Avaya Solution & Interoperability Test Lab

Application Notes for configuring novaconf from novalink with Avaya Aura® Communication Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration for connecting the novalink novaconf via SIP Trunks to Avaya Aura® Communication Manager using Avaya Aura® Session Manager.

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

The purpose of this document is to describe the configuration for connecting the novalink novaconf, via a SIP trunk interface, to Avaya Aura® Session Manager, in order for various endpoints on Avaya Aura® Communication Manager dial into or receive calls from novaconf in order to establish a conference call.

novaconf is an application which is used in a health care, hotel or industrial environment for to allow users setup conference calls using an existing telephone system such as Communication Manager. novaconf offers all the conferencing possibilities, which make it easier to reach the persons required. Thus, the Conference Server is able to call and look for anyone at various telephone numbers. Some of the features of novaconf include:

- Dial Out.
  With conferences programmed to a certain time, a person is automatically called by the Server and connected to the conference.
- Dial In.
  Alternately, one can dial into the conference using the specific access data, received in an email.
- Ad Hoc.
  With the simple and clear desktop ad-hoc conferences can be setup on the spot.

2. General Test Approach and Test Results

This section describes the compliance testing used to verify interoperability of novaconf with Communication Manager and covers the general test approach and the test results. Calls were made to novaconf over SIP trunks connecting Session Manager and novaconf. novaconf was configured as a SIP Entity on Session Manager allowing calls route between novaconf and Communication Manager via Session Manager.

novaconf was manually configured using the web interface to setup conference calls for Communication Manager endpoints.

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 testing evaluated the ability of novaconf to handle conference calls. These conferences are then accessed by Communication Manager users over SIP trunks. Test cases are selected to exercise a sufficiently broad segment of functionality to have a reasonable expectation of interoperability in production configurations. Serviceability testing will also be conducted to assess the reliability of the solution. These included accessing the conference bridge on novaconf from Avaya SIP/H.323/Digital endpoints.

- Dialing into a conference.
- Having novaconf dial out to initiate a conference.
- Serviceability testing consisted of verifying the ability of novaconf to recover from power or network interruption to both Communication Manager and novaconf.

2.2. Test Results

All test cases were executed successfully.

2.3. Support

Technical support can be obtained for novaconf from the website http://www.novalink.ch/en/ or from the following.

novalink GmbH
Businesstower
Zuercherstrasse 310
8500 Frauenfeld
Switzerland
helpdesk@novalink.ch
Phone: +41 52 762 66 77
Fax: +41 52 762 66 99

3. Reference Configuration

The configuration in **Figure 1** is used to compliance test novaconf with Communication Manager registering with Session Manager as a third party SIP entity. Calls are made to novaconf using SIP trunks.

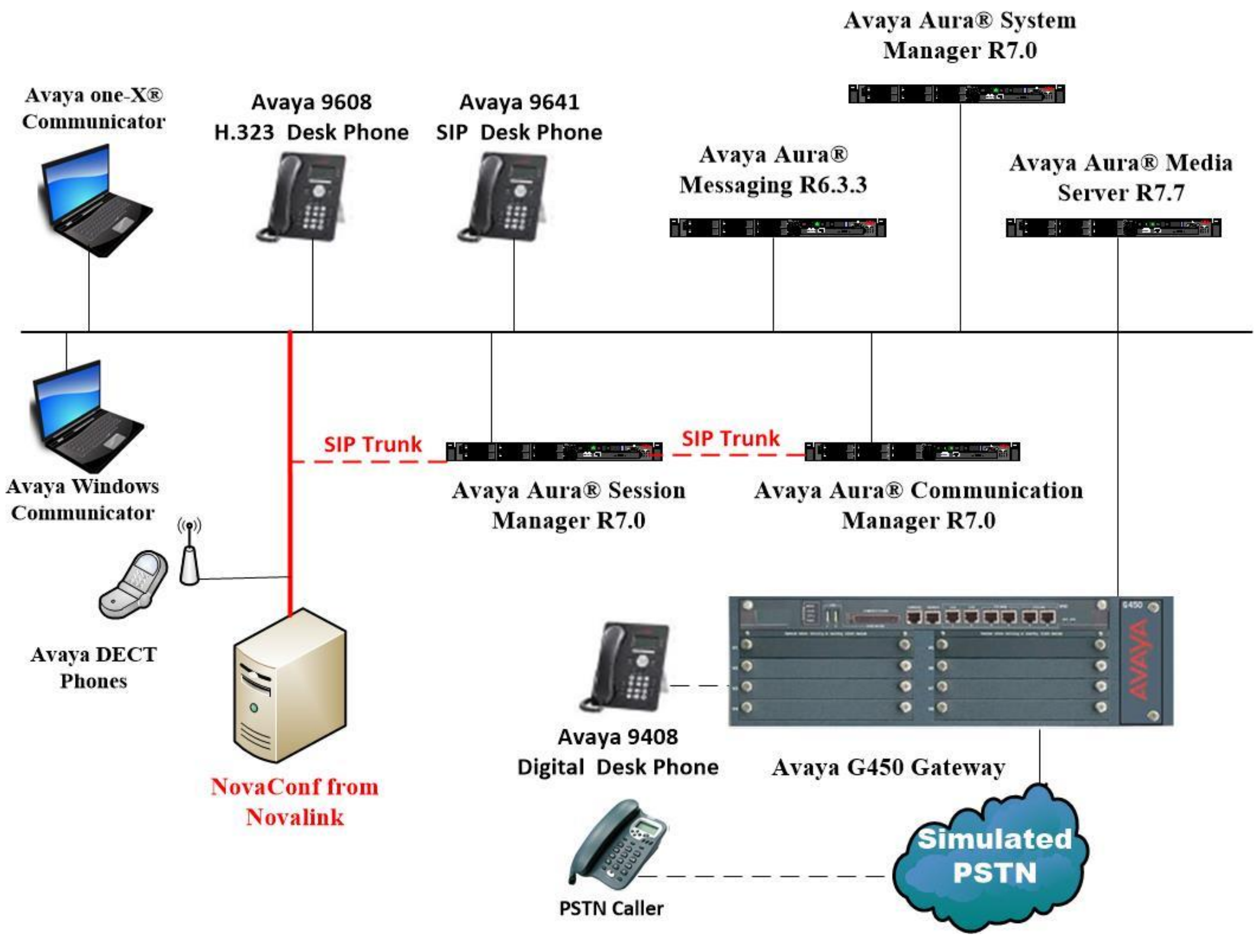


Figure 1: Connection of novaconf from novalink with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager running on a virtual server	System Manager 7.0.1.1 Build No. - 7.0.0.0.16266 Software Update Revision No: 7.0.1.1.065378 Service Pack 1
Avaya Aura® Session Manager running on a virtual server	Session Manager R7.0 SP1 Build No. – 7.0.1.1.701114
Avaya Aura® Communication Manager running on a virtual server	R7.0 R017x.00.0.441.0 00.0.441.0-23169
Avaya Media Server running on a virtual server	Media Server SYSTEM R7.7.0.8 Media Server R7.7.0.200
Avaya G450 Gateway	37.19.0 /1
Avaya Aura® Messaging	R6.3.3
Avaya 9608 H323 Deskphone	96x1 H323 Release 6.6.028
Avaya 9641 SIP Deskphone	96x1 SIP Release 7.0.0.39
Avaya 9408 Digital Deskphone	V2.0
Avaya DECT Handsets	3725 DH4 (R3.3.11) 3720 DH3 (R3.3.11)
Avaya one-X® Communicator H.323	R6.2.4.07-FP4
Avaya Communicator for Windows	R2.1.3.80
novalink novaconf running on a Windows 2012 virtual server	9.8

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

The configuration operations described in this section can be summarized as follows:

- Verify System Parameters Customer Options.
- System Features and Access Codes.
- Administer Dial Plan.
- Administer Route Selection for calls to novaconf.
- Configure Network Region and IP Codec.

Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

5.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that **Maximum Administered SIP Trunks** has sufficient capacity.

```
display system-parameters customer-options                     Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 250
           Maximum Concurrently Registered IP Stations: 18000 2
             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: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 319
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

On **Page 3**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? y
```

On **Page 5**, ensure that **Uniform Dialing Plan** is set to **y**.

```
display system-parameters customer-options                      Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                     Multiple Locations? n
                                            System Management Data Transfer? n
          Personal Station Access (PSA)? y            Tenant Partitioning? y
                        PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                   Port Network Support? y            Time of Day Routing? y
                       Posted Messages? y       TN2501 VAL Maximum Capacity? y
                                                      Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
```

5.2. System Features and Access Codes

For the testing, **Trunk-to Trunk Transfer** was set to **all** on **page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 10** for supporting documentation.

```
display 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
```

Use the **display feature-access-codes** command to verify that a FAC (feature access code) has been defined for both AAR and ARS. Note that **8** is used for AAR and **9** for ARS routing.

```
display feature-access-codes                                         Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *25    Deactivation: #25
```

5.3. Administer Dial Plan

It was decided for compliance testing that all calls beginning with 49 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager and therefore to novaconf. In order to achieve this, automatic alternate routing (aar) would be used to route the calls. The dial plan and aar routing analysis need to be changed to allow this.

Type **change dialplan analysis**, in order to make changes to the dial plan. Ensure that **4** is added with a **Total Length** of **4** and a **Call Type** of **udp**.

```
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           4   ext
   3           4   ext
   4           4   udp
   5           4   ext
   6           4   udp
   7           3   dac
   8           1   fac
   9           1   fac
   *           3   fac
   #           3   fac
```

5.4. Administer Route Selection for novaconf Calls

As digits**49xx** were defined in the dial plan as udp (**Section 5.3**) use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to numbers beginning with **49** that are **4** digits in length will be matched. No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

```
change uniform-dialplan 4                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert               Node
  Pattern        Len Del     Digits      Net Conv Num
  49              4   0                  aar   n
                                               n
```

Use the **change aar analysis** x command to further configure the routing of the dialed digits. Calls to and from novaconf begin with **49** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 1**, which contains the outbound SIP Trunk Group.

```
change aar analysis 49                                         Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed                  Total      Route     Call   Node  ANI
    String                 Min  Max   Pattern    Type   Num   Reqd
    49                     4    4      1         unku         n
```

Use the **change route-pattern *n*** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, **Route Pattern Number 1** is used to route calls to trunk group **(Grp No) 1**, this is the SIP Trunk configured in **Appendix**.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: SIPTRK
                             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            unre                                             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
 6: y y y y y n  n            rest                                             none
```

5.5. Configure Network Region and IP Codec

In the Node Names IP form, note the IP Address of the **procr** and the Session Manager (**sm70vmpg**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

```
display node-names ip                                               Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AMS77vmpg           10.10.40.17
CMS18vmpg           10.10.40.36
IPO500V2            10.10.40.20
IPOSE               10.10.40.25
PGDECT              10.10.40.50
aes70vmpg           10.10.40.26
default             0.0.0.0
procr               10.10.40.13
procr6              ::
sm70vmpg            10.10.40.12
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.2**. In this configuration, the domain name is **devconnect.local**. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session manager as **ip-network region 1** is specified in the SIP signaling group.

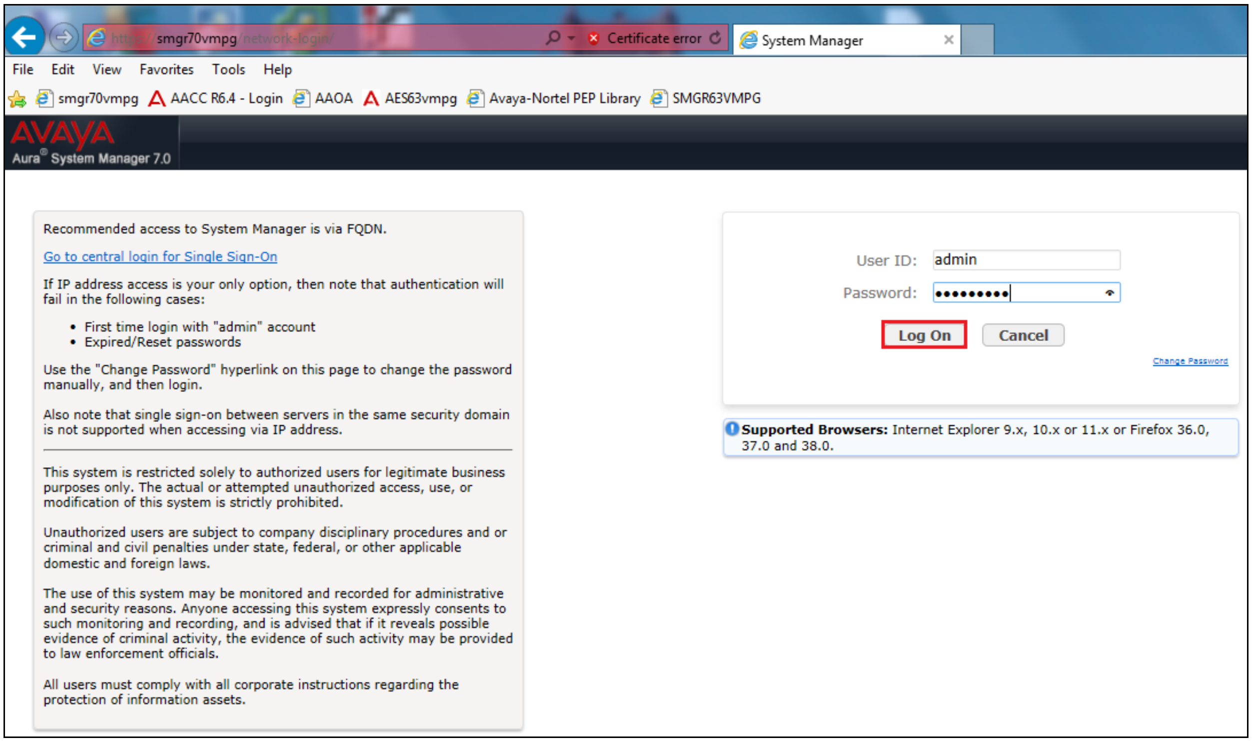
```
display ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devconnect.local
    Name: Default region
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        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
```

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk to and from novaconf. The form is accessed via the **change ip-codec-set n** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the example below includes **G.711A** (a-law), which is supported by novaconf. Note the **Media Encryption** has been set to **none**. This ensures that no media is encrypted.

```
change 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.711A           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption                  Encrypted SRTCP:
 1: none
 2:
 3:
 4:
 5:
```

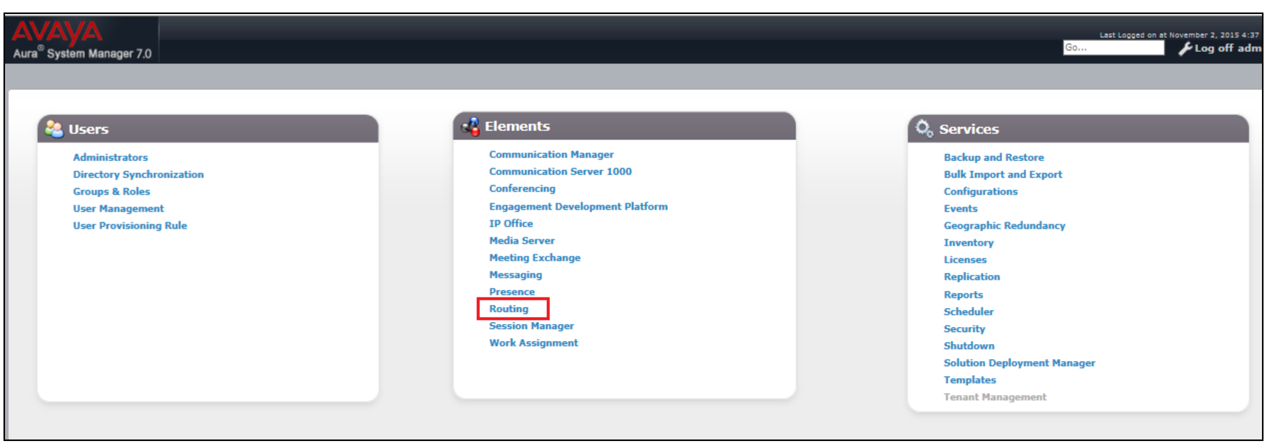
6. Configure Avaya Aura® Session Manager

In order to make changes in Session Manager, a web session to System Manager is opened. Navigate to http://<System Manager IP Address>/SMGR, enter the appropriate credentials and click on **Log On** as shown below.

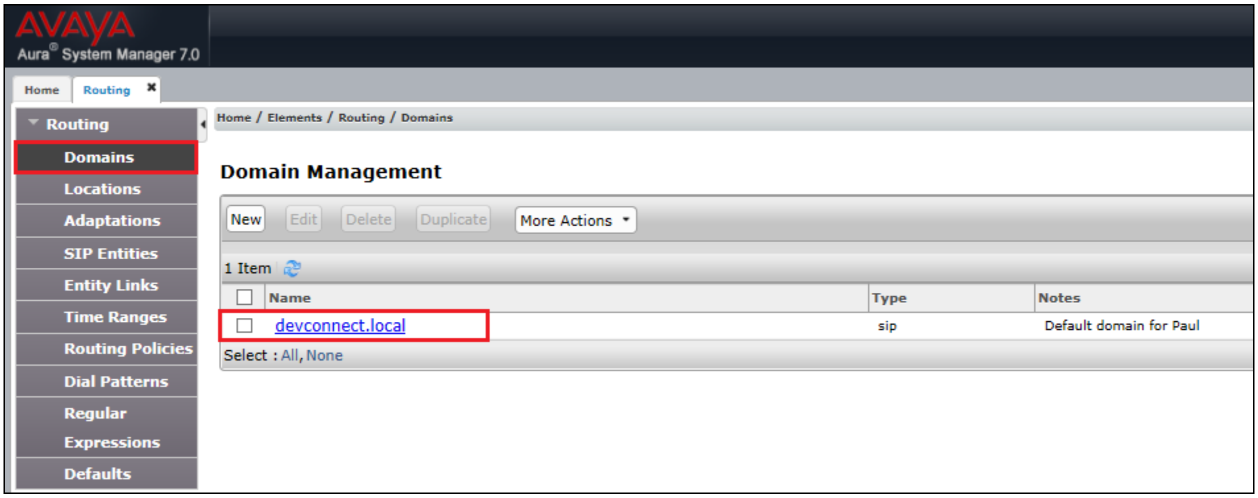


6.1. Configuration of a Domain

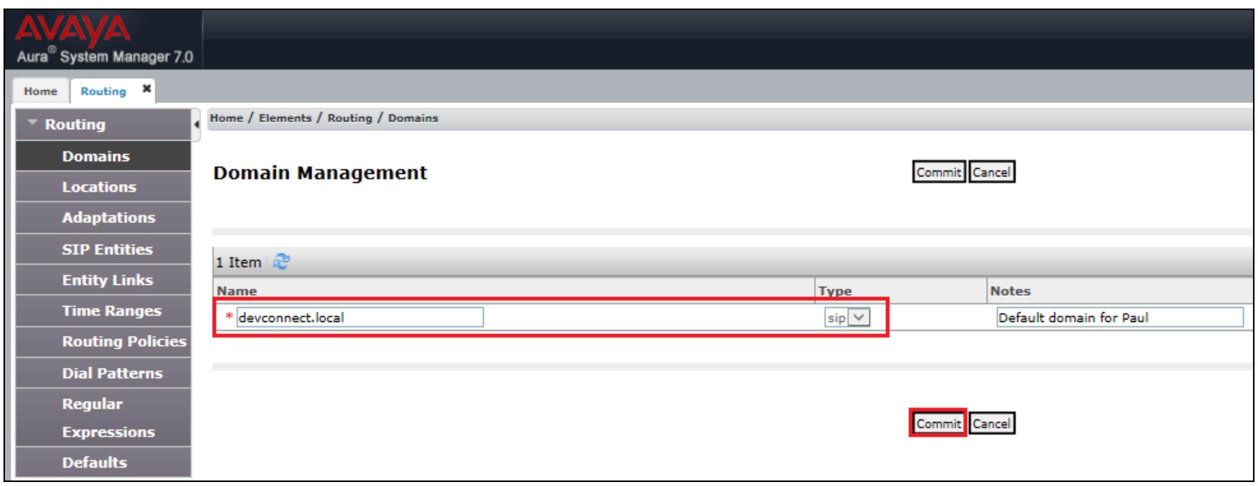
Click on **Routing** highlighted below.



Click on **Domains** in the left window. If there is not a domain already configured click on **New**. In the example below there exists a domain called devconnect.local which has been already configured.

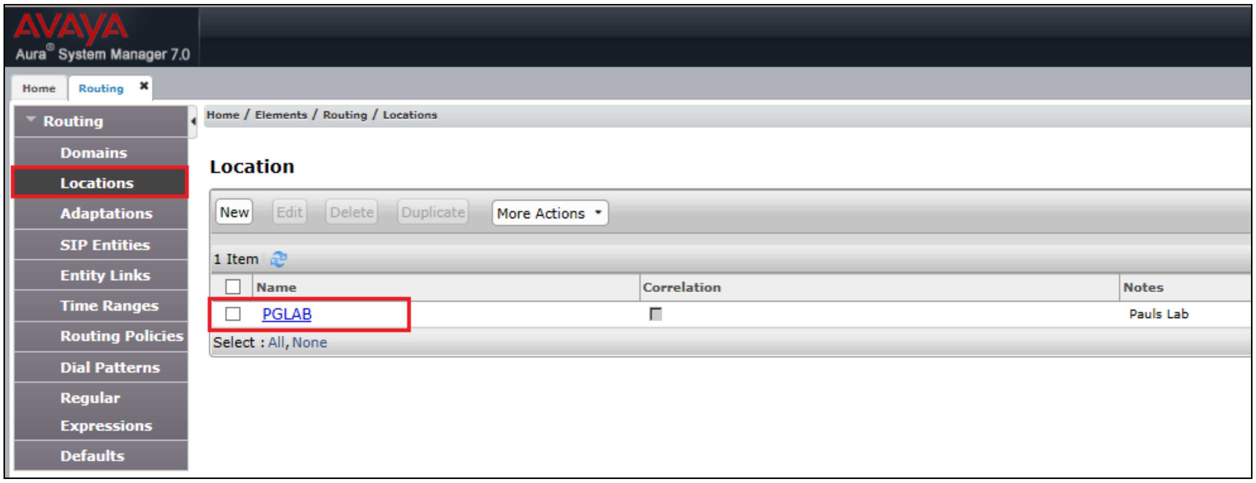


Clicking on the domain name above will open the following window; this is simply to show an example of such a domain. When entering a new domain the following should be entered, once the domain name is entered click on **Commit** to save this.

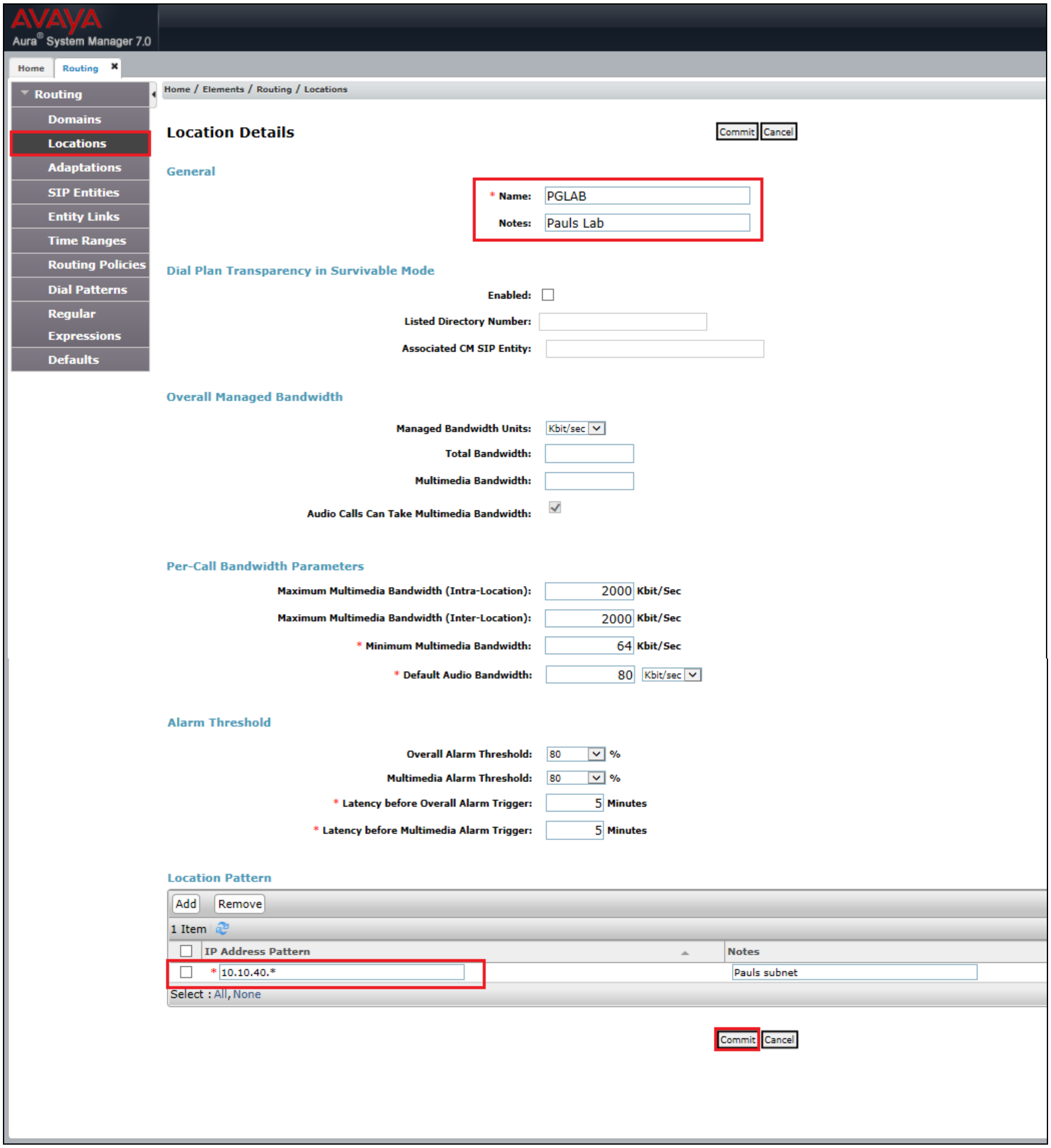


6.2. Configuration of a Location

Click on **Locations** in the left window and if there is no Location already configured then click on **New**, however in the screen below a location called **PGLAB** is already setup and configured and clicking into this will show its contents.



The Location below shows a suitable **Name** with a **Location Pattern** of **10.10.40.***. Once this is configured, click on **Commit**.

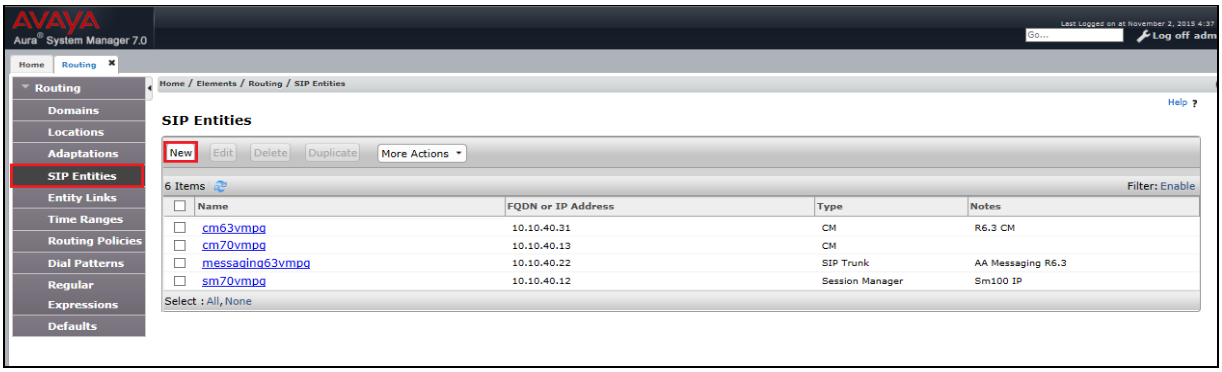


6.3. Configuration of SIP Entities

Clicking on **SIP Entities** in the left window shows what SIP Entities have been added to the system and allows the addition of any new SIP Entity that may be required. Please note the SIP Entities already present for the compliance testing of novaconf.

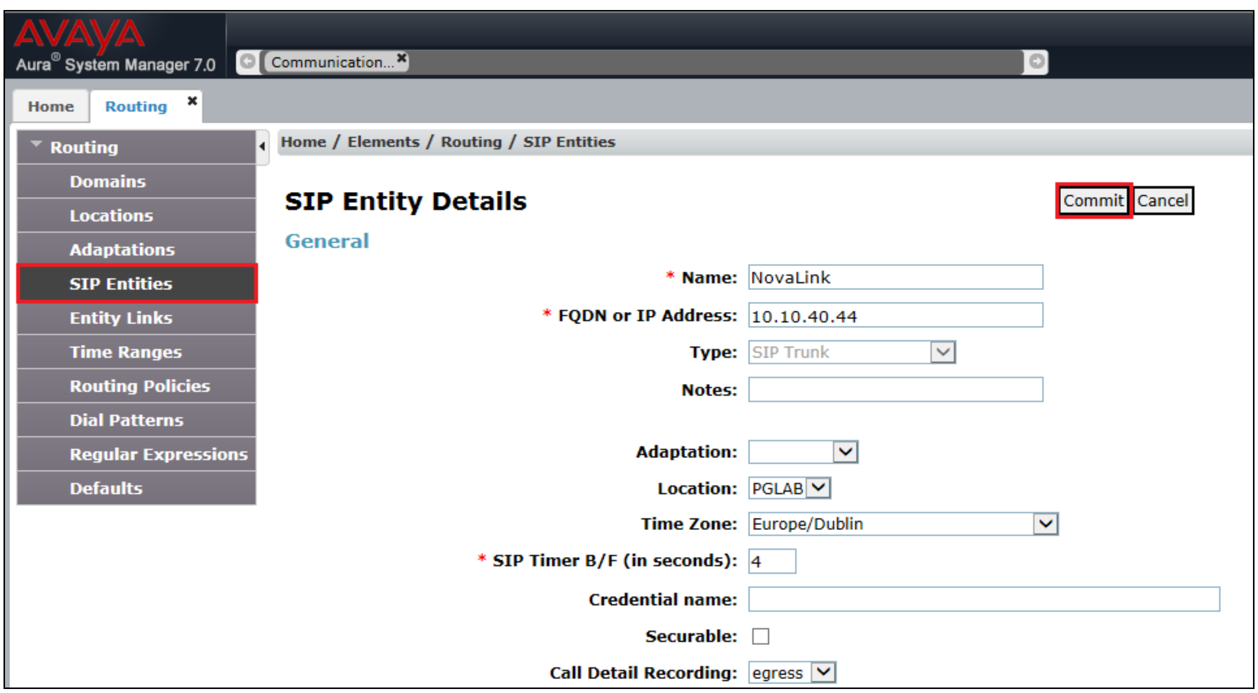
- Communication Manager SIP Entity (cm70vmpg)
- Session Manager SIP Entity (sm70vmpg)

To add a SIP entity, click on **New**.

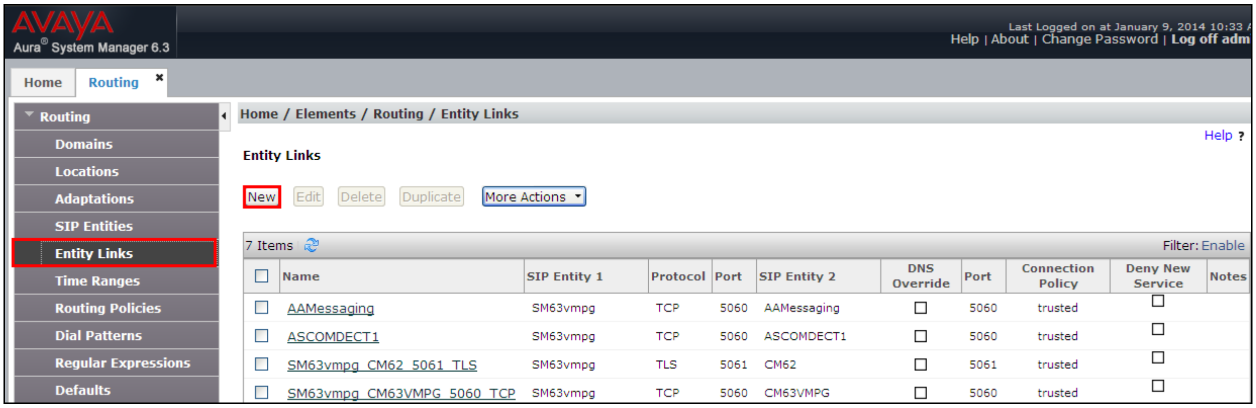


Enter a suitable **Name** as well as the **IP Address** of novaconf. Select **SIP Trunk** as the **Type**. Click on **Commit** once completed.

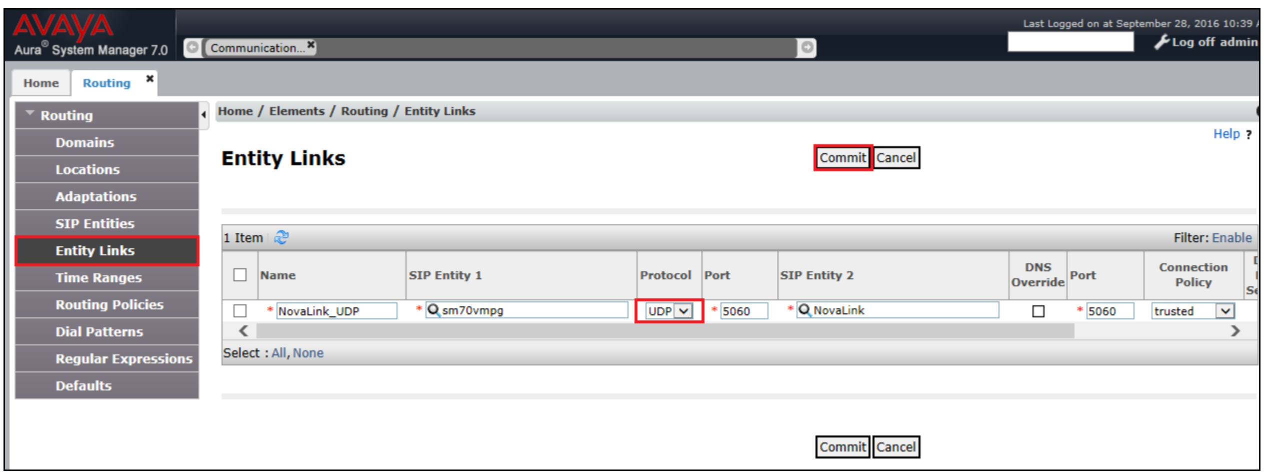
Note: In the remainder of this section including the screen shots below novaconf may also be referred to as novalink.



An Entity Link between novaconf and Session Manager is required, click on Entity Links in the left column and then on **New**.

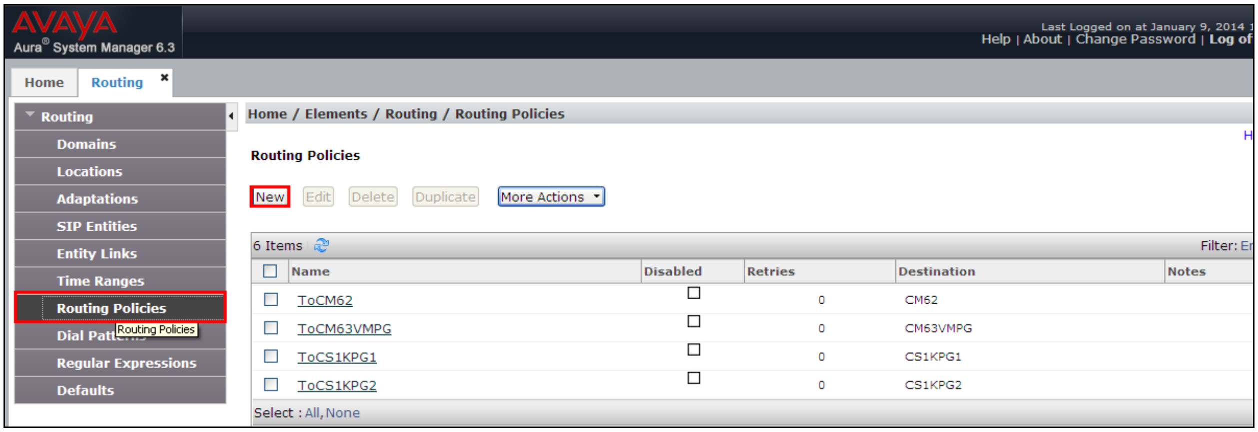


Enter a suitable **Name** and ensure that **UDP** is selected for the **Protocol** and **5060** for the **Port**. The **Connection Policy** must be setup as **trusted** as shown below. Click on **Commit** once completed.

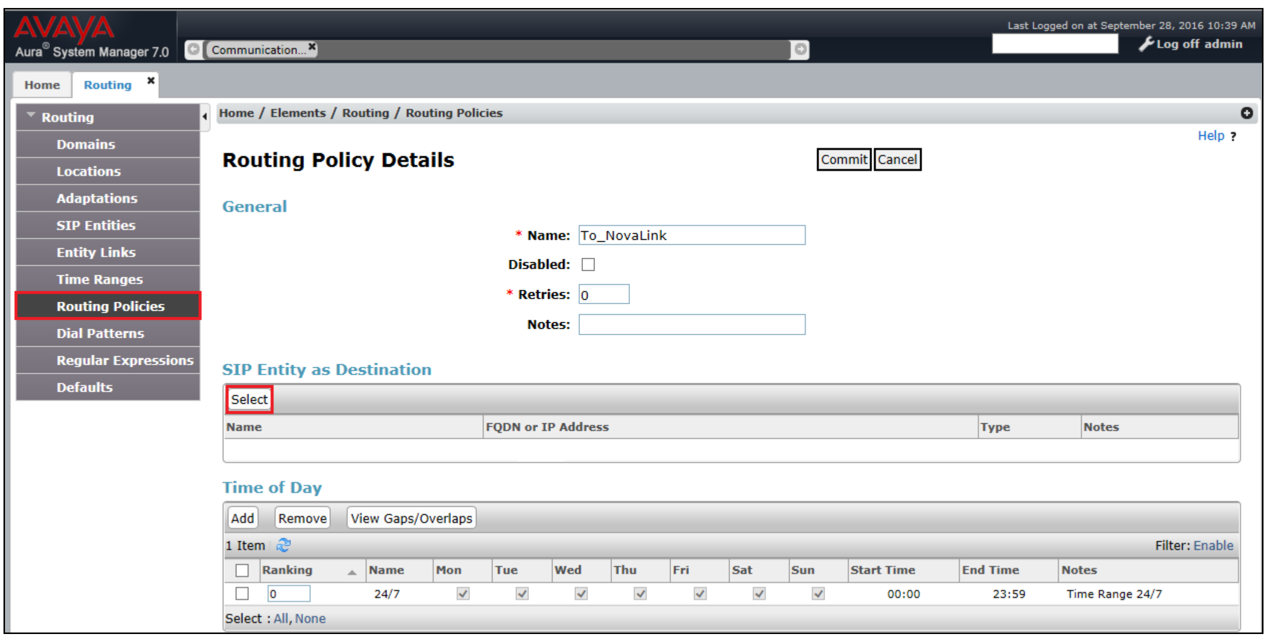


6.4. Configure Routing Policy for novalink

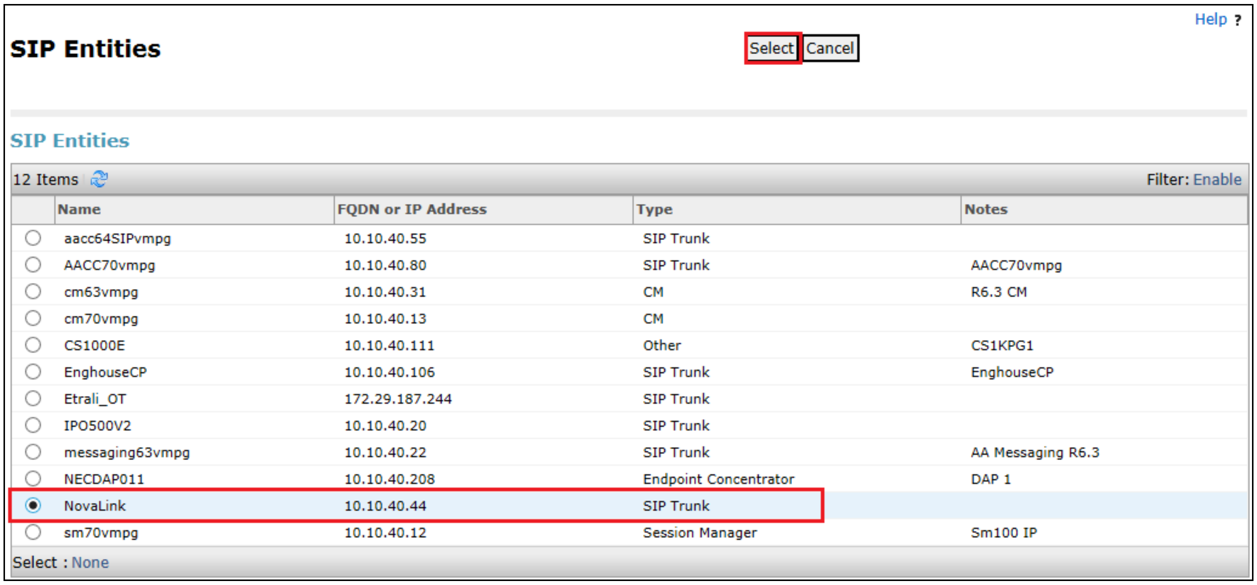
Select **Routing Policies** from the left window and click on **New** in the main window.



Enter a suitable **Name** and click on **Select** highlighted in order to associate this routing policy with a SIP Entity.

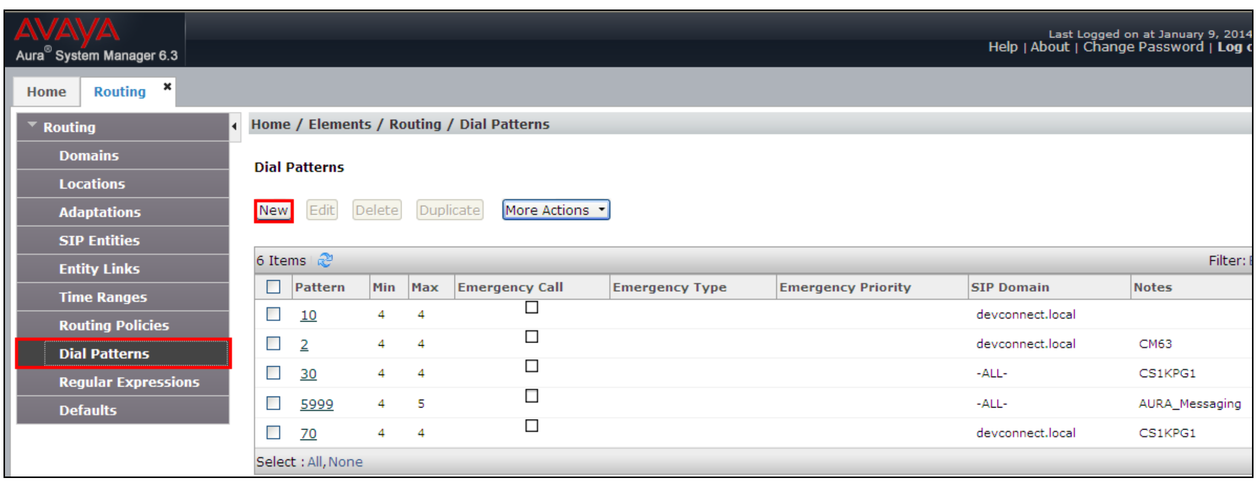


Select the **novalink** SIP Entity created in **Section 6.3** and click on **Commit** when done (not shown).

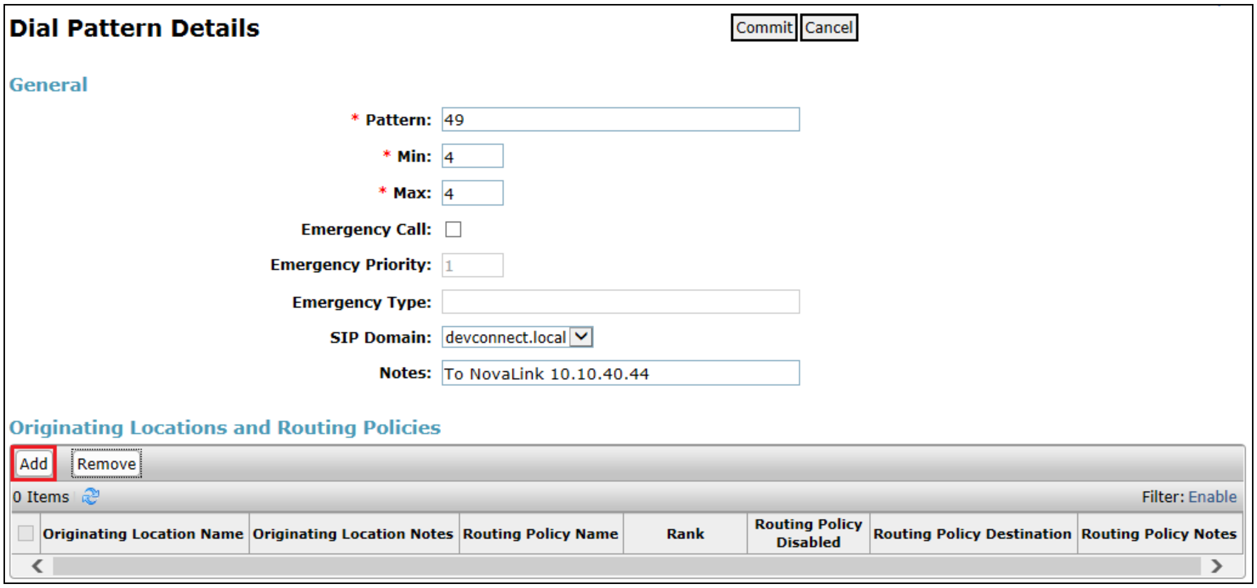


6.5. Configure Dial Pattern for novalink

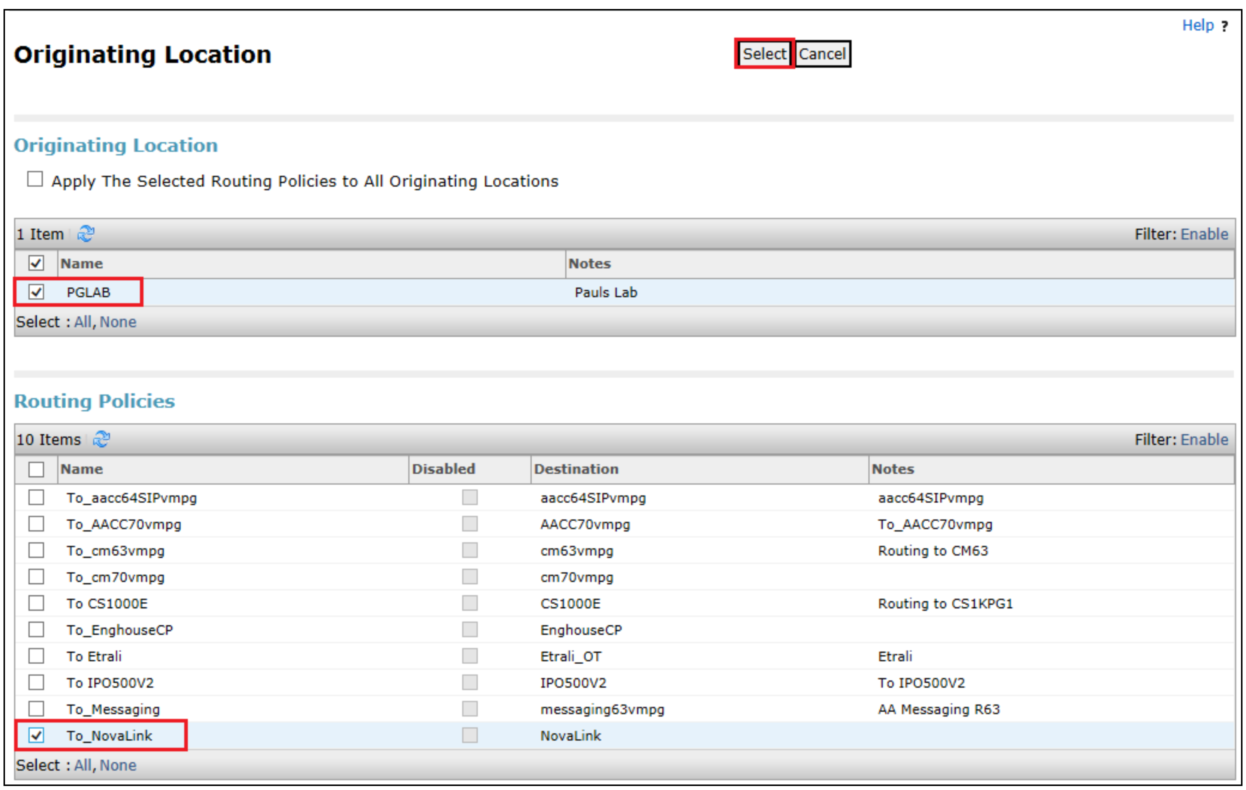
In order to route calls to novaconf a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.



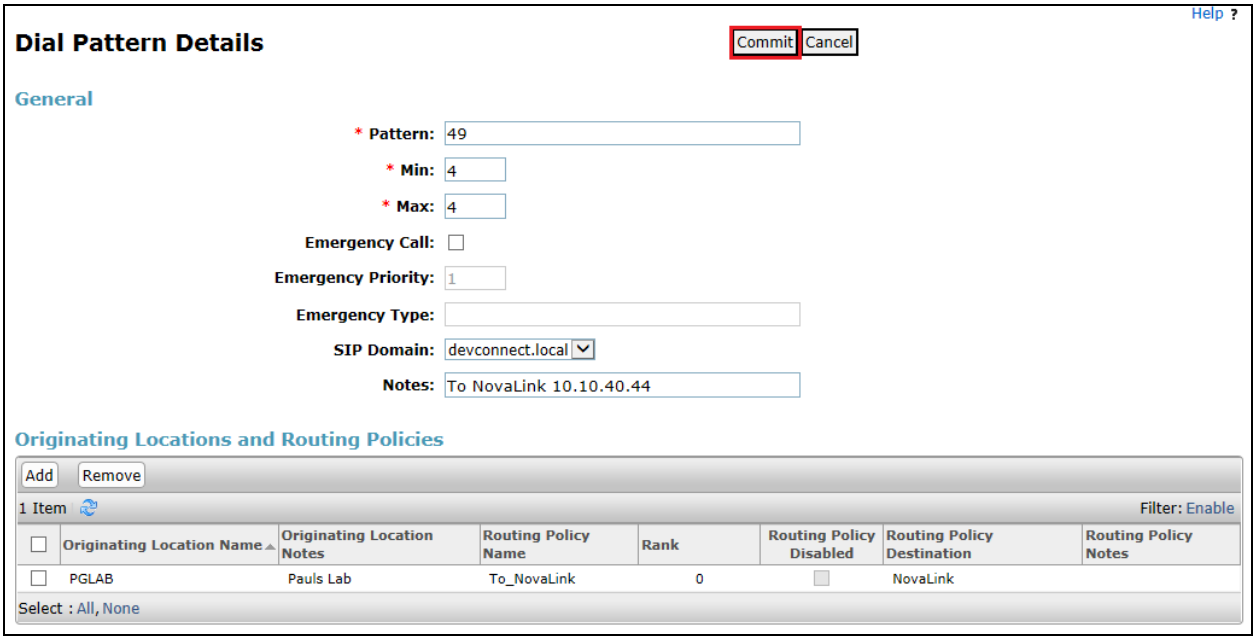
Enter the number to be routed noting this will be the same number outlined in **Section 5.4**. Note the **SIP Domain** is that configured in **Section 6.2**. Click on **Add** to select the SIP Entity.



Tick on the **Originating Location** as shown below and select the **novalink** Routing Policy. Click on **Select** once complete.



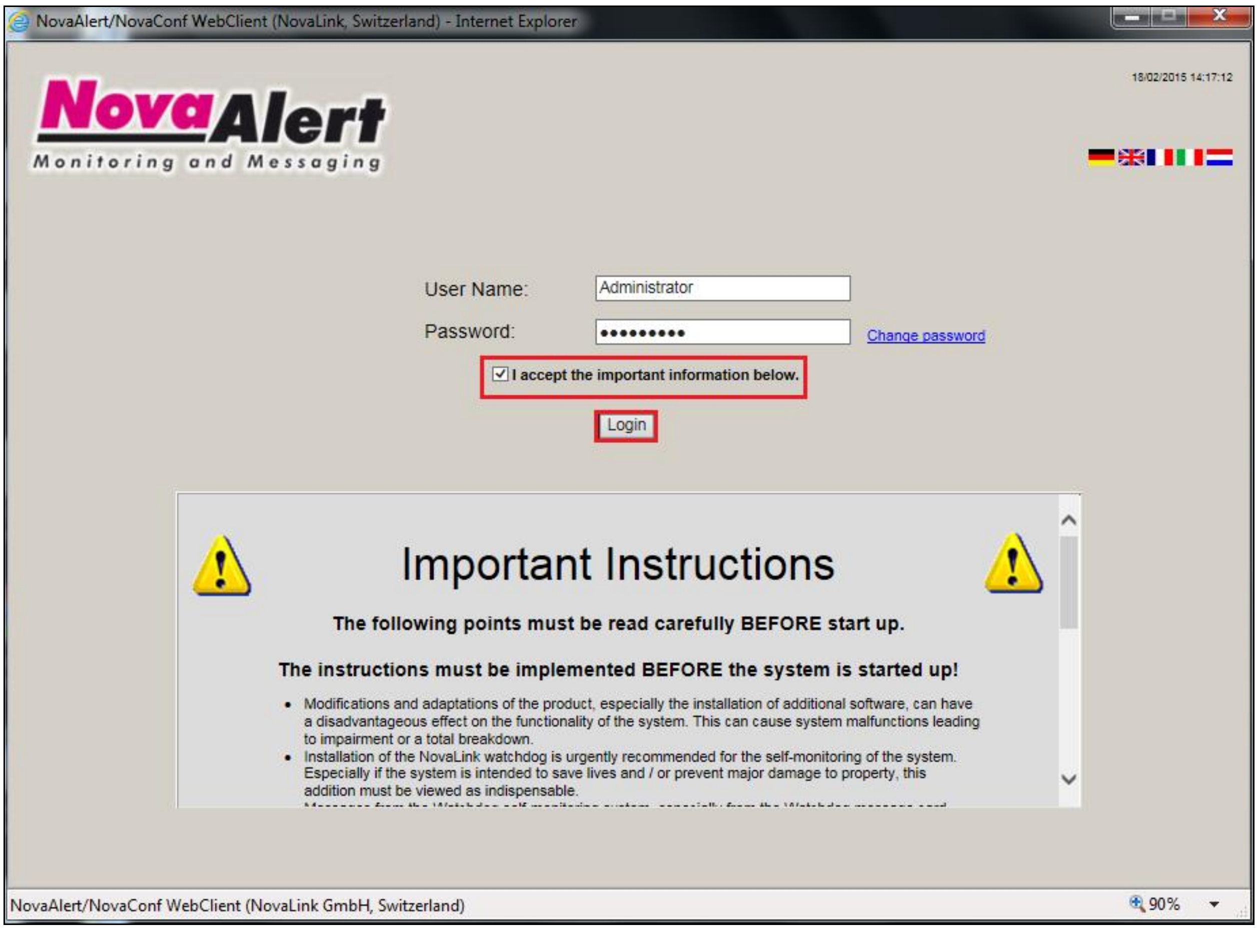
With the new Routing Policy in place, click on **Commit** as shown below.



7. Configure novaconf

The following sections describe the steps required to configure novaconf in order to successfully connect to Session Manager using SIP trunks. All configuration changes are made to novaconf using a web browser session to the novaconf server. Open a web browser session to the IP Address of the novaconf server followed by /novaconf. For example what was used for compliance testing was **http://10.10.40.44/novaconf**. The following screen is shown asking for the **User Name** and **Password**. Enter these and click on the tick box as shown and click on the **Login** button.

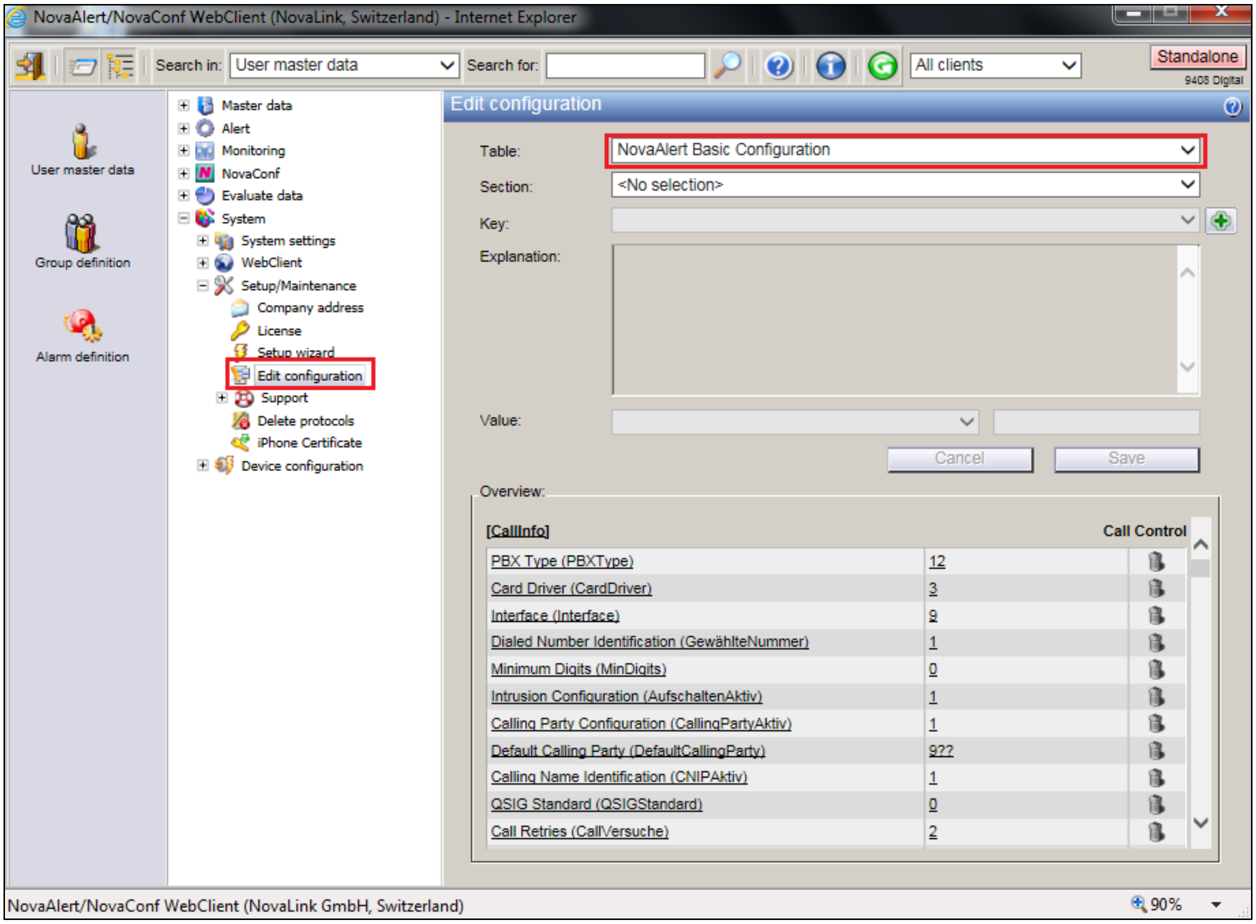
Note: novaconf and novaalert are similar modules from novalink. The following screen shots will show novaalert and this is because novaconf uses novaalert for the connection to Session Manager.



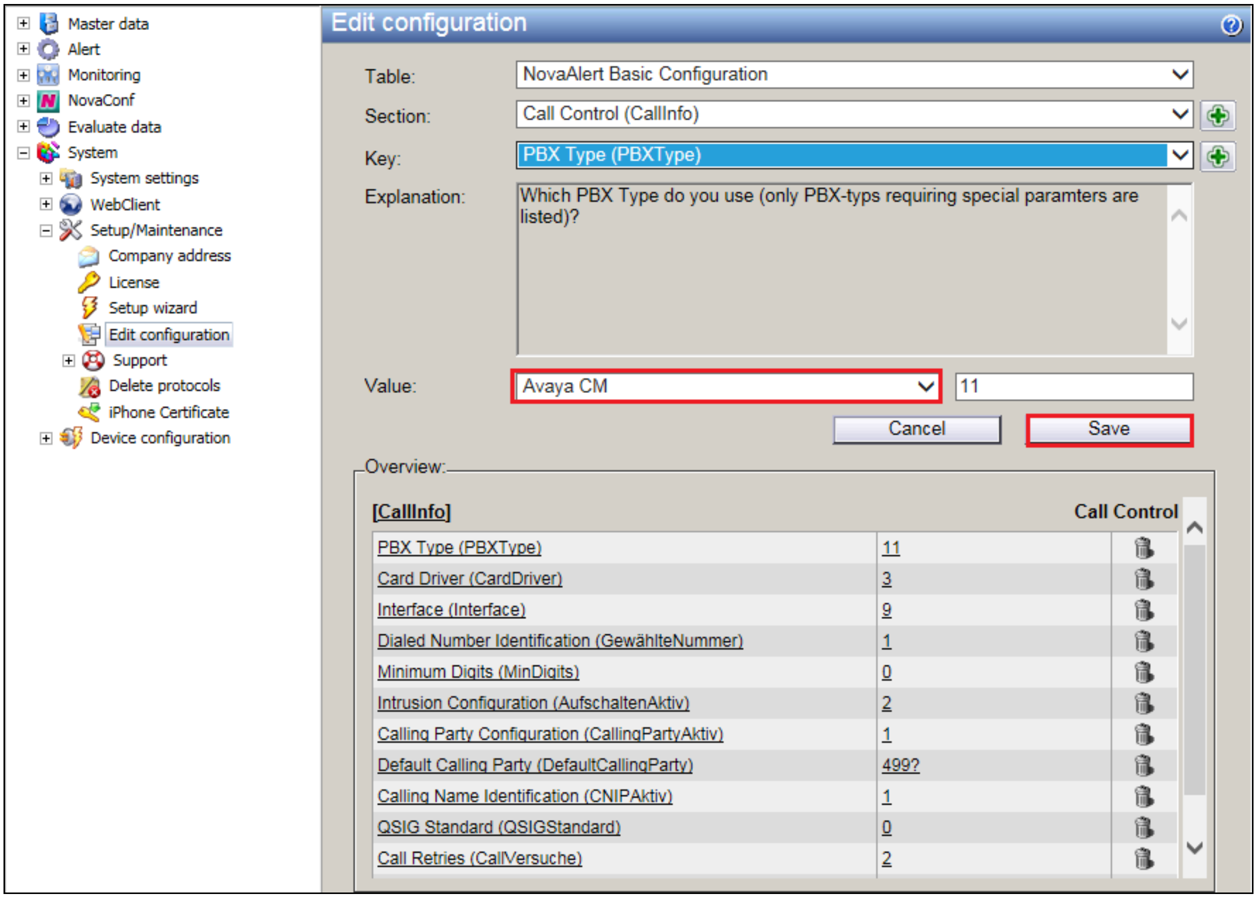
Once logged in the following screen is presented to the user.

7.1. Configure novaconf SIP Trunk Connection

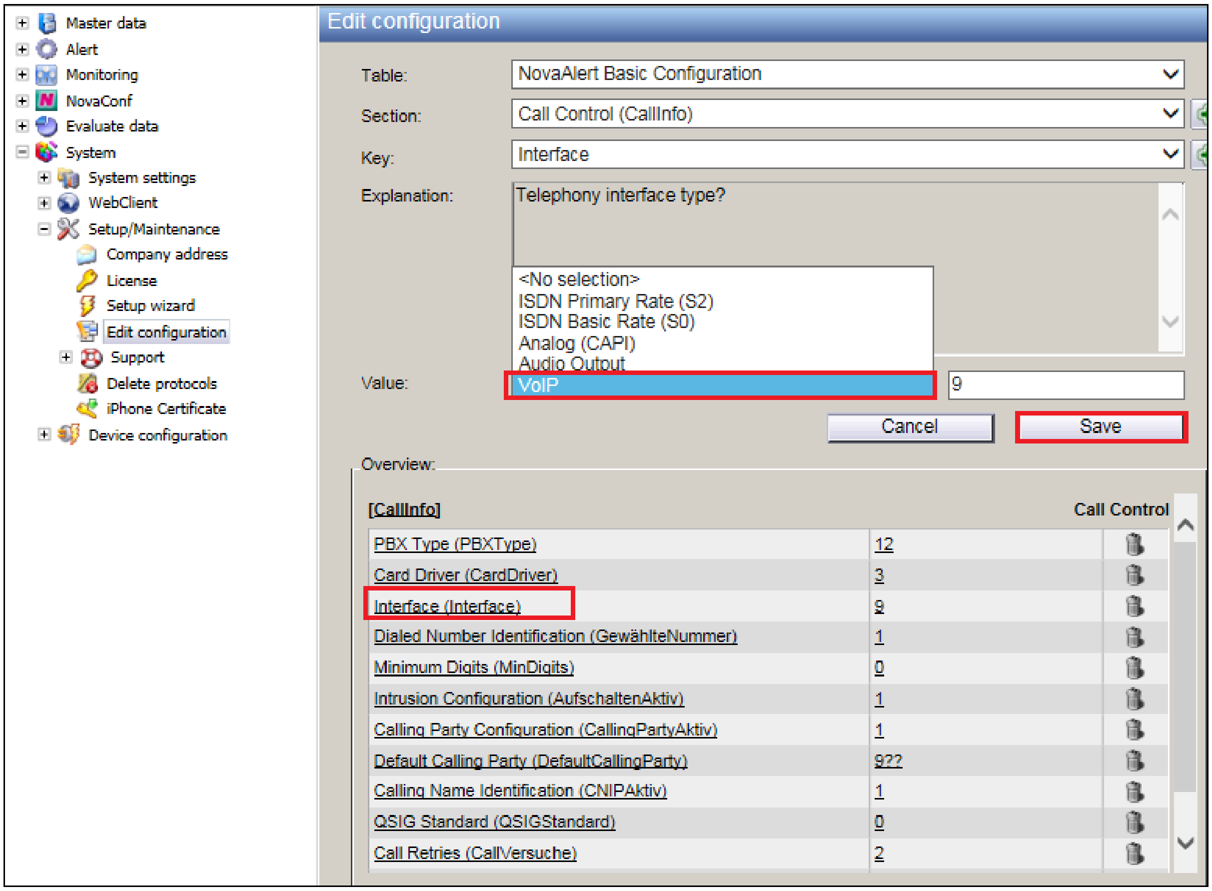
To begin the configuration of novaconf in order to connect to Session Manager using SIP trunks, from the main menu, expand **System → Setup/Maintenance** and click on **Edit configuration**. From the main window select the **Table**, **novaconf Basic Configuration**, from the drop-down menu.



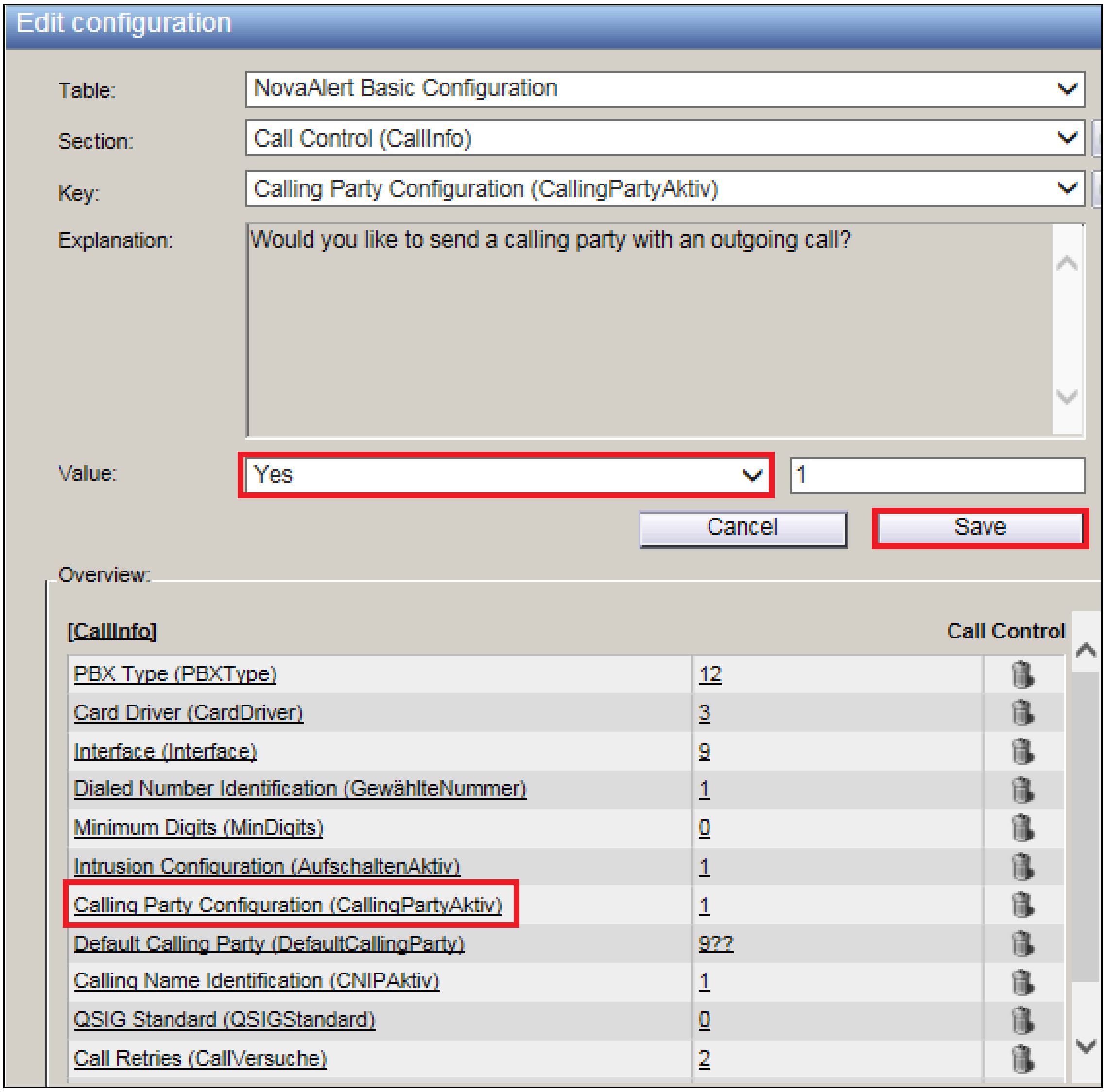
Select **Call Control (CallInfo)** from the **Section** drop-down menu. Select **PBX Type** from the **Key** drop-down menu or click on **PBX Type** highlighted at the bottom of the screen. Ensure that the **Value** is set to **Avaya CM** and click on **Save**.



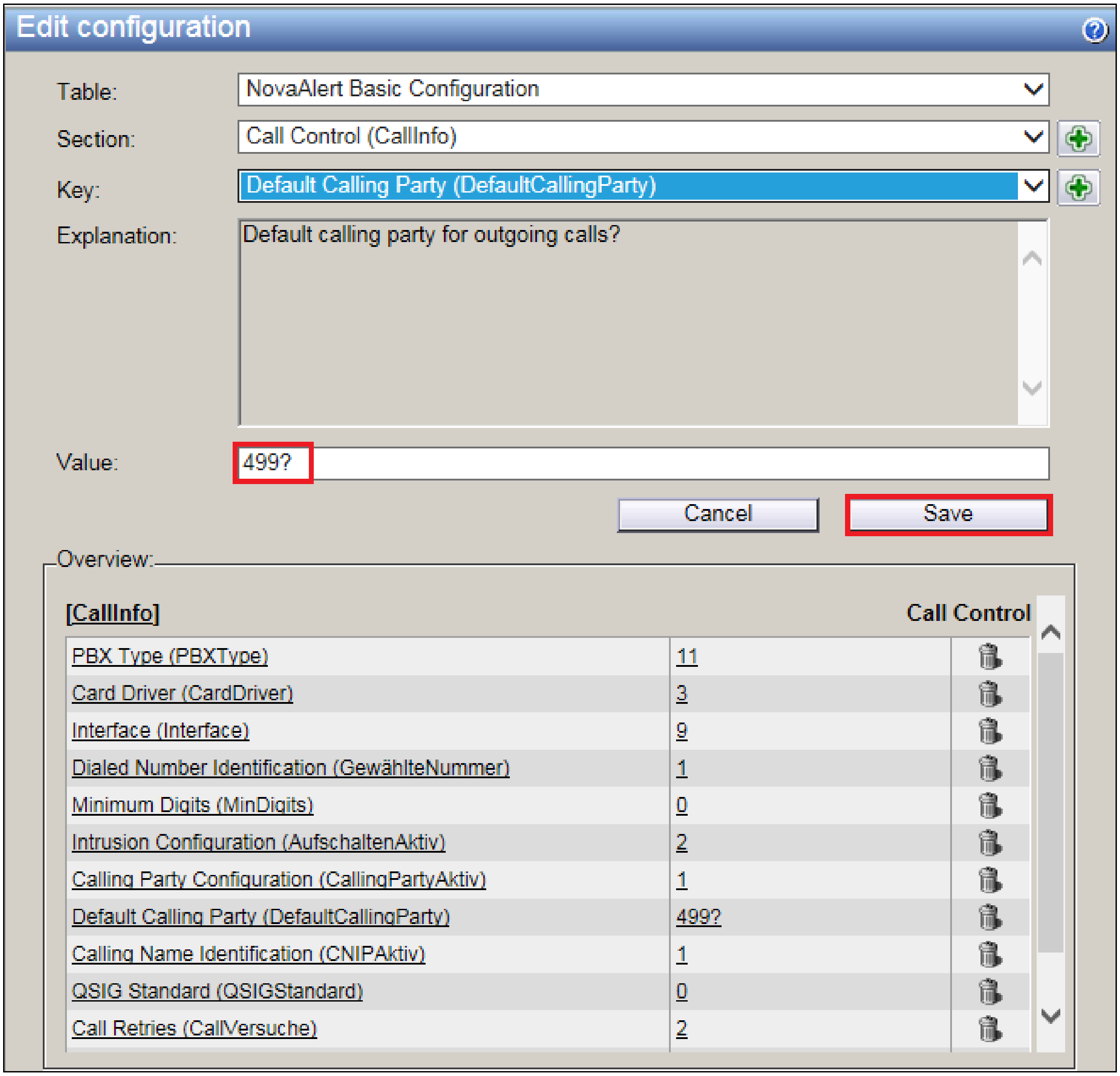
Remaining in the same **Section**, select **Interface** from the **Key** drop-down menu and ensure that the **Value** is set to **VoIP**. Click on **Save** to complete.



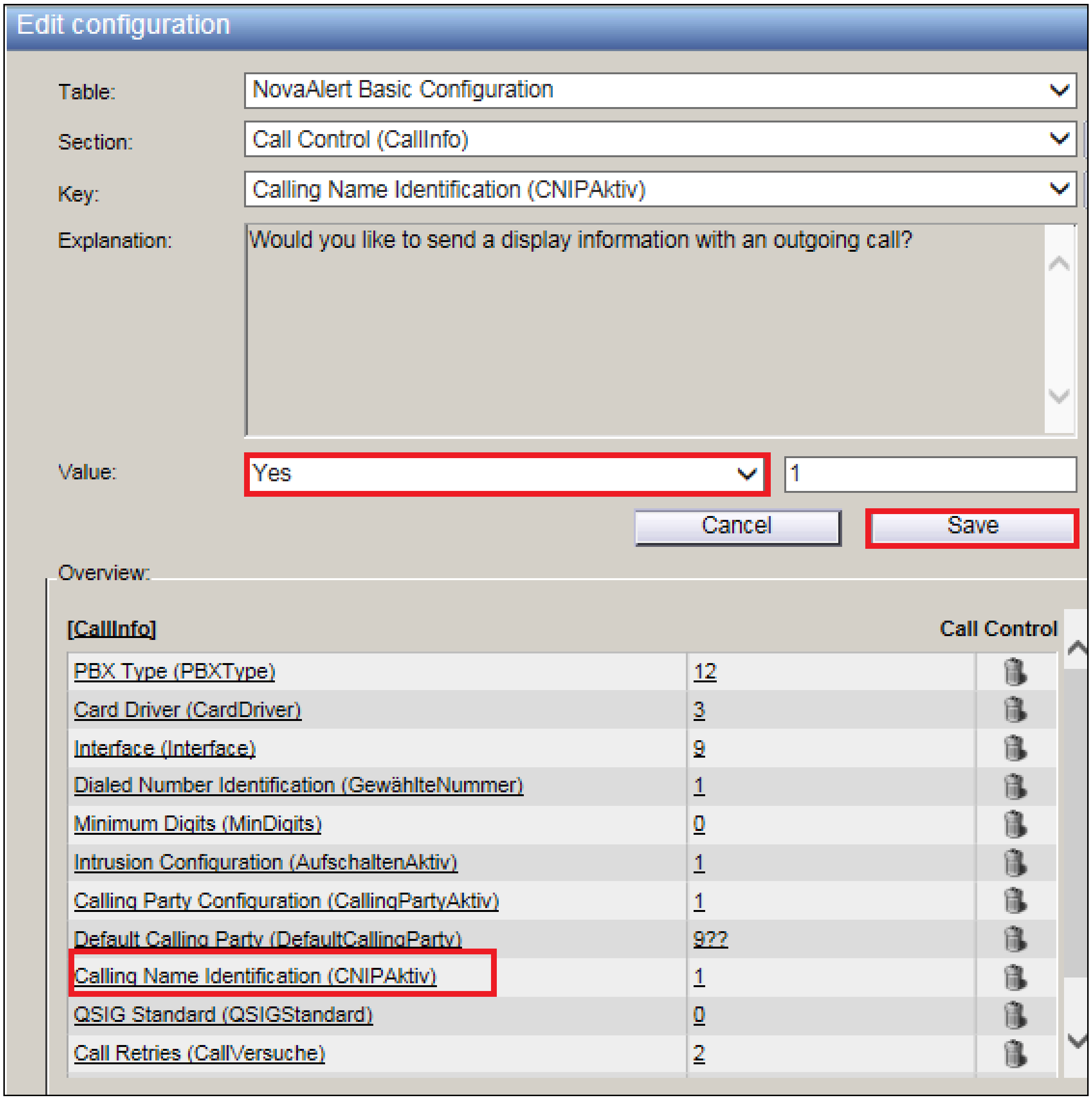
In the same **Section** select the **Calling Party Configuration (CallingPartyAktiv) Key**. Set the **Value** to **Yes** and click on **Save**. This will send the calling party with the outgoing call.



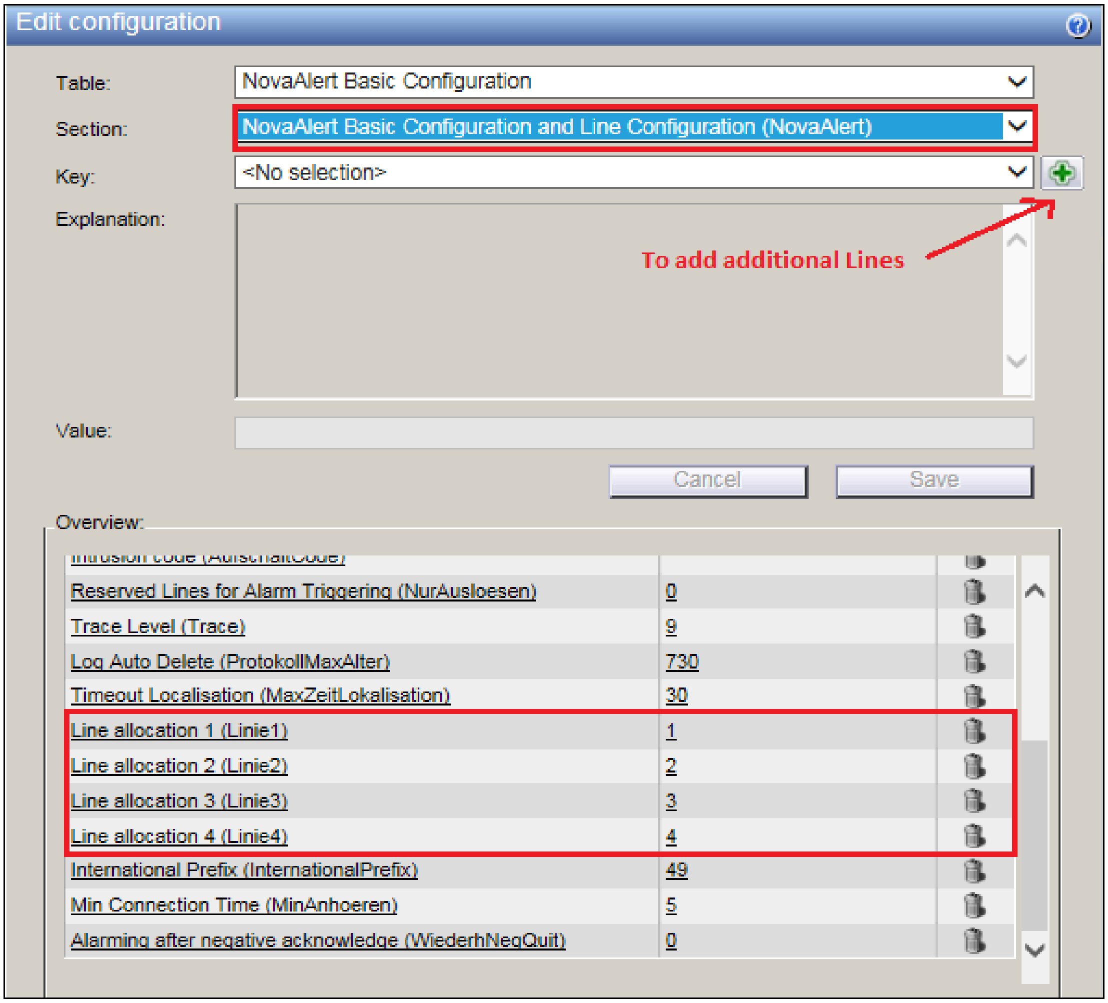
In the same **Section** select the **Default Calling Party (DefaultCallingParty) Key**. Set the **Value** to **499?** and click on **Save.** Note this value will be used for dialing out from Communication Manager.



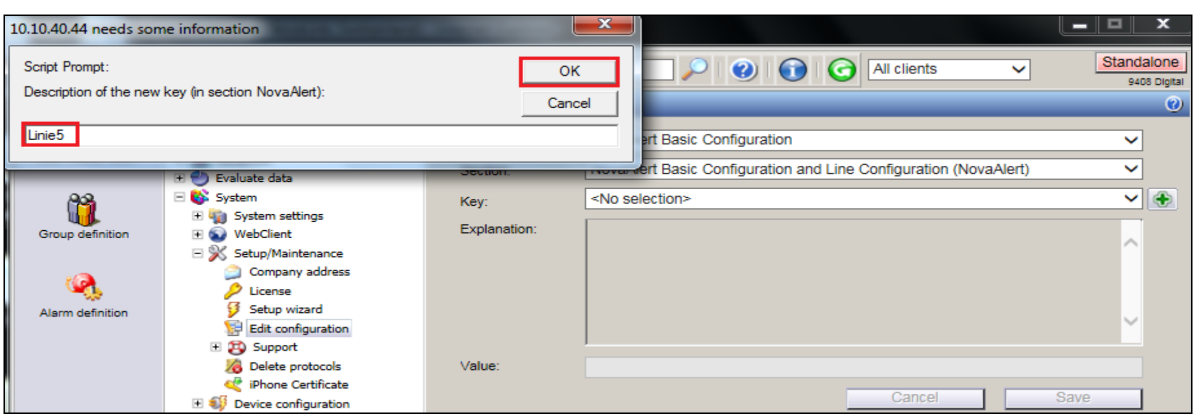
In the same **Section** select the **Calling Name Identification (CNIPAktiv) Key**. Set the **Value** to **Yes** and click on **Save**. This will send the CLID info on the outgoing call.



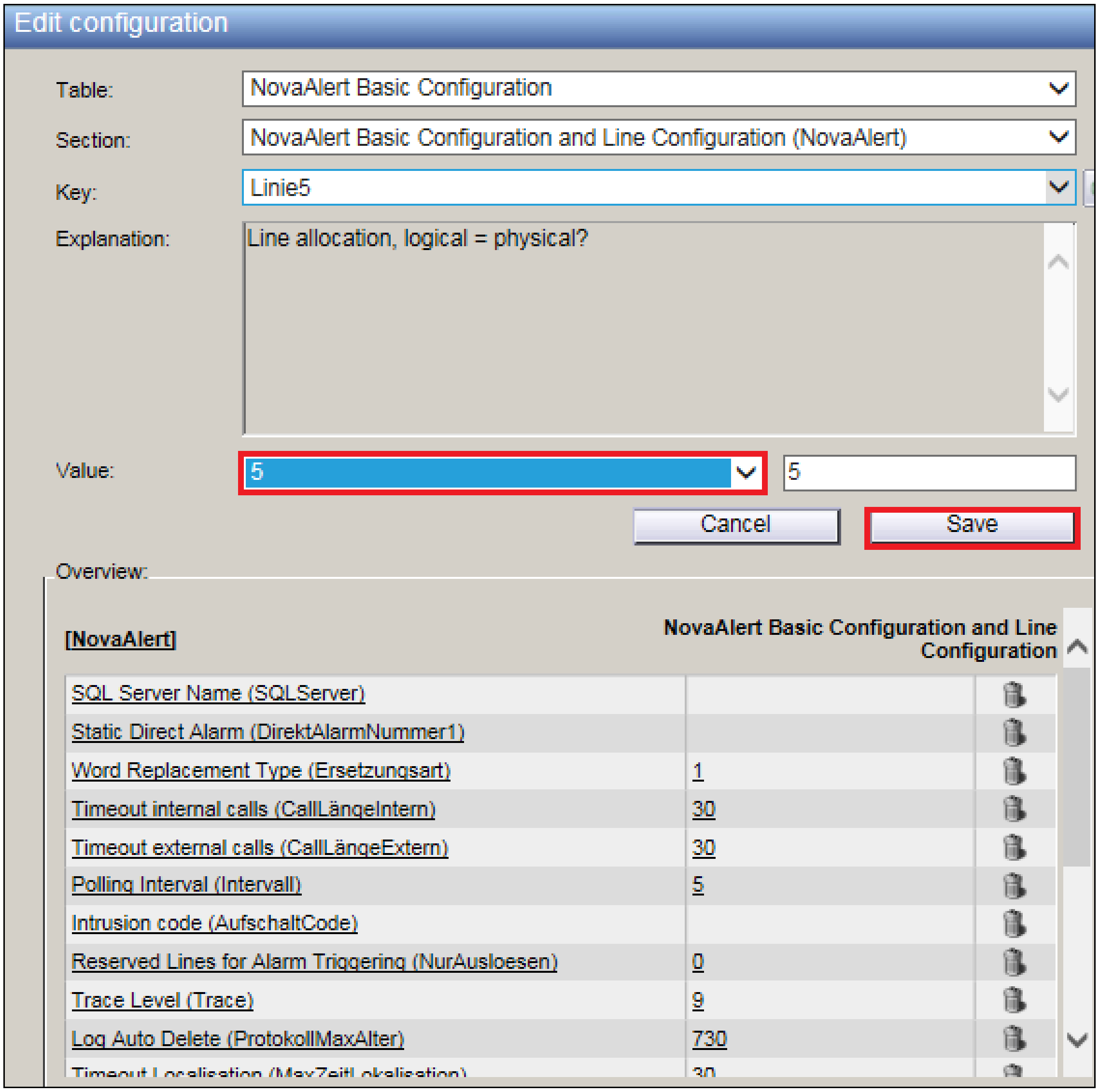
Select **novaalert Basic Configuration and Line Configuration (novaalert)** from the **Section** drop-down menu. In order to add lines to any existing lines shown in the **Overview** window, click on the **+** icon to the right of the **Key** drop down menu, as is shown below.



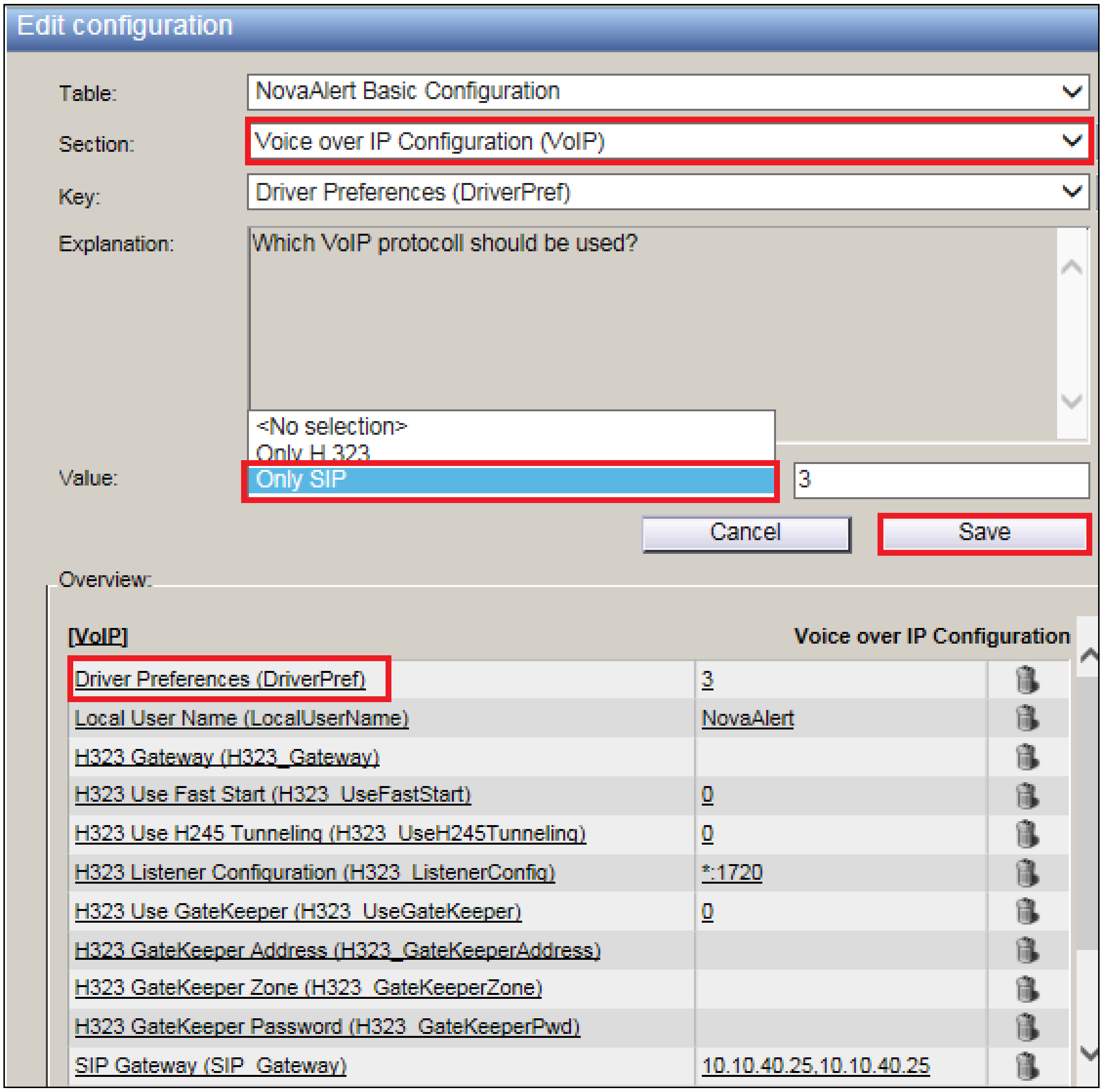
The following window opens, enter **LinieX** into the window and click on **OK**, where X is the next line number to be added.



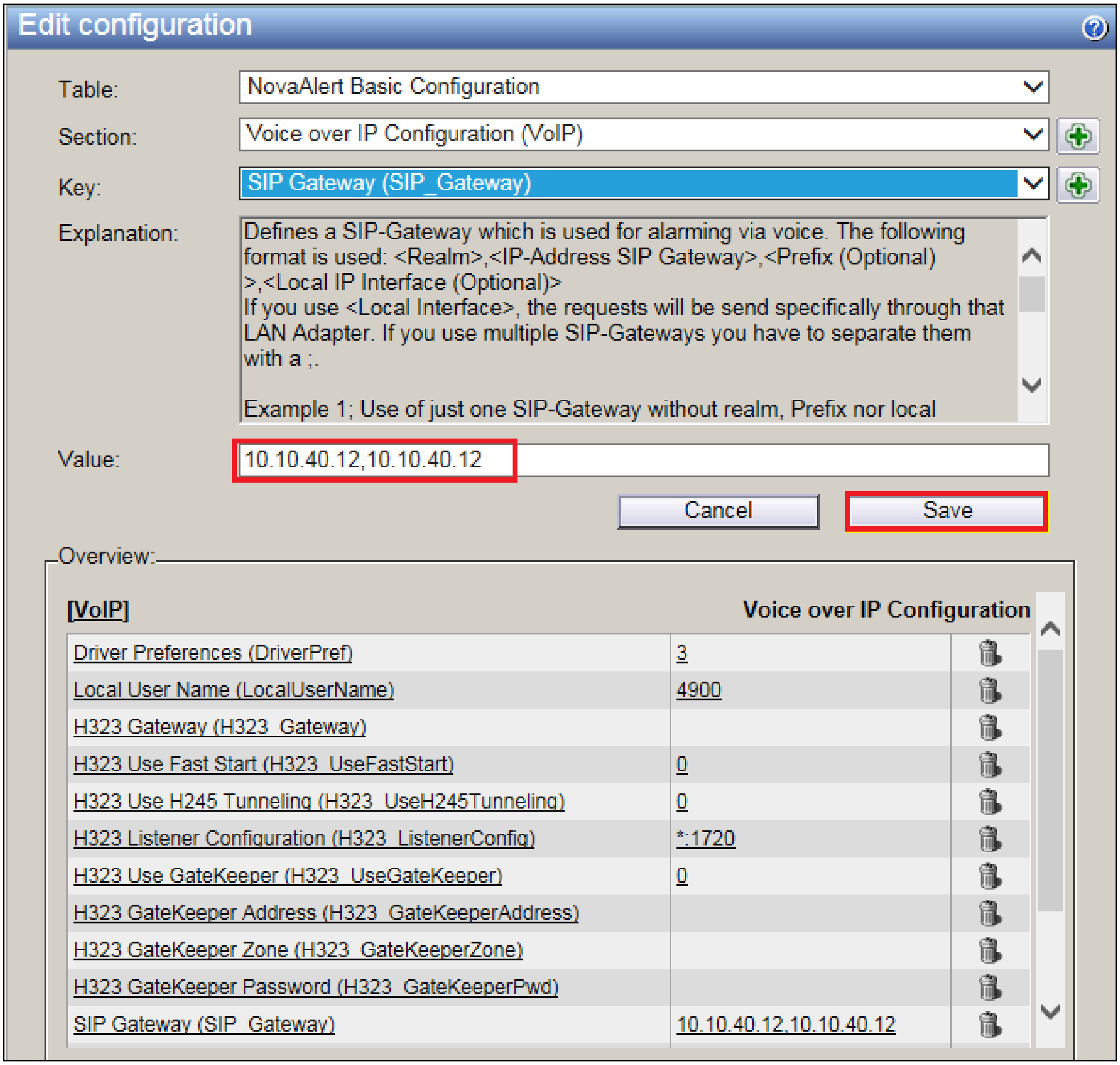
The Key added above, Linie5 should now populate the **Key** menu. Enter the **Value** X where X is the next line number to be added; in this case it is **5**. Click on **Save** to continue.



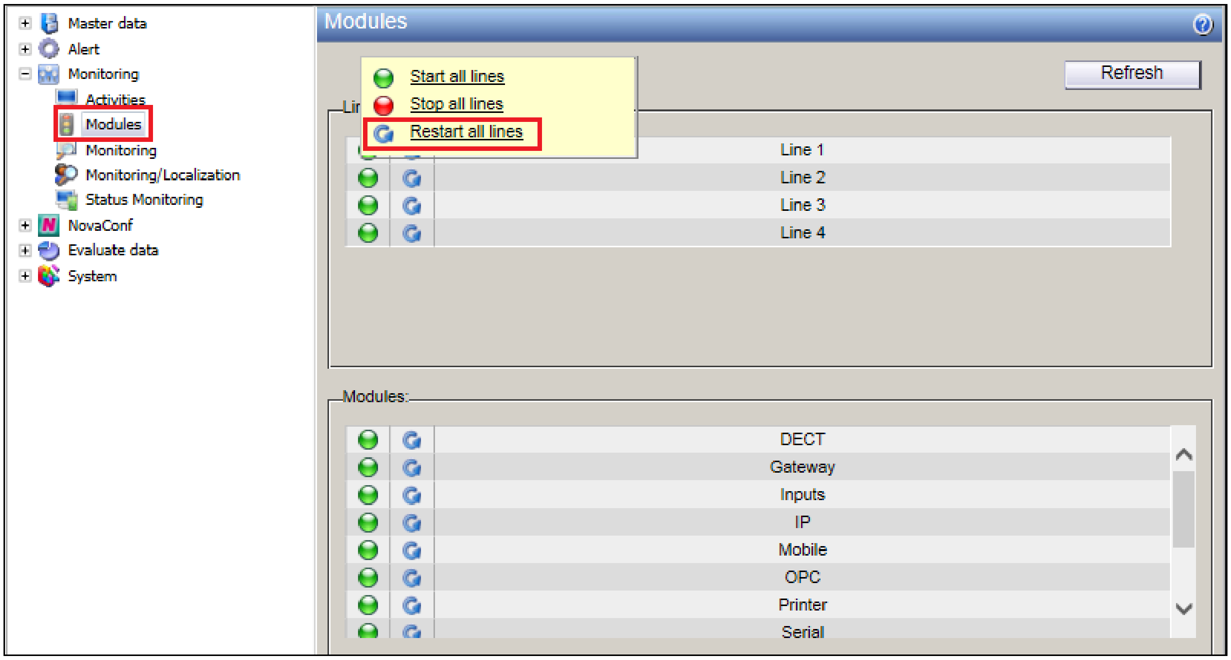
Choose a new section, **Voice over IP Configuration (VoIP)** from the **Section** drop-down menu. Select **Driver Preferences (DriverPref)** from the **Key** drop-down menu. Select **Only SIP** from the drop-down menu for **Value** and click on **Save** to continue.



Staying with the same **Section**, using the drop-down menu change the **Key** to **SIP Gateway (SIP_Gateway)**. Enter the **Value** for the SIP Gateway which will be the IP address of Session Manager. This is entered in the format IP Address, IP Address or **10.10.40.12**, **10.10.40.12** as is shown below. Click on **Save** to continue.

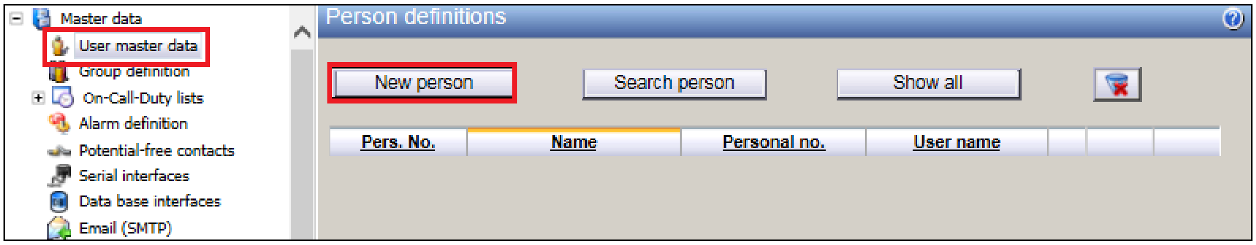


To finish out the configuration a restart of the lines is required. From the menu section navigate to **Monitoring → Modules** and from the main window click on the **refresh icon** beside any of the lines and select **Restart all lines**, as shown below.

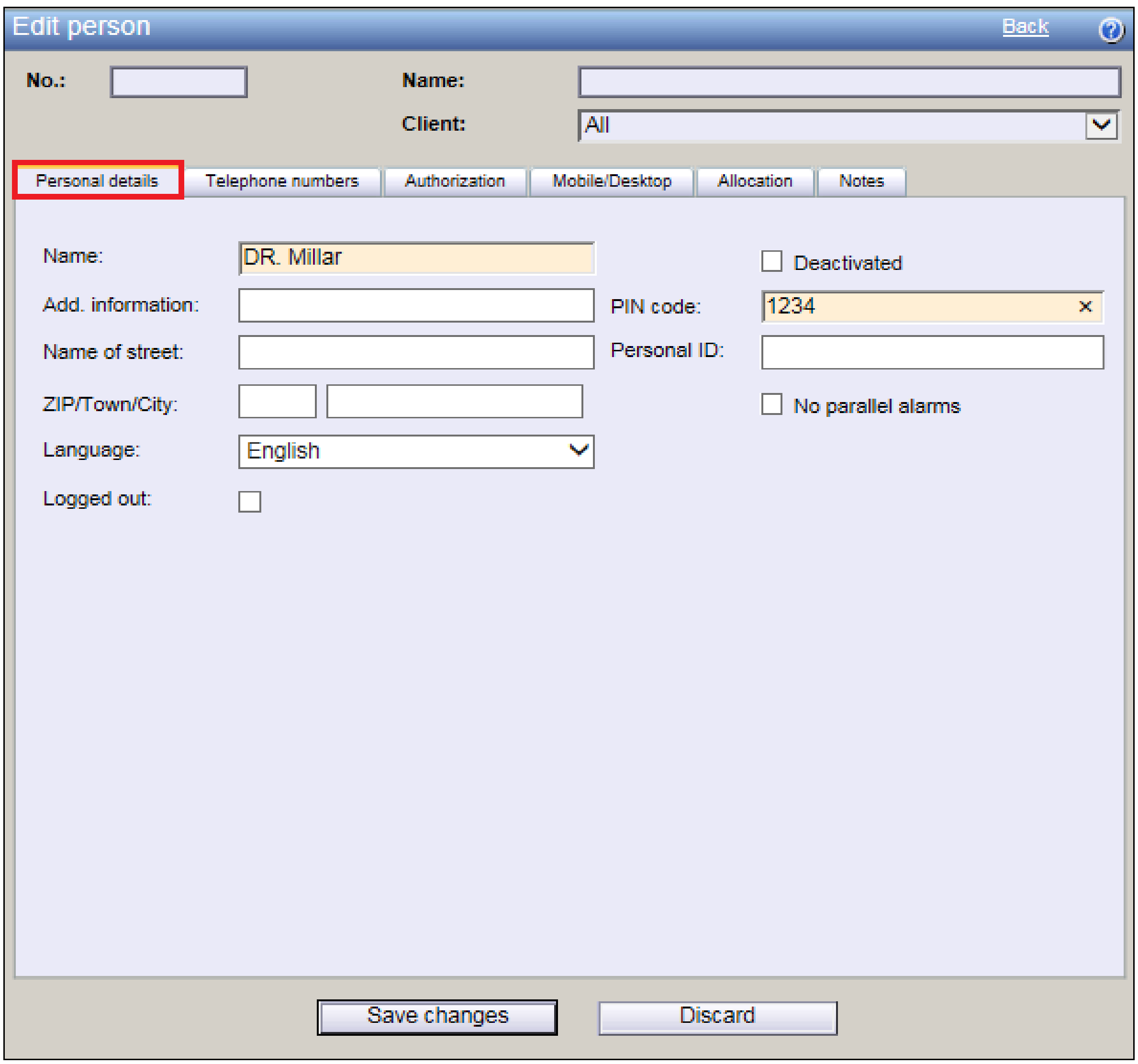


7.2. Add an Avaya Communication Manager extension for Conference

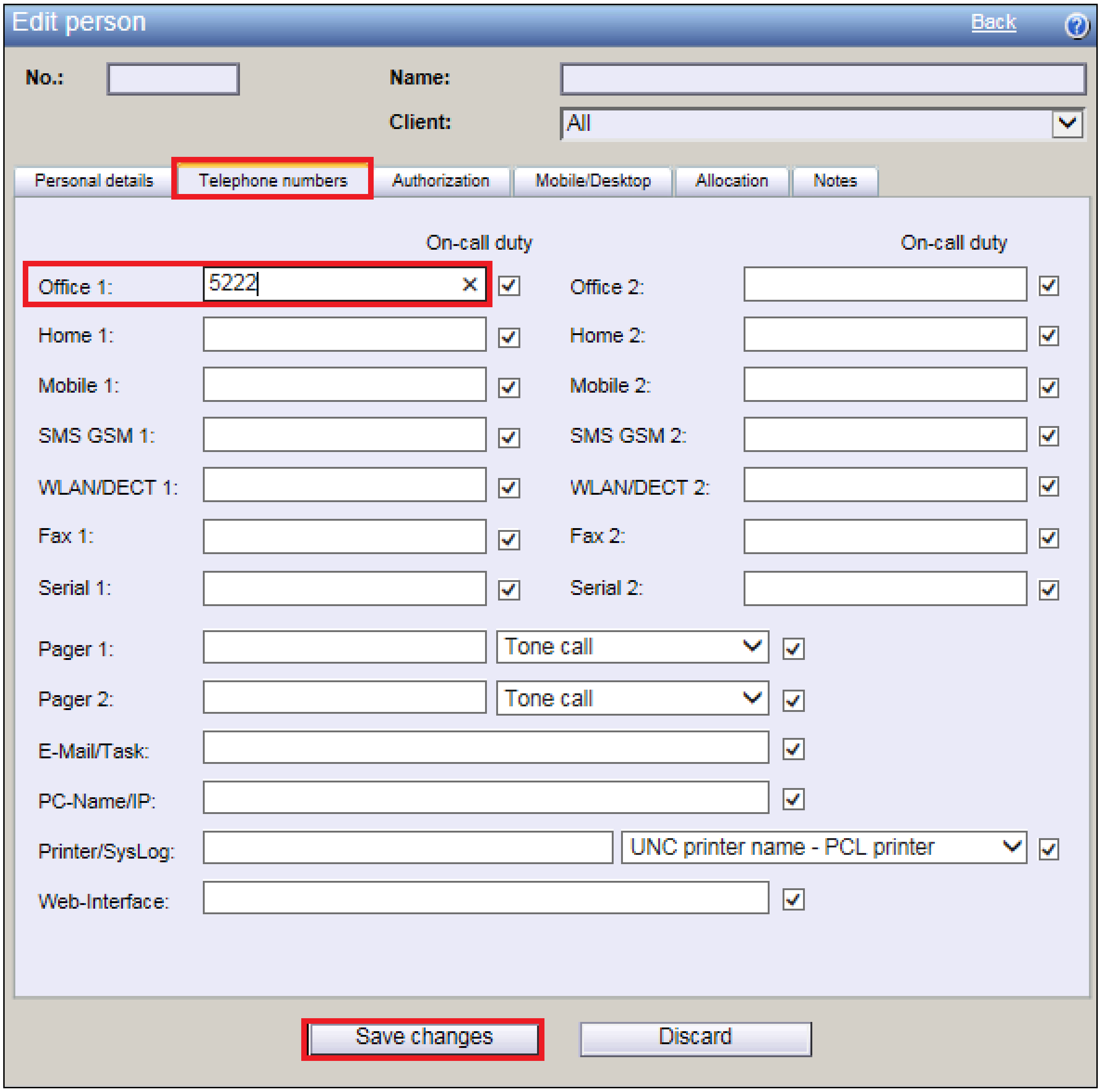
In order setup a conference, Communication Manager extensions need to be added. From the main menu, navigate to **Master data → User master data**. In the main window select **New person** as shown below.



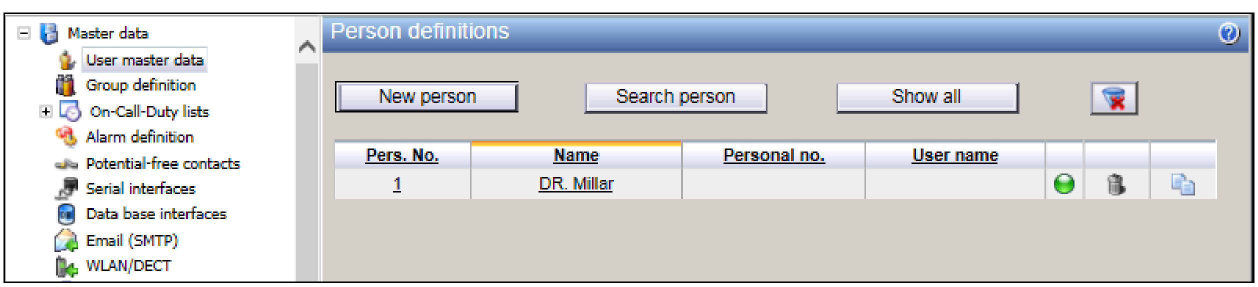
Click on the **Personal details** tab and enter a suitable **Name** and **Pin code**.



Click on the **Telephone numbers** tab and enter the Communication Manager telephone number for this user and click on **Save Changes** at the bottom of the screen.



The new user/extension is now clearly shown.

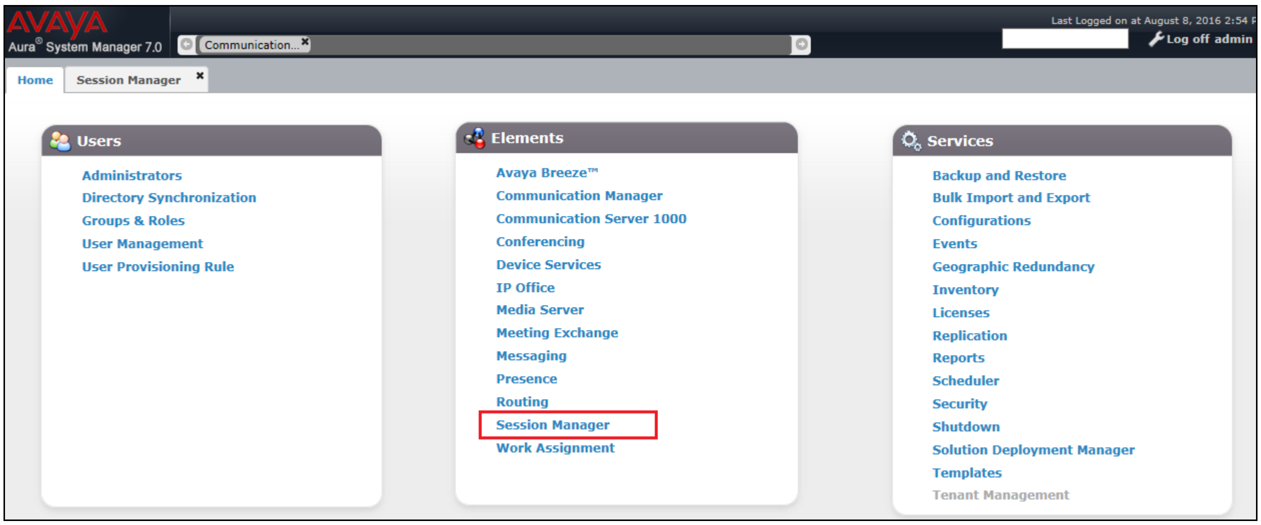


8. Verification Steps

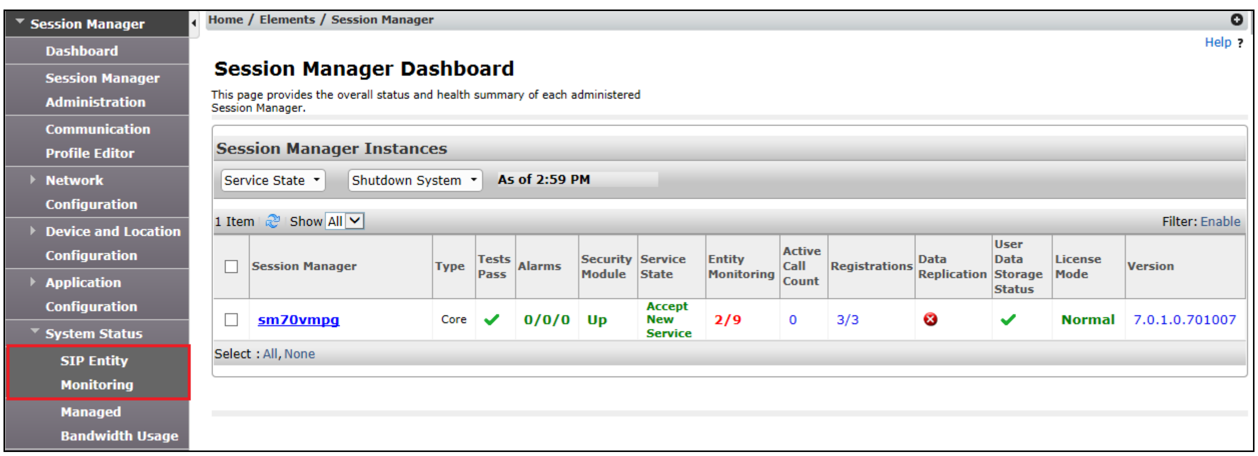
This section illustrates the steps necessary to verify that the novaconf is configured correctly to allow extensions on Communication Manager dial in and use the conference facilities over SIP trunks.

8.1. Verify Link on Session Manager

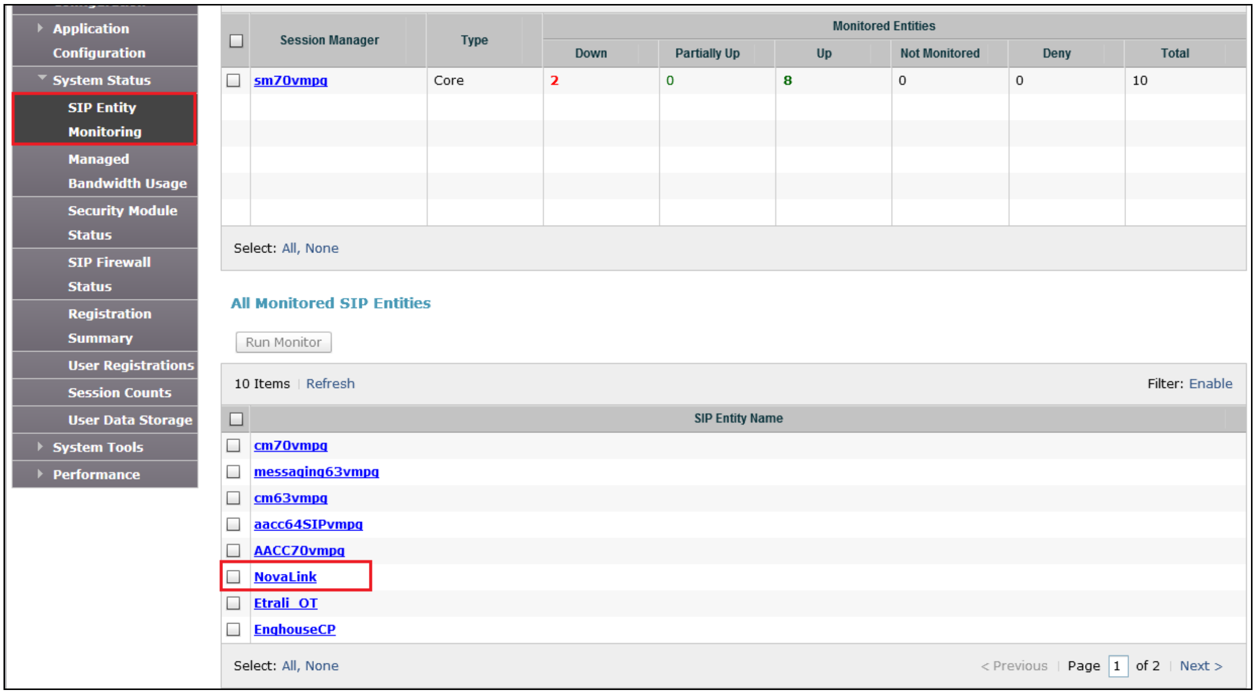
Log in to System Manager as per **Section 6**. From the main menu select Session Manager as shown below.



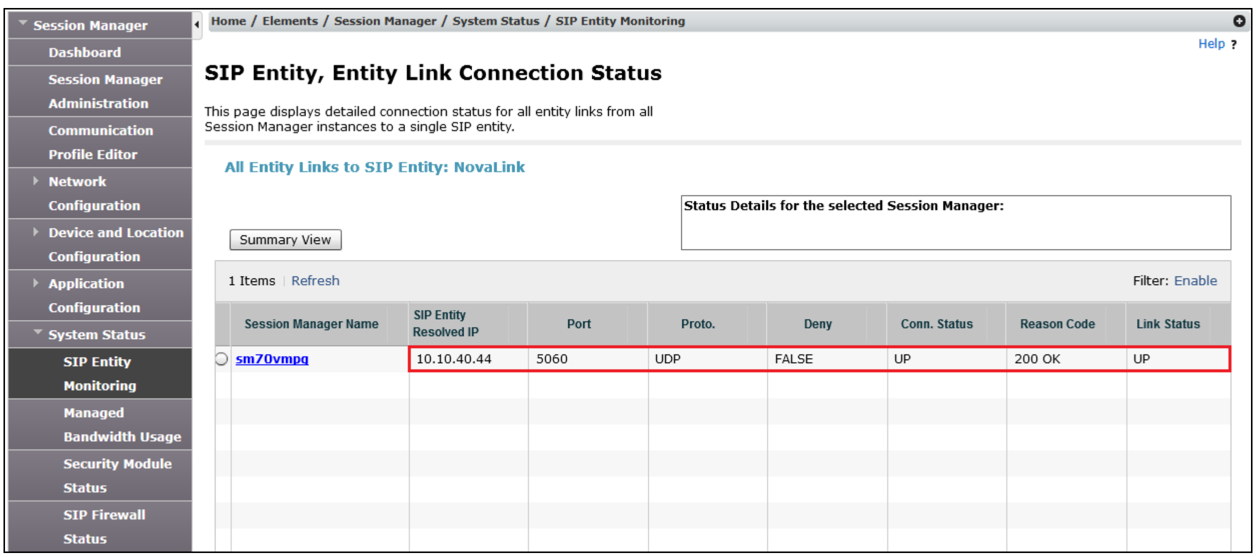
Navigate to **System Status → SIP Entity Monitoring**.



Choose the **novalink** SIP entity as shown below.

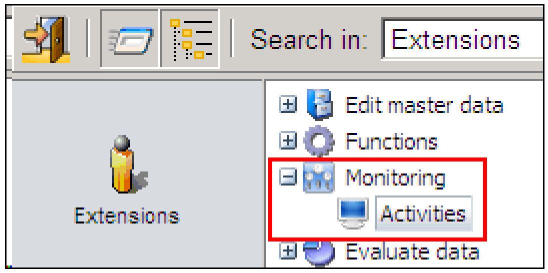


The **Link Status** and **Conn. Status** should both show as **UP** as is shown below.

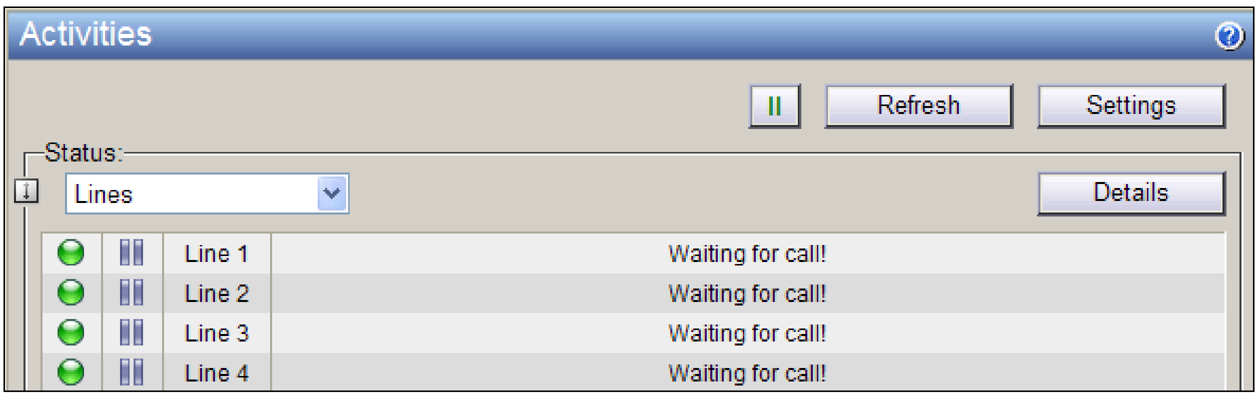


8.2. Verify novalink novaconf on NovaBox Status

From the novaconf web interface (not shown), navigate to **Monitoring** → **Activities**.

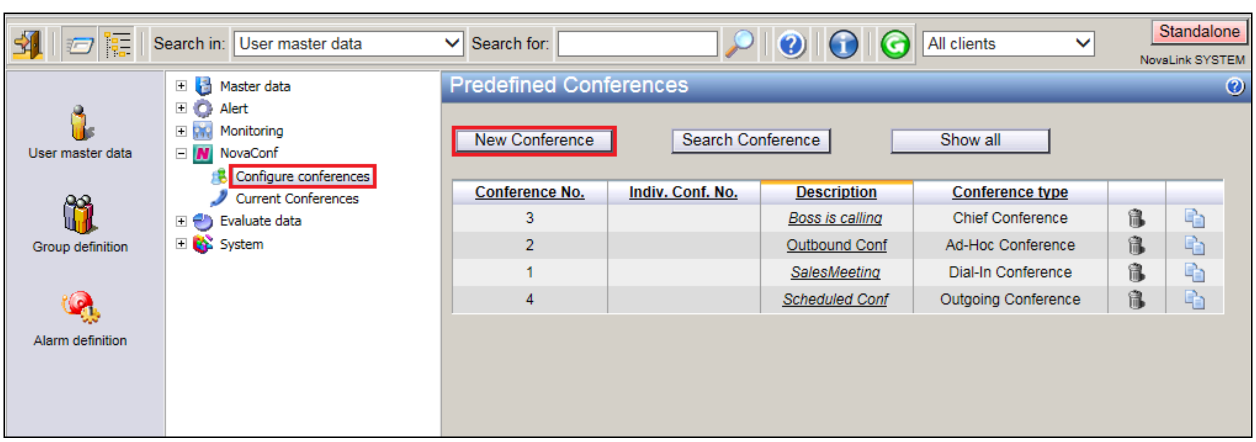


Verify that the icon in the left column is green indicating that the SIP trunks are in service and Session Manager can be reached.

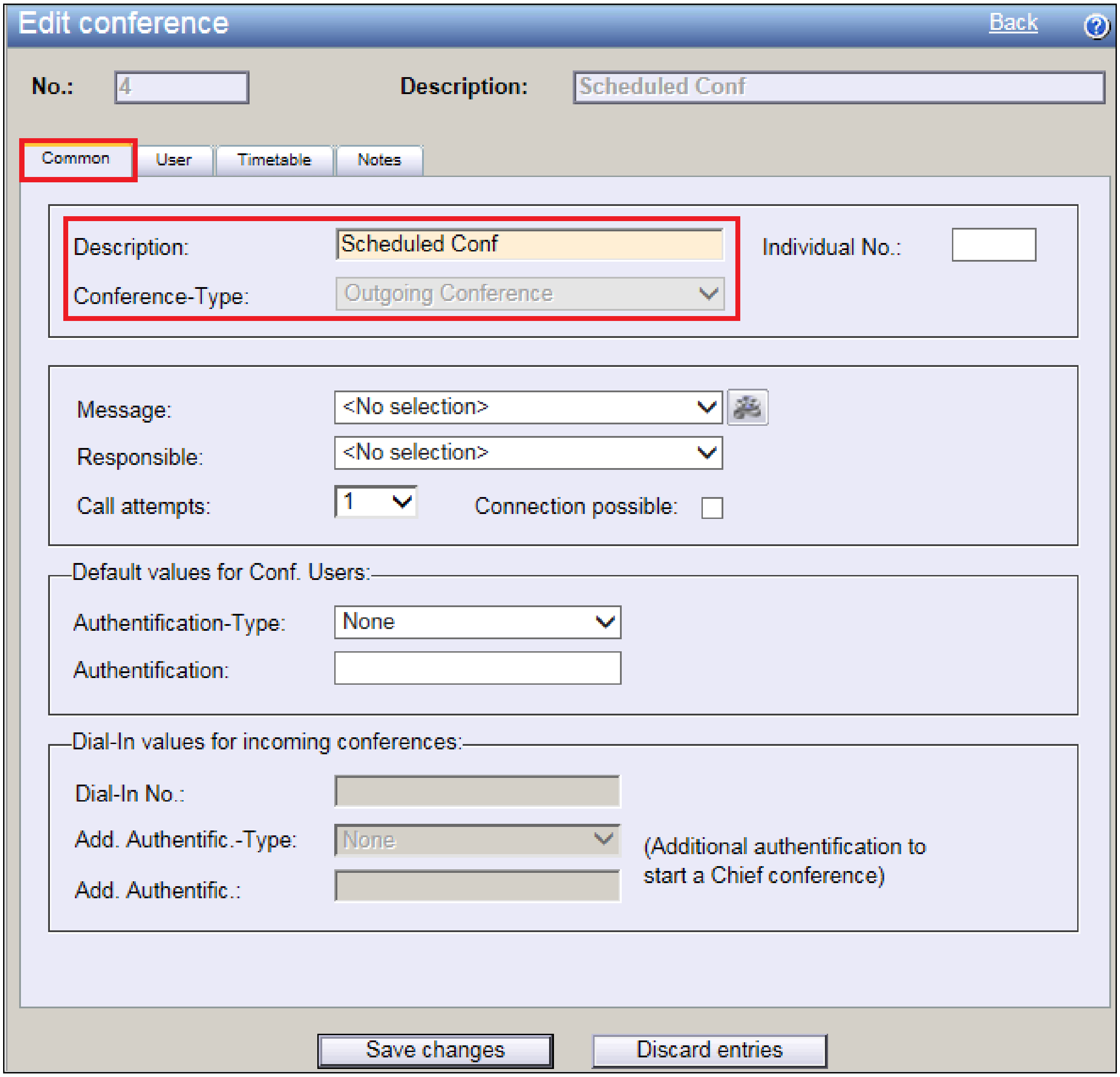


8.3. Create a New Conference on novaconf

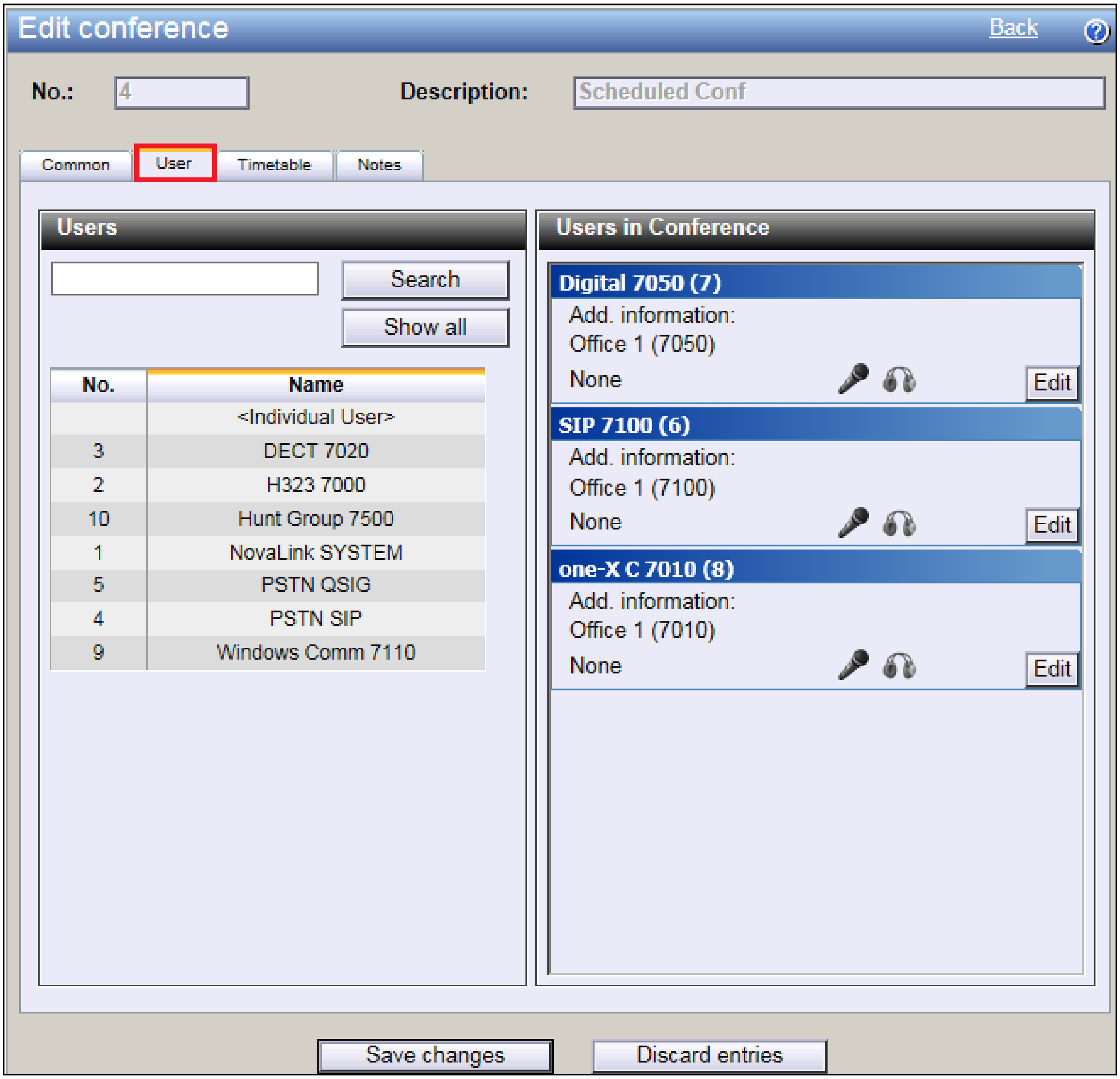
Navigate to **novaconf** → **Configure conferences** in the left window, select **New Conference** from the main window.



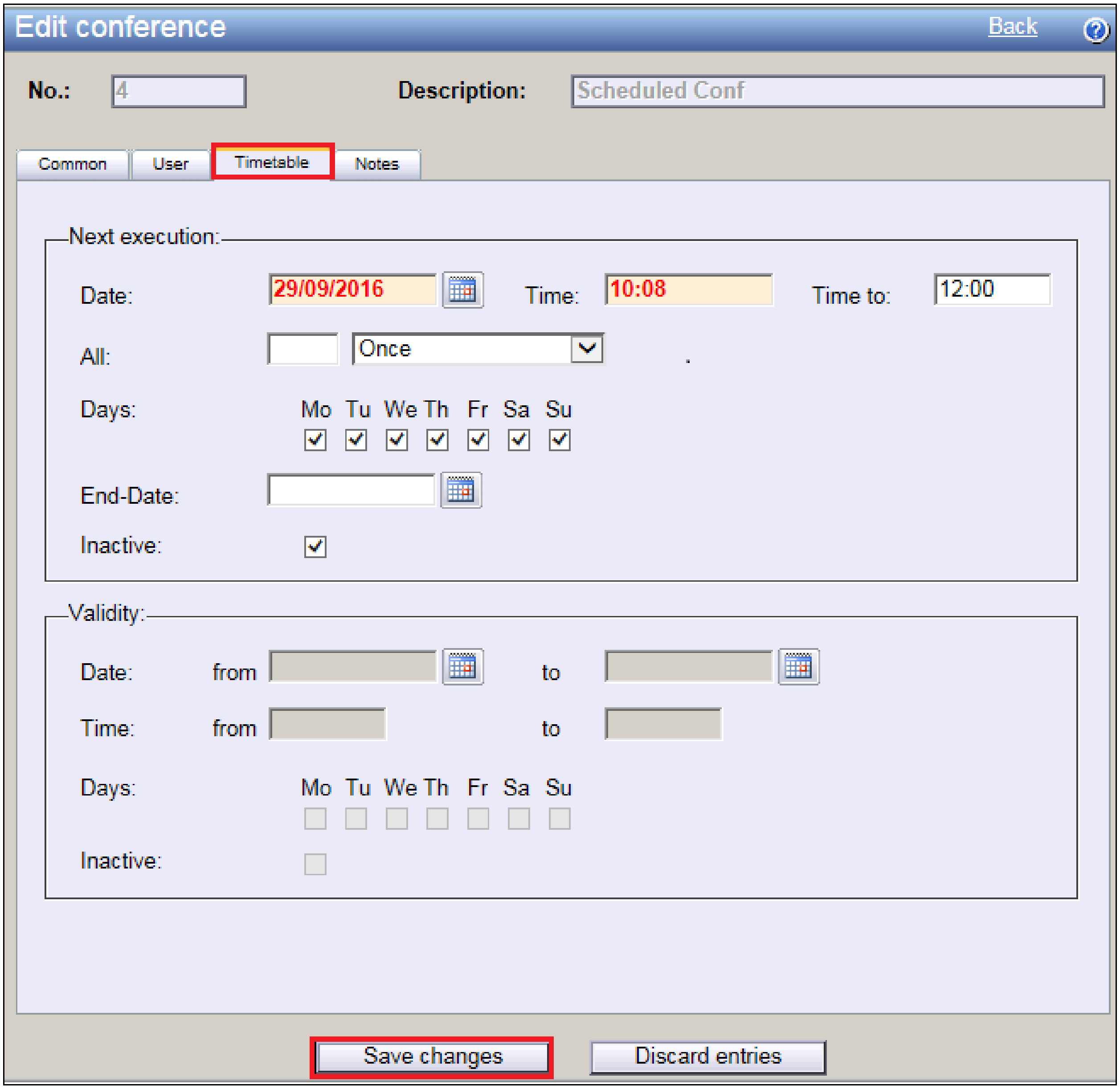
Under the Common Tab, enter a suitable **Description** for the conference and choose **Outgoing Conference** as the **Conference-Type**.



Under the **User** tab, click and drag the required users from the left column into the right column.



Under the **Timetable** tab, enter the time for the conference to start and click on **Save Changes** at the bottom of the screen. This should setup the conference to call out to the users on the previous page at **10:08**.



9. Conclusion

These Application Notes describe the configuration steps required for novaconf from novalink to successfully interoperate with Avaya Aura® Communication Manager. All feature test cases were completed successfully with any observations noted in **Section 2.2**.

10. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com where the following documents can be obtained.

[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] *Implementing Avaya Aura® Session Manager* Document ID 03-603473

[4] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324

Technical support can be obtained for novaconf from the website http://www.novalink.ch/en/ or from ftp://support.novalink.ch/Technikerhandbuch/English/Technikerhandbuch novalink GmbH EN.chm (please request Login and Password from novalink).

Appendix

Configure SIP Trunk between Session Manager and Communication Manager

The following shows the SIP Signalling Group and SIP trunk that was used during compliance testing.

- Set the **Group Type** field to **sip**.
- For compliance testing **Transport Method** was set to **tls**.
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively.
- Set the **Near-end Node Name** to **procr**. Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **sm70vmpg**), as per **Section 5.5**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5.** This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- **Far-end Domain** was set to the domain used during compliance testing.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** was set to **N** for compliance testing.
- The default values for the other fields may be used.

```
change signaling-group 1                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
   IMS Enabled? n        Transport Method: tls
         Q-SIP? n
      IP Video? n                                  Enforce SIPS URI for SRTP? n
  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: sm70vmpg
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain: devconnect.local
                                             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? y
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
```

Configure the Trunk Group form as shown below. This trunk group is used for calls to and from novaconf. Enter a descriptive name in the Group Name field. Set the Group Type field to sip. Enter a TAC code compatible with the Communication Manager dial plan. Set the Service Type field to tie. Specify the signaling group associated with this trunk group in the Signaling Group field, and specify the Number of Members supported by this SIP trunk group. Accept the default values for the remaining fields.

```
change trunk-group 1                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: r
  Group Name: SIPTRK                     COR: 1        TN: 1        TAC: *801
   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: 1
                                                  Number of Members: 10
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with NEC to prevent unnecessary SIP messages during call setup. Session refresh is used throughout the duration of the call, to check the other side has not gone away, for the compliance test a value of **600** was used.

```
change trunk-group 1                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                           Redirect On OPTIM Failure: 5000

            SCCAN? n                              Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

Settings on **Page 3** can be left as default. However the **Numbering Format** in the example below is set to **private**.

```
change trunk-group 1                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



   Suppress # Outpulsing? n  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
```

Settings on **Page 4** are as follows.

```
change trunk-group 1                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? y
                                  Network Call Redirection? y
          Build Refer-To URI of REFER From Contact For NCR? n
                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 120


                        Convert 180 to 183 for Early Media? n
                  Always Use re-INVITE for Display Updates? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
                                              Enable Q-SIP? n

          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

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.