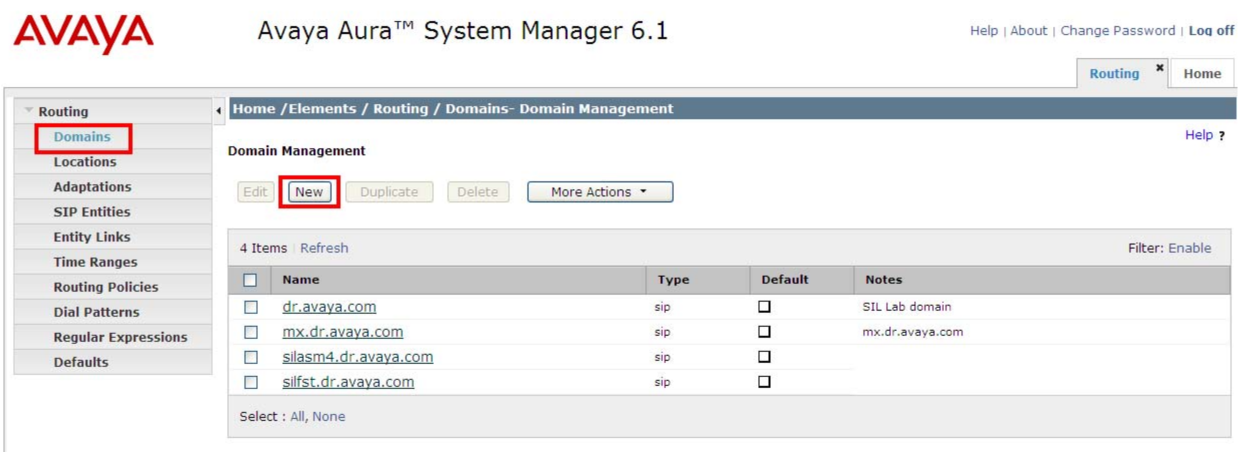
This section provides the procedures for configuring Session Manager. These instructions assume the System Manager and the Session Managers are already installed and functioning properly. Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**http://<FQDN>/SMGR**", where "**<FQDN>**" is the Fully Qualified Domain Name of System Manager. Log in to the System Manager with the appropriate credentials. The menu below is displayed. Select the link for **Routing** under **Elements.**

The menu below is displayed. The sub-menus in the left column will be used for configuration.

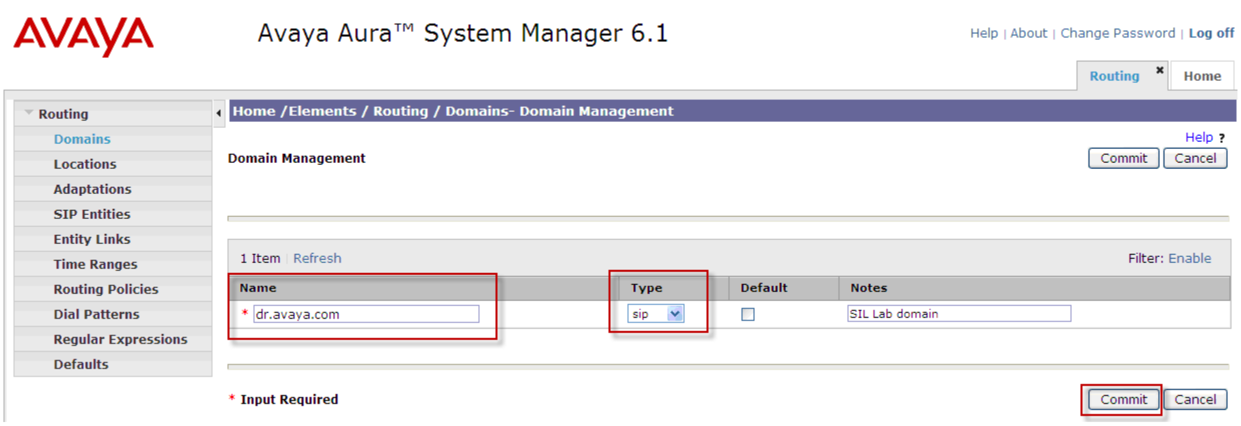


4.1 Specify SIP Domains

Configure the SIP Domain appropriately. Select **Domains** from the left hand window. Click **New**.



In the **General** section, under **Name** add the name of the SIP domain in use. Select **sip** for the **Type**. Under **Notes** add a brief description. Click **Commit** to save.

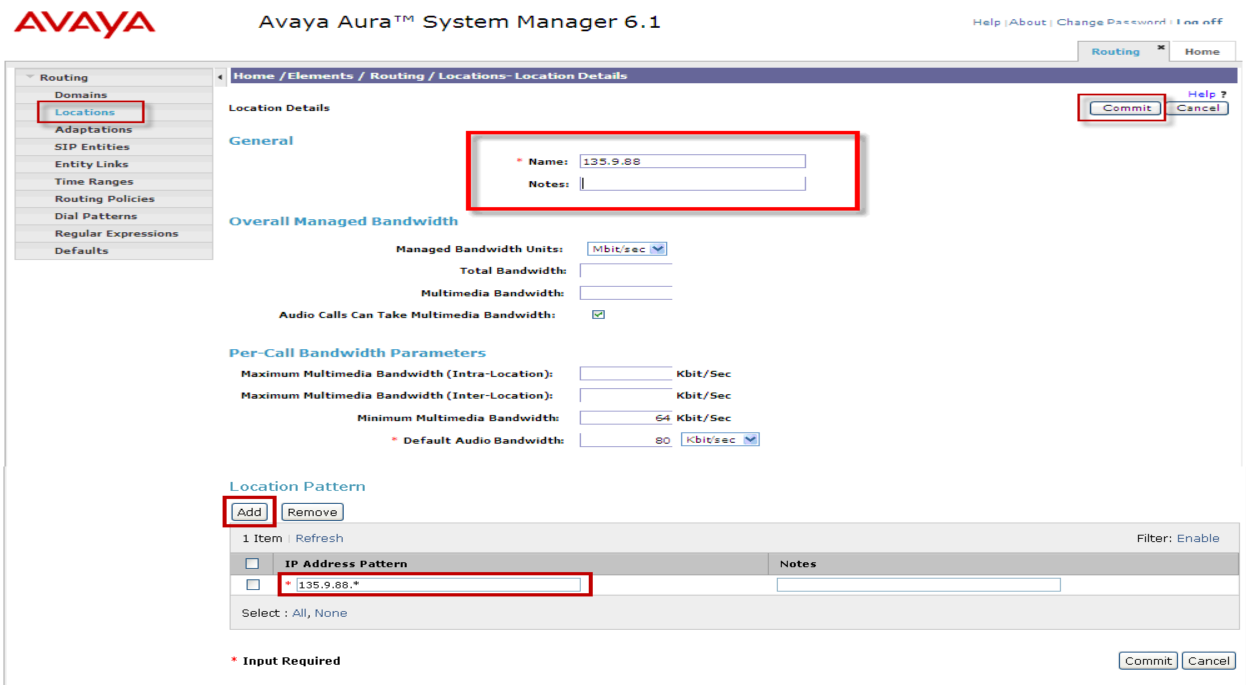


4.2 Add Locations

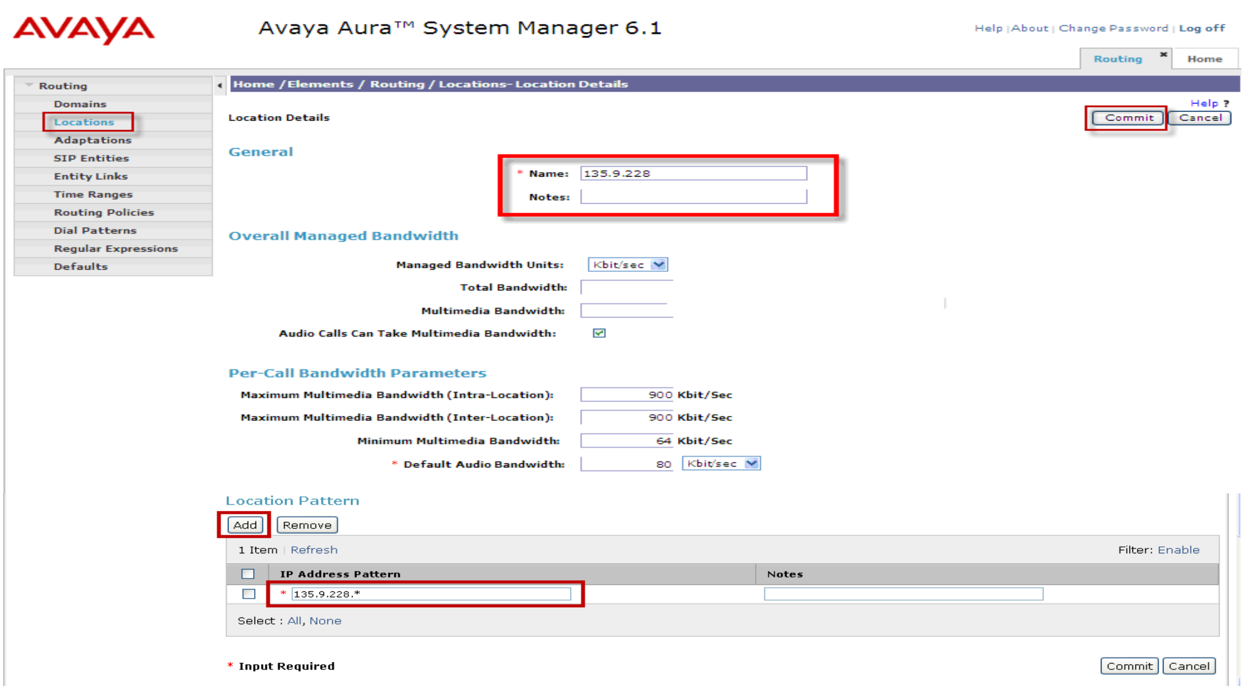
Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. For our environment we are adding two separate locations. The locations specified are based on the network ID. The first location is for our *sending* Communication Manager subnet 135.9.88 (CM4 depicted in **Figure 1**). The second location is for our *receiving* Communication Manager subnet 135.9.228 (CM5 depicted in Figure 1).

Select **Locations** and click **New** (not shown)**.** In the **General** section, under **Name** add a descriptive name. Under **Notes** add a brief description. Under **Location Pattern** click **Add** and enter the IP address pattern. Click **Commit** to save.

Add 135.9.88 Location:



Add 135.9.228 Location:



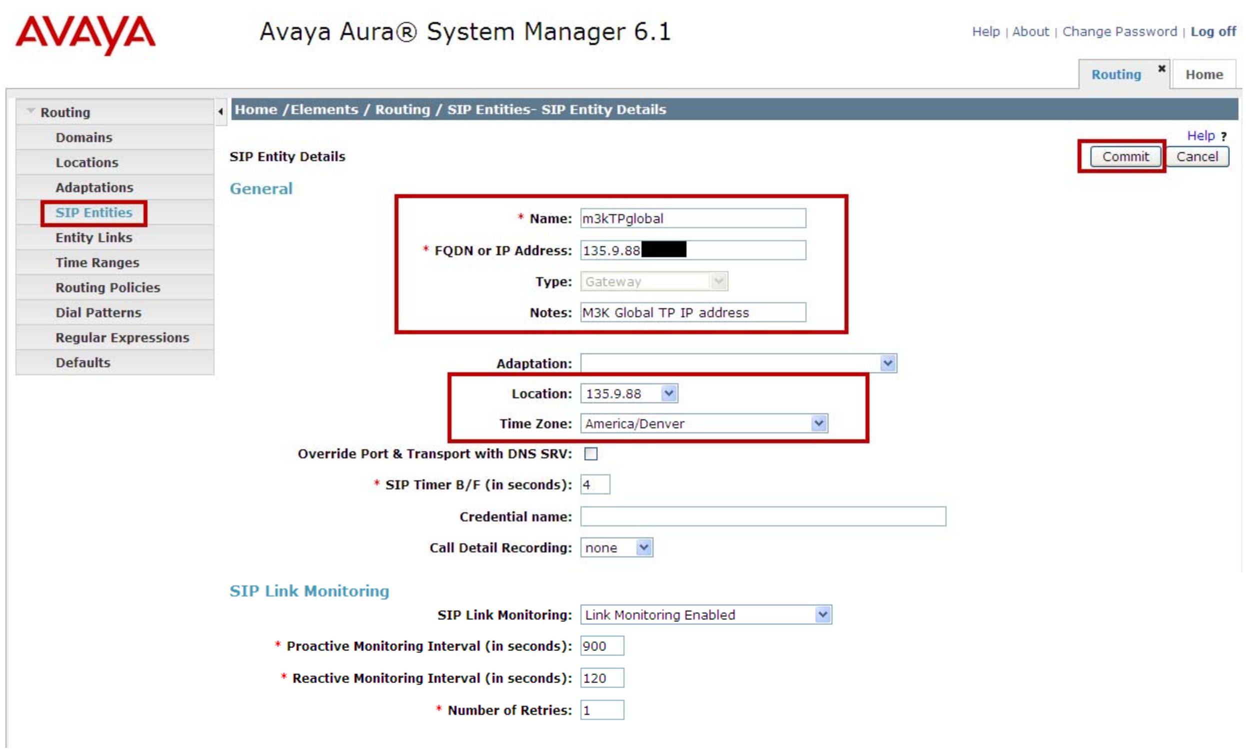
4.3 Add the AudioCodes Mediant™ 3000 Gateway SIP Entity

Configure the SIP Entity for the AudioCodes Mediant™ 3000 Gateway (M3K). Select **Routing** and select **SIP Entities.** Click **New** (not shown). In the **General** Section, add the following:

- **Name:** Add an identifier for the M3K
- **FQDN or IP Address:** Enter the IP Address of M3K
- **Type:** Select **Gateway**
- **Notes:** Add a brief description
- **Location:** From the drop down select the ***first*** Location added in **Section 4.2**
- **Time Zone:** From the drop down select the appropriate time zone

Click **Commit** to save.

Note: IP addresses have been partially hidden for security.



