

SAM SIP Integration with Avaya Session Manager

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Overview

This document outlines the configuration steps to integrate the Smart Assist by Mutare(SAM) using Session Initiation Protocol (SIP) with the Avaya Aura Communication Manager (CM) and Avaya Session Manager (ASM)

Site Configuration

Avaya Aura Communication Manager must be at release 5.1 or higher.

For this document, the configuration was as follows:

- Avaya Communication Manager 7.0 (CM) virtualized.
- Avaya Media Server (AMS) Virtualized
- Avaya Session Manager 7.0.1 (ASM) virtualized
- Avaya System Manager 7.0.1 (SMGR) virtualized

For the purposes of the configuration examples below, the following IP configuration was used:

- Mutare SAM- 192.168.1.79
- Avaya CM – 192.168.1.206
- Avaya Session Manager- 192.168.1.208
- Avaya Session Manager Security Module- 192.168.1.215

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Configure Avaya Aura Communication Manager

Communication Manager License

Use the **display system-parameters customer-options** command to verify that the Communication Manager license has sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

```
display system-parameters customer-options                               Page 2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
      Maximum Administered H.323 Trunks: 12      11
    Maximum Concurrently Registered IP Stations: 450 16
      Maximum Administered Remote Office Trunks: 450 0
Maximum Concurrently Registered Remote Office Stations: 450 0
      Maximum Concurrently Registered IP eCons: 0 0
    Max Concur Registered Unauthenticated H.323 Stations: 0 0
      Maximum Video Capable H.323 Stations: 0 0
      Maximum Video Capable IP Softphones: 0 0
      Maximum Administered SIP Trunks: 450 37
Maximum Administered Ad-hoc Video Conferencing Ports: 0 0
    Maximum Number of DS1 Boards with Echo Cancellation: 80 0
      Maximum TN2501 VAL Boards: 0 0
      Maximum Media Gateway VAL Sources: 50 1
      Maximum TN2602 Boards with 80 VoIP Channels: 0 0
      Maximum TN2602 Boards with 320 VoIP Channels: 0 0
    Maximum Number of Expanded Meet-me Conference Ports: 0 0
```

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IP Interfaces

Use the list ip-interface all command to identify which IP interfaces are located in which network region.

```
list ip-interface all
```

IP INTERFACES

ON	Type	Slot	Code/Sfx	Node Name/ IP-Address	Mask	Gateway Node	Net Rgn	VLAN
---	-----	-----	-----	-----	-----	-----	---	----
y	PROCR			192.168.1.206	/24	192.168.1.1	1	

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IP Network Region

The configuration of the IP network regions is assumed to be already in place and is included here for clarity. Use **display ip-network-region** command to view these settings. Important fields:

- The **Authoritative Domain** field is configured to match the domain name configured on the Avaya SES. This name appears in the "From" header of SIP messages originating from this IP region.
- **IP-IP Direct Audio** (media shuffling) was enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway.
- The **Codec Set** field was set to the IP codec set to be used for calls within this IP network region.

```
display ip-network-region 1                                     Page 1 of 19
                                                              IP NETWORK REGION
Region: 1
Location: 1           Authoritative Domain: mutaresip.com
Name: main
MEDIA PARAMETERS           Intra-region IP-IP Direct Audio: yes
                          Codec Set: 1           Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048        IP Audio Hairpinning? n
UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS   RTCP Reporting Enabled? y
Call Control PHB Value: 34 RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46       Use Default Server Parameters? y
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
Audio 802.1p Priority: 6
Video 802.1p Priority: 5   AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS        RSVP Enabled? n
H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```



Customer Initials:

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Codecs

Use the **change ip-codec-set** to verify that the codec is configured to G.711MU.

```
display ip-codec-set 1 Page 1 of 2
```

IP Codec Set

Codec Set: 1

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

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Signaling Group

The signaling group and the associated SIP trunk group are used for routing calls to/from the CM to the ASM. Important fields:

- **Group Type:** sip.
- **Transport Method:** tcp (Transport Layer Security).
- **Near-end Node Name:** This will be **procr**
- **Far-end Node Name:** Node name of the ASM, in this case, **ASM**.
- **Near-end Listen Port:** This will default to **5060**
- **Far-end Listen Port:** Change to **5060**.
- **Far-end Network Region:** This should be set to the network region which contains the ASM.
- **DTMF over IP:** Set to the default value of **rtp-payload**, which allows the CM to send DTMF using RFC 2833.
- **Direct IP-IP Audio Connections:** Set to **n** to disable media shuffling on the trunk level

SIGNALING GROUP

```
Group Number: 1                Group Type: sip
IMS Enabled? n                 Transport Method: tcp
Q-SIP? n
IP Video? n                    Enforce SIPS URI for SRTP? y
Peer Detection Enabled? n     Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr      Far-end Node Name: ASM
Near-end Listen Port: 5060     Far-end Listen Port: 5060
Far-end Network Region: 1

Far-end Domain:
Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload     Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3 IP Audio Hairpinning? n
Enable Layer 3 Test? y        Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n Alternate Route Timer(sec): 6
```

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Trunk Group

The trunk group should be configured as follows. Important fields:

- **Group Type:** sip
- **Group Name:** Use a descriptive name.
- **Direction:** two-way
- **Service Type:** public-ntwrk
- **Signaling Group:** Use the signaling group configured in the previous step.
- **Number of Members:** Enter the number of trunks desired for the application.

```
TRUNK GROUP

Group Number: 1                Group Type: sip                CDR Reports: y
Group Name: SMGR                COR: 1                TN: 1                TAC: 100
Direction: two-way            Outgoing Display? n
Dial Access? n                Night Service:
Queue Length: 0
Service Type: public-ntwrk    Auth Code? n
                                Member Assignment Method: auto
                                Signaling Group: 1
                                Number of Members: 20
```

- **Numbering Format:** public. This field specifies the format of the calling party number sent to the far-end.

```
display trunk-group 4                Page 3 of 21
TRUNK FEATURES
    ACA Assignment? n                Measured: none
                                Maintenance Tests? y

    Numbering Format: public
                                UUI Treatment: service-provider
                                Replace Restricted Numbers? n
                                Replace Unavailable Numbers? n

Show ANSWERED BY on Display? y
```

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- **Send Transferring Party Information:** y
- **Send Diversion Header:** y

PROTOCOL VARIATIONS

Mark Users as Phone? n
Prepend '+' to Calling Number? n
Send Transferring Party Information? y

Send Diversion Header? y
Support Request History? y
Telephone Event Payload Type:

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Create a Hunt Group

Add a hunt group using the following command: **add hunt-group X (desired group number)**

Hunt group should be configured as following: Important fields:

- **-Group Number:** Use number according to dial plan
- **-Group Name:** Use distinctive group name
- **-Group Extension:** Use desired extension to match dial plan
- **-Group Type:** ucd-mia
- **-Message Center:** sip-adjunct
- **-Voice Mail Number:** Use Hunt Group number
- **-Voice Mail Handle:** This will represent the SIP Header (ex. 5999@mutare.com) Use a unique identifier
- **-Routing Digits:** Use AAR feature access code

```
display hunt-group 20                                     Page 1 of
60
                                     HUNT GROUP

      Group Number: 1                                     ACD? n
      Group Name: SAM - SIP                               Queue? n
Group Extension: 7000                                   Vector? n
      Group Type: ucd-mia                                Coverage Path:
      TN: 1                                              Night Service Destination:
      COR: 81                                           MM Early Answer? n
      Security Code:                                    Local Agent Preference? n
ISDN/SIP Caller Display: mbr-name
```

```
change hunt-group 20                                     Page 2 of
60
                                     HUNT GROUP

      Message Center: sip-adjunct
Voice Mail Number      Voice Mail Handle      Routing Digits
(e.g., AAR/ARS Access Code)
5999                   5999                                           *9
```

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Create Coverage Path:

Create a coverage path using the command: **add coverage path X** (desired number)

Coverage path should be configured as follows: Important fields:

- **Coverage Criteria: Active:**
 - Inside Call-n Outside Call-n
 - Busy: Inside Call-y Outside call-y
 - Don't Answer: Inside Call-y Outside Call-y
 - All: Inside Call-n Outside Call-n
 - DND/SAC/Goto cover: Inside Call-y Outside Call-y
 - Holiday Coverage: Inside Call-n Outside Call-n
- **Number of Rings: 3**
- **Coverage Point 1:** h(hunt group number created above)

```
                                COVERAGE PATH

                                Coverage Path Number: 1
                                Cvg Enabled for VDN Route-To Party? n          Hunt after Coverage? n
                                Next Path Number:                            Linkage

COVERAGE CRITERIA

                                Station/Group Status   Inside Call   Outside Call
                                Active?                n             n
                                Busy?                  y             y
                                Don't Answer?          y             y          Number of Rings: 3
                                All?                  n             n
                                DND/SAC/Goto Cover?    y             y
                                Holiday Coverage?      n             n

COVERAGE POINTS

                                Terminate to Coverage Pts. with Bridged Appearances? y
                                Point1: h1             Rng:          Point2:
                                Point3:                 Point4:
                                Point5:                 Point6:
```

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Add Route-Pattern

Add a route-pattern using the following command: **add route-pattern X (match dial plan)**

Route-Pattern should be configured as follows: Important fields

- **Route Pattern Name:** Use unique identifier to match dialplan
- **Group Number:** Select the Signalling group number (this should match the SIP trunk)
- **FRL:** Choose an FRL that is equal to or lower than your station COR

```

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

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

```

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Create entry in AAR (OR ARS, depending on your standard configuration):

Create an entry in the AAR using the following command: change aar analysis X (this should be the start of your hunt group number)

Dialed string should be configured as follows: Important Fields:

- **Dialed String:** Add hunt group number
- **Total:** Total length min and max of your hunt group (per dial plan)
- **Route Pattern:** Enter the newly created route-pattern
- **Call Type:** AAR

AAR DIGIT ANALYSIS TABLE						
Location: all				Percent Full: 2		
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
5	7	7	254	aar	n	n
5999	4	4	2	aar	n	n
52000	5	5	18	aar	n	n
54000	5	5	104	aar	n	n
55000	5	5	9	unku	n	n
56000	5	5	50	aar	n	n
57000	5	5	103	aar	n	n
58000	5	5	101	aar	n	n
59000	5	5	199	aar	n	n
6	7	7	254	aar	n	n
7	7	7	254	aar	n	n
8	7	7	254	aar	n	n
84749	5	5	13	aar	n	n
9	7	7	254	aar	n	n

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Assign Coverage Path to Station

Change all applicable stations to cover to newly created coverage path using the following command:
change station X (Station number)

Stations should be configure as follows: Important Fields:

- **Coverage Path 1:** Assign newly created coverage path

```
STATION
Extension: 69057                Lock Messages? n                BCC: 0
Type: 9640                      Security Code: *                TN: 1
Port: S00048                    Coverage Path 1: 1              COR: 6
Name: Brown, John                Coverage Path 2:                COS: 2
                                Hunt-to Station:

STATION OPTIONS
                                Time of Day Lock Table:
Loss Group: 19                    Personalized Ringing Pattern: 1
                                Message Lamp Ext: 69057
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
                                Customizable Labels?
```

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Change System Features:

Check the following fields using the command: **change system-parameters features**

Configure features as follows: Important fields:

- **Trunk-to-Trunk Transfer: all**

change system-parameters features

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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

Abbreviated Dial Programming by Assigned Lists? n

Auto Abbreviated/Delayed Transition Interval (rings): 2

Protocol for Caller ID Analog Terminals: Bellcore

Display Calling Number for Room to Room Caller ID Calls? n

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Avaya System Manager Configuration

Log into SMGR.

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:
Password:

[Change Password](#)

Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

Under "Elements" click on 'Routing'

Home

Users

- Administrators
- Directory Synchronization
- Groups & Roles
- User Management
- User Provisioning Rule

Elements

- Avaya Breeze™
- Communication Manager
- Communication Server 1000
- Conferencing
- Device Services
- IP Office
- Media Server
- Meeting Exchange
- Messaging
- Presence
- Routing
- Session Manager
- Work Assignment

Services

- Backup and Restore
- Bulk Import and Export
- Configurations
- Events
- Geographic Redundancy
- Inventory
- Licenses
- Replication
- Reports
- Scheduler
- Security
- Shutdown
- Solution Deployment Manager
- Templates
- Tenant Management

Last Logged on at: September 29, 2016 3:27 AM

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Select 'Adaptations'

Home / Elements / Routing / Adaptations

Adaptations

2 Items Filter: Enable

<input type="checkbox"/>	Name	Module Name	Module Parameters	Egress URI Parameters	Notes
<input type="checkbox"/>	SAM	DigitConversionAdapter	fromto=true odstd=olaf.mutare.com osrcd=breezesman.mutare.com reduceRthdrs=true iosrcd=breezecm.mutare.com		
<input type="checkbox"/>	Standard	DigitConversionAdapter			

Select : All, None

Add a new adaptation by clicking 'New'

Home / Elements / Routing / Adaptations

Adaptation Details

Commit Cancel

General

* Adaptation Name: SAM

* Module Name: DigitConversionAdapter

Module Parameter Type: Name-Value Parameter

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	fromto	true
<input type="checkbox"/>	iosrcd	breezecm.mutare.com
<input type="checkbox"/>	odstd	olaf.mutare.com

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
--------------------------	------------------	-----	-----	---------------	---------------	---------------	-------------------	-----------------	-------

Digit Conversion for Outgoing Calls from SM

Add Remove

2 Items Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*5003	*4	*4		*0		both		
<input type="checkbox"/>	*5999	*4	*4		*0		both		

Select : All, None

Commit Cancel

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Adaptation should be configured as follows. Important Fields:

- Adaptation Name: Set this to your desired name, in this case 'SAM'
- Module Name: Selcet DigitConversionAdapter from the drop down menu
- Module Parameter Type: Select Name-Value Parameter from the drop down menu
Add the following parameters:
 1. Name: fromto Value: true
 2. Name:iosrcd Value: the FQDN of your CM (Not a requirement but recommended)
 3. Name:odstd Value: The FQDN or IP address of the SAM server (ex. Olaf.mutare.com)

Click 'Commit' to save changes

Add a SIP Entity

Select "SIP Entity" from the left side menu and Click 'New'

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', the Mutare logo, and a 'Log off admin' button. The left sidebar contains a menu with 'Routing' selected, and sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entities' and shows a table with 5 items. The table has columns for Name, FQDN or IP Address, Type, and Notes. Below the table, there is a 'Select : All, None' option.

Name	FQDN or IP Address	Type	Notes
Breeze EDP	192.168.1.214	Avaya Breeze	
Communication Manager	192.168.1.206	CM	
Flowroute	192.168.1.201	SIP Trunk	
SAM	192.168.1.79	SIP Trunk	
Session Manager	192.168.1.215	Session Manager	

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SIP Entity should be configured as follows. Important fields:

SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Securable:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

Loop Count Threshold:

Loop Detection Interval (in msec):

- Name: Input name for SAM entity (ex. SAM)
- FQDN or IP Address: Enter IP address of SAM server (ex. 192.168.1.79)
- Adaptation: Select SAM adaptation created above
- Entity Link: add a new entity link as depicted below (Note the transport method is UDP)

Entity Links

Override Port & Transport with DNS SRV:

Add		Remove						Filter: Enable	
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	
<input checked="" type="checkbox"/>	* SAM	Session Manager	UDP	* 5060	SAM	* 5060	trusted	<input type="checkbox"/>	

Select : All, None

Commit the changes to save.

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Create a Routing Policy

Click on Routing Policies on the left hand menu. Click 'new' to create a new routing policy.

[Help ?](#)

Routing Policy Details

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Select			
Name	FQDN or IP Address	Type	Notes
SAM	192.168.1.79	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps											
1 Item Filter: Enable											
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

Dial Patterns

Add Remove						
1 Item Filter: Enable						
Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/> 5	4	4	<input type="checkbox"/>	-ALL-	Rolling Meadows	

Select : All, None

Regular Expressions

Add Remove			
1 Item Filter: Enable			
Pattern	Rank Order	Deny	Notes
<input type="checkbox"/> 5	0	<input type="checkbox"/>	

Select : All, None

Routing Policy should be configured as follows. Relevent Fields:

- Name: add descriptive name for a ASM to SAM policy
- Sip Entity as Destination: Click 'new' and add the SAM server as the destination
- Dial Pattern: Click add and create a dial pattern that matches the inbound number (ex. 5999)

Click commit to save changes

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Create New Dial Pattern

Select Dial Patterns from the left side menu.

Click 'New' to add new dial pattern.

Dial Pattern Details

General

* Pattern:

* Min:

* Max:

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

<input type="checkbox"/>	Originating Location Name ^	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Rolling Meadows		SM to CM	0	<input type="checkbox"/>	Communication Manager	
<input type="checkbox"/>	Rolling Meadows		SM to SAM	0	<input type="checkbox"/>	SAM	
<input type="checkbox"/>	Rolling Meadows		CM to SM	0	<input type="checkbox"/>	Session Manager	

Select : All, None

Dial patterns should be configured as followed. Relevant fields:

- Pattern: Enter a matching pattern of the SAM routing number (ex. 5999 or 5)
- Min: Select minimum expected digits
- Max: Select maximum expected digits
- Sip Domain: Select all
- Originating locations and routing policy: Click 'add' and add all desired locations and routing policies that were created above for SAM.

Click commit to save changes.