



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 (UII) with Network Call Redirection (NCR) utilizing SIP REFER messages via Look-Ahead Interflow (LAI) over SIP Trunks with UII 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.

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1. Introduction

These Application Notes focus on a sample configuration for passing non-shared User-to-User Information (UII) with Network Call Redirection (NCR) utilizing Look-Ahead Interflow (LAI) over SIP trunks via SIP REFER messages with the UII 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 UII is contained within the SIP REFER message. Non-shared UII transports only ASAI UII.

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

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 UII. 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 UII 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 UII data is present as applicable, displaying the ASAI UII data on the agent's phone display when the UII-Info button on the agent's phone is pressed, correct ASAI UII 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.

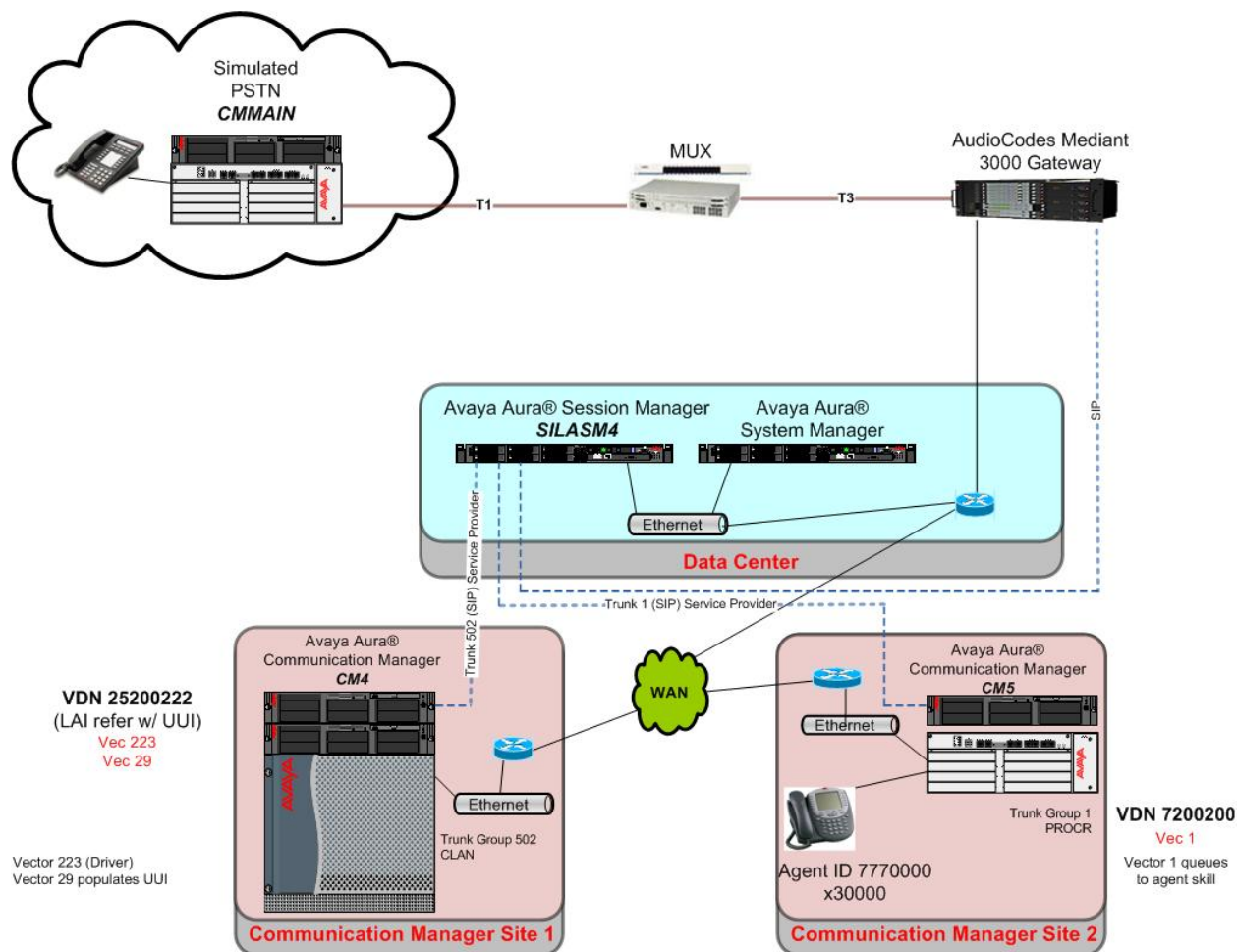


Figure 1

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 25220220 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
UII	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 <ul style="list-style-type: none">• 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	
VECTORIZING	
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 Codec	Silence Suppression	Frames Per Pkt	Packet 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		
H.323 IP ENDPOINTS	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 Link Bounce Recovery? y	RSVP Enabled? n	
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**
- **UI Treatment** **service provider**

add trunk-group 502	Page 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Numbering Format: public	
UI 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 UII 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
Basic? y	EAS? y
Prompting? y	LAI? 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 UII on Sending Communication Manager

Administer the vector to populate simulated ASAI UII 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 UII data. As vector steps 1 thru 5 build the first 80 bytes of ASAI UII from the VDN Variables, step 6 will add the remaining 16 bytes of ASAI UII 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 UII
Multimedia? n	Attendant Vectoring? n
Basic? y	EAS? y
Prompting? y	LAI? 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 UII Data on Sending Communication Manager

Administer the VDN for populating non-shared UII. 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 UII
      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 UII data. Based on the protocol up to 96 bytes of ASAI UII can be passed. The first 80 bytes of ASAI UII are created using the VDN Variables. The remaining bytes will be appended based on vector programming.

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

add vdn 5200222	VECTOR DIRECTORY NUMBER	Page 3 of 3																														
	VDN VARIABLES*																															
	<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="width: 10%;">Var</th> <th style="width: 40%;">Description</th> <th style="width: 50%;">Assignment</th> </tr> </thead> <tbody> <tr> <td>V1</td> <td>ASAI 1-16</td> <td>5200221520022152</td> </tr> <tr> <td>V2</td> <td>ASAI 17-32</td> <td>0022152002215200</td> </tr> <tr> <td>V3</td> <td>ASAI 33-48</td> <td>2215200221520022</td> </tr> <tr> <td>V4</td> <td>ASAI 49-64</td> <td>1520022152002215</td> </tr> <tr> <td>V5</td> <td>ASAI 65-80</td> <td>2002215200221520</td> </tr> <tr> <td>V6</td> <td></td> <td></td> </tr> <tr> <td>V7</td> <td></td> <td></td> </tr> <tr> <td>V8</td> <td></td> <td></td> </tr> <tr> <td>V9</td> <td></td> <td></td> </tr> </tbody> </table>	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			
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 NCRUII	
Multimedia? n	Attendant Vectoring? n	Meet-me Conf? n
Basic? y	EAS? y	G3V4 Enhanced? y
Prompting? y	LAI? y	G3V4 Adv Route? y
Variables? y	3.0 Enhanced? y	CINFO? y
		BSR? y
		Holidays? y
01 wait-time	2 secs hearing ringback	
02 queue-to	skill 1 pri 1	
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
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								
DIAL PLAN ANALYSIS TABLE								
Location: all								
Percent Full: 4								
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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 UI 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
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.

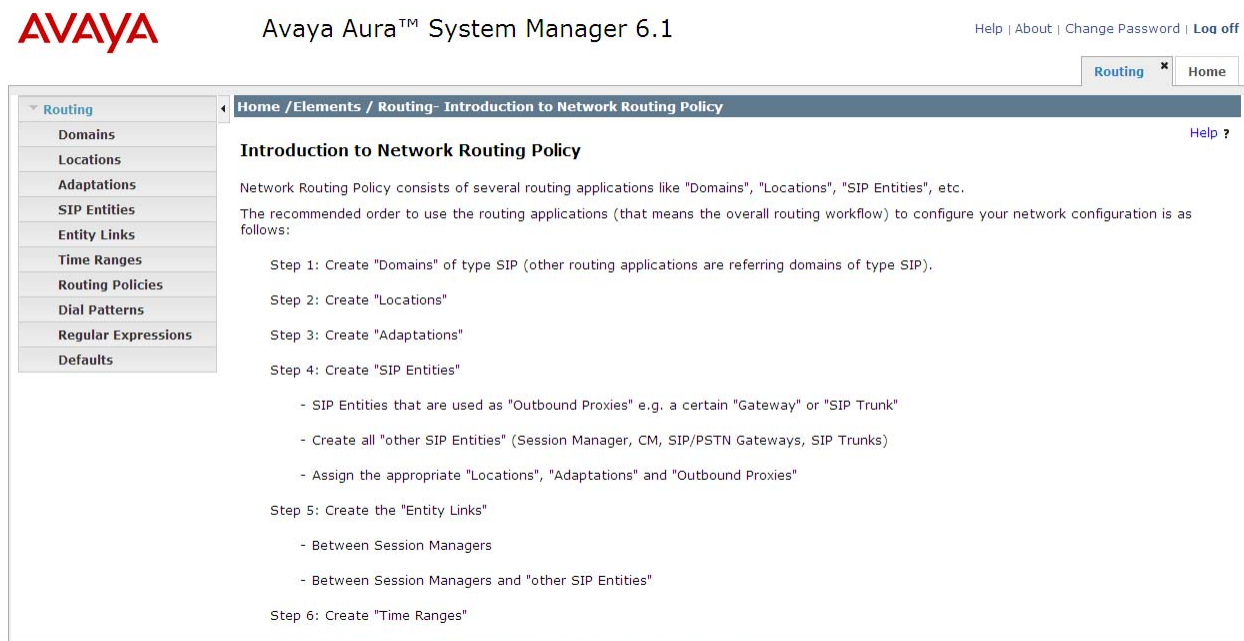
change feature-access-codes	Page 5 of 11
FEATURE ACCESS CODE (FAC)	
Call Center Features	
AGENT WORK MODES	
After Call Work Access Code: *75	
Assist Access Code: *76	
Auto-In Access Code: *71	
Aux Work Access Code: *72	
Login Access Code: *88	
Logout Access Code: *89	
Manual-in Access Code: *73	
SERVICE 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**.



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	Name	Type	Default	Notes
<input type="checkbox"/>	dr.avaya.com	sip	<input type="checkbox"/>	SIL Lab domain
<input type="checkbox"/>	mx.dr.avaya.com	sip	<input type="checkbox"/>	mx.dr.avaya.com
<input type="checkbox"/>	silasm4.dr.avaya.com	sip	<input type="checkbox"/>	
<input type="checkbox"/>	silfst.dr.avaya.com	sip	<input type="checkbox"/>	

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.



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Name	Type	Default	Notes
<input type="text" value="* dr.avaya.com"/>	<input type="text" value="sip"/>	<input type="checkbox"/>	<input type="text" value="SIL Lab domain"/>

* Input Required

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:

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar has a tree view with 'Locations' selected. The main content area is titled 'Home / Elements / Routing / Locations- Location Details'. In the 'General' section, the 'Name' field is set to '135.9.88'. The 'Location Pattern' section has an 'Add' button and a table with one entry: '135.9.88.*'. The 'Commit' button is highlighted in the top right corner.

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Routing | Home

Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Locations- Location Details

Location Details

General

* Name: 135.9.88
Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Mbit/sec
Total Bandwidth:
Multimedia Bandwidth:
Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec
Minimum Multimedia Bandwidth: 64 Kbit/Sec
* Default Audio Bandwidth: 80 Kbit/Sec

Location Pattern

Add Remove

1 Item Refresh Filter: Enable

IP Address Pattern	Notes
* 135.9.88.*	

Select : All, None

* Input Required

Commit Cancel

Add 135.9.228 Location:

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar has a tree view with 'Locations' selected. The main content area is titled 'Home / Elements / Routing / Locations- Location Details'. In the 'General' section, the 'Name' field is set to '135.9.228'. The 'Location Pattern' section has an 'Add' button and a table with one entry: '135.9.228.*'. The 'Commit' button is highlighted in the top right corner.

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Routing | Home

Routing
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Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

Home / Elements / Routing / Locations- Location Details

Location Details

General

* Name: 135.9.228
Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec
Total Bandwidth:
Multimedia Bandwidth:
Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 900 Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location): 900 Kbit/Sec
Minimum Multimedia Bandwidth: 64 Kbit/Sec
* Default Audio Bandwidth: 80 Kbit/Sec

Location Pattern

Add Remove

1 Item Refresh Filter: Enable

IP Address Pattern	Notes
* 135.9.228.*	

Select : All, None

* Input Required

Commit Cancel

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.



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Routing x Home

Home / Elements / Routing / SIP Entities- SIP Entity Details

SIP Entity Details

General

* Name: m3kTPglobal

* FQDN or IP Address: 135.9.88

Type: Gateway

Notes: M3K Global TP IP address

Adaptation: [v]

Location: 135.9.88 [v]

Time Zone: America/Denver [v]

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name: [v]

Call Detail Recording: none [v]

SIP Link Monitoring

SIP Link Monitoring: Link Monitoring Enabled [v]

* Proactive Monitoring Interval (in seconds): 900

* Reactive Monitoring Interval (in seconds): 120

* Number of Retries: 1

Commit Cancel

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.



