Mutare Voice™ SIP Integration with Avaya Session Manager

This document outlines the requirements for an on premise giSTT Enterprise server for speech to text conversions.

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 MVoice- 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

Customer Initials:

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
```

Customer Initials:

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                       192.168.1.206     /24   192.168.1.1      1
```

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:

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 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
```

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.

```
                                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
```

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:
```

Customer Initials:

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: MVoice - 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
```

Customer Initials:

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:
```

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: 2   Pattern Name: MVoice
                             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                                                             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 (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           Total      Route     Call    Node   ANI
           String          Min   Max  Pattern    Type    Num    Reqd
     5                      7     7     254      aar            n
     5999                   4     4     2        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 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?
```

Customer Initials:

Change System Features:

Check the following fields using the command: **change system-parameters features**
Configure features as follows: Important fields:

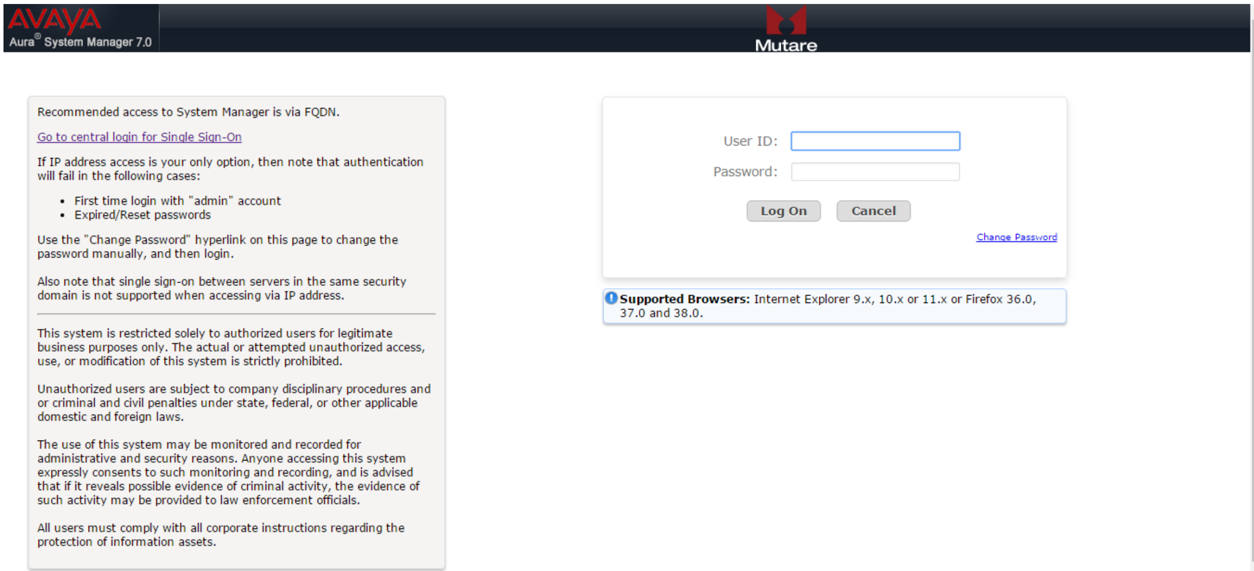
- **Trunk-to-Trunk Transfer**: all

