Avaya Solution & Interoperability Test Lab

Application Notes for configuring Funktel DPx SIP Trunk Solution with Avaya Aura® Communication Manager R7.0 and Avaya Aura® Session Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Funktel DPx to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay particular attention to the scope of testing as outlined in Section 2.1, as well as 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 for provisioning Funktel DPx to interoperate with Avaya Aura® Communication Manager R7.0 and Avaya Aura® Session Manager. Funktel DPx is configured to connect to Session Manager via a SIP Trunk and hosts the Funktel f.airnet IP DECT Handsets. Calls are made to and from the f.airnet DECT handsets over the SIP Trunk.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Funktel DPx to send and receive calls using the f.airnet DECT handsets. Calls to and from Avaya H.323 and SIP deskphones were made during the testing.

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 compliance testing included the test scenarios shown below. Note that when applicable, all tests were performed with Avaya SIP deskphones, Avaya H.323 deskphones, Funktel f.airnet DECT endpoints and PSTN endpoints.

- Basic Calls
- Hold and Retrieve
- Attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy (Controlled on DPx)
- Call Waiting
- Call Park/Pickup
- Conference
- Calling Line Name/Identification
- Codec Support
- DTMF Support

2.2. Test Results

All tests carried out passed successfully.

2.3. Support

Support from Avaya is available by visiting the website http://support.avaya.com and a list of product documentation can be found in **Section 11** of these Application Notes.

Technical support from Funktel can be obtained through the following:
Marcel Schwiebert
Phone: +49 5341 223 5313
E-mail: marcel.schwiebert@funktel.com
Web : www.funktel.com

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. The Funktel f.airnet DECT handsets connect to the Funktel f.airnet DECT base station and registered to the Funktel DPx which is placed on the LAN. The DPx connect directly with Session Manager as a SIP Trunk in order to be able to make/receive calls to and from the Avaya H.323 deskphones on Communication Manager and SIP deskphones registered with Session Manager.

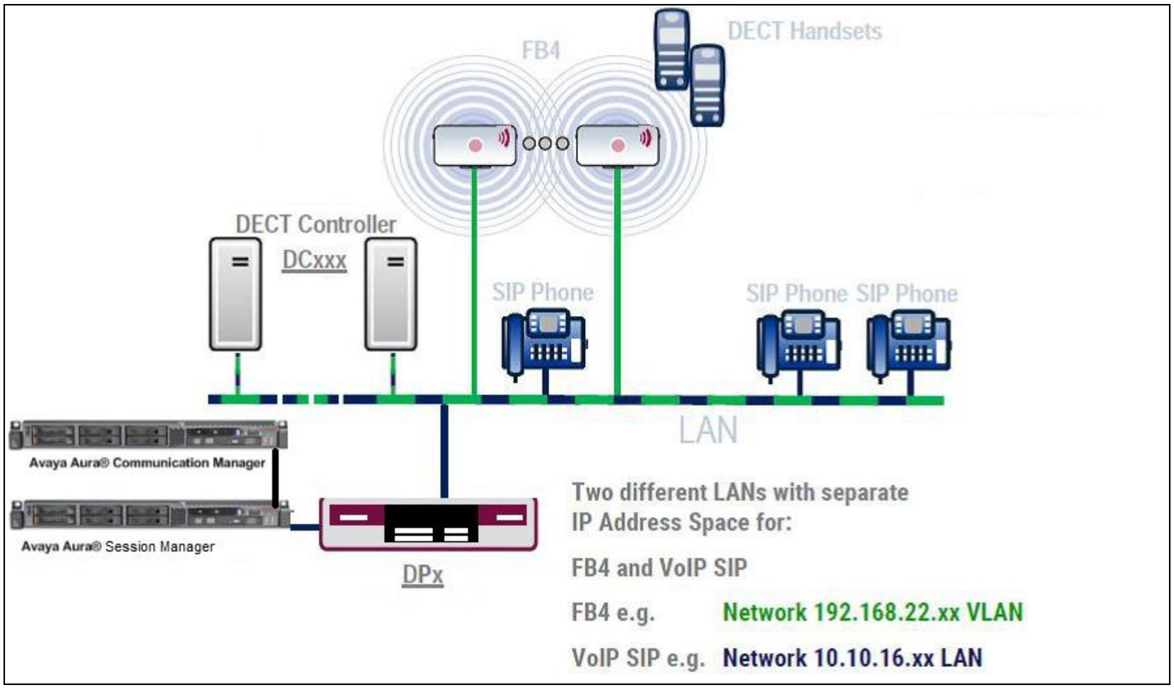


Figure 1: Network Solution of Funktel DPx hosting Funktel f.airnet DECT Handsets with Avaya Aura® Communication Manager R7.0 and Avaya Aura® Session Manager.

4. Equipment and Software Validated

The following equipment and software was used for the compliance test.

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtual Server	R7.0.1.2 R017x.00.0.441.0 Version 7.0.1.2.0.441.23523 Patch: Kernel-2.6.32.3.1.e16.AV4 PLAT-rhel6.5-0050
Avaya Aura® Session Manager running on Virtual Server	7.0.1.2.701230
Avaya Aura® System Manager	7.0.1.2 Build No: 7.0.0.0.16266 Software Update Revision No: 7.0.1.2.086007 Service Pack 2
Avaya G430 Gateway	37.41.0 /1
Avaya 9611G/9641G H323 Deskphones	6.6229
Avaya 9641G SIP Deskphone Avaya 9611G SIP Deskphone	7.0.1.1
Funktel Equipment	**Software / Firmware Version**
FC4 Handset	3.2.x
D11 light grey	3.2.x
FC11 (blue)	3.2.x
FB4 IP TP	V5.1.x
BSIP1 ikon IP DECT Basisst.	V5.1.x
DC 200 DoIP-Controller	5.1.x
DPx	5.30a

5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing. For further information on the configuration of Communication Manager please see **Section 11** of these Application Notes.

- Customer Options.
- Sip Trunk.
- Network Region.
- IP Codec.

5.1. Check SIP Trunk License

Use the **display system-parameters customer-options** command to check the SIP Trunk licensing. Go to page 2 and make sure that there are sufficient numbers for **Maximum Administered SIP Trunks**.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000  22
           Maximum Concurrently Registered IP Stations: 2400  4
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  3
                   Maximum Video Capable IP Softphones: 2400  5
                       Maximum Administered SIP Trunks: 4000  1290
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

5.2. Configure a SIP Trunk

To allow calls to be made between Communication Manager and the Funktel DPx a SIP Trunk must be administered. Use change node-names ip to add the Session Manager. In this example the Session Manager is called **SM1677**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AMS1616             10.10.16.16
CMS1640             10.10.16.40
SM1677              10.10.16.77
```

Use the **add signaling group x** command where x is an unused trunk group number from 1-999. Set the **Group Type** to **sip**, set the **Near-end Node Name** to a Communication Manager Clan entry or as below using the **procr**. The **Near-end Listen Port** is automatically set to **5060**. Set the **Far-end Node Name** to the entry added to node-names ip and set the **Far-end Listen Port** to **5060**. Set the **Far-end Network Region** to **1**

```
add signaling-group 20                                            Page   1 of   2
                                SIGNALING GROUP

 Group Number: 20             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: Others
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y
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? 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
```

Use **add trunk-group x** where x is a valid Trunk group number. In this test the same number was used for trunk and signaling group. On **Page 1** set the **Group Type** as **sip** and give it a descriptive **Group Name**. Set a valid **TAC** and set the **Service Type** as **tie**. Set the **Member Assignment Method** as **auto**, set the **Signaling Group** as administered above and set the **Number of Members** to the number required (max 255)

```
add trunk-group 20                                              Page   1 of  22
                                TRUNK GROUP

Group Number: 20                   Group Type: sip           CDR Reports: y
  Group Name: SM1677                     COR: 1        TN: 1        TAC: 720
   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: 20
                                                     Number of Members: 255
```

On Page 3 set the **Numbering Format** as **private** so that calls sent over this trunk will not be proceeded by a +.

```
add trunk-group 20                                              Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? n
```

Use the **change ip-network-region x** (where x is the network region to be configured) command to assign an appropriate domain name to be used by Communication Manager, in the example below **devconnect.local** is used.

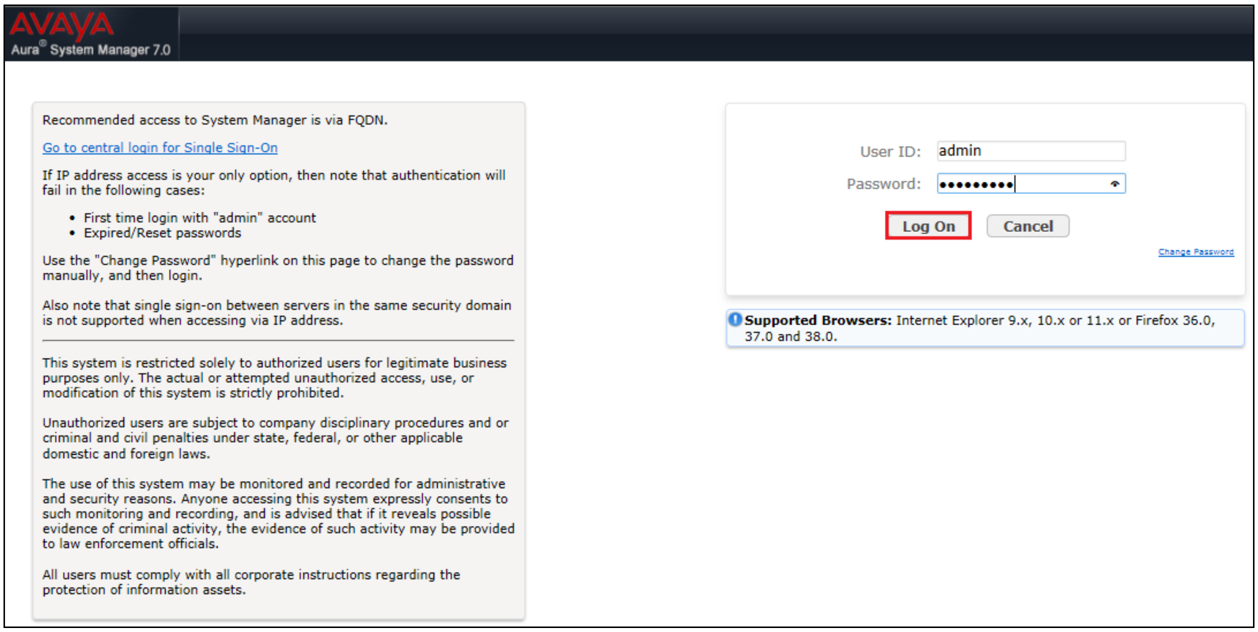
```
change ip-network-region 1                                  Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devconnect.local
    Name: default NR
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   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
```

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with the Funktel f.airnet Handsets, which support both **G.711A** and **G.729A**.

```
change 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:
```

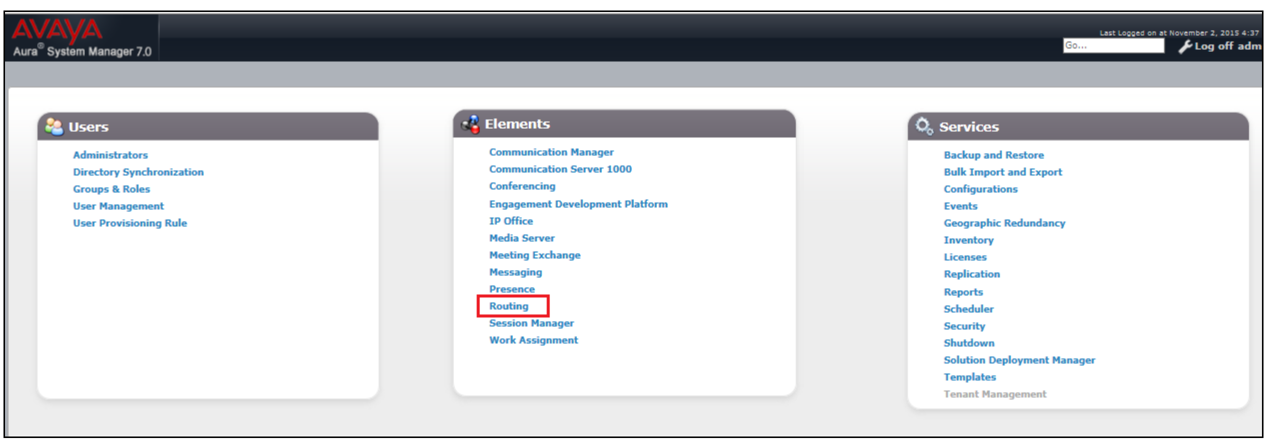
6. Configure Avaya Aura® Session Manager

The Funktel DPx is added to Session Manager as a SIP Entity and an Entity Link, Routing Policy and Dial Pattern is added to route calls to the f.airnet DECT Handsets. In order 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.

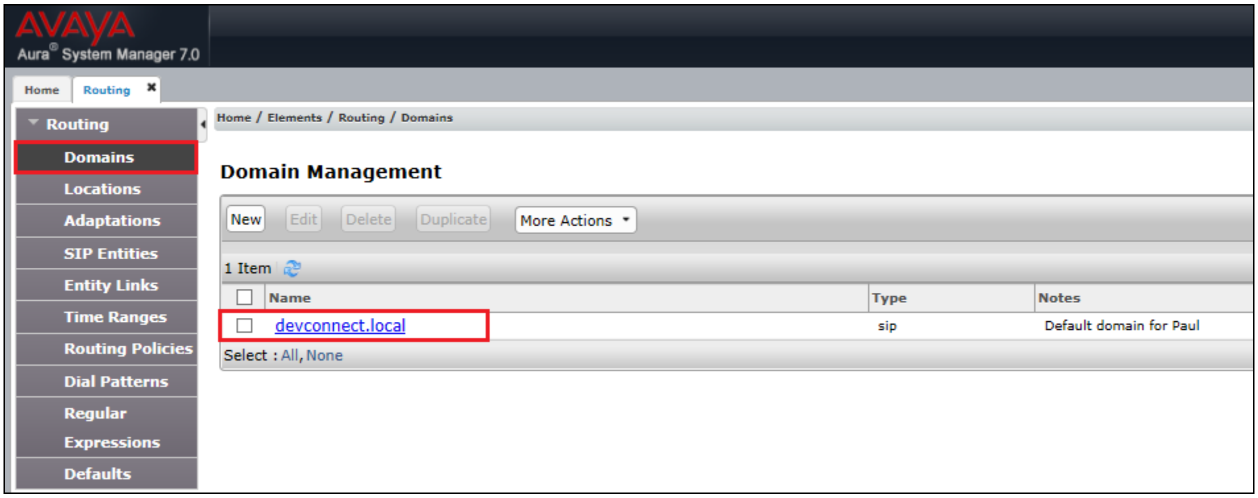


6.1. Configuration of a Domain

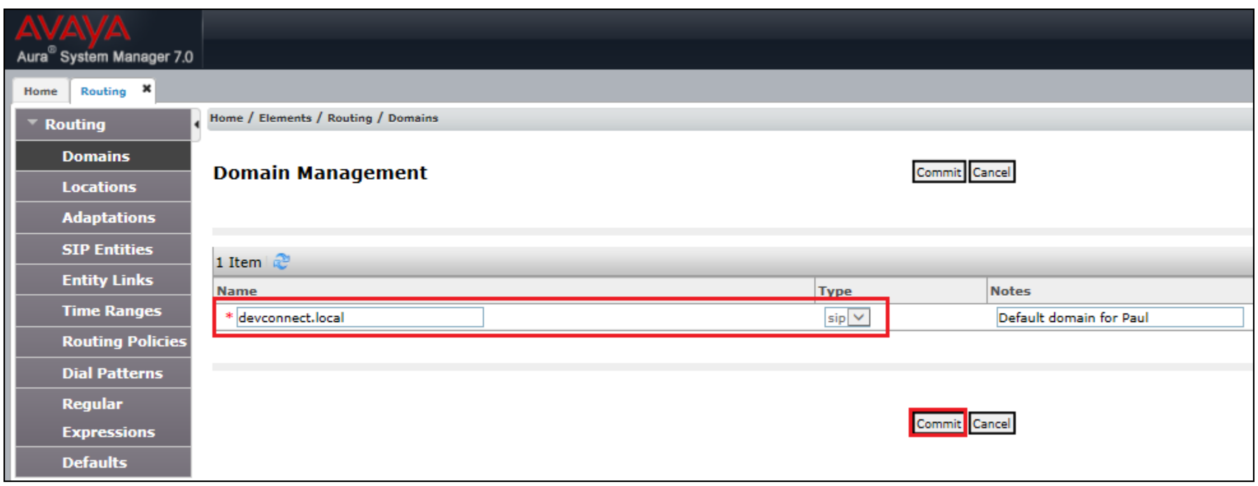
Click on **Routing** highlighted below.



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.

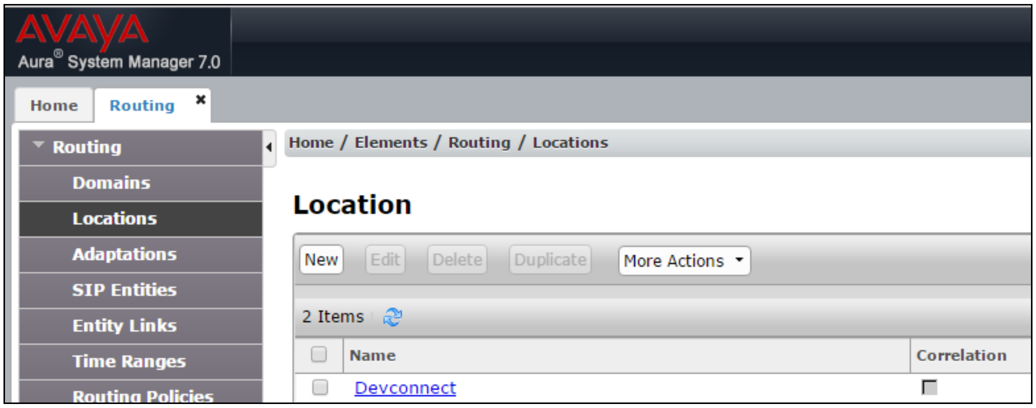


Clicking on the domain name above will open the following window; this is simply to show an example of such a domain. When entering a new domain the following should be entered, once the domain name is entered click on **Commit** to save this.

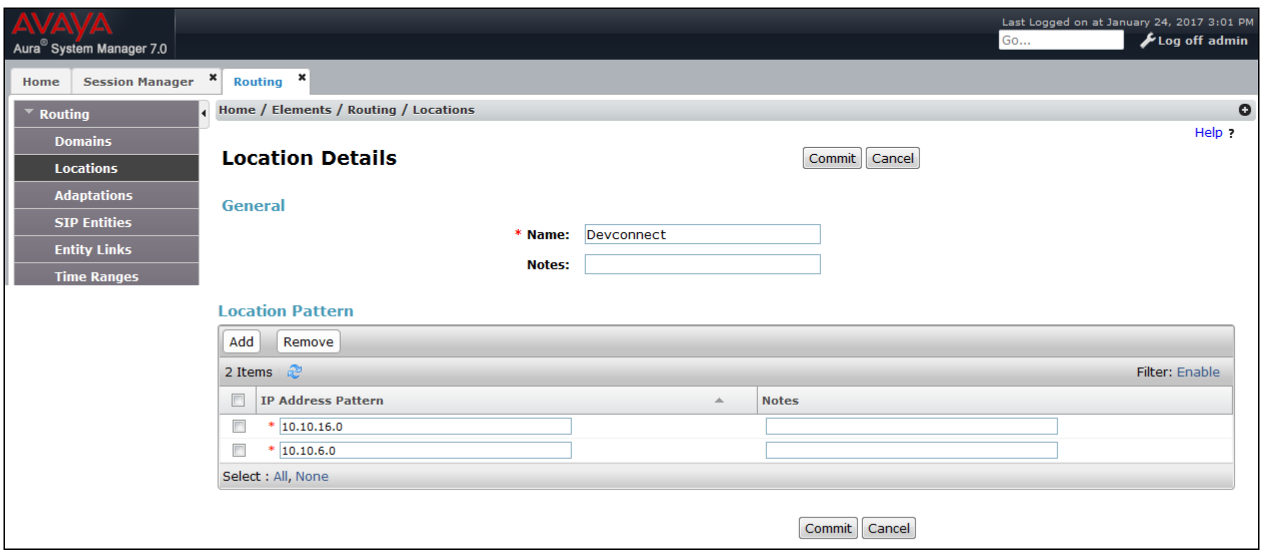


6.2. Configuration of a Location

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 **Devconnect** is already setup and configured and clicking into this will show its contents.

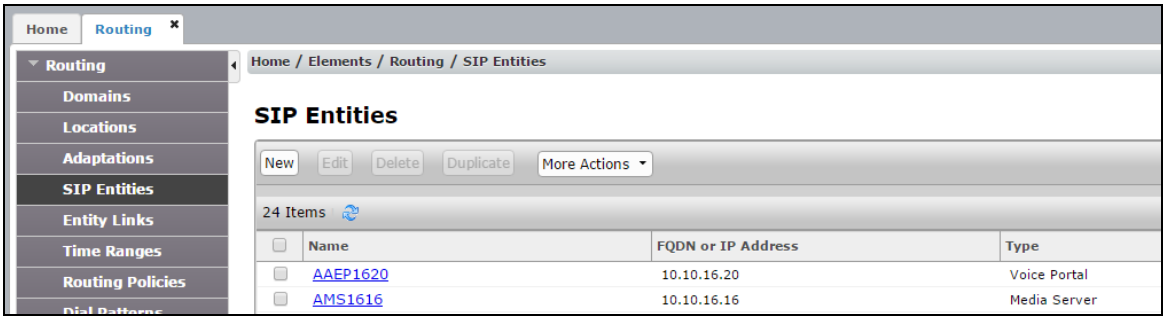


The Location below shows a suitable **Name** with a **Location Pattern** of **10.10.16.0**. Once this is configured click on **Commit**.

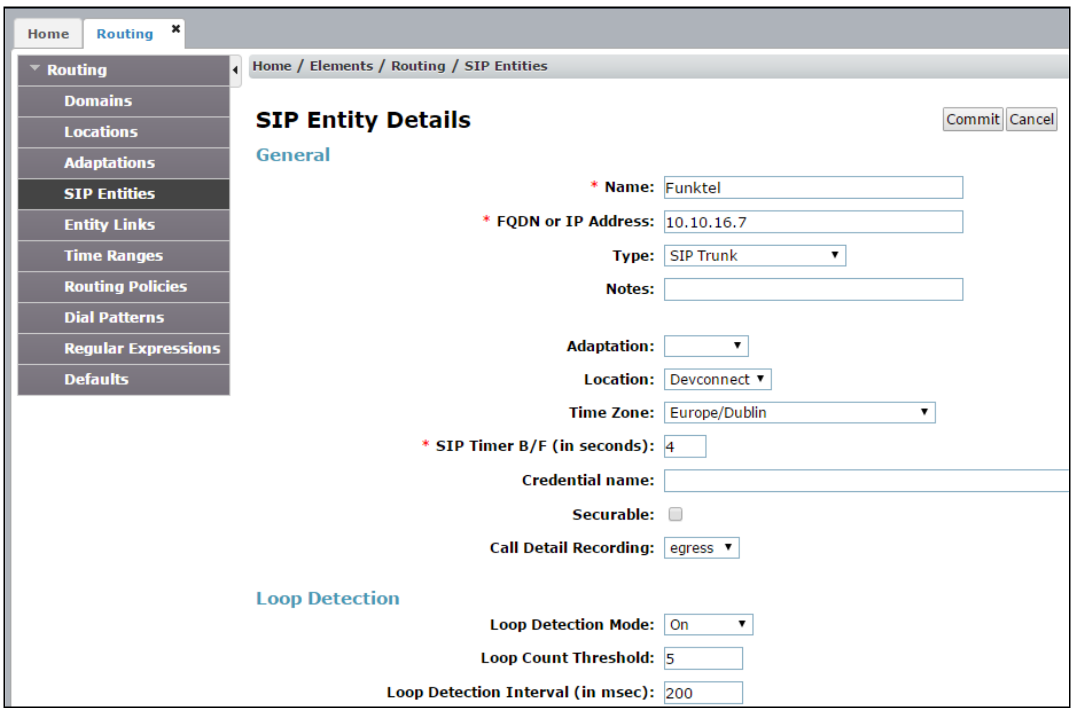


6.3. Configuration of the Funktel DPx as a SIP Entity

From the left hand menu select **SIP Entities** and click on **New.**

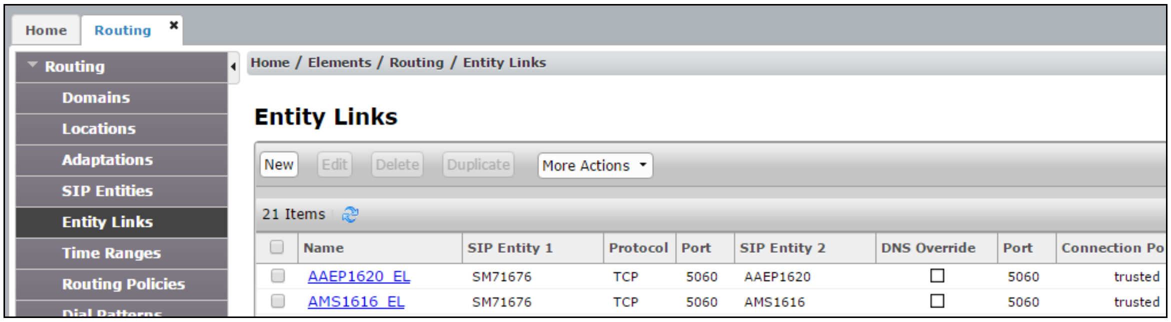


Enter a descriptive **Name:**, The **IP address** of the Funktel DPx and Select **SIP Trunk** from the **Type:** drop down. Select the Location added in **Section 6.2**and Click on Commit to save the changes

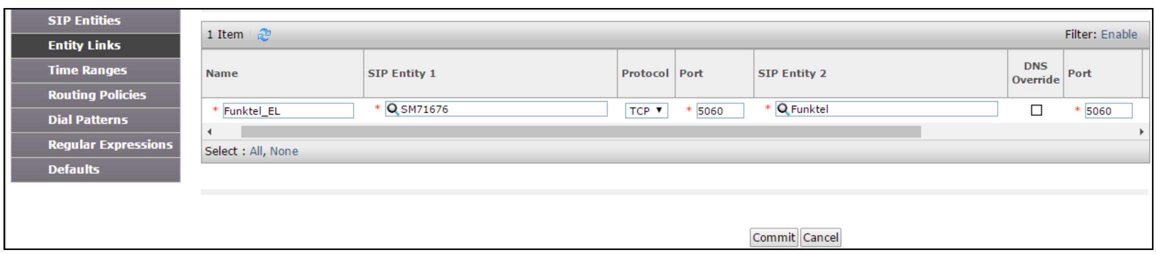


6.4. Configuration of the entity Link between Session Manager and the Funktel DPx

From the left hand menu select **Entity Links.** Click on **New.**

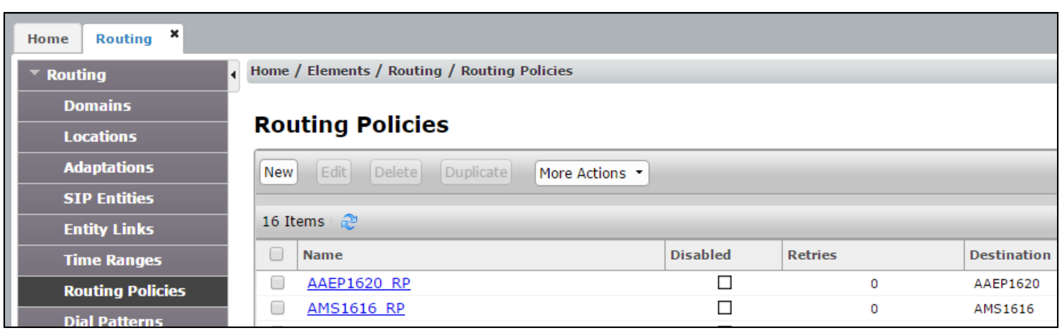


Enter a descriptive **Name**. Select the Session Manager entity connected to the Communication manager as **SIP Entity 1**. Select the SIP entity added in **Section 6.3** as **SIP Entity 2**. Set the protocol to **TCP** and the default port will update automatically.



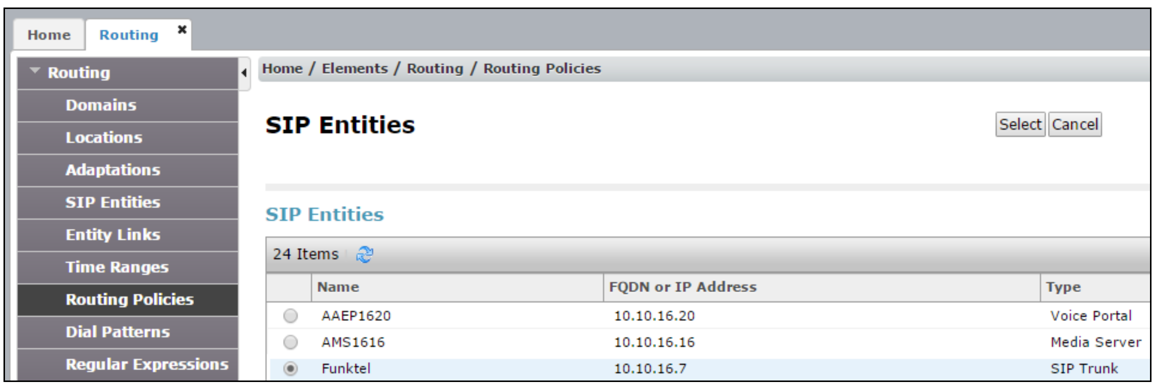
6.5. Configuration of the Routing Policy

Select **Routing Policies** from the left hand menu and click on **New**.

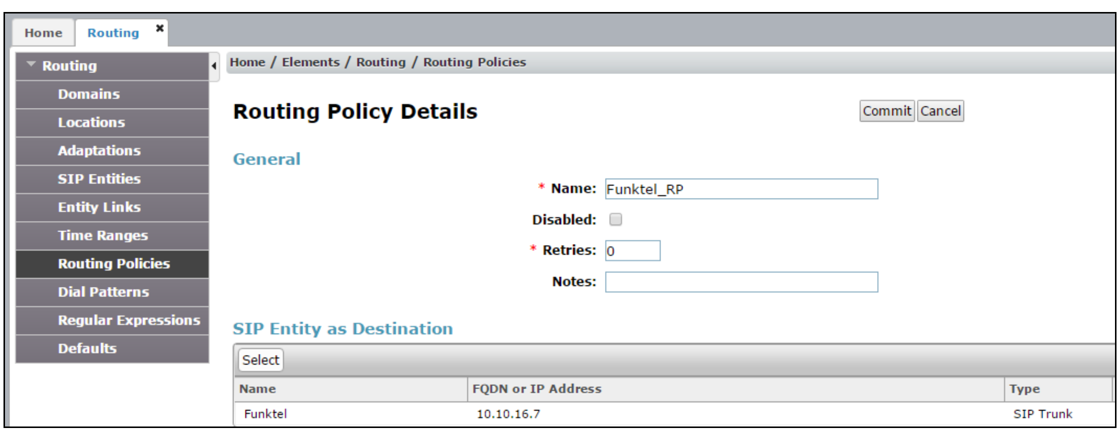


Enter a descriptive **Name:** and click on **Select** under **SIP entity as Destination**.

Select the **SIP Entity** added in **Section 6.3** and click on **Select**.

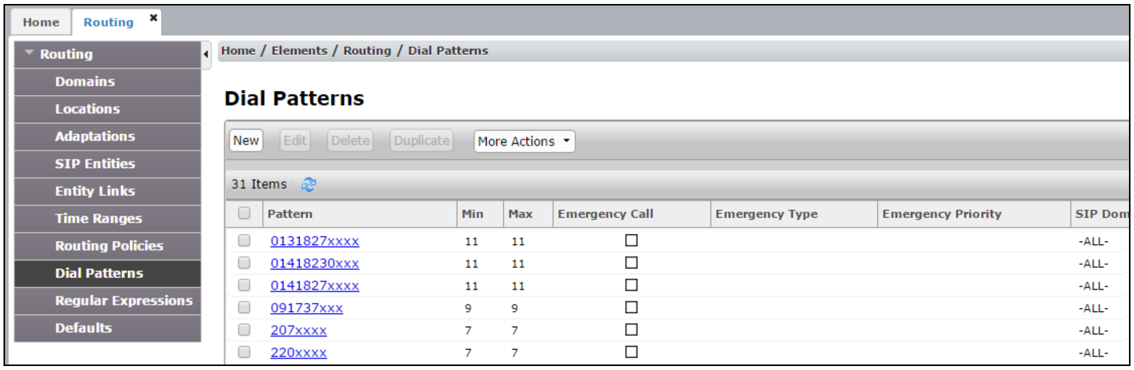


Click on **Commit**.

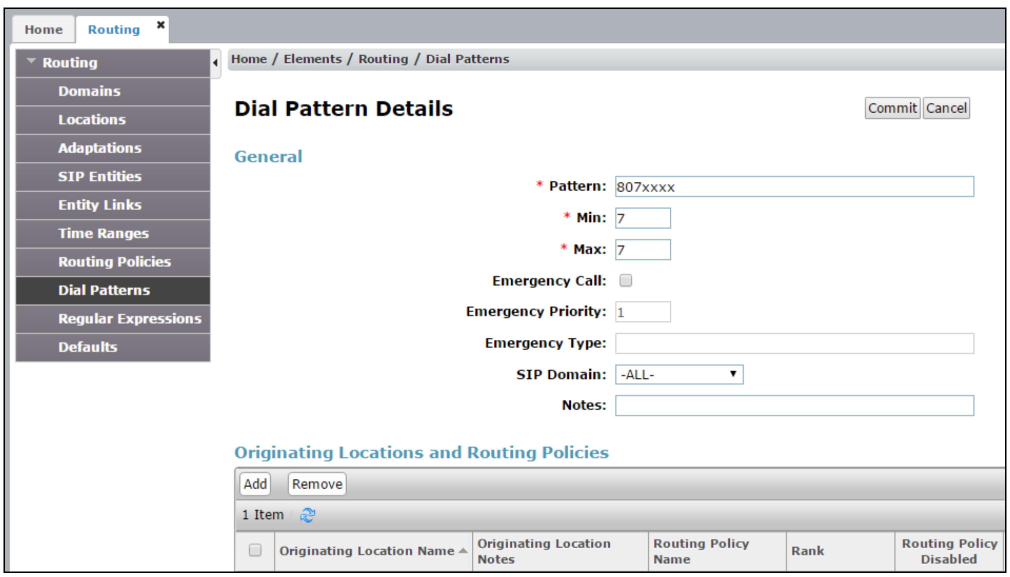


6.6. Configuration of a Dial Pattern to route calls to the Funktel DPx

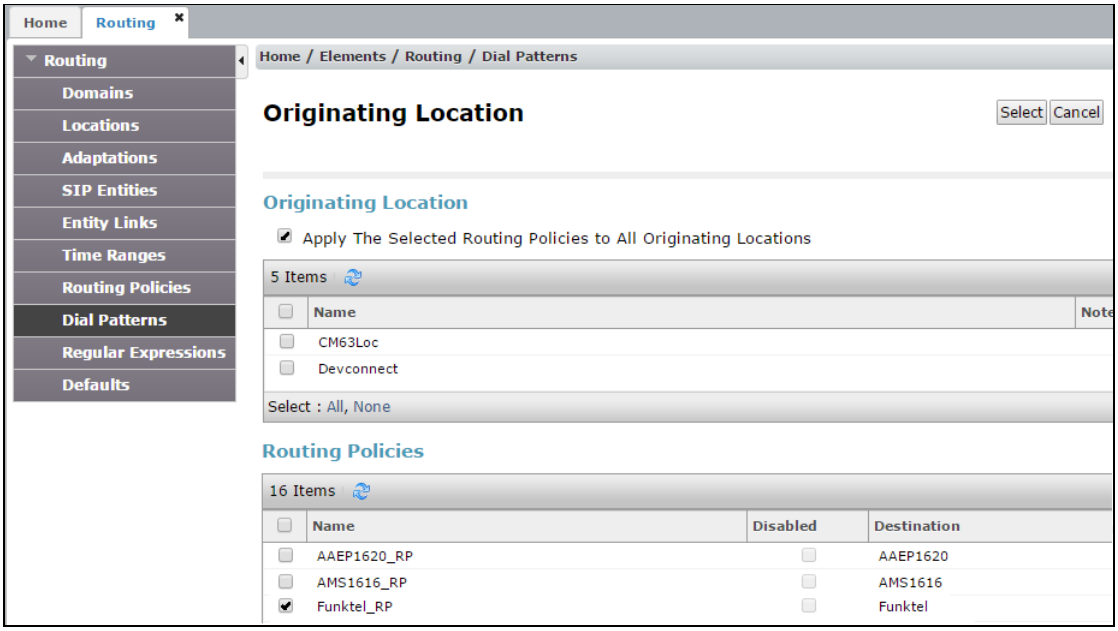
From the left hand menu select **Dial Patterns** and click on **New**.



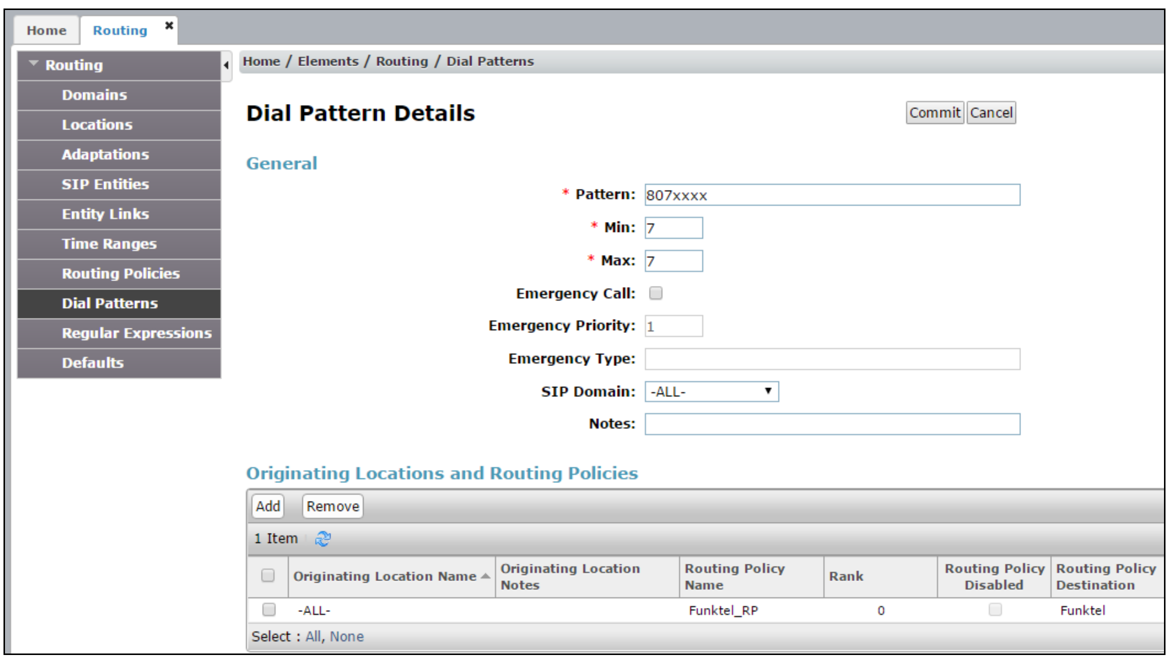
Enter the **Pattern:** that corresponds with the Extension numbers administered in **Section 7.4**. Enter the **Min** and **Max** digits for the extensions. Click on **Add** under **Originating Locations and Routing Policies**.



Select **Apply the Selected Routing Policies to All Originating Locations**. Select the Routing Policy administered in **Section 6.5**



Click on **Commit**.



7. Configure DPx SIP Trunk

7.1. About the configuration software

The Java-based DPx Admin configuration software has the following system requirements:

- Windows-PC or Linux-PC
- Java Version 8 (tested with Oracle JRE)

The configuration screens are grouped in topics, for the sake of greater clarity. Each screen presents a clear list structure, which can be further enhanced with the aid of filters. "Tool tips" are built into the configuration screens… If you place the cursor on a term, a tool tip regarding this term is soon superimposed, which will facilitate configuration. All configured data sets can be deactivated individually on a temporary basis without the respective data set being lost.

7.2. Network

As standard, the first Ethernet interface NET 1 is configured with the IP address 192.168.0.253 and the network mask with 255.255.255.0. No DHCP server is configured on this interface.

DHCP-Server

The second Ethernet interface NET 2, if available, serves as a DHCP server and makes numbers 172.20.153.x available in the IP network, with the network mask 255.255.255.0 Its own address is 172.20.153.1. Addresses 172.20.153.2 to 172.20.153.9 are not allocated via DHCP.
Ensure that no other DHCP server is operating on the network. The operation of the two DHCP servers in the same network is not supported!

7.3. Exclusive Ethernet connection to a PC

This connection is required if you wish to configure the DPx, before integrating into the network. This is also the recommended method.
Use an Ethernet cable (CAT-5, crossed), to connect a PC to interface NET2 of the appliance. In order to be able to establish an exclusive Ethernet connection with the appliance, the PC must satisfy the following requirements.

- The TCP/IP protocol must be installed on the PC.
  Check the TCP/IP settings
- The network addresses of both systems must match each other. In doing so the following options are available:
- No change to the network settings of the appliance is required as the PC will acquire its IP address dynamically. This will be a fixed IP address in the 172.20.153.10–172.20.153.254 range and set the sub-network mask to 255.255.255.0
- The IP address of the DPx appliance is changed in the course of the basic configuration and have adapted the conditions according to the circumstances of the network. Set the same sub-network mask and a fixed IP address in the same address range on the PC. The IP addresses of both systems may only differ from each other in the final address block. 172.20.153.1 and PC: 172.20.153.20). Call up the DPx Admin Interface by means of your web browser and start the configuration.

Further information in this regard can be found in the chapter titled "Calling up the DPx as Administrator".

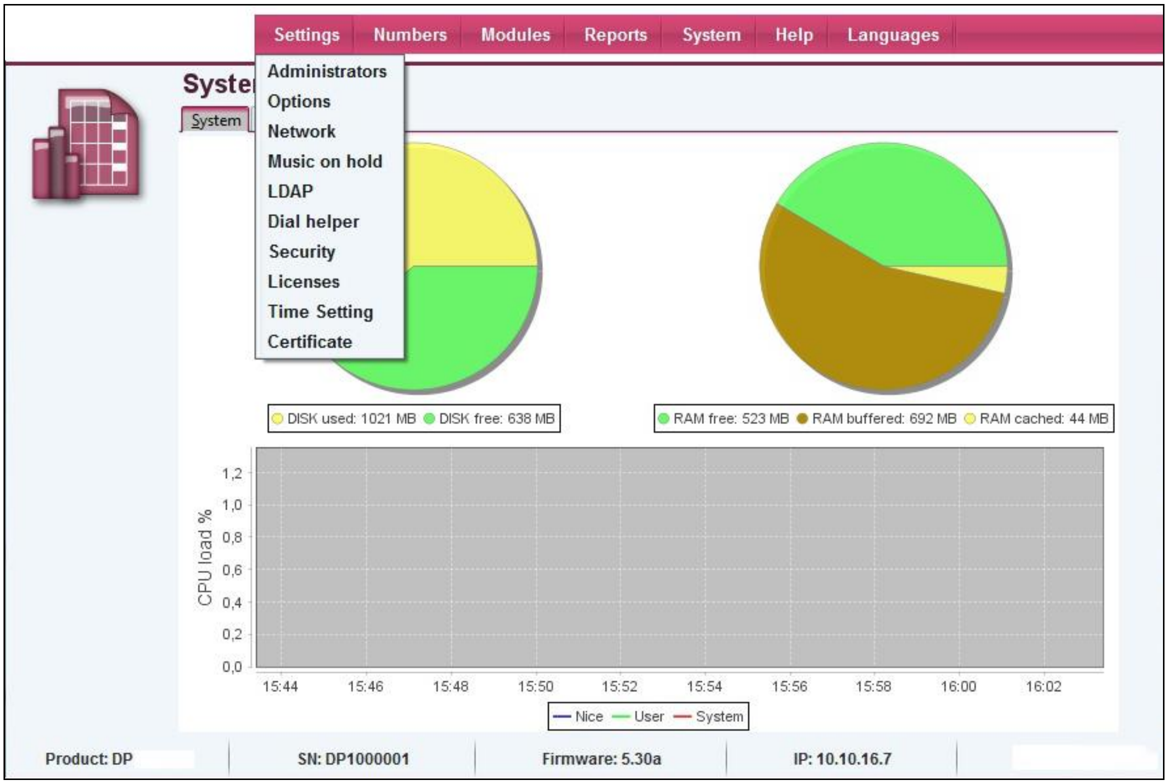
7.3.1. Calling up DPx as Administrator (DPx Admin)

Start your web browser (e.g. MS Internet Explorer, Firefox, Chrome or Safari) and enter the IP number of the system in the address line (see previous section). The DPx Admin program is downloaded from DPx and started.
The login screen appears:

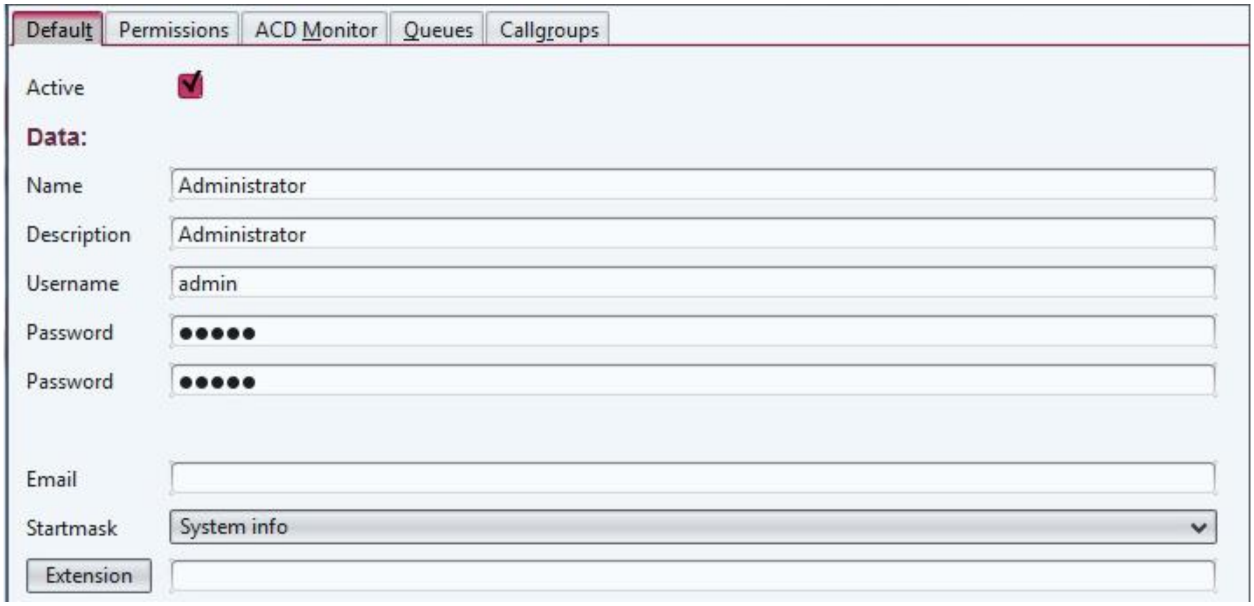
As the administrator, use an appropriate login and password.
The default password must be changed during the initial installation. Make a note of the new password and store it in a safe place. Should you forget the password, the system must be reset to the factory settings by the customer service department.
After clicking on **Login** access is granted to the DPx.
The main menu is always in the top blue bar. Pull-down sub-menus will be shown when clicking on the menu items.

7.3.2. Settings

Under **Settings** for the drop-down menu shows **Administrators**, **Options**, **Network**, **Music on Hold**, **LDAP**, **Dial helper**, **Security**, **Licenses**, **Time Setting** and **Certificate** sub menus.



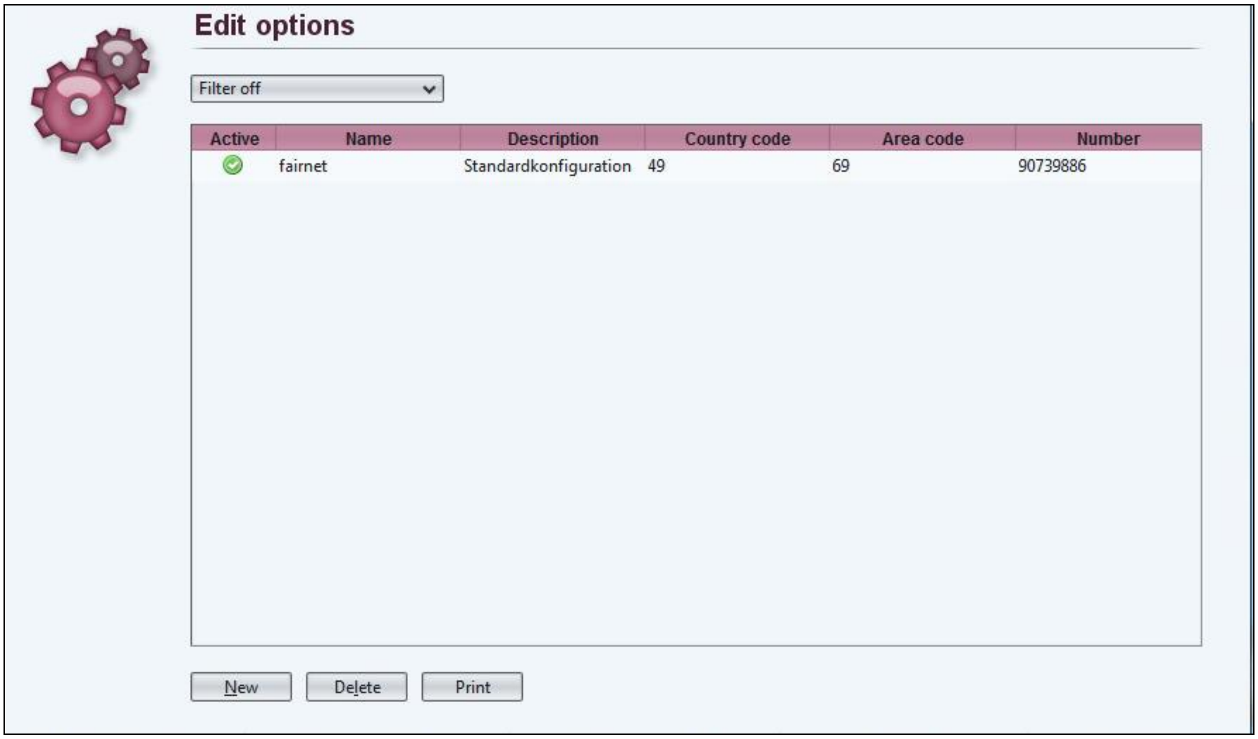
7.3.2.1 Default



Active	Select the checkbox, if this administrator is to be active. By deactivating, login by means of the stored access data is not possible.
Data:	
Name	Enter a name for this administrator.
Description	Enter a description for this administrator if necessary.
Username	Enter the username for the login process.
Password	The password required for the login process.
Email	An e-mail address can be stored for the administrator.
Startmask	the start screen stored at this point appears after a successful login.
Extension	An extension can be allocated to the administrator. This is necessary if the administrator has SiMo (silent monitoring) rights for the ACD monitor, among others.

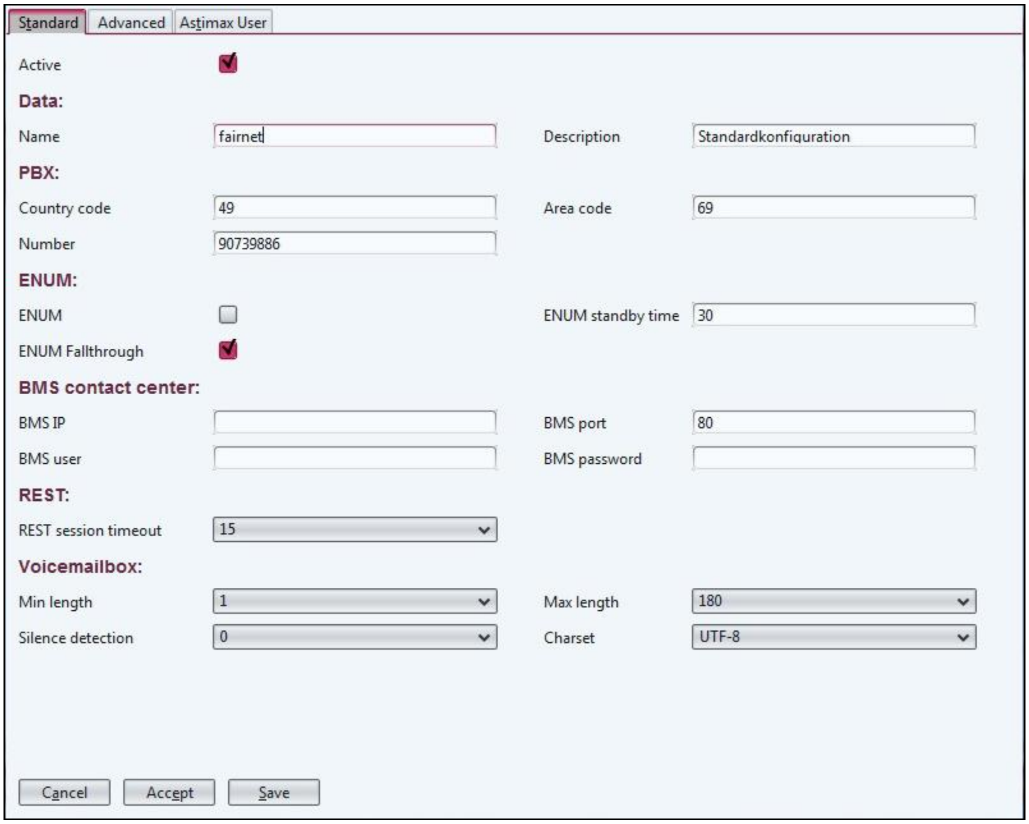
7.3.2.2 Network

Multiple configurations can be defined, of which only one can be active at any one time. Usually, only one configuration data set is required.



By double-clicking the entry above the Standard entry screen is opened. By clicking on **New** an entry screens opens and new entries can be added as follows.

7.3.2.3 Standard

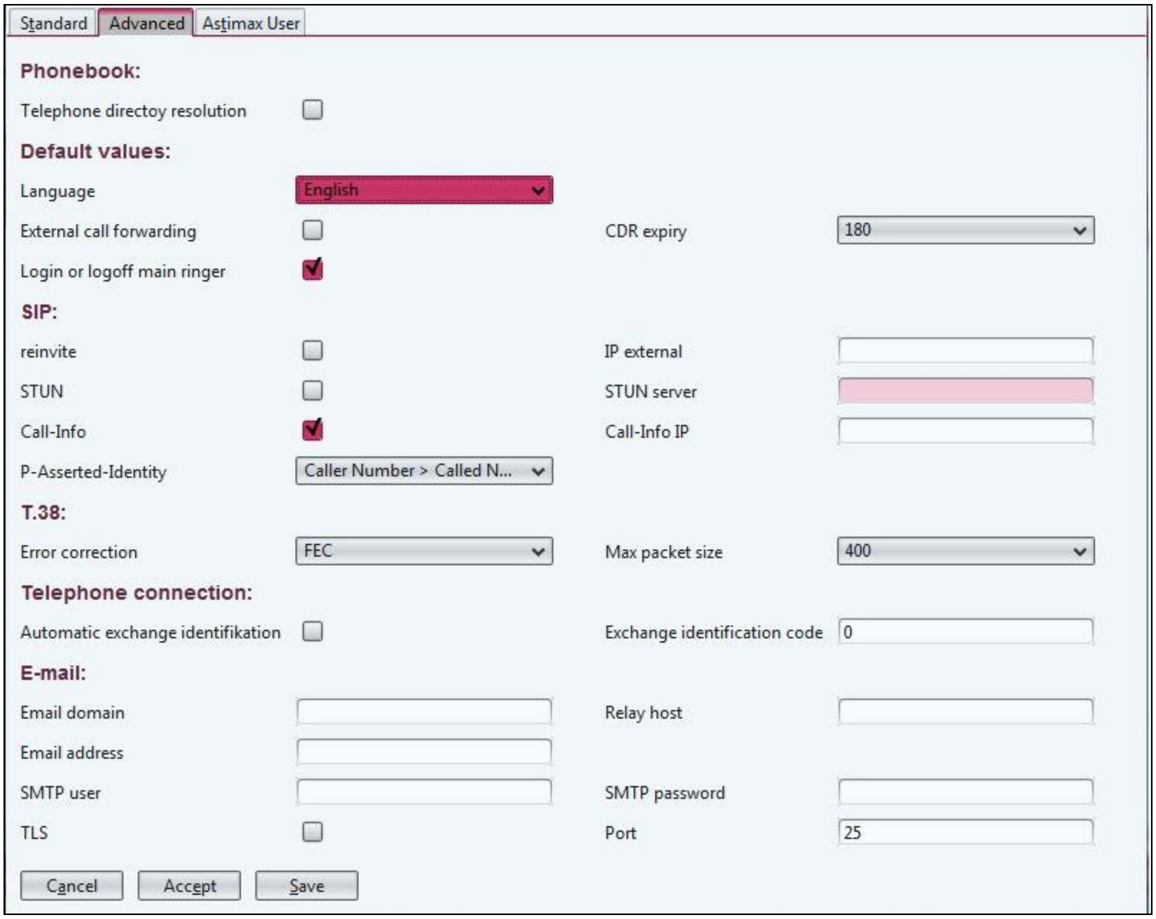


Active	Mark the checkbox, if this setting is to be active.
Data:	
Name	Enter a name for this configuration in the field.
Description	Enter a description for this configuration if necessary.
PBX:	
Country code	Entry of the international country code (Germany = **49**, Austria = **43**, Switzerland = **41** etc.).
Area code	Entry of the area code without a zero prefix (example: Kiel = **431**).
Number	The fixed line or connection number supplied by your telephone service provider.
ENUM:	
ENUM	Mark the checkbox to switch on the ENUM functionality.
ENUM standby time	Here the time (in seconds) can be set.
ENUM Fallthrough	Mark the checkbox if in the case of a non-existent ENUM entry, the calls is to be made via the configured IP provider.
BMS Contact Centre:	
BMS IP	IP address of the DPx BMS Contact Centre application server.
BMS port	User port of the DPx BMS Contact Centre application server.

BMS user	The user name for the DPx BMS server. The BMS server must use this data to log in to the DPx IP DPx
BMS password	The password for the DPx BMS server.
REST:	
REST session timeout	The automatic session end of a connection by means of the REST API is set here. If an exchange of data no longer takes place via this connection, then the connection is disconnected automatically after the preset number of minutes.
Voice mailbox:	
Min length	Enter the minimum duration in seconds of a voice message here. The minimum duration must always be greater than the time required for the detection of silence.
Max length	Enter the maximum duration of a voice message here. The maximum duration must always be greater than the minimum duration.
Detection of silence	Here how many seconds the voice mailbox must recognise the end of a voice message is set. If, for example, nothing is said for the set period of time, then the recording is ended automatically. The time required for recognition of silence must always be less than the minimum duration.
Charset	Select the character set coding for e-mail notifications of the voice mailbox.

Note:ENUM stands for telephone number mapping and can be understood to be similar to the domain name system (DNS). A completely normal telephone number is entered into database at DENIC and can then be accessed by Internet via the IP protocol. This can only happen if the connection also controls IP telephony. Should this not be the case, then the DPx DPx will simply dial in the conventional way via the telephone network.
ENUM has the advantage that the subscriber does not have to remember various numbers. Nowadays, service providers offer a cross-over from IP telephony to the fixed line network. However, if a customer of one operator now wishes to call a subscriber of the other, then he/she must know the other's VoIP number. The interconnection of the networks does not help in this instance. Use ENUM here. You simply dial the known fixed-line number and the DPx looks up the desired route automatically. If the ENUM entry is positive, then a purely Internet-based connection is used; if the entry is negative, then dialling takes place via one of the configured VoIP providers.

7.3.2.4 Advanced



Phonebook:

Phone directory resolution	Mark the checkbox, if a name entered in the phone book is to be shown in the telephone display.

Default values:

Language	Select the default language for voice messages. If, for example, another language has been selected for a voice mailbox, then the language that has been configured for the voice mailbox is used.
External call forwarding	Activate this function if every extension to be able to set the call forwarding function of any other extension. Call forwarding is set by means of code number dialing.
CDR expiry	Set the number of days after which the CDR data is to be deleted automatically.
Recording storage period	Set the number of days after which the recordings are to be deleted automatically.
Login or logoff main ringer	When using secondary extensions, the main extension can be logged in or logged out by calling on its own extension number. Activate this option, if you wish to allow this function.

SIP:

Reinvite	Mark the checkbox, to allow this function.
Reinvite	*By using this SIP option one can ensure that the RTP data of a VoIP call can be transferred directly between the terminals. This can reduce the data packet propagation times and free up the DPx. One should be careful when using this option on networks that are operated behind a NAT gateway to the Internet and via which IP phone calls are also made.*
IP external	This IP address is used for all outgoing SIP connections. This is helpful if the DPx is operated behind a NAT gateway and, for example, calls are to be made via an SIP service provider This IP address is not used in IP sub-networks, for which the DPx has his own IP address.
STUN	Activate this function, if the DPx is operated behind a NAT gateway and a STUN server is to be used. The IP external function is thereby deactivated.
STUN Server	Enter the IP address or the host name of a STUN server.
Call-Info	Activate this function to activate the Call-Info header in SIP notifications. The DPx uses this, for example, to transfer an image to the terminal when a call is made.
Call-Info IP	As an option, the IP address of the image source that is to be used in the Call-Info header can be entered here. If no IP address is stored, then the IP address of network interface NET 1 is used. Normally, nothing needs to be entered here.
P-Asserted-Identify	In the case of calls to an extension, the P-Asserted-Identity header is transferred to the terminals in SIP. Here you can determine the content that is to be conveyed by this header. Most terminals are in a position to evaluate and further process this header.

T.38

Error correction	Select between the various errors corrections for the T.38 protocol. Standard: FEC Maximum packet size set the maximum permissible packet size. Standard: 400

Telephone connection

Automatic exchange ID	Mark the checkbox, if an external dial-up line is to be assigned automatically (without the input of an **exchange code**). Internal connections must then begin with *****
Exchange ID code	Configure an alternative **exchange code** here. Exchange codes may contain up to 3 digits. The 0 exchange code is always active.

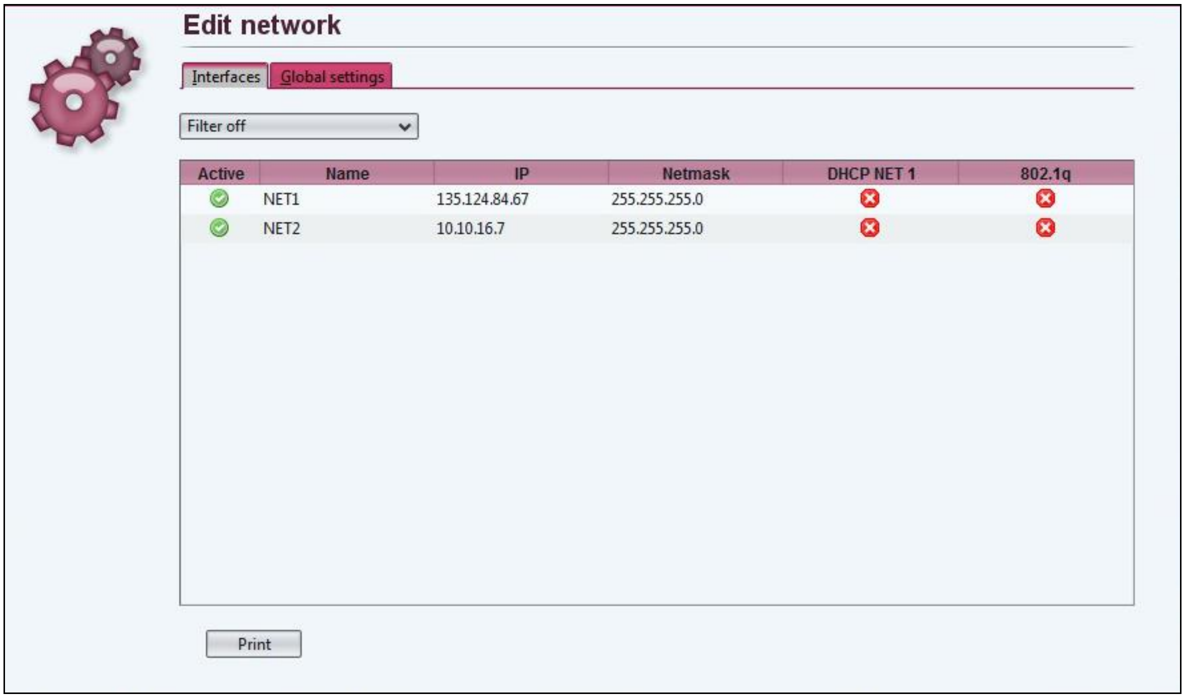
E-mail

Email domain	Domain name that is to be entered as the sender when sending voice messages.
Relay host	A relay host determines the default mail server to which the DPx sends e-mail. Here, the host must be entered with the full domain name (FQDN) or with the IP address.

	Take care that the name has been entered in the DNS server under **Network options**, is entered and that the DPx can resolve this as an IP address.
Email address	E-mail address that is to be used as the sender in the case of voice messages and system notifications.
SMTP user	Only to be filled in if the SMTP Server requires authentication.
SMTP password	Only to be filled in if the SMTP Server requires authentication.
TLS	Activate the TLS to set up a TLS-encrypted connection with the mail server.
Port	Enter the port number for the SMTP connections of the mail server. Port 25 is used as standard.

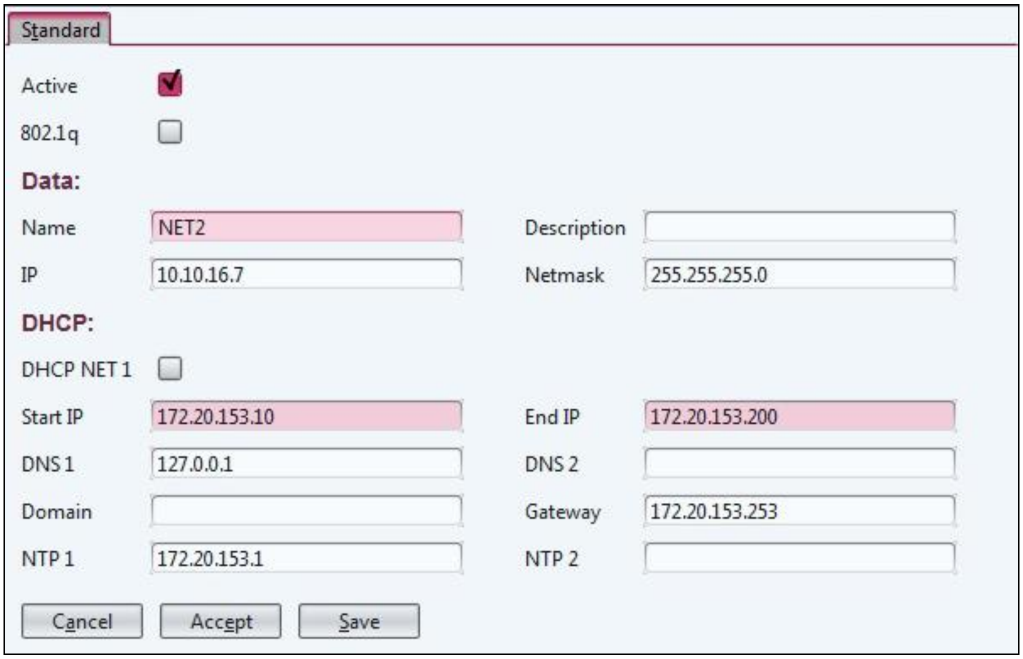
7.3.3. Network

Settings for configuration of the network interfaces and the DHCP server can be found in the menu below.



Depending on the type of system, from 1 to 6 individual interfaces can be configured. In addition, there is the option of IEEE 802.1q-compliant VLAN tagging.
The network interfaces available in each case are listed. These are opened for editing by double-clicking on the respective entry.
The list of network interfaces can be printed by clicking on **Print**.

7.3.3.1 Standard



Active	Mark the checkbox, if this network interface is to be active.
802.1q	Activate VLAN tagging here. A new tab marked 802.1q then appears, in which individual VLANs can configured. Network addresses can then only be configured in this tab. At least one VLAN ID must be present.

Data

Name	Enter a name for this network interface in the field.
Description	Enter a description for this network interface if necessary.
IP	Enter the IP address for this network interface.
Netmask	The corresponding network screen for this network interface is configured here.

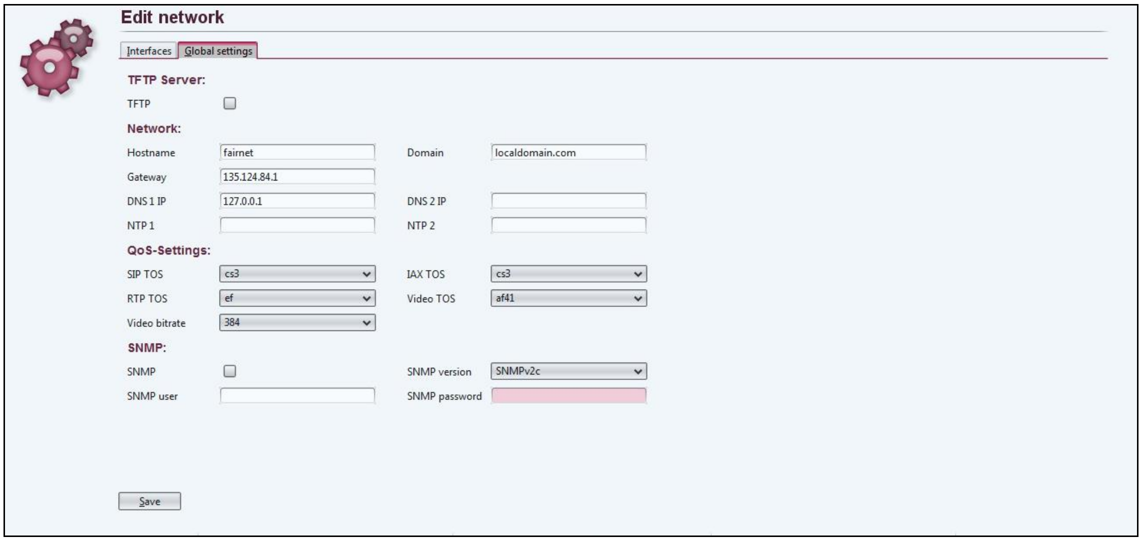
DHCP

DHCP NET 1	A DHCP server can be activated on the network interface.
Start IP	The first IP address of the DHCP pool is configured here.
End IP	The last IP address of the DHCP pool is configured here.
DNS 1…2	The DNS server to which the DHCP clients are to be assigned must be entered here.
Domain	The domain name of the network interfaces is configured here.
Gateway	Enter the standard gateway to which the DHCP clients are to be assigned, here.
NTP 1…2	Enter the NTP server that is to be used by the clients, here.

Not all interfaces must be configured In the case of the DPx, up to 6 interfaces. At least one interface must, however, show a valid configuration.

To save the information, click on **Save** on the lower bar. An instruction follows, that the system must be restarted.

7.3.3.2 Global settings



TFTP-Server

TFTP	Activate the integrated TFTP server of the DPx here. For example, firmware files for terminals can be made available by means of the TFTP server.

Network

Hostname	Name of the DPx
Domain	Internet Domain, if available
Gateway	The gateway of the IP-network. The data traffic of the DPx is routed further via this gateway.
DSN 1…2 IP	IP addresses of 1. and 2. respectively Name server. Two independent DNS servers must always be entered.
NTP 1…2	IP address or host name of the time server (an alternative can be entered for NTP 2), from which the system must request the current time. The 127.0.0.1 setting refers to the internal clock of the system.

QoS settings

Data traffic can be classified by the system into various service classes. A network that is configured for QoS, is thus in a position to priorities certain data ahead of other.

SIP TOS	From the list, select the class of the SIP signalling packets.
IAX TOS	From the list, select the class of the IAX signalling and voice data packets.
RTP TOS	From the list, select the class of the RIP voice data packets.
Video TOS	From the list, select the class of video data.
Video bitrate	Select the maximum video bitrate for a single video call. The maximum bandwidth selectable is 2048 kbps.

SNMP

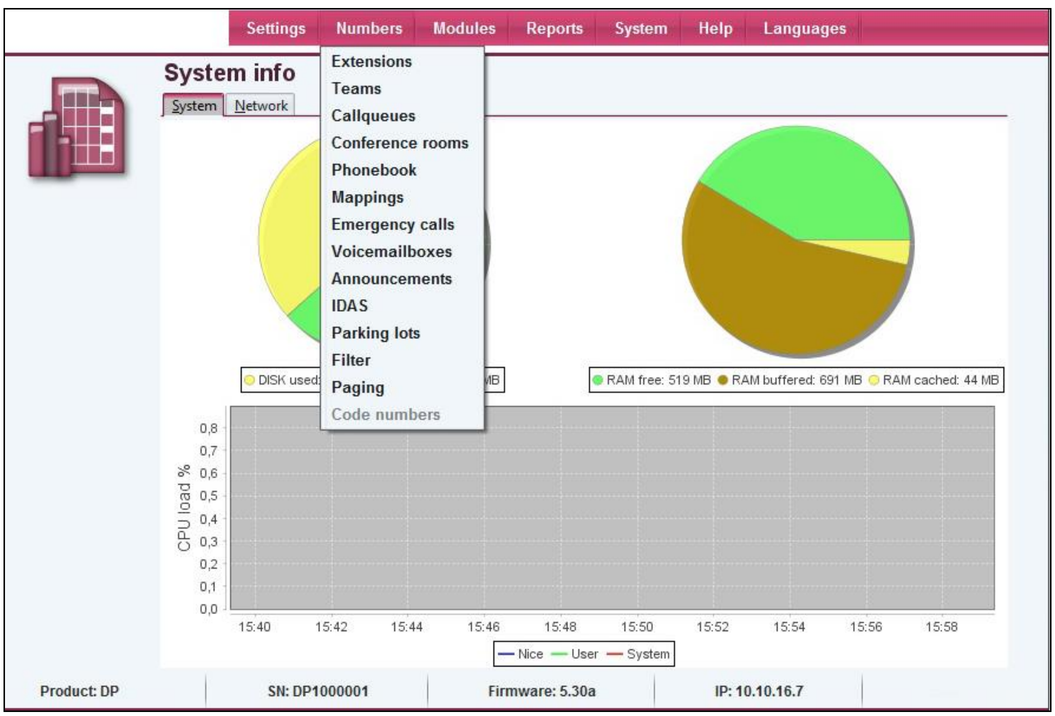
SNMP	Activate the SNMP service if to integrate the DPx DPx in a network monitoring system.
SNMP version	Select the SNMP version to be supported.
SNMP user	The SNMP client must log in to the DPx with a user name or a community.
SNMP password	A password is required for SNMPv3.

A phone call cannot be connected without a correct network configuration. Check if the DNS and NTP servers that have been entered can be accessed and are functioning. Telephony is not possible via SIP service providers (ITSPs) without a functioning DNS. Kindly contact funktel support in the event of problems. If the IP address of the system has been changed, log in to the system under the new IP after a restart.

To save the information, click on **Save** on the lower bar. An instruction follows, that the system must be restarted.

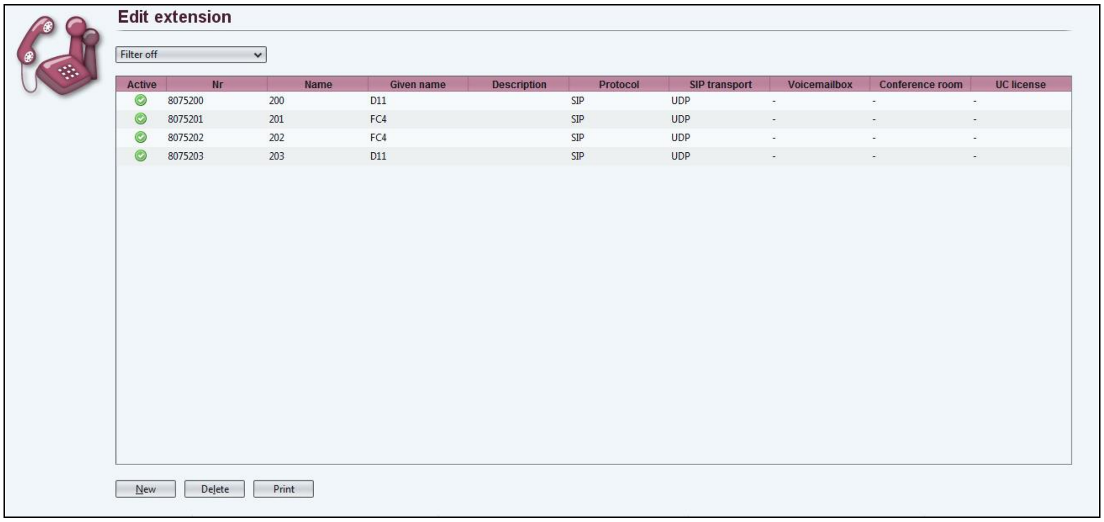
7.4. Numbers

The numbering plan used can be administered under the **Numbers** menu.



7.4.1. Extensions

Extensions can be managed under **Extensions**.

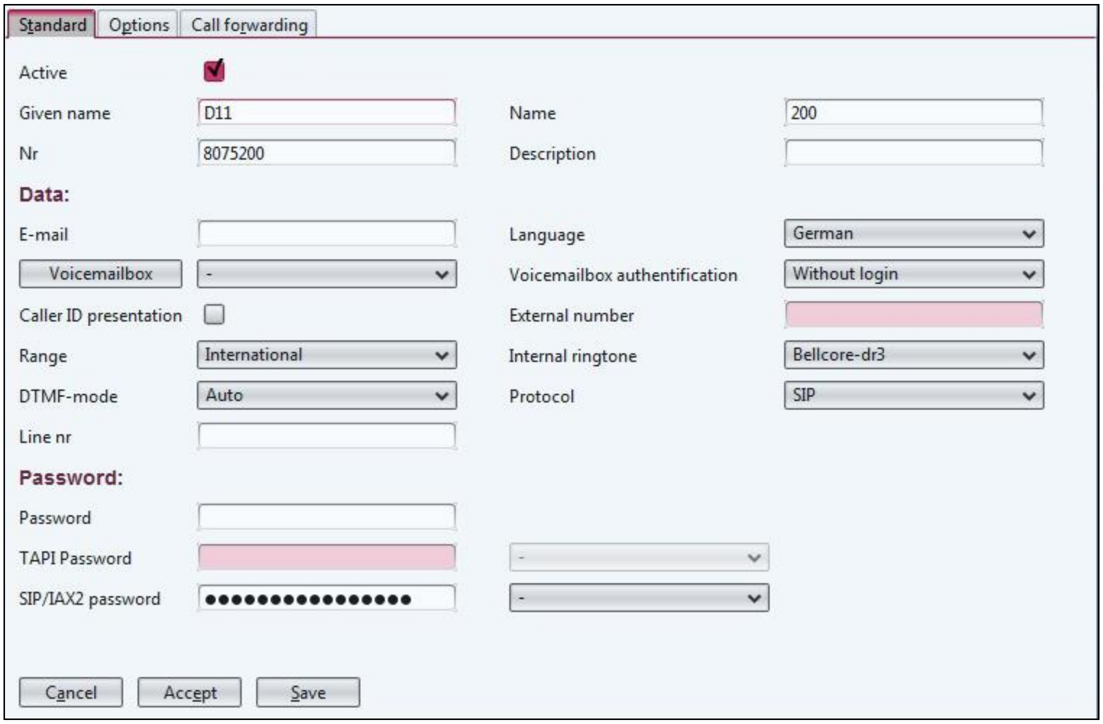


The extensions that have already been set up are listed. These are opened for editing by means of a double-click on the respective entry.

Click on **New** to set up a new extension.

7.4.1.1 Standard

When the New extension screen is open the first tab is **Standard**



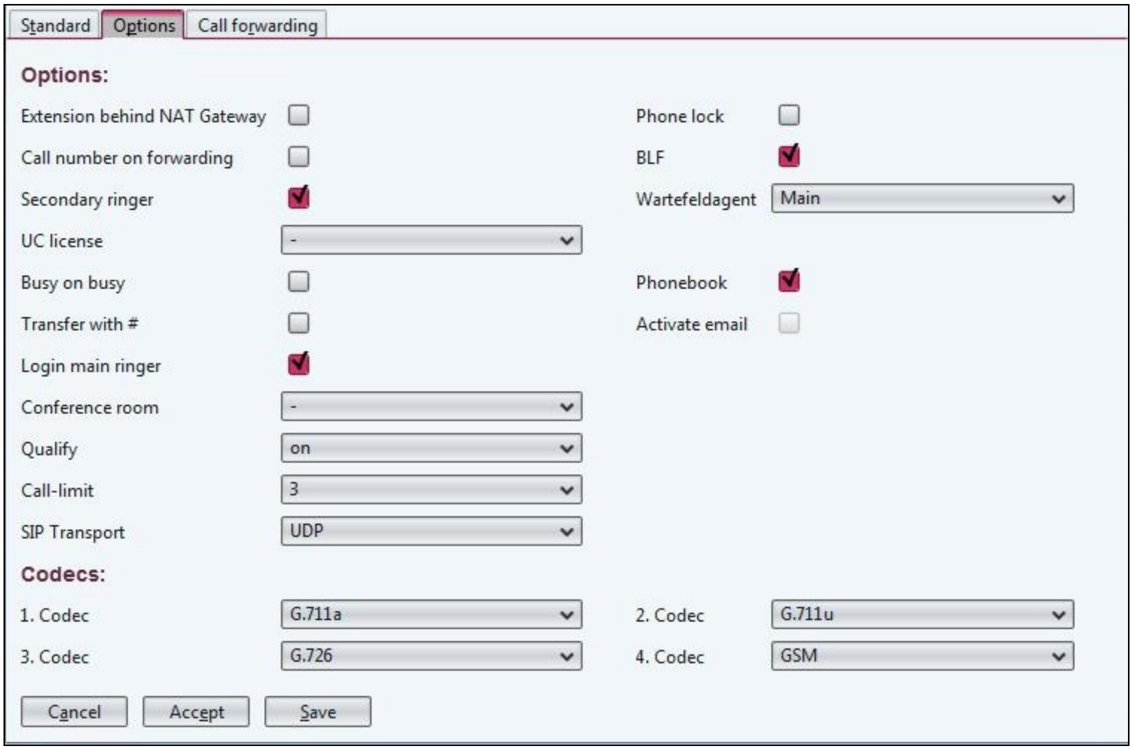
Active	Mark the checkbox, if this extension is to be active.
Given name	Enter the first names of the subscriber.
Name	Enter the surname of the subscriber.
Nr	Enter the internal number of the extension.
Description	Enter a description for this extension if necessary.
Data	
E-mail	Enter the E-mail address of the subscriber. The e-mail address is used for notification e-mails such as information regarding missed calls. The entry of the e-mail address still does not lead to automatic notification. In addition, this must be configured in the options of the extension
Language	Select the default language that is to be used by the DPx for this extension for notification texts. The language configured at this point can be overwritten by other configurations if required.
Voice mailbox	After clicking on the button, it is possible to create or edit a new mailbox. The selection list enables an existing voice mailbox to be assigned to this extension. All the existing voice mailboxes in the list are shown. In the same manner, a single mailbox can be assigned to multiple extensions.

Voicemailbox authentication	This option sets the login parameters for the specific voice mailbox. • Without login: No check when accessing mailbox, if the request is made from this extension. • PIN only: Only the PIN is requested if the voice mailbox is accessed from this extension.
Caller IP presentation	Mark this checkbox if the number of the subscriber is to be displayed.
External number	Enter the number that is to be transferred to the receiving device in the event of the number identification function being activated. The format must be national or international (e.g. 0049 431 123456… or 0431 123456…).
Range	Specify the outbound call rights for this subscriber. • Internal: Subscriber may only call internally • Local: Subscriber may only call internally and in the local area code. • National: Subscriber may call in own country (country code). • International: Subscriber has no restrictions and may call worldwide.
Internal ringtone	Select a ringtone by which internal calls will be signaled (support by terminal required).
DTMF-mode	Select from the list the manner in which DTMF tones are to be relayed in SIP. If you select "Inband" then G.711 must be selected as the audio codec, as otherwise DTMF will not be possible. The standard setting is Auto.
Protocol	Extension can be external numbers, SIP or IAX2 terminals or a Fax2Mail service. If "External“ is selected, then fixed-line numbers can be stored under Extension Number These must in turn be in national or international format. The external number is routed according to the routing configuration.
Line nr	Input field for the number when "External" is set in the Protocol field (see above). For further information, see INFO at the end of this section.
Passwords	
Password	The password is a freely selectable character string (at least 6 and a maximum of 24 characters), by which access to the DPx user web interface is secured. If it is also used for the unblocking of the telephone, only digits are permissible.
TAPI Password	The TAPI password is a freely-definable string that contains at least 6 and a maximum of 24 characters. It is used to connect a TAPI client to the DPx. The TAPI interface can only be accessed if a valid TAPI connector license is available.
SIP/IAX2 Password	The SIP password is the password that is used by the terminal to log in to the DPx. This password is used for authorization in the web interface or in the configuration of the terminal.

Protocol	*If Fax2Mail is selected as the protocol, it is possible to receive faxes on this extension number and to relay them by e-mail. The faxes are sent to the e-mail addresses stored in the E-mail field as a PDF attachment. Multiple numbers can be set up with the Fax2Mail function. A maximum of two faxes can be received simultaneously.* *The receipt of faxes is only possible via the T.38 or with the aid of the G.711 audio codec. Other codecs do not support receiving.*
Extension No.	*The default value is "no entry" and should always be left blank during setup. This option makes it possible to influence the connection channel. If, for example, an SIP user is set up with the number 1234, then the connection channel is SIP/1234. If the string 9876 is entered under the connection number, then its connection channel is SIP/9876. If a string is entered under the connection number, it is no longer possible for an SIP terminal to register on this account. This enables one, amongst other things, to define two numbers (virtual extensions) that use the same terminal (connection channel). It does, however, seize the call forwarding setting of the number called in each case, even though the same terminal is used.* *In addition it is possible to define users, in which - for example - the protocol is set to external and any external number (without exchange code) is entered as the connection number. The call forwarding settings also come into action in this case. If the external subscriber is engaged, then the call can be diverted accordingly. This subscriber can also be a member of a team.* *If the Fax2Mail function is selected under Protocol, no entry may be made in the port number field.*

7.4.1.2 Options

Click on the **Options** tab.



Options

Ext. behind NAT Gateway	Mark the checkbox, to switch on synchronous RTP. This is useful and necessary in the case of many terminals that are connected behind a NAT gateway. By default this value is "off" and should only be activated in special cases (firewall without SIP inspection).
Phone lock	If the phone lock is activated, only emergency calls can be made. The phone lock can be disabled on the telephone by dialing a code and entering the PIN.
Call number on forwarding	When a permanent ACD is activated, the own number is signaled instead of that of the original caller.
BLF	If the checkbox is selected, it is possible (in the case of suitable terminals) to signal the current status of the terminal on a busy lamp field. If required, this function must also be activated for specific TAPI functions and when the terminal is used for the BMS Contact Centre.

Secondary ringer	If the checkbox is selected, it is possible to register up to two additional SIP terminals with specific accounts for this number. These can be reached at the same number. The additional automatic SIP usernames have the syntax za<extension number> and zb<extension number. The password is the same for all accounts.
Wartefeldagent	Specify which terminal is to be used in the waiting fields. Only SIP and DPx UC terminals are permissible.
UC license	Assign a DPx UC license to the extension, if this DPx UC is to be used.
Busy on Busy	If the checkbox is selected, the callers receive the busy tone when subscribers are busy. This function is helpful when the secondary extension is used simultaneously. It is not supported for external extensions.
Phonebook	If this checkbox is marked, then this extension is listed in the internal phone book - for example in the DPx User book or in the DPx phone book for SIP terminals.
Transfer with #	During a call the option exists of transferring the call by pressing the □ key and then dialing the destination number.
Activate email	Notifications regarding missed calls are sent to the e-mail address stored in the E-mail Address field if this checkbox is marked.
Login main ringer	The main extension can, as an option, be deactivated when the secondary extension is activated. Activation and deactivation is also possible by calling the primary extension on the on the extension's own number.
Conference room	This extension can be assigned to a personal conference room. The subscriber can then manage this conference room in DPx User and DPx Live Conference.
Qualify	If this feature is switched on, a regular check of the connection runtime to the server takes place every 90sec. (Keep alive). If a terminal can no longer be reached, no further calls are assigned to this device automatically, so that there is no time consuming wait for an answer from the device when a call is put through. Activate this feature, if DPx is operated behind a NAT firewall. Not all terminals are able to process these notifications correctly, meaning that this function may have to be deactivated.
Call-limit	By means of this setting it can be determined how many simultaneous conversations may be conducted via this extension. *If one call is selected, it is no longer possible to transfer calls.*
SIP Transport	The DPx supports UDP, TCP as well as TLS as protocol for SIP. UDP is selected as the default. The activation of TLS also activates the encryption of audio data by means of SRTP.

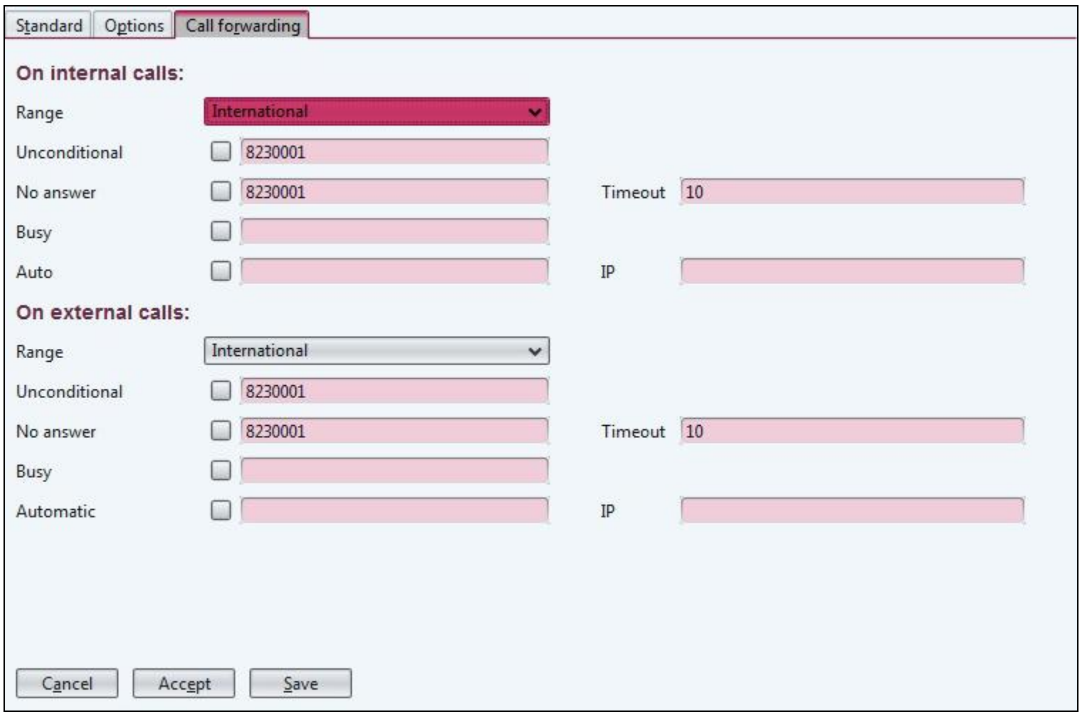
Codecs

1. … 4. Codec	Select the codec to be used for voice coding. The pre-set corresponds to the current setting.

DPx supports the G.711a, G.711u, G.722, G.726, GSM and Opus codecs for the extensions. Additionally, the H.261, H.263, H.263p, H.264 and VP8 video codecs can be used. Transcoding does not take place between various video codecs

7.4.1.3 Call forwarding

Call forwarding is the final tab for a new extension.



In the event of internal calls / In the event of external calls

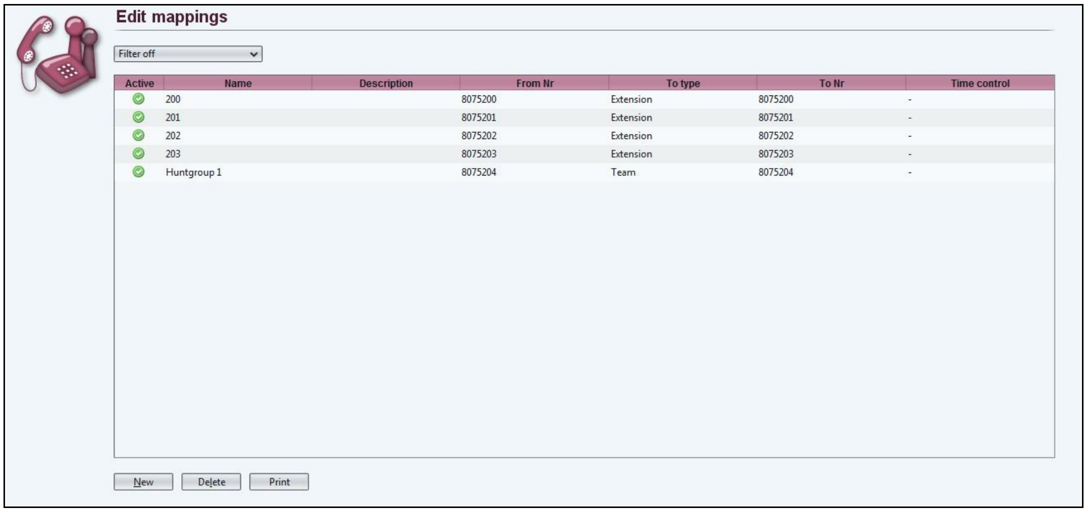
Range	Specify what rights the extensions should have to set up call forwarding (internal, local, national, international).
Unconditional	Activate the checkbox and enter a number if to set up a permanent call forwarding setting. If a permanent call forwarding set is activated, then call forwarding no longer functions in the event of no answer and busy.
No answer	Activate the checkbox and enter a number if to set up a permanent call forwarding setting in the event of **No answer**.
Timeout	Enter the time (in seconds), after which automatic call forwarding takes place in the event of **No answer**.
Busy	Activate the checkbox and enter a number if to set up call forwarding **When Busy**.

Auto	If the checkbox is marked, the system checks when a call is made to this extension whether a PC or other device is available at the specified IP address. If no device is detected, the call is diverted to the destination phone. This is of particular interest with regard to soft phones (e.g. in a call centre). Calls are only connected if the PC is ready to operate. In this way a check can also be made as to whether a workstation (phone) is in fact busy and it enables an automatic diversion to a defined number, if the workstation is switched off. This type of Automatic Call Forward has the highest priority and is evaluated ahead of all others.
IP	Enter an IP-address for automatic Automatic Call Forward.

To save the information, click on **Save** on the lower bar.

7.4.2. Mappings

The allocation of numbers (mapping) is edited in this menu. In doing so, the external public numbers are assigned to the internal numbers.

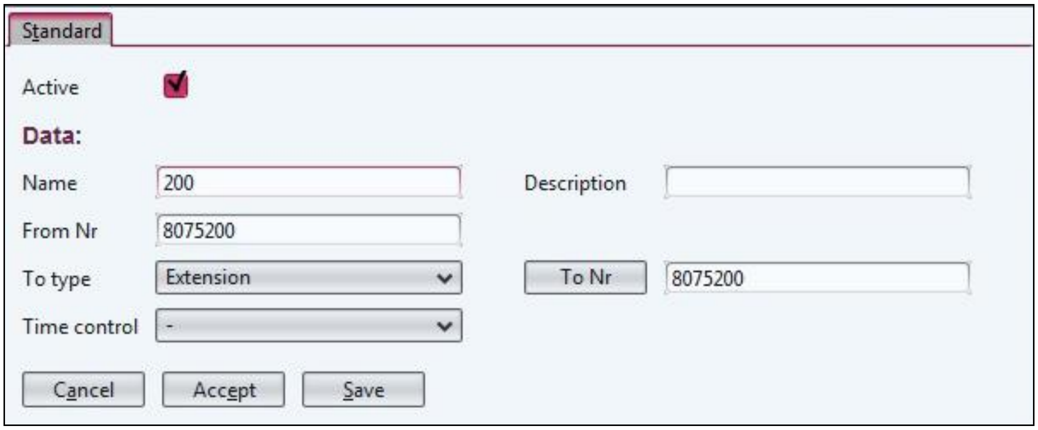


The allocation that has been entered is listed. These are opened for editing by double-clicking on the respective entry.
A new entry can be created by clicking on **New**

The mapping tool is a convenient way and means of allocating telephone numbers. Inbound call (external) can be assigned to an internal number. A number can also be inbound at a SIP provider. By means of this function a number of a SIP provider can be assigned to an internal number. In this way, multiple SIP gateways can also be assigned directly to the respective subscribers. In addition to the subscribers, an entire team can be dialed.
Even when no mapping has been added, a series of specified mappings is listed. All subscribers who have already been set up in the user area, are assigned on a 1:1 basis automatically. A subscriber with extension 258 is assigned automatically by the system in such a way that an external call on the international number will also reach extension 258. This standard mapping facilitates setting up and can be changed at any time.

7.4.2.1 Standard

When a new Mapping is selected the **Standard** tab is shown.

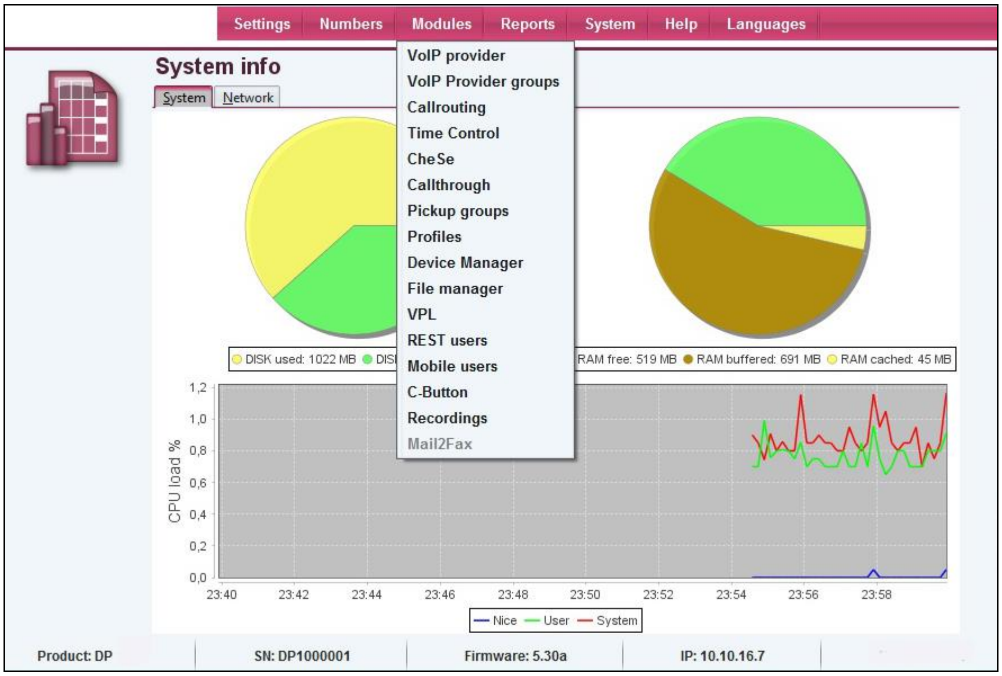


Active	Mark the checkbox, if the entry is to be active.
Data	
Name	Enter a name for the entry, e.g. the location
Description	Enter a description if necessary.
From Nr	If required, enter a particular number, which is to be assigned to a particular destination. As rule, a number is stored here in the international format without the 00 prefix. Numbers can only be processed correctly in this format. The * symbol is permitted at the end of a From Nr and indicates that any digits may follow the preceding number.
To type	Select the type of destination (extension, conference room, mailbox …)
To Nr	Enter the number of the destination. By clicking on the To-No., a selection list with possible destination number is opened. A destination can be assigned directly by a double-click within the selection list.
Time control	If time control is configured, it can be assigned to each mapping entry. Within time control, specific destinations can be assigned for various time frames. If an inbound call takes place outside of the defined time frame, the standard destination takes the call new destinations can be controlled centrally by time control.

To save the information, click on **Save** on the lower bar.

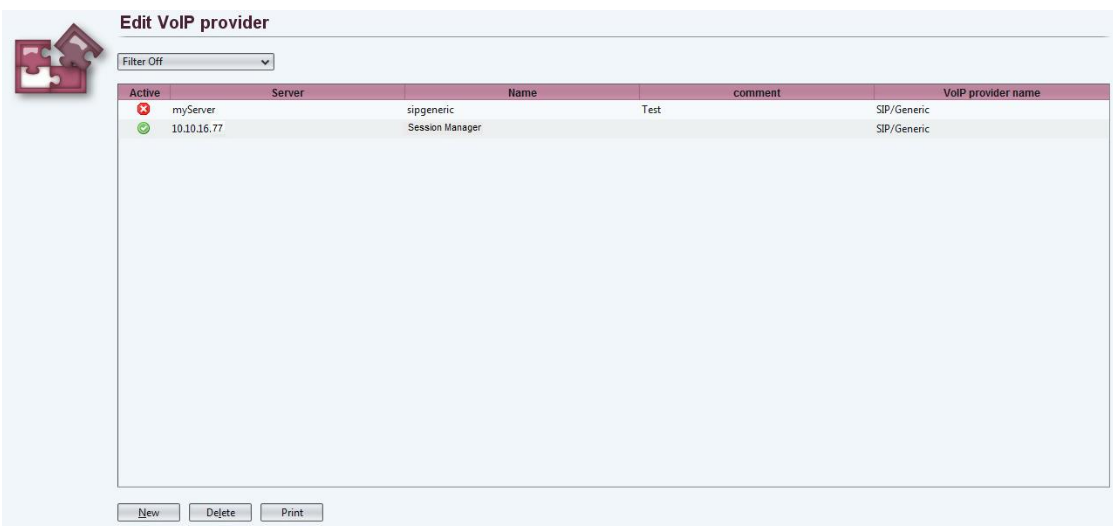
7.5. Modules

In this menu you will find the settings for Callrouting and terminal management as well as all settings for special functions of the DPx.



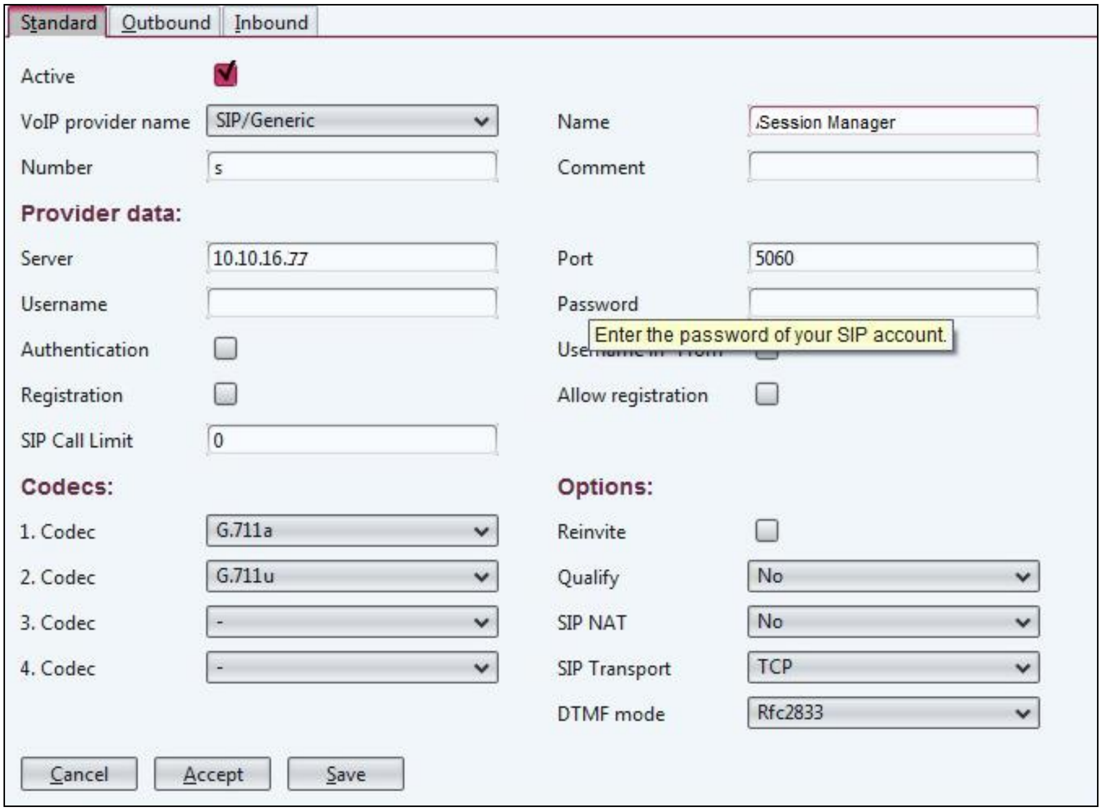
7.5.1. VoIP provider

The providers that have already been set up are listed. These are opened for editing by means of a double-click on the respective entry.



New providers can be created by clicking on **New**.

7.5.1.1 Standard

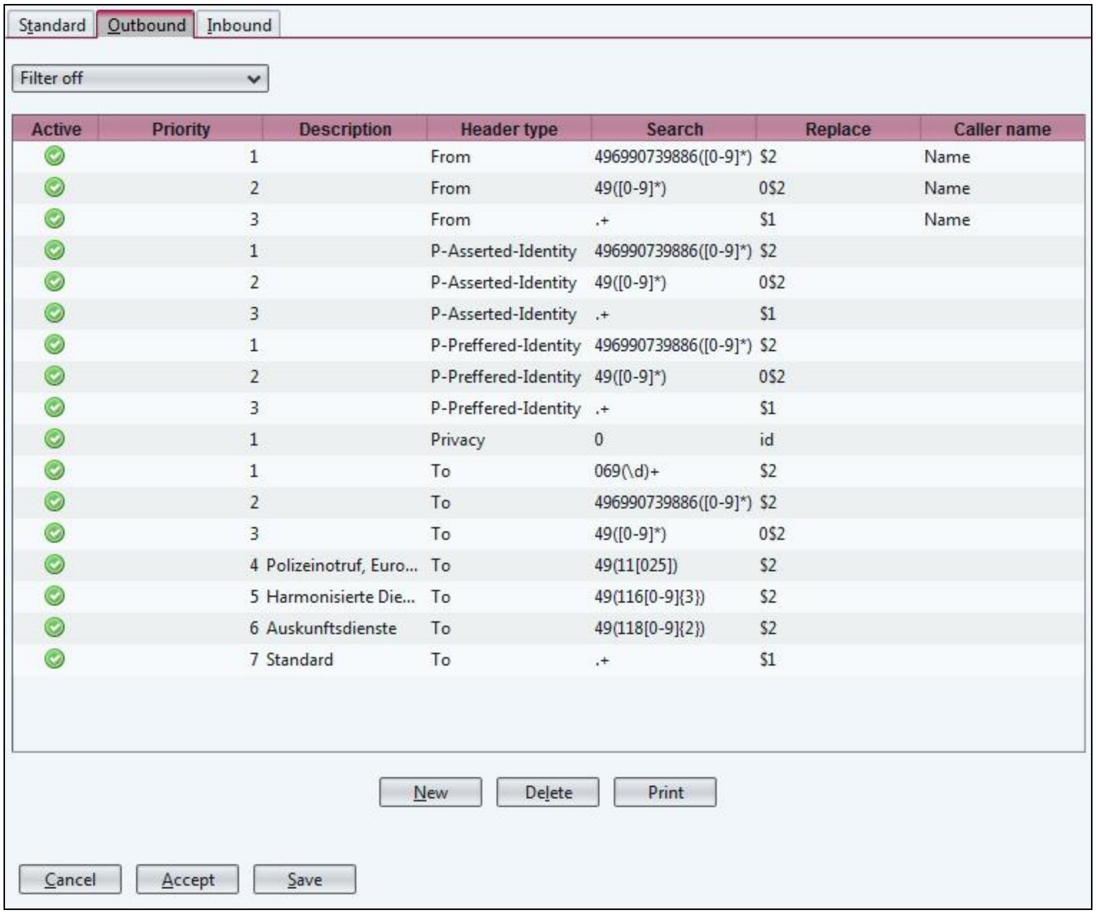


Active	Mark the checkbox, if this provider is to be active.
VoIP provider name	Select an entry from the list.
Name	Give a unique name to this provider. This name, in conjunction with the prefix "prov" is also used for the optional authentication of the provider to the PBX.
Number	A number assigned by the provider is stored here. In most cases the standard entry s is correct.
Comment	Enter a comment for this entry if necessary.
Provider Data	
Server	Enter the address of the server or of the media gateway.
Port	Enter the port. For SIP providers or media gateways this is usually Port 5060. Port 50601 is usually used in the case of SIP TLS being used.
Username	User name for provider login
Password	Password for provider login
Authentication	Mark the checkbox, if to request authentication of the provider in the case of inbound calls.
Registration	Registers the respective user with the VoIP service provider. This is necessary for most service providers in order that inbound calls can be connected.

SIP Call Limit	Specify how many simultaneous calls via this VoIP provider are to be allowed. The value 0 cancels a limit.
Username in "From"	This option places the user name in the From-header in SIP protocol instead of the number.
Allow registration	This option permits a terminal - for example, a media gateway - to be registered on the DPx.
Codecs	
Codecs 1. … 4.	Codec selection of the voice coding in use. The pre-set corresponds to the current setting.
Options	
Reinvite	The Reinvite option makes it possible to swap speech data (RTP) directly between the terminals. In this case, the DPx serves only as an SIP signaling gateway a compatible network infrastructure is required for this option. If required, contact a specialist dealer.
Qualify	If the checkbox is marked, a regular check of the connection runtime takes place.
SIP NAT	Select "Yes" if the system is used behind a firewall when connected to SIP providers.
SIP Transport	Select from UDP, TCP or TLS as a transfer method. The correct setting can be obtained from a VoIP provider or from the documentation of the media gateway. When TLS is selected, the encryption of audio data by means of SRTP is also activated.
DTMF mode	Select here whether DTFM signals are to be transferred.

7.5.1.2 Outbound

In this screen, the various SIP headers can be adapted to the respective VoIP service providers or the respective media gateway with the aid of POSIX-compliant, regular expressions.

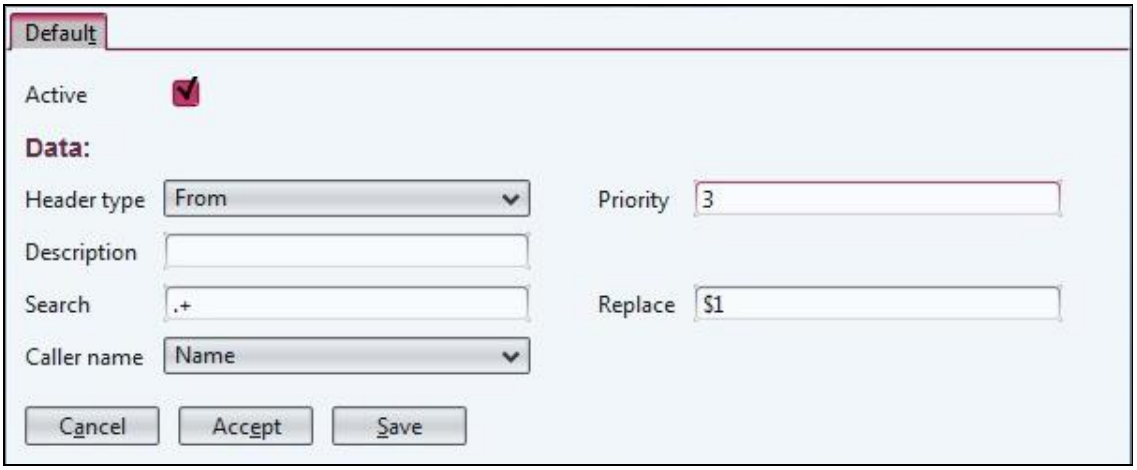


Active	Priority	Description	Header type	Search	Replace	Caller name
	1		From	496990739886([0-9]*)	$2	Name
	2		From	49([0-9]*)	0$2	Name
	3		From	.+	$1	Name
	1		P-Asserted-Identity	496990739886([0-9]*)	$2	
	2		P-Asserted-Identity	49([0-9]*)	0$2	
	3		P-Asserted-Identity	.+	$1	
	1		P-Preffered-Identity	496990739886([0-9]*)	$2	
	2		P-Preffered-Identity	49([0-9]*)	0$2	
	3		P-Preffered-Identity	.+	$1	
	1		Privacy	0	id	
	1		To	069(\d)+	$2	
	2		To	496990739886([0-9]*)	$2	
	3		To	49([0-9]*)	0$2	
	4	Polizeinotruf, Euro...	To	49(11[025])	$2	
	5	Harmonisierte Die...	To	49(116[0-9]{3})	$2	
	6	Auskunftsdienste	To	49(118[0-9]{2})	$2	
	7	Standard	To	.+	$1	

The outbound rules that have already been set up are listed. These are opened for editing by double-clicking on the respective entry.

Before changing the rules under the Outbound tab, the newly created VoIP service provider must be stored and re-opened. Depending on the selected profile, a standard set of rules then becomes available.

Click on **New** to set up a new rule. Complete the fields carefully.



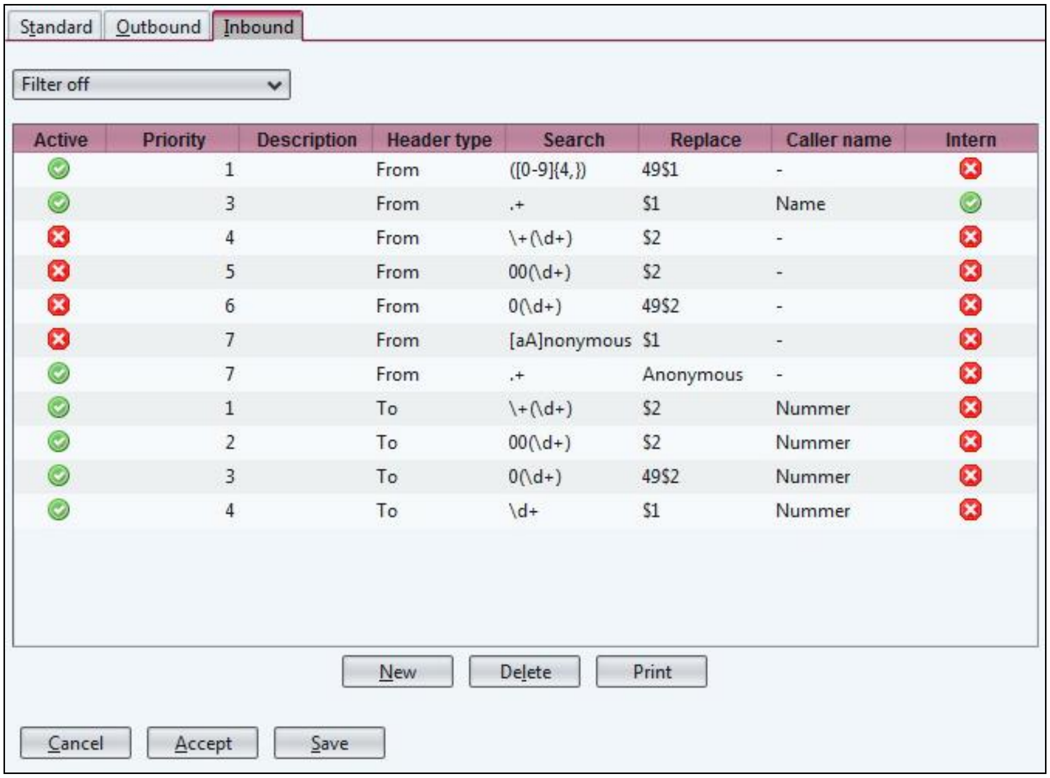
Active	Mark the checkbox, if this rule is to be active.
Data	
Header type	Select the SIP-Header to be set.
Priority	In the event of there being multiple rules for the same header. The header can be prioritized.
Description	Enter a description for this entry if necessary.
Search	A search for the destination or source number can be made with the aid of regular expressions, depending on the selected header type.
Replace	The source number or the destination number can be modified for the selected header type.

Some examples for regular expressions.

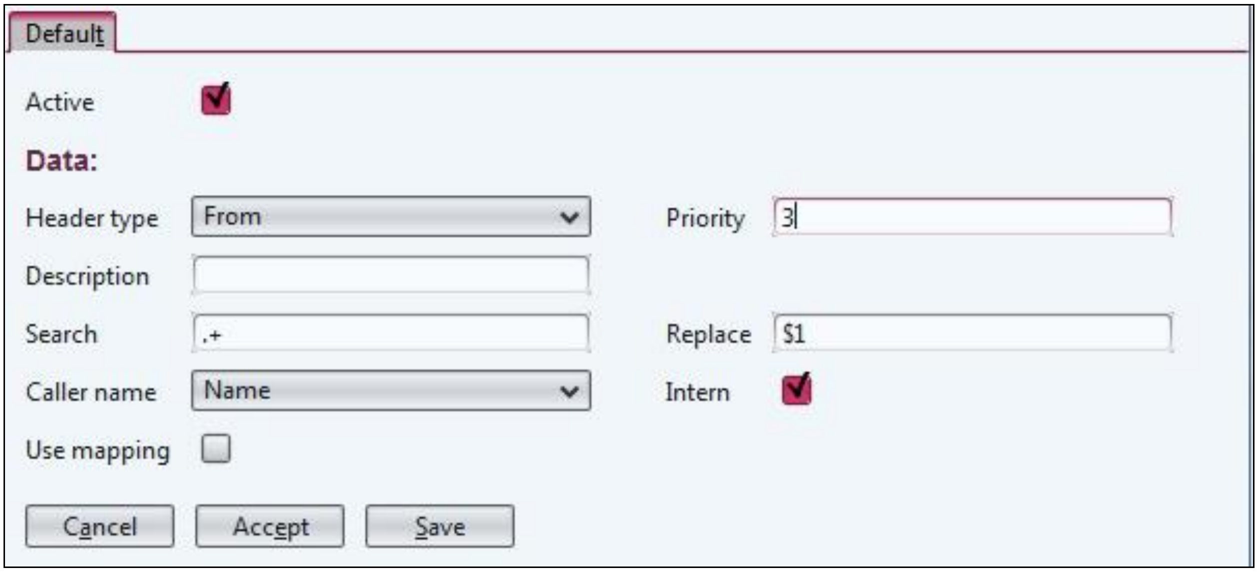
.+	*One or more of any characters*
*.**	*No or any characters, as desired*
{3}	*precisely 3 characters of the prefixed expression*
{2.4}	*precisely 4 characters of the prefixed expression*
\d	*Digits from 0 to 9*
[0,-9]	*Digits from 0 to 9*
[a-z]	*alphabetic characters from a to z*
[0,4,7]	*Digits 0, 4 and 7*
\+	*The + sign*

7.5.1.3 Inbound

In this screen, the various SIP headers can be interpreted as destination or source numbers with the aid of POSIX-compliant, regular expressions.



The inbound rules that have already been set up are listed. These are opened for editing by double-clicking on the respective entry.
Before changing the rules under the Inbound tab, the newly created VoIP provider must be stored and re-opened. Depending on the selected profile, a standard set of rules then becomes available.
Click on **New** to set up a new rule.



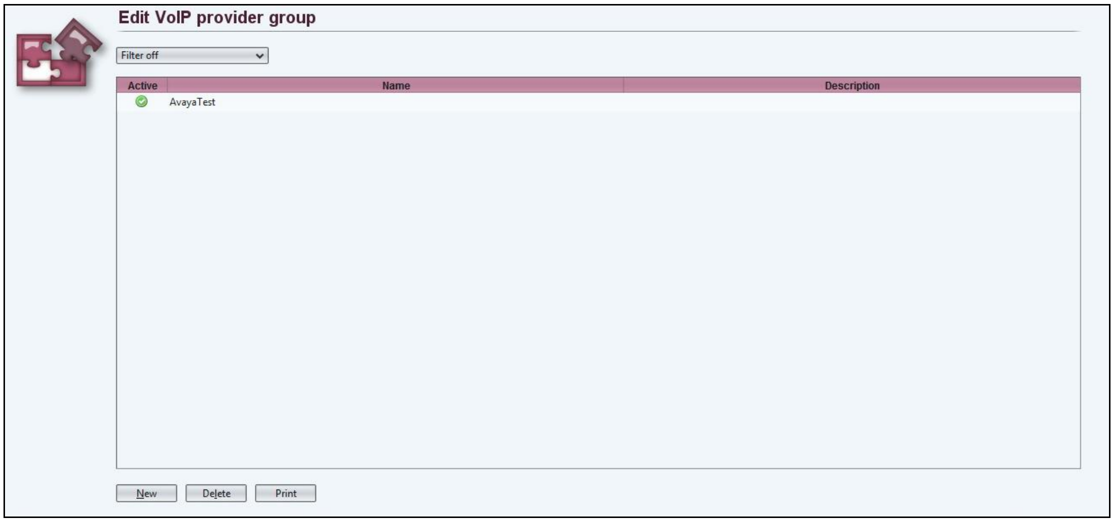
Active	Mark the checkbox, if this rule is to be active.
Data	
Header type	Select the SIP header to be analyzed.
Priority	In the event of there being multiple rules for the same header. The header can be prioritized.
Description	Enter a description for this entry if necessary.
Search	A search for the destination or source number can be made with the aid of regular expressions, depending on the selected header type.
Replace	The source number or the destination number can be modified for the selected header type.
Caller name	Specify if the caller name is to be taken from the corresponding SIP header (Name), or replaced by the number of the caller (Number) or not placed at all.
Intern	If the DPx is operated as a sub-system it is then possible to mask inbound calls via this VoIP service provider as internal calls. These calls are then not made via the mapping but rather directly via the internal routing to the number dialed.

Likewise, calls masked as internal are not captured by time control. Masking only takes place if the stored expression in the search field matches the header for which the search is being made.

To save the information, click on **Save** on the lower bar.

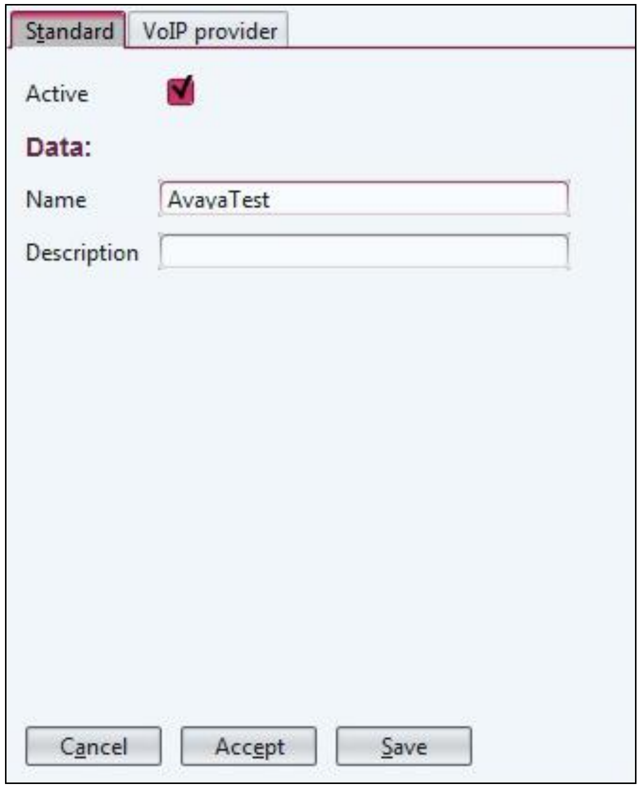
7.5.2. VoIP provider group

You can manage the provider groups in this menu. The provider groups combine one or many VoIP providers into a single virtual VoIP service provider. The VoIP service provider groups form the basis for callrouting.



The providers that have already been set up are listed. These are opened for editing by means of a double-click on the respective entry.
New providers can be created by clicking on **New**.

7.5.2.1 Standard



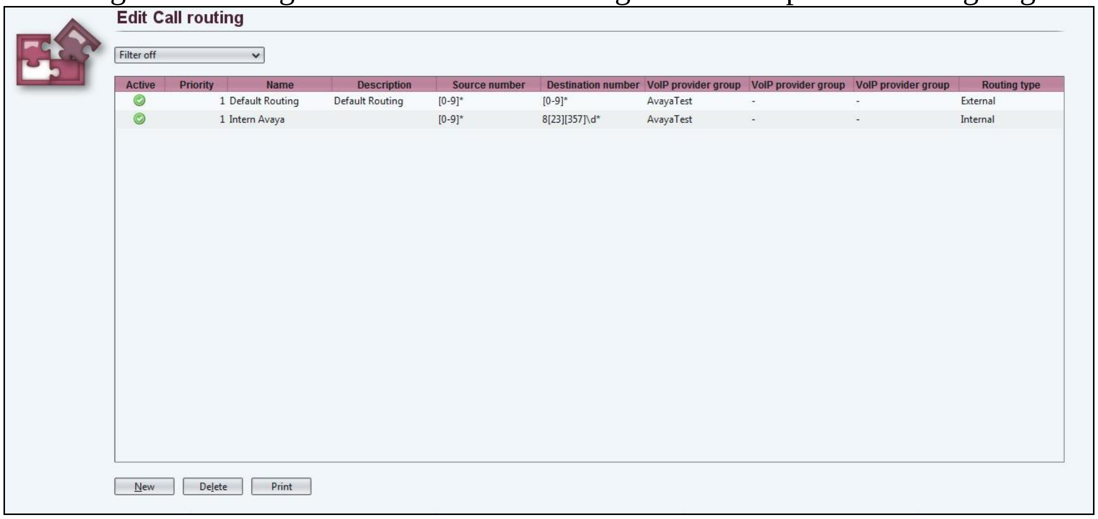
Active	Mark the checkbox, if this provider is to be active.
Data	
Name	Give a unique name to this provider group. This name serves as a reference in call routing.
Description	Enter a description for this entry if necessary.

7.5.2.2 VoIP provider

By marking the **Select** checkbox, select which VoIP providers should belong to the provider group.
To save the information, click on **Save** on the lower bar.

7.5.3. Callrouting

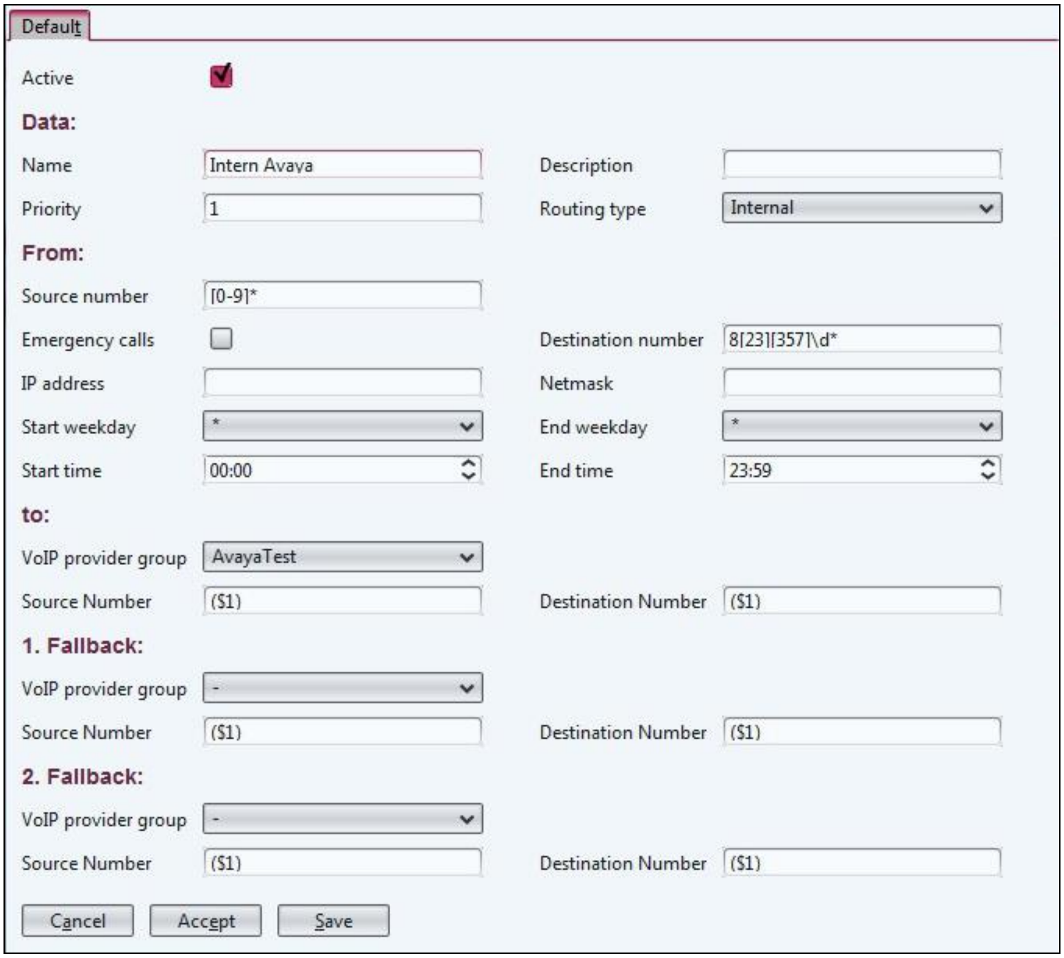
You can manage call routing in this menu. Call routing defines the path of an outgoing call.



The call routings that have already been set up are listed. These are opened for editing by double-clicking on the respective entry.

A new call routing can be created by clicking on **New**.

7.5.3.1 Default



Active	Mark the checkbox, if this routing rule is to be active.
Data	
Name	Give a unique name to this routing rule.
Description	Enter a description for this entry if necessary.
Priority	Enter the priority of this routing rule. In the event that multiple rules satisfy the same conditions, then the rule with the highest priority, therefore the lowest priority number, is selected.
Routing type	If the DPx is operated as a sub-system, then individual internal number or ranges of numbers can be directed to the primary DPx via an SIP trunk. In this case the routing type is to be set as Internal. The External setting must be selected for normal routing to a media gateway or a VoIP provider.

From

Source number	The source of a routing is always an internal number of an extension. Groups of extensions can be summarized with the aid of regular printouts to POSIX. Example 1([0-9]*): All internal extensions, whose numbers begin with 1. Bracketing in round brackets is important, if one wishes to use this part of the number to change the number at a later stage.
Emergency calls	Mark the checkbox, if this routing rule is to apply to all emergency numbers defined in the emergency number list. In doing so, the configuration in the destination number field is not taken into consideration.
Destination number	International numbers in the E.164 format can be stored as destinations. The routing rule is then valid for the destination numbers stored here.
IP Address	Enter an IP address if this routing is to be valid only for terminals from a particular IP network. Otherwise this field remains empty.
Netmask	Enter the corresponding network screen.
Weekday from/to	Select the range of weekdays.
Time from/to	Specify the time range for which this rule is to be valid.

To

VoIP provider group	Select the VoIP service provider group via which the calls are to be routed.
Source Number	To change the source number, the relevant modifications can be stored here. ($1) denotes the entire number from the above-mentioned source number entry. If no changes are made, then the CLIP number stored for the relevant extension is taken over, or else the internal number The individual bracketed areas of the above-mentioned source number are numbered in turn and are available with ($2), ($3), … Example: The extension has number 123 and is to be changed to number 98713. To do so, enter 987($1). The result of the source number must always be a number in the E.164 format in order to guarantee correct routing.
Destination Number	The destination number can be modified in a manner analogous to that of the source number. The standard input is ($1).

Fallback 1. and 2.

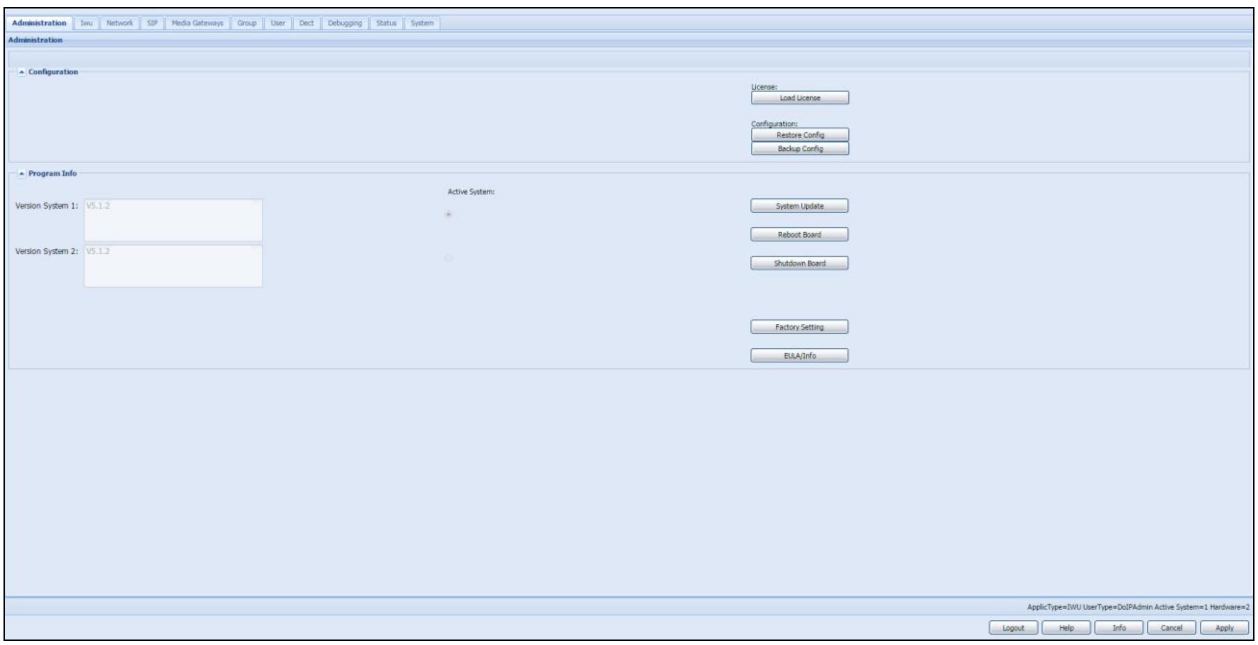
VoIP provider group	If the specified VoIP service provider group cannot be reached for technical reasons, then a fallback can be defined for routing via an alternative VoIP provider group.
Source Number	The source number can be modified.
Destination Number	The destination number can be modified.

To save the information, click on [**Save**] on the lower bar.

8. Configure Funktel f.airnet DECT

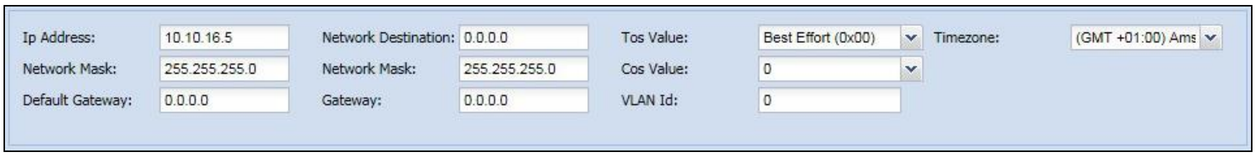
8.1. Configure a DECT Controller

The DECT Controller is accessible via its factory default IP address 192.168.2.1. To access the controller Web Based Management(WBM) you have to configure an IP address in the network 92.168.2.0/255.255.255.0, e.g. 192.168.2.101 on the maintenance PC. If the IP address 192.168.2.1 is already used in your network, directly connect the maintenance PC and the DECT Controller via Ethernet. This may be accomplished by using a direct Ethernet cable or via a separate Ethernet switch where only the maintenance PC and the DECT Controller are connected.
Test via ping, if the DECT Controller is replying to the ping requests (ping 192.168.2.1). If not, check all cabling, switch settings, (e.g. VLAN configuration). Ensure that the configured local IP of the maintenance PC address is up (e.g. ping 192.168.2.101).
Start the Web browser at the maintenance PC.
Access the WBM at the following URL: http://192.168.2.1
Log in to the WBM with appropriate credentials:
Press the **Login** button (not shown). The configuration page for the IP-DECT System Server appears.



8.2. Configuration of VoIP (Infrastructure) Network

Since configuration of the IP VoIP (Infrastructure) Network settings requires a reboot of the DECT Controller, these settings are configured initially.
To allow direct IP communication between the DECT Controller and the PBX both devices have to be located in the same IP network. Therefore it is necessary to adapt the IP address of the DECT Controller to the network of the VoIP (Infrastructure). You need at least one unused IP address of the Infrastructure network, which has to be configured at the DECT Controller.
Select the **configuration page Network** (not shown)**.**
Change the configuration in the bottom frame to the designated values of the DECT Controller (not shown).



Change the following values as needed:

IP address

Configure the IP address at which the DECT Controller should be available inside the VoIP (Infrastructure) Network.
This configuration example uses ip address 192.168.100.1.

Network mask

Enter the corresponding netmask for the IP address as configured above. (Default for Class C networks: 255.255.255.0). This configuration example uses the network mask 255.255.255.0.

Routing Configuration

If routing to another network is necessary (e.g. access from Maintenance PC to IP VoIP (Infrastructure) network) or if infrastructure components (e.g. PBX, NTP servers) are located behind other routers, routing may be configured using a **Default Gateway** or by a specific network route (**Network destination**).
The usage of a **Default gateway** is the recommended routing method.
Using the method **Network destination**, the values for **Network destination**, **Network mask** and **Gateway** have to be configured. At the Maintenance PC a corresponding route has to be configured. This configuration example uses the Default Gateway IP address 192.168.100.83.

Time Server IP

The time settings have to be set according a NTP (or SNTP) time server. After activating the NTP Server, it may take some minutes upon activation of the NTP service. This is due to the nature of time synchronization between NTP server and NTP client.

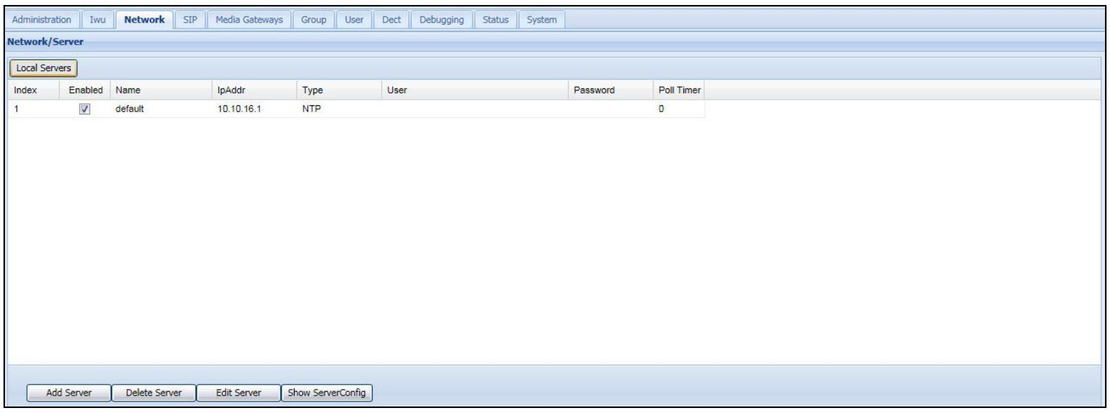
The Server has a built-in hardware clock. However, usage of an accurate time by using NTP is suggested. This configuration example uses the NTP Server ip address 192.168.201.94.

Set the Time zone to an appropriate zone (e.g. " (GMT +01:00) Amsterdam, Berlin, Rome, Stockholm, Vienna").

Switch to configuration page **Network → Local Servers.**

Add an NTP server entry by clicking on **Add Server**

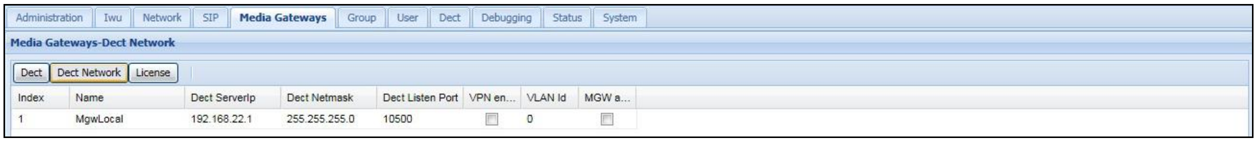
Enable the new entry, configure the ip address of the NTP Server in filed **IpAddr**, select **NTP** as Type. All other filed in this row may be left unchanged.



***Important:** Ensure that the **ip configuration** is configured **correctly**. Otherwise - after rebooting the DECT Controller - it may not be accessible without resetting it to its factory defaults (which have a fixed IP setting of 192.168.1.1 or 192.168.2.1).*

8.3. Configuration of DECT Network

Since configuration of the DECT Network settings requires a reboot of the DECT Controller, the settings are configured before the DECT Controller hardware is attached to the designated network segment.
Select configuration page **Media Gateways → Dect Network**.



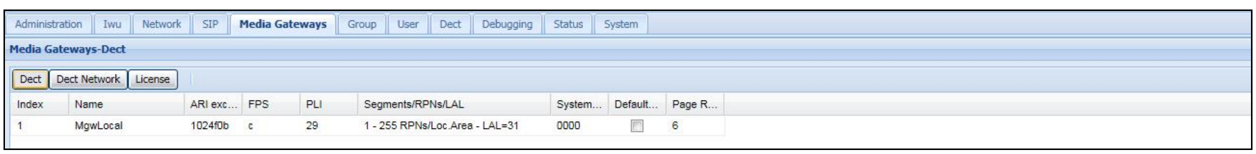
Change the configuration of entry **MgwLocal** in the table to the designated values of the DECT Controller.

Dect ServerIp

This field contains the IP address of the server (the DECT Controller) in the DECT network. It is used for communication between all DECT base stations and the DECT Controller. This configuration example uses the Server IP address 192.168.10.1.

Dect Netmask

In this field the corresponding network mask which is assigned to the Server Ip address is configured. (The default value for a Class-C network is 255.255.255.0). This configuration example uses the network mask 255.255.255.0.
Select configuration page **Media Gateways → Dect**.

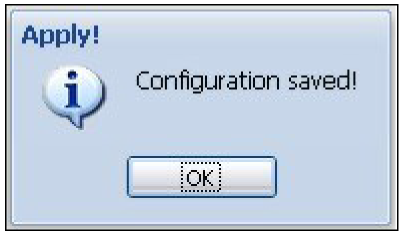


ARI excl FPS, FPS

In the field **ARI excl FPS** and **FPS** the System ARI (DECT ID) which has to be unique at each DECT system is configured. The System Ari is provided by the license dongle. Supported System Ari classes are Class B Ari.
Hint: All handset registrations are bound to a specific System ARI. If the System ARI is changed, all handsets lose their registration at the DECT Controller. To achieve system functionality, the handsets have to be registered again at the DECT Controller.
ARI excl FPS = **101b2ff**
FPS = **7**
This configuration example uses the System ARI 101b2ff7.

System Pin

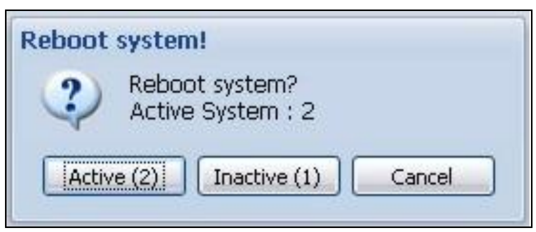
The **PIN** is a 8-digit number and it is needed for the registration of handsets. It is preconfigured with "00000000" and may be configured system wide here. Change the system pin to another decimal value if required. This configuration example uses the System Pin **1234**.
Apply the changes by clicking the **Apply** button(not shown) at the bottom section.



As stated above, for the changes to become active, the DECT Controller has to be rebooted.

Important: *After rebooting, the DECT Controller will not be accessible by its the IP address 192.168.2.1 anymore. Instead it is accessible by the IP address of the IP VoIP (Infrastructure) Network (configured at the step above). If you have attached the DECT Controller directly via an Ethernet cable you have to attach the DECT Controller physically to the designated network segment* ***after rebooting*** *it.*

Select the configuration page **Administration**(not shown).
Initiate the reboot by clicking on the **left button** which is labeled [**Active (x)**]. The "x" is a placeholder for the active partition number, in this example x=2.



Start the reboot process by clicking on button **Active (2)**, if the current active partition is System 2 (as this example used - see output of **Active System:**). If the active system of your DECT Controller is 1, then click the **left button** which is labeled **Active (1)**.
Wait about 2 minutes for the DECT Controller to come up again.
Access the WBM (Web based management) by the IP address you have configured before for the IP VoIP (Infrastructure) Network.

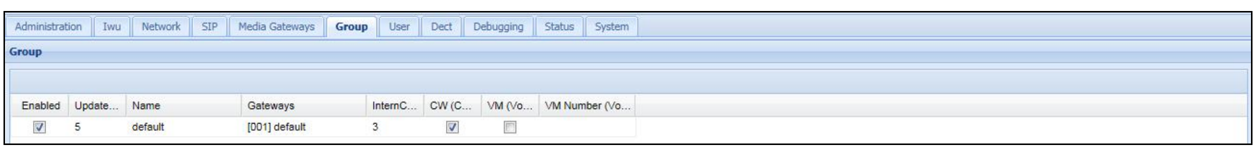
8.4. Configuration of Users at the PBX

It is assumed that the VoIP users at the PBX are already configured. Configuration of Users at the DECT Controller.

8.5. Gateway and Group

Access the WBM of the DECT Controller via the web browser at the maintenance PC.
Example: **http://192.168.100.10**
Log in to the WBM with the appropriate credentials.
Switch to configuration page **"SIP → General"**.
Add a new gateway entry by clicking on the button **Add Gateway**(not shown).

Change the following values:
Gateway Type
The **Gateway Type** is set to "Default".
SIP Server Id and Netmask
Change the preconfigured **SIP Server Id** from 0.0.0.0 to the IP address of the used **PBX** as well as the corresponding **Netmask**. This configuration example uses the IP address 192.168.100.50 and the network mask 255.255.255.0.
Switch to configuration page tab **Group**
Groups are the connecting link between Gateways and VoIP Users. A User is assigned to a Group and a Group is assigned to a Gateway.
Add a new group by clicking on the button **Add Group**.



Name
Change the preconfigured name of the Group (e.g. to the name of the corresponding PBX). This name is only used for the internal configuration of the DECT Controller Software. This configuration example uses the default Group name "default".

Gateways
Select the gateway from the dropdown field which you have configured in the last step.

InternCallLength

With this setting the maximum number of digits of the calling Party number for internal calls is configured. Calls with a larger number of digits are signaled as external calls at the handset. The default value is "3".

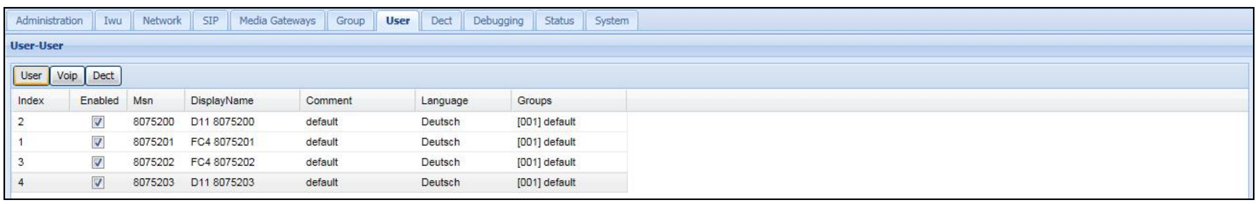
Note: *Alternatively it is possible to register several users at once using the Bulk Registration Mode (see "Multi-Register (Bulk Registering) of Handsets"). Since this method is out of scope of a quick start, the manual method is used here.*

Switch to configuration page **User**, sub page **User** (User → User).

- Set up one or several **Users** according the user configuration at the **PBX** for the connection with the DECT Controller. Please take care of the consistency of the entries between the PBX and the DECT Controller.

Add a new user by clicking on the button **Add User**.

Note: To add a new user entry (even during running system services) it is necessary to select an existing user (otherwise an error message will appear) and then click on button [Add User]. A new entry with default values is inserted above the selected user entry..



Change the contents of the following fields:

MSN (necessary)

The MSN has to correlate with the Call number of the User at the PBX. This configuration example uses MSN 8075200, 8075201, 8075202 and 8075203.

DisplayName (necessary)

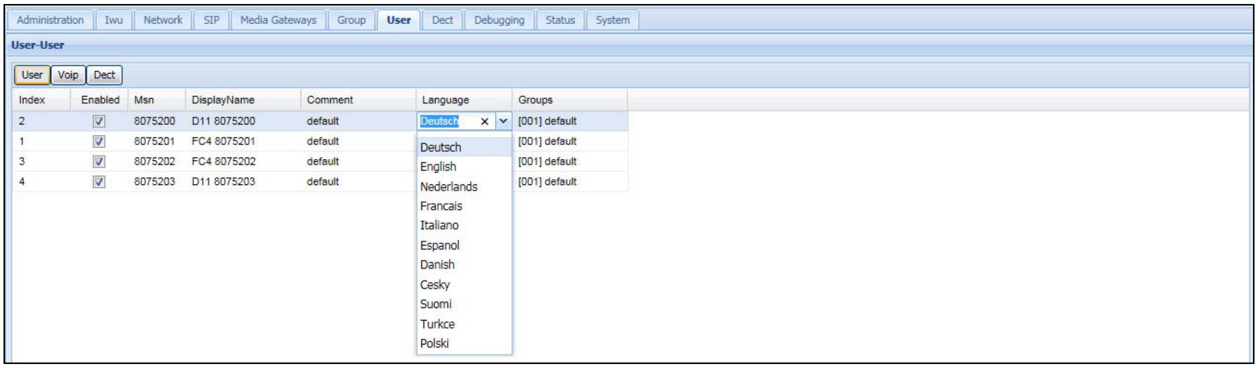
This information is shown at the idle display of the corresponding handset. This configuration example uses MSN D11 8075200, FC4 8075201, FC4 8075202 and D11 8075203.

Comment (optional)

Enter any desired text for administration purposes e.g. "Sales", "Marketing" and "Support". This configuration example uses the values "default".

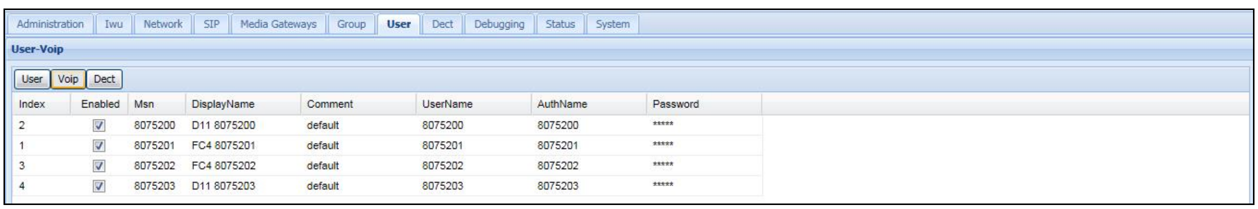
Language

The language used for display messages of the handset can is selected here. This configuration example uses language "Deutsch". The supported languages are shown in the picture.



Groups

Choose a Group (and with that a Gateway) from the dropdown box to which the user is associated to. This configuration example uses the default Group name "default".
Switch to configuration page **User**, sub page **Voip** (**User → Voip**).



Change the contents of the following fields:

UserName (necessary)

Configure the Name or Number for the registration of the User at the PBX. This configuration example uses the UserName 8075200, 8075201, 8075202 and 8075203.

AuthName (optional)

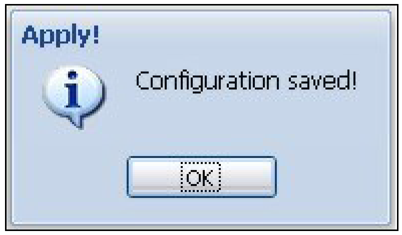
Configure the AuthName which is used for the authentication at the PBX (together with "Password"). This configuration example uses the AuthName 8075200, 8075201, 8075202 and 8075203.

Password

Optional, but necessary if an **AuthName** is configured. The password which is used for the authentication at the PBX (together with **AuthName**). This configuration example uses the Passwords 0000 but shown as ****.

Note: *The entries in the password field are visible only at time of adding or overwriting a password. After applying the changes, the password fields are masked out and not visible anymore.*

Apply the changes by clicking the Apply button(not shown) at the bottom section.



Confirm the message box by clicking on **OK**.
To append further users to the configuration, repeat the steps above.

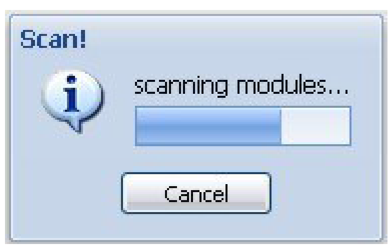
8.6. Configure DECT-FB4 IP Base Stations to the System

Attach at least two DECT-FB4 IP base stations to a PoE port of the network switch of the DECT network. If you use a standard port of the network switch without PoE, use a separate power supply.
Inside this configuration example the first DECT-FB4 IP base station is configured as the synchronization master for over-air synchronization. The second DECT-FB4 IP base station and all further base stations are configured as synchronization slave.
Wait about 2 minutes until the LED states at the DECT-FB4 IP base stations change to permanently red.

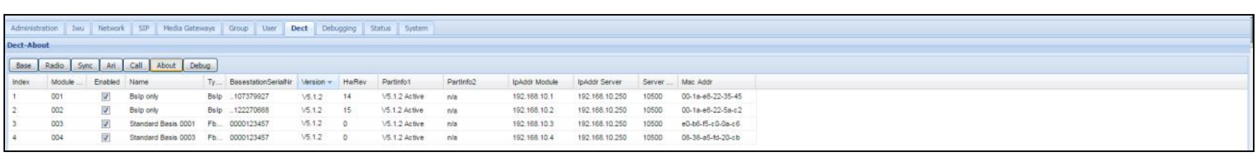
DECT About

At the WBM of the DECT Controller switch to configuration page **Dect-About**(not sohwn).

- To scan the newly attached second DECT-FB4 IP click on the button **Scan**(not shown).



The newly attached DECT FB4 IP should be found automatically and a record will be appended for it in the table of DECT devices.



Index	Module ...	Enabled	Name	Ty...	BasestationSerialNr	Version	HwRev	PartInfo1	PartInfo2	IpAddr Module	IpAddr Server	Server ...	Mac Addr
1	001	☑	Bsip only	Bsip	..107379927	V5.1.2	14	V5.1.2 Active	n/a	192.168.10.1	192.168.10.250	10500	00-1a-e8-22-35-45
2	002	☑	Bsip only	Bsip	..122270668	V5.1.2	15	V5.1.2 Active	n/a	192.168.10.2	192.168.10.250	10500	00-1a-e8-22-5a-c2
3	003	☑	Standard Basis 0001	Fb...	0000123457	V5.1.2	0	V5.1.2 Active	n/a	192.168.10.3	192.168.10.250	10500	e0-b6-f5-c0-0a-c6
4	004	☑	Standard Basis 0003	Fb...	0000123457	V5.1.2	0	V5.1.2 Active	n/a	192.168.10.4	192.168.10.250	10500	08-38-a5-fd-20-cb

DECT Base

At the WBM of the DECT Controller switch to configuration page **Dect-Base**.
The DECT-FB4 IP base station has to be configured for usage within the IP-DECT System.
Select the newly created entry for DECT-FB4 IP base station and change the contents of the following fields:

Enabled (necessary)

Set to Enabled for usage within the IP-DECT System.

Name (necessary)

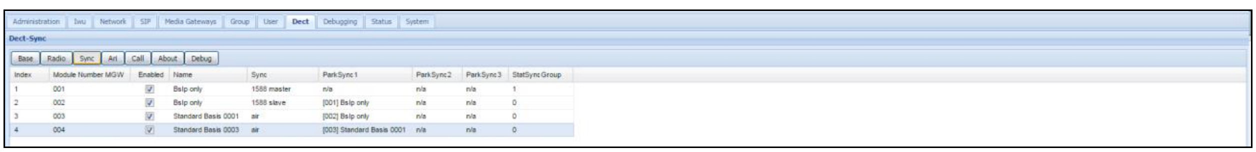
Configure a descriptive name for the DECT base station. Change the preconfigured name of the DECT base station (e.g. to the name of the physical location it is designated for). This name is only used for the internal configuration of the base station.

IpAddrModule (necessary)

Configure an IP address for this DECT RBS inside the DECT network. Hint: The IP-DECT System-System automatically suggests a valid IP address.

DECT Sync

At the WBM of the DECT Controller switch to configuration page **Dect-Sync**.



Select the newly created entry for DECT Controller and change the contents of the following fields:

Sync (necessary)

Set **Sync** to **air** to synchronize the actual base station to another base station via air.

ParkSync1 (necessary)

Select from the dropdown to which base station the selected base station should be synchronized to. This configuration example uses Module **[001] BSIPIWU Local** as the sync master.

DECT ARI

At the WBM of the DECT Controller switch to configuration page **Dect-Ari**.

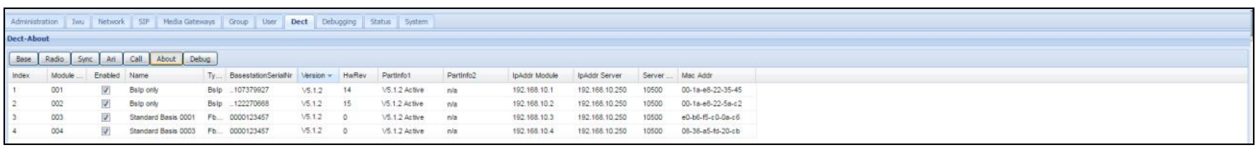
Select the newly created entry for base station and change the contents of the following fields:

Segment Relative Index

Using this dropdown box, the DECT module has to be assigned a relative Index related to the selected location segment inside the MGW. The combination of **Segment** and **Segment Relative Index** will be calculated by the WBM to the resulting RPN.
Note: The IP-DECT System-System automatically suggests a valid RPN Segment Relative Index.
Note*: When using several base stations they have to be configured with a unique "Segment Relative Index" different from "0". Using the same Segment Relative Index as the DECT module number (Index) is very feasible.*

DECT About

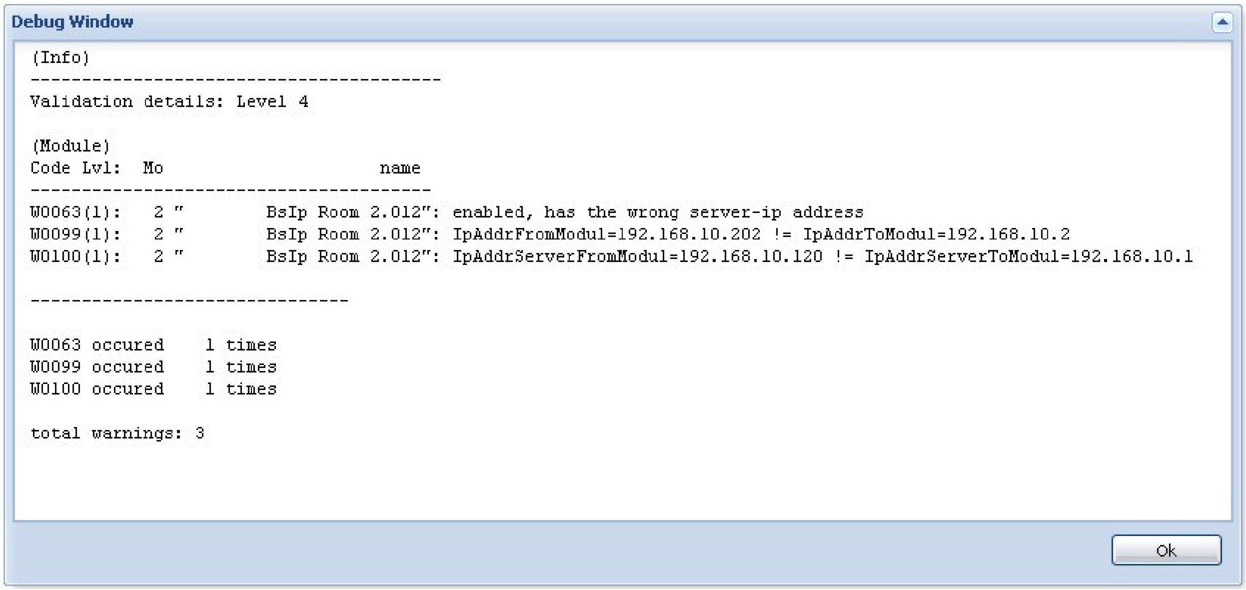
At the WBM of the DECT Controller switch to configuration page **Dect-About.**



The fields **IpAddr Module** and **IpAddr Server** display the current (default) values for the selected base station.
To apply all the changes to all base stations, the base stations have to be synchronized. During the synchronization process the base station will receive its configuration values from the DECT Controller.
Apply the changes by clicking the Apply button(not shown) at the bottom section.

Confirm the message box by clicking on **OK**.

A debug window will appear.



Confirm the Debug window by clicking on **OK**.
Start the synchronization process by clicking the **Sync** button(not shown) at the bottom section.
A message box will appear.

Confirm the message box by clicking on **OK**.
The following process will transfer the new settings to all base stations. Therefore, the base stations will be rebooted automatically by the system.
Wait about **2 minutes** until the base station is started completely (LED states should be green/red)
To verify the configured values of the previous steps, click on the **SCAN** button(not shown).

At the WBM of the DECT Controller switch to configuration page **Dect-About**(not shown).
Verify, if the values **IpAddr Module** and **IpAddr Server** contain the correct values which have been configured before (IpAddrModule → IpAddr Module, Server Ip → IpAddr Server).

8.7. Start System Services and Register Handsets

8.7.1. Start System Services

Switch to configuration page **System**.
To start the functionality of the DECT Controller, first start the system services.
Start the services by clicking the **System Start** button(not shown) at the bottom section. Starting the services requires some time.
Check if the system services are running by clicking at the **Refresh** button(not shown) several times.

Enabled	Applic Name	Status	License	Up since	Service
☑	capisrv.exe	Stopped	OK	23.02.2017 13:12	☑
☑	iwu.exe	Stopped	OK	23.02.2017 13:13	☑
☑	EtpRouter.exe	Stopped	OK	23.02.2017 13:13	☑

If the system services are started correctly (all states displayed as **Running**, the LED state at the DECT base station should change from flashing yellow to flashing green.
If a time server is configured correctly and can be contacted, the field "Up since" should display actual local time values, otherwise time will start at "01.01.1970 00:00".

8.7.2. Register Handsets

Note: *Alternatively it is possible to register several users at once using the Bulk Registration Mode. Since this method is out of scope of a quick start, the manual method is used here.*

User-DECT

Switch to configuration page **User-Dect**.

- Prepare one handset for the registration process.
- Attention: Do not confirm the following procedure at the Handset right now! Start the **Registration procedure** via the **menu at the Handset**.
  If the handset requires a PARK, leave the PARK field empty and confirm with **OK**.
  – If the handset requires an Access Code, enter the DECT Controller system PIN which has been pre-configured to 0000 at configuration page DECT as Access Code.
  Attention: Do not confirm this entry yet!
- Select the corresponding user in the WBM to which the handset has to be assigned to.

- Activate the Registration procedure at the WBM by clicking Register(not shown) at the bottom section of page **User-Dect**.

- Now confirm the already entered PIN at the Handset (normally with soft button "OK").
- The WBM displays the successful Registration of the Handset.

The IPUI of the registered handset is displayed in hexadecimal notation in the user entry.

Index	Ena...	Msn	DisplayName	Comment	HandsetType	Ipui	UAK
1	☑	8075200	FC4 8075200	default	DoIP	0176008285	
2	☑	8075201	FC4 8075201	default	DoIP	017601B262	
3	☑	8075202	FC11 8075202	default	DoIP	0011E24DC5	
4	☑	8075203	D11 8075203	default	DoIP	017604949B	

After registering several users, telephony functionality should be available. You should be able to establish calls between the handsets.

9. Verification Steps

The following steps can be taken to ensure that connections between Funktel f.airnet DECT handsets and Session Manager and Communication Manager are up.

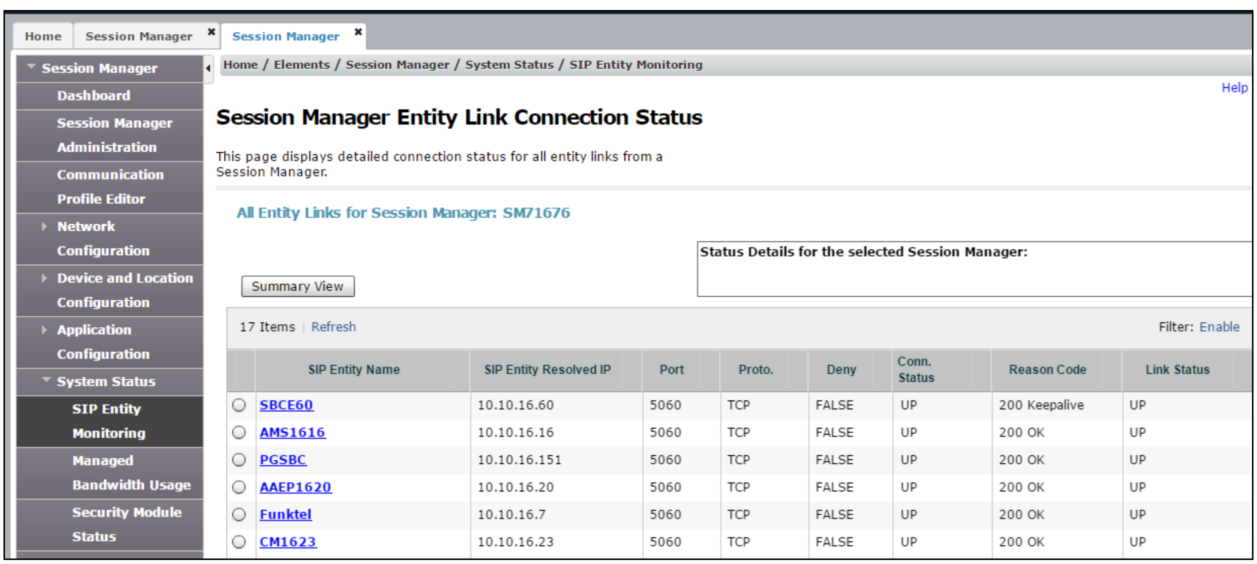
9.1. Communication Manager SIP Trunk status

Use the status trunk n where n is the SIP trunk number. Make sure that all trunks are showing as **in-service/idle**. Make a Call into the Communication Manager and make sure that the call can be answered.

```
status trunk 11

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0011/001 T00266   in-service/idle    no
0011/002 T00267   in-service/idle    no
0011/003 T00268   in-service/idle    no
0011/004 T00269   in-service/idle    no
0011/005 T00270   in-service/idle    no
0011/006 T00271   in-service/idle    no
0011/007 T00272   in-service/idle    no
0011/008 T00273   in-service/idle    no
0011/009 T00274   in-service/idle    no
0011/010 T00275   in-service/idle    no
```

9.2. Session Manager Entity Status

From the System Manager Dashboard select Session Manager → SIP Entity Monitoring. Make sure that the Funktel Entity shows **UP** under **Conn. Status** and **Link Status**.



10. Conclusion

These Application Notes describe the configuration steps required for Funktel DPx and DECT Handsets to successfully interoperate with Avaya Aura® Communication Manager R7.0 and Avaya Aura® Session Manager via a SIP Trunk. Please refer to **Section 2.2** for test results and observations.

11. 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] Administering Avaya Aura® Session Manager, Release 7.0, Issue 1 August 2015
[4] Administering Avaya Aura® System Manager, Release 7.0, Issue 1, August, 2015

Product Documentation for Funktel Products can be obtained from Funktel.

[1] *DECT IP Controller Administration Manual*, Document ID 5010790011
[2] *DPx DevConnect Setup guide*
[3] *Bedienungsanltg. FC4/FC4Ex(HS)*, Document ID 5000807201
[4] *Bedienungsanleitung D11/FC11*, Document ID 5000807226

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.