Avaya Aura® System Manager 6.1

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

The screenshot displays the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off'. The main interface is divided into a left sidebar and a central content area. The sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted under the 'Routing' category. The central content area has a breadcrumb trail: 'Home / Elements / Routing / SIP Entities - SIP Entity Details'. The 'SIP Entity Details' section is titled 'General' and contains several configuration fields. A red box highlights the 'Name' field (cm4 CLAN 01a11), the 'FQDN or IP Address' field (135.9.88. [redacted]), the 'Type' dropdown (CM), and the 'Notes' field. Another red box highlights the 'Location' dropdown (135.9.88) and the 'Time Zone' dropdown (America/Denver). Below these, there are checkboxes for 'Override Port & Transport with DNS SRV' and 'SIP Timer B/F (in seconds): 4'. A 'Credential name' field and a 'Call Detail Recording' dropdown (none) are also visible. At the bottom, the 'SIP Link Monitoring' section shows a dropdown set to 'Use Session Manager Configuration'. In the top right corner of the main content area, there are 'Commit' and 'Cancel' buttons, with the 'Commit' button highlighted by a red box.

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.



Avaya Aura® System Manager 6.1

[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

The screenshot shows the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off'. The main interface has a sidebar on the left with a tree view containing 'Routing', 'Domains', 'Locations', 'Adaptations', 'SIP Entities' (highlighted), 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. The 'SIP Entity Details' section contains the following fields: 'Name' (silcm5-procr), 'FQDN or IP Address' (135.9.228. [redacted]), 'Type' (CM), 'Notes' (silcm5-procr for NCR/UUI Shared [redacted]), 'Adaptation' (dropdown), 'Location' (135.9.228 [redacted]), 'Time Zone' (America/Denver [redacted]), 'Override Port & Transport with DNS SRV' (checkbox), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (text field), 'Call Detail Recording' (none), and 'SIP Link Monitoring' (Use Session Manager Configuration). The 'Commit' button is highlighted with a red box.

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.

The screenshot displays the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 6.1", and links for "Help | About | Change Password | Log off". Below the navigation bar, the breadcrumb trail reads "Home / Elements / Routing / SIP Entities- SIP Entity Details". The left sidebar contains a menu with "Routing" selected, and sub-items including "Domains", "Locations", "Adaptations", "SIP Entities" (highlighted with a red box), "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The main content area is titled "SIP Entity Details" and shows the "General" tab. A red rectangular box highlights the form fields for creating a new SIP entity. The fields are: "Name" (text input with value "silas m#"), "FQDN or IP Address" (text input with value "135.9.88"), "Type" (dropdown menu with "Session Manager" selected), "Notes" (text input), "Location" (dropdown menu with "135.9.88" selected), "Outbound Proxy" (dropdown menu), "Time Zone" (dropdown menu with "America/Denver" selected), and "Credential name" (text input). To the right of the form are "Commit" and "Cancel" buttons, with a "Help ?" link above them. Below the form, the "SIP Link Monitoring" section shows a dropdown menu set to "Use Session Manager Configuration".

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.

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Routing * Home

Entity Links

Commit Cancel

1 Item Refresh Filter Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
silas m4-to-m3kT Pgl	silas m4	TCP	* 5060	m3kTPglobal	* 5060	<input checked="" type="checkbox"/>	

* Input Required

Commit Cancel

Enter a descriptive name

Select near-end entity

Select far-end entity

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.

AVAYA Avaya Aura™ System Manager 6.1

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Routing * Home

Entity Links

Commit Cancel

1 Item Refresh Filter Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
silasm4_cm4 CLAN	silasm4	TCP	* 5060	cm4 CLAN 01a11	* 5060	<input checked="" type="checkbox"/>	

* Input Required

Commit Cancel

Enter a descriptive name Select near-end entity Select far-end entity

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.

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Routing * Home

Entity Links

Commit Cancel

1 Item Refresh Filter Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
silasm4 to silcm5-procr	silasm4	TCP	* 5060	silcm5-procr	* 5060	<input checked="" type="checkbox"/>	Interface from silasm4

Enter a descriptive name Select near-end entity Select far-end entity

* Input Required

Commit Cancel

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.

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Routing x Home

Home / Elements / Routing / Time Ranges- Time Ranges

Time Ranges

Edit New Duplicate Delete More Actions

1 Item Refresh Filter Enable

<input type="checkbox"/>	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.

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Routing x Home

Home / Elements / Routing / Routing Policies- Routing Policy Details

Routing Policy Details

Commit Cancel

General

Name: to_m3kTPglobal

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
m3kTPglobal	135.9.89	Gateway	M3K Global TP IP address

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter Enable

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.

SIP Entities

29 Items Refresh		Filter Enable		
	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	cat-icrserver		Other	
<input type="radio"/>	cm4		CM	
<input type="radio"/>	cm4 CLAN 01e10		CM	
<input type="radio"/>	cm4 CLAN 01e11		CM	
<input type="radio"/>	cm4procr		CM	
<input type="radio"/>	CMFS1		CM	CMFS 6.0
<input type="radio"/>	CMFSTG		CM	s8800-CMFS-non-IMS
<input type="radio"/>	Conferencing AAC Server 6.0		Other	Conferencing 6.0 Standard
<input type="radio"/>	CS1K_Rel7_5		SIP Trunk	
<input type="radio"/>	GS60-OC3-TP8		Gateway	
<input type="radio"/>	GS60-OC3-TP9		Gateway	
<input type="radio"/>	IBMSUT		Other	
<input type="radio"/>	lras-MPPs		Voice Portal	
<input checked="" type="radio"/>	m3kTPglobal	135.9.88.	Gateway	M3K Global TP IP address
<input type="radio"/>	Presence-Element		Other	

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.



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Routing

Domains

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Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies- Routing Policy Details

Routing Policy Details

Commit

Cancel

Help ?

General

* Name:

to m3kTPglobal

Disabled:

☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
m3kTPglobal	135.9.88.89	Gateway	M3K Global TP IP address

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item Refresh

Filter Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.

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Routing Policy Details

General

Name: Disabled: ☐ Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
cm4 CLAN 01a11	135.9.88.1	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
1	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.

