Avaya Solution & Interoperability Test Lab

Application Notes for New Voice Technologies Mobicall 8.0.3 with Avaya Aura® Communication Manager 7.0 using Avaya Aura Session Manager 7.0 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate New Voice Technologies Mobicall with Avaya Aura® Communication Manager using Avaya Aura Session Manager. Mobicall is an Alarm generation and distribution solution that connects to Session Manager as a SIP entity.

Readers should pay attention to section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to successfully integrate New Voice Technologies Mobicall with Avaya Aura® Communication Manager using Avaya Aura Session Manager. Mobicall is an Alarm generation and distribution solution that connects to Session Manager as a SIP entity. System alarms are recorded on the Mobicall server and distributed to Communication Manager endpoints.

2. General Test Approach and Test Results

The general test approach was to configure the Mobicall Server to communicate with Communication Manager (CM) via a SIP Trunk connected to Session Manager. Stations present on CM were configured on the Mobicall server and a number was configured to dial Mobicall and create and initiate alarms.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on setting and distributing alarms in different call scenarios with good quality audio. The tests included:

- Mobicall SIP trunk is connected and in Service.
- Mobicall can route Alarms to SIP, Digital and H.323 endpoints.
- Alarms can be set and distributed from CM to Mobicall.
- Priority calling and Whisper Paging functionality can be initiated from Mobicall
- Failover/Service – Tests the behaviour of Eurocross Connect Client during certain failed conditions.

2.2. Test Results

All test cases were passed.

2.3. Support

Provide details for technical support for the member. Please include web sites and telephone number.

Telefon	+41 58 750 11 11
Fax	+41 58 750 11 12
Email	support@newvoice.ch
Internet	mobilisierung.com

3. Reference Configuration

The configuration shown in Figure 1 was used during the compliance test of New Voice Technologies Mobicall with Communication Manager via Session Manager. Mobicall utilizes a SIP trunk to communicate with Communication Manager handsets

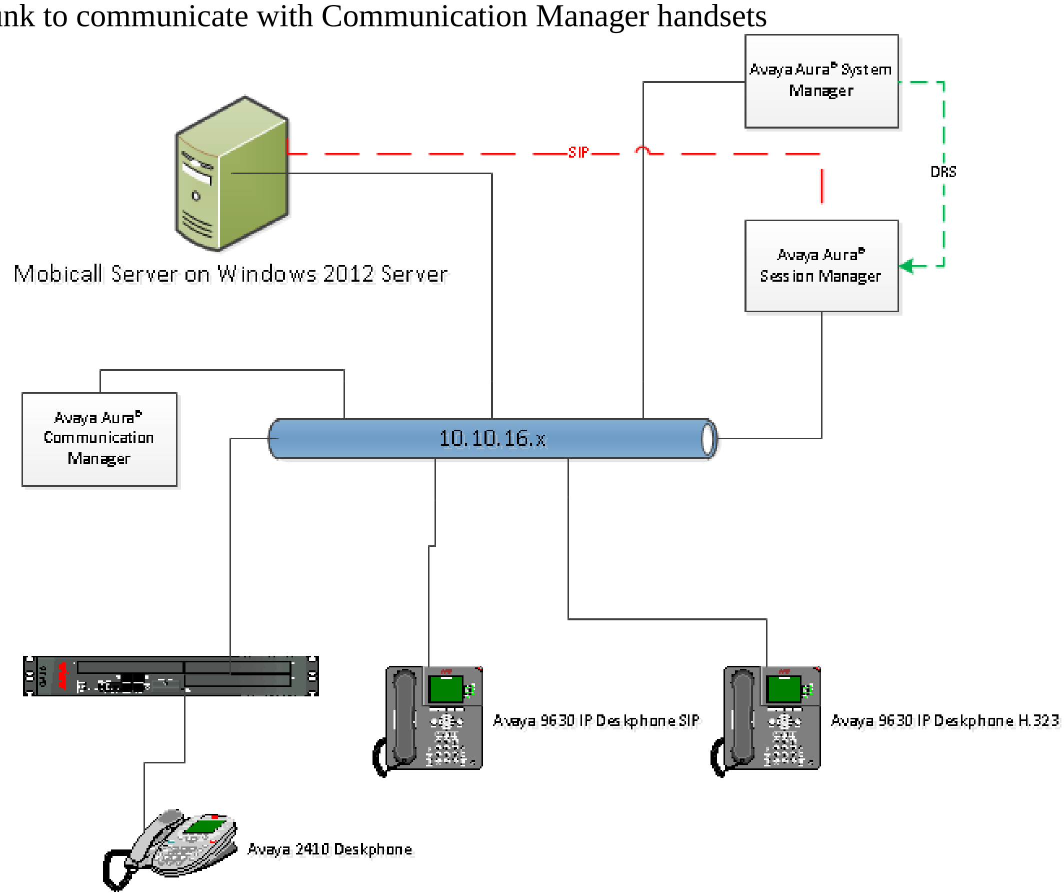


Figure 1: Connection of Mobicall with Avaya Aura® Communication Manager via Session Manager

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® Communicaiton Manager running on VMware Virtual Machine	R7 SP1 R017.00.0.441.22438
Avaya Aura® Session Manager	R7.0 7.0.0.0.700007
Avaya Aura® System Manager	R7.0 7.0.0.0.1626-7.0.9.912
Avaya G430 Media Gateway	FW 37.19.0
Avaya 96x1 Series IP Deskphones H.323	6.6.0.29
Avaya 96x1 Series IP Deskphones SIP	6.5.0
Avaya 2410 Digital Deskphone	N/A
Mobicall	8.0.3

5. Configure Avaya Aura® Communication Manager

This section describes the steps required to allow Communication Manager to communicate with Mobicall. Is it assumed that Communication Manager is installed and configured before implementing the configuration step below. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

The configuration illustrated in this section was performed using Communication Manager System Administration Terminal (SAT).

Configuration steps include:

- Check SIP Trunk Licensing.
- Add entries in the Dial Plan for use with SIP Trunk and routing to Mobicall.
- SIP Trunk Administration (to Session Manager).
- Adding Mobicall Route Pattern.
- Adding Mobicall Access number.
- Setting feature Access Codes.

Using the ***display system-parameters customer-options*** command go to **page 2** and check that the system is sufficiently licensed for SIP Trunks.

```
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 3
             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: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 10
```

Use the ***change node-names ip*** command to add Session Manager

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
SM1677              10.10.16.77
default             0.0.0.0
procr               10.10.16.27
procr6              ::-
```

Use ***change dialplan analysis*** to add a **3** digit dial access code(**dac**) for use in the SIP Trunk, a unform dial plan(**udp**) entry for calling Mobicall and check that there is an entry for feature access codes(**fac**).

```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 2

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   2           7   udp
   7           3   dac
   8           5   udp
   8           7   udp
   827         7   ext
   9           1   fac
   *           3   fac
   #           3   fac
```

Use ***add-signaling-group x*** where x is the number of the group required. Set **Transport Method** to **tcp, Near-end Node Name** to **procr** and **Far-end Node Name** to the entry added in **node-names**. Set the **Far-end Network Region** to **1** and **Direct IP-IP Audio Connections?** to **n**

```
add signaling-group 76                                     Page   1 of   2
                                SIGNALING GROUP

 Group Number: 76             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  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: SM1677
 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
                                                  Alternate Route Timer(sec): 6
```

Use ***add trunk-group x*** where x is the number administered for the signaling group. On **Page 1** set the **Group Type** to **sip**. Set the **TAC** to suitable entry based on the dial plan **dac** administered above. Set the **Service Type** to **tie**, **Signaling group** to the one administered above and **Number of Members** to a number satisfactory for call routing required (**255** shown is the max for this type of trunk group).

```
add trunk-group 76                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 76                   Group Type: sip          CDR Reports: y
  Group Name: ToSM7                       COR: 1        TN: 1        TAC: 776
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 76
                                                    Number of Members: 255
```

On **page 3** set the **Numbering Format** to **private**

```
add trunk-group 76                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

Next a route pattern needs to be added so that call can be routed out of Communication Manager to Session Manager. Use ***change route-pattern x*** where x is the number of the SIP trunk created. Enter the Trunk group created above beside the first **Grp No,** an **FRL** of **0** and **Numbering Format** of **lev0-pvt**

```
change route-pattern 76                                          Page   1 of   3
                    Pattern Number: 76     Pattern Name: ToSM7
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 76   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 Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  none
```

Next a number must be created that Communication Manager can use to connect to Mobicall. Use ***change uniform-dialplan x*** where x is the first digit used in the dialplan for type udp. Set **Matching Pattern** to **x**, **Len** to the length of the digit string to be dialed, **Del** to **0** and **Net** to **aar**.

```
change uniform-dialplan 8                                        Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                               Percent Full: 0

  Matching                   Insert              Node
  Pattern       Len Del      Digits      Net Conv Num
 8               5   0                   aar  n
```

Now an Automatic Alternate Route(aar) entry must be made for this number. Use ***change aar analysis x*** where x is the number used above. Set **Dialed String** to **x**, **Total Min/Max** to the entry in Len above, **Route Pattern** to the one added above and **Call Type** to **aar.**

```
change aar analysis 8                                            Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    8                     5    5     76       aar          n
```

Mobicall utilizes the Priority and Whisper Paging features on Communication Manager to allow Alarms to barge in on active calls. Use ***change feature-access-codes*** to enter these settings based on valid dial plan entries entered above. On page 3 enter a valid code for **Priority Calling Access Code**.

```
change feature-access-codes                                      Page   3 of  10
                               FEATURE ACCESS CODE (FAC)
                PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code:
      Per Call CPN Unblocking Code Access Code:
                    Posted Messages Activation:        Deactivation:
                  Priority Calling Access Code: *79
```

On page 4 enter a valid code for **Whisper Page Activation Access Code**.

```
change feature-access-codes                                   Page   4 of  10
                               FEATURE ACCESS CODE (FAC)
                             Station Lock Activation:      Deactivation:
            Station Security Code Change Access Code:
                    Station User Admin of FBI Assign:      Remove:
        Station User Button Ring Control Access Code:
                   Terminal Dial-Up Test Access Code:
      Terminal Translation Initialization Merge Code:      Separation Code:
                  Transfer to Voice Mail Access Code:
                Trunk Answer Any Station Access Code:
                    User Control Restrict Activation:      Deactivation:
        Voice Coverage Message Retrieval Access Code:
       Voice Principal Message Retrieval Access Code:
                Whisper Page Activation Access Code: *80
```

Mobicall uses a virtual extension that must be valid on the Communication Manager for Alarm routing to Mobicall for distribution to Communication Manager endpoints. Use ***add station x*** where x is a valid extension on Communication Manager but is not an extension that will be used to register a physical endpoint. On **Page 1** enter the **Extension, Type and Name**.

```
add station 8270999                                        Page   1 of   5
                                     STATION

Extension: 827-0999                      Lock Messages? n               BCC: 0
     Type: 9620                          Security Code:                  TN: 1
     Port: S00020                      Coverage Path 1:                 COR: 1
     Name: New Voice Virtual           Coverage Path 2:                 COS: 1
                                       Hunt-to Station:               Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 827-0999
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
                              Short/Prefixed Registration Allowed: default

                                              Customizable Labels? y
```

On page 2 set **EC500 State** to **enabled**

```
change station 8270999                                        Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe           Auto Select Any Idle Appearance? n
          LWC Activation? y                      Coverage Msg Retrieval? y
  LWC Log External Calls? n                                 Auto Answer: none
             CDR Privacy? n                            Data Restriction? n
   Redirect Notification? y                  Idle Appearance Preference? n
 Per Button Ring Control? n                Bridged Idle Line Preference? n
   Bridged Call Alerting? n                    Restrict Last Appearance? y
  Active Station Ringing: single
                                                        EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                    EC500 State: enabled
```

Next use **change off-pbx-station-mapping x** where x is the virtual station added above. On Page 1 set **Application** as **EC500**, **Phone number** as the number used to call Mobicall and **Trunk Selection** as **aar**.

```
change off-pbx-telephone station-mapping 8270999              Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number    Trunk        Config  Dual
 Extension                  Prefix                     Selection    Set     Mode
 827-0999          EC500        -      88888            aar         3
```

On page 2 set **Call Limit** to the number of trunks configured on Mobicall and check that **Mapping Mode** and **Bridged Calls** are set to **both**

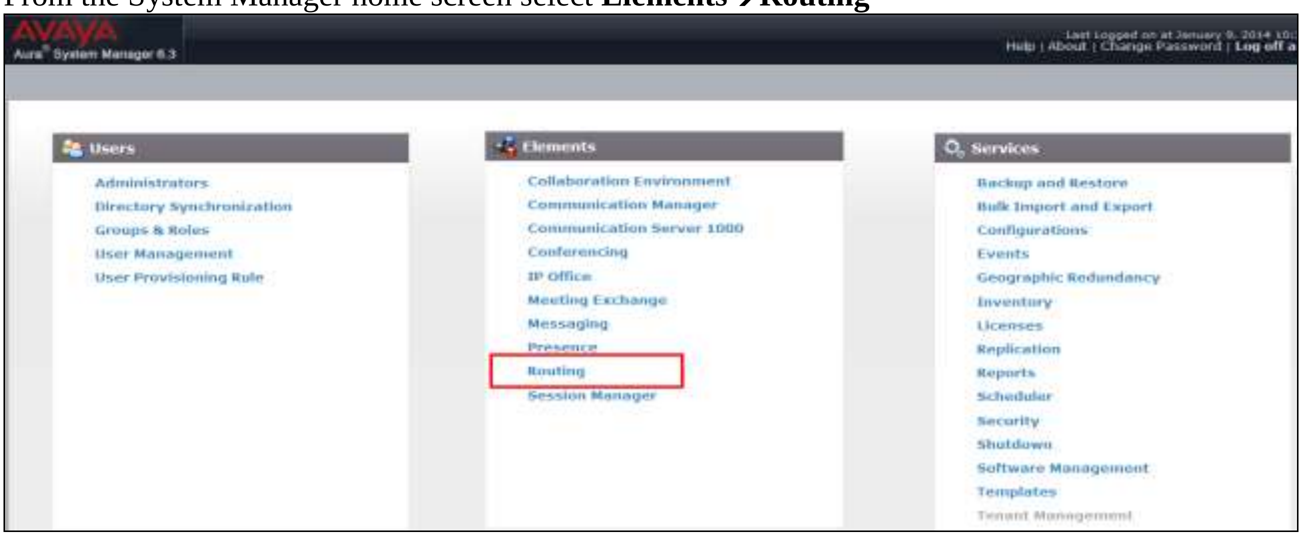
```
change off-pbx-telephone station-mapping 8270999              Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Appl     Call        Mapping    Calls        Bridged       Location
 Extension      Name     Limit       Mode       Allowed      Calls
 827-0999       EC500     10        both        all          both
```

6. Configure Avaya Aura® Session Manager

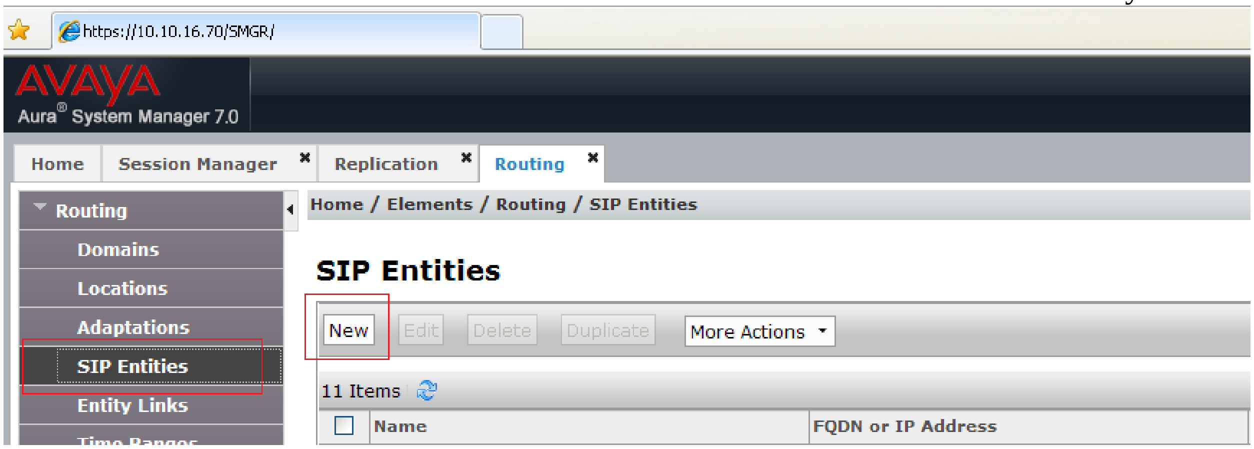
In this section the configuration steps required to connect Mobicall to Session Manager as a SIP entity are described. It is assumed that an existing Session manager instance has already been installed and configured as this is out with the scope of this document. All Configuration steps were carried out using Avaya Aura® System Manager. Configuration steps include:

- Adding Mobicall SIP Entity.
- Adding an Entity Link.
- Adding a Routing Policy.
- Adding a Dial Pattern.

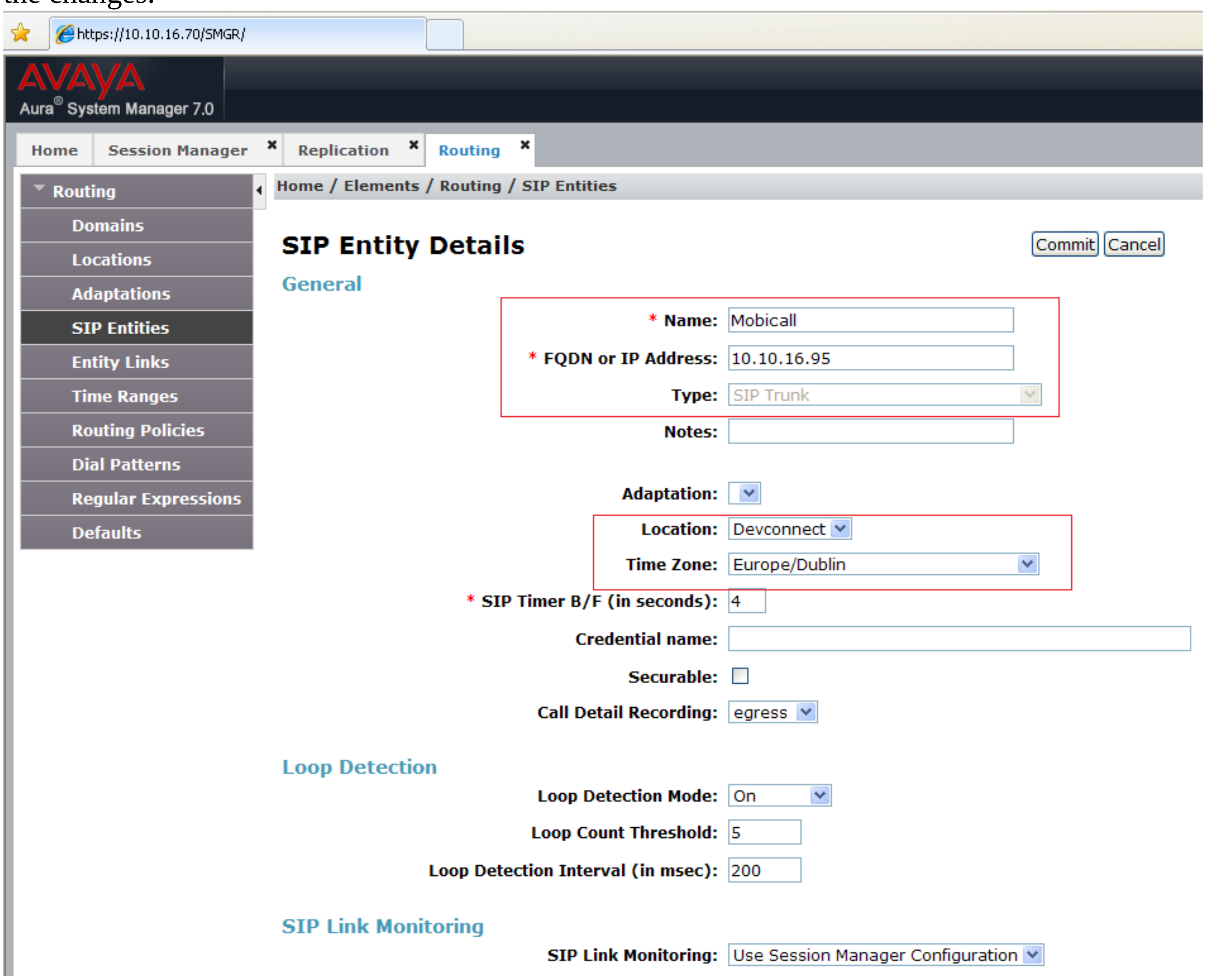
From the System Manager home screen select **Elements→Routing**



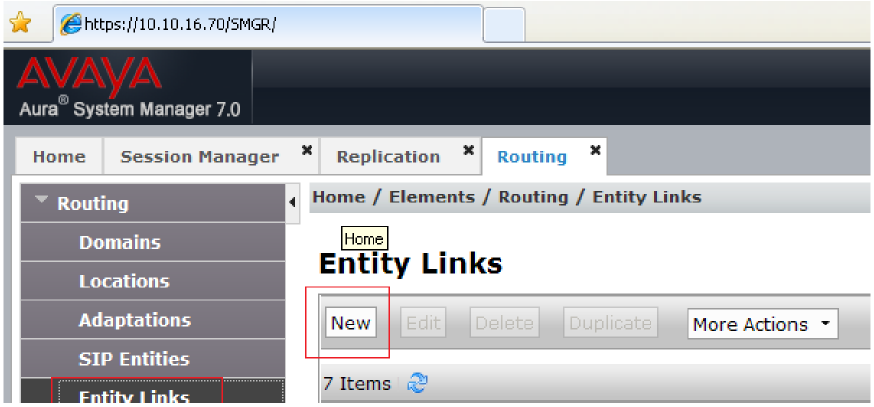
Select **SIP Entities** from the left hand menu and click on **New** to add the Mobicall entity



Enter a descriptive **Name** and the **IP Address** of the Mobicall Server. Set **Type** as **SIP Trunk** and choose a **Location** and **Time Zone** from the drop down menus. Click on **Commit** to save the changes.

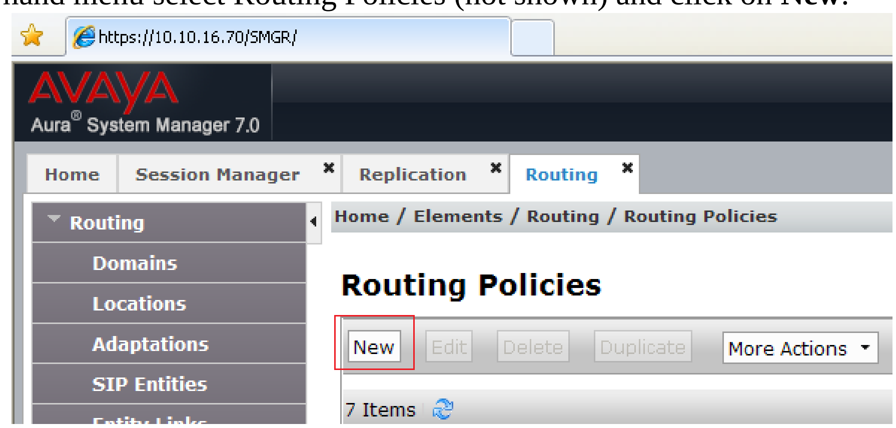


Next add an Entity link between the Mobicall and Session Manager entities. Select **Entity Links** from the left hand menu and click on **New.**



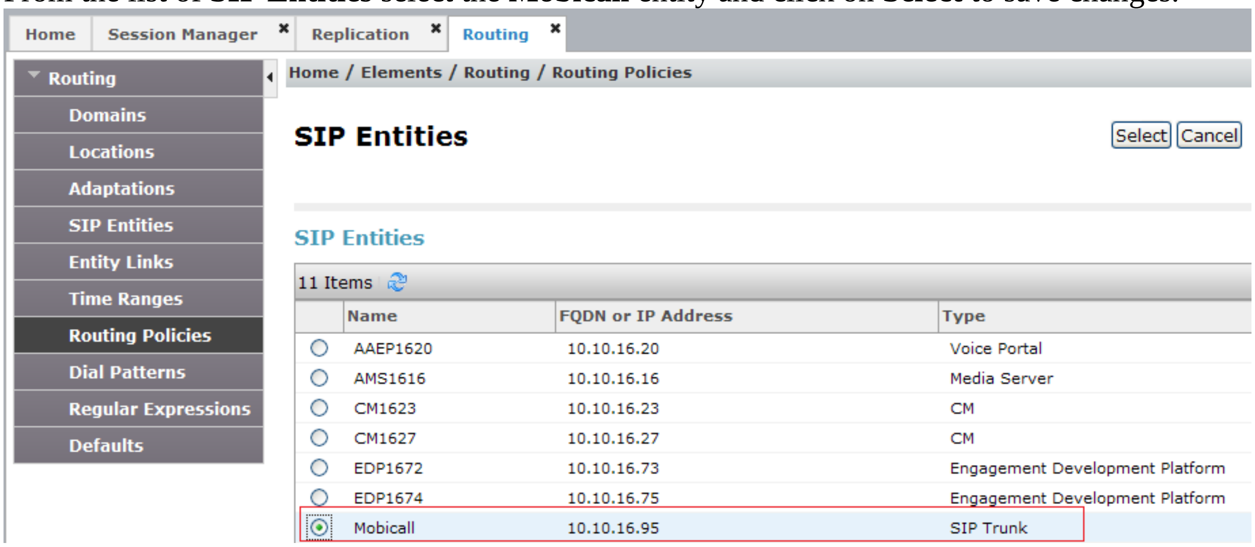
Enter a descriptive **Name** and then select the Session Manager as **SIP Entity 1** from the drop down. Select the **Mobicall** entity as **SIP Entity 2**. Select the **Protocol** administered on the mobicall server. **TCP** was used during testing. The ports will automatically change to the default **5060**. Click on **Commit** to save changes.

From the left hand menu select Routing Policies (not shown) and click on **New**.

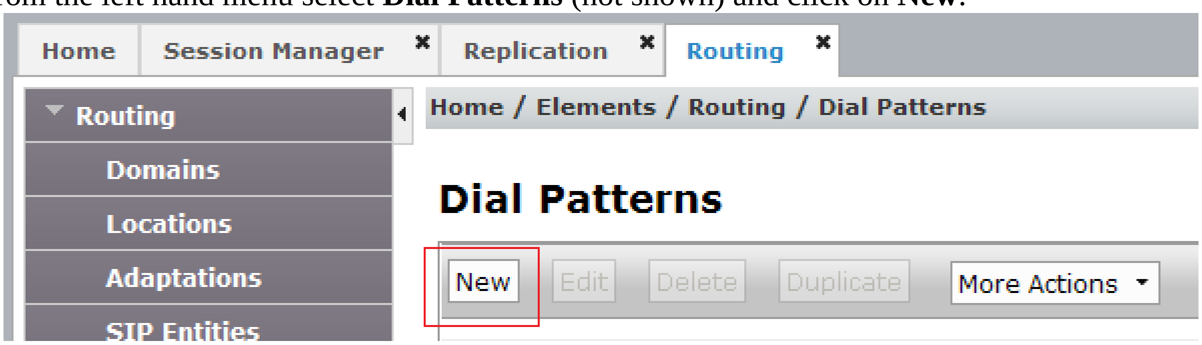


Enter a descriptive **Name** and under **SIP Entity as Destination** click on **Select**.

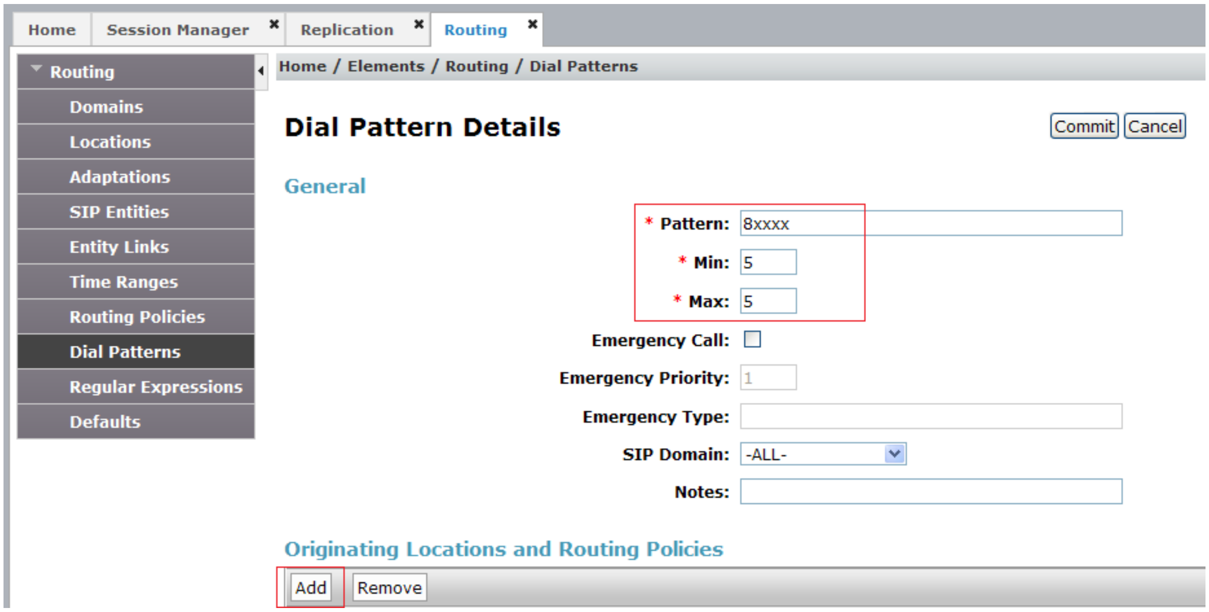
From the list of **SIP Entities** select the **Mobicall** entity and click on **Select** to save changes.



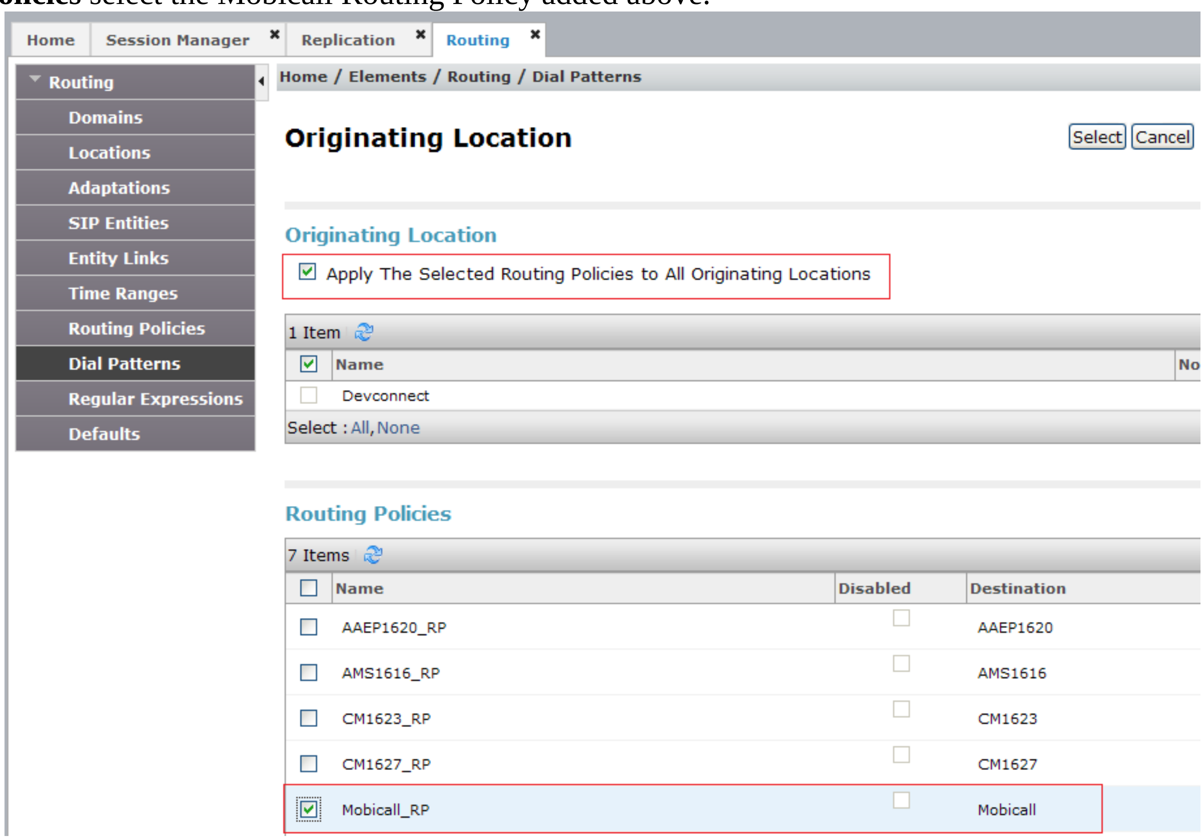
From the left hand menu select **Dial Patterns** (not shown) and click on **New**.



Enter the **Pattern** that will route calls to the Mobicall server and set the **Min** and **Max** to the length of the number to be dialed. Under **Originating Location and Routing Policies** click on **Add**.



Select **Apply the Selected Routing Policy to All Originating Locations** and under **Routing Policies** select the Mobicall Routing Policy added above.

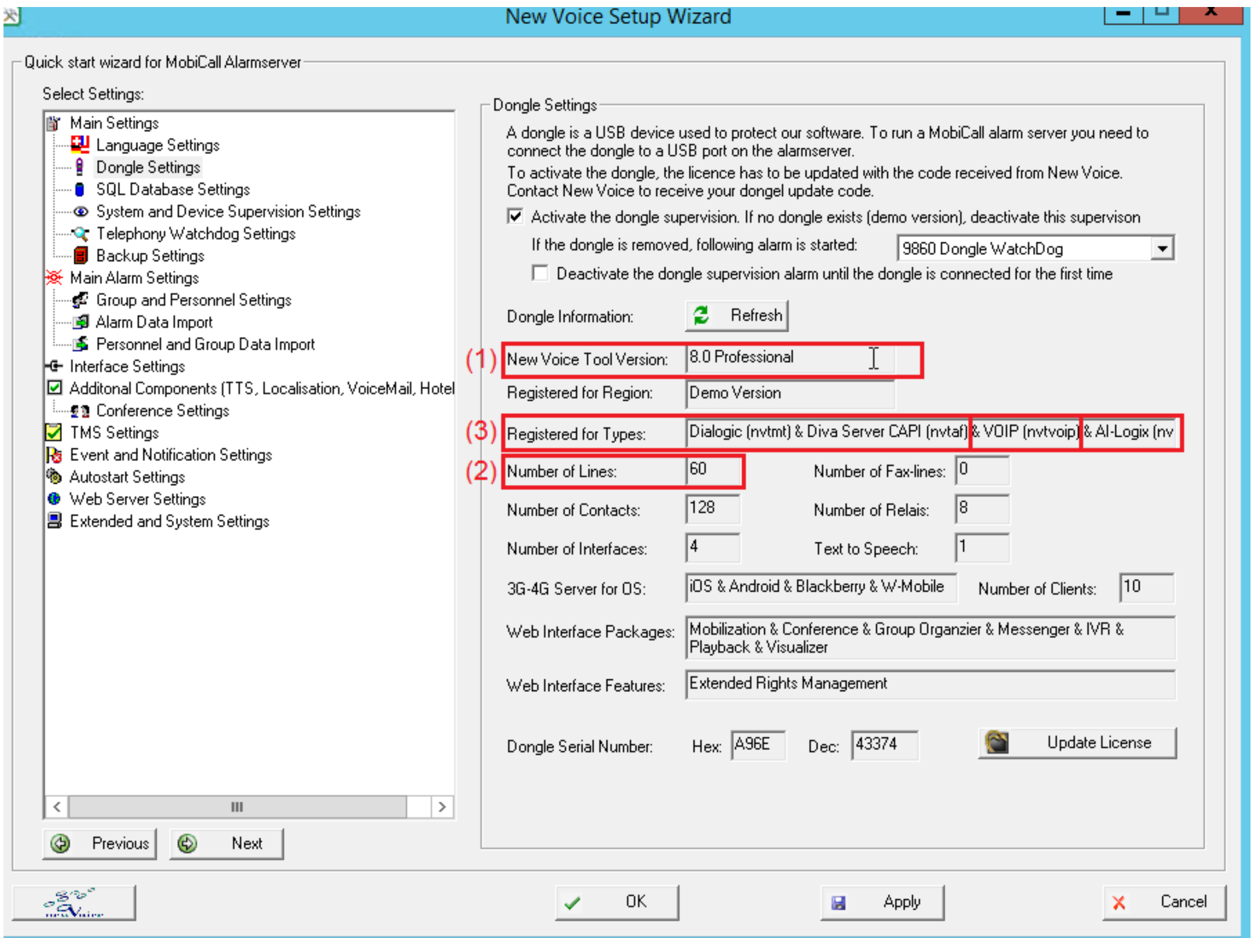


7. Configure NewVoice Technology MobiCall

Setting up the MobiCall installation is not described here. Please refer to the product documents in **Section 10**. All configuration is carried out in this section using the New Voice Setup Wizard

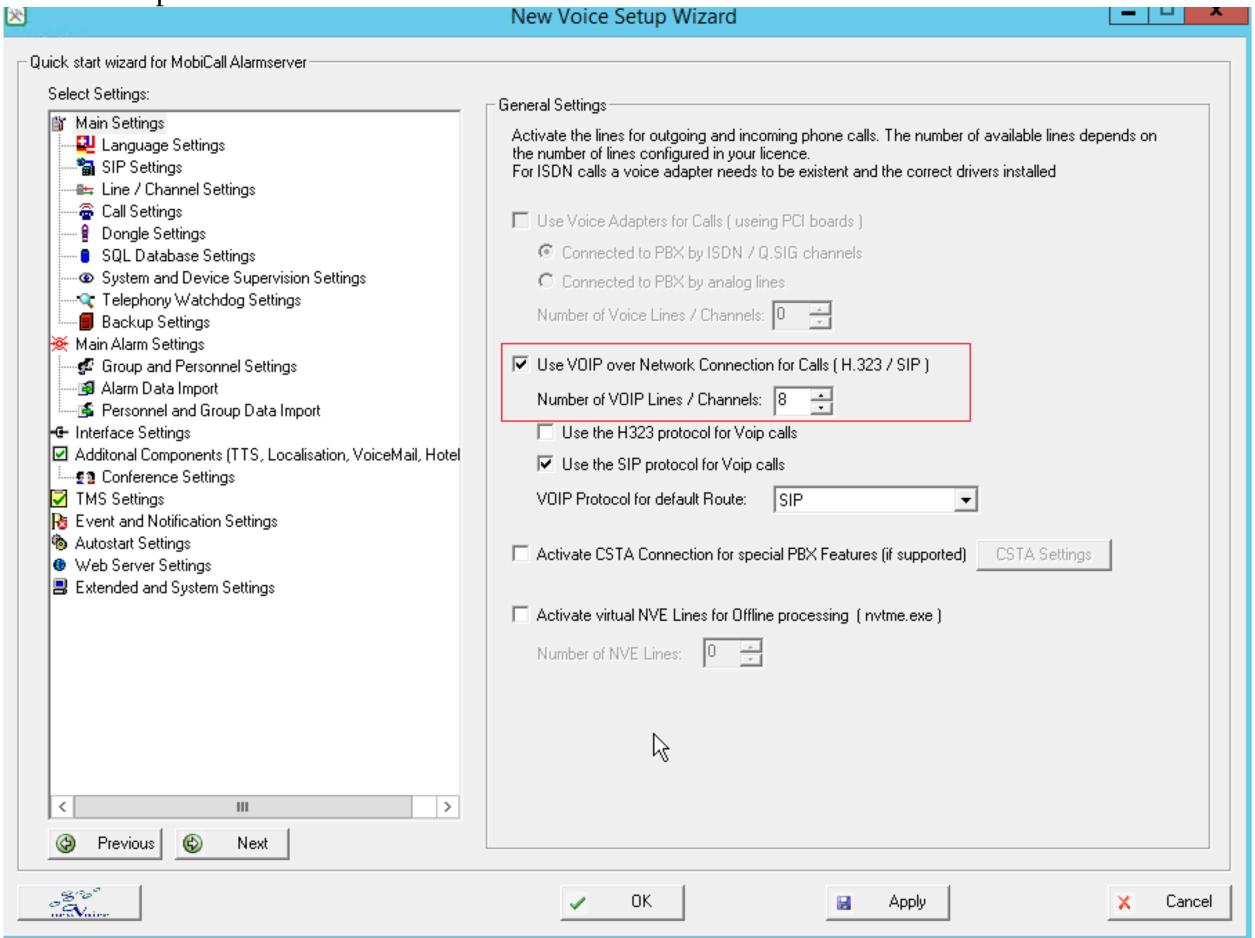
7.1. SIP Settings

Select **Main Settings→Dongle Settings**
Check the licenses on USB Dongle
The minimum required licenses are (1) **NewVoice Tool Version 8.x** with (2) **Number of Lines 2** and (3) **Registered for Types nvtvoip**.



Select **Main Settings**

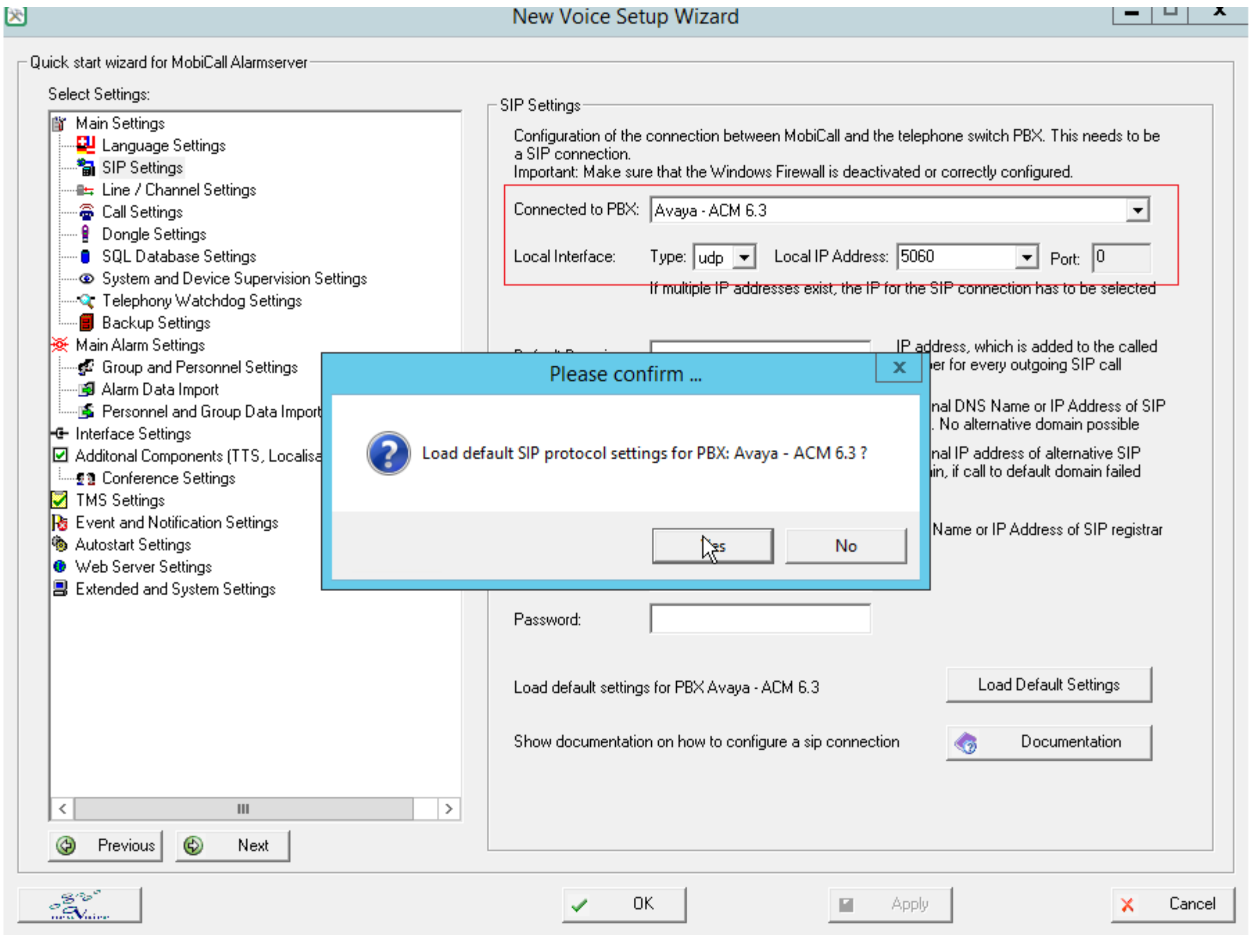
Select Use VOIP over Network Connection for Calls. Set **Number of VOIP Lines / Channels** up to the licensed number



Select **Main Setting→SIP Settings**

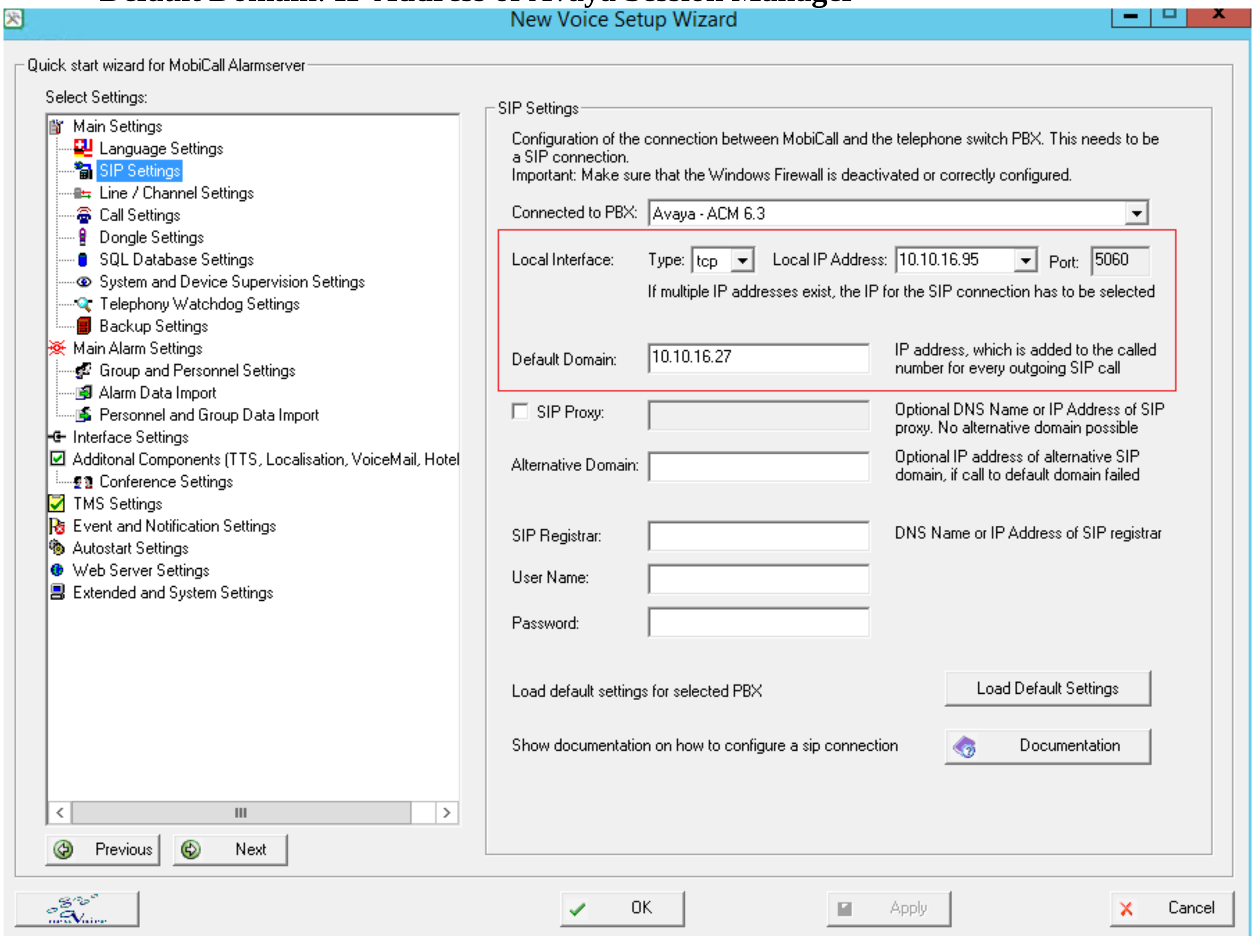
Select Avaya-ACM 6.3 from the Connected to PBX drop down. When asked to Confirm click on **Yes.**

If there is no exact entry for the pbx software version, select the one below.

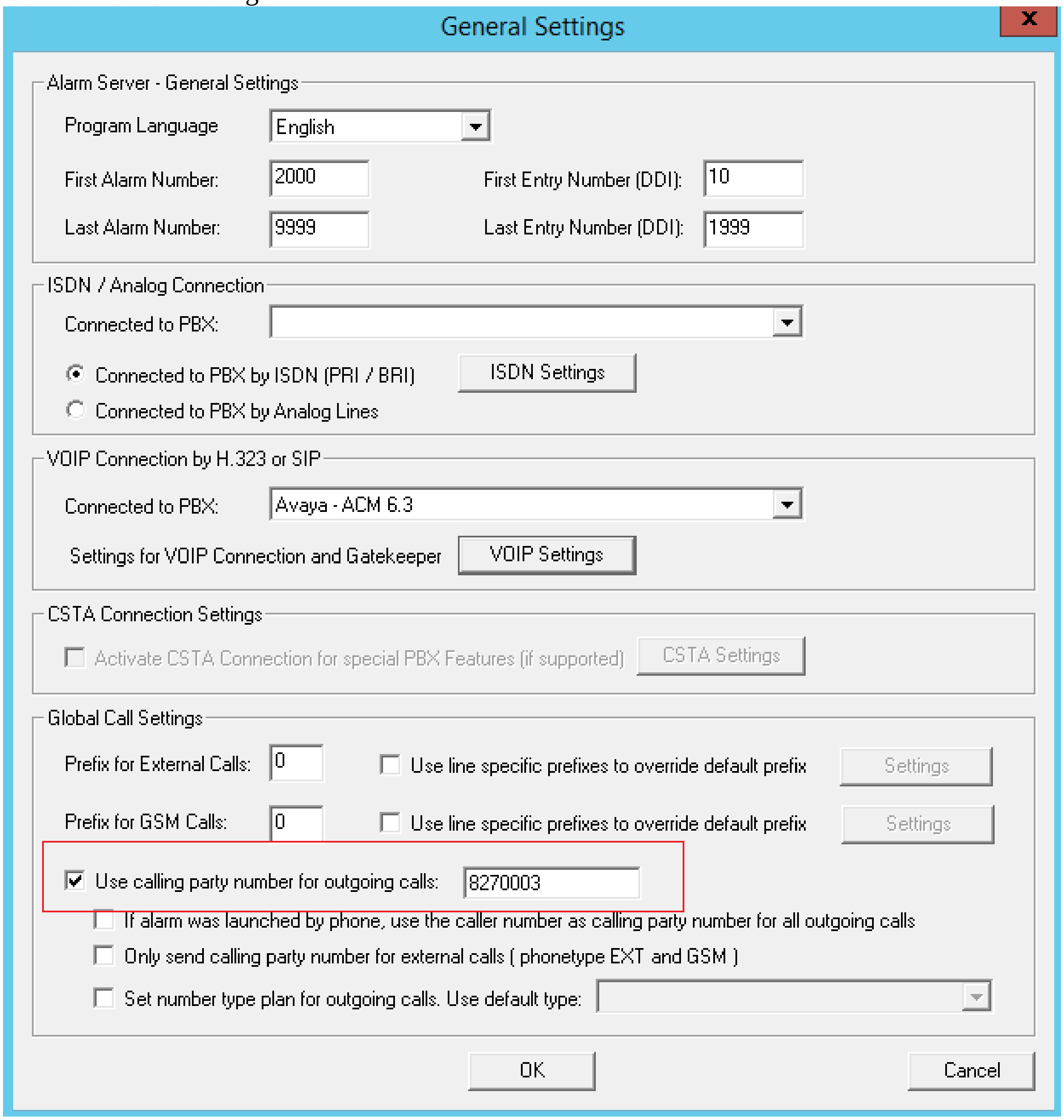


When the Profile is loaded enter the following required settings

- **Local Interface Type**: **tcp**
- **Local Interface IP Address**: **IP Address of MobiCall**
- **Local Interface Port**: **5060**
- **Default Domain**: **IP Address of Avaya Session Manager**



Select **Main Alarm Settings**
Select **Use Calling party number for outgoing calls** and enter and unused extension on the Communication Manager



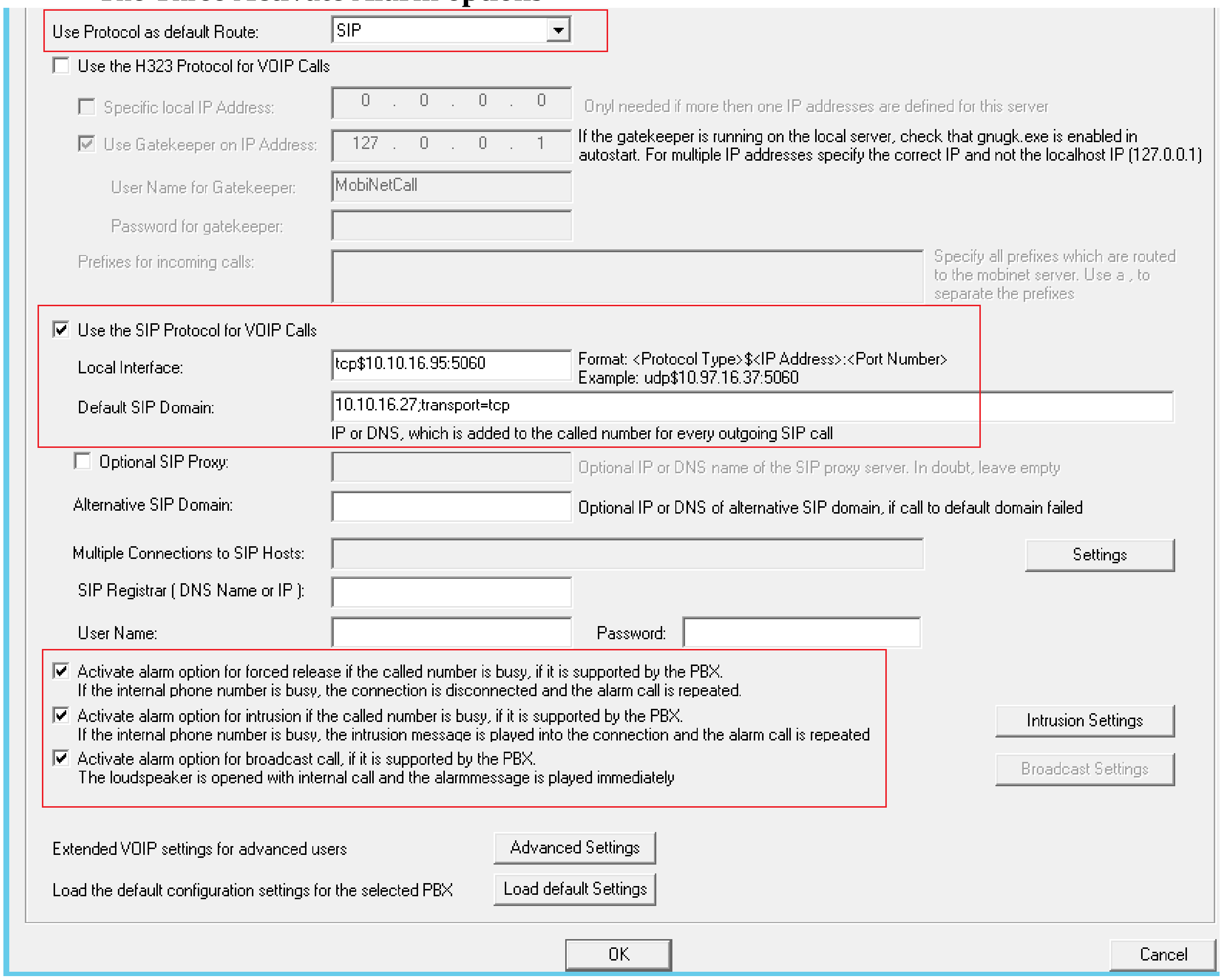
7.2. Feature activation and configuration

All configuration in this section is carried out using the New Voice Alarm Central

Select **Main Settings→ VOIP Settings**

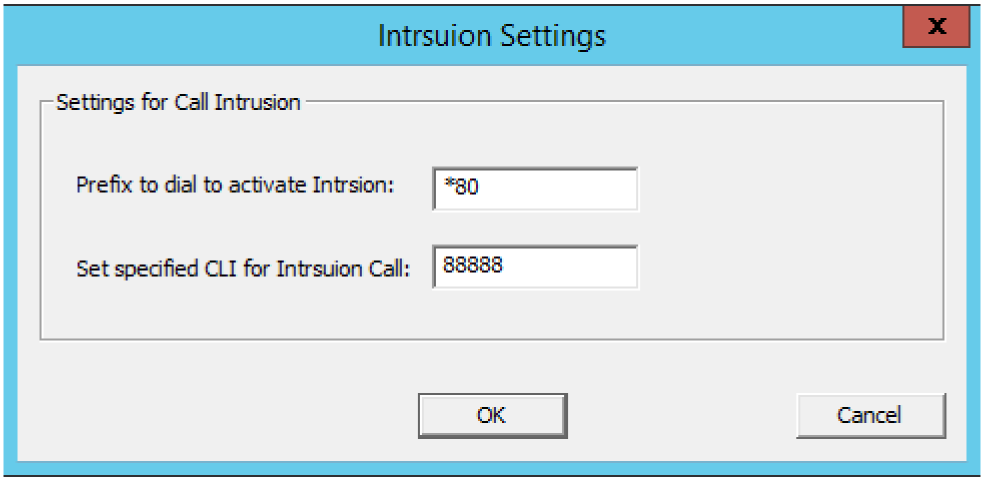
Activate the following features as shown.

- **Use Protocol as default Route**
- **Use the SIP Protocol for VOIP Calls**
- The Three **Activate Alarm options**



)

Select Main Settings → VOIP Settings → Intrusion Settings
Set Prefix to dial to activate Intrusion: section 5: ***change feature-access-codes***
Set specified CLI for Intrusion Call: section 5: ***off-pbx-station-mapping***



8. Verification Steps

This section describes the checks that can be carried out to verify the connection between Mobicall and Communication Manager.

8.1. Session Manager

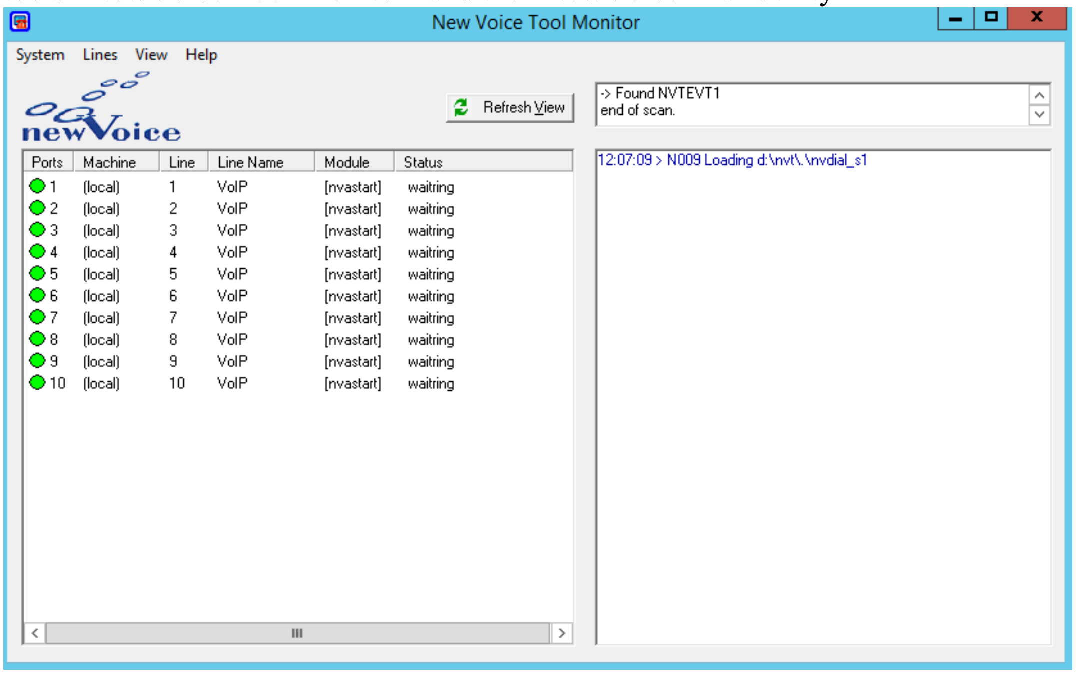
Select Elements → Session Manager from the Home screen (not shown) and click on the value under Entity Monitoring

Verify that the Mobicall entry is UP.

○	MSG1689	10.10.16.89	5060	TCP	FALSE	UP	200 OK	UP
○	SM214	10.10.16.214	5060	TCP	FALSE	UP	200 OK	UP
○	CM1627	10.10.16.27	5060	TCP	FALSE	UP	200 OK	UP
○	Mobicall	10.10.16.95	5060	TCP	FALSE	UP	200 OK	UP
○	CM1623	10.10.16.23	5060	TCP	FALSE	UP	200 OK	UP

8.2. MobiCall Line Monitor and Dial utility

Newvoice Tool Monitor can be found as shortcut on the desktop.
A successful communication between MobiCall and the Session Manager can be verified via the tools “NewVoice Tool Monitor” and the “NewVoice Dial Utility”



Alarmcentral – Extras – Dial Utility
Enter an internal number and press **Dial** to start an outgoing call

9. Conclusion

These Application Notes describe the configuration steps required for New Voice Technologies Mobicall to interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. All feature functionality and serviceability test cases were completed successfully as outlined in Section 2.2.

10. Additional References

This section references the Avaya and Enghouse product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at *http://support.avaya.com*.

[1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509

[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205

[3] *Administering Avaya Aura® Session Manager*, Release 6.3, 03-603324

Product documentation for NewVoice Technologies MobiCall can be obtained by visiting the following website www.mobilisierung.com

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Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.