Mutare Voice™ SIP Integration with Avaya Communication Manager

Overview

This document outlines the configuration steps to integrate Mutare Voice 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 is 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 is used:

- Mutare Voice- 192.168.1.79
- Avaya CM – 192.168.1.71

Configure Avaya Aura Communication Manager

Communication Manager License

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

Customer Initials:

```
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
```

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

Customer Initials:

```
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
```

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

                                                                    Net
ON Type   Slot  Code/Sfx       Node Name/       Mask  Gateway Node     Rgn  VLAN
                               IP-Address
-- ------ ----- --------       ---------------  ----  ---------------  ---  ----
 y PROCR                       192.168.1.71     /24   192.168.1.1      1
```

Customer Initials:

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) is 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 is 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:

IP Node Names

Use the change node-names ip command to create a node name that maps to the Mutare Voice 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
MutareVoice         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
```

Customer Initials:

IP Network Map

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

```
change ip-network-map                                          Page   1 of  63
                              IP ADDRESS MAPPING


                                              Subnet Network     Emergency
 IP Address                                   Bits   Region VLAN 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
```

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:
```

Customer Initials:

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 Mutare Voice, in this case, Mutare Voice.
- **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 Mutare Voice. This domain is sent in the headers of the SIP INVITE messages for calls originating from and terminating to the Mutare Voice 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 n to disable 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: MutareVoice
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
Far-end Domain: 192.168.1.79



                                             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? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

Customer Initials:

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: MutareVoice                 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
```

Customer Initials:

- **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:
```

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: MutareVoice - 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
```

Customer Initials:

- **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                                                    Page   2 of
60
                                  HUNT GROUP






                      Message Center: sip-adjunct
     Voice Mail Number         Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     51020                     Mutare                   *99
```

Customer Initials:

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:
```

Customer Initials:

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: MutareVoice
                             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
```

Customer Initials:

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            Total     Route     Call   Node   ANI
      String          Min   Max   Pattern   Type   Num    Reqd
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
```

Customer Initials:

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 configured 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?
```

Customer Initials:

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
SipTCPEnabled=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>

Customer Initials: