

SAM SIP Integration with Avaya Communication 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).

Site Configuration

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

For this document, the configuration was as follows:

- Avaya S8300 running Communication Manager (CM) in a G450 Gateway.
- On the Avaya G450 Gateway, the signaling and media resources needed to support SIP and H.323 trunks are integrated directly on the media gateway processor.

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

- Mutare SAM- 192.168.1.79
- Avaya CM – 192.168.1.71

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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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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                               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

      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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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			192.168.1.71	/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
```

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IP Node Names

Use the **change node-names ip** command to create a node name that maps to the SAM IP address.

```
change node-names ip                                     Page 1 of 2
                                                    IP NODE NAMES
Name                IP Address
FreeSwitch          54.200.128.127
HVM                 63.255.49.7
HVM1                68.70.168.8
HVM2                63.255.49.14
SMGR                192.168.1.223
aam                 192.168.1.227
cmm                 192.168.1.73
default             0.0.0.0
ivr                 192.168.1.81
marth               192.168.1.188
msgserver           192.168.1.38
msgserver2          192.168.1.36
SAM                 192.168.1.79
procr               192.168.1.71
sip-proxy           192.168.1.71
zipdx               71.116.126.81
( 16 of 16  administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

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IP Network Map

Use the **change ip-network map** command to ensure that the proper Network Region is assigned to the SAM IP address.

```
change ip-network-map                                     Page 1 of 63
```

IP ADDRESS MAPPING				
IP Address	Subnet Bits	Network Region	VLAN	Emergency Location Ext
FROM: 10.10.1.0	/24	5	n	
TO: 10.10.1.255				
FROM: 172.16.10.0	/24	5	n	
TO: 172.16.10.255				
FROM: 192.168.1.79	/	1	n	
TO: 192.168.1.79				

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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 Mutare MCS. Important fields:

- **Group Type:** sip.
- **Transport Method:** tcp (Transport Layer Security)
- **Co-Resident SES:** Set to **y** for this system. May not apply in your configuration.
- **Near-end Node Name:** This will be either **procr** if co-resident SES or use the node name that maps to the IP address of the CLAN circuit pack used to connect to the MCS.
- **Far-end Node Name:** Node name of the SAM, in this case, **SAM**.
- **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 MCS.
- **Far-end Domain:** This is set to the IP address assigned to the SAM. This domain is sent in the headers of the SIP INVITE messages for calls originating from and terminating to the SAM using this signaling group.
- **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 **y** to enable media shuffling on the trunk level.

```
display signaling-group 20
                                SIGNALING GROUP
Group Number: 20                Group Type: sip
                                Transport Method: tcp
IMS Enabled? n                 Co-Resident SES? y
Near-end Node Name: procr       Far-end Node Name: SAM
Near-end Listen Port: 5060      Far-end Listen Port: 5060
                                Far-end Network Region: 1
Far-end Domain: 192.168.1.79
Incoming Dialog Loopbacks: eliminate
                                Bypass If IP Threshold Exceeded? n
                                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? n
                                Direct IP-IP Early 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.

```
display trunk-group 20                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 20                Group Type: sip                CDR Reports: y
  Group Name: SAM                COR: 81                TN: 1                TAC: 187
  Direction: two-way            Outgoing Display? y
Dial Access? n                Night Service:
Queue Length: 0
Service Type: public-ntwrk        Auth Code? n
                                     Signaling Group: 20
                                     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

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

      Group Number: 20                                     ACD? n
      Group Name: SAM - SIP                               Queue? n
      Group Extension: 51020                             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
```

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- **-Message Center:** sip-adjunct
- **-Voice Mail Number:** Use Hunt Group number
- **-Voice Mail Handle:** This will represent the SIP Header (ex. Mutare@192.178.1.71) Use a unique identifier
- **Routing Digits:** Use AAR feature access code

change hunt-group 20

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HUNT GROUP

Message Center: sip-adjunct

Voice Mail Number

Voice Mail Handle

Routing Digits

(e.g., AAR/ARS Access Code)

51020

Mutare

*99

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Create Coverage Path:

Create a coverage path using the command: **change 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: 20
                                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: h20           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: 120 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: 20  0                                     n  user
2:                                     n  user
3:                                     n  user
4:                                     n  user
5:                                     n  user
6:                                     n  user

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

```

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Create entry in AAR:

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		Route Pattern	Call Type	Node Num	ANI Req'd
	Min	Max				
5	7	7	254	aar		n
51020	5	5	120	aar		n
52000	5	5	18	aar		n
54000	5	5	104	aar		n
55000	5	5	9	unku		n
56000	5	5	50	aar		n
57000	5	5	103	aar		n
58000	5	5	101	aar		n
59000	5	5	199	aar		n
6	7	7	254	aar		n
7	7	7	254	aar		n
8	7	7	254	aar		n
84749	5	5	13	aar		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: 20	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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Configure MCS

Mutare will configure the MCS for interfacing to the CM. The following section is for informational purposes only.

VBVConfig

```
[VoIP]
AcceptReinvite=1
SipDefaultTransportProtocol=TCP
SipTCPEntered=1
RetrieveSipHeader=Diversion      <--- Important to get the Diversion Header!

[Intel]
SIPDTMFMode=default
SIPSignalingPort=5060

[VoiceCard]
UseAlaw=0
Type=DialogicHMP
GetRawCallInfo=1 <--- Important to get the Diversion Header!

[Intel]
SIPCallControl=Dialogic

[Data]
NumofDATASessions=0
```

Outbound ANI

DID@<IP Address of MCS>

Outbound Calls

S<phone number>@<IP Address of CM>