```
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
```

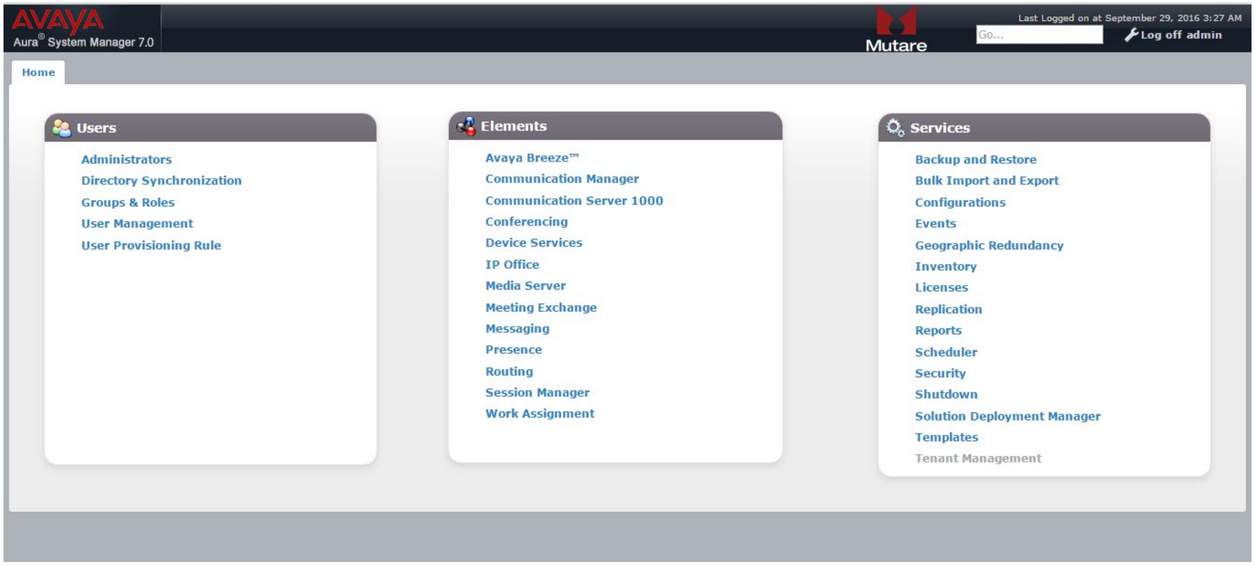
Customer Initials:

Avaya System Manager Configuration

Log into SMGR.

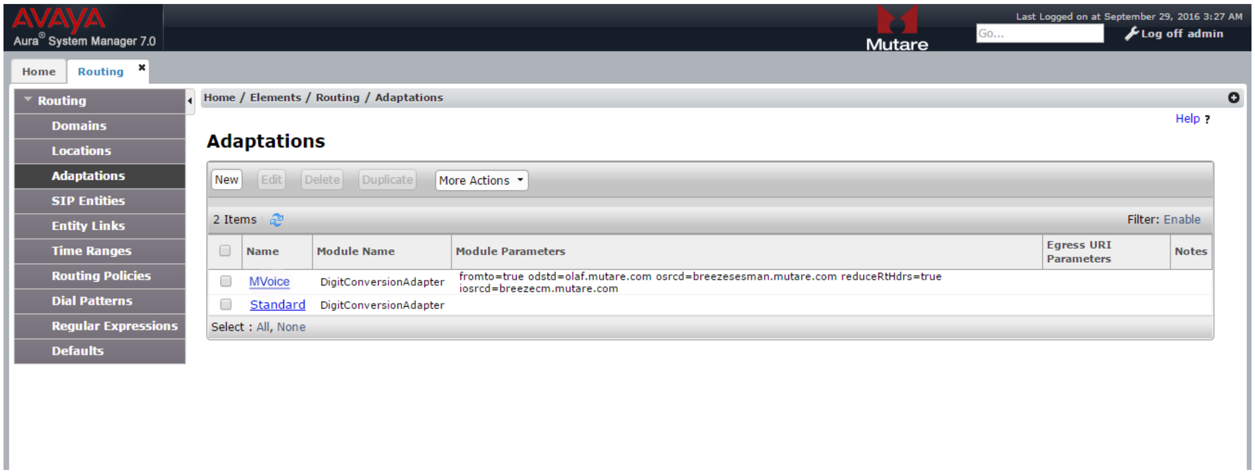


Under "Elements' click on 'Routing'

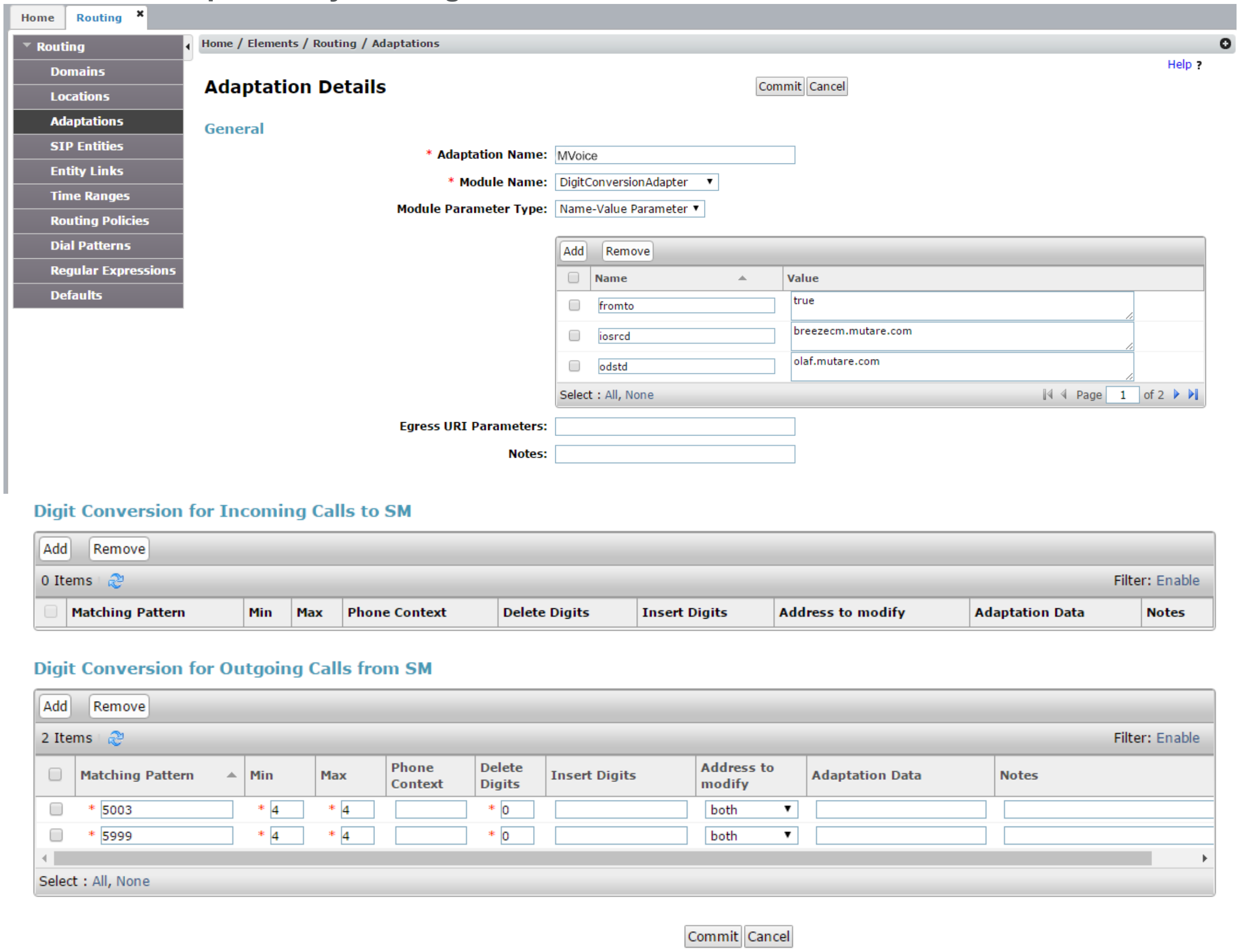


Customer Initials:

Select ‘Adaptations’



Add a new adaptation by clicking ‘New”