SIP Entity List

SIP Entities				
29 Items Refresh Filter Enable				
	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	cato-lordsserver		Other	
<input type="radio"/>	cm4		CM	
<input type="radio"/>	cm4 CLAN 01a10		CM	
<input checked="" type="radio"/>	cm4 CLAN 01a11	135.9.88.1	CM	
<input type="radio"/>	cm4procr		CM	
<input type="radio"/>	CMFS1		CM	CMFS 6.0
<input type="radio"/>	CMFSTG		CM	e8800-CMFS-non-IMS
<input type="radio"/>	Conferencing AAC Server 6.0		Other	Conferencing 6.0 Standard
<input type="radio"/>	CS1K_Rail7_5		SIP Trunk	
<input type="radio"/>	G860-OC3-TP8		Gateway	
<input type="radio"/>	G860-OC3-TP9		Gateway	
<input type="radio"/>	IBMSUT		Other	
<input type="radio"/>	inas-MPPs		Voice Portal	
<input type="radio"/>	m3kTPglobal		Gateway	M3K Global TP IP address
<input type="radio"/>	Presence-Element		Other	8

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.



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Routing Policy Details

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
cm4 CLAN 01a11	135.9.98. [REDACTED]	CM	

Time of Day

1 Item Refresh Filter Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
1	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.



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Routing Policy Details

General

* Name: to_silcm5-procr

Disabled: ☐

Notes: NCR/UII Shared and SP trunks

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
silcm5-procr	135.9.228.100	CM	silcm5-procr for NCR/UII Shared and Service Provider Trunks

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
1	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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.

SIP Entity List

Select Cancel

SIP Entities

29 Items Refresh Filter: Enable

Name	FQDN or IP Address	Type	Notes
s8800_cmes		CM	CMES 6.0
silasm3		Session Manager	Mixed Enterprise SM
silasm4		Session Manager	SM 6.0 Sprint 35
silasm5		Session Manager	
silasm6		Session Manager	
silasm1-cm		CM	LSP
silasm1-sip		Session Manager	silasm1-sip
SILCM2		CM	
silcm5		CM	
silcm5_MixedEnterprise		CM	Mixed Enterprise CM
silcm5-procr	135.9.228.100	CM	silcm5-procr for NCR/UII Shared and Service Provider Trunks
SIL LAB CM		CM	
sil-sbc		Gateway	
StackMM		Modular Messaging	

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.



[Routing](#) [Home](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies- Routing Policy Details

Routing Policy Details

Commit

Cancel

Help ?

General

* Name:

to: silcm5-procr

Disabled:

☐

Notes:

NCR/UI Shared and SP trunks

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
silcm5-procr	135.9.228. [REDACTED]	CM	silcm5-procr for NCR/UI Shared and Service Provider Trunks

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item

Refresh

Filter Enable

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

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

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura™ System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off". Below the navigation bar, there are tabs for "Routing" and "Home". The left sidebar contains a tree view with the following items: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, **Dial Patterns** (highlighted with a red box), Regular Expressions, and Defaults. The main content area is titled "Home / Elements / Routing / Dial Patterns- Dial Pattern Details". It features a "Dial Pattern Details" section with a "General" tab. The form fields are as follows: "Pattern" (text box with "2522"), "Min" (text box with "8"), "Max" (text box with "8"), "Emergency Call" (checkbox), "SIP Domain" (dropdown menu with "dr.avaya.com"), and "Notes" (text box with "for NCR/UUI Config (Shared)"). There are "Commit" and "Cancel" buttons at the top right of the form area.

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.



Home / Elements / Routing / Dial Patterns- Originating Location and Routing Policy List

Originating Location and Routing Policy List Select Cancel

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

6 Items Refresh Filter Enable

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>	135.9.228	
<input type="checkbox"/>	135.9.52	
<input checked="" type="checkbox"/>	135.9.88	
<input type="checkbox"/>	20.20.20	Data VLAN
<input type="checkbox"/>	IBM SUT	IBM Sametime Unified Telephony
<input type="checkbox"/>	Lec SILICR	

Select : All, None

Routing Policies

18 Items Refresh Filter Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	CMPS1-IMS	<input type="checkbox"/>	CMPS1	(s8800_cmfr1_G450)
<input type="checkbox"/>	CMPS1-nen-IMS	<input type="checkbox"/>	CMPS1Q	CMPS1-nen-IMS
<input type="checkbox"/>	Conference Server	<input type="checkbox"/>	Conferencing AAC Server 6.0	to Conferencing Server
<input type="checkbox"/>	CS1000 Rel 7.5	<input type="checkbox"/>	CS1K_Rel7_5	
<input type="checkbox"/>	stackmm	<input type="checkbox"/>	StackMM	Mod Max
<input type="checkbox"/>	to_CM4	<input type="checkbox"/>	cm4	
<input type="checkbox"/>	to cm4 CLAN 01x10	<input type="checkbox"/>	cm4 CLAN 01x10	CLAN 1x10 Service Provider
<input checked="" type="checkbox"/>	to cm4 CLAN 01x11	<input type="checkbox"/>	cm4 CLAN 01x11	CLAN 1x11 Shared
<input type="checkbox"/>	To_CM4procr	<input type="checkbox"/>	cm4procr	IBM dev connect testing
<input type="checkbox"/>	to CMES	<input type="checkbox"/>	s8800_cmes	Route to CMES aka cm4
<input type="checkbox"/>	To_IBMSUT	<input type="checkbox"/>	IBMSUT	
<input type="checkbox"/>	to-irsa-MPPs	<input type="checkbox"/>	irsa-MPPs	
<input type="checkbox"/>	to m3kTPglobal	<input type="checkbox"/>	m3kTPglobal	silasm3 to m3kTPglobal
<input type="checkbox"/>	to silcm2	<input type="checkbox"/>	SILCM2	
<input type="checkbox"/>	to silcm5	<input type="checkbox"/>	silcm5	

Select : All, None < Previous Page 1 of 2 Next >

Select Cancel

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

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off". Below the navigation bar, there is a breadcrumb trail: "Home / Elements / Routing / Dial Patterns- Dial Pattern Details". The left sidebar contains a tree view with the following items: "Routing", "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns" (highlighted with a red box), "Regular Expressions", and "Defaults". The main content area is titled "Dial Pattern Details" and has a "General" tab selected. The form fields are as follows: "Pattern" (text box with value "72002", highlighted with a red box), "Min" (text box with value "7", highlighted with a red box), "Max" (text box with value "7", highlighted with a red box), "Emergency Call" (checkbox, unchecked), "SIP Domain" (dropdown menu with value "-ALL-", highlighted with a red box), and "Notes" (text box with value "silcm5-procr NCR/UII", highlighted with a red box). At the bottom of the form, there is a section titled "Originating Locations and Routing Policies" with "Add" and "Remove" buttons. The "Add" button is highlighted with a red box. There are also "Commit" and "Cancel" buttons at the top right of the form area.

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.



Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns- Originating Location and Routing Policy List

Originating Location and Routing Policy List

SelectCancel

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

6 Items RefreshFilter: Enable

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>	135.9.228	
<input type="checkbox"/>	135.9.52	
<input checked="" type="checkbox"/>	135.9.88	
<input type="checkbox"/>	20.20.20	Data VLAN
<input type="checkbox"/>	IBM SUT	IBM Sametime Unified Telephony
<input type="checkbox"/>	Loc SILICR	

Select : All, None

Routing Policies

18 Items RefreshFilter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	to silcm5-procr	<input type="checkbox"/>	silcm5-procr	NCR/UII Shared and SP trunks
<input type="checkbox"/>	TO SIL LAB CM	<input type="checkbox"/>	SIL LAB CM	
<input type="checkbox"/>	to-sil-sbc	<input type="checkbox"/>	sil-sbc	

Select : All, None

< PreviousPage 2 of 2Next >

SelectCancel

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

5. Configure the AudioCodes Mediant™ 3000 Gateway

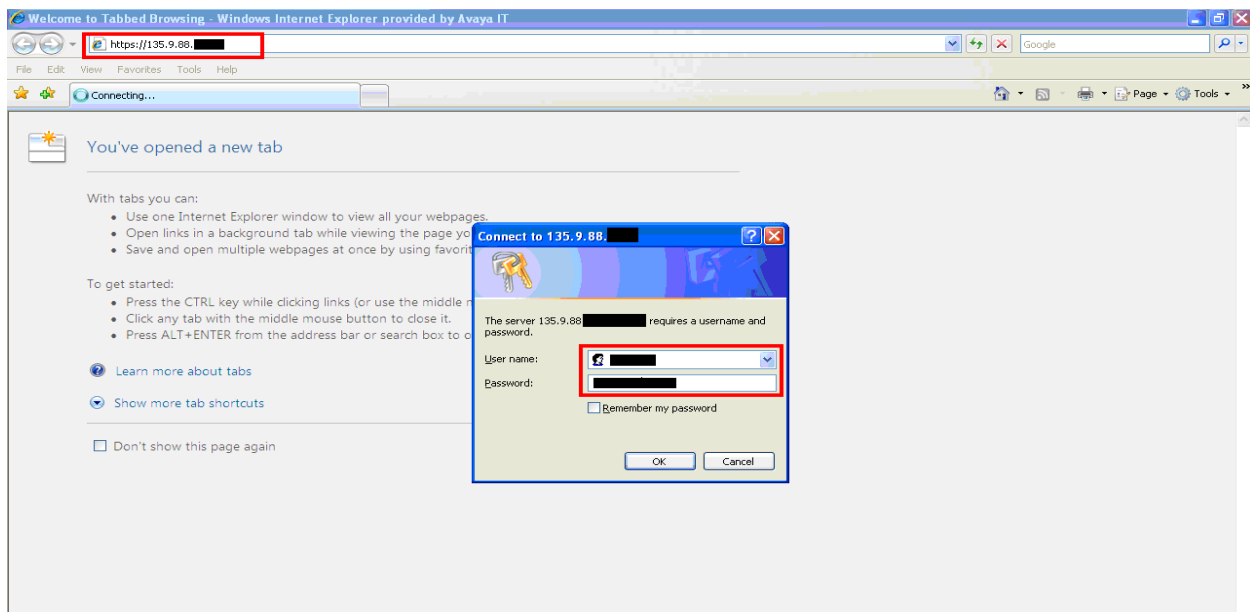
This section provides the procedures for configuring M3K using the web based graphical user interface. The procedures include the following areas:

- 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.

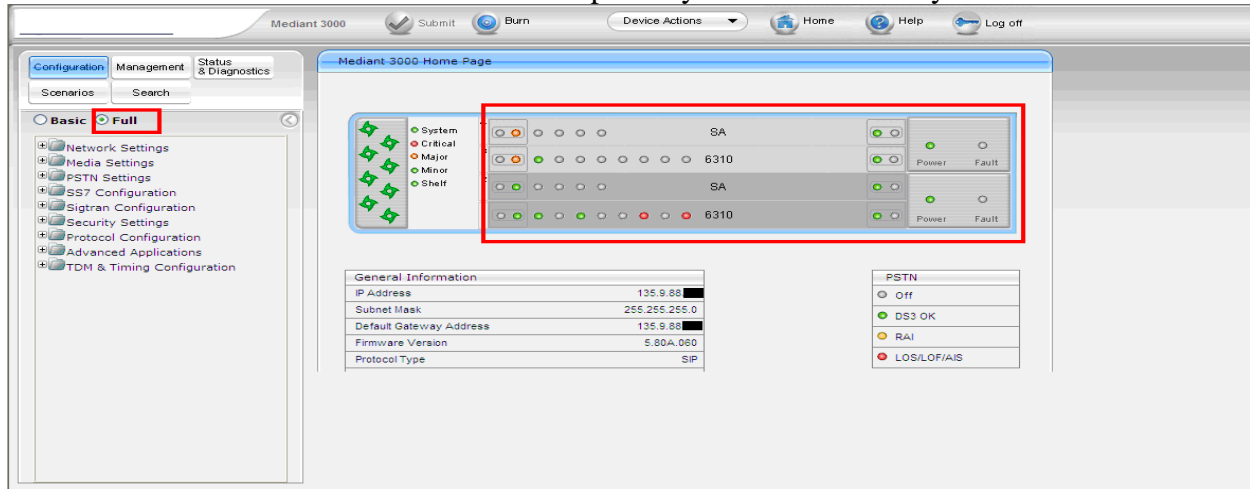
Note: IP addresses and usernames have been partially hidden for security.



5.1 Display all configurable parameters

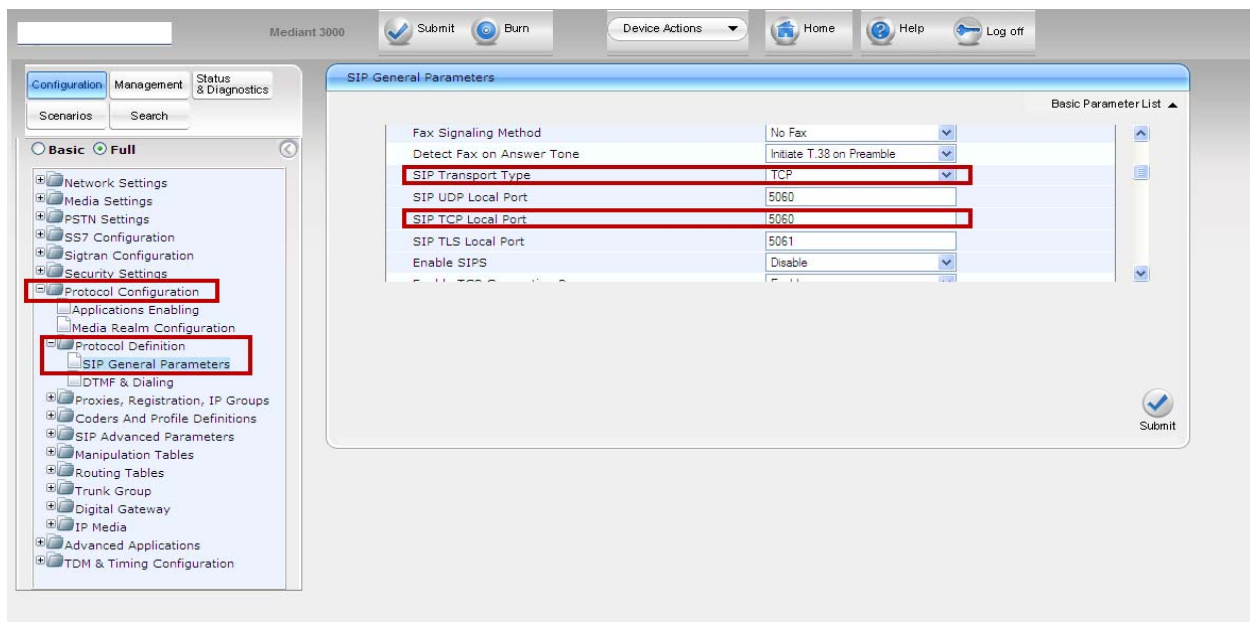
The Mediant 3000 Home Page will be displayed. To display all configurable parameters, select **Full** in the left pane. In the screenshot below, both TP6310 and SA boards are shown.

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.

The screenshot shows the Mediant 3000 configuration interface. On the left is a navigation tree with categories like Basic, Full, Network Settings, Media Settings, PSTN Settings, SS7 Configuration, Sigtran Configuration, Security Settings, Protocol Configuration, Applications Enabling, Media Realm Configuration, Protocol Definition, SIP General Parameters, DTMF & Dialing, Proxies, Registration, IP Groups, Coders And Profile Definitions, SIP Advanced Parameters, Manipulation Tables, Routing Tables, Trunk Group, Digital Gateway, IP Media, Advanced Applications, and TDM & Timing Configuration. The 'SIP General Parameters' page is active, displaying a table of parameters. Two rows are highlighted with red boxes: 'SIP Destination Port' set to 5060 and 'Use user=phone in SIP URL' set to Yes. Other parameters include SIP UDP Local Port (5060), SIP TCP Local Port (5060), SIP TLS Local Port (5061), Enable SIPS (Disable), Enable TCP Connection Reuse (Enable), TCP Timeout (0), Use user=phone in From Header (No), Use Tel URI for Asserted Identity (Disable), Tel to IP No Answer Timeout (180), and Enable Remote Party ID (Disable). A 'Submit' button is at the bottom right.

SIP General Parameters	
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use user=phone in SIP URL	Yes
Use user=phone in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180
Enable Remote Party ID	Disable

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

This screenshot shows the same configuration interface as the previous one, but with different settings. The 'Burn' button in the top toolbar is highlighted with a red box. The 'SIP General Parameters' page is still active, but the parameters are different. A 'Submit' button at the bottom right is also highlighted with a red box. The parameters listed are: NAT IP Address (0.0.0.0), PRACK Mode (Supported), Channel Select Mode (Cyclic Ascending), Enable Early Media (Enable), 183 Message Behavior (Progress), Session-Expires Time (0), Minimum Session-Expires (90), Session Expires Method (Re-INVITE), Asserted Identity Mode (Disabled), Fax Signaling Method (No Fax), and Detect Fax on Answer Tone (Initiate T.38 on Preamble).

SIP General Parameters	
NAT IP Address	0.0.0.0
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	No Fax
Detect Fax on Answer Tone	Initiate T.38 on Preamble

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 in **Section 3.3**.

The screenshot displays the Mediant 3000 configuration interface. On the left, the 'Configuration' pane is expanded to show 'Protocol Configuration' and 'Coders And Profile Definitions', with 'Coders' selected. The right pane shows the 'Coders Table' with the following data:

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled
G.729	20	8	18	Disabled

A 'Submit' button is located at the bottom right of the table area.

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.

Mediant 3000 Submit Burn Device Actions Home Help Log off

Configuration Management Status & Diagnostics
Scenarios Search

Basic Full

Network Settings
Media Settings
PSTN Settings
SS7 Configuration
Sigtran Configuration
Security Settings
Protocol Configuration
Applications Enabling
Media Realm Configuration
Protocol Definition
Proxies, Registration, IP Groups
Codex And Profile Definitions
SIP Advanced Parameters
Manipulation Tables
Routing Tables
Routing General Parameters
Tel to IP Routing
IP to Trunk Group Routing
Internal DNS Table
Internal SRV Table
Release Cause Mapping
Alternative Routing

Tel to IP Routing

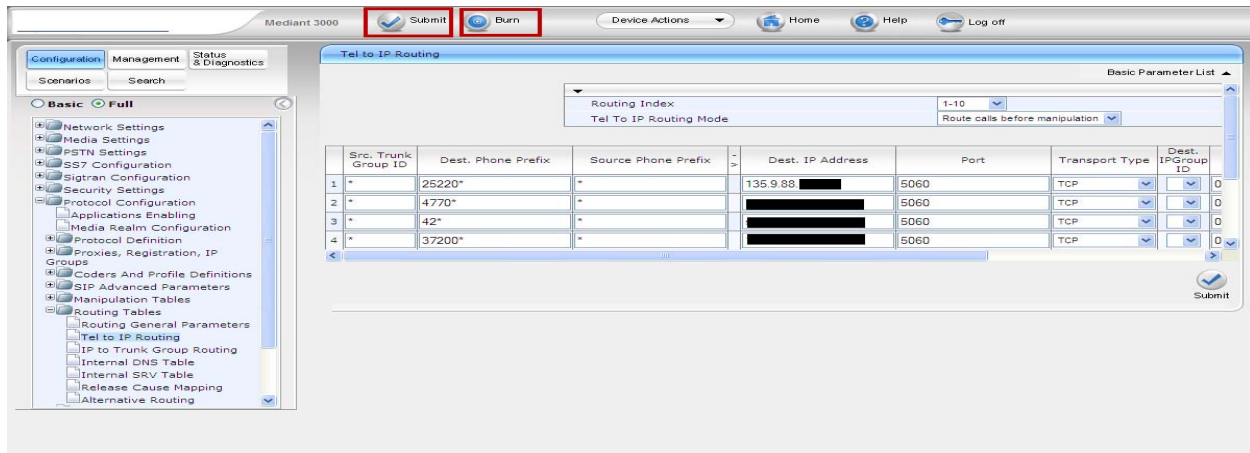
Basic Parameter List

Routing Index 1-10
Tel To IP Routing Mode Route calls before manipulation

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IP Group ID
1	*	25220*	*	135.9.88. [REDACTED]	5060	TCP	0
2	*	4770*	*	[REDACTED]	5060	TCP	0
3	*	42*	*	[REDACTED]	5060	TCP	0
4	*	37200*	*	[REDACTED]	5060	TCP	0

Submit

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

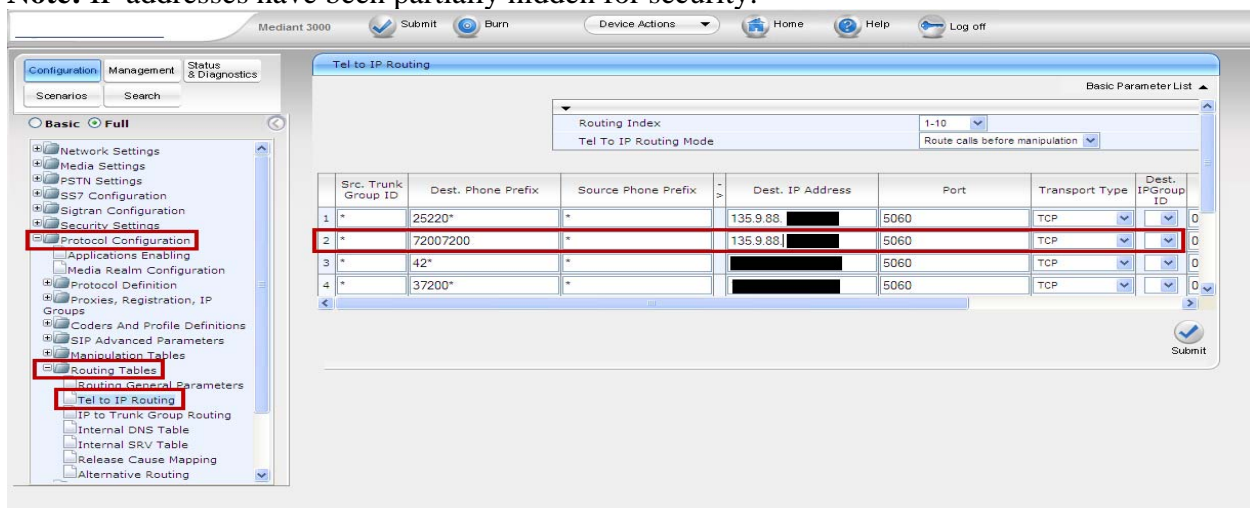


5.5 Setup Tel to IP Routing to Route Calls between Session Manager and Receiving Communication Manager

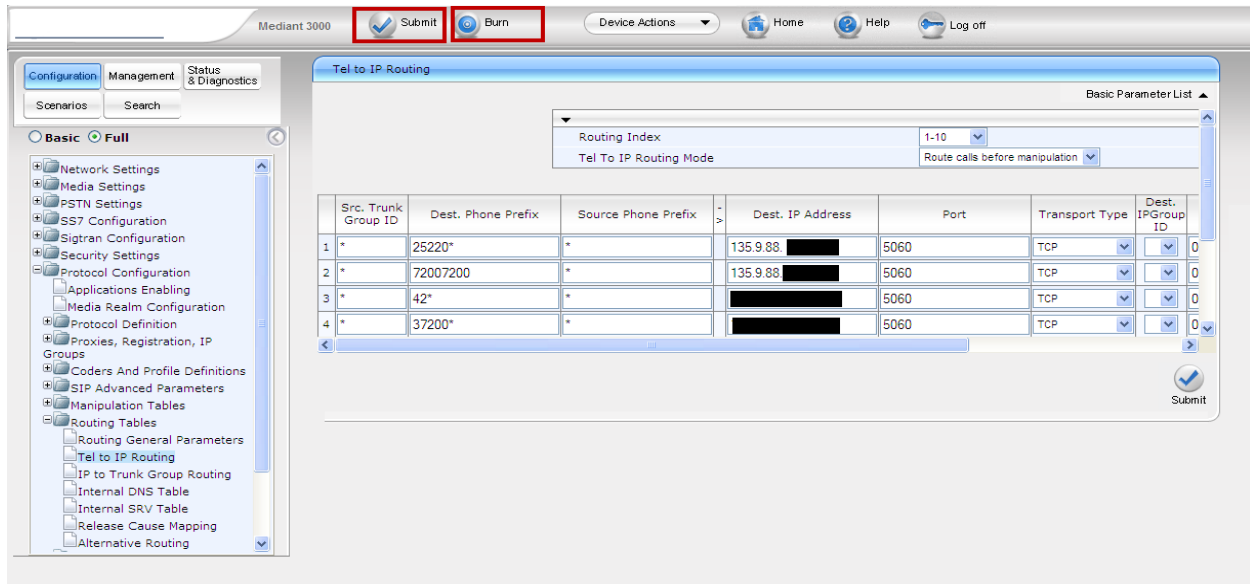
Expand **Protocol Configuration** and **Routing Tables** in the left pane. Select **Tel to IP Routing**. The **Tel to IP Routing** table is displayed in the right pane. Select the desired row of the table and enter routing information:

- **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.



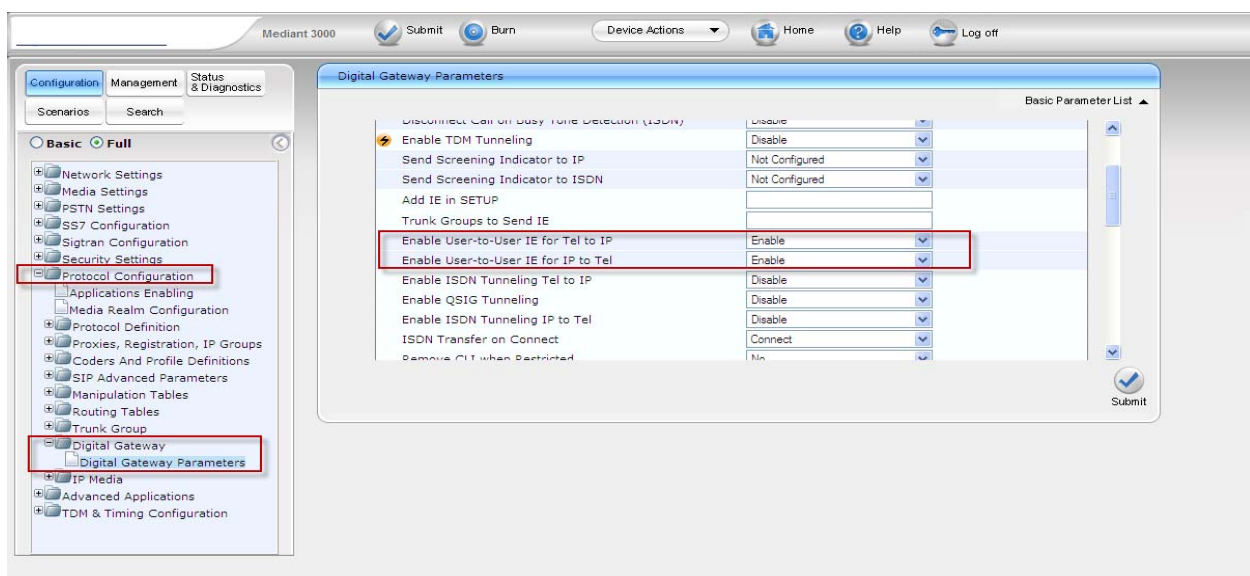
Click **Submit** and then **Burn**.



5.6 Verify the Protocol Configuration

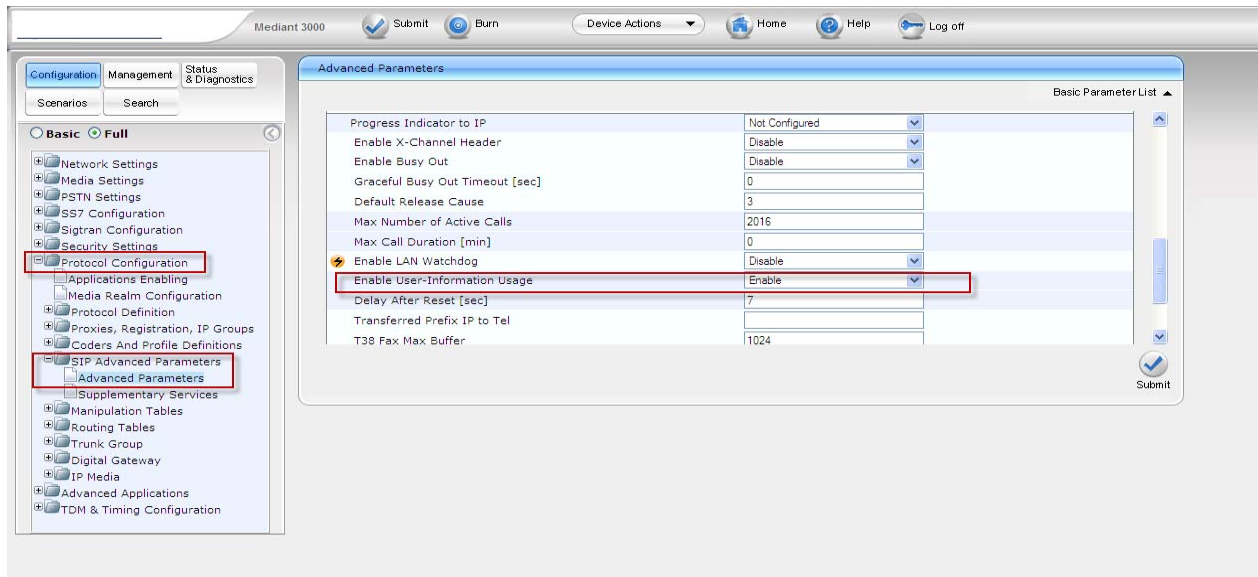
5.6.1 Verify the Digital Gateway Parameters

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

AVAYA Avaya Aura® System Manager 6.1

Session Manager x Home

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

SIP Entity, Entity Link Connection Status [Help ?](#)

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: silasm4

Summary View

1 Item Refresh

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	silasm3	135.9.88	5060	TCP	Up	200 OK	Up

Filter: Enable

Session Manager

- Dashboard
- Session Manager
- Administration
- Communication Profile Editor
- Network Configuration
- Device and Location Configuration
- Application Configuration
- System Status
 - SIP Entity Monitoring
 - Managed Bandwidth Usage
 - Security Module Status
 - Registration Summary
 - User Registrations

- 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.



Avaya Aura® System Manager 6.1

Session Manager x
Home

- ▼ Session Manager
 - Dashboard
 - Session Manager Administration
 - Communication Profile Editor
 - ▶ Network Configuration
 - ▶ Device and Location Configuration
 - ▶ Application Configuration
 - ▶ System Status
 - ▼ System Tools
 - Maintenance Tests
 - SIP Tracer Configuration**
 - SIP Trace Viewer**
 - Call Routing Test
 - ▶ Performance

Home / Elements / Session Manager / System Tools-

System Tools

Sub Pages

Action	Description	Help
Maintenance Tests	Issue on-demand maintenance tests against this System Manager server or any Session Manager.	Maintenance Tests Page Fields
SIP Tracer Configuration	Configure SIP call tracing settings.	Tracer Configuration Page Fields
SIP Trace Viewer	View SIP call trace output.	Trace Viewer Page Fields
Call Routing Test	Test SIP routing algorithms on Session Manager instances. Enter information about a SIP INVITE to learn how it will be routed based on current administration.	Call Routing Testing Page Fields

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**

[illegible]

- **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=z9hG4bK8709583DFFFFFFFFFB8119ADD0716608

Via: SIP/2.0/TCP 135.9.88.XX_SESSIONMANGERADDRESS:15070;branch=z9hG4bK8709583DFFFFFFFFFB8119ADD1716606

Via: SIP/2.0/TCP 135.9.88.XX_SESSIONMANGERADDRESS:15070;branch=z9hG4bK8709583DFFFFFFFFFB8119ADD1716605

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@>

To: <sip:7200200@dr.avaya.com>

Call-ID: 1540223
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 UII" <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:

```
04353230303232323532303032323235323030323232353230303232323532303032323235323030323232353230303232  
32353230303232323532303032323235323031323334353637383930313233343536;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 Manager 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

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