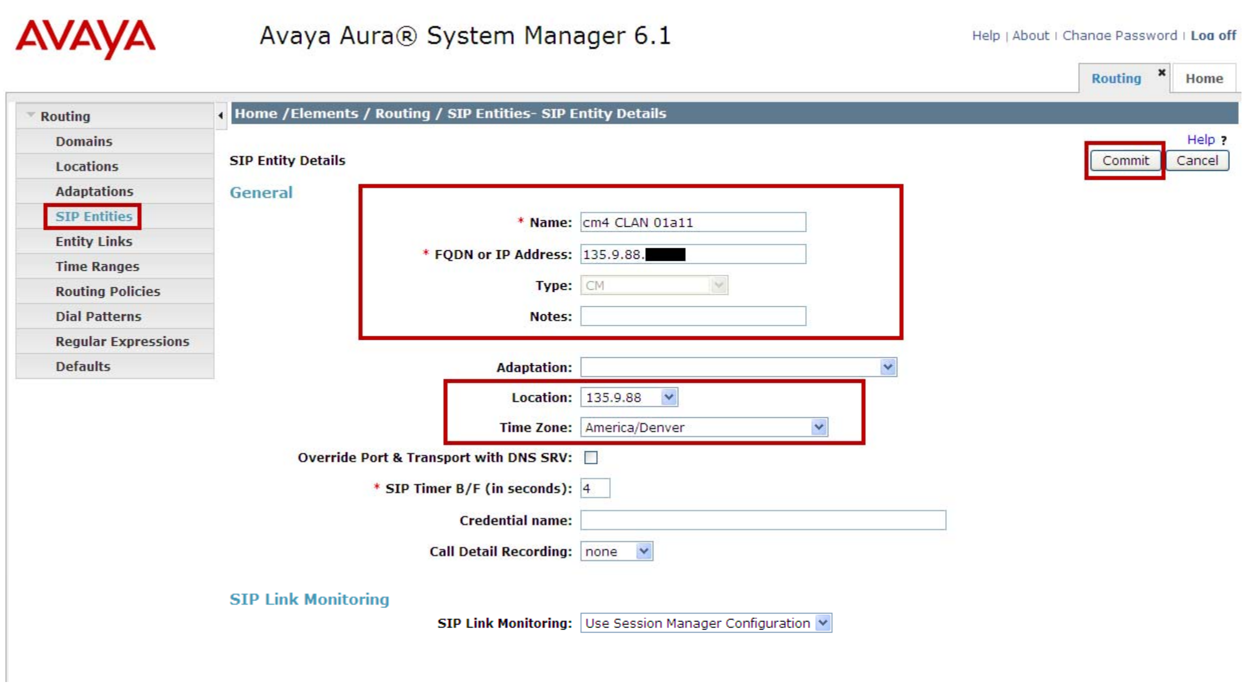
4.4 Add the CLAN interface for Avaya Aura® Communication Manager as a SIP Entity

Configure the SIP Entity for the sending Communication Manager (CM4 as depicted in **Figure 1**). Select **Routing** and select **SIP Entities.** Click **New** (not shown)**.** In the **General** Section, add the following:

- **Name:** Add an identifier for the Communication Manager
- **FQDN or IP Address:** Enter the IP Address of Communication Manager CLAN
- **Type:** Select **CM**
- **Notes:** Add a brief description
- **Location:** From the drop down select the ***first*** Location added in **Section 4.2**
- **Time Zone:** From the drop down select the appropriate time zone

Click **Commit** to save.

Note: IP addresses have been partially hidden for security.



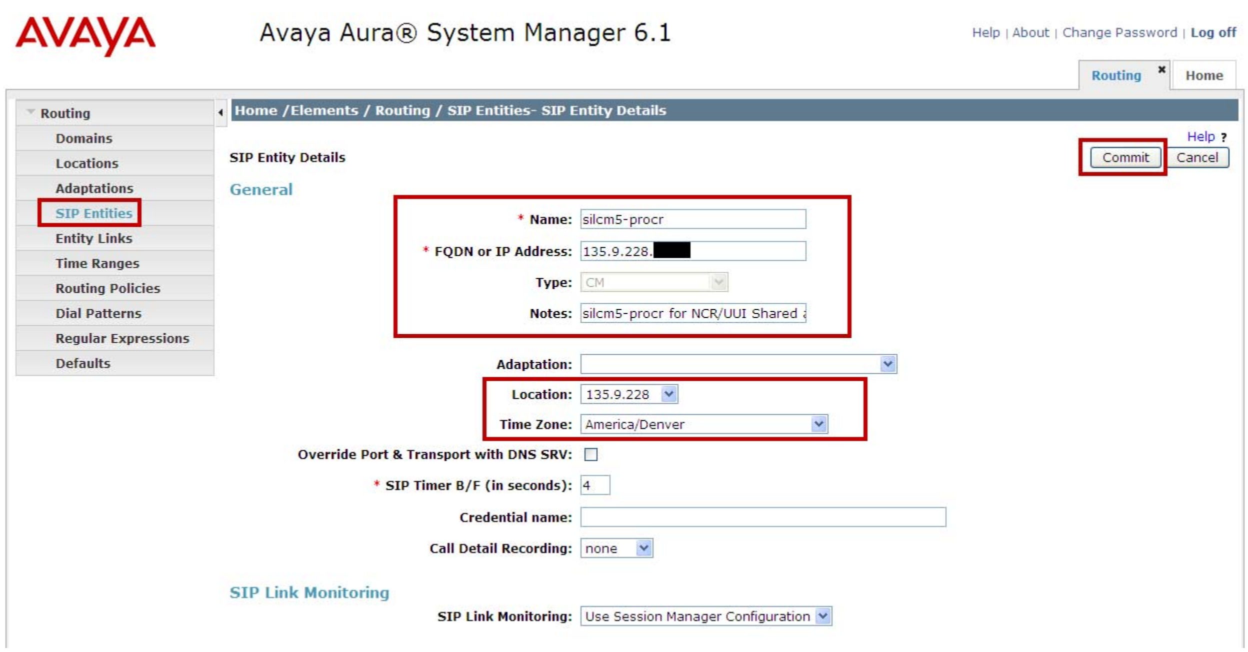
4.5 Add the PROCR interface for Avaya Aura® Communication Manager as a SIP Entity

Configure the SIP Entity for the receiving Communication Manager (CM5 depicted in **Figure 1**). Select **Routing** and select **SIP Entities.** Click **New** (not shown)**.** In the **General** Section, add the following:

- **Name:** Add an identifier for the Communication Manager
- **FQDN or IP Address:** Enter the IP Address of Communication Manager PROCR interface
- **Type:** Select **CM**
- **Notes:** Add a brief description
- **Location:** From the drop down select the ***second*** Location added in **Section 4.2**
- **Time Zone:** From the drop down select the appropriate time zone

Click **Commit** to save.

Note: IP addresses have been partially hidden for security.

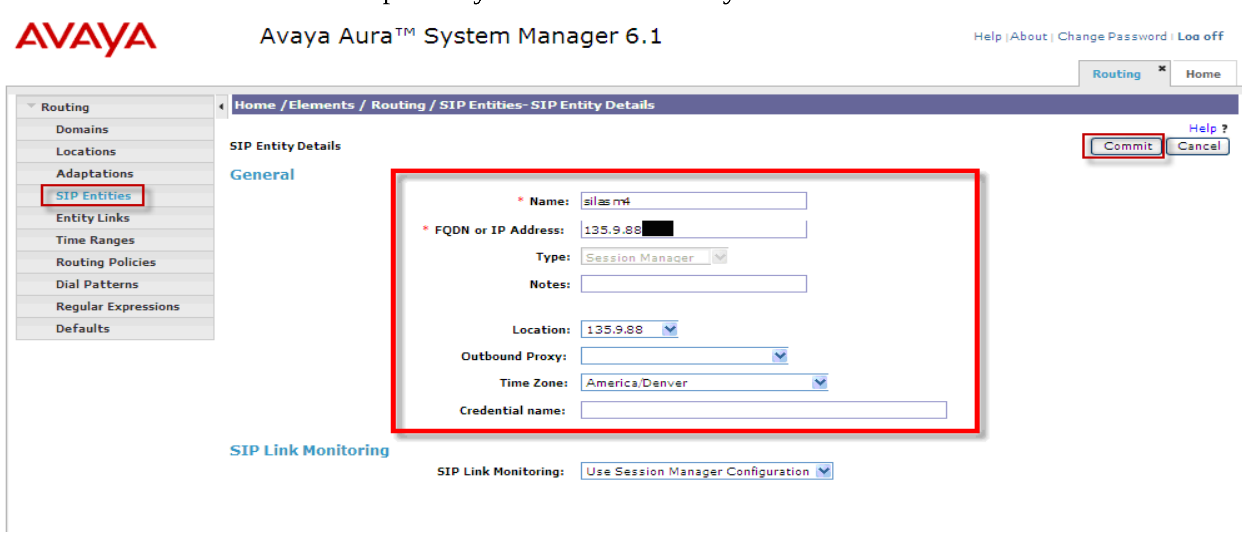


4.6 Add Avaya Aura® Session Manager as a SIP Entity

One of the first steps in properly setting up Session Manager and System Manager is to add Session Manager as a SIP Entity. Generally, this is done during the initial installation of Session Manager and System Manager. Select **Routing** and select **SIP Entities.** Click **New** (not shown)**.** In the **General** Section, add the following:

- **Name:** Add an identifier for the Session Manager
- **FQDN or IP Address:** Enter the IP Address of Session Manager SIP interface
- **Type:** Select **Session Manager**
- **Notes:** Add a brief description
- **Location:** From the drop down select the ***first*** Location added in **Section 4.2**
- **Time Zone:** From the drop down select the appropriate time zone

Note: IP addresses have been partially hidden for security.

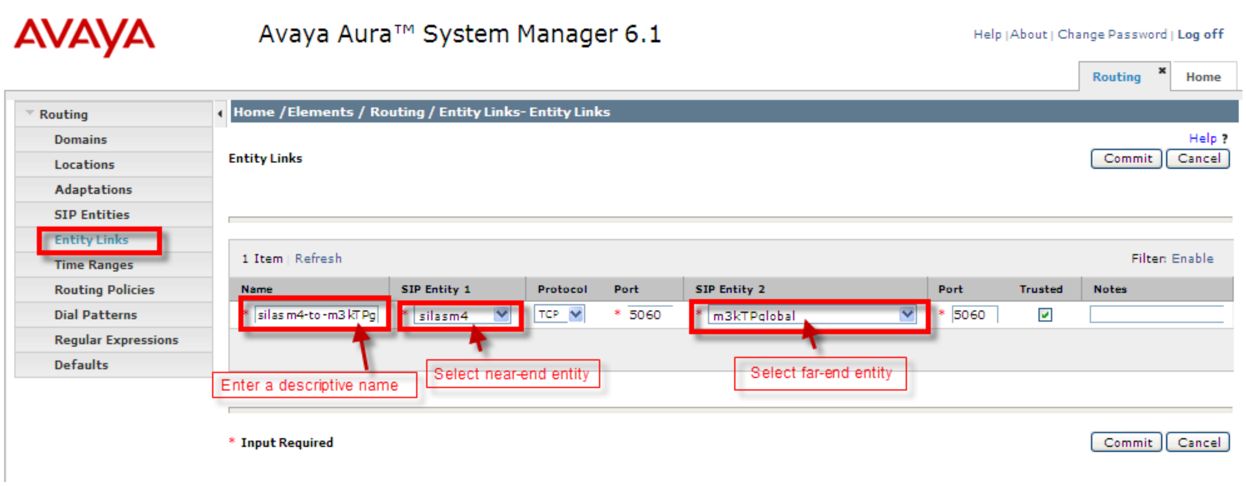


4.7 Define an Entity Link for the AudioCodes Mediant™ 3000 Gateway

Configure Entity Link for the M3K. Select **Routing** and select **Entity Links**. Click New (not shown). In the **Entity Links** section, add the following:

- **Name:** Enter an identifier for M3K
- **SIP Element 1:** From dropdown, select the SIP Entity added in **Section 4.6** for Session Manager
- **Protocol:** From dropdown select the required protocol
- **Port:** Enter the correct port for the Session Manager
- **SIP Element 2:** From dropdown, select the SIP Entity added in **Section 4.3** for M3K
- **Port:** Enter the correct port for M3K
- **Trusted:** Ensure the ticked box is clicked
- **Notes:** Add a brief description

Click **Commit** to save.

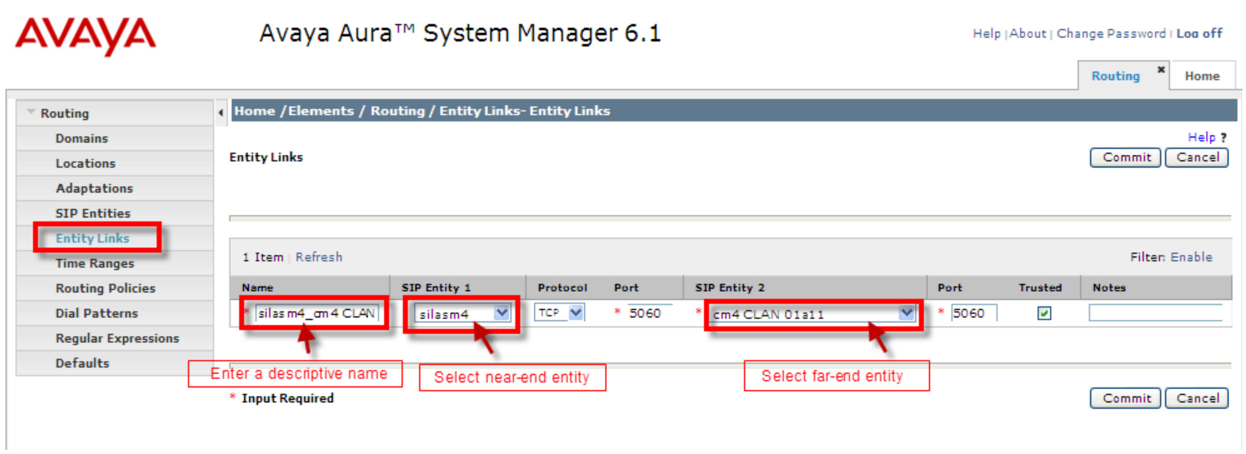


4.8 Define an Entity Link for Avaya Aura® Communication Manager

Configure Entity Link for the *sending* Communication Manager (CM4 depicted in **Figure 1**). Select **Routing** and select **Entity Links**. Click **New** (not shown). In the **Entity Links** section, add the following:

- **Name:** Enter an identifier for sending Communication Manager
- **SIP Element 1:** From dropdown, select the SIP Entity added in **Section 4.6** for Session Manager
- **Protocol:** From dropdown select the required protocol
- **Port:** Enter the correct port for the Session Manager
- **SIP Element 2:** From dropdown, select the SIP Entity added in **Section 4.4** for sending Communication Manager
- **Port:** Enter the correct port
- **Trusted:** Ensure the ticked box is clicked
- **Notes:** Add a brief description

Click **Commit** to save.

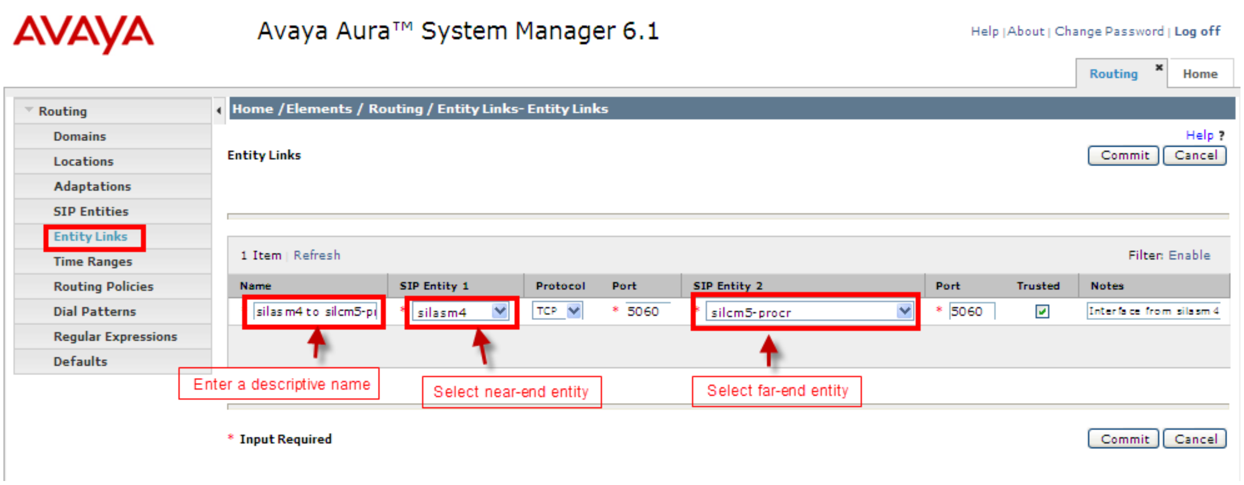


4.9 Define an Entity Link for Avaya Aura® the Communication Manager

Configure Entity Link for the *receiving* Communication Manager (CM5 depicted in **Figure 1**). Select **Routing** and select **Entity Links**. Click **New (**not shown). In the **Entity Links** section, add the following:

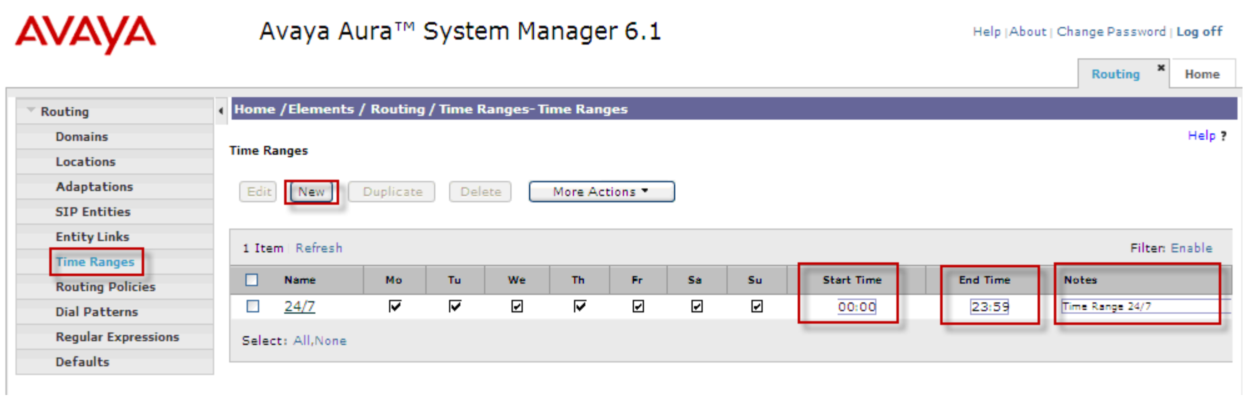
- **Name:** Enter an identifier for CM5 Communication Manager
- **SIP Element 1:** From dropdown, select the SIP Entity added in **Section 4.6** for Session Manager
- **Protocol:** From dropdown select the required protocol
- **Port:** Enter the correct port for the Session Manager
- **SIP Element 2:** From dropdown, select the SIP Entity added in **Section 4.5** for CM5 Communication Manager
- **Port:** Enter the correct port
- **Trusted:** Ensure the ticked box is clicked
- **Notes:** Add a brief description

Click **Commit** to save.



4.10 Setup Time Ranges

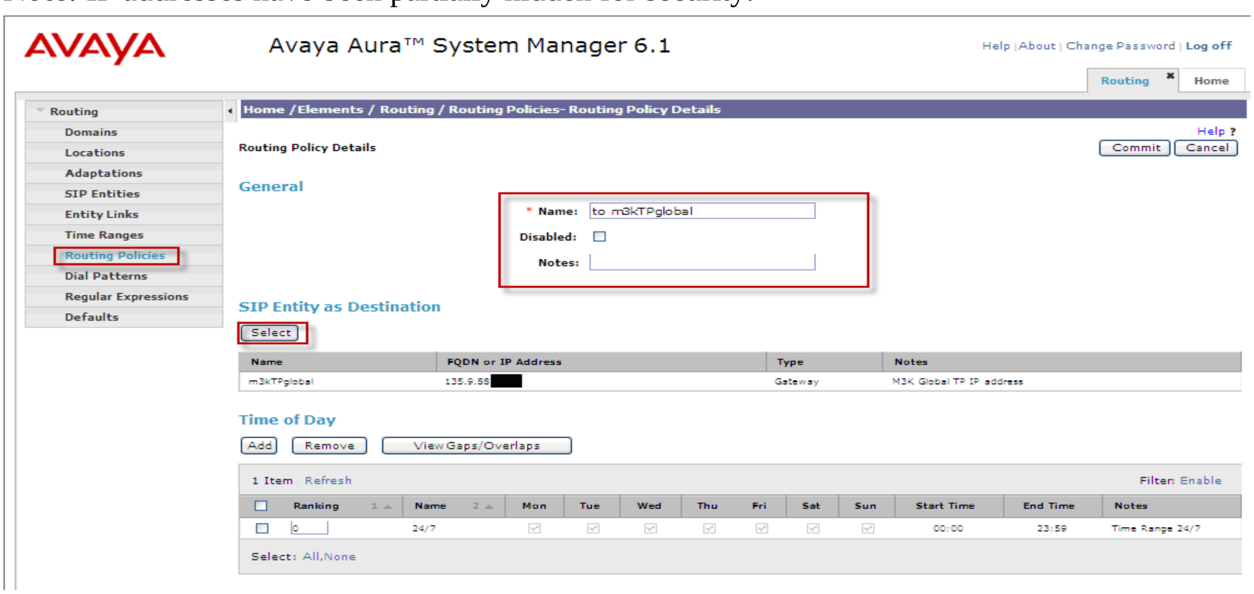
Configure the Time Ranges. Select **Routing** and select **Time Ranges**. Click **New**. Under **Name** enter an identifier. Select the days of the week, enter time values for **Start Time** and **End Time**. Under **Notes** add a brief description. When completed, click **Commit** (not shown) to save. Screenshot below shows the updated information.



4.11 Define Policies and Time of Day for the AudioCodes Mediant™ 3000 Gateway

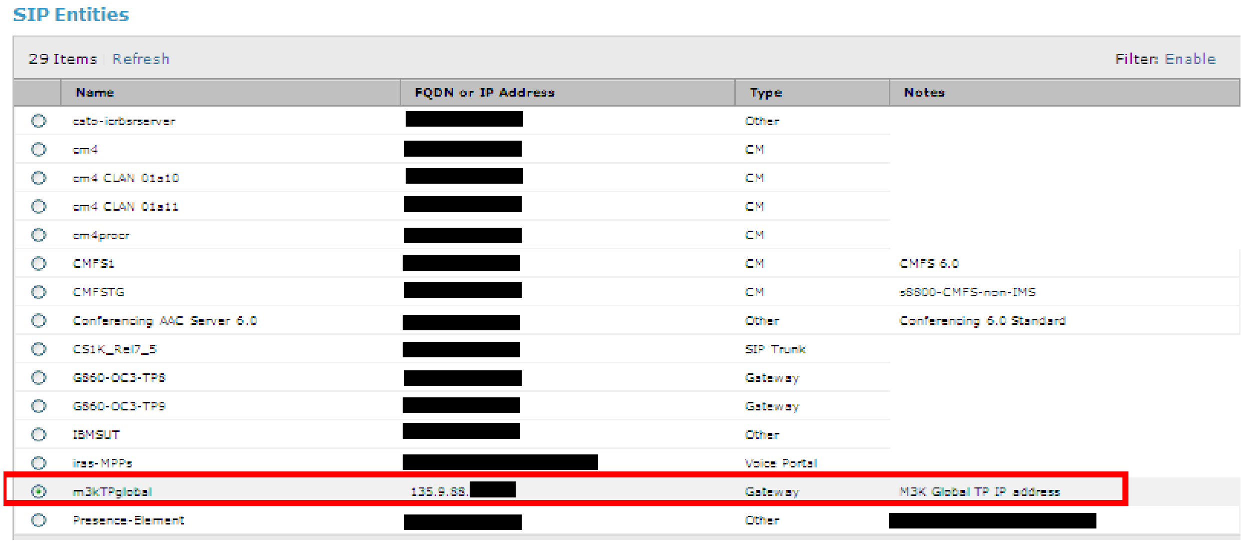
Configure Policies and Time of Day for the M3K. Select **Routing** and select **Routing Policies**. Click **New** (not shown). In the **General** section under **Name**, add an identifier to define the routing policy for the M3K. Under **Notes** add a brief description. In the **SIP Entity as Destination** section click on **Select**.

Note: IP addresses have been partially hidden for security.



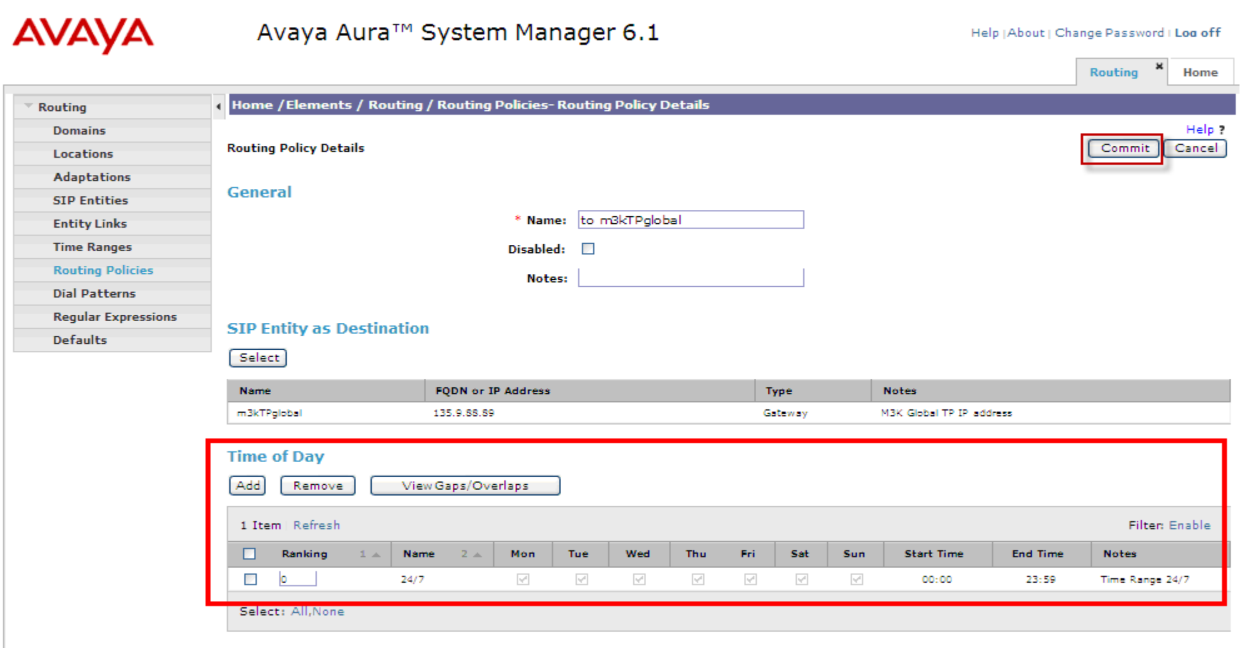
The **SIP Entity List** page opens. Select the entry of the M3K added in **Section 4.3**, and click on **Select** (not shown)**.**

Note: IP addresses have been partially hidden for security.



The selected SIP Entity displays on the **Routing Policy Details** page. In the **Time of Day** section, click on **Add**, and from the **Time Ranges List** page (not shown), select the desired **Time Range** (not shown) and click on **Select** (not shown).

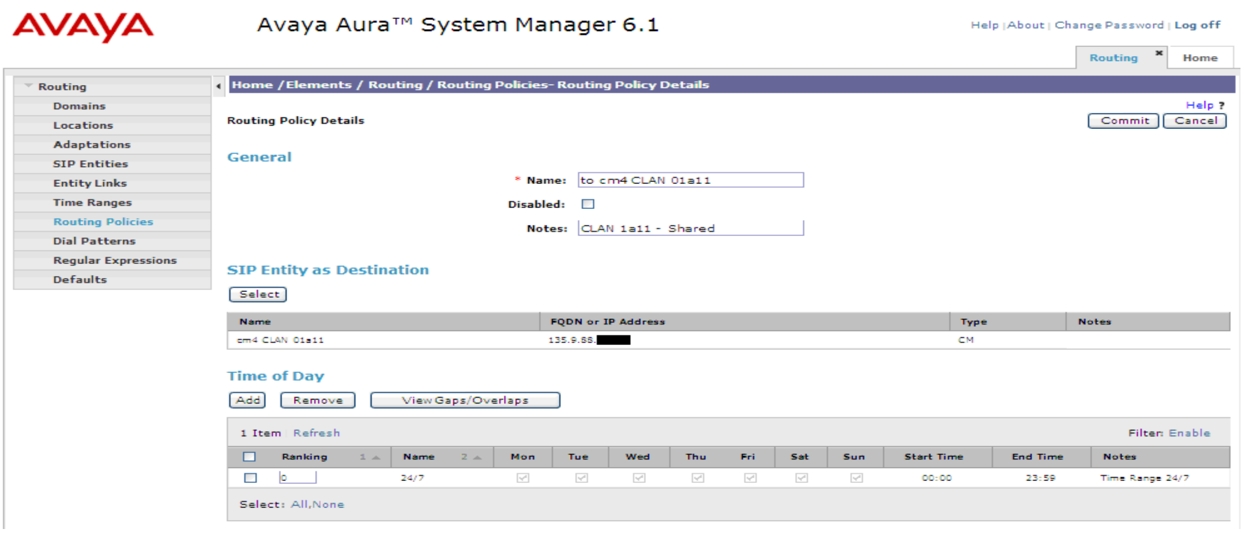
The selected Time Range displays on the **Routing Policy Details** page. Click on **Commit** to save.



4.12 Define Policies and Time of Day for Sending Communication Manager

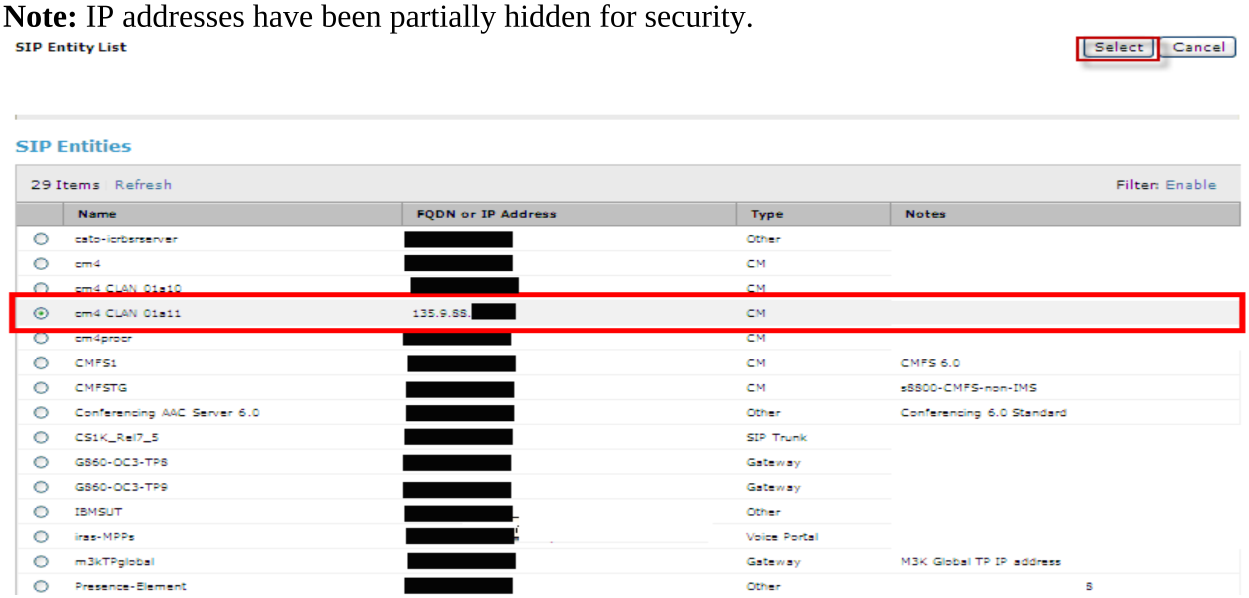
Configure Policies and Time of Day for the *sending* Communication Manager (CM4 depicted in Figure 1). Select **Routing** and select **Routing Policies**. Click **New** (not shown). In the **General** section under **Name**, add an identifier to define the routing policy for the CM4 Communication Manager. Under **Notes** add a brief description. In the **SIP Entity as Destination** section click on **Select**.

Note: IP addresses have been partially hidden for security.



The **SIP Entity List** page opens. Select the entry of the sending Communication Manager added in **Section 4.4**, and click on **Select.**

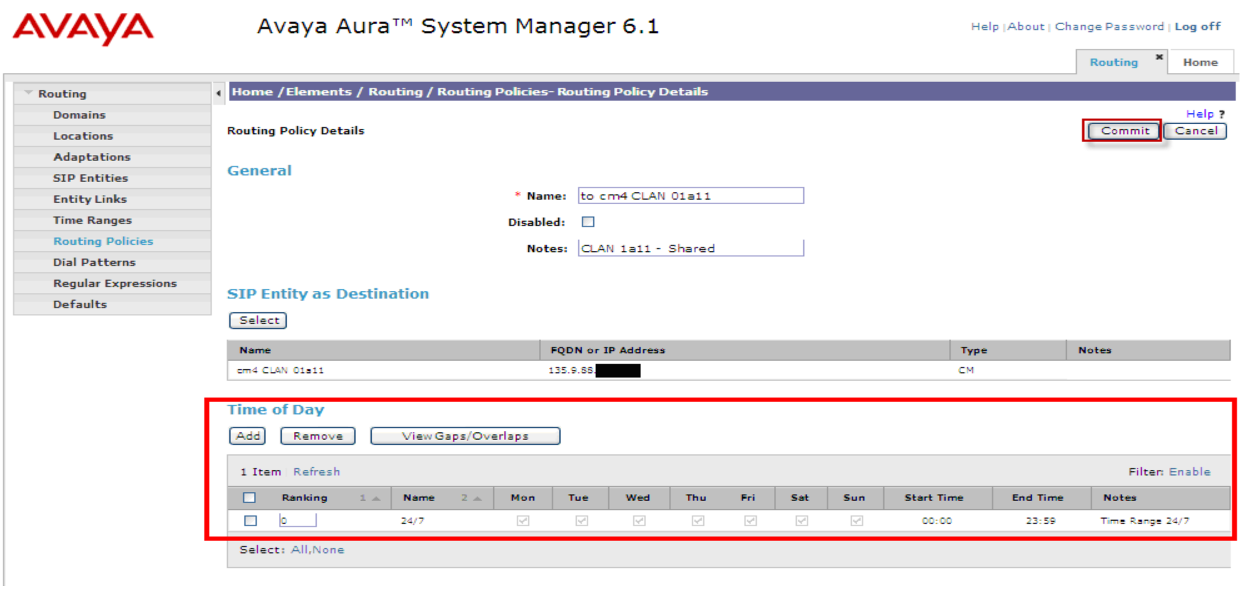
Note: IP addresses have been partially hidden for security.



The selected SIP Entity displays on the **Routing Policy Details** page. In the **Time of Day** section, click on **Add**, and from the **Time Ranges List** page (not shown), select the desired **Time Range** and click on **Select**.

The selected Time Range displays on the **Routing Policy Details** page. Click on **Commit** to save.

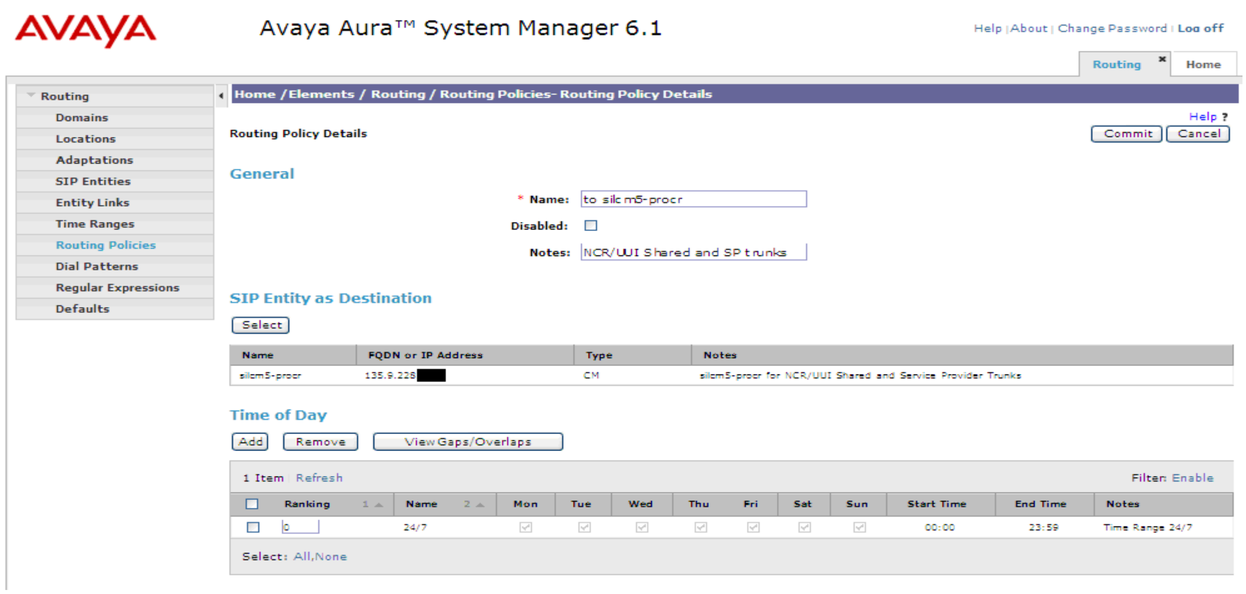
Note: IP addresses have been partially hidden for security.



4.13 Define Policies and Time of Day for Receiving Communication Manager

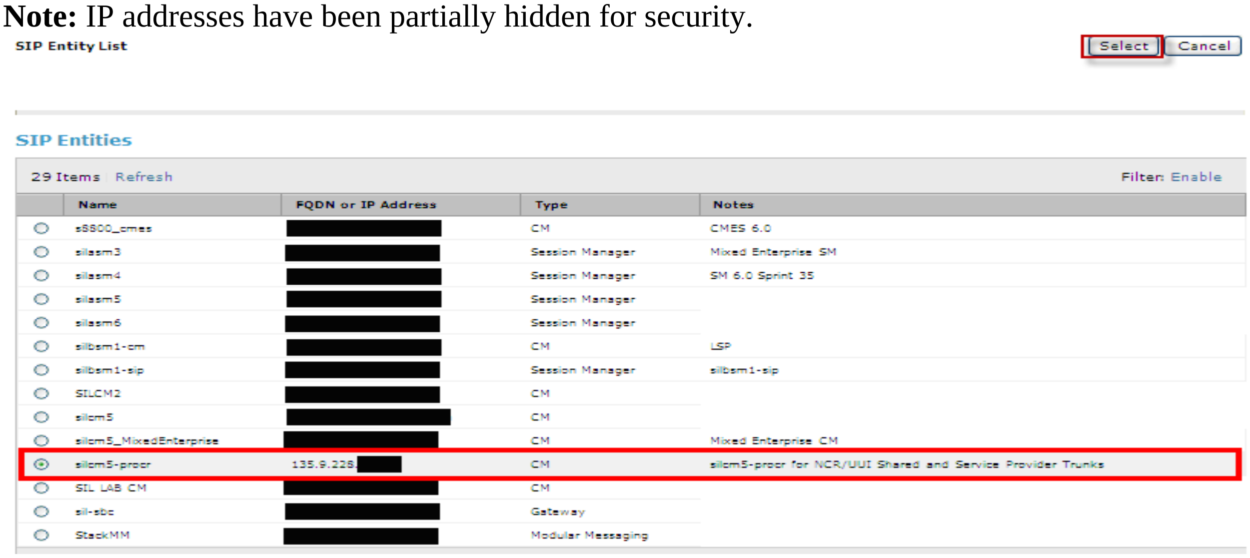
Configure Policies and Time of Day for the *receiving* Communication Manager. Select **Routing** and select **Routing Policies**. Click **New** (not shown). In the **General** section under Name, add an identifier to define the routing policy for the Communication Manager. Under **Notes** add a brief description. In the **SIP Entity as Destination** section click on Select.

Note: IP addresses have been partially hidden for security.



The **SIP Entity List** page opens. Select the entry of the Communication Manager added in **Section 4.5**, and click on **Select.**

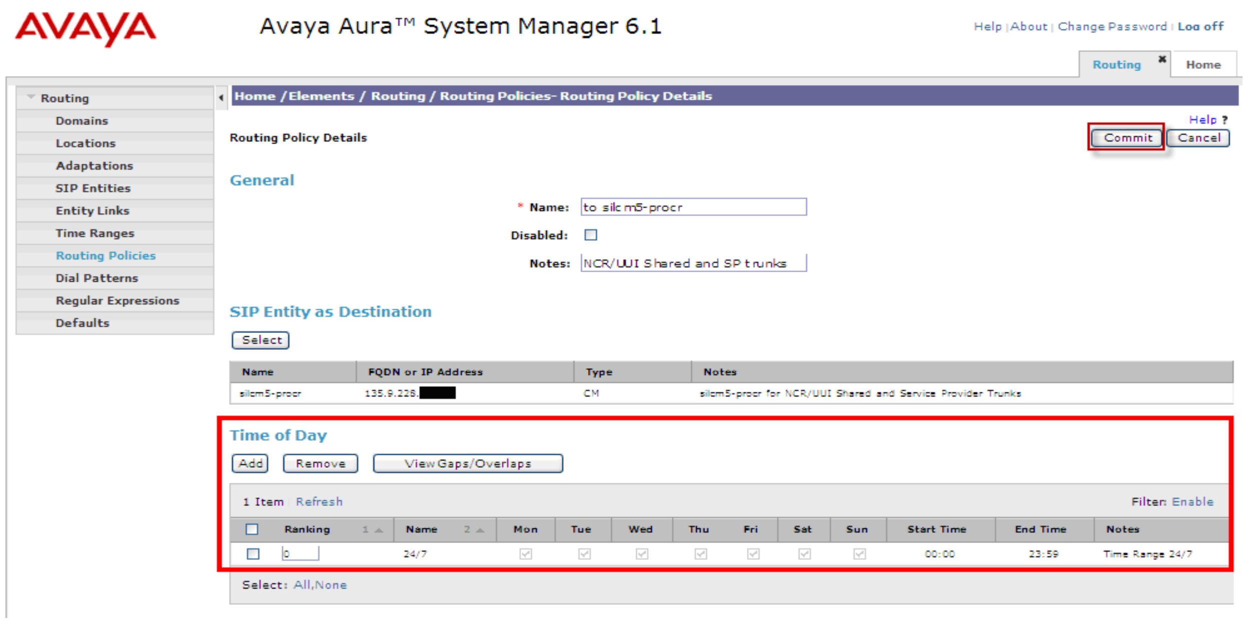
Note: IP addresses have been partially hidden for security.



The selected SIP Entity displays on the **Routing Policy Details** page. In the **Time of Day** section, click on **Add**, and from the **Time Ranges List** page (not shown), select the desired **Time Range** and click on **Select**.

The selected Time Range displays on the **Routing Policy Details** page. Click on **Commit** to save.