Adaptation should be configured as follows. Important Fields:

- **Adaptation Name:** Set this to your desired name, in this case ‘MVoice
- **Module Name:** Selcet DigitConversionAdapter from the drop down menu
- **Module Parameter Type:** Select Name-Value Parameter from the drop down menu

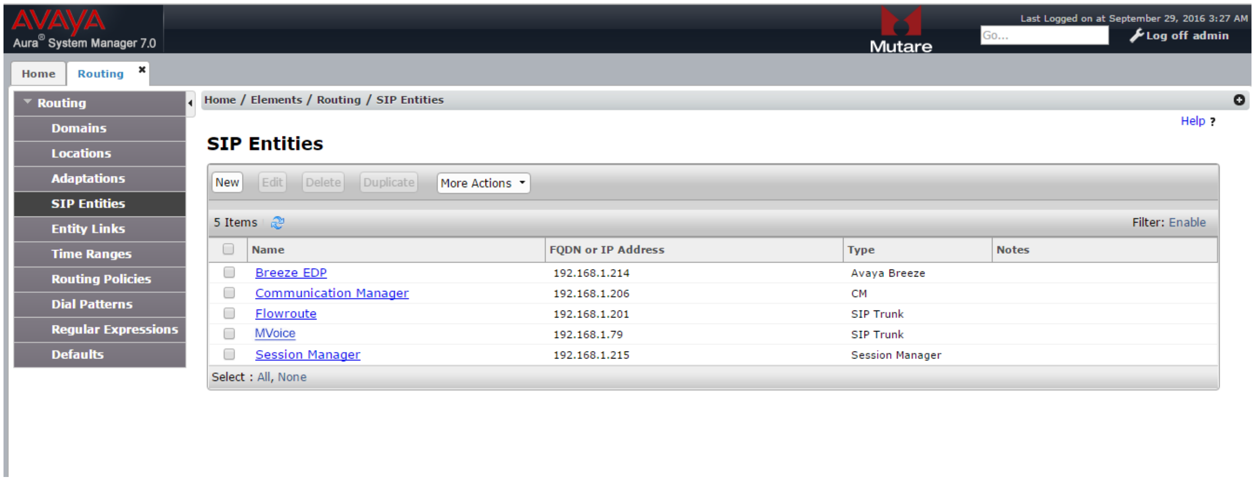
  Add the following parameters:

  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 MVoice server (ex. Olaf.mutare.com)