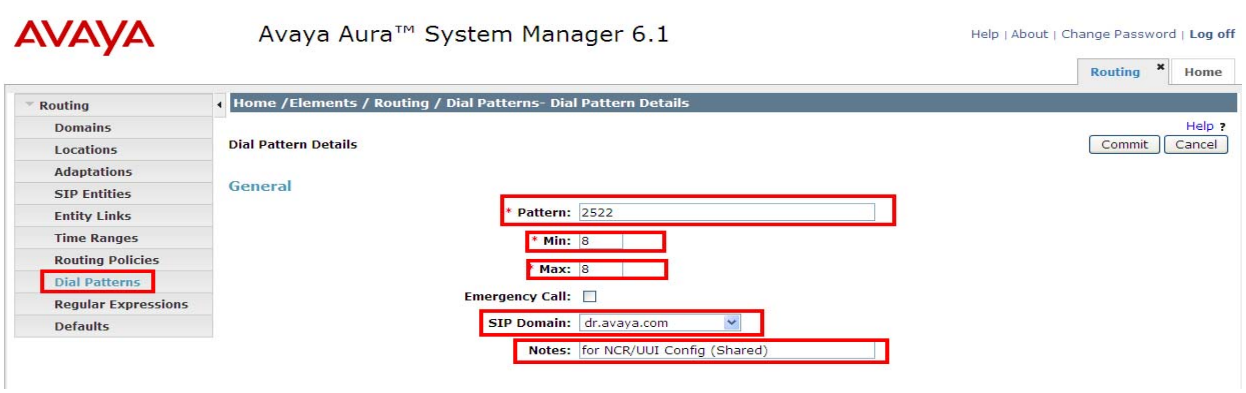
Note: IP addresses have been partially hidden for security.



4.14 Define Dial Plan for Sending Avaya Aura® Communication Manager

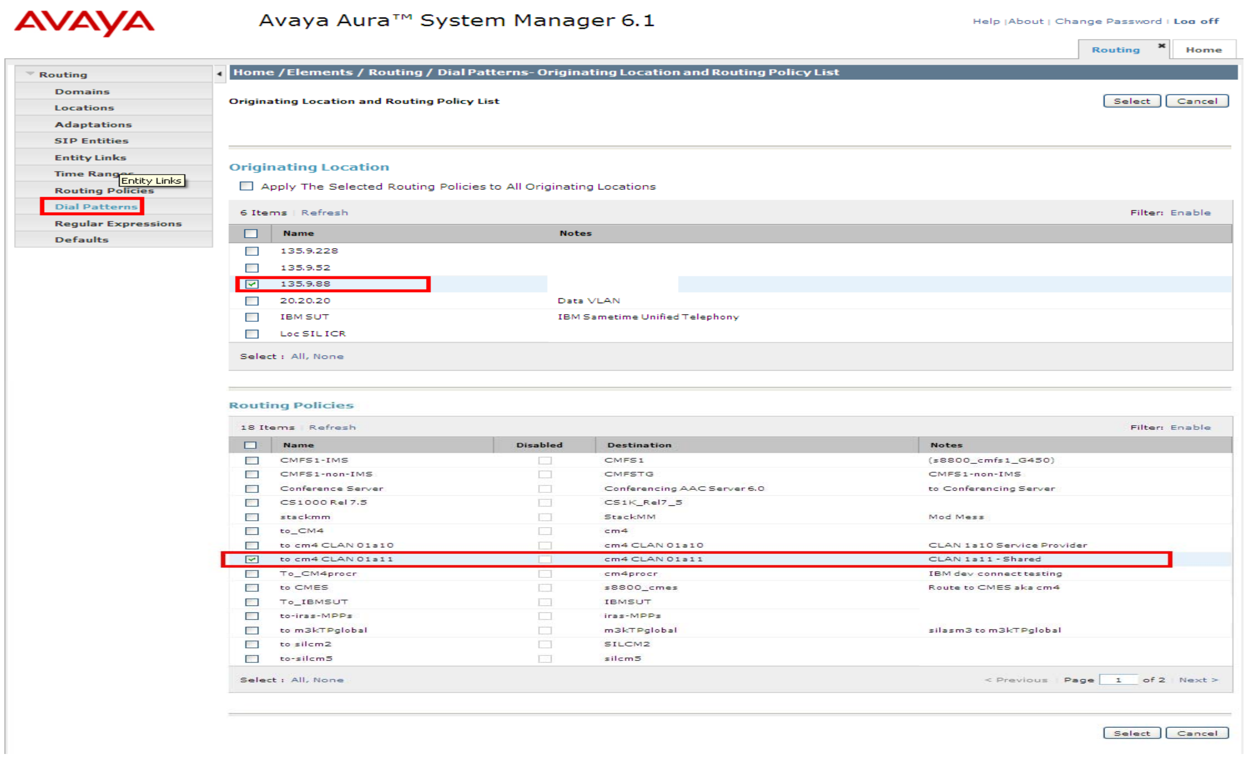
Configure the Dial Plan pattern. Select **Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, add the following:

- **Pattern:** 2522
- **Min:** 8
- **Max:** 8
- **Emergency Call:** Select if applicable
- **SIP Domain:** Select the SIP Domain added in **Section 4.1**
- **Notes:** Brief description



In the **Originating Locations and Routing Policies** section click **Add** (not shown).

In the **Originating Location** section, select the appropriate location. In the **Routing Policies** section, select the appropriate policy for routing. Click the **Select** button.

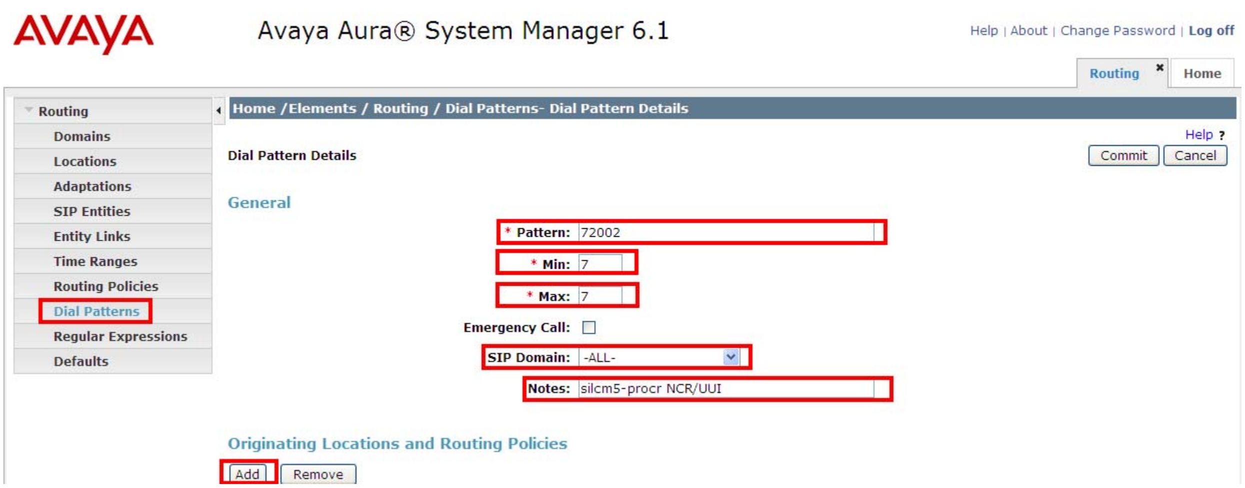


Click **Commit** to save new Dial Plan.

4.15 Define Dial Plan for Receiving Avaya Aura® Communication Manager

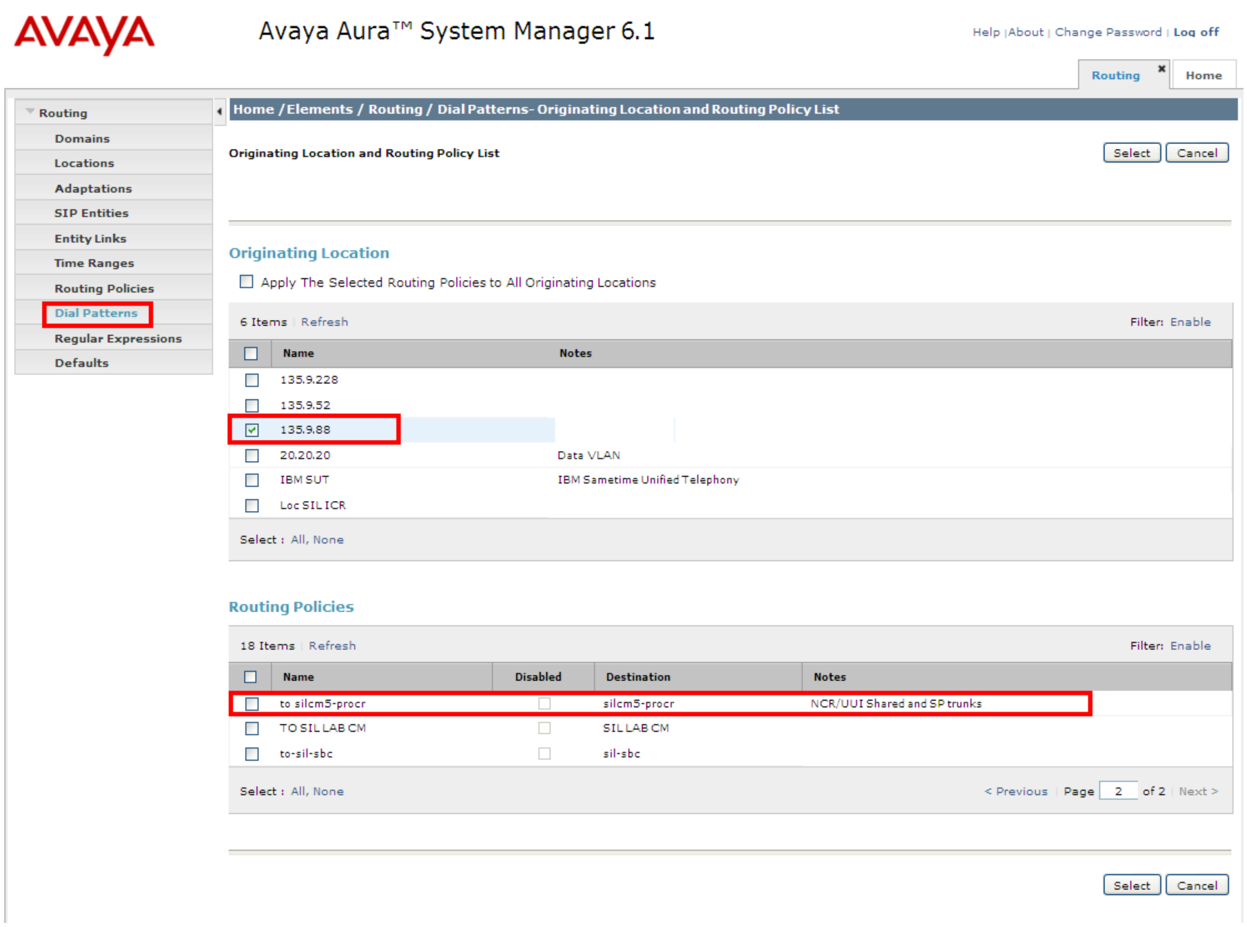
Configure the Dial Plan pattern. Select **Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, add the following:

- **Pattern:** 72002
- **Min:** 7
- **Max:** 7
- **Emergency Call:** Select if applicable
- **SIP Domain:** ALL
- **Notes:** Brief description



In the **Originating Locations and Routing Policies** section click **Add.**

In the **Originating Location** section, select the appropriate location. In the **Routing Policies** section, select the appropriate policy for routing. Click the **Select** button.



Click **Commit** to save new Dial Plan.

5. Configure the AudioCodes Mediant™ 3000 Gateway

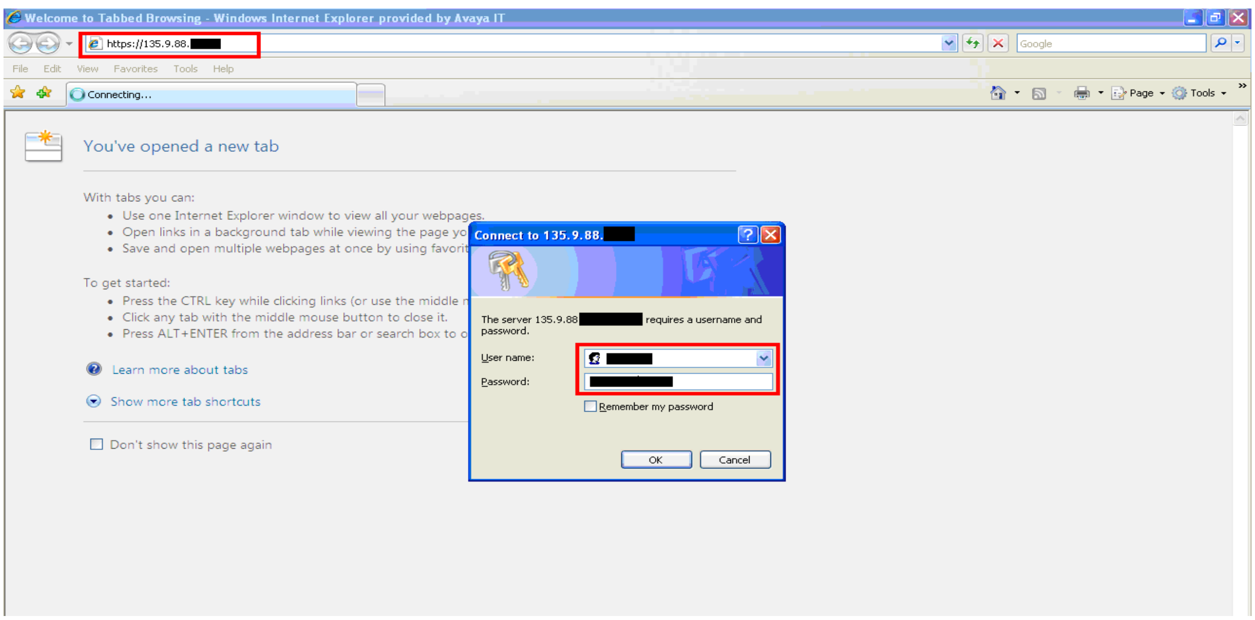
This section provides the procedures for configuring M3K using the web based graphical user interface. The procedures include the following areas:

- Select all configurable parameters
- Setting SIP Protocols and ports
- Setup Codec preferences
- Add a SIP Trunk Group
- Setup Tel to IP routing
- Setup Internal DNS Table
- Setup Tel-to-IP Alternative Routing

These Application Notes assume the M3K is already installed and functioning properly. See the reference documents listed in **Section 8** for the AudioCodes Mediant™ 3000 Gateway documents.

Log in to the M3K via an internet browser using the IP address or FQDN of the M3K with a login name having administrative rights.

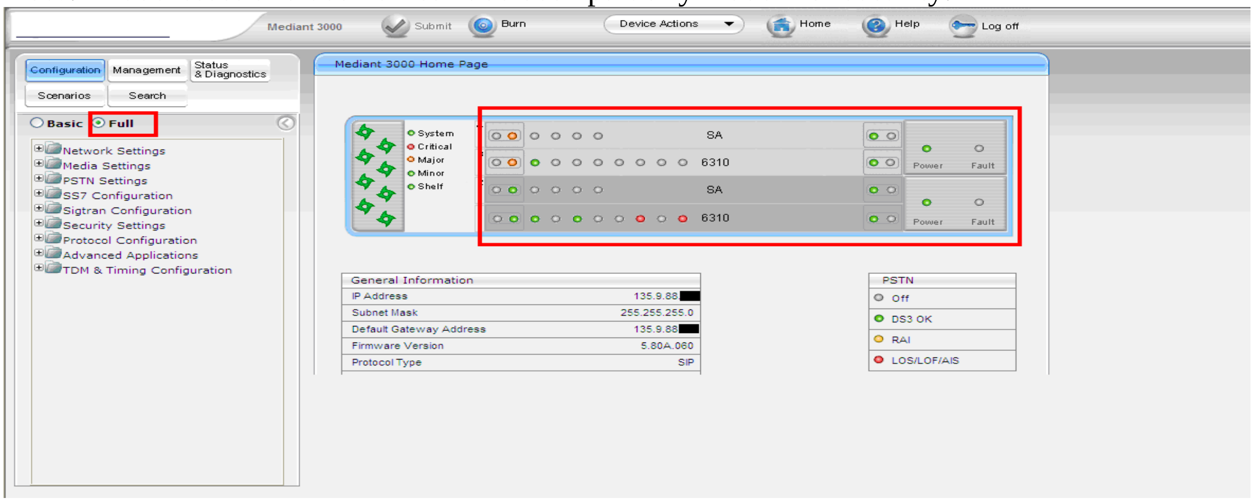
Note: IP addresses and usernames have been partially hidden for security.



5.1 Display all configurable parameters

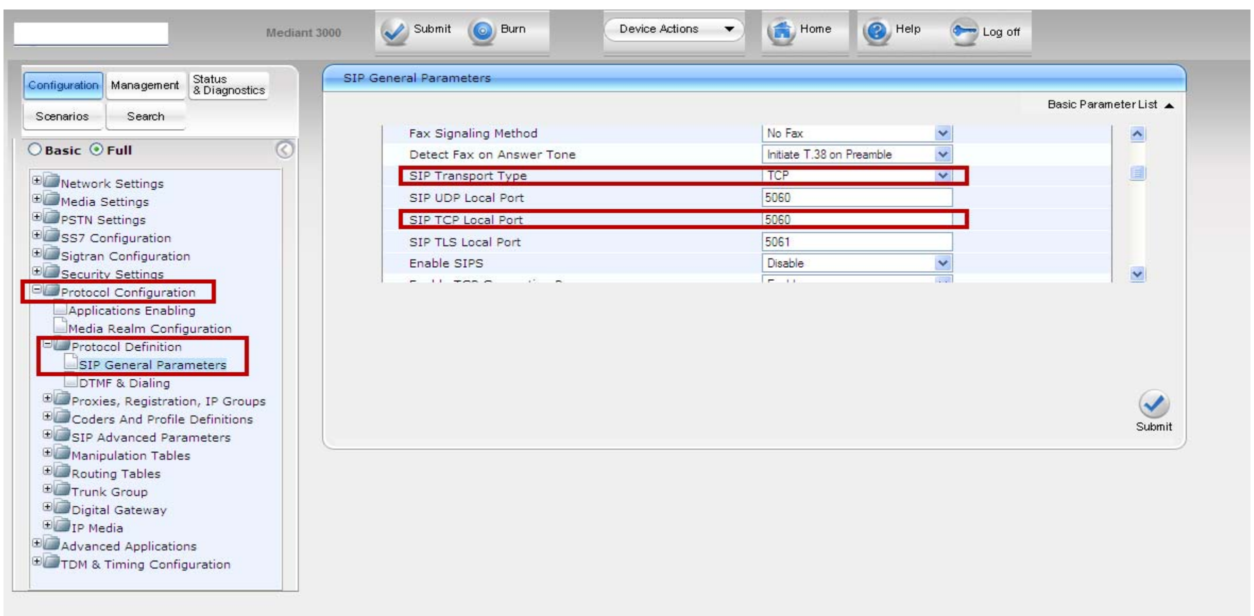
The Mediant 3000 Home Page will be displayed. To display all configurable parameters, select **Full** in the left pane. In the screenshot below, both TP6310 and SA boards are shown.

Note: IP addresses and usernames have been partially hidden for security.

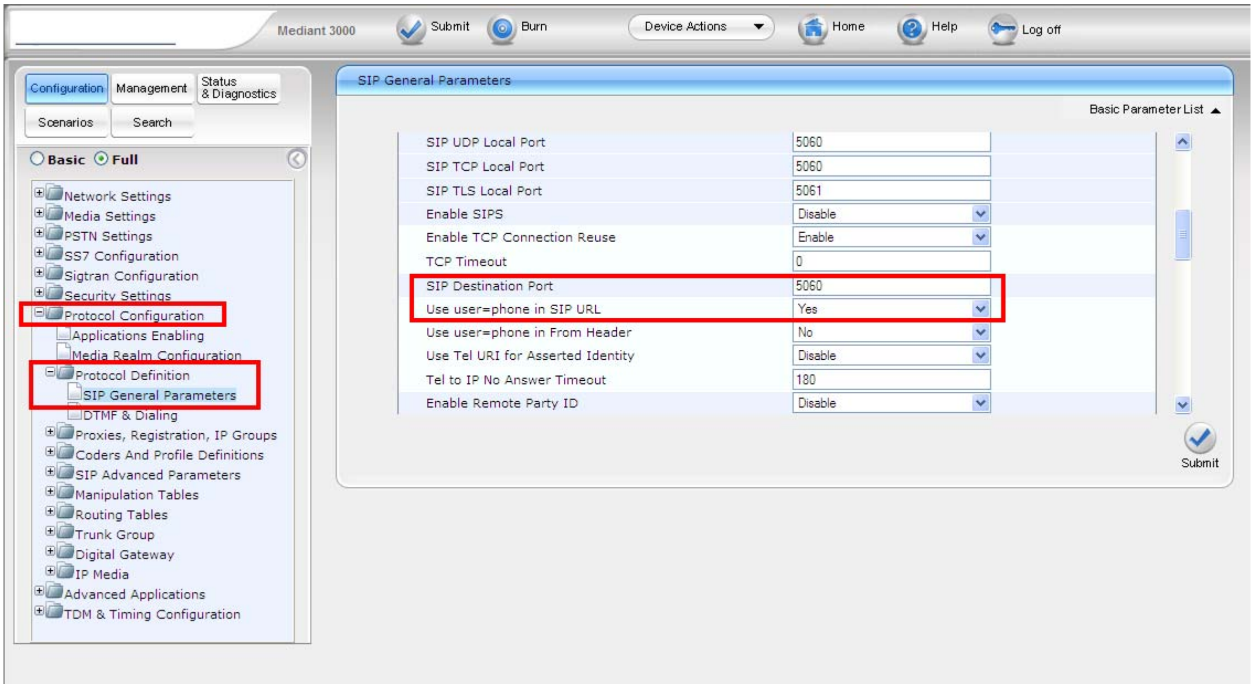


5.2 Set SIP Protocols and Ports

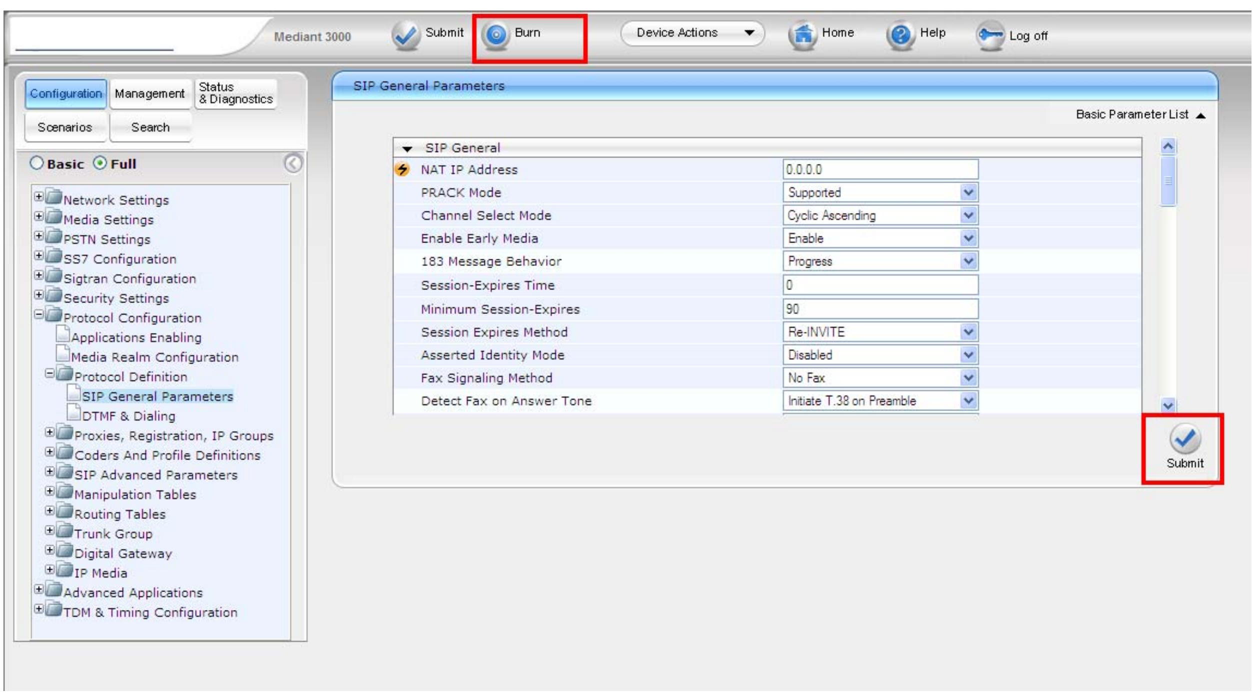
Double-click on **Protocol Configuration,** expand **Protocol Definition** and select **SIP General Parameters.** This will display the SIP General Parameters in the right pane. Scroll down and set the **SIP Transport Type** and **SIP TCP Local Port** to the values administered in **Section 4.7**



Scroll down further and set the **SIP Destination Port** and **Use user=phone in SIP URL** fields.

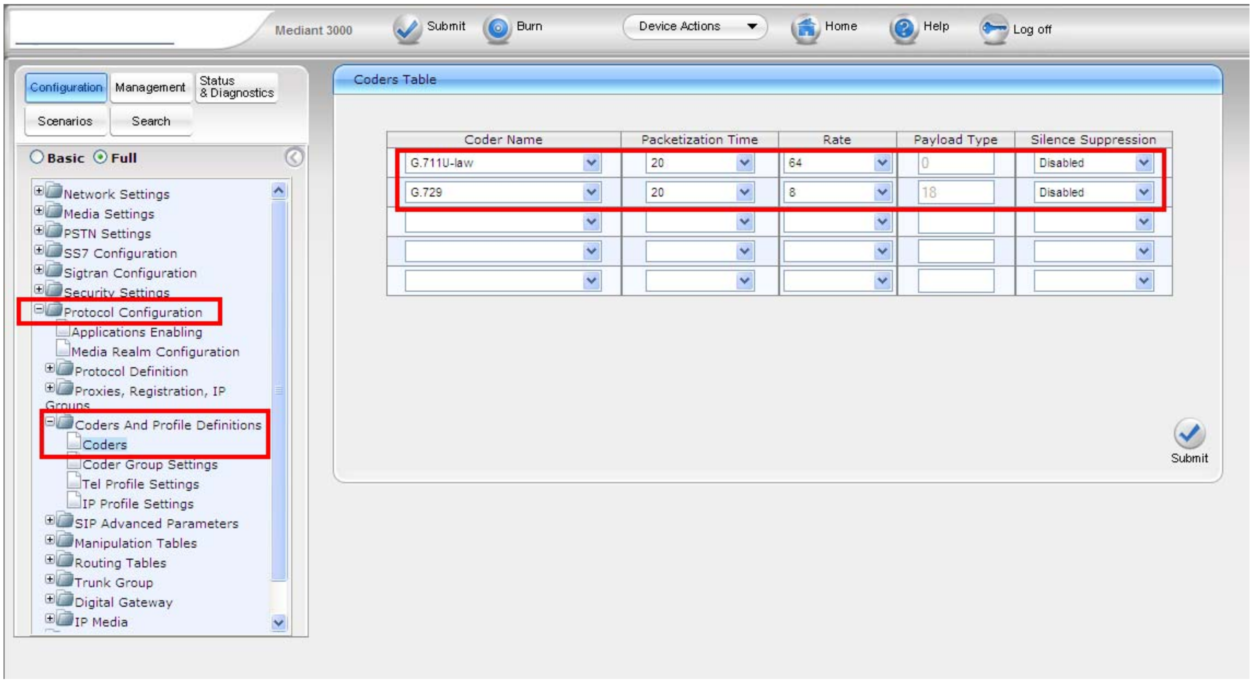


When finished editing the **SIP General Parameters**, click **Submit** and then **Burn** to save the changes.



5.3 Setup Codec Preferences/Order

Expand **Protocol Configuration** and **Coders And Profile Definitions** in the left pane. Click on **Coders** to display the **Coders Table** in the right pane. Add entries for the codecs administered is **Section 3.3**.

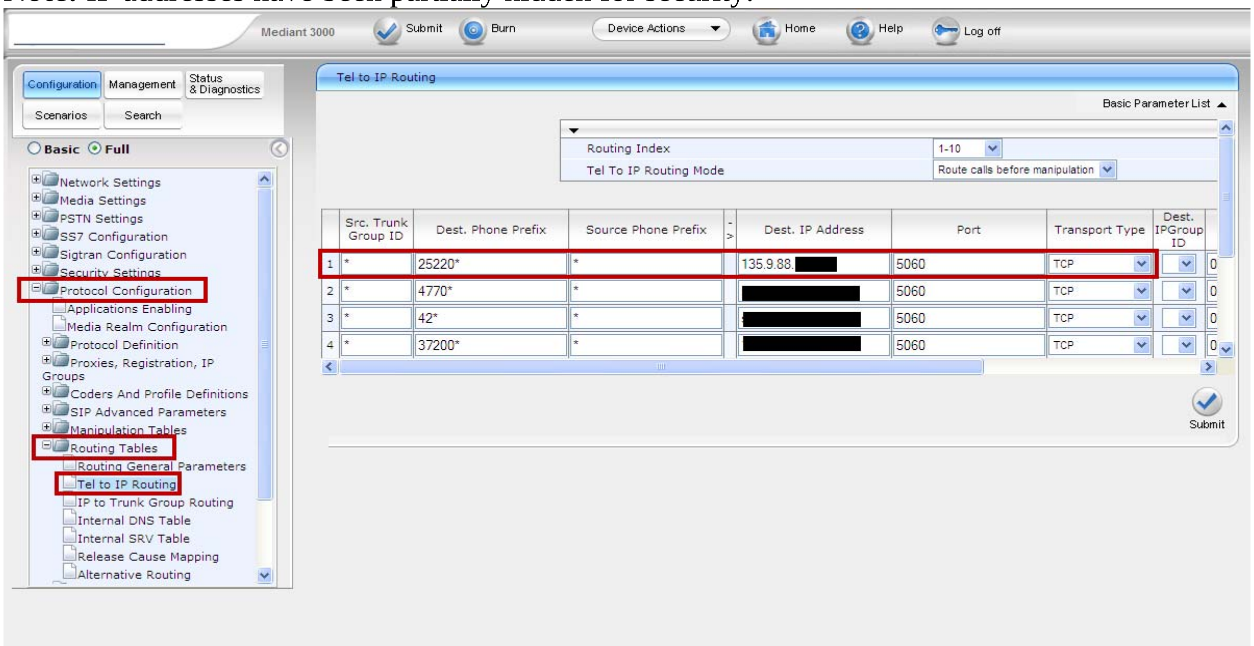


5.4 Setup Tel to IP Routing to Route Calls between Session Manager and Sending Communication Manager

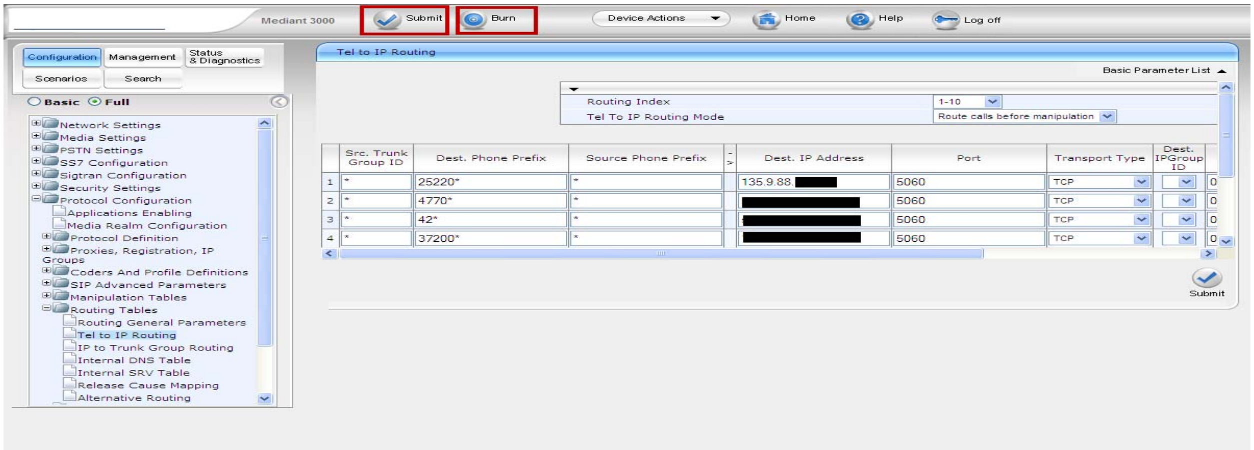
Expand **Protocol Configuration** and **Routing Tables** in the left pane. Select **Tel to IP Routing.** The **Tel to IP Routing** table is displayed in the right pane. Select the desired row of the table and enter routing information:

- **Src. Trunk Group ID:** *
- **Dest. Phone Prefix:** 25220*
- **Source Phone Prefix:** *
- **Dest. IP Address:** IP or FQDN of Avaya Aura® Session Manager
- **Port:** Port administered from **Section 4.7**
- **Transport Type:** Transport Protocol administered from **Section 4.7**

Note: IP addresses have been partially hidden for security.



Click **Submit** and then **Burn**.

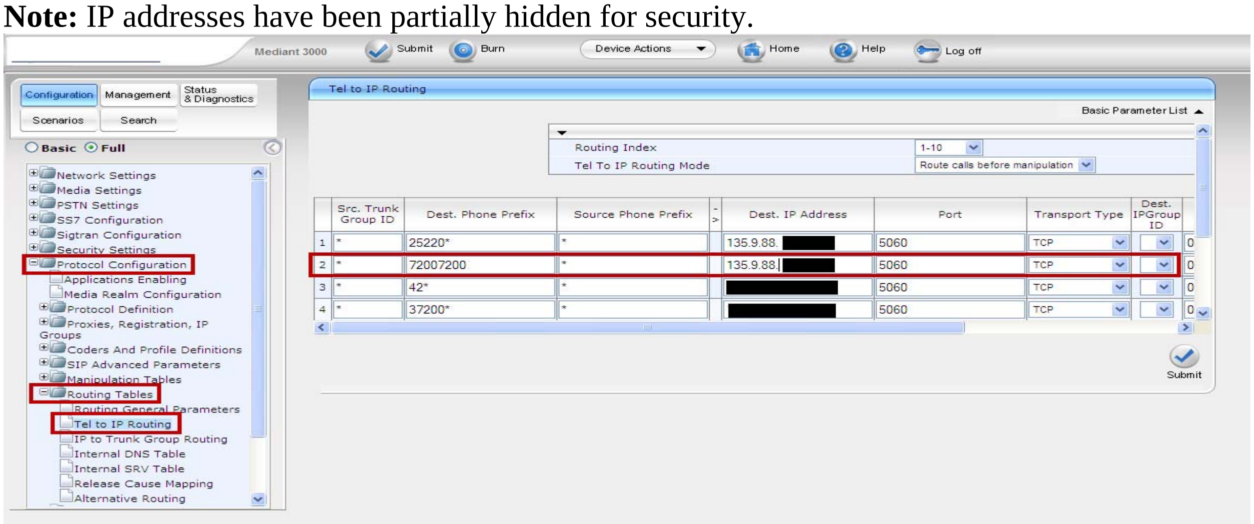


5.5 Setup Tel to IP Routing to Route Calls between Session Manager and Receiving Communication Manager

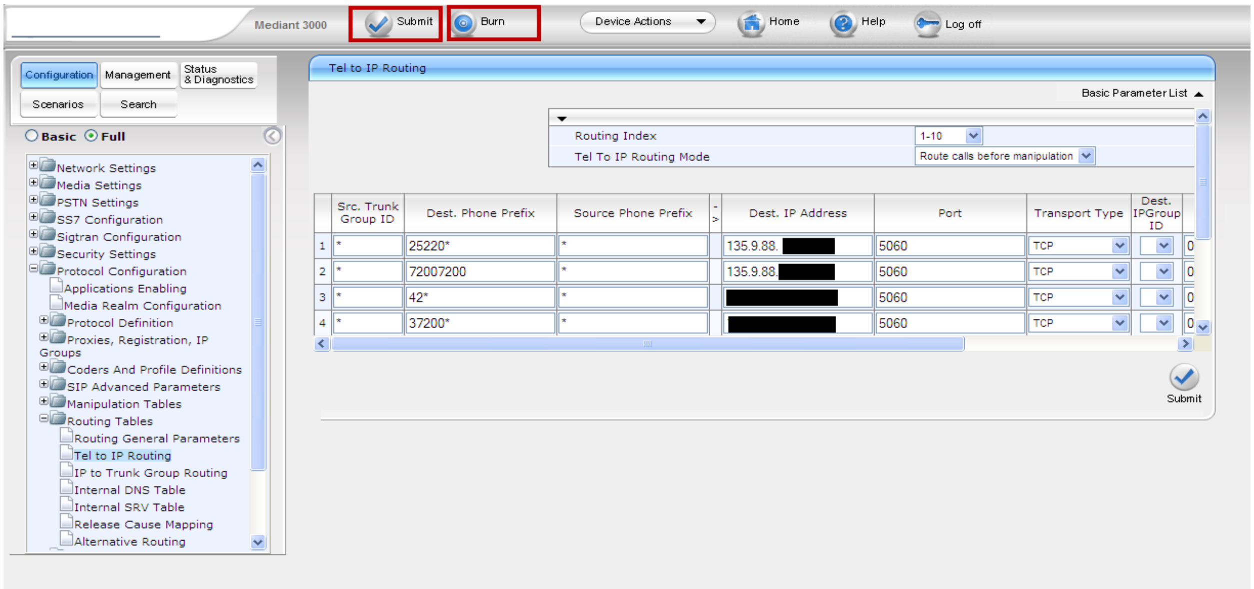
Expand **Protocol Configuration** and **Routing Tables** in the left pane. Select **Tel to IP Routing**. The **Tel to IP Routing** table is displayed in the right pane. Select the desired row of the table and enter routing information:

- **Src. Trunk Group ID:** *
- **Dest. Phone Prefix:** 7200200
- **Source Phone Prefix:** *
- **Dest. IP Address:** IP or FQDN of Avaya Aura® Session Manager
- **Port:** Port administered from **Section 4.7**
- **Transport Type:** Transport Protocol administered from **Section 4.7**

Note: IP addresses have been partially hidden for security.



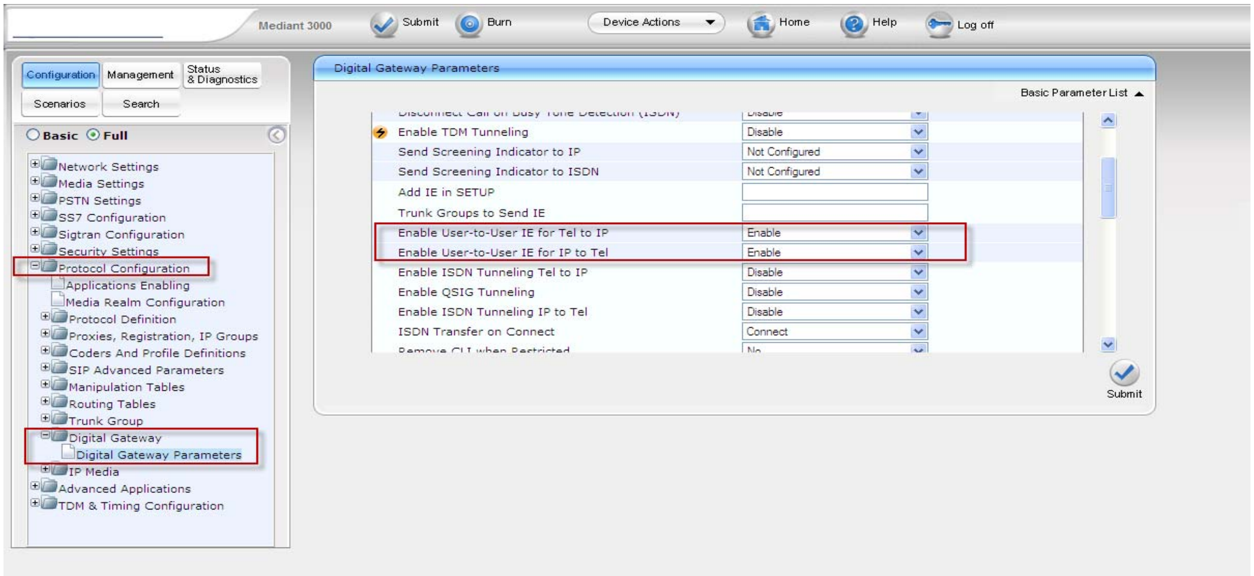
Click **Submit** and then **Burn**.



5.6 Verify the Protocol Configuration

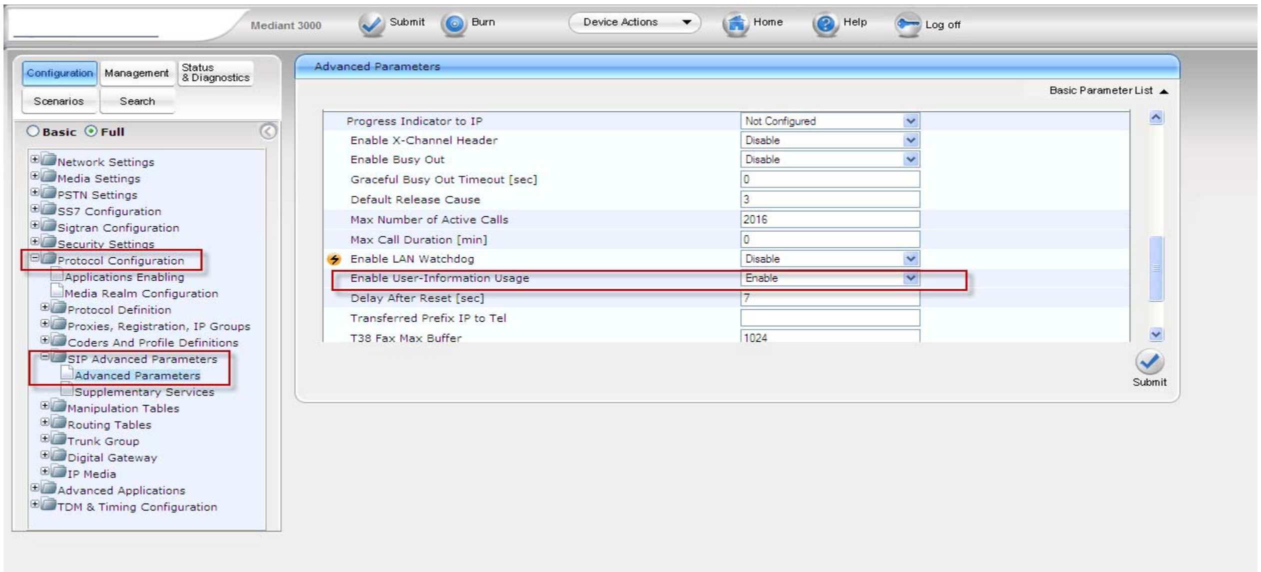
5.6.1 Verify the Digital Gateway Parameters

Expand **Protocol Configuration** and **Digital Gateway** in the left pane. Select **Digital Gateway Parameters**. The **Digital Gateway Parameters** table is displayed in the right pane. Scroll down and verify **Enable User-to-User IE for Tel to IP** and **Enable User-to-User IE for IP to Tel** are both enabled.



5.6.2 Verify the SIP Advanced Parameters

Expand **Protocol Configuration** and **SIP Advanced Parameters** in the left pane. Select **Advanced Parameters.** The **Advanced Parameters** table is displayed in the right pane. Scroll down and verify **Enable User-Information Usage** is enabled.



6. Verification Steps

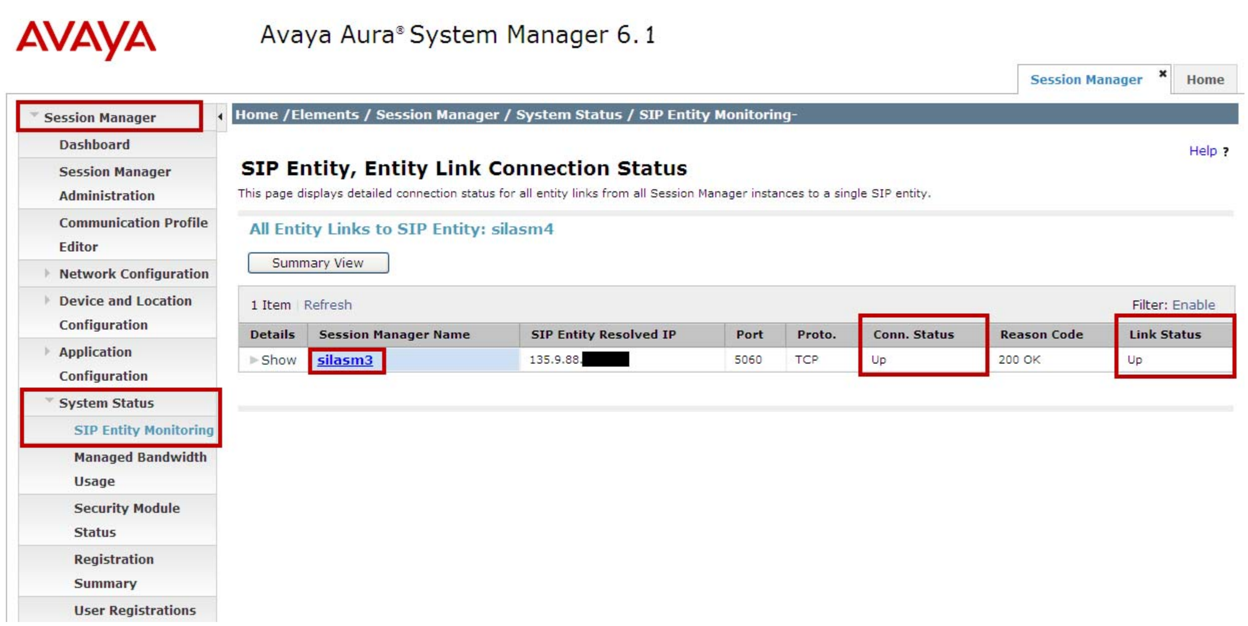
6.1 Verify AudioCodes Mediant™ 3000 Gateway Configuration

- Log into the web interface and click **Statistics & Diagnostics** from the left pane. Expand **Status & Diagnostics** (not shown) and select **Message Log**.
- Place initial test customer call and verify proper M3K routing by inspecting the SIP messages and SIP message sequence.

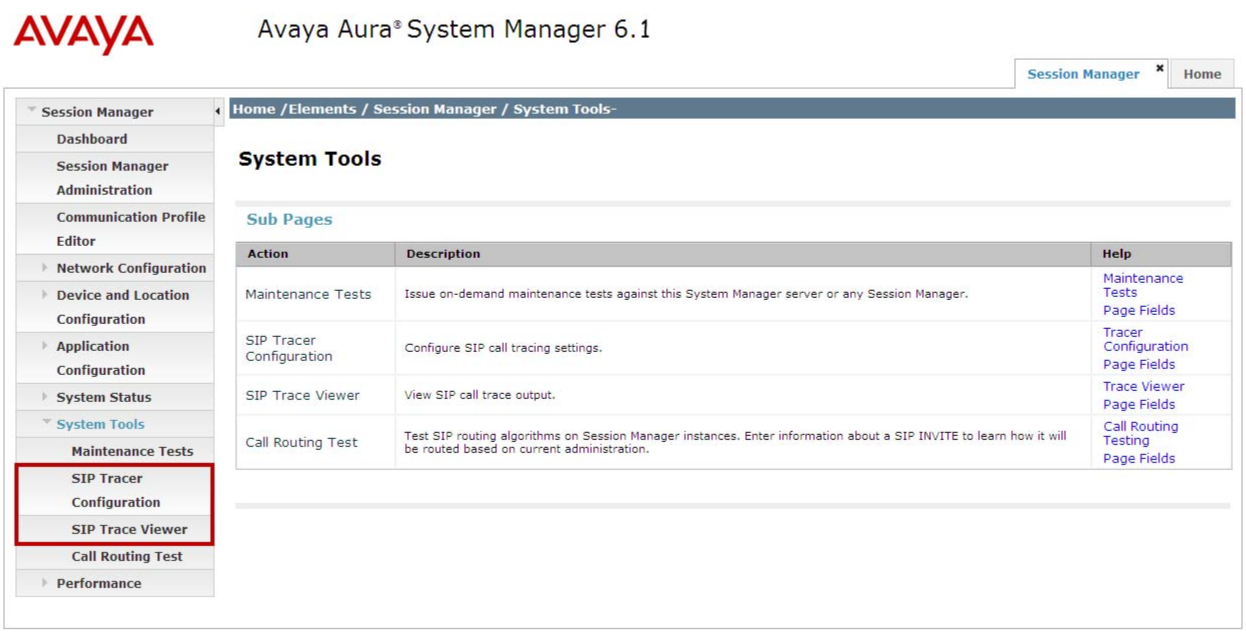
6.2 Verify Avaya Aura® Session Manager Configuration

- Log in to System Manager web interface. If **SIP Monitoring** is **enabled**, select **Session Manager** from the left pane. Select **SIP Monitoring** from the sub-menu. Drill down into the applicable entity to verify status.

  The following example shows the **Connection Status** and **Link Status** as **UP** for the Avaya Aura® Session Manager



- Use the SIP Message Trace Tool available on Avaya Aura® System Manager to view SIP messages. The trace tool is located under the Session Manager Element under System Tools.



For more information regarding how to configure and use the SIP Tracer, see **Reference [9].**

6.2.1 SIP Trace Excerpt

The following excerpt showing a SIP REFER message with non-shared UUI followed by the subsequent SIP INVITE:

Note: IP addresses have been partially hidden for security.

- **SIP REFER from Sending Communication Manager**

2011-07-11 10:45:00,443 CallLogs INFO - : Incoming Message
Transport: TCP : ip=135.9.88.XX***_SESSIONMANGERADDRESS***, port=28215,
REFER sip:1156@135.9.88.XX***_M3KADDRESS*** :5060;transport=tcp SIP/2.0
From: <sip:25220220@135.9.88. XX***_SESSIONMANGERADDRESS***;user=phone>;tag=0c496773db8e01f8a44e352ba400
To: "Station 4000" <sip:1156@135.9.88.XX***_M3KADDRESS*** >;tag=1c1466213091
Call-ID: 146621225711720111164455@135.9.88.XX***_M3KADDRESS***
CSeq: 1 REFER
P-Av-Transport: AP;fe=135.9.88.XX***_CLANADDRESS***:13381;ne=135.9.88.XX***_SESSIONMANGERADDRESS***:5060;tt=TCP;th
Max-Forwards: 69
Via: SIP/2.0/TCP 135.9.88.XX***_SESSIONMANGERADDRESS***;branch=z9hG4bK01ef9793db8e01f9a44e352ba400-AP;ft=42626
Via: SIP/2.0/TCP 135.9.88.XX***_CLANADDRESS***;branch=z9hG4bK01ef9793db8e01f9a44e352ba400
User-Agent: Avaya CM/R016x.00.1.510.1
Contact: "NCR LAI UUI" <sip:25220220@135.9.88.XX***_CLANADDRESS*** ;transport=tcp>
Route: <sip:135.9.88.XX***_SESSIONMANGERADDRESS***:15060;transport=tcp;lr;sap=-1610992167*1*016asm-callprocessing.sar1029179527~1310402695991~-751678143~1>
Route: <sip:5dee6dc7@135.9.88.XX***_SESSIONMANGERADDRESS***;transport=tcp;lr>
Refer-To: <sip:7200200@dr.avaya.com?**User-to-User=0435323030323232353230303232323532303032323235323030323232353230303232323532303032323235323030323232353230303232323532303 032323235323030323232353230303232323532303032323235323031323334353637383930313233343536%3Bencoding%3Dhex**>
Referred-By: "NCR LAI UUI" <sip:25220220@dr.avaya.com>
Content-Length: 0

- **SIP INVITE to Receiving Communication Manager**

2011-07-11 10:45:00,487 CallLogs INFO - : Outgoing Message
Transport: TCP : ip=135.9.88. XX_***SESSIONMANGERADDRESS***, port=5060,

INVITE sip:7200200@dr.avaya.com;routeinfo=0-0 SIP/2.0
Record-Route: <sip:135.9.88.XX_***SESSIONMANGERADDRESS***:15060;lr;sap=-1610992167*1*016asm-callprocessing.sar1029179527~1310402700482~-751678127~1;transport=tcp>
Record-Route: <sip:5dee6dc7@135.9.88.XX_***SESSIONMANGERADDRESS***;transport=tcp;lr>
Via: SIP/2.0/TCP 135.9.88.XX_***SESSIONMANGERADDRESS***:15070;branch=z9hG4bK8709583DFFFFFFFFB8119ADD0716608
Via: SIP/2.0/TCP 135.9.88.XX_***SESSIONMANGERADDRESS***:15070;branch=z9hG4bK8709583DFFFFFFFFB8119ADD1716606
Via: SIP/2.0/TCP 135.9.88.XX_***SESSIONMANGERADDRESS***:15070;branch=z9hG4bK8709583DFFFFFFFFB8119ADD1716605
Via: SIP/2.0/TCP 135.9.88.XX_***SESSIONMANGERADDRESS***;branch=z9hG4bKac1540234860-AP;ft=47632
Via: SIP/2.0/TCP 135.9.88.XX_***M3KADDRESS***;branch=z9hG4bKac1540234860;alias
From: "Station 4000" <sip:1156@135.9.88. XX_***M3KADDRESS*** >;tag=1c1540224117
To: <sip:7200200@dr.avaya.com>
Call-ID: 15402232611172011164450@135.9.88. XX_***M3KADDRESS***
CSeq: 1 INVITE
Contact: <sip:1156@135.9.88.XX_***M3KADDRESS***:5060;transport=tcp>
Supported: em,100rel,timer,replaces,path,early-session,resource-priority,sdp-anat
Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
Referred-By: "NCR LAI UUI" <sip:25220220@dr.avaya.com>
User-Agent: Audiocodes-Sip-Gateway-Mediant 3000/v.5.80A.060 AVAYA-SM-6.1.2.0.612009
User-to-User:
0435323030323232353230303232323532303032323235323030323232353230303232323532303032323235323030323232353230303232
32353230303232323532303032323235323030323232353230313233343536373839303132333435 36;encoding=hex
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 276
P-Asserted-Identity: "Station 4000" <sip:1156@dr.avaya.com>
Route: <sip:135.9.88.XX_***SESSIONMANGERADDRESS***;transport=tcp;lr>
Route: <sip:135.9.228.XX_***SESSIONMANGERADDRESS***;transport=tcp;lr;phase=terminating>
P-AV-Transport:
AP;fe=135.9.88.XX_***M3KADDRESS***:61643;ne=135.9.88.XX_***SESSIONMANGERADDRESS***:5060;tt=TCP;th;timerB=4
P-Location: SM;origlocname="135.9.88";termlocname="135.9.228"
Max-Forwards: 67

6.3 Verify Avaya Aura® Communication Manager Configuration

- To verify Call Flow, log in an agent at the receiving Communication Manager and place a call like a customer would through the solution. Select appropriate prompts. Verify call is delivered to the agent.
- To verify UUI data delivered with the call use the **uui-info** button programmed on the agent's phone.
- Use the CM SAT command **list trace tac x** where x is the TAC for the desired trunk group to be monitored. Verify the SIP routing to the CM SIP trunk connected to CM. **Note:** use the **status trunk** command on the CM SAT to see trunk member activity.
- Use the CM SAT command **list trace vec x** where x is the vector defined in **Section 3.9** and **Section 3.11** to trace for the call scenario. Verify applicable vector steps are processed.
- Use the CM SAT command **list trace vdn x** where x is the VDN defined in **Section 3.10** and **Section 3.12** to trace for the call scenario.
- Use the CM SAT command **display events** to display applicable failures logged by the Communication Manager.

7. Conclusion

As illustrated here in these Application Notes, User-to-User data can be passed successfully for Network Call Redirection using Avaya Aura® Communication Manager and the Look-Ahead Interflow feature over SIP trunks that are integrated with Avaya Aura® Session Manager and AudioCodes Mediant™ 3000 Gateway.

8. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Aura® Communication Manger 6.0.x:

[1] *Avaya Aura®™ Communication Manager Feature Description and Implementation*, DOC ID 555-245-205, available at http://support.avaya.com.

[2] *Administering Avaya Aura®Communication Manager*, DOC ID 03-300509, available at http://support.avaya.com.

[3] *Avaya Aura®™ Communication Manager Feature Description and Implementation*, DOC ID 555-245-205, available at http://support.avaya.com.

[4] *Avaya Toll Fraud Security Guide*, DOC ID 555-025-600, available at http://support.avaya.com.

Avaya Aura® Call Center 6.0.x

[5] *Administering Avaya Aura®™ Call Center Features*, Release 6.0 November 2010, available at http://support.avaya.com.

[6] *Programming Call Vectors in Avaya Aura™ Call Center*, Release 6.0 June 2010, available at http://support.avaya.com.

[7] *Avaya Aura®™ Call Center 6.0 Overview*, Release 6.0 April 2011, available at http://support.avaya.com.

[8] *Avaya Aura®™ Call Center Feature Reference*, Release 6.0 November 2010, available at http://support.avaya.com.

Avaya Aura® Session Manager 6.1.x

[9] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, DOC ID 03-603325, available at http://support.avaya.com.

[10] *Administering Avaya Aura® Session Manager*, DOC ID 03-603324, available at http://support.avaya.com.

[11] *Avaya Aura®™ Session Manager Overview*, DOC ID 100068105, available at http://support.avaya.com.

AudioCodes Mediant™ 3000

[12] *Mediant 3000 Media Gateway Setup*, March 2011, available at http://support.avaya.com.

9. Change History

This section is required only if the Application Notes are being re-issued. It should contain the following table listing previous issue numbers and the dates of issue.

Issue	Date	Reason
0.1	05/23/2011	Initial issue
0.2	07/07/2011	Updating document based on review/comments
0.3	07/20/2011	Updating document based on review/comments
0.4	08/11/2011	Updating document based on review/comments

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com