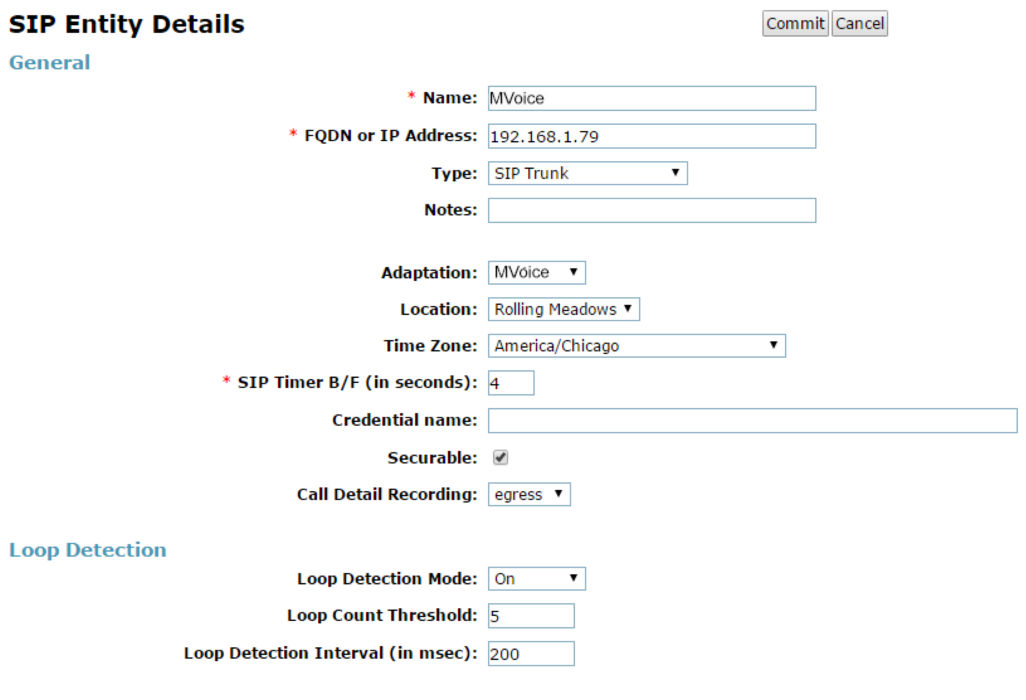
  Click ‘Commit’ to save changes

Add a SIP Entity

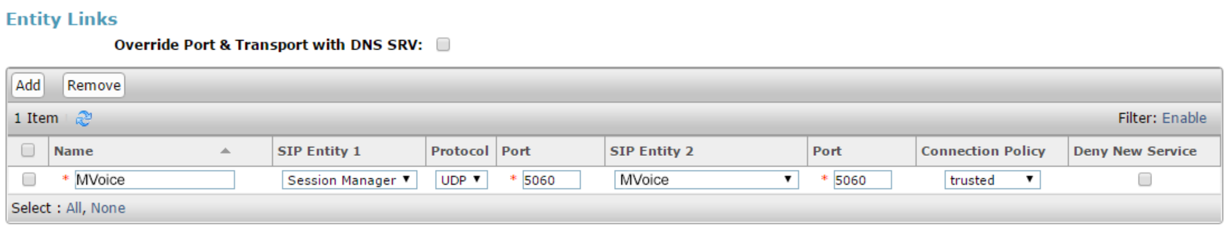
Select “SIP Entity” from the left side menu and Click ‘New”



SIP Entity should be configured as follows. Important fields:



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

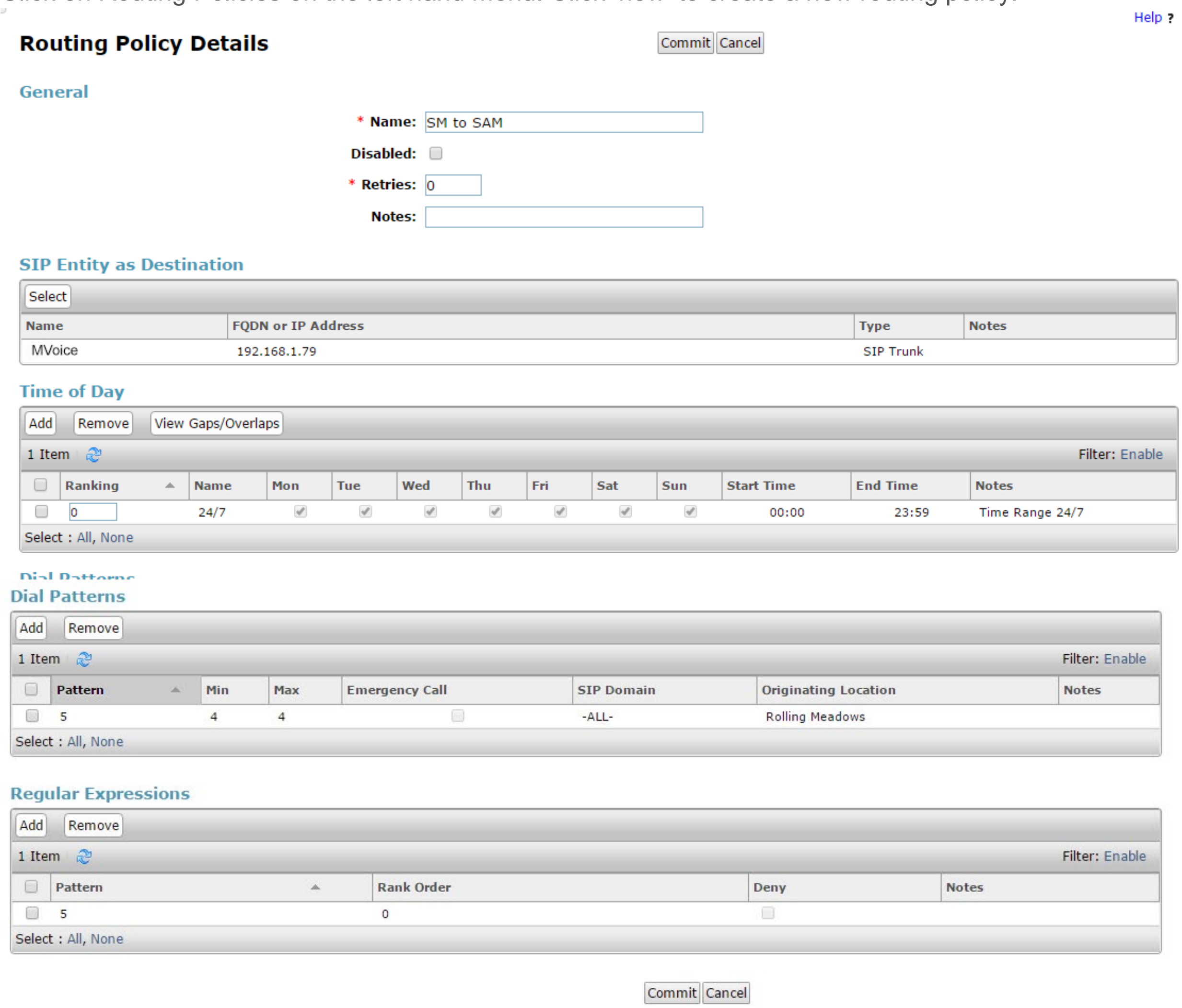


Commit the changes to save.

Customer Initials:

Create a Routing Policy

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



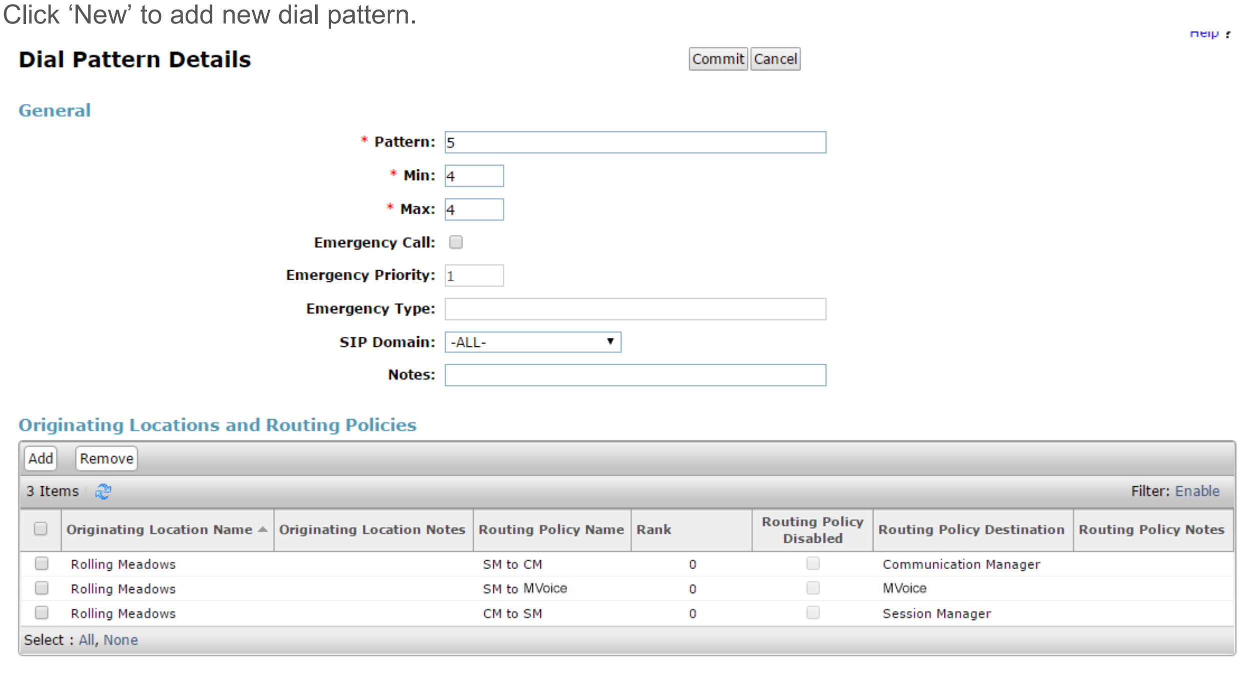
Routing Policy should be configured as follows. Relevant Fields:

- **Name:** add descriptive name for a ASM to MVoice policy
- **Sip Entity as Destination:** Click 'new" and add the MVoice 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

Create New Dial Pattern

Select Dial Patterns from the left side menu.
Click ‘New’ to add new dial pattern.



Dial patterns should be configured as followed. Relevant fields:

- **Pattern: Enter a matching pattern of the MVoice routing number (ex. 5999 or 5)**
- **Min: Selct 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 MVoice.**

Click commit to save changes.

Customer Initials: