Documentation

OpenScape Voice
OpenStage 15, OpenStage 20, OpenStage 40, OpenStage 60, OpenStage 80

Administration Manual

A31003-O1010-M100-18-76A9

Communication for the open minded

Siemens Enterprise Communications
www.siemens-enterprise.com

SIEMENS

Our Quality and Environmental Management Systems are implemented according to the requirements of the ISO9001 and ISO14001 standard certified by an external certification company.

Reference No.: A31003-O1010-M100-18-76A9

Communication for the open minded

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Content

1 Overview

1.1 Important Notes

Do not operate the equipment in environments where there is a danger of explosions.

For safety reasons the phone should only be operating using the supplied plug in power unit.

Use only original Siemens accessories!

Using other accessories may be dangerous, and will invalidate the warranty, extended manufacturer's liability and the CE mark.

Never open the telephone or add-on equipment. If you encounter any problems, contact System Support.

Installation requirement for USA, Canada, Norway, Finland and Sweden: Connection to networks which use outside cables is prohibited. Only in-house networks are permitted.

For USA and Canada only:
This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This product is a UL Listed Accessory, I.T.E., in U.S.A. and Canada.
This equipment also complies with the Part 68 of the FCC Rules and the Industrie Canada CS-03.

1.2 Maintenance Notes

Do not operate the telephone in environments where there is a danger of explosions.

Use only original Siemens accessories. Using other accessories may be dangerous, and will invalidate the warranty and the CE mark.

Never open the telephone or a key module. If you encounter any problems, contact System Support.

1.3 About the Manual

The instructions within this manual will help you in administering and maintaining the OpenStage phone. The instructions contain important information for safe and proper operation of the phones. Follow them carefully to avoid improper operation and get the most out of your multi-function telephone in a network environment.

This guide is intended for service providers and network administrators who administer VoIP services using the OpenStage phone and who have a fundamental understanding of SIP. The tasks described in this guide are not intended for end users. Many of these tasks affect the ability of a phone to function on the network and require an understanding of IP networking and telephony concepts.

These instructions are laid out in a user-oriented manner, which means that you are led through the functions of the OpenStage phone step by step, wherever expedient. For the users, a separate manual is provided.

You can find further information on the official Siemens Enterprise Communications website (http://www.enterprise-communications.siemens.com) and on the Siemens Enterprise Wiki (http://wiki.siemens-enterprise.com).

1.4 Conventions for this Document

The terms for parameters and functions used in this document are derived from the web interface (WBM). In some cases, the the phone's local menu uses shorter, less specific terms and abbreviations. In a few cases the terminologies differ in wording. If so, the local menu term is added with a preceding "/".

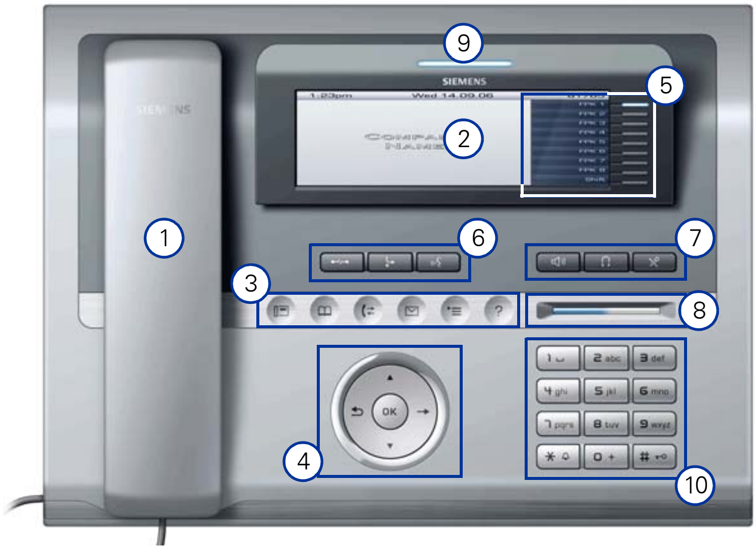
For the parameter described in this document, a WBM screenshot and the path in the local phone menu is provided. All WBM screenshots are taken from OpenStage 60/80. As some WBM input masks have been changed with firmware updates, the screenshots are selected after the following rules:

- If a later version contains more or less parameters compared to previous software versions, the screenshot of the older version is shown.
- If the title of a mask (e.g. "Pixel saver" vs. "Energy saving") or the name of a parameter (e.g. "Time Zone" vs. "DST zone") has changed, the later version is shown.
- If a parameter has moved from one mask to another, both older and later versions are shown. The same is true for the local menu paths.

The focus of this document comprehends the software versions from V1R5 onwards.

1.5 The OpenStage Family

1.5.1 OpenStage 60/80

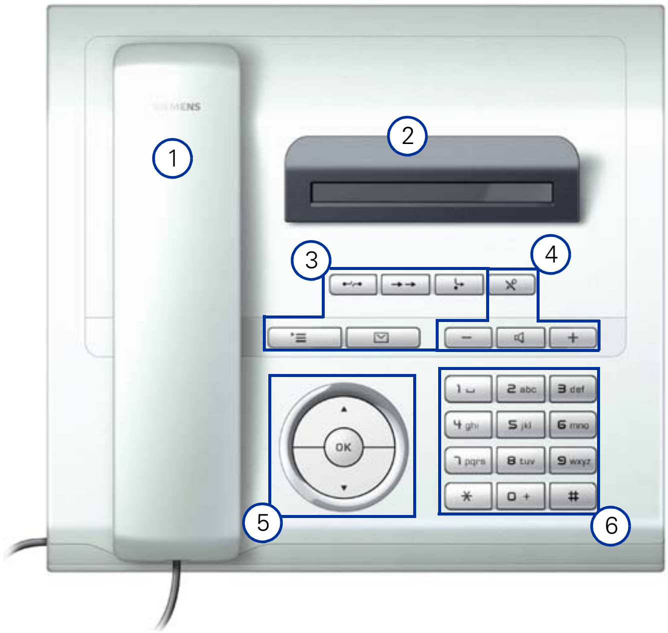


1	With the **handset**, the user can pick up and dial calls in the usual manner.
2	The **graphic display** provides intuitive support for telephone operation.
3	The **mode keys** provide easy access to the phone's applications.
4	With the **TouchGuide**, the user/administrator can navigate in the phone functions, applications, and configuration menus.
5	The **free programmable keys** enable the user to customize the telephone in line with his/her personal needs by assigning individual phone numbers and functions.
6	The **fixed function keys** provide access to frequently used telephony functions.
7	With the **audio keys**, the user can control the audio settings.
8	With the **TouchSlider**, the user can adjust the volume, e.g. of ringtones.
9	Inbound calls are visually signaled via the **call display**.
10	The **keypad** is used for entering phone numbers and text.

1.5.2 OpenStage 40

1	With the **handset**, the user can pick up and dial calls in the usual manner.
2	The **graphic display** provides intuitive support for telephone operation.
3	The **fixed function keys** provide access to frequently used telephony functions.
4	With the **5-way navigator**, the user/administrator can navigate in the various phone functions, applications, and configuration menus.
5	The **free programmable keys** enable the user to customize the telephone in line with his/her personal needs by assigning individual phone numbers and functions.
6	With the **audio keys**, the user can control the audio settings.
7	Inbound calls are visually signaled via the **call display**.
8	The **keypad** is used for entering phone numbers and text.

1.5.3 OpenStage 20



1	With the **handset**, the user can pick up and dial calls in the usual manner.
2	The **display** provides intuitive support for telephone operation.
3	The **fixed function keys** provide access to frequently used telephony functions.
4	With the **audio keys**, the user can control the audio settings.
5	With the **3-way navigator**, the user/administrator can navigate in the various phone functions, applications, and configuration menus.
6	The **keypad** is used for entering phone numbers and text.

1.5.4 OpenStage 15

1	With the **handset**, the user can pick up and dial calls in the usual manner.
2	The **display** provides intuitive support for telephone operation.
3	With the **audio keys**, the user can control the audio settings.
4	The **fixed function keys** provide access to frequently used telephony functions.
5	The **keypad** is used for entering phone numbers and text.
6	With the **navigation keys**, the user/administrator can navigate in the various phone functions, applications, and configuration menus.
7	The **free programmable keys** enable the user to customize the telephone in line with his/her personal needs by assigning individual phone numbers and functions.

1.6 Administration Interfaces

You can configure the OpenStage phone by using any of the methods described in this chapter.

1.6.1 Web-based Management (WBM)

This method employs a web browser for communication with the phone via HTTP or HTTPS. It is applicable for remote configuration of individual IP phones in your network. Direct access to the phone is not required.

> To use this method, the phone must first obtain IP connectivity.

1.6.2 DLS (Deployment Service)

The Deployment Service (DLS) is a HiPath Management application for administering phones and soft clients in both HiPath and non-HiPath networks. It has a Java-supported, web-based user interface, which runs on an internet browser. For further information, please refer to the Deployment Service Administration Guide.

1.6.3 Local Phone Menu

This method provides direct configuration of an the OpenStage phone. Direct access to the phone is required.

> As long as the IP connection is not properly configured, you have to use this method to set up the phone.

2 Startup

2.1 Prerequisites

The OpenStage phone acts as an endpoint client on an IP telephony network, and has the following network requirements:

- An Ethernet connection to a network with SIP clients and servers.

> ⚠ Only use **switches** in the LAN, to which the OpenStage phone is connected. An operation at hubs can cause serious malfunctions in the hub and in the whole network.

- OpenScape Voice server.
- An FTP Server for file transfer, e. g. firmware, configuration data, application software.
- A DHCP (Dynamic Host Configuration Protocol) server (recommended).
- DLS (Deployment Service) for advanced configuration and software deployment (recommended).

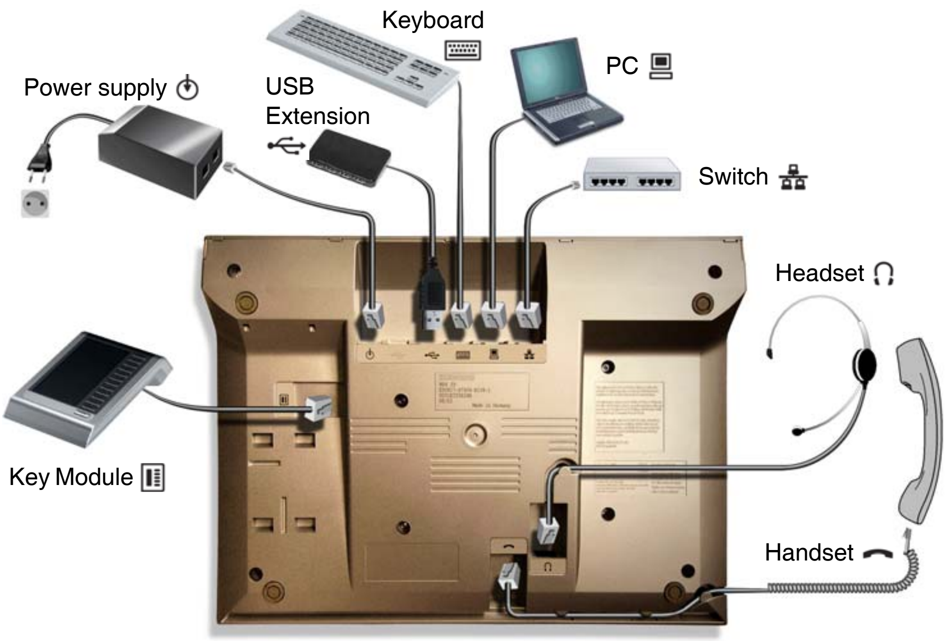
2.2 Assembling and Installing the Phone

2.2.1 Shipment

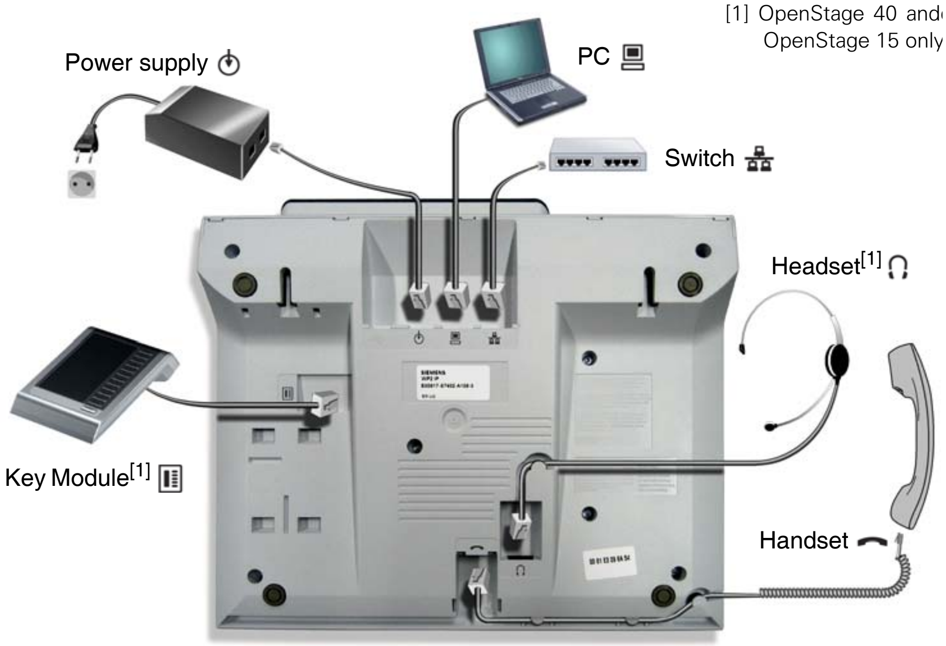
- **Phone**
- **Handset**
- **Handset cable**
- Subpackage:
  - **Document "Information and Important Operating Procedures"**
  - **Emergency number sticker**
- **Emergency Number Sticker**

2.2.2 Connectors at the bottom side

OpenStage 60



OpenStage 40 (OpenStage 15 and 20 similar, except [1])



2.2.3 Assembly

1. **Handset**

   Insert the plug on the long end of the handset cable into the jack on the base of the telephone and press the cable into the groove provided for it. Next, insert the plug on the short end of the handset cable into the jack on the handset.

2. **Emergency Number Sticker**

   Write your telephone number and those for the fire and police departments on the included label and attach it to the telephone housing underneath the handset (see arrow).

2.2.4 Connecting the Phone

1. Plug the LAN cable into the connector at the bottom of the telephone and connect the cable to the LAN resp. switch. If PoE (Power over Ethernet) is to be used, the PSE (Power Sourcing Equipment) must meet the IEEE 802.3af specification.
   For details about the required power supply, see the following table:

Model	Power Consumption/Supply
OpenStage 15[1]	Power Class 1
OpenStage 20 E	Power Class 1
OpenStage 20	Power Class 1
OpenStage 20 G	Power Class 2
OpenStage 40[2]	Power Class 2
OpenStage 40 + 2nd Key Module	Power Class 2
OpenStage 40 G[2]	Power Class 3
OpenStage 40 G + 2nd Key Module	Power Class 3
OpenStage 60/80[3]	Power Class 3
OpenStage 60/80 + 2nd Key Module	Power Class 3
OpenStage 60/80 G[3]	Power Class 3
OpenStage 60/80 G + 2nd Key Module	External power unit required

1 Includes 1 Key Module 15.
2 Includes 1 Key Module.
3 Includes 1 Key Module + USB-Extension with Acoustic Unit.

2. Only if Power over Ethernet (PoE) is **NOT** supported:

> ⚠ The order no. for the plug-in power supply is region specific:
> EU: C39280-Z4-C510
> UK: C39280-Z4-C512
> USA: C39280-Z4-C511

Plug the power supply unit into the mains. Connect the plug-in power supply unit to the jack at the bottom of the phone.

3. If applicable, connect the following optional jacks:
   - LAN connection to PC
   - Headset (accessory)
   - Connection to add-on device (accessory)
   - Connection to external keyboard (accessory)
   - USB master for connection to a USB device (e. g. accessory USB Acoustic Adapter)

> ⚠ **To prevent damage on the OpenStage phone, connect an USB stick using the adapter cable C39195-Z7704-A5.**

> ⚠ **Do not connect a USB hub to the phone's USB port, as this may lead to stability problems.**

2.3 Quick Start

This section describes a typical case: the setup of an OpenStage endpoint in an environment using a DHCP server and the web interface. For different scenarios, cross-references to the corresponding section of the administration chapter are given.

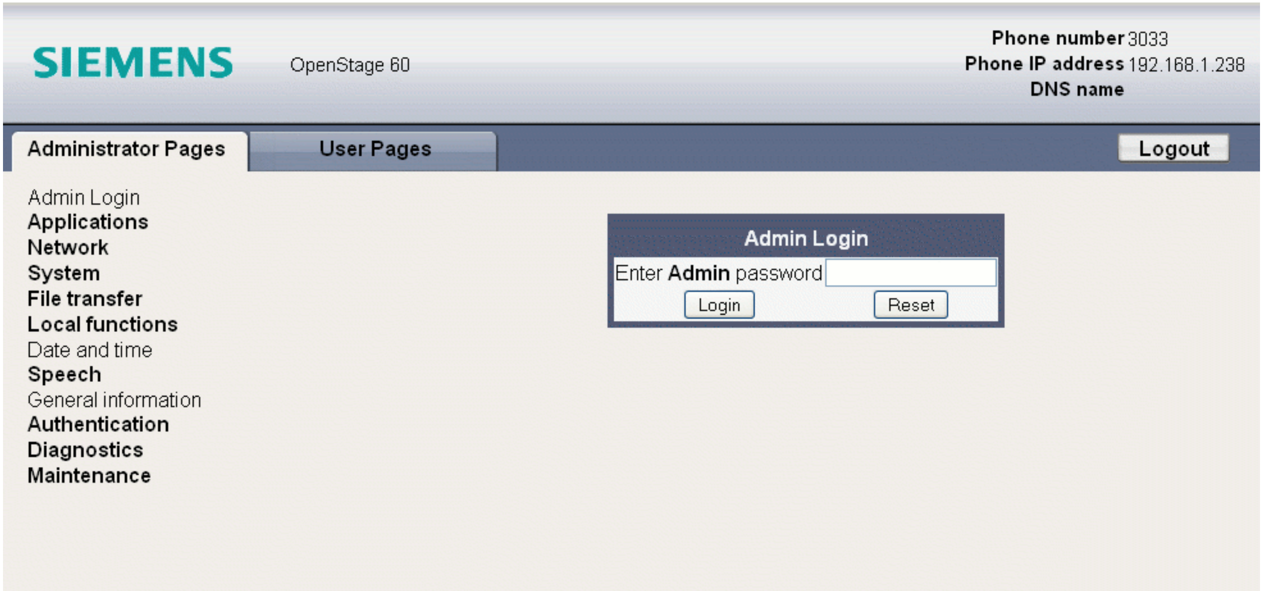
> Alternatively, the DLS (Deployment Service) administration tool can be used. Its Plug & Play functionality allows to provide the phone with configuration data by assigning an existing data profile to the phone's MAC address or E.164 number. For further information, see the Deployment Service Administration Manual.

> Any settings made by a DHCP server are not configurable by other configuration tools.

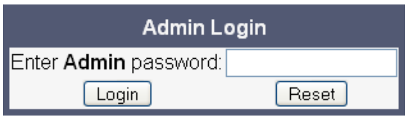
2.3.1 Access the Web Interface (WBM)

1. Open your web browser (MS Internet Explorer or Firefox) and enter the appropriate URL. Example: `https://192.168.1.15` or `https://myphone.phones` (firmware V2)
   For configuring the phone's DNS name, which is possible which firmware V2, please refer to Section 3.3.6.3, "Terminal Hostname (V2)".

If the browser displays a certificate notification, accept it. The start page of the web interface appears. In the upper right corner, the phone number, the phone's IP address, as well as the DNS name assigned to the phone are displayed. The left corner contains the user menu tree.



2. Click on the tab "Administrator Pages". In the dialog box, enter the admin password:

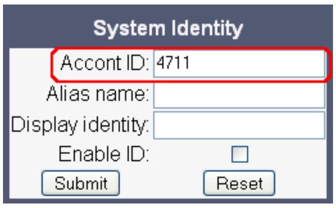


3. The administration main page opens. The left column contains the menu tree. If you click on an item which is printed in normal style, the corresponding dialog opens in the center of the page. If you click on an item printed in bold letters, a sub-menu opens in the right column.

2.3.2 Set the Terminal Number

If the user and administrator menus are needed in the course of setup, the terminal number, which by default is identical with the phone number, must be configured first. When the phone is in delivery status, the terminal number input form is presented to the user/administrator right after booting, unless the Plug&Play facility of the DLS is used. For further information about this setting, please refer to Section 3.5.1.1, “Terminal Identity”. With the WBM, the terminal number is configured as follows:

In the left column, select System > System Identity to open the "System Identity" dialog. Enter the terminal number, i. e. the SIP name / phone number.



2.3.3 Basic Network Configuration

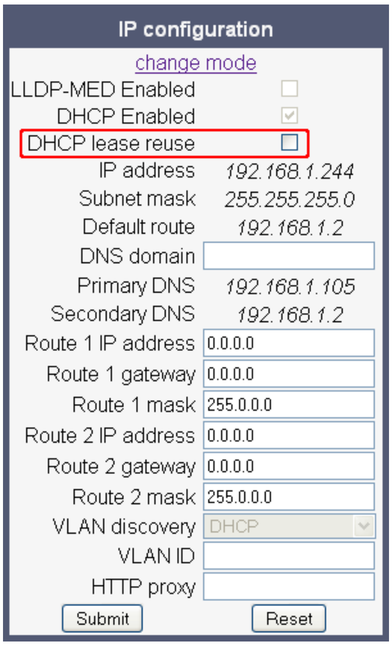
For basic functionality, DHCP must provide the following parameters:

- **IP Address**: IP Address for the phone.
- **Subnet Mask (option #1)**: Subnet mask of the phone.
- **Default Route (option #3 "Router")**: IP Address of the default gateway which is used for connections beyond the subnet.
- **DNS IP Addresses (option #6 "Domain Server")**: IP Addresses of the primary and secondary DNS servers.

If no DHCP server is present, see Section 3.3.3, “IP Address - Manual Configuration” for IP address and subnet mask, and Section 3.3.4, “Default Route/Gateway” for the default route.

2.3.4 DHCP Resilience (V2R1)

With firmware version V2R1, it is possible to sustain network connectivity in case of DHCP server failure. If **DHCP lease reuse** is activated, the phone will keep its DHCP-based IP address even if the lease expires. To prevent address conflicts, the phone will send ARP requests in 5 second intervals. Additionally, it will send discovery messages periodically to obtain a new DHCP lease.



2.3.5 Date and Time / SNTP

An SNTP (Simple Network Time Protocol) server provides the current date and time for network clients. The IP address of an SNTP server can be given by DHCP.

In order to provide the correct time, it is required to give the timezone offset, i.e. the shift in hours to be added to the UTC time provided by the SNTP server.

The following DHCP options are required:

- **SNTP IP Address (option #42 "NTP Servers")**: IP Address or hostname of the SNTP server to be used by the phone.
- **Timezone offset (option #2 "Time Offset")**: Offset in seconds in relationship to the UTC time provided by the SNTP server.

For manual configuration of date and time see Section 3.5.4, “Date and Time”.

2.3.6 SIP Server Address

The IP Address or hostname of the SIP server can be provided by DHCP.

The option's name and code are as follows:

- **option #120 "SIP Servers DHCP Option"**

For manual configuration of the SIP server address see Section 3.5.5.1, "SIP Addresses".

2.3.7 Extended Network Configuration

To have constant access to other subnets, you can enter a total of two more network destinations. For each further domain/subnet you wish to use, first the IP address for the destination, and then that of the router must be given. The option's name and code are as follows:

- **option #33 "Static Routing Table"**

For manual configuration of specific/static routing see Section 3.3.5, "Specific IP Routing".

Also the DNS domain wherein the phone is located can be specified by DHCP. The option's name and code are as follows:

- **option #15 "Domain Name"**

For manual configuration of the DNS domain name see Section 3.3.6.1, "DNS Domain Name".

2.3.8 Vendor Specific: VLAN Discovery And DLS Address

> The VLAN ID can also be configured by LLDP-MED (see Section 3.2.2.2, "Automatic VLAN discovery using LLDP-MED").

If the phone is to be located in a VLAN (Virtual LAN), a VLAN ID must be assigned. In case the VLAN shall be provided by DHCP, **VLAN Discovery** must be set to "DHCP" (see Section 3.2.2.1, "Automatic VLAN discovery using DHCP").

If a DLS (Deployment Service) server is in use, its IP address must be provided. It is recommended to configure the DLS server address by DCHP, as this method enables full Plug & Play: having received the DLS address from DHCP, the phone will contact the DLS during startup. Provided that the DLS is configured appropriately, it will send all necessary configuration data to the phone. Additionally, this method is relevant to security, as it ensures the authenticity of the DLS server.

For manual configuration of the DLS server address see Section 3.3.7, "Configuration & Update Service (DLS)".

For the configuration of vendor-specific settings by DHCP, there are two alternative methods: 1) the use of a vendor class, or 2) the use of DHCP option 43.

2.3.8.1 Using a Vendor Class

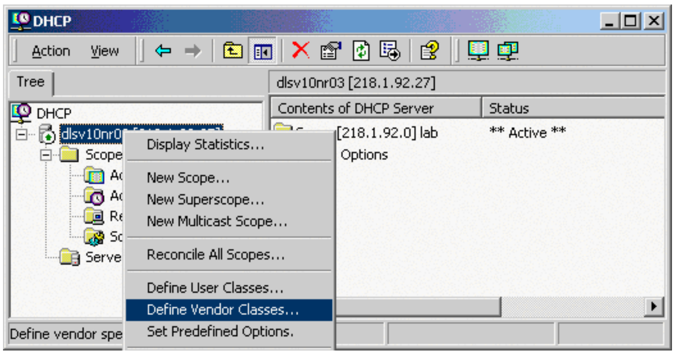
It is recommended to define a vendor class on the DHCP server, thus enabling server and phone to exchange vendor-specific data exclusively. The data is disclosed from other clients.

In the following, the configuration of vendor classes is explained both for a Windows DHCP Server and for Unix/Linux.

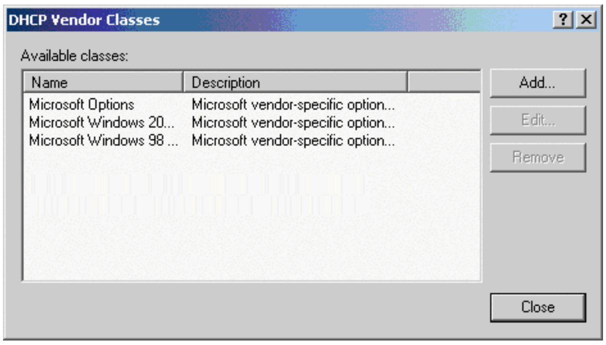
Configuration of the Windows DHCP Server

> For DHCP servers on a pre-SP2 Windows 2003 Server:
>
> Windows 2003 Server contains a bug that prevents you from using the DHCP console to create an option with the ID 1 for a user-defined vendor class. Instead, this entry must be created with the `netsh` tool in the command line (DOS shell).
>
> You can use the following command to set the required option (without error message), so that it will appear in the DHCP console afterwards:
>
> ```
> netsh dhcp server add optiondef 1 "Optipoint element 001"
> STRING 0 vendor=OptiIpPhone comment="Tag 001 for Optipoint"
> ```
>
> The value "Siemens" for optiPoint Element 1 can then be re-assigned using the DHCP console.
>
> This error was corrected in Windows 2003 Server SP2.

1. In the Windows Start menu, select **Start** > **Programs** > **Administrative Tools** > **DHCP**.
2. In the DHCP console menu, right-click the DHCP server in question and select **Define Vendor Classes...** in the context menu.

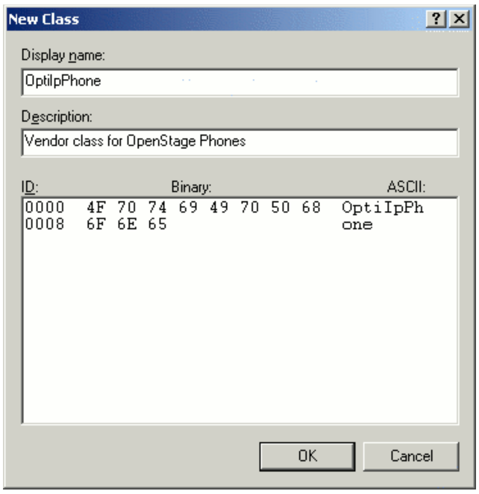


3. A dialog window opens with a list of the classes that are already available.

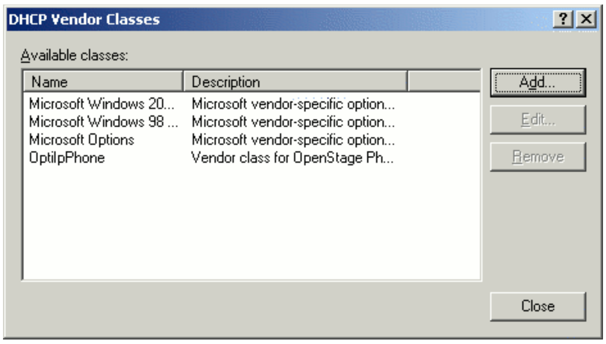


4. Press **Add...** to define a new vendor class.

5. Enter "OptiIpPhone" as **Display name** and give a description of this class. Provide the class name proper by setting the cursor underneath **ASCII** and typing "OptiIpPhone". The binary value is displayed simultaneously.

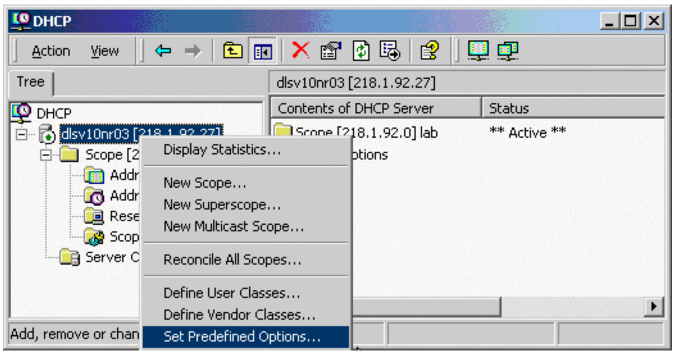


Click **OK** to apply the changes. The new vendor class now appears in the list:

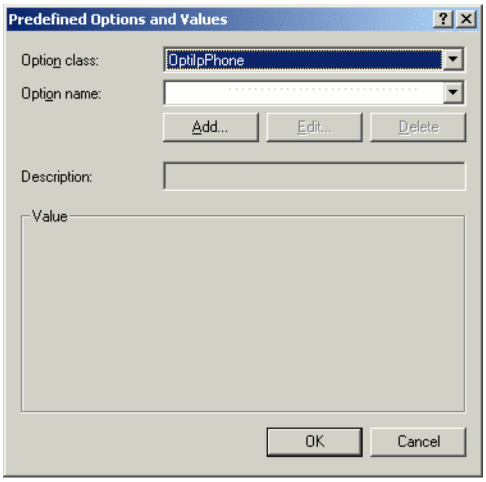


6. Exit the window with **Close**.

7. In the DHCP console menu, right-click the DHCP server in question and select **Set Pre-defined Options** from the context menu.

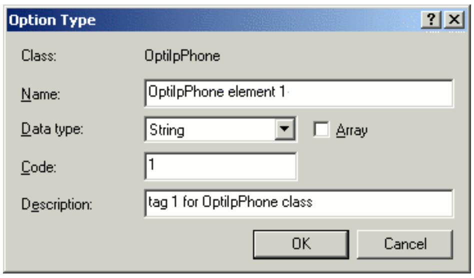


8. In the dialog, select the previously defined **OptiIpPhone** class and click on **Add...** to add a new option. (If the workaround for a pre-SP2 Windows 2003 Server has been applied, the first option will be there already.)



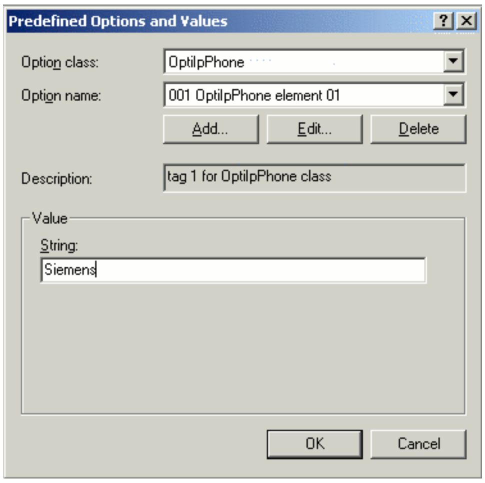
9. In the following dialog, specify the option type as follows. (If the workaround for a pre-SP2 Windows 2003 Server has been applied, the option type dialog will be skipped for the first option.)

   - **Name:** Free text, e. g. "OptiIpPhone element 01".
   - **Data type:** "String".
   - **Code:** "1"**.**
   - **Description:** Free text, e. g. "tag 1 for OptiIpPhone class".



   Click **OK** to return to the previous window.

10. The newly created option is displayed now. Enter "Siemens" in the **Value** field.

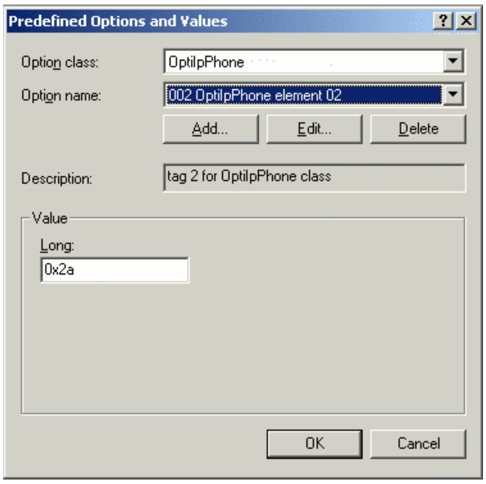


11. If the VLAN is to be provided by DHCP: Repeat step 7 and 8, and then specify the option type as follows. If you want to proceed to the configuration of the DLS address, continue with step 13.
    - **Name:** Free text, e. g. "OptiIpPhone element 02"
    - **Data type:** "Long"
    - **Code:** "2"
    - **Description:** Free text, e. g. "tag 2 for OptiIpPhone class".

    Click **OK** to return to the previous window.

12. The newly created option is displayed now. Enter the VLAN ID as a hexadecimal number in the **Value** field. In the example, the VLAN ID is 10 (Hex: 2A).

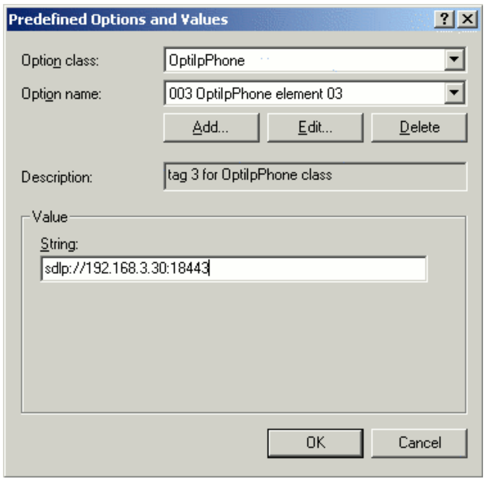
    

    If you do not intend to configure the DLS address, click OK and continue with step 15.

13. If the DLS address is to be provided by DHCP: Repeat step 7 and 8, and then specify the option type as follows.
    - **Name:** Free text, e. g. "OptiIpPhone element 03".
    - **Data type:** "String".
    - **Code:** "3"**.**
    - **Description:** Free text, e. g. "tag 3 for OptiIpPhone class".

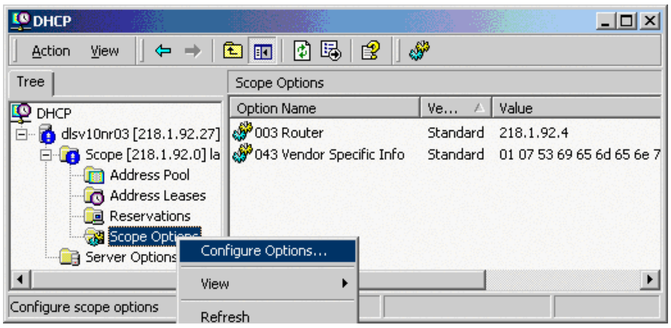
    Click **OK** to return to the previous window.

14. The newly created option is displayed now. Enter the DLS address in the **Value** field, using the following format:
    <PROTOCOL>:://<IP ADDRESS OF DLS SERVER>:<PORT NUMBER>
    In the example, the DLS address is "sdlp://192.168.3.30:18443".

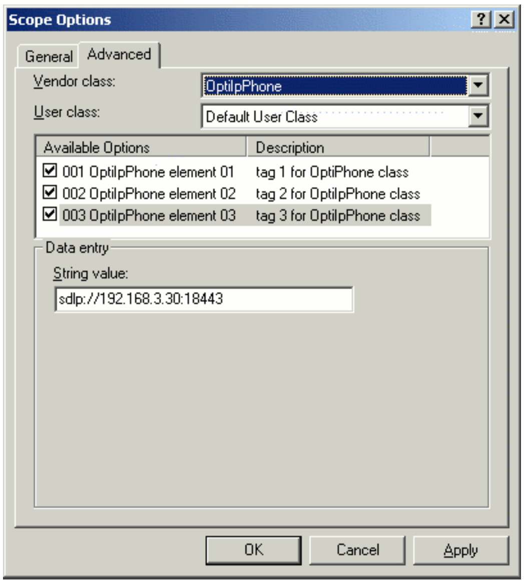


    Click **OK**.

15. To define a scope, select the DHCP server in question, and then **Scope**, and right-click **Scope Options**. Select **Configure Options...** in the context menu.

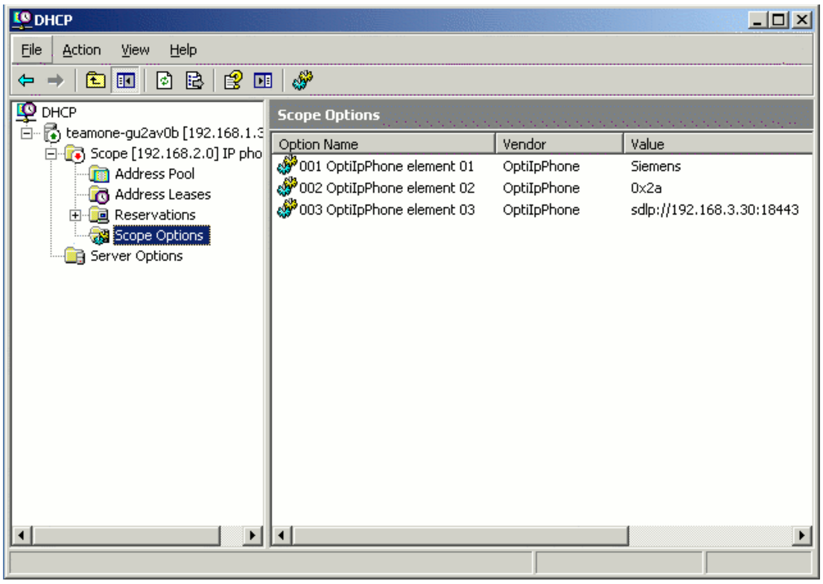


16. Select the **Advanced** tab. Under **Vendor class**, select the class that you previously defined (**OptiIpPhone**) and, under **User class**, select **Default User Class**.



Activate the check boxes for the options that you want to assign to the scope (in the example, **001**, **002**, and **003**). Click **OK**.

17. The DHCP console now shows the information that will be transmitted to the corresponding workpoints. Information from the **Standard** vendor is transmitted to all clients, whereas information from the **OptiIpPhone** vendor is transmitted only to the clients (workpoints) in this vendor class.



Setup using a DHCP server on Unix/Linux

The following snippet from a DHCP configuration file (usually dhcpd.conf) shows how to set up a configuration using a vendor class and the "vendor-encapsulated-options" option.

```
class "OptiIpPhone" {
   option vendor-encapsulated-options
   # The vendor encapsulated options consist of hexadecimal values for
the option number (for instance, 01), the length of the value (for in-
stance, 07), and the value (for instance, 53:69:65:6D:65:6E:73). The
options can be written in separate lines; the last option must be fol-
lowed by a ';' instead of a ':'.
   # Tag/Option #1: Vendor must be "Siemens"
   #1  7  S  i  e  m  e  n  s
    01:07:53:69:65:6D:65:6E:73:
   # Tag/Option #2: VLAN ID
   # 2  4  0  0  0 10
    02:04:00:00:00:0A;
   # Tag/Option #3: DLS IP Address (here: sdlp://192.168.3.30:18443)
   # 3 25 s  d  l  p  :  / /  1  9  2  .  1  6  8  .  3  . (...etc.)
    03:19:73:64:6C:70:3A:2F:2F:31:39:32:2E:31:36:38:2E:33:2E:33:30:
3A:31:38:34:34:33;
   match if substring (option vendor-class-identifier, 0, 11) =
   "OptiIpPhone";
}
```

2.3.8.2 Using Option #43 "Vendor Specific"

Alternatively, option #43 can be used for setting up the VLAN ID and DLS address. The following tags are used:

- **Tag 1: Vendor name**
- **Tag 2: VLAN ID**
- **Tag 3: DLS address**

Optionally, the DLS address can be given in an alternative way:

- **Tag 4: DLS hostname**

The Vendor name tag is coded as follows (the first line indicates the ASCII values, the second line contains the hexadecimal values):

Code	Length	Vendor name						
1	7	S	i	e	m	e	n	s
01	07	53	69	65	6D	65	6E	73

The following example shows a VLAN ID with the decimal value "10". Providing:

Code	Length	VLAN ID			
2	4	0	0	1	0
02	04	00	00	00	0A

For manual configuration of the VLAN ID see Section 3.2.2.3, “Manual configuration of a VLAN ID”.

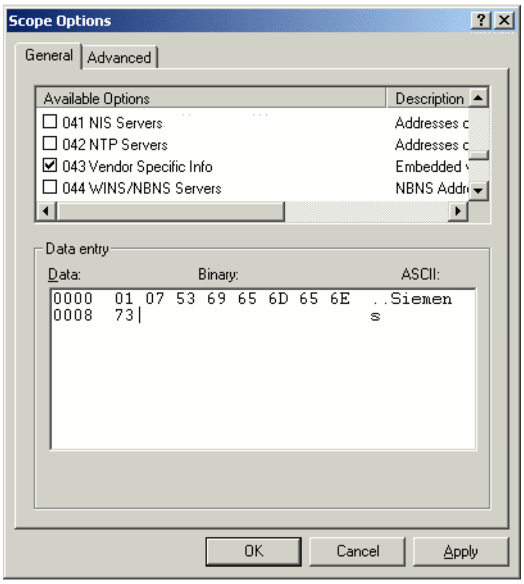
The DLS IP address tag consists of the protocol prefix "sdlp://", the IP address of the DLS server, and the DLS port number, which is "18443" by default. The following example illustrates the syntax:

Code	Length	DLS IP address																								
3	25	s	d	l	p	:	/	/	1	9	2	.	1	6	8	.	3	.	3	0	:	1	8	4	4	3
03	19	73	64	6C	70	3A	2F	2F	31	39	32	2E	31	36	38	2E	33	2E	33	30	3A	31	38	34	34	33

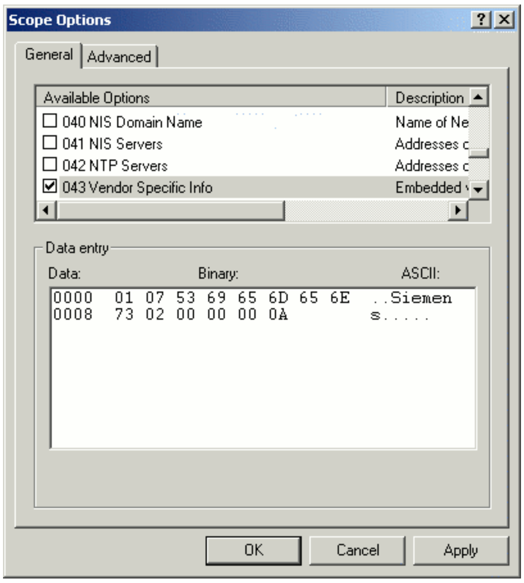
Setup using the Windows DHCP Server

1. In the Windows Start menu, select **Start** > **Programs** > **Administrative Tools** > **DHCP**.
2. Select the DHCP server and the scope. Choose **Configure Options** in the context menu using the right mouse button.

3. Enter tag 1, that is the vendor tag. The value has to be "Siemens".



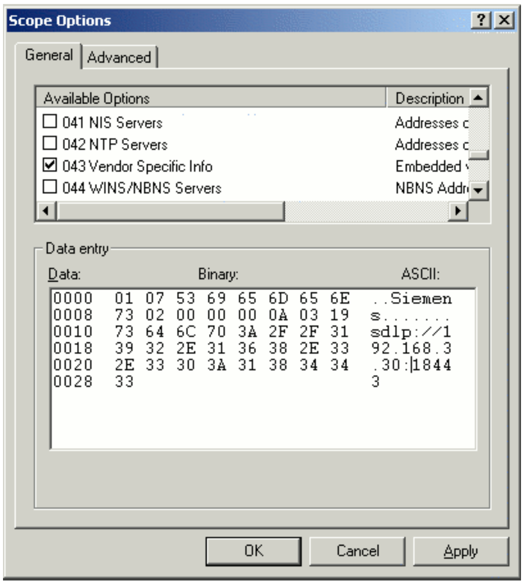
4. If the VLAN ID is to be provided by DHCP: Enter the hexadecimal value in **Data entry**. Providing the length is not required here, as the VLAN ID is always 4 Bytes long. In the example, the VLAN ID is 10 (Hex: 2A).



5. If the DLS address is to be provided by DHCP: Enter the DLS address in the **Value** field, using the following format:
<PROTOCOL>:://<IP ADDRESS OF DLS SERVER>:<PORT NUMBER>

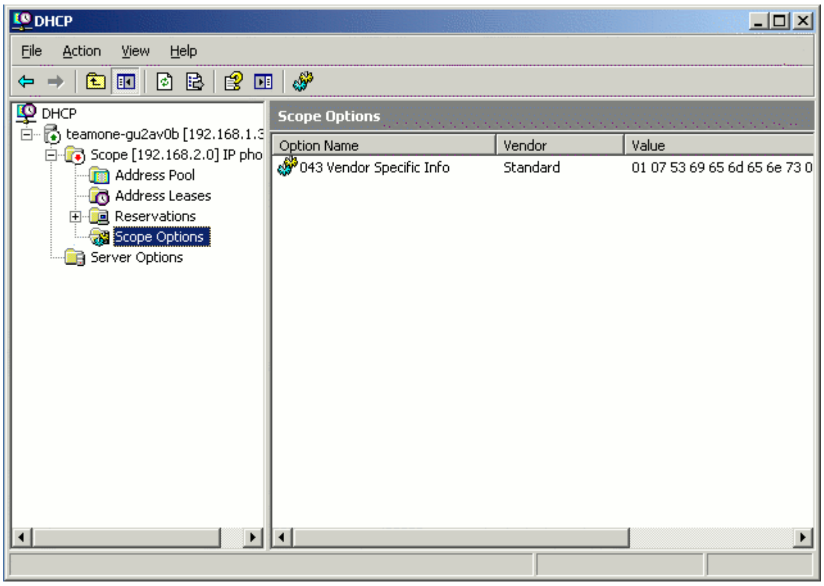
> For ensuring proper functionality, the port number should not be followed by any character.

In the example, the DLS address is "sdlp://192.168.3.30:18443".
Note that the screenshot also shows the VLAN ID described in step 4.



Click **OK**.

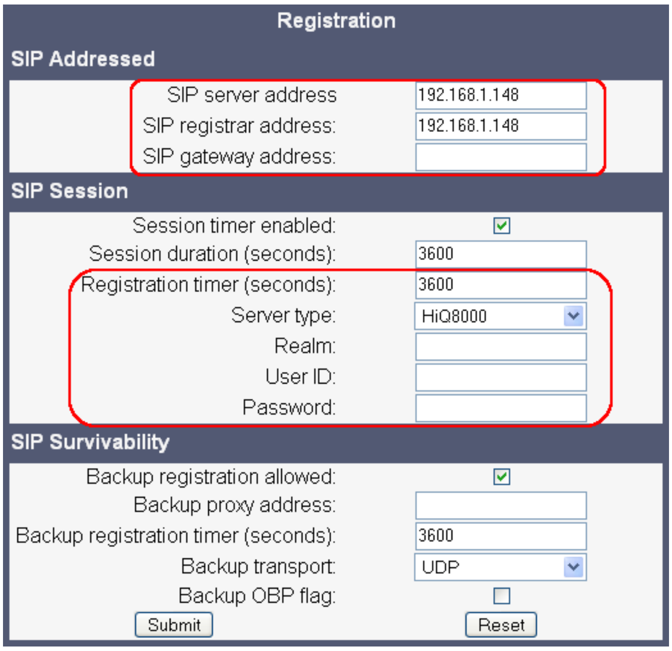
6. The DHCP console now shows the information that will be transmitted to the corresponding workpoints.



2.3.9 Registering at OpenScape Voice

For registration at the OpenScape Voice SIP server, a SIP user ID and passwort must be provided by the phone. The following procedure describes the configuration using the web interface (see Section 2.3.1, “Access the Web Interface (WBM)”; if the web interface is not applicable, please refer to Section 3.5.6, “Authenticated Registration”) for configuration via the local menu.

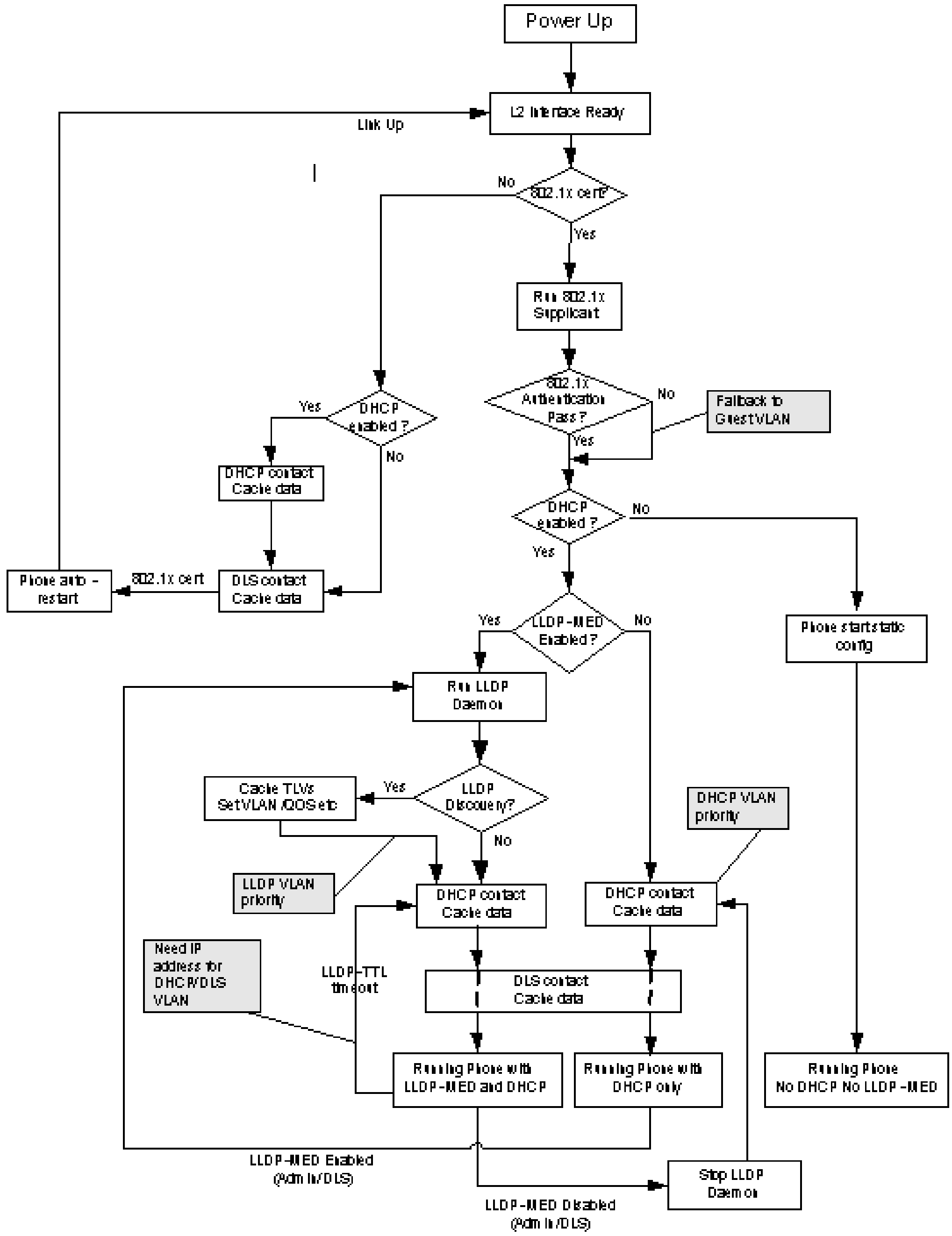
1. In the administration menu, select System > Registration. The **Registration** dialog opens.



2. Make sure that **SIP server address** and **SIP registrar address** contain the IP address of your OpenScape Voice server. If not provided by DHCP or DLS, enter the appropriate values. If the phone is to register with a gateway, enter the appropriate **SIP Gateway address**.
3. In the **Server type** field, select "OS Voice".
4. In **Realm**, enter the SIP realm the targeted user/password combination refers to.
5. In the **User ID** and **Password** fields, enter the user name/password combination for the phone.

2.4 Startup Procedure

The following flowchart shows the startup process for OpenStage phones:



3 Administration

This chapter describes the configuration of every parameter available on the OpenStage phones. For access via the local phone menu, see the following; for access using the web interface, please refer to Section 2.3.1, “Access the Web Interface (WBM)”.

3.1 Access via Local Phone

> The data entered in input fields is parsed and controlled by the phone. Thus, data is accepted only if it complies to the value range.

1. **Access the Administration Menu**

   OpenStage 60/80:

   The menu key toggles between the Settings menu, the Applications menu, and the applications currently running. Press the key repeatedly until the "Settings" tab is active. (The key toggles between the Settings menu, the Applications menu, and the applications currently running.)

   OpenStage 15/20/40:

   Press the keys ≡, ▾, and OK consecutively to select the administration menu.

2. **Enter Password**

   When the Admin menu is active, you will be prompted to enter the administrator password. The default admin password is "123456". It is highly recommended to change the password (see Section 3.18, “Password”) after your first login.

   For entering passwords with non-numeric characters, please consider the following:

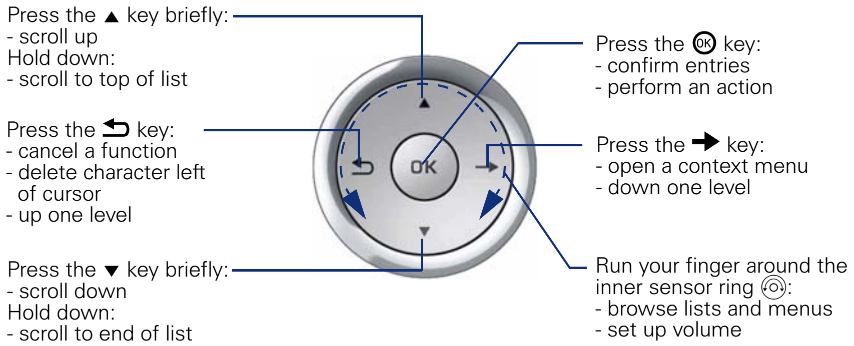
   By default, password entry is in numeric mode. For changing the mode, press the # key once or repeatedly, depending on the desired character. The # key cycles around the input modes as follows:

   (Abc) -> (abc) -> (123) -> (ABC) -> back to start.

3. Navigate within the Administration Menu

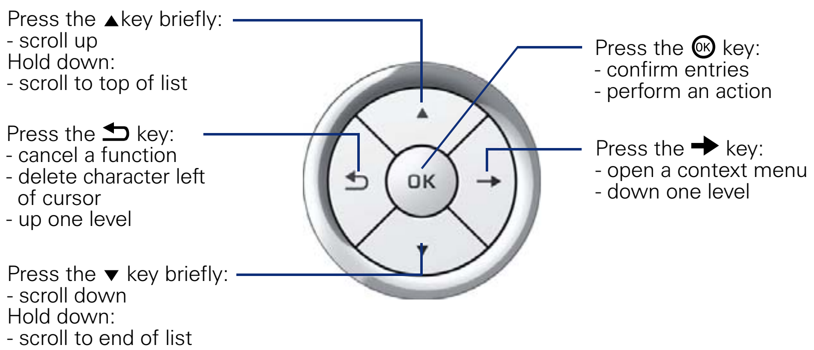
OpenStage 60/80

Use the TouchGuide to navigate and execute administrative actions in the administration menu.



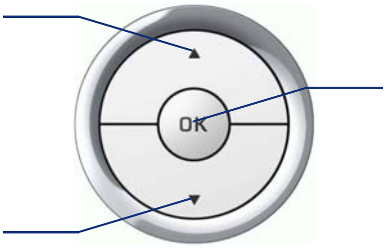
OpenStage 40

Use the 5-way navigator to navigate and execute administrative actions in the administration menu.



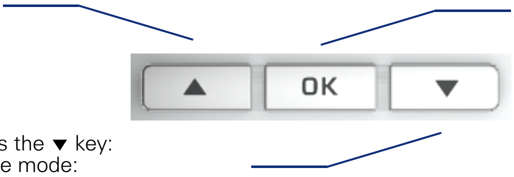
OpenStage 20

Use the 3-way navigator to navigate and execute administrative actions in the administration menu.



OpenStage 15

Use the navigation keys to navigate and execute administrative actions in the administration menu.



4. **Select a parameter**
   If a parameter is set by choosing a value from a selective list, an arrow symbol appears in the parameter field that has the focus. Press the OK key to enter the selective list. Use the Sensor Wheel resp. the ▲ and ▼ key to scroll up and down in the selective list. To select a list entry, press the OK key.

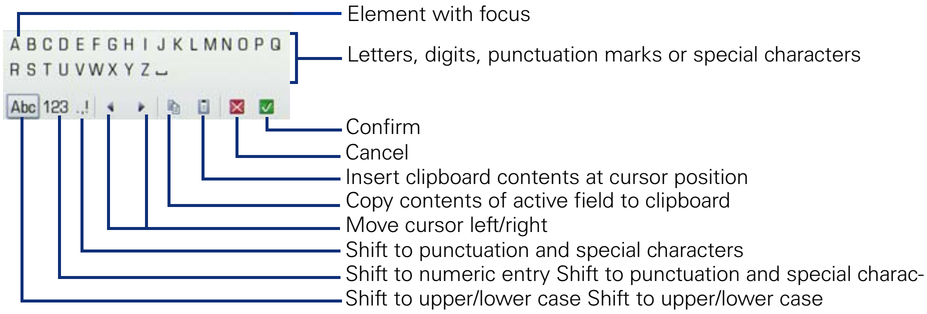
5. **Enter the parameter value**
   For selecting numbers and characters, you can use special keys. See the following table:

Key	Function
*	Switch to punctuation and special characters.
#	Toggle between lowercase characters, uppercase characters, and digits in the following order: (Abc) -> (abc) -> (123) -> (ABC) -> back to start.

OpenStage 60/80

If a parameter is set by entering a number or character data, the onscreen keypad is used. Press the OK key to enter the editor. Within the editor, solely use the key numbers or the Sensor Wheel for selecting numbers, characters, or groups of characters. The ⮌ key deletes one character in the input field, and the ➔ key moves the cursor to the OK field.

The following figure describes the elements of the onscreen keypad and their functions:



Additionally, you can use the following keys on the keypad as shortcuts for the selection of character groups

Element	Function
*	Switch to punctuation and special characters.
#	Toggle between lowercase characters, uppercase characters, and digits.

OpenStage 15/20/40

With the OpenStage 15/20/40, use the keypad for entering parameters. With the 3 way/5 way navigator, you can enter, delete, copy and paste characters and numbers as well as navigate within an entry and toggle the input mode.

6. Save and exit

When you are done, select **Save & exit** and press OK.

3.2 LAN Settings

3.2.1 LAN Port Settings

The OpenStage phone provides an integrated switch which connects the LAN, the phone itself and a PC port. By default, the switch will auto negotiate transfer rate (10/100 Mb/s, 1000 Mb/s with OpenStage 20/40/60/80 G) and duplex method (full or half duplex) with whatever equipment is connected. Optionally, the required transfer rate and duplex mode can be specified manually using the **LAN port speed** parameter.

> In the default configuration, the LAN port supports automatic detection of cable configuration (pass through or crossover cable) and will reconfigure itself as needed to connect to the network. If the phone is set up to manually configure the switch port settings, the cable detection mechanism is disabled. In this case, care must be taken to use the correct cable type.

The PC Ethernet port is controlled by the **PC port mode** parameter. If set to "Disabled", the PC port is inactive; if set to "Enabled", it is active. If set to "Mirror", the data traffic at the LAN port is mirrored at the PC port. This setting is for diagnostic purposes. If, for instance, a PC running Ethereal/Wireshark is connected to the PC port, all network activities at the phone’s LAN port can be captured.

> Removing the power from the phone, or a phone reset/reboot will result in the temporary loss of the network connection to the PC port.

When **PC port autoMDIX** is enabled, the switch determines automatically whether a regular MDI connector or a MDI-X (crossover) connector is needed, and configures the connector accordingly.

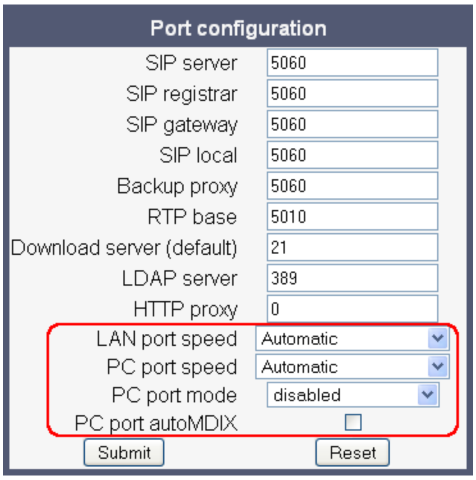
Data required

- **LAN port speed** / **LAN port type**: Settings for the ethernet port connected to a LAN switch.
  Value range: "Automatic," "10 Mbps half duplex", "10 Mbps full duplex", "100 Mbps half duplex", "100 Mbps full duplex", and, additionally, for OpenStage 20/40/60/80 G, "1 Gbps full duplex"
  Default: "Automatic"
- **PC port speed** / **PC port type**: Settings for the ethernet port connected to a PC.
  Value range: "Automatic," "10 Mbps half duplex", "10 Mbps full duplex", "100 Mbps half duplex", "100 Mbps full duplex", and, additionally, for OpenStage 20/40/60/80 G, "1 Gbps full duplex"
  Default: "Automatic"

- **PC port mode** / **PC port status**: Controls the PC port.
  Value range: "disabled", "enabled", "mirror".
  Default: "disabled"
- **PC port autoMDIX**: Switches between MDI and MDI-X automatically.
  Value range: "On", "Off"
  Default: "Off"

Administration via WBM

Network > Port configuration



Administration via Local Phone

|--- Administration
    |--- Network
        |--- Port Configuration
            |--- **LAN port type**
            |--- **PC port status**
            |--- **PC port type**
            |--- **PC port autoMDIX**

3.2.2 VLAN

VLAN (Virtual Local Area Network) is a technology that allows network administrators to partition one physical network into a set of virtual networks (or broadcast domains).

Physically partitioning the LAN into separate VLANs allows a network administrator to build a more robust network infrastructure. A good example is a separation of the data and voice networks into data and voice VLANs. This isolates the two networks and helps shield the endpoints within the voice network from disturbances in the data network and vice versa.

> The implementation of a voice network based on VLANs requires the network infrastructure (the switch fabric) to support VLANs.

In a layer 1 VLAN, the ports of a VLAN-aware switch are assigned to a VLAN statically. The switch only forwards traffic to a particular port if that port is a member of the VLAN that the traffic is allocated to. Any device connected to a VLAN-assigned port is automatically a member of this VLAN, without being a VLAN aware device itself. If two or more network clients are connected to one port, they cannot be assigned to different VLANs. When a network client is moving from one switch to another, the switches' ports have to be updated accordingly by hand.

With a layer 2 VLAN, the assignment of VLANs to network clients is realized by the MAC addresses of the network devices. In some environments, the mapping of VLANs and MAC addresses can be stored and managed by a central database. Alternatively, the VLAN ID, which defines the VLAN whereof the device is a member, can be assigned directly to the device, e. g. by DHCP. The task of determining the VLAN for which an Ethernet packet is destined is carried out by VLAN tags within each Ethernet frame. As the MAC addresses are (more or less) wired to the devices, mobility does not require any administrator action, as opposed to layer 1 VLAN. It is possible to assign one device, i.e. one MAC address, to different VLANs.

It is important that every switch connected to a PC is VLAN-capable. This is also true for the integrated switch of the OpenStage. The phone must be configured as a VLAN aware endpoint if the phone itself is a member of the voice VLAN, and the PC connected to the phone's PC port is a member of the data VLAN.

There are 3 ways for configuring the VLAN ID:

- Manually
- By DHCP
- By LLDP-MED

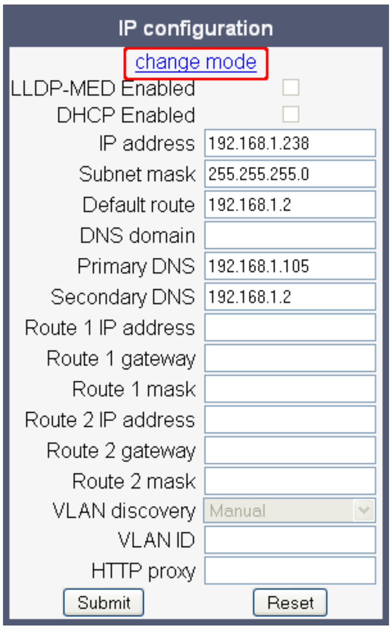
3.2.2.1 Automatic VLAN discovery using DHCP

To automatically discover a VLAN ID using DHCP, the phone must be configured as DHCP enabled, and **VLAN discovery** mode must be set to "DHCP". This is the default configuration. The DHCP server must be configured to supply the Vendor Unique Option in the correct Siemens VLAN over DHCP format. If a phone configured for VLAN discovery by DHCP fails to discover its VLAN, it will proceed to configure itself from the DHCP within the non-tagged LAN. Under these circumstances, network routing may probably not be correct.

Administration via WBM

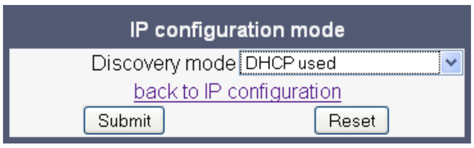
Network > IP configuration

First, click on **change mode**. Afterwards, the **IP configuration mode** dialog opens.



Network > IP configuration > **change mode**

To enable VLAN discovery by DHCP, select **DHCP used** in the **Discovery mode** menu. Afterwards, click **Submit**.



Administration via Local Phone

To enable VLAN discovery by DHCP, select **DHCP used** in the **Discovery mode** menu.

|— Administration
    |— Network
        |— IP Configuration
            |— **Discovery mode**

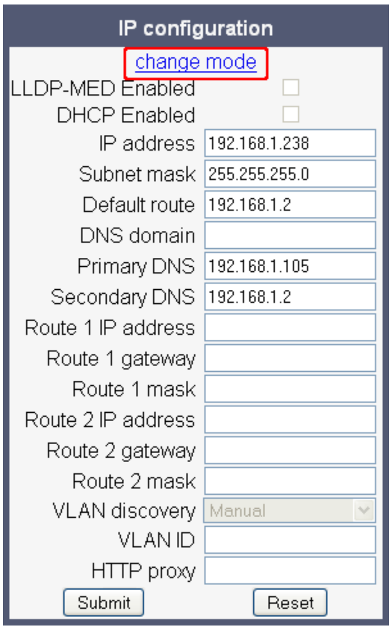
3.2.2.2 Automatic VLAN discovery using LLDP-MED

As an alternative, the VLAN ID can be configured by the network switch using LLDP-MED (Link Layer Discovery Protocol-Media Endpoint Discovery). If this option is selected, and the switch provides an appropriate TLV (Type-Length-Value) element containing the VLAN ID, this VLAN ID will be used. If no appropriate TLV is received, DHCP will be used for VLAN discovery.

Administration via WBM

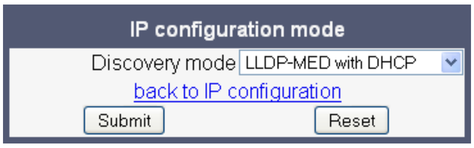
Network > IP configuration

First, click on **change mode**. Afterwards, the **IP configuration mode** dialog opens.



Network > IP configuration > **change mode**

To enable VLAN discovery by LLDP-MED, select **LLDP-MED with DHCP** in the **Discovery mode** menu. Afterwards, click **Submit**.



Administration via Local Phone

To enable VLAN discovery by DHCP, select **LLDP-MED with DHCP** in the **Discovery mode** menu.

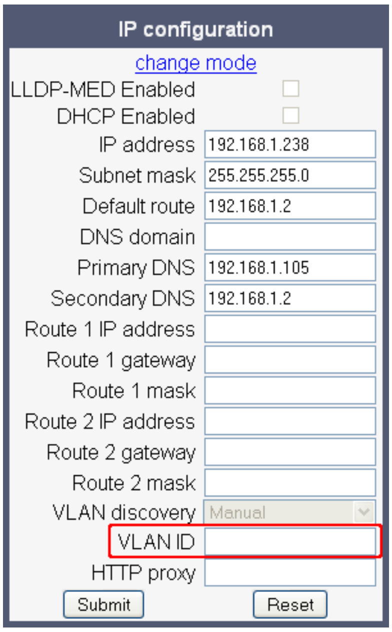
|— Administration
    |— Network
        |— IP Configuration
            |— **Discovery mode**

3.2.2.3 Manual configuration of a VLAN ID

To configure layer 2 VLAN manually, first make shure that VLAN discovery is set to "Manual" (see Section 3.2.2.1, “Automatic VLAN discovery using DHCP”). Then, the phone must be provided with a VLAN ID between 1 and 4095. If you mis-configure a phone to an incorrect VLAN, the phone will possibly not connect to the network. In DHCP mode it will behave as though the DHCP server cannot be found, in fixed IP mode no server connections will be possible.

Administration via WBM

Network > IP configuration



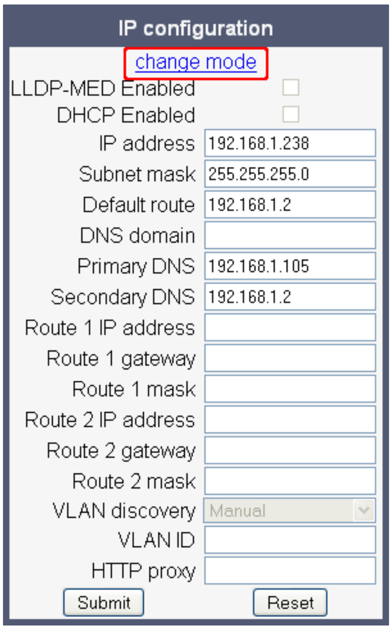
Administration via Local Phone

|--- Administration
    |--- Network
        |--- IP Configuration
            |--- **VLAN ID**

Administration via WBM

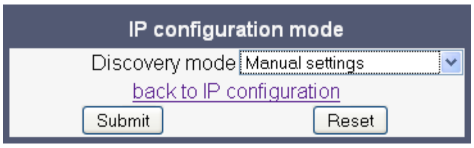
Network > IP configuration

First, click on **change mode**. Afterwards, the **IP configuration mode** dialog opens.



Network > IP configuration > **change mode**

To enable manual VLAN configuration, select **Manual settings** in the **Discovery mode** menu. Afterwards, click **Submit**.



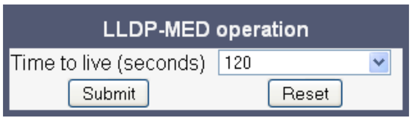
3.2.3 LLDP-MED Operation

OpenStage phones support LLDP-MED (Link Layer Discovery Protocol-Media Endpoint Discovery) for auto-configuration and network management. The auto-configurable parameters are VLAN ID (see Section 3.2.2, “VLAN”) and Quality of Service parameters (see Section 3.3.1, “Quality of Service (QoS)”).

The data sent by a network device is stored in neighboring network devices in MIB (Manegement Information Base) format. In order to keep this information up-to-date, a specific TTL (Time To Live) is specified in LLDP. This value tells a device how long the received information is valid. For OpenStage phones, the value range is **40**, **60**, **80**, **100**, **110**, **120**, **140**, **180**, **240**, **320**, **400**.

An example for LLDP-MED operation an OpenStage phones can be found in Section 5.4, “An LLDP-Med Example”.

Administration via WBM



Administration via Local Phone

|--- Administration
    |--- Network
        |--- LLDP-MED operation
            |--- TTL
                |--- **TTL**

3.3 IP Network Parameters

3.3.1 Quality of Service (QoS)

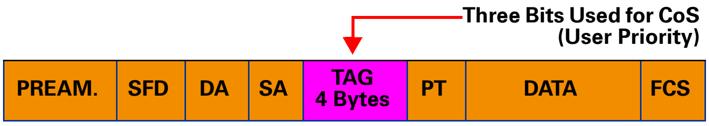
The QoS technology based on layer 2 and the two QoS technologies Diffserv and TOS/IP Precedence based on layer 3 are allowing the VoIP application to request and receive predictable service levels in terms of data throughput capacity (bandwidth), latency variations (jitter), and delay.

> Layer 2 and 3 QoS for voice transmission can be set via LLDP-MED (see Section 3.24.3, “LLDP-MED”). If so, the value can not be changed by any other interface.

3.3.1.1 Layer 2 / 802.1p

QoS on layer 2 is using 3 Bits in the 802.1q/p 4-Byte VLAN tag which has to be added in the Ethernet header.

The CoS (class of service) value can be set from 0 to 7. 7 is describing the highest priority and is reserved for network management. 5 is used for voice (RTP-streams) by default. 3 is used for signaling by default.

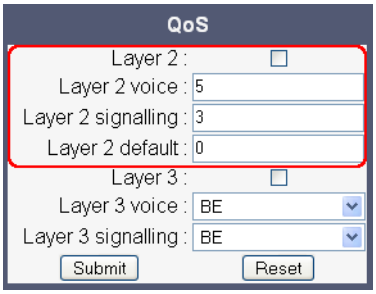


Data required

- **Layer 2**: Activates or deactivates QoS on layer 2.
  Value range: "Yes", "No"
  Default: "Yes"
- **Layer 2 voice**: Sets the CoS (Class of Service) value for voice data (RTP streams).
  Value range: 0-7
  Default: 5
- **Layer 2 signalling**: Sets the CoS (Class of Service) value for signaling.
  Value range: 0-7
  Default: 3
- **Layer 2 default**: Sets the default CoS (Class of Service) value.
  Value range: 0-7
  Default: 0

Administration via WBM

Network > QoS



Administration via Local Phone

```
|-- Administration
    |-- Network
        |-- QoS
            |-- Service
                |-- Layer 2
                |-- Layer 2 voice
                |-- Layer 2 signalling
                |-- Layer 2 default
```

3.3.1.2 Layer 3 / Diffserv

Diffserv assigns a class of service to an IP packet by adding an entry in the IP header.

Traffic flows are classified into 3 per-hop behavior groups:

1. Default
   Any traffic that does not meet the requirements of any of the other defined classes is placed in the default per-hop behaviour group. Typically, the forwarding has best-effort forwarding characteristics. The DSCP (Diffserv Codepoint) value for Default is "0 0 0 0 0 0".
2. Expedited Forwarding (EF referred to RFC 3246)
   Expedited Forwarding is used for voice (RTP streams) by default. It effectively creates a special low-latency path in the network. The DSCP (Diffserv Codepoint) value for EF is "1 0 1 1 1 0".
3. Assured Forwarding (AF referred to RFC 2597)
   Assured forwarding is used for signaling messages by default (AF31). It is less stringent than EF in a multiple dropping system. The AF values are containing two digits X and Y (AFXY), where X is describing the priority class and Y the drop level.
   Four classes X are reserved for AFXY: AF1Y (high priority), AF2Y, AF3Y and AF4Y (low priority).

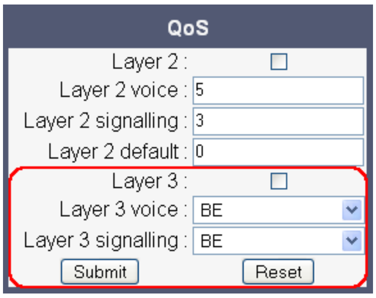
Three drop levels Y are reserved for AFXY: AFX1 (low drop probability), AFX2 and AFX3 (High drop probability). In the case of low drop level, packets are buffered over an extended period in the case of high drop level, packets are promptly rejected if they cannot be forwarded.

Data required

- **Layer 3**: Activates or deactivates QoS on layer 3.
  Value range: "Yes", "No"
  Default: "Yes"
- **Layer 3 voice**: Sets the CoS (Class of Service) value for voice data (RTP streams).
  Value range: "AF11", "AF12", "AF13", "AF21", "AF22", "AF23", "AF31", "AF32", "AF33", "AF41", "AF42", "AF43", "EF", "CS7", "CS3", "CS4", "CS5"
  Default: "EF"
- **Layer 3 signalling**: Sets the CoS (Class of Service) value for signaling.
  Value range: "AF11", "AF12", "AF13", "AF21", "AF22", "AF23", "AF31", "AF32", "AF33", "AF41", "AF42", "AF43", "EF", "CS7", "CS3", "CS4", "CS5"
  Default: "AF31"

Administration via WBM

Network > QoS



Administration via Local Phone

```
|--- Administration
     |--- Network
          |--- QoS
               |--- Service
                    |--- Layer 3
                    |--- Layer 3 voice
                    |--- Layer 3 signalling
```

3.3.2 Use DHCP

If this parameter is set to "Yes" (default), the phone will search for a DHCP server on startup and try to obtain IP data and further configuration parameters from that central server.

If no DHCP server is available in the IP network, please deactivate this option. In this case, the IP address, subnet mask and default gateway/route must be defined manually.

> With firmware version V2R1 onwards, the phone is able to maintain its IP connection even in case of DHCP server failure. For further information, please refer to Section 2.3.4, “DHCP Resilience (V2R1)”.

The following parameters can be obtained by DHCP:

Basic Configuration

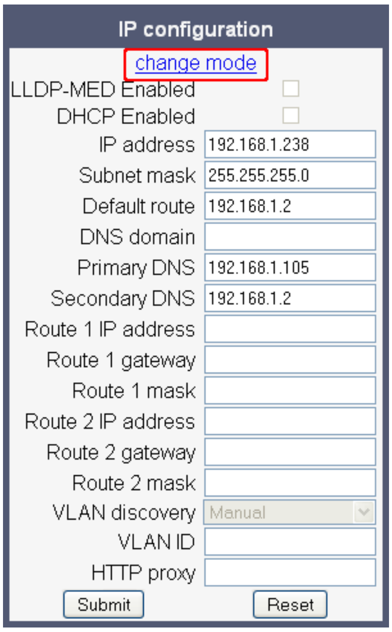
- IP Address
- Subnet Mask

Optional Configuration

- Default Route (Routers option 3)
- IP Routing/Route 1 & 2 (Static Routes option 33)
- SNTP IP Address (NTP Server option 42)
- Timezone offset (Time Server Offset option 2)
- Primary/Secondary IP Addresses (DNS Server option 6)
- DNS Domain Name (DNS Domain option 15)
- SIP Addresses / SIP Server & Registrar (SIP Server option 120)
- Vendor Unique (option 43)

Administration via WBM

Network > IP configuration



Administration via Local Phone

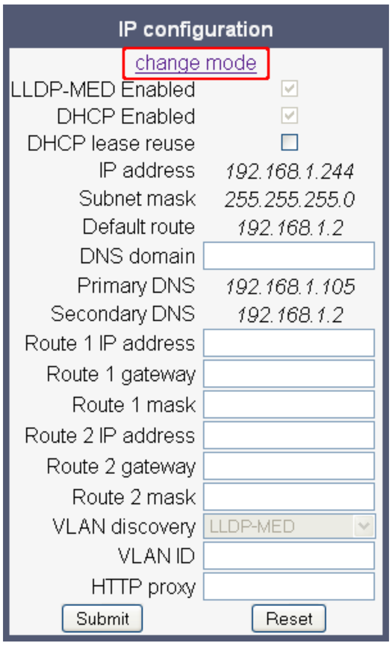
|--- Administration
    |--- Network
        |--- IP Configuration
            |--- **Use DHCP**

3.3.3 IP Address - Manual Configuration

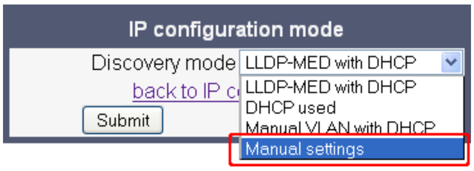
If not provided by DHCP dynamically, the phone's IP address and subnet mask must be specified manually.

By default, IP configuration by DHCP and LLDP-MED is enabled. For manual IP configuration, please proceed as follows:

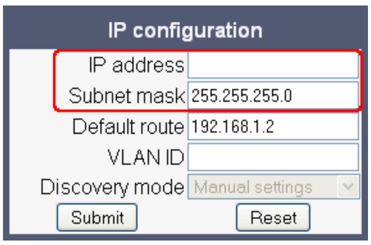
1. Navigate to **Network** > **IP configuration** and click **change mode**.



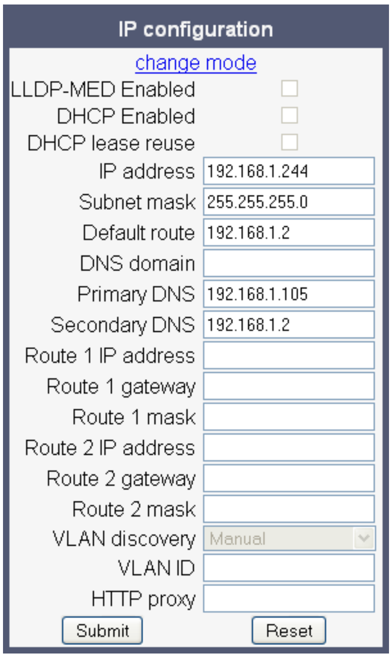
2. The dialog window **IP configuration mode** appears. In the **Discovery mode** menu, select **Manual settings**.



3. The dialog window **IP configuration** appears, with a reduced choice of parameters. Enter the **IP address** and the **Subnet mask**. If applicable, enter the **Default route** and the **VLAN ID**. When finished, click **Submit**.



4. After the phone's network service has restarted, the other IP parameters can be configured.



Administration via Local Phone

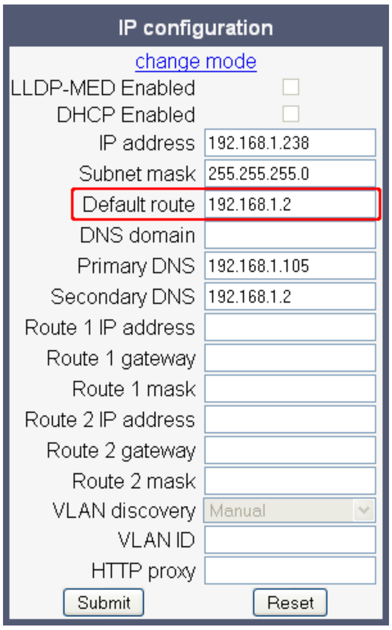
|--- Administration
    |--- Network
        |--- IP Configuration
            |--- **IP address**
            |--- **Subnet mask**

3.3.4 Default Route/Gateway

If not provided by DHCP dynamically (see Section 3.3.2, “Use DHCP”), enter the IP address of the router that links your IP network to other networks. If the value was assigned by DHCP, it can only be read.

Administration via WBM

Network > IP configuration



Administration via Local Phone

|--- Administration
    |--- Network
        |--- IP Configuration
            |--- **Route (default)**

3.3.5 Specific IP Routing

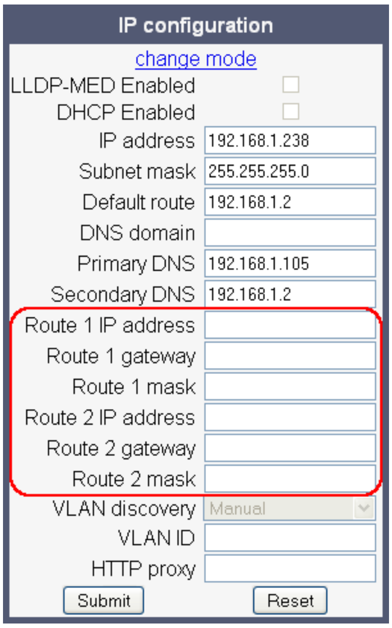
To have constant access to network subscribers of other domains, you can enter a total of two more network destinations, in addition to the default route/gateway. This is useful if the LAN has more than one router or if the LAN is divided into subnets.

Data required

- **Route 1/2 IP address**: IP address of the selected route.
- **Route 1/2 gateway**: IP address of the gateway for the selected route.
- **Route 1/2 mask**: Network mask for the selected route.

Administration via WBM

Network > IP configuration



Administration via Local Phone

```
|— Administration
    |— Network
        |— IP Configuration
            |— Route 1 IP
            |— Route 1 gateway
            |— Route 1 mask
            |— Route 2 IP
            |— Route 2 gateway
            |— Route 2 mask
```

3.3.6 DNS

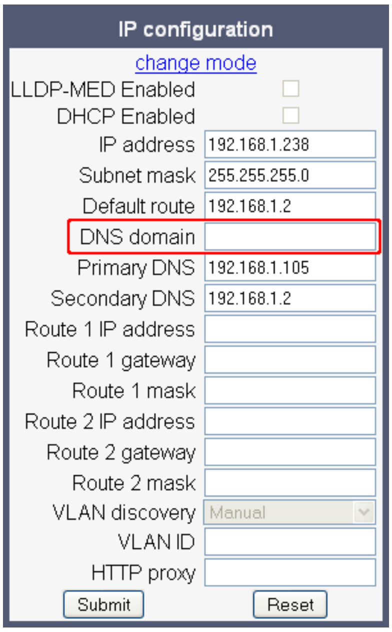
The main task of the domain name system (DNS) is to translate domain names to IP addresses. For some features and functions of the OpenStage phone, it is necessary to configure the DNS domain the phone belongs to, as well as the nameservers needed for DNS resolving.

3.3.6.1 DNS Domain Name

This is the name of the phone’s local domain.

Administration via WBM

Network > IP configuration



Administration via Local Phone

|--- Administration
  |--- Network
    |--- IP Configuration
      |--- **DNS domain**

3.3.6.2 DNS Servers

If not provided by DHCP automatically, a primary and a secondary DNS server can be configured.

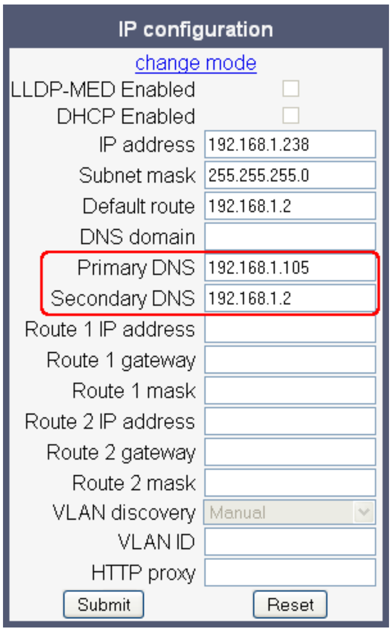
> With firmware V2, enhanced survivability using DNS SRV is available. To make use of it, a special configuration is required. For details, please refer to Section 3.5.9, “Resilience and Survivability”.

Data required

- **Primary DNS**: IP address of the primary DNS server.
- **Secondary DNS**: IP address of the secondary DNS server.

Administration via WBM

Network > IP configuration



Administration via Local Phone

```
|— Administration
    |— Network
        |— IP Configuration
            |— Primary DNS
            |— Secondary DNS
```

3.3.6.3 Terminal Hostname (V2)

With OpenStage firmware V2, the phone’s hostname for registration with the DNS server can be customised. The phone will send the specified hostname to the DNS server using DDNS. Therefore, the DNS server must support DDNS.

The corresponding DNS domain is configured in Network > IP configuration > DNS domain (see Section 3.3.6.1, “DNS Domain Name”).

The current DNS name of the phone is displayed at the right-hand side of the banner of the admin and user web pages, under **DNS name**. To see configuration changes, the web page must be reloaded.

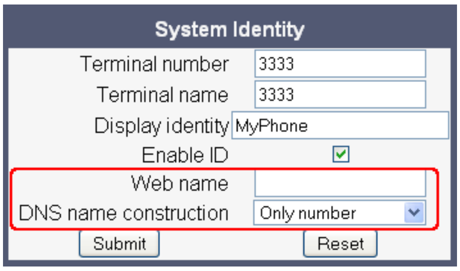
> It is recommended to inform the user about the DNS name of his/her phone. The complete WBM address can be found under User menu > Network information > Web address.

The DNS name can be constructed from pre-defined parameters and free text. Its composition is defined by the **DNS name construction** parameter. The following options are available:

- "None": The phone does not attempt to change its DNS name via DDNS.
- "MAC based": The DNS name is built from the prefix "OIP" followed by the phone’s MAC address.
- "Web name": The DNS name is set to the the string entered in **Web name**.
- "Only number": The DNS name is set to the **Terminal number**, that is, the phone’s call number (see Section 3.5.1, “Terminal and User Identity”).
- "Prefix number": The DNS name is constructed from the the string entered in **Web name**, followed by the **Terminal number**.

Administration via WBM

System > System Identity



Administration via Local Phone

```
|— Administration
     |— Identity
          |— Web name
          |— DDNS hostname
```

3.3.7 Configuration & Update Service (DLS)

The Deployment Service (DLS) is a HiPath Management application for administering workpoints in both HiPath and non-HiPath networks. Amongst the most important features are: security (e.g. PSS generation and distribution within an SRTP security domain), mobility for optiPoint and OpenStage SIP phones, software deployment, plug&play support, as well as error and activity logging.

DLS address, i.e. the IP address or hostname of the DLS server, and **DLS port**, i.e. the port on which the DLS server is listening, are required to enable proper communication between phone and DLS.

The **Contact gap** parameter controls a security function. It specifies a minimum time interval that must elapse between individual HTTP requests from the phone which are responding to a ContactMe request from the DLS. The ContactMe request is sent by the DLS each time the DLS wants to execute an action on the phone, e. g. software deployment, or a configuration change. Any requests coming within that time will be ignored. The purpose is to prevent DoS (Denial of Service) attacks on the phone.

The **Security mode** determines whether the communication between the phone and the DLS is secure. A secure connection is established by exchanging credentials between the DLS and the phone for mutual authentication. After this, the communication is encrypted, and a different port is used.

> With firmware V2, it is possible to operate the DLS server behind a firewall or NAT (Network Address Translation), which prevents the DLS from sending ContactMe messages directly to the phone. Only outbound connections from the phone are allowed. To overcome this restriction, a DLS Contact-Me proxy (DCMP) can be deployed. The phone periodically polls the DCMP (DLS Contact-Me Proxy), which is placed outside of the phone's network, for pending contact requests from the DLS. If there are contact requests, the phone will send a request to the DLS in order to obtain the update, just as with a regular DLS connection.
> The URI of the DCMP, as well as the polling interval, are configured by the DLS. For this purpose, it is necessary that the phone establishes a first contact to the DLS, e. g. by phone restart or local configuration change.

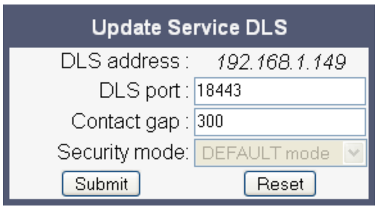
Data required

- **DLS address**: IP address or hostname of the server on which the Deployment Service is running.
- **DLS port**: Port on which the DLS Deployment Service is listening.
  Default: 18443
- **Contact gap**: Minimum time interval in seconds that must elapse between responses to a ContactMe request from the DLS, in order to prevent DoS attacks.
  Default: 300

- **Security mode** / **Security status**: Determines whether the communication between the phone and the DLS is secure.
  Value range: "Default mode", "Secure mode"
  Default: "Default"

Administration via WBM

Network > Update Service (DLS)



Administration via Local Phone

|--- Administration
    |--- Network
        |--- Update Service (DLS)
            |--- **DLS address**
            |--- **DLS port**
            |--- **Contact gap**
            |--- **Security status**

3.3.8 SNMP

The Simple Network Management Protocol is used by network management systems for monitoring network-attached devices for conditions that warrant administrative attention. An SNMP manager surveys and, if needed, configures several SNMP elements, e.g. VoIP phones.

OpenStage phones support SNMPv1.

There are currently 4 trap categories that can be sent by the phones:

Standard SNMP traps

OpenStage phones support the following types of standard SNMP traps, as defined in RFC 1157:

- **coldStart**: sent if the phone does a full restart.
- **warmStart**: sent if only the phone software is restarted.
- **linkUp**: sent when IP connectivity is restored.

QoS Related traps

These traps are designed specifically for receipt and interpretation by the QDC collection system. The traps are common to SIP phones, HFA phones, Gateways, etc.

Traps for important high level SIP related problems

Currently, these traps are related to problems in registering with a SIP Server and to a failure in remotely logging off a mobile user. These traps are aimed at a non-expert user (e.g. a standard Network Management System) to highlight important telephony related problems.

Traps specific to OpenStage phones

Currently, the following traps are defined:

TraceEventFatal: sent if severe trace events occur; aimed at expert users.

TraceEventError: sent if severe trace events occur; aimed at expert users.

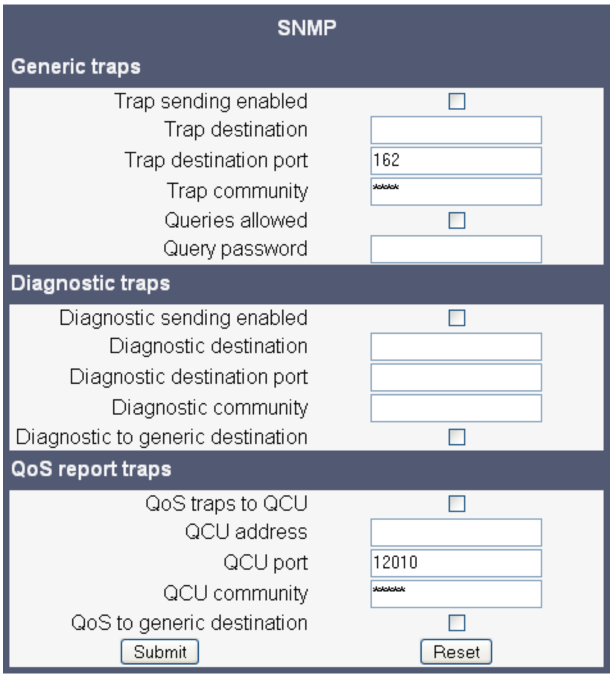
Data required

- **Trap sending enabled**: Enables or disables the sending of a TRAP message to the SNMP manager.
  Value range: "Yes", "No"
  Default: "No"
- **Trap destination**: IP address or hostname of the SNMP manager that receives traps.
- **Trap destination port**: Port on which the SNMP manager is receiving TRAP messages.
- Default: 162

- **Trap community**: SNMP community string for the SNMP manager receiving TRAP messages.
  Default: "snmp"
- **Queries allowed**: Allows or disallows queries by the SNMP manager.
- **Query password**: Password for the execution of a query by the SNMP manager.
- **Diagnostic sending enabled**: Enables or disables the sending of diagnostic data to the SNMP manager.
  Value range: "Yes", "No"
  Default: "No"
- **Diagnostic destination**: IP address or hostname of the SNMP manager receiving diagnostic data.
- **Diagnostic destination port**: Port on which the SNMP manager is receiving diagnostic data.
- **Diagnostic community**: SNMP community string for the SNMP manager receiving diagnostic data.
- **QoS traps to QCU**: Enables or disables the sending of TRAP messages to the QCU server.
  Value range: "Yes", "No"
  Default: "No"
- **QCU address**: IP address of the QCU server.
- **QCU port**: Port on which the QCU server is listening for messages.
  Default: 12010.
- **QCU community**: QCU community string.
  Default: "QOSCD".
- **QoS to generic destination / QoS to generic device**: Enables or disables the sending of QoS traps to a generic destination.
  Value range: "Yes", "No"
  Default: "No"

Administration via WBM

System > SNMP



Administration via Local Phone

```
|— Administration
   |— System
      |— SNMP
         |— Queries allowed
         |— Query password
         |— Trap sending enabled
         |— Trap destination
         |— Trap destination port
         |— Trap community
         |— Diag sending enabled
         |— Diag destination
         |— Diag destination port
         |— Diag community
         |— QoS traps to QCU
         |— QCU address
         |— QCU port
         |— QCU community
         |— QoS to generic device
```

3.4 Security

OpenStage phones support secure speech transmission via SRTP. For enabling secure calls, a TLS connection to the OpenScape Voice server is required.

If **Use secure calls** is activated , the encryption of outgoing calls is enabled, and the phone is capable of receiving encrypted calls. When the phone is connected to an OpenScape Voice system, call security is communicated to the user as follows:

- An icon in the call view tells the user whether a call is secure or not.
- If an active call changes from secure to insecure, e. g. after a transfer, a popup window and an alert tone will notify the user.

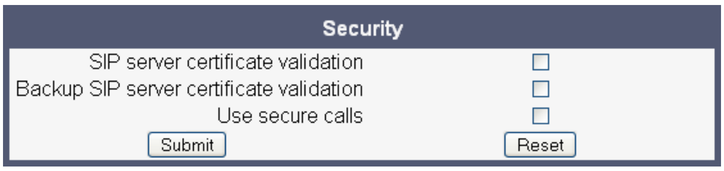
> For secure calls, it is required that both endpoints support SRTP. The secure call indication tells the user that the other endpoint has acknowledged the secure connection.

> In order to use SRTP, the phone must be configured for NTP (for further information please see Section 3.5.4, “Date and Time”). The reason is that the key generation (MIKEY) uses the system time of the particular device as a basis. Thus, encryption will only work correctly if all devices have the same UTC time.

If **SIP server certificate validation** resp. **Backup SIP server certificate validation** is activated, the phone will validate the server certificate sent by the OpenScape Voice server in order to establish a TLS connection. The server certificate is validated against the root certificate from the trusted certificate authority (CA), which must be stored on the phone first. For delivering the root certificate, a DLS (Deployment Software) server is required.

Administration via WBM (up to V2R2)

System > Security



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Security
               |--- Server cerfificate
               |--- Backup certificate
               |--- Use secure calls
```

3.4.1 Authentication Policy (V2R2 onwards)

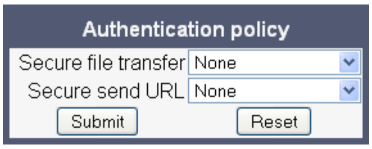
For individual certificates provided by specific servers, the level of authentication can be configured. When "None" is selected, no certificate check is performed. With "Trusted", the certificate is only checked against the signature credentials provided by the remote entity for signature, and the expiry date is checked. When "Full" is selected, the certificate is fully checked against the credentials provided by the remote entity for signature, the fields must match the requested subject/usage, and the expiry date is checked.

Secure file transfer sets the authentication level for the HTTPS server to be used (see Section 3.14.2, “Common FTP/HTTPS Settings”).

Secure send URL sets the authentication level for the server to which special HTTP requests are sent on key press ("Send URL" function, see Section 3.7.29, “Send Request via HTTP/HTTPS (V2)”).

Administration via WBM

Security and Policies > Certificates > Authentication policy



Administration via Local Phone (Version V2R2 onwards)

```
|--- Administration
     |--- Security & policie
          |--- Certificates
               |--- Authentication policy
                    |--- Secure file transfer
                    |--- Secure send URL
```

3.5 System Settings

3.5.1 Terminal and User Identity

3.5.1.1 Terminal Identity

Within a SIP environment, both Terminal Number and Terminal Name may serve as a phone number. The values are used in the userinfo part of SIP URIs.

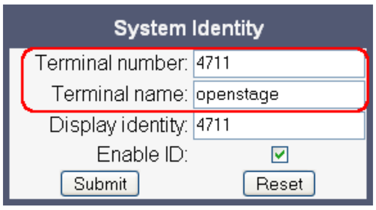
In order to register with a SIP registrar, the phone sends REGISTER messages to the registrar containing the contents of **Terminal number**.

Data required

- **Terminal number**: Number to be registered at the SIP registrar.
- **Terminal name**: Name to be registered at the SIP registrar.

Administration via WBM

System > System Identity



Administration via Local Phone

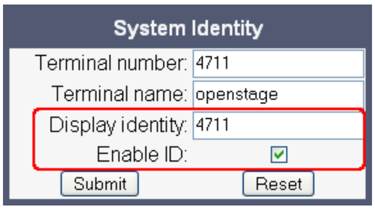
```
|-- Administration
    |-- System
        |-- Identity
            |-- Terminal number
            |-- Terminal name
```

3.5.1.2 Display Identity

If an individual name oder number is entered as **Display identity**, and **Enable ID** is activated, it is displayed in the phone's status bar instead of the Terminal number or Terminal name.

Administration via WBM

System > System Identity



Administration via Local Phone

|--- Administration
    |--- System
        |--- Identity
            |--- **Display identity**
            |--- **Enable ID**

3.5.2 Emergency and Voice Mail

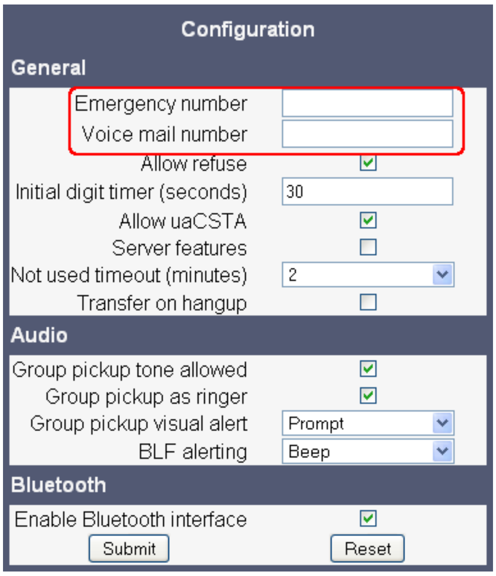
It is important to have an **Emergency number** configured. If the phone is locked, a clickable area for making an emergency call is created.

> If more than one emergency number is needed, additional numbers can be configured in the canonical dial settings (Section 3.11.1, "Canonical Dialing Configuration").

If a mailbox located at a remote server shall be used, its **Voice mail number** must be entered.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

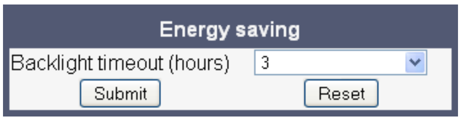
```
|— Administration
    |— System
        |— Features
            |— Configuration
                |— General
                    |— Emergency number
                    |— Voicemail number
```

3.5.3 Energy Saving (OpenStage 40/60/80)

After the phone has been inactive within the timespan specified here, the display backlight is switched off to save energy. The length of this timespan ranges from 2 hours to 8 hours. The default value is 3 hours. With OpenStage 40 (firmware version V2R2), this parameter can also be configured by the user.

Administration via WBM

Local functions > Energy saving



Administration via Local Phone

```
|--- Administration
     |--- Local Functions
          |--- Energy saving
               |--- Backlight timeout
```

3.5.4 Date and Time

If the DHCP server in your network provides the IP address of the SNTP server, no manual configuration is necessary. If not, you have to set the **SNTP IP address** parameter manually.

For correct display of the current time, the **Timezone offset** must be set appropriately. This is the time offset from UTC (Coordinated Universal Time). If, for instance, the phone is located in Munich, Germany, the offset is +1 (or simply 1); if it is located in Los Angeles, USA, the offset is -8. For countries or areas with half-our time zones, like South Australia or India, non-integer values can be used, for example 10.5 for South Australia (UTC +10:30).

If the phone is located in a country with daylight saving, the administrator can choose whether daylight saving time is activated manually or automatically. If **Daylight saving** is enabled, and **Auto time change** is disabled, daylight saving time (DST) is in effect immediately. If **Auto time change** is enabled, daylight saving is controlled by the **Time zone** parameter. This selects the daylight saving time zone which is characterized by the start and end date for daylight saving time.

The **Difference (minutes)** provides the time difference for daylight saving time in minutes. This parameter is required also when **Auto time change** is enabled. In Germany, for instance, as in most countries, this is +60.

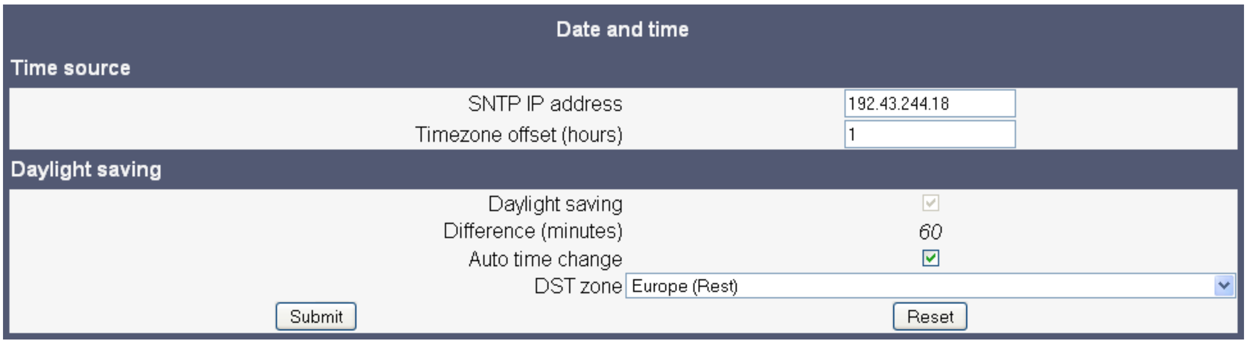
3.5.4.1 SNTP is available, but no automatic configuration by DHCP server

Data required

- **SNTP IP address**: IP address or hostname of the SNTP server.
- **Timezone offset (hours)**: Shift in hours corresponding to UTC.
- **Daylight saving**: Enables or disables daylight saving time in conjunction with **Auto time change**.
  Value range: "Yes", "No"
- **Difference (minutes)**: Time difference when daylight saving time is in effect.
- **Auto time change** / **Auto DST**: Enables or disables automatic control of daylight saving time according to the **Time zone**.
  Value range: "Yes", "No"
- **Time zone** / **DST zone**: Area with common start and end date for daylight saving time.
  Value range: "Australia 2007 (ACT, South Australia, Tasmania, Victoria)", "Australia 2007 (New South Wales)", "Australia (Western Australia)", "Australia 2008+ (ACT, New South Wales, South Australia, Tasmania, Victoria)", "Brazil", "Canada", "Canada (Newfoundland)", "Europe (Portugal, United Kingdom)", "Europe (Finland)", "Europe (Rest)", "Mexico", "United States"

Administration via WBM

Date and Time



Administration via Local Phone

|— Administration
    |— Date and Time
        |— **SNTP IP address**
        |— **Timezone offset**

3.5.4.2 No SNTP server available

If no SNTP server is available, date and time must be set manually.

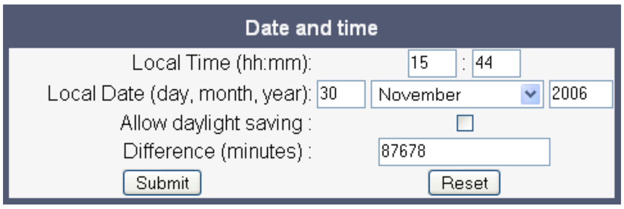
> The manual setting of time and date is located in the user menu, not in the administrator menu.

Data required

- **Local time (hh:mm)**: Local time.
- **Local date (day, month, year)**: Local date.
- **Allow daylight saving**: Defines whether there is daylight is set.
- **Difference (minutes)**: Timezone offset in minutes.

Administration via WBM

(User pages >) Date and time



Administration via Local Phone

```
|— Menu
    |— Date and Time
        |— Time
        |— Date
        |— Daylight saving
        |— Difference (mins)
```

3.5.5 SIP Addresses and Ports

3.5.5.1 SIP Addresses

In this group of parameters, the IP addresses or host names for the SIP server, the SIP registrar, and the SIP gateway are defined.

SIP server address provides the IP address or host name of the SIP proxy server (OpenScape Voice). This is necessary for outgoing calls. **SIP registrar address** contains the IP address or host name of the registration server, to which the phone will send REGISTER messages. When registered, the phone is ready to receive incoming calls. **SIP gateway address** gives the IP address or host name of the SIP gateway. If configured, the SIP gateway is used for outgoing calls; otherwise the server specified in **SIP server address** is used. A SIP gateway is able to perform a conversion of SIP to TDM, which enables to send calls directly into the public network.

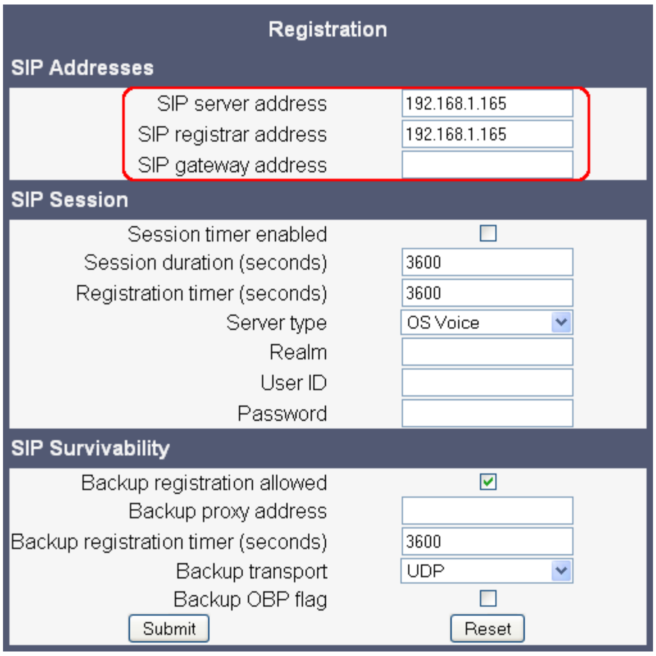
> With firmware V2, enhanced survivability using DNS SRV is available. To make use of it, a special configuration is required. For details, please refer to Section 3.5.9, “Resilience and Survivability”.

Data required

- **SIP server address**: IP address or host name of the SIP proxy server.
- **SIP registrar address**: IP address or host name of the registration server.
- **SIP gateway address**: IP address or host name of the SIP gateway.

Administration via WBM

System > Registration



Administration via Local Phone

```
|— Administration
    |— System
        |— Registration
            |— SIP Addresses
                |— SIP server
                |— SIP registrar
                |— SIP gateway
```

3.5.5.2 SIP Ports

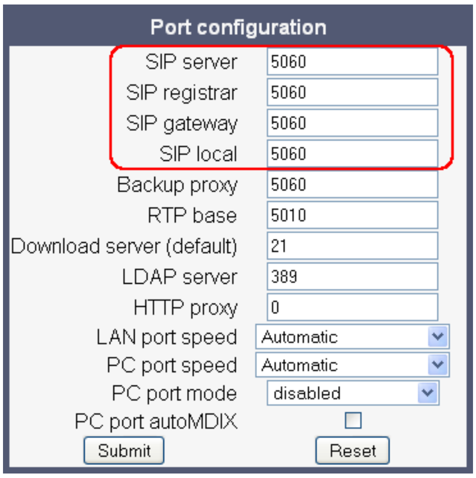
In this group of parameters, the ports for the SIP server, the SIP registrar, and the SIP gateway are defined (for further information see Section 3.5.5.1, “SIP Addresses”), as well as the SIP port used by the phone (**SIP local**).

Data required

- **SIP server**: Port of the SIP proxy server.
  Default: 5060.
- **SIP registrar**: Port of the server at which the phone registers.
  Default: 5060.
- **SIP gateway**: Port of the SIP gateway.
  Default: 5060.
- **SIP local**: Port used by the phone for sending and receiving SIP messages.
  Default: 5060.

Administration via WBM

Network > Port configuration



Administration via Local Phone

```
|— Administration
   |— Network
      |— Port Configuration
         |— SIP server
         |— SIP registrar
         |— SIP gateway
         |— SIP local
```

3.5.6 SIP Registration

Registration is the process by which centralized SIP Server/Registrars become aware of the existence and readiness of an endpoint to make and receive calls. The phone supports a number of configuration parameters to allow this to happen. Registration can be authenticated or un-authenticated depending on how the server and phone is configured.

For operation with an OpenScape Voice server, set **Server type** to "OS Voice". The expiry time of a registration can be specified by **Registration timer**.

Unauthenticated Registration

For unauthenticated registration, the following parameters must be set on the phone: Terminal number or Terminal name (see Section 3.5.1.1, “Terminal Identity”), SIP server and SIP registrar address (see Section 3.5.5.1, “SIP Addresses”).

In unauthenticated mode, the server must pre-authenticate the user. This procedure is server specific and is not described here.

Authenticated Registration

The phone supports the digest authentication scheme and requires some parameters to be configured in addition to those for unauthenticated registration. By providing a **User ID** and a **Password** which match with a corresponding account on the SIP registrar, the phone authenticates itself. Optionally, a **Realm** can be added. This parameter specifies the protection domain wherein the SIP authentication is meaningful. The protection domain is globally unique, so that each protection domain has its own arbitrary usernames and passwords.

> A challenge from the server for authentication information is not only restricted to the REGISTER message, but can also occur in response to other SIP messages, e. g. INVITE.

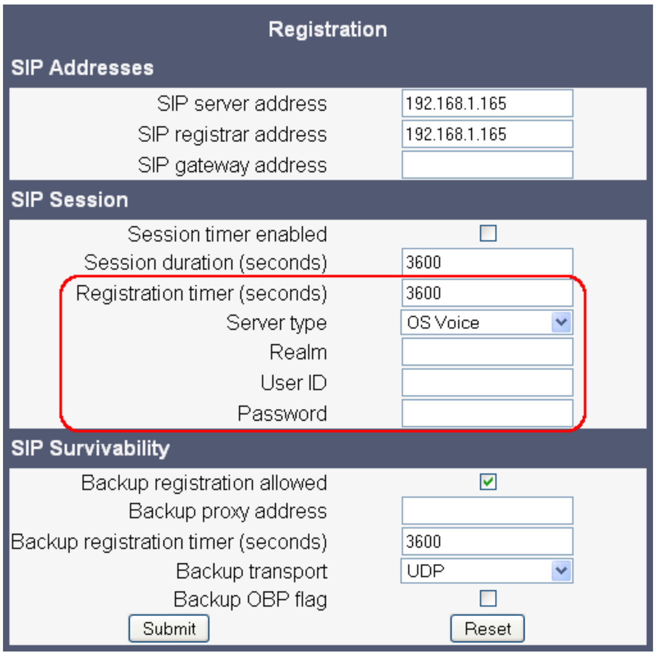
> If registration has not succeeded at startup or registration fails after having been previously successfully registered the phone will try to re-register every 30 seconds. This is not configurable.

Data required

- **Registration timer (seconds)**: Expiry time of the registration in seconds.
  Default value: 3600.
- **Server type**: Type of server the phone will register to.
  Value range: "Other", "OS Voice"
  Default value: "OS Voice"
- **Realm**: Protection domain for authentication.
- **User ID**: Username required for an authenticated registration.
- **Password**: Password required for an authenticated registration.

Administration via WBM

System > Registration



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Registration
               |--- SIP Session
                    |--- Registration timer
                    |--- Server type
                    |--- Realm
                    |--- User ID
                    |--- Password
```

3.5.7 SIP Communication

3.5.7.1 Outbound Proxy

If this option set to "Yes", the phone routes outbond requests to the configured proxy. The outbound proxy will fulfill the task of resolving the domain contained in the SIP request. If "No" is set, the phone will attempt to resolve the domain by itself.

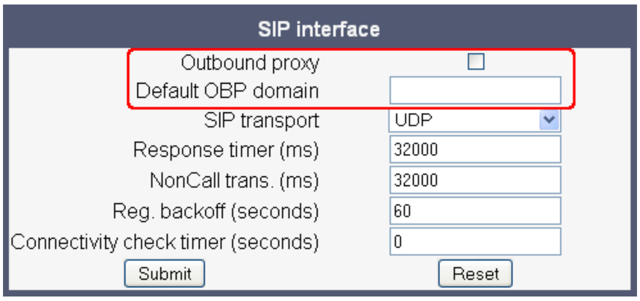
If a **Default OBP** (Outbound Proxy) **domain** is set and the number or name dialed by the user does not provide a domain, this value will be appended to the name or number. Otherwise, the domain of the outbound proxy will be appended.

Data required

- **Outbound proxy**: Determines whether an outbound proxy is used or not.
  Value range: "Yes", "No"
  Default: "No"
- **Default OBP domain**: Alternative value for the domain that is given in the outbound request.

Administration via WBM

System > SIP interface



Administration via Local Phone

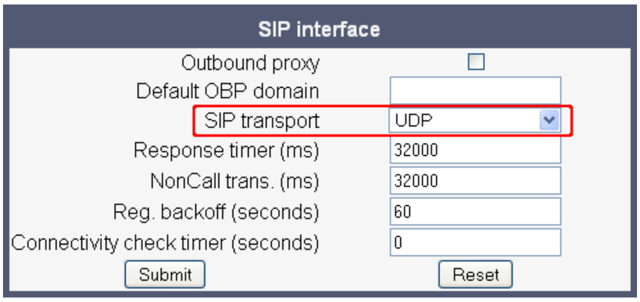
|— Administration
  |— System
    |— SIP Interface
      |— **Outbound proxy**
      |— **Default OBP domain**

3.5.7.2 SIP Transport Protocol

Selects the transport protocol to be used for SIP messages. The values "UDP", "TCP", and "TLS" are available. The default is "UDP".

Administration via WBM

System > SIP interface



Administration via Local Phone

|--- Administration
    |--- System
        |--- SIP Interface
            |--- **SIP transport**

3.5.8 SIP Session Timer

Session timers provide a basic keep-alive mechanism between 2 user agents or phones. This mechanism can be useful to the endpoints concerned or for stateful proxies to determine that a session is still alive. This is achieved by the phone sending periodic re-INVITEs to keep the session alive. If no re-INVITE is received before the interval passes, the session is considered terminated. Both phones are supposed to terminate the call, and stateful proxies can remove any state for the call.

This feature is sufficiently backward compatible such that only one end of a call needs to implement the SIP extension for it to work.

The parameter **Session timer enabled** determines whether the mechanism shall be used, and **Session duration (seconds)** sets the expiration time, and thus the interval between refresh re-INVITEs.

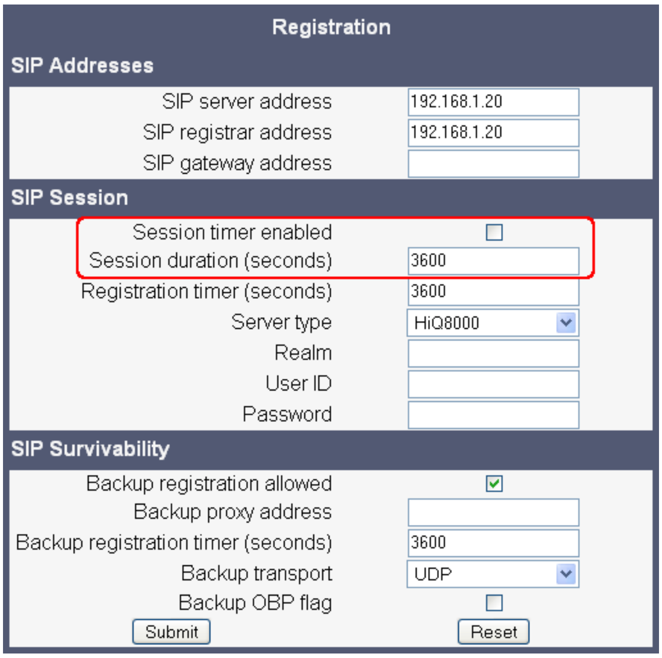
> Some server environments support their own mechanism for auditing the health of a session. In these cases, the **Session timer** must be deactivated.

Data required

- **Session timer enabled**: Activates or deactivates the session timer mechanism.
  Value range: "Yes", "No"
  Default value: "No"
- **Session duration (seconds)**: Sets the expiration time for a SIP session.
  Default: 3600

Administration via WBM

System > Registration



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Registration
               |--- SIP session
                    |--- Session timer
                    |--- Session duration
```

3.5.9 Resilience and Survivability

To allow for stable operation even in case of network or server failure, OpenStage phones have the capability of switching to a fallback system. The switchover is controlled by various configurable check and timeout intervals.

Survivability is achieved in 3 different ways:

1. With firmware V2, DNS SRV can be used for enhanced survivability, either in a scenario with a survivability proxy, or in a scenario with multiple primary SIP servers. The DNS server provides the phone with a prioritized list of SIP servers via DNS SRV. The phone fetches this list periodically from the server, depending on the TTL (time to live) specified for the DNS SRV records.
   To enable DNS SVR requests from the phone, please make the following settings:
   - Specifiy the IP address of the DNS server that provides the server list via DNS SRV. The web interface path is Network > IP configuration > Primary DNS. For details, see Section 3.3.6.2, “DNS Servers”.
   - Enable the use of an outbound proxy for routing outbound requests. The web interface path is System > SIP interface > Outbound proxy. For details, see Section 3.5.7.1, “Outbound Proxy”.
   - Set the SIP gateway port to 0. The web interface path is Network > Port configuration > SIP gateway. Alternatively, if the SIP server otherwise specified in System > Registration > SIP server address is to be configured by DNS SRV, set the SIP erver port to 0. The web interface path is Network > Port configuration > SIP server. For details, see Section 3.5.5.2, “SIP Ports”.
   - As SIP gateway address, enter the DNS domain name for which the DNS SRV records are valid. The web interface path is System > Registration > SIP gateway address. Alternatively, if the SIP server otherwise specified in System > Registration > SIP server address is to be configured by DNS SRV, set the mentioned parmeter to the DNS domain name for which the DNS SRV records are valid. For details, see Section 3.5.5.1, “SIP Addresses”.

   A survivability proxy acts as a relay between the phone and the primary SIP server. Thus, the address of the survivability proxy is specified as gateway or SIP server at the phone (see Section 3.5.6, “SIP Registration”). When the TLS connection between the survivability proxy and the SIP server breaks down, e. g. because of server failure, the survivable proxy itself acts as a replacement for the primary SIP server. Vice versa, in case the phone can not reach the survivability proxy itself, it will register directly with the primary SIP server, provided that it is specified in the DNS SRV server list.
   The survivability proxy notifies the phone whenever the survivability changes, so it can indicate possible feature limitations to the user. Furthermore, to enhance survivability, the phone will be kept up-to-date about the current survivability state even after a restart.

Another way to realize survivability is the use of multiple, geographically separated SIP servers. Normally, the phone is registered with that server that has the highest priority in the DNS SRV server list. If the highest priority server fails to respond to the TLS connectivity check (see Section 3.5.9.1, “TLS Connectivity Check”), the phone will register with the server that has the second highest priority.

2. Use of a Backup SIP Server. Along with the registration at the primary SIP server, the phone is registered with a backup SIP server. In normal operation, the phone uses the primary server for outgoing calls. If the phone detects that the connection to the primary SIP server is lost, it uses the backup server for outgoing calls. This connection check is realized by 2 timers; for details, see Section 3.5.9.2, “Response Timer” and Section 3.5.9.3, “Non-INVITE Transaction Timer”. For configuring the backup server, please refer to Section 3.5.9.5, “Backup SIP Server”.

In survivability mode, some features will presumably not be available. The user will be informed by a message in the Call View display.

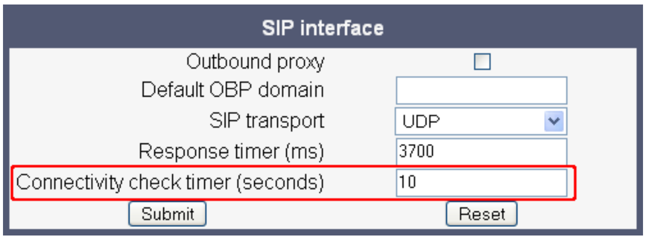
3.5.9.1 TLS Connectivity Check

A regular check ensures that the TLS link to the main SIP server is active. When the **Connectivity check timer** is set to a non-zero value, test messages will be sent at the defined interval. If the link is found to be dead, the phone uses DNS SRV to find another SIP server. Certainly, the DNS SRV records must be properly configured in the DNS server.

If no other primary SIP server is found via DNS SRV, the phone will switch over to a backup server for making receiving calls. For configuring the backup server, please refer to Section 3.5.9.5, “Backup SIP Server”.

Administration via WBM

System > SIP interface



3.5.9.2 Response Timer

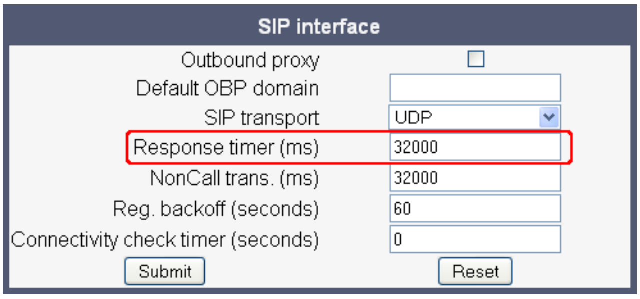
The **Response Timer** resp. **Call trans** timer is started whenever the phone sends a new INVITE message to the SIP server.

If the call transaction timer expires before the phone gets a response from the SIP server, the phone assumes that the server had died and then attempts to contact the backup server, if configured. If there is no backup server configured, the phone just tidies up internally.

The data is given in milliseconds. The default value is 32 000; for OpenScape Voice, the recommended setting is 3.7 seconds (3700 ms).

Administration via WBM

System > SIP interface



Administration via Local Phone

|--- Administration
    |--- System
        |--- SIP Interface
            |--- **Call trans. (ms)**

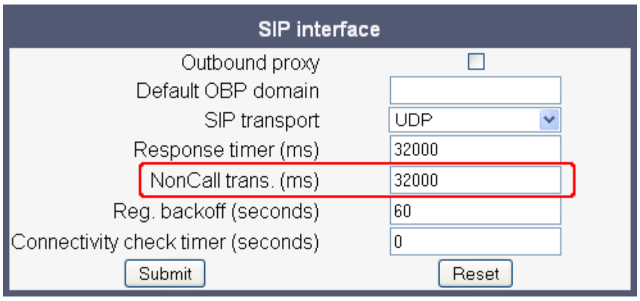
3.5.9.3 Non-INVITE Transaction Timer

The **NonCall trans** timer is started whenever the phone sends a non-INVITE message to the SIP server. If the timer expires before the phone gets a response from the SIP server, the phone assumes that the server had died and then attempts to contact the backup server, if configured. If no backup server is configured, the phone will just tidy up internally.

The data is given in milliseconds. The default value is 32 000; for OpenScape Voice, the recommended setting is 6 seconds (6000 ms).

Administration via WBM

System > SIP interface



Administration via Local Phone

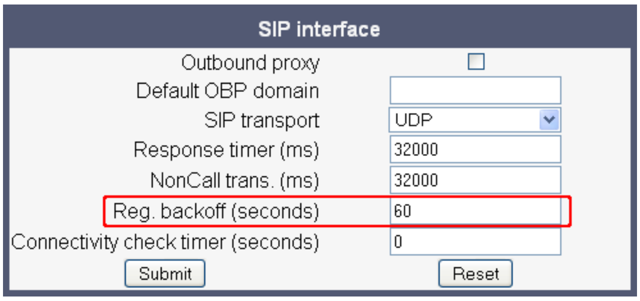
|--- Administration
    |--- System
        |--- SIP Interface
            |--- **NonCall transactions (ms)**

3.5.9.4 Maximum Registration Backoff Timer

If a registration attempt should result in a timeout, the phone waits a random time before sending another REGISTER message. The **Reg. backoff (seconds)** parameter determines the maximum waiting time.

Administration via WBM

System > SIP interface



Administration via Local Phone

|--- Administration
  |--- System
    |--- SIP Interface
      |--- **Reg. backoff**

3.5.9.5 Backup SIP Server

The **Backup registration flag** indicates whether or not the phone treats the backup proxy server as a SIP registrar. If set to "Yes", the phone tries to register its SIP address with the server whose IP address or hostname is specified by **Backup proxy address**.

The **Backup registration timer** determines the duration of a registration with the backup SIP server.

The **Backup transport** option displays the current transport protocol used to carry SIP messages to the Backup proxy server.

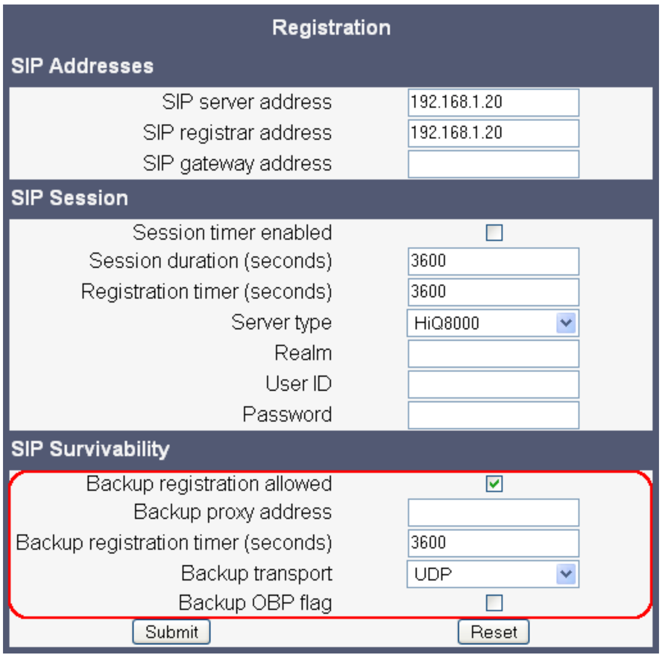
The **Backup OBP flag** indicates whether or not the Backup proxy server is used as an outbound proxy.

Data required

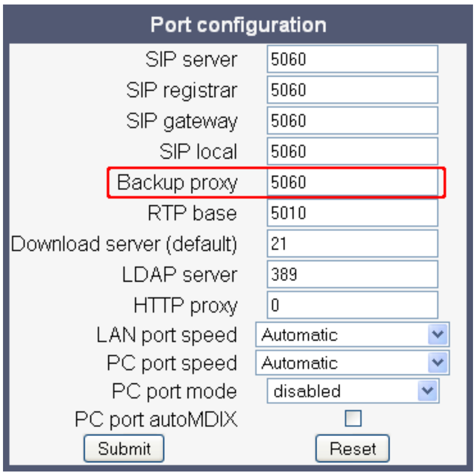
- **Backup registration allowed / Backup registration flag**: Determines whether or not the backup proxy is used as a SIP Registrar.
  Value Range: "Yes", "No"
  Default: "Yes"
- **Backup proxy address**: IP address or hostname of the backup proxy server.
- **Backup registration timer**: Expiry time of the registration in seconds.
  Default: 3600
- **Backup transport**: Transport protocol to be used for messages to the backup proxy.
  Value range: "TCP", "UDP"
  Default: "UDP"
- **Backup OBP flag**: Determines whether or not the backup proxy is used as an outbound proxy.
  Value range: "Yes", "No"
  Default: "No"
- Network > Port Configuration > **Backup proxy**: Port of the backup proxy server.
  Default: 5060

Administration via WBM

System > Registration



Network > Port configuration



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Registration
               |--- SIP Session
               |--- SIP Survivability
                    |--- Backup registration flag
                    |--- Backup proxy address
                    |--- Backup transport
                    |--- OBP flag
```

```
|--- Administration
     |--- Network
          |--- Port Configuration
               |--- Backup proxy
```

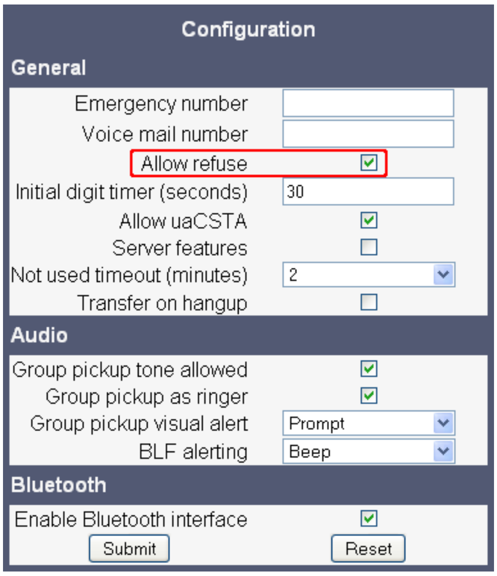
3.6 Feature Configuration

3.6.1 Allow Refuse

This parameter defines whether the Refuse Call feature is available on the phone. The possible values are "Yes" or "No". The default is "Yes".

Administration via WBM

System > Features > Configuration



Administration via Local Phone

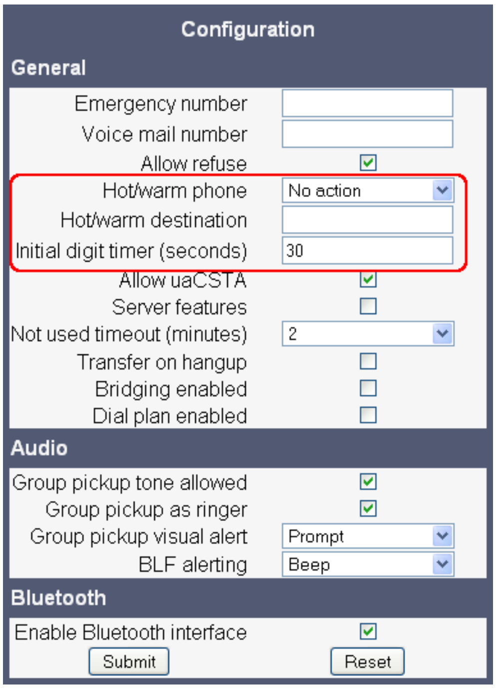
|--- Administration
    |--- System
        |--- Features
            |--- Configuration
                |--- General
                    |--- **Allow refuse**

3.6.2 Hot/Warm Phone (V2)

With firmware V2, hot/warm phone functionality is available. If the phone is configured as hot phone, the number specified in **Hot warm destination** is dialed immediately when the user goes off-hook. For this purpose, **Hot warm phone** must be set to "Hot phone". If set to "Warm phone", the specified destination number is dialed after a delay which is defined in **Initial digit timer (seconds)** (for details, see Section 3.6.3, “Initial Digit Timer”). During the delay period, the user can dial a number which will be used instead of the hot/warm destination. In addition, the user will be provided with a dial tone during the delay period. With the setting "No action", hot phone or warm phone functionality is disabled.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

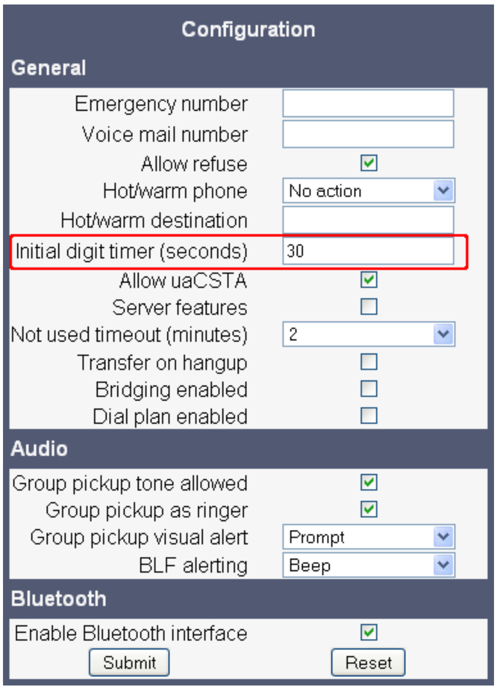
```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                         |--- Hot / warm phone
                         |--- Hot / warm destination
                         |--- Initial digit timer
```

3.6.3 Initial Digit Timer

This timer is started when the user goes off-hook, and the dial tone sounds. When the user has not entered a digit until timer expiry, the dial tone is turned off, and the phone changes to idle mode. The **Initial digit timer (seconds)** parameter defines the duration of this timespan.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

|--- Administration
    |--- System
        |--- Features
            |--- Configuration
                |--- General
                    |--- **Initial digit timer**

3.6.4 Group Pickup

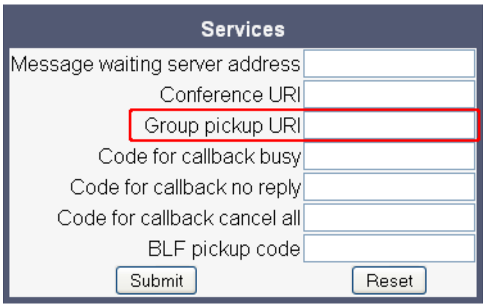
3.6.4.1 Feature Code

This feature allows a user to collect a call from any ringing phone that is in the same pickup group. To be a member of a Call Pickup group, the phone must be configured with the corresponding URI of the Call Pickup group service provided by the server. An example pickup URI is "**3".

Administration via WBM

> The BLF pickup code parameter is only relevant when the phone is connected to an Asterisk server.

System > Features > Services



3.6.4.2 Pickup alert

If desired, an incoming call for the pickup group can be indicated acoustically.

The **Group pickup tone allowed** parameter activates or deactivates the generation of an acoustic signal for incoming pickup group calls. The default is "Yes". If this is activated, **Group pickup as ringer** determines whether the current ring tone or an alert beep is used. If set to "Yes", a pickup group call will be signaled by a short ring tone; the currently selected rigtone is used. If set to "No", a pickup group call will be signaled by an alert tone. The default is "Yes".

Depending on the phone state and the setting for **Group pickup as ringer**, the group pickup tone comes from the loudspeaker, the handset, or the headset. The volumes can be set in the local user menu, under Audio > Volumes.

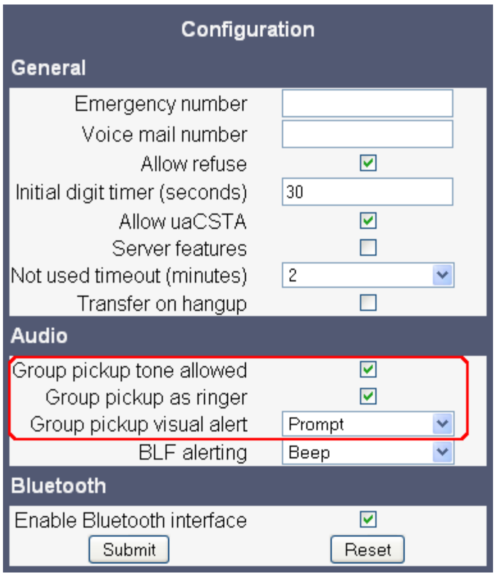
The following table shows the group pickup alert behaviour for each possible scenario:

Phone State			**Group pickup as ringer=yes**	**Group pickup as ringer=no**
Ringer on	**Idle**		Ring tone Speaker	Beep Speaker
	In call	**Handset**	Ring tone Speaker	Beep Handset
		Handset **Open listening**	Beep Handset and Speaker	Beep Handset and Speaker
		Headset	Ring tone Speaker	Beep Headset
		Headset **Open listening**	Beep Headset and Speaker	Beep Headset and Speaker
		Hands-free	Beep Speaker	Beep Speaker
Ringer off	**Idle**		Nothing	Nothing
	In call	**Handset**	Nothing	Beep Handset
		Handset **Open listening**	Beep Handset and Speaker	Beep Handset and Speaker
		Headset	Nothing	Beep Headset
		Headset **Open listening**	Beep Headset and Speaker	Beep Headset and Speaker
		Hands-free	Beep Speaker	Beep Speaker

Group pickup visual alert defines the user action required to accept a pickup call. If "Prompt" is selected, an incoming pickup call is signaled by an alert on the phone GUI. As soon as the user goes off-hook or presses the speaker key, the pickup call is accepted. Alternatively, the user can press the corresponding function key, if configured. If "Notify" is selected, an incoming pickup call is signaled by an alert on the phone GUI. To accept the call, the user must confirm the alert or press the corresponding function key, if configured.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

|--- Administration
    |--- System
        |--- Features
            |--- Group pickup
                |--- **Group pickup tone**
                |--- **Group pickup as ringer**
                |--- **Group pickup visual**

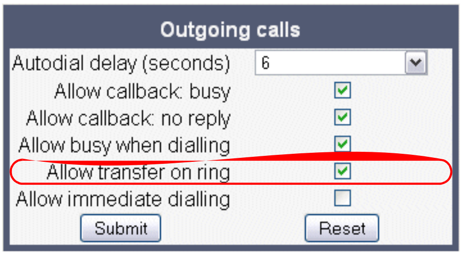
3.6.5 Call Transfer

3.6.5.1 Transfer on Ring

If this function is active, a call can be transferred after the user has dialled the third participant's number, but before the third party has answered the call. This feature is enabled or disabled in the User menu. The default is "Yes".

Administration via WBM

(User) Configuration > Outgoing calls



Administration via Local Phone

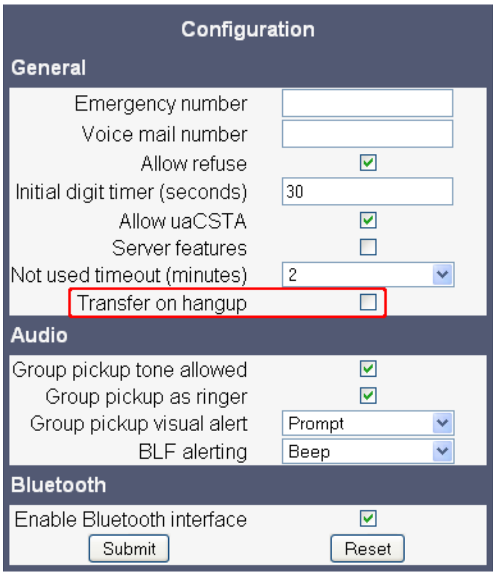
|— User
    |— Configuration
        |— Outgoing calls
            |— **Transfer on ring**

3.6.5.2 Transfer on Hangup

This feature applies to the following scenario: While A is talking to B, C calls A. A accepts the call, so B is on hold and the call between A and C is active. If **Transfer on hangup** is enabled, and A goes on-hook, B gets connected to C. If disabled, C will be released when A hangs up, and A has the possibility to reconnect to B. By default, the feature is disabled.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

|--- Administration
    |--- System
        |--- Features
            |--- Configuration
                |--- General
                    |--- **Transfer on hangup**

3.6.6 Callback URIs

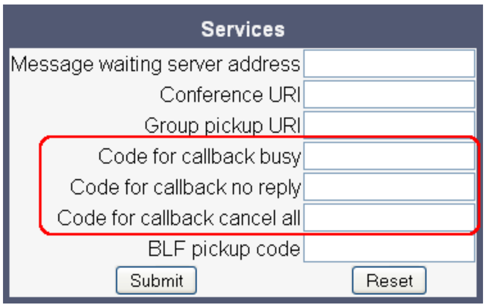
The Callback option allows the user to request a callback on certain conditions. The callback request is sent to the SIP server. The **Code for callback busy** requests a callback if the line is busy, i. e. if there is a conversation on the remote phone. **Code for callback no reply** applies when the call is not answered, i. e. if nobody lifts the handset or accepts the call in another way. The **Code for callback cancel all** all deletes all the callback requests stored previously on the telephone system/SIP server.

Data required

- **Code for callback busy** / **Callback: Busy**: Access code that is sent to the server if the line is busy.
- **Code for callback no reply** / **Callback: No reply**: Access code that is sent to the server if the callee does not reply.
- **Code for callback cancel all** / **Callback: Cancel all**: Access code for canceling all callback requests on the server.

Administration via WBM

System > Features > Services



Administration via Local Phone

```
|— Administration
   |— System
      |— Features
         |— Addressing
            |— Callback: Busy
            |— Callback: No reply
            |— Callback: Cancel all
```

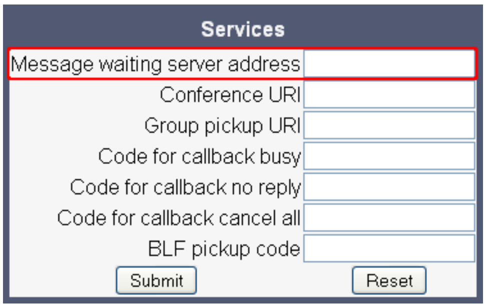
3.6.7 Message Waiting Address

The MWI (Message Waiting Indicator) is an optical signal which indicates that voicemail messages are on the server. Depending on the SIP server / gateway in use, the **Message waiting server address**, that is the address or host name of the server that sends message waiting notifications to the phone, must be configured.

With OpenScape Voice, this setting is not typically necessary for enabling MWI functionality.

Administration via WBM

System > Features > Services



Administration via Local Phone

|— Administration
  |— System
    |— Features
      |— Addressing
        |— **MWI server URI**

3.6.8 Indicate Messages (V2)

With firmware version V2, the indication of old and new messages on the display can be configured. There are 4 categories of voicemail messages: new, new urgent, old, and old urgent. For each category, the administrator can define whether the message count is shown or hidden, and set a header for the category.

Data required

- **New items**: Determines whether new items are indicated.
  Fixed Value: "Show".
- **Alternative label**: Label for new items.
- **New urgent items**: Determines whether new urgent items are indicated.
  Value range: "Show", "Hide"
- **Alternative label**: Label for new urgent items.
- **Old items**: Determines whether new urgent items are indicated.
  Value range: "Show", "Hide"
- **Alternative label**: Label for old items.
- **Old urgent items**: Determines whether old urgent items are indicated.
  Value range: "Show", "Hide"
- **Alternative label**: Label for old urgent items.

Administration via WBM

Local functions > Messages settings

Administration via Local Phone

```
|--- Administration
     |--- Locatl functions
          |--- Messages settings
               |--- New items
               |--- Alternative label
               |--- New urgent items
               |--- Alternative label
               |--- Old items
               |--- Alternative label
               |--- Old urgent items
               |--- Alternative label
```

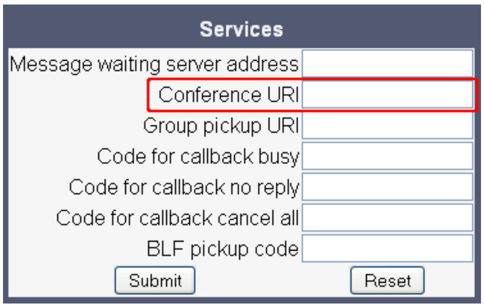
3.6.9 System Based Conference

The **Conference URI** provides the number/URI used for system based conferences, which can involve more than three members. This feature is not available with every system.

> It is recommended not to enter the full URI, but only the user part. For instance, enter "123", not "123@<SIP SERVER ADDRESS>". A full address in this place might cause a conflict when OpenScape Voice uses multiple nodes.

Administration via WBM

System > Features > Services



3.6.10 Call Recording (V2R2)

When call recording is activated, the audio of an established call is transmitted to a central voice recorder, which acts as a regular SIP endpoint. The audio mixing is done by the phone, like in a local conference. The behaviour of the phone is configurable.

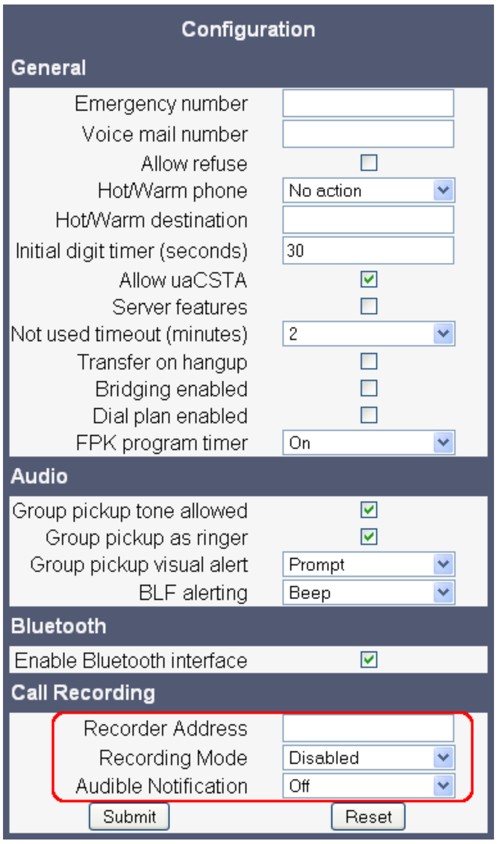
The **Recorder adress** is the SIP DN, or call number, of the voice recorder.

Recording mode determines if and how call recording will be activated. When set to "Disabled", no call will be recorded, and the corresponding FPK function (see Section 3.7.24, “Call recording (V2R2)”) is disabled. When set to "Manual", call recording can be started and stopped with the FPK function. With "Auto-start", call recording is started when a call is established, and can be stopped with the FPK function. When "All calls" is selected, call recording is started when a call is established, and can not be stopped with the FPK function.

Audible notification determines if and how the user is notified when a call is being recorded. When set to "Off", the user will not notice that a call is being recorded. When "Single-shot" is selected, a single short beep tone is played through the handset, headset or loudspeaker when call recording starts, i.e. when the connection to the voice recorder has been established. When "Repeated" is selected, a short beep tone is played repeatedly through the handset, headset or loudspeaker when call recording starts.

Administration via WBM

System > Features > Services



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                    |--- Call recording
                         |--- Recorder number
                         |--- Recording mode
                         |--- Audible notification
```

3.6.11 Server Based Features

> Please note that the **Server features** parameter, despite the name similarity, is not related to the Server feature functionality as described in Section 3.7.26, “Server Feature”.

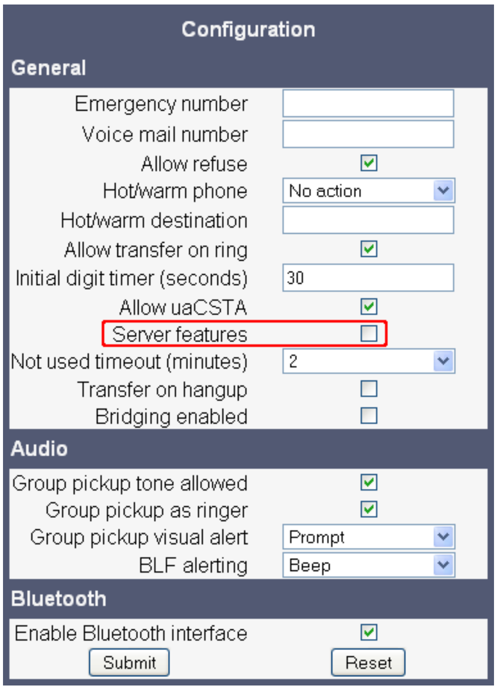
The use of server based call forwarding is enabled or disabled here. When phone based DND and phone based call forwarding are to be used, **Server features** must be deactivated. This is the default setting. For using server based Call Forwarding, it must be activated.

> Before switching **Server features** on or off, please ensure that both Call Forwarding and DND are not activated. Otherwise, the user will not be able to control the feature any more.
> It is recommended to set **Server features** when setting up the phone, and avoid further changes, as possible.

> To enable server based features, uaCSTA must be allowed (see Section 3.6.13, “uaCSTA Interface”).

3.6.12 Administration via WBM

System > Features > Configuration



Administration via Local Phone

|--- Administration
    |--- System
        |--- Features
            |--- Configuration
                |--- General
                    |--- **Server features**

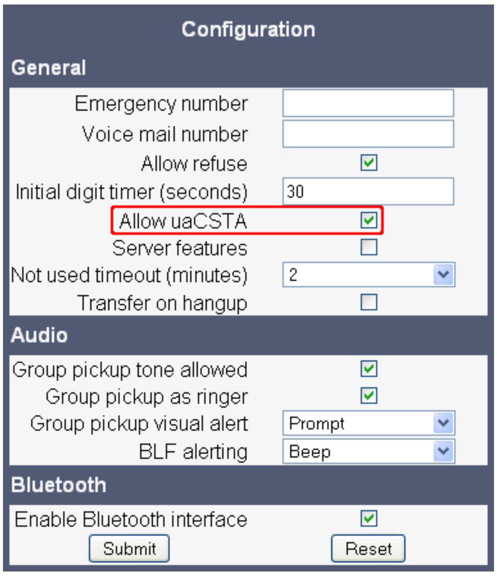
3.6.13 uaCSTA Interface

User Agent CSTA (uaCSTA) is a limited subset of the CSTA protocol, which allows external CTI applications to interact with the phone.

If **Allow uaCSTA** is enabled, applications which support the uaCSTA standard will have access to the OpenStage phone. The default is "Yes".

Administration via WBM

System > Features > Configuration



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                         |--- Allow uaCSTA
```

3.6.14 Local Menu Timeout

The timeout for the local user and admin menu is configurable. When the time interval is over, the menu is closed and the administrator/user is logged out.

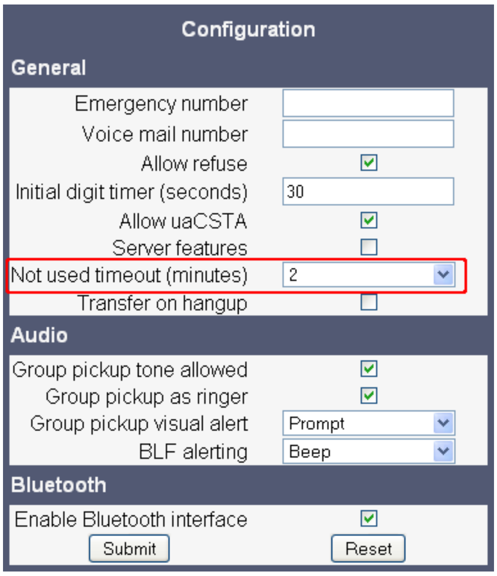
The timeout may be helpful in case a user does a long press on a line key unintentionally, and thereby invokes the key configuration menu. The menu will close after the timeout, and the key will return to normal line key operation.

> With firmware version V2, the current position in the user or admin menu is kept in case the user/admin has exited the menu, e.g. for receiving a call. Thus, if the user/ admin re-enters the menu, he is directed to exactly that submenu, or parameter, which he had been editing before.

The timeout ranges from 1 to 5 minutes. The default value is 2.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                         |--- Not used timeout
```

3.7 Free Programmable Keys

OpenStage 15/40/60/80 phones feature free programmable keys (FPKs) which can be associated with special phone functions.

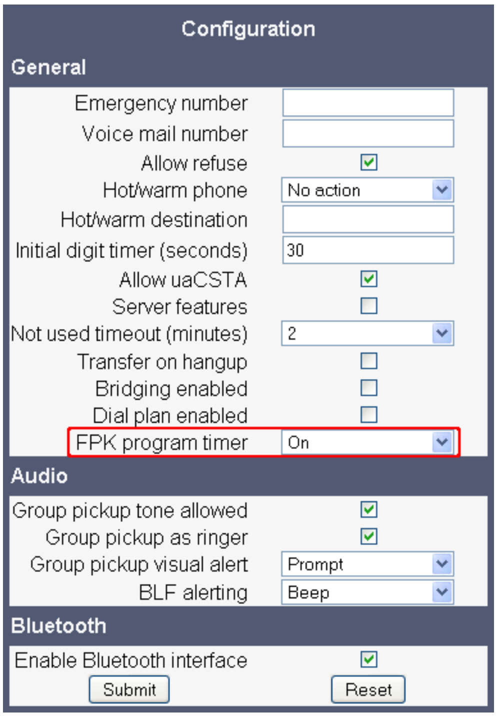
In the Administrator pages of the WBM, the programmable keys menu can be accessed via System > Features > Program keys.

At the phone, the configuration menu for a specific key is called by a long press on the related key. With firmware version V2R1, this can be disabled by deactivating **FPK program timer**. When this parameter is disabled, it is not possible to enter programming mode by long key press. However, the other methods for key programming remain enabled.

The functions available and their parameters are described in the following sub-sections. For keyset and DSS functionality, please refer to Section 3.9, “Multiline Appearance/Keyset”.

Administration via WBM (V2R1)

System > Features > Configuration > General



Administration via Local Phone

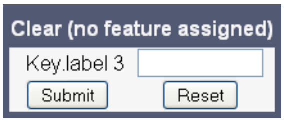
```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                         |--- FPK prog. timer
```

3.7.1 Clear (no feature assigned)

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys



3.7.2 Selected Dialing

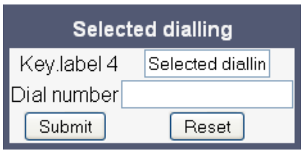
On key press, a pre-defined call number is called.

The label displayed to the left of the key is defined in **Key label <key number>**.

The call number defined in the **Dial number** parameter is dialed on key press.

Administration via WBM

System > Features > Program keys > Selected dialling



3.7.3 Repeat Dialing

On key press, the call number that has been dialed lastly is dialed again.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Repeat dialling

3.7.4 Call Forwarding

This key function controls phone based call forwarding. If forwarding is enabled, the phone will forward incoming calls to the predefined call number, depending on the current situation.

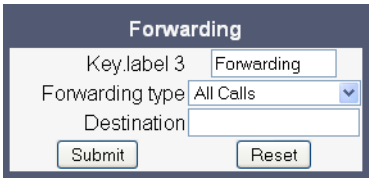
> To use phone based call forwarding, **Server features** must be switched off (see Section 3.6.11, “Server Based Features”).

The label displayed to the left of the key is defined in **Key label <key number>**.

The **Forwarding type** parameter determines the forwarding behaviour. If "All calls" is selected, any incoming call will be forwarded. If "On no reply" is set, the call will be forwarded when the user has not answered within a specified timespan. The timespan is configured in the WBM user pages under Configuration > Incoming calls > Forwarding > No replay delay (seconds). If "On busy" is selected, incoming calls will be forwarded when the phone is busy.

Administration via WBM

System > Features > Program keys > Forwarding



3.7.5 Ringer Off

Turns off the ring tone. Incoming calls are indicated via LEDs and display only.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Ringer off

3.7.6 Hold

The call currently selected or active is put on hold.

With firmware version V2R1, a held call can be retrieved by pressing the key a second time.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Hold

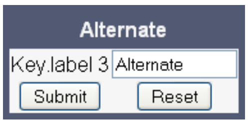
3.7.7 Alternate

Toggles between two calls; the currently active call is put on hold.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Alternate



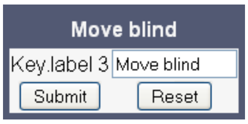
3.7.8 Blind Call Transfer / Move Blind

A call is transferred without consultation, as soon as the phone goes on-hook or the target phone goes off-hook.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Move blind



3.7.9 Join Two Calls

Call transfer, applicable when there is one active call and one call on hold. The active call and the held call are connected to each other, while the phone that has initiated the transfer is disconnected.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Join

3.7.10 Deflect a Call

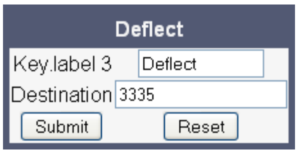
On key press, an incoming call is deflected to the specified destination.

The label displayed to the left of the key is defined in **Key label <key number>**.

The target destination is defined in the **Destination** parameter.

Administration via WBM

System > Features > Program keys > Deflect



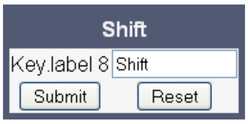
3.7.11 Shift Level

Shift the level for the programmable keys. When activated, the functions assigned to the shifted level are available on the keys.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Shift



3.7.12 Phone-Based Conference

Establishes a three-party conference from an active call and held call.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Conference

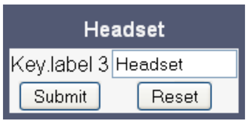
3.7.13 Accept Call via Headset (OpenStage 40/60/80)

On key press, an incoming call is accepted via headset.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Headset



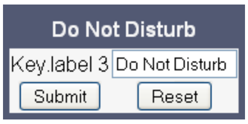
3.7.14 Do Not Disturb

If this feature is activated, incoming calls will not be indicated to the user.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Do Not Disturb



3.7.15 Group Pickup

On key press, a call for a different destination within the same pickup group is answered.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Group pickup

3.7.16 Repertory Dial

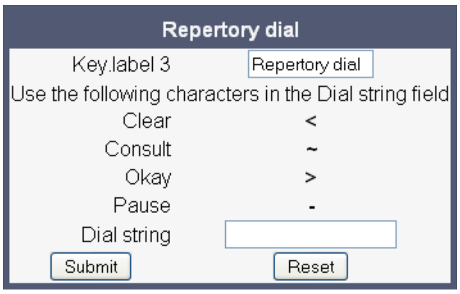
This feature is similar to the selected dialing function, but additionally, special calling functions are possible. The desired number and/or function is selected via the **Dial string** parameter.

The following call functions are available:
- "<" disconnect a call.
- "~" start a consultation call. Example: "~3333>"
- ">" (preceded by a call number) start a call. Example: "3333>"
- "-" enter a pause, e. g. for exit-code or international dialing. Example: "0-011511234567>"

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Repertory dial



3.7.17 Hunt Group: Send Busy Status

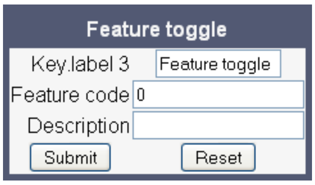
This feature is relevant for hunt groups. If the user is a member of a hunt group and wants another member of the hunt group to pick up an incoming call, he can signal Busy status using the Feature toggle function.

The label displayed to the left of the key is defined in **Key label <key number>**.

The **Feature code** parameter is the OpenScape Voice code for Busy status. In the **Description** field, an appropriate description for the feature can be entered.

Administration via WBM

System > Features > Program keys > Feature toggle



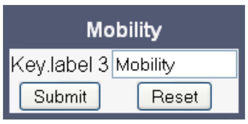
3.7.18 Mobile User Logon

The mobility feature enables users to transfer their personal settings, such as their key layout, or personal phonebook, from one phone to another. The data is stored and managed by the DLS (Deployment Service).

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Mobility



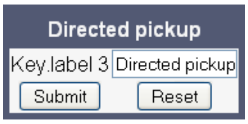
3.7.19 Directed Pickup

This feature enables the user to pick up a call which is ringing at another phone. On pressing the key, a menu opens which requests the call number of the target phone.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Directed pickup



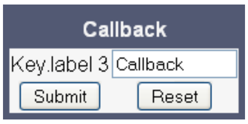
3.7.20 Callback

When the remote phone called is busy does not reply, the user can send a callback request to the server by pressing this key.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Callback



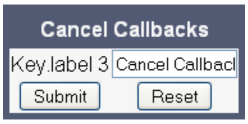
3.7.21 Cancel Callbacks

With this this function, the user can cancel all callback requests on the server.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Cancel callbacks



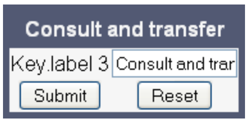
3.7.22 Consult and Transfer

When the phone is engaged in an active call, this function opens a dialing menu to make a consulation call.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Consult and transfer



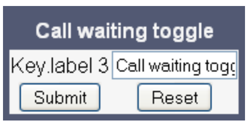
3.7.23 Toggle Call Waiting

Enables or disables the call waiting feature. If enabled, calls from a third party are allowed during an active call.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Call waiting toggle



3.7.24 Call recording (V2R2)

Starts or stops call recording (for configuring call recording, see Section 3.6.10, “Call Recording (V2R2)”).

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Call recording

3.7.25 Auto Answer With Zip Tone (V2)

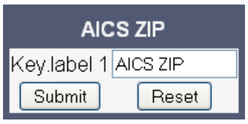
This feature is primarily designed for call centers. If activated, and a headset is used, the phone will automatically accept incoming calls without ringing and without the necessity to press a key. Moreover, additional signalling information from OpenScape Voice is not required.

To indicate a new call to the user, a zip tone is played through the headset when the call is accepted.

> The feature is available for OpenStage 40/60/80, which provide a headset jack; it only operates if the headset is plugged in. In case the key for feature activation has been pressed before the headset is connected, the feature will be automatically activated when the headset is plugged in.

Administration via WBM

System > Features > Program keys > AICS Zip tone



3.7.26 Server Feature

Invokes a feature on the SIP server. The status of the feature can be monitored via the LED associated to the key.

> This function is intended primarily for operation with an Asterisk SIP server. For details, please refer to the Administration Manual for OpenStage 15/20/40/60/80 on Asterisk.

3.7.27 BLF Key

This function offers the possibility to monitor another extension, and to pick up calls for the monitored extension.

> This function is intended primarily for operation with an Asterisk SIP server. For details, please refer to the Administration Manual for OpenStage 15/20/40/60/80 on Asterisk.

3.7.28 Start Application

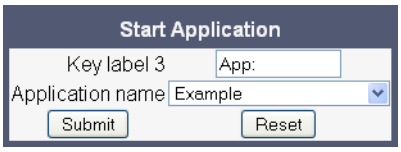
With this key, the user can start a pre-defined XML application (see Section 3.17, “Applications”). XML applications are available for OpenStage 60/80 phones.

The label displayed to the left of the key is defined in **Key label <key number>**.

The **Application name** parameter selectes the XML application to be started.

Administration via WBM

System > Features > Program keys



3.7.29 Send Request via HTTP/HTTPS (V2)

With this function, the phone can send a specific HTTP or HTTPS request to a server. The function is available at any time, irrespective of registration and call state. Possible uses are HTTP-controlled features on the system, or functions on a web server that can only be triggered by HTTP/HTTPS request, e. g. login/logout for flexible working hours.

The **Protocol** parameter defines whether HTTP or HTTPS is to be used for sending the URL to the server.

The **Web server address** is the IP address or DNS name of the remote server to which the URL is to be sent.

The **Port** is the target port at the server to which the URL is to be sent.

The **Path** is the server-side path to the desired function, i. e. the part of the URL that follows the IP address or DNS name. Example: `webpage/checkin.html`

In the **Parameters** field, one or more key/value pairs in the format `"<key>=<value>"` can be added to the request, separated by an ampersand (`&`).
Example: `phonenumber=3338&action=huntGroupLogon`

> The question mark will be automatically added between the path and the parameters. If a question mark has been entered at the start of the parameters, it will be stripped off automatically.

The **Method** parameter determines the HTTP method to be used, which can either be GET or POST. If GET is selected, the additional parameters (**Parameters**) and the user id/password (**Web server user ID**/**Web server password**) are part of the URL. If POST is selected, these data form the body of the message.

In case the web server requires user authentication, the parameters **Web server user ID** and **Web server password** can be used. If not null, the values are appended between the server-side path (**Path**) and the additional parameters (**Parameter**).

If the **LED controller URI** is given, the LED associated with this key indicates the state of the call number or SIP URI specified, provided the SIP server sends a notification:

- Busy notification: LED is glowing.
- Ringing notification: LED is blinking.
- Idle notification (state=terminated): LED is dark.

> When assigning the function described here to the release key, please consider that this key has no LED.

With firmware version V2R2, the **Push support** parameter is available. If activated, the LED is controllable by a combination of an HTTP push request and an XML document. For further information, see the XML Applications Developer’s Guide.

> If you want to use the HTTP push solution, please ensure that the **LED controller URI** field is empty. Otherwise, the phone will only use the SIP mechanism for LED control, and ignore the push request.

The **Symbolic name**, which is available with firmware version V2R2, is used to assign a push request from the application server to the appropriate free programmable key resp. fixed function key. This value must be unique for all keys involved.

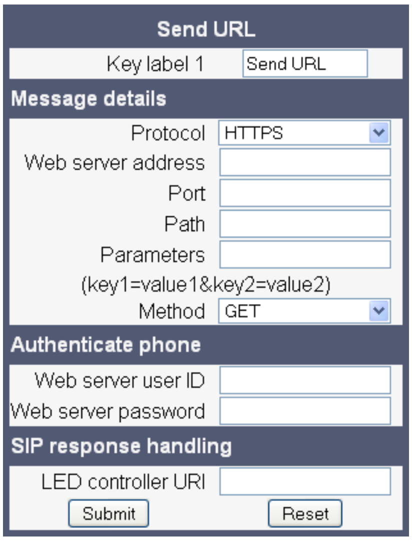
Data required

- **Key label <n>**: Label for the key.
- **Protocol**: Transfer protocol to be used.
  Value range: "HTTP", "HTTPS"
- **Web server address**: IP address or DNS name of the remote server.
- **Port**: Target port at the server.
- **Path**: Server-side path to the function.
- **Parameters**: Optional parameters to be sent to the server.
- **Method:** HTTP method used for transfer.
  Value range: "GET", "POST"
- **Web server user ID**: User id for user authentication at the server.
- **Web server password**: Password for user authentication at the server.
- **LED controller URI:** Indicates the state of the call number specified.

- **Push support** (V2R2): Enables or disables LED control by push requests from the server.
- **Symbolic name** (V2R2): Assigns a push request to the appropriate free programmable key resp. fixed function key.

Administration via WBM

System > Features > Program keys > Send URL



3.7.30 Built-in Forwarding (V2R2)

This function is equivalent to the function described in Section 3.8.1, “Programmable Call Forwarding Key (V2)”. As a programmable key function, this is relevant for OpenStage 15 phones, which have no fixed forwarding key.

System > Features > Program keys

3.7.31 Start Phonebook (OpenStage 40 with V2R1 only)

This key function opens a menu which enables the user to start the local or the corporate phonebook. For further information about the local phonebook, please refer to the user guide for OpenStage 40 phones. For information about the corparate phonebook, please see Section 3.15, “Corporate Phonebook: Directory Settings”.

Administration via WBM

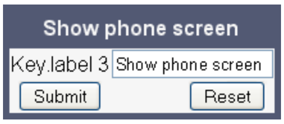
System > Features > Program keys > Start Phonebook

3.7.32 Show phone screen (OpenStage 15 and OpenStage 40 only)

On pressing this key, the phone display switches to call view mode.

Administration via WBM

System > Features > Program keys > Show phone screen



3.7.33 Mute (OpenStage 15 Only)

On pressing this key, the microphone is turned off. This programmable key function is available only for OpenStage 15 phones, which have no fixed mute key.

Administration via WBM

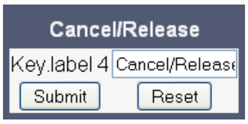
System > Features > Program keys > Mute

3.7.34 Release (OpenStage 15 Only)

On pressing this key, the current call is disconnected. This programmable key function is available only for OpenStage 15 phones, which have no fixed release key.

Administration via WBM

System > Features > Program keys > Release



3.8 Fixed Function Keys

For the forwarding key, the release key, and the voice recognition key, specific SIP or HTTP based functions can be defined. These functions can be employed as an alternative to the built-in functions.

> The programming of fixed function keys is intended primarily for operation with an Asterisk SIP server. For details, please refer to the Administration Manual for OpenStage 15/20/40/60/80 on Asterisk.

3.8.1 Programmable Call Forwarding Key (V2)

> This feature is available for all OpenStage phones except OpenStage 15, which has no forwarding key. With firmware version V2R2, a free programmable key can be configured as forwarding key (Section 3.7.30, “Built-in Forwarding (V2R2)”).

By default, the fixed forwarding key controls the phone’s built-in forwarding functionality. Alternatively, server-based forwarding can be assigned to this key. For this purpose, an appropriate feature code or DTMF signal is sent to the SIP server in order to toggle forwarding. The parameters **Feature code**, **DTMF digits**, and **LED control** are the same as with the server feature key; pleaser refer to Section 3.7.26, “Server Feature”.

3.9 Multiline Appearance/Keyset

> This feature is available only on OpenStage 15, OpenStage 40 and OpenStage 60/80 phones.

A phone that has more than one line associated to it, and therefore works as a multiline phone, is referred to as "keyset". The lines are assigned to the phone by setting up a separate line key for each line.

The multiline appearance feature allows for multiple lines to be assigned to a keyset and for a line to be assigned to multiple keysets. This feature requires configuration in OpenScape Voice and in the telephone, and is particularly useful for executive-assistant arrangements.

> In order to configure the phone as a keyset, it is required to
>
> - use an outbound proxy (System > SIP interface > Outbound proxy, see Section 3.5.7.1, “Outbound Proxy”), and
> - set the server type to "OS Voice" (System > Registration > Server type, see Section 3.5.6, “SIP Registration”).

For each keyset, a Primary Line/Main DN is required. The primary line is the dialing number for that keyset.

There are two types of line:
- **Private line**: A line that appears on only one keyset.
- **Shared line**: A line that is shared between keysets.

3.9.1 Line key configuration

> It is recommended to configure primary lines only on keys 1 to 6, or 1 to 5, if a shift key is needed. This ensures that the lines are still accessible when the user migrates to a different phone with fewer keys via the mobility feature.

A line corresponds to a SIP address of record (AoR), which can have a form similar to an E-mail address, or can be a phone number. It is defined by the **Address** parameter. For registration of the line, a corresponding entry must exist on the SIP server resp. the SIP registrar server.

A label can be assigned to the line key by setting its **Key label**.

Every keyset must necessarily have a line key for the primary line. To configure the key of the primary line, set **Primary line** to "true".

If **Ring on/off** is checked, the line will ring when an incoming call occurs, and a popup will appear on the display. If the option is not checked, the incoming call will be indicated only by the blinking of the key's LED. If it is desired that the line ring with a delay, the time interval in seconds can be configured by **Ring delay**.

When the user lifts the handset in order to initiate a call, the line to be used is determined by selection rules. To each line, a priority is assigned by the **Selection order** parameter. A line with the rank 1 is the first line to be considered for use. If more than one line have the same rank, the selection is made according to the key number. Note that **Selection order** is a mandatory setting; it is also relevant to the **Terminating line preference**, as well as to other functions.

The **Address** (Address of Record) parameter is the phone number resp. SIP name corresponding to the entry in the SIP registrar at which the line is to be registered.

> For the configuration of line keys, the use of the DLS (Deployment Service) is recommended. For operating the DLS, please refer to the DLS user's guide. Alternatively, the web interface or the local menu can be used. Note that the creation of a new line key and the configuration of some parameters can not be accomplished by the phone's local menu.
> Generally, it is advisable to restrict the user's possibilities to modify line keys. This can be achieved solely by the DLS. For further instructions, see the DLS Administration Guide.

The **Realm**, a protection domain used for authenticated access to the SIP server, works as a name space. Any combination of user id and password is meaningful only within the realm it is assigned to. The other parameters necessary for authenticated access are **User Identifier** and **Password**. For all three parameters, there must be corresponding entries on the SIP server.

The **Shared type** parameter determines whether the line is a shared line, i. e. shared with other endpoints, or a private line, i .e. available exclusively for this endpoint. A line that is configured as primary line on one phone can be configured as secondary line on other phones.

When **Allow in overview** is set to "Yes", the line will be visible in the line overview on the phone's display.

With firmware V2, hot/warm line functionality is available. If a line is configured as hot line, the number indicated in **Hot warm destination** is dialed immediately when the user goes off-hook. This number is configurerd in the user menu under **Configuration > Keyset > Lines > Hot/warm destination**. To create a hot line, **Hot warm action** must be set to "hot line". If set to "Warm phone", the specified destination number is dialed after a delay which is defined in **Initial digit timer (seconds)** (for details, see Section 3.6.3, “Initial Digit Timer”). During the delay period, it is possible for the user to dial a different number which will be used instead of the hot/warm line destination. In addition, the user will be provided with a dial tone during the delay period. With the setting "No action", the line key will not have hot line or warm line functionality.

Data required

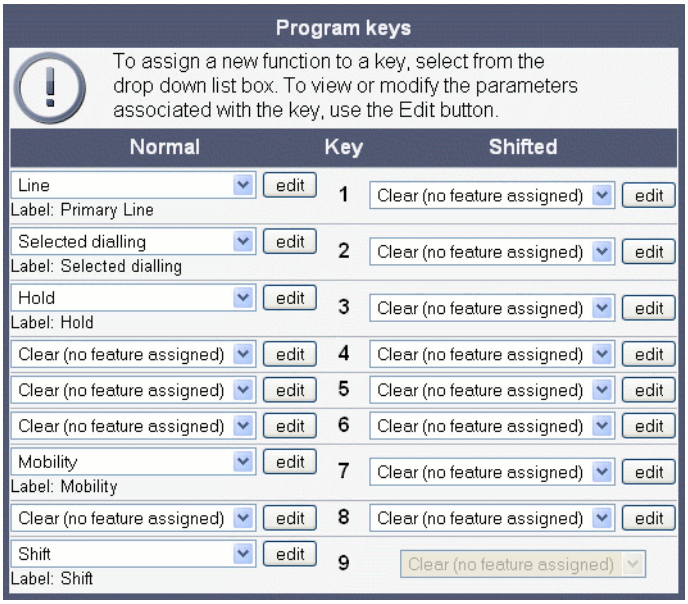
- **Key label <n>**: Set the label of the line key with the key number <n>.
  Default: "Line"
- **Primary line**: Determines whether the line is the primary line.
  Value range: "Yes", "No"
  Default: "No"
- **Ring on/off**: Determines whether the line rings on an incoming call.
  Value range: "On", "Off"
  Default: "On"
- **Ring delay**: Time interval in seconds after which the line starts ringing on an incoming call.
  Default: 0
- **Selection order**: Priority assigned to the line for the selection of an outgoing line.
  Default: 0
- **Address**: Address/phone number which has a corresponding entry on the SIP server/ registrar.
- **Realm**: Domain wherein user id and password are valid.
- **User Identifier**: User name for authentication with the SIP server.
- **Password**: Password for authentication with the SIP server.
- **Shared type**: Determines whether the line is a shared line (shared by multiple endpoints) or a private line (only available for this endpoint).
  Value range: "shared", "private", "unknown".
  Default: "shared"
- **Hot/Warm line type**: Determines whether the line is a hot line or a warm line.
  Value range: „hot line“, „warm line“
- **Hot/Warm line destination**: Number to be dialed when the phone is in hotline or warmline mode.
- **Allow in Overview**: Determines whether the line appears in the phone’s line overview.
  Value range: "Yes", "No"
  Default: "Yes"
- **Hot warm action** (V2): Determines if the line is a regular line, a hot line, or a warm line.
  Value range: "No action", "hot line", "warm line"
- **Hot warm destination** (V2): The destination to be dialed from the hot/warm line when the user goes off-hook.

> A new line key can only be added by use of the WBM or, preferably, the DLS. Once a line key exists, it can also be configured by the local menu.

Administration via WBM

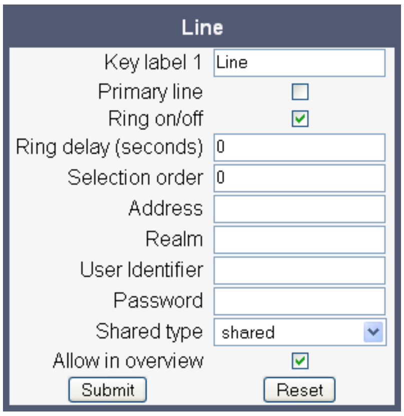
1. Invoke the "Phone keys" dialog and select "line" in the pulldown menu of the key you want to configure. Next, press "Edit...".

Features > Program keys

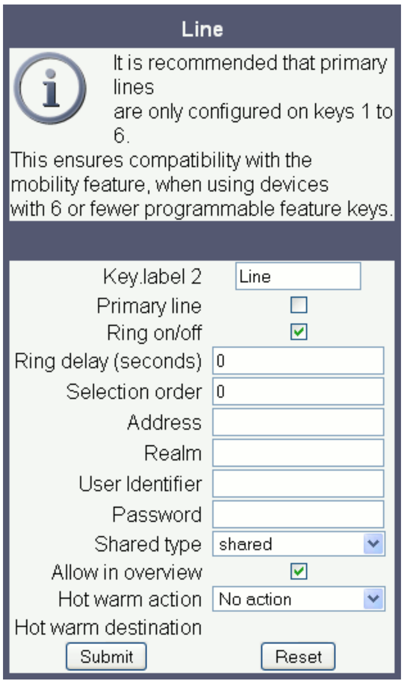


2. In the "Line" dialog, set the specific parameters for the line key.

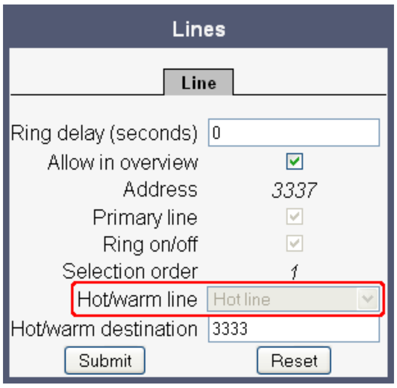
Firmware version V1R5:



Firmware version V2:



3. (Only relevant if warm line / hot line is to be configured:) The destination for warm line or hot line is set in User menu > Configuration > Keyset > Lines:



In the local menu, the menu path is the same.

Administration via Local Phone

The configuration of a line via Local phone is only possible when the line key has been created via Web interface or DLS before.

```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- Keyset Lines
                         |--- Details For Keyset Line <xx>
                              |--- Address
                              |--- Ring on/off
                              |--- Selection order
```

3.9.2 Configure Keyset Operation

The following parameters provide general settings which are common for all keyset lines.

The **Rollover ring** setting will be used when, during an active call, an incoming call arrives on a different line. If "no ring" is selected, the incoming call will not initiate a ring. If "alert ring" is selected, a 3 seconds burst of the configured ring tone is activated on an incoming call; "alert beep" selects a beep instead of a ring tone. "Standard ring tone" selects the default ringer.

LED on registration determines whether the line LEDs will be lit for a few seconds if they have been registered successfully with the SIP server on phone startup.

The **Originating line preference** parameter determines which line will be used when the user goes off-hook or starts on-hook dialing.

> When a terminating call exists, the terminating line preference takes priority over originating line preference.

The following preferences can be configured:

- "idle line": An idle line is selected. The selection is based on the **Hunt ranking** parameter assigned to each line (see Section 3.9.1, “Line key configuration”).
- "primary": The designated Primary Line/Main DN is always selected for originating calls.
- "last": The line selected for originating calls is the line that has been used for the last call (originating or terminating).
- "none": The user manually selects a line by pressing its line key before going off-hook, or by pressing the speaker key, to originate a call. Manual line selection overrides automatic line preferences.

The **Terminating line preference** parameter decides which terminating line, i. e. line with an incoming call, is selected when the user goes off-hook.

The following preferences can be configured:

- "ringing line": The line in the alerting or audible ringing state is automatically selected when the user goes off-hook. In the case of multiple lines alerting or ringing, the lines are selected on the one that has been alerting the longest.
- "ringing PLP": The line in the alerting or audible ringing state is automatically selected when the user goes off-hook. However, if the prime line is alerting, it is given priority.
- "incoming": The earliest line to start audible ringing is selected, or else the earliest alerting (ringing suppression ignored) line is selected.

- "incoming PLP": The earliest line to start audible ringing is selected, or else the earliest alerting (ringing suppression ignored) line is selected. However, if the prime line is alerting, it is given priority.
- "none": To answer a call, the user manually selects a line by pressing its line key before going off-hook, or by pressing the speaker key. Manual line selection overrides automatic line preferences.

Line action mode determines the consequence for an established connection when the line key is pressed. If "hold" is selected, the call currently active is set to hold as soon as the line key is activated. The user has two options: 1) to reconnect to the remote phone by pressing the line key that corresponds to that call, or 2) to initiate another call from the newly selected line. If "release" is selected, the previously established call is ended.

If **Show Focus** is checked, the LED of a line key flutters when the line is in use. If it is not checked, the line key is lit steady when it is in use.

The **Reservation timer** sets the period after which the reservation of a line is canceled. A line is automatically reserved for the keyset whenever the user has selected a line for an outgoing call and hears a dial tone. The reservation of a line is accomplished by the OpenScape Voiceserver, which notifies all the endpoints sharing this line. If set to 0, the reservation timer is deactivated.

Forward indication activates or deactivates the indication of station forwarding, i. e. the forwarding function of OpenScape Voice. If **Forward indication** is activated and station forwarding is active for the corresponding line, the LED of the line key blinks.

Preselect mode determines the phone’s behaviour when a call is active, and another call is ringing. If the parameter is set to "Single button", the user can accept the call a single press on the line key. If it is set to "Preselection", the user must first press the line key to select it and then press it a second time to accept the call. In both cases, the options for a ringing call are presented to the user: "Accept", "Reject", "Deflect".

Preselect timer is relevant if **Preselect mode** is set to "Preselection". The parameter sets the timeout in seconds for the second key press that is required to accept the call. After the timeout has expired, the call is no longer available.

With firmware V2, call bridging is available. When **Bridging enabled** is activated, the user may join into an existing call on a shared line by pressing the corresponding line key. On key press, the OpenScape Voice builds a server based conference from the existing call parties and the user. If the call has already been in a server based conference, the user is added to this conference.

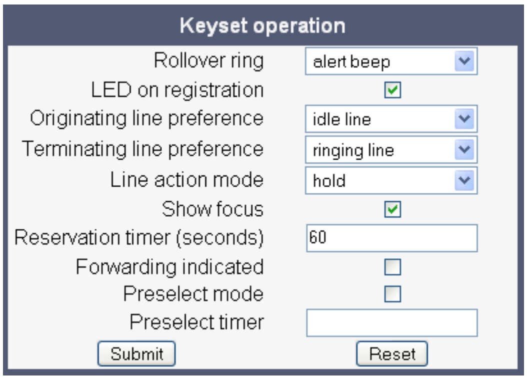
> When bridging shall be used, it is highly recommended to configure the phone for a system based conferece (see Section 3.6.9, “System Based Conference”). This enables adding more users to a system based conference that has been initiated by bridging.

Data required

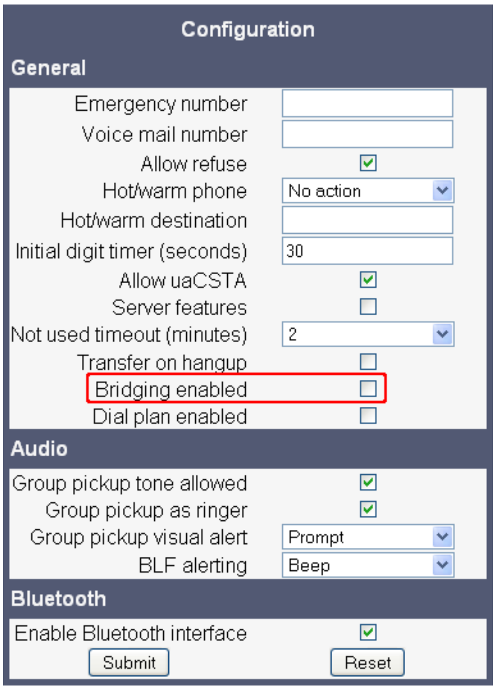
- **Rollover ring**: Determines if a ring tone will signal an incoming call while a call is active.
  Value range: "No ring", "Alert beep", "Alert ring"
  Default: "Alert beep"
- **LED on registration**: Determines if line LEDs will signal SIP registration.
  Value range: "Yes", "No"
  Default: "Yes"
- **Originating line preference**: Selects the line to be used for outgoing calls.
  Value range: "Idle line", "Primary", "Last", "None"
  Default: "Idle line"
- **Terminating line preference**: Determines which line with an incoming call shall be selected for answering.
  Value range: "Ringing line", "Incoming", "Incoming PLP", "Ringing PLP", "None"
  Default: "Idle line"
- **Line action mode**: Determines the consequence for an established connection when the line key is pressed.
  Value range: "Hold", "Release"
  Default: "Hold"
- **Show focus**: Determines whether the line key LED blinks or is steady when it is in use.
  Value range: "Yes", "No"
  Default: "Yes"
- **Reservation timer**: Sets the period in seconds after which a line reservation is cancelled. If set to 0, the reservation timer is deactivated.
  Default: 60
- **Forward indication**: Activates or deactivates the indication of station forwarding.
  Value range: "Yes", "No"
  Default: "No"
- **Preselect mode**: Determines whether an incoming call is accepted by a single press on the corresponding line key or a double press is needed.
  Value range: "Single button", "Preselection"
  Default: "Single button"
- **Preselect timer**: Sets the timeout in seconds for accepting an incoming call.
- **Bridging enabled** (V2): When set to "Yes", the user is allowed to join a call on a shared line. For this purpose, a server based conference is established.

Administration via WBM

System > Features > Keyset Operation



System > Features > Configuration



Administration via Local Phone

```
|— Administration
   |— System
      |— Features
         |— Keyset operation
            |— Rollover ring
            |— LED on registration
            |— Originating line preference
            |— Terminating line preference
            |— Line action mode
            |— Show focus
            |— Reservation timer
            |— Forward indicated
            |— Preselect mode
            |— Preselect timer
```

Administration via Local Phone

```
|— Administration
   |— System
      |— Features
         Configuration
            |— General
               |— Bridging enabled
```

3.9.3 Line Preview (V2)

This key enables the preview mode, which allows the user to preview a line before using it.

When preview mode is active, the line keys behave similar to when the keyset configuration is set to preselection for line keys (see Section 3.9.2, “Configure Keyset Operation”). On pressing the line key (not DSS key!), the call activity on the corresponding line is shown. Unlike with a preselected line, there will be no change to the phone’s current line connections. The LED indicates whether line preview is active or not.

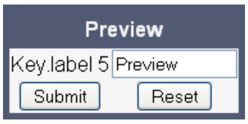
The information shown to the user depends on the ring/alert configuration for the line in question. If the line is configured to alert only, the preview will only show the state of the call, not the identity of the call party. If the line is configured to ring, the identity of the call party will be revealed.

The preview mode can be configured as temporary or as permanent. If **System** > **Features** > **Keyset operation** > **Preview mode** is disabled, preview mode will end when the user uses the previewed line, or a new call is started in any other way, or if the focus is changed away from call view. If the parameter is enabled, preview mode remains active until the user cancels it by pressing the key again.

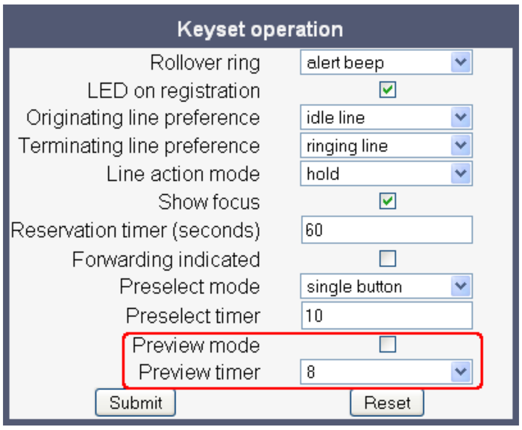
The **Preview timer** parameter determines the timespan during which the line preview will remain on the screen.

Administration via WBM

System > Features > Program keys > Preview



System > Features > Keyset operation



Administration via Local Phone

|--- Administration
    |--- System
        |--- Features
            |--- Keyset operation
                |--- **Preview mode**
                |--- **Preview timer**

3.9.4 Immediate Ring

Enables or disables the preset delay for all line keys. This feature only applies to keyset lines.

The label displayed to the left of the key is defined in **Key label <key number>**.

Administration via WBM

System > Features > Program keys > Immediate ring

3.9.5 Direct Station Select (DSS)

> This feature is available only on OpenStage 15/40/60/80, and requires HiPath V 3.0.

A DSS key is a special variant of a line key. It enables a direct connection to a target phone, allowing the user to pick up or forward a call alerting the DSS target and make/complete a call to the DSS target.

3.9.5.1 General DSS Settings

These parameters define the behaviour of all DSS keys.

> Generally, it is advisable to restrict the user's possibilities to modify line keys, including DSS keys. This can be achieved solely by the DLS. For further instructions, see the DLS Administration Guide.

If the user picks up an incoming call for the DSS target by pressing the associated DSS key, the call is forwarded to the user's primary line. Thereafter, the user's phone rings, and the user can accept the call.

> To enable the immediate answering of a call via the DSS key, **Allow auto-answer** in the user menu must be activated. The complete path on the WBM is:
> User Pages > Configuration > Incoming calls > CTI calls > Allow auto-answer.

The value of **Call pickup detect timer (seconds)** determines the time interval in which the deflected call is expected at the primary line. When the call arrives whithin this interval, it is given special priority and handling. If a second call arrives on the primary line during this interval, it will be rejected. If a second call arrives outside the interval, it will be treated just like any other incoming call. The default is 3.

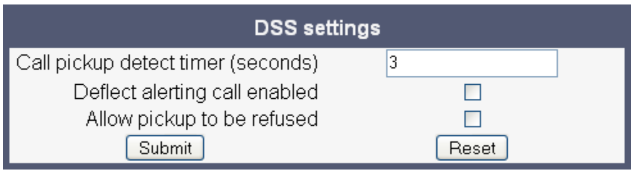
If **Deflecting call enabled** is checked, the user can forward an alerting call to the DSS target by pressing the DSS key. The default is "No".

If **Allow pickup to be refused** is checked, the user is enabled to reject a call alerting on the line associated with the DSS key. The default is "No".

With firmware version V2, the DSS key can be configured to indicate the call forwarding state of the number represented by the DSS key. This feature is activated when **Forwarding shown** is enabled.

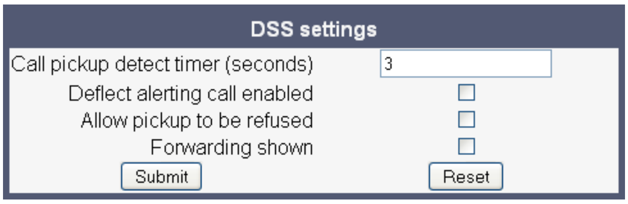
Administration via WBM (V1R5)

System > Features > DSS Settings



Administration via WBM (V2)

System > Features > DSS Settings



Administration via Local Phone (V1R5)

```
|— Administration
   |— System
      |— Features
         |— Feature Access
            |— Call establish
               |— Deflect to DSS
               |— Refuse DSS pickup
```

Deflect to DSS, **Refuse DSS pickup**

```
|— Administration
   |— System
      |— Features
         |— Configuration
            |— General
               |— DSS Pickup timer
```

DSS Pickup timer

Administration via Local Phone (V2)

```
|--- Administration
     |--- System
          |--- Features
               |--- DSS operation
                    |--- Deflect to DSS
                    |--- Refuse DSS pickup
                    |--- Forwarding shown
```

```
|--- Administration
     |--- System
          |--- Features
               |--- Configuration
                    |--- General
                         |--- DSS Pickup timer
```

3.9.5.2 Settings for a DSS key

The **Key label <n>** parameter provides the DSS key with a label that is displayed on the graphic display on a OpenStage 60/80 phone. The label is also user configurable.

Address contains the call number of the line associated with the DSS key.

The **Realm** parameter stores the SIP Realm of the line associated with the DSS key.

User Identifier gives the SIP user ID of the line associated with the DSS key.

Password provides the password corresponding to the SIP user ID.

The **Outgoing calls** parameter determines the behaviour of a call over the DSS line at the target phone. If set to "Direct", any forwarding and Do not Disturb settings on the target phone will be overridden, so that a call will always alert. If set to Line type is set to "Normal", this is not the case, and the call will be treated like a regular call.

Action on calls defines the handling of an active call when pressing the DSS key. If set to "Consult", the user has an option to start a consultation with the DSS target. If set to "Transfer", the user can only transfer the call to the DSS target. If "No action" is selected, pressing the DSS key will have no effect.

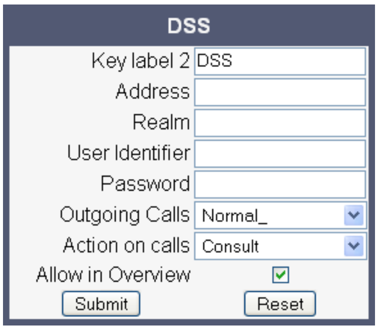
When **Allow in Overview** is set to "Yes", the line corresponding to the DSS key will be visible in the line overview on the phone's display.

Data required

- **Key label <key number>**: Label to be displayed on the display.
  Default: "DSS"
- **Address**: SIP Address of Record of the destination that is assigned to the DSS key.
- **Realm**: SIP Realm of the DSS destination.
- **User ID**: SIP user ID of the DSS destination.
- **Password**: Password corresponding to the SIP user ID.
- **Outgoing calls**: Determines whether forwarding and DND at the target phone will be overridden on a DSS call.
  Value range: "Normal", "Direct"
  Default: "Normal"
- **Action on calls**: Handling of an active call when pressing the DSS key. "Consult": the user can start a consultation with the DSS target; "Transfer": the user can transfer the call to the DSS target.
  Value range: "Consult", "Transfer", "No action"
  Default: "Consult"
- **Allow in Overview**: Determines whether the line appears in the phone’s line overview.
  Value range: "Yes", "No"
  Default: "Yes"

Administration via WBM

System > Features > Program keys > [edit]



3.10 Key Modules

A key module provides 12 additional free programable keys. Key modules are available for OpenStage 15/40/60/80 phones. A maximum of 2 key modules can be connected to one phone.

The following table shows which key modules can be connected to the particular phone types.

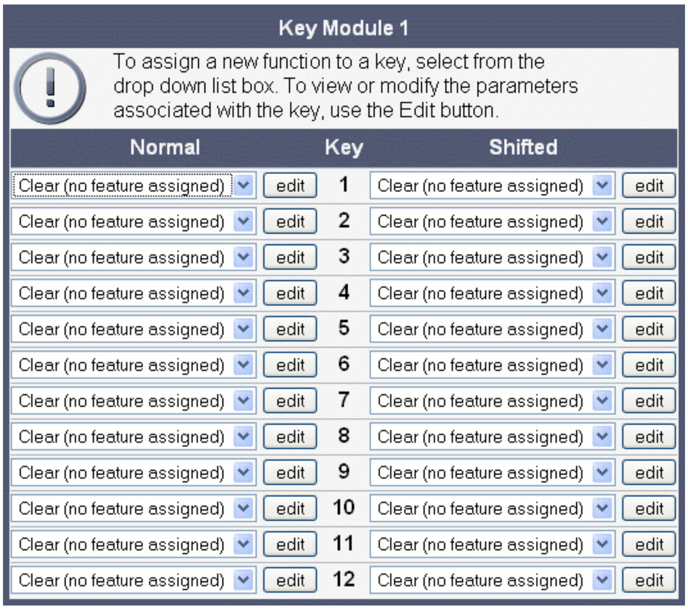
Phone Type	OpenStag Key Module 15	OpenStage Key Module
OpenStage 15	1	-
OpenStage 40	1	2
OpenStage 60/80	-	2

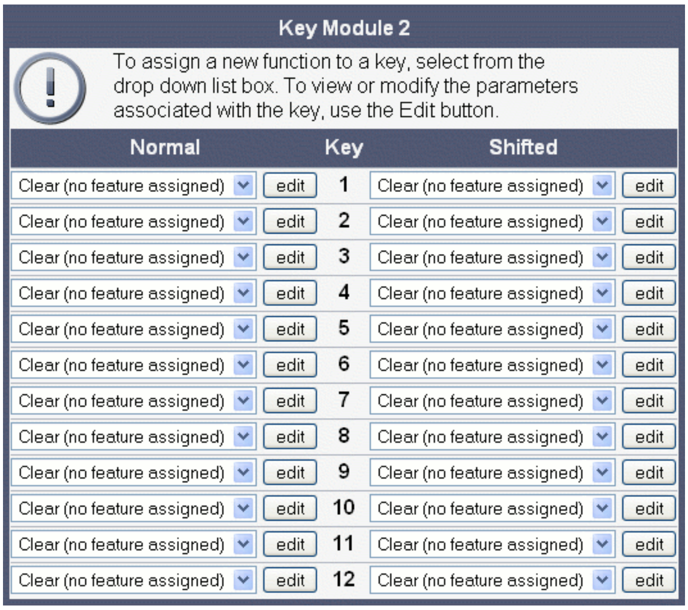
> Please note that OpenStage Key Modules (self-labeling) and OpenStage Key Module 15 (paper label) can not be combined. For key labelling, a special tool is available; please refer to:
> http://wiki.siemens-enterprise.com/index.php/Key_Labelling_Tool

The configuration of a key on the key module is exactly the same as the configuration of a phone key.

Administration via WBM

System > Features > Key module 1/2




Key Module 2
To assign a new function to a key, select from the drop down list box. To view or modify the parameters associated with the key, use the Edit button.
Normal
Key
Shifted
Clear (no feature assigned) edit 1 Clear (no feature assigned) edit
Clear (no feature assigned) edit 2 Clear (no feature assigned) edit
Clear (no feature assigned) edit 3 Clear (no feature assigned) edit
Clear (no feature assigned) edit 4 Clear (no feature assigned) edit
Clear (no feature assigned) edit 5 Clear (no feature assigned) edit
Clear (no feature assigned) edit 6 Clear (no feature assigned) edit
Clear (no feature assigned) edit 7 Clear (no feature assigned) edit
Clear (no feature assigned) edit 8 Clear (no feature assigned) edit
Clear (no feature assigned) edit 9 Clear (no feature assigned) edit
Clear (no feature assigned) edit 10 Clear (no feature assigned) edit
Clear (no feature assigned) edit 11 Clear (no feature assigned) edit
Clear (no feature assigned) edit 12 Clear (no feature assigned) edit

3.11 Dialing

3.11.1 Canonical Dialing Configuration

Call numbers taken from a directory application, LDAP for instance, are mostly expressed in canonical format. Moreover, call numbers entered into the local phone book are automatically converted and stored in canonical format, thereby adding "+", **Local country code**, **Local national code**, and **Local enterprise number** as prefixes. If, for instance, the user enters the extension "1234", the local country code is "49", the local national code is "89", and the local enterprise number is "722", the resulting number in canonical format is "+49897221234".

For generating an appropiate dial string, a conversion from canonical format to a different format may be required. The following parameters determine the local settings of the phone, like **Local country code** or **Local national code**, and define rules for converting from canonical format to the format required by the PBX.

> To enable the number conversion, all parameters not marked as optional must be provided, and the canonical dial lookup settings must be configured (see Section 3.11.2, “Canonical Dial Lookup”).

Data required

- **Local country code**: E.164 Country code, e.g. "49" for Germany, "44" for United Kingdom. Maximum length: 5
- **National prefix digit**: Prefix for national connections, e.g. "0" in Germany and United Kingdom.
  Maximum length: 5
- **Local national code**: Local area code or city code, e.g. "89" for Munich, "20" for London. Maximum length: 6
- **Minimal local number length**: Minimum number of digits in a local PSTN number, e.g. 3335333 = 7 digits.
- **Local enterprise number**: Number of the company/PBX wherein the phone is residing. Maximum length: 10 (Optional)
- **PSTN access code**: Access code used for dialing out from a PBX to a PSTN. Maximum length: 10 (Optional)
- **International access code**: International prefix used to dial to another country, e.g. "00" in Germany and United Kingdom. Maximum length: 5
- **Operator codes**: List of extension numbers for a connection to the operator. The numbers entered here are not converted to canonical format. Maximum length: 50 (Optional)
- **Emergency number**: List of emergency numbers to be used for the phone. If there are more than one numbers, they must be separated by commas. The numbers entered here are not converted to canonical format. Maximum length: 50 (Optional)

These emergency numbers can also be dialed when the phone is locked, in line with the emergency number configured in **Features** > **Configuration** (see Section 3.5.2, “Emergency and Voice Mail”).

- **Initial extension digits** / **Initial digits**: List of initial digits of all possible extensions in the local enterprise network. When a call number could not be matched as a public network number, the phone checks if it is part of the local enterprise network. This is done by comparing the first digit of the call number to the value(s) given here. If it matches, the call number is recognized as a local enterprise number and processed accordingly.
  If, for instance, the extensions 3000-5999 are configured in OpenScape Voice, each number will start with 3, 4, or 5. Therefore, the digits to be entered are 3, 4, 5.
- **Internal numbers**

> To enable the phone to discern internal numbers from external numbers, it is crucial that a canonical lookup table is provided (Section 3.11.2, “Canonical Dial Lookup”).

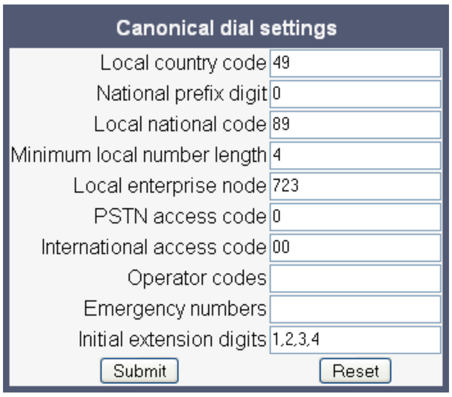
  - "Local enterprise form": Default value. Any extension number is dialled in its simplest form. For an extension on the local enterprise node, the node ID is omitted. If the extension is on a different enterprise node, then the appropriate node ID is prefixed to the extension number. Numbers that do not correspond to an enterprise node extension are treated as external numbers.
  - "Always add node": Numbers that correspond to an enterprise node extension are always prefixed with the node ID, even those on the local node. Numbers that do not correspond to an enterprise node extension are treated as external numbers.
  - "Use external numbers": All numbers are dialled using the external number form.
- **External numbers**
  - "Local public form": Default value. All external numbers are dialled in their simplest form. Thus a number in the local public network region does not have the region code prefix. Numbers in the same country but not in the local region are dialled as national numbers. Numbers for a different country are dialled using the international format.
  - "National public form": All numbers within the current country are dialled as national numbers, thus even local numbers will have a region code prefix (as dialling from a mobile). Numbers for a different country are dialled using the international format.
  - "International form": All numbers are dialled using their full international number format.
- **External access code**
  - "Not required": The access code to allow a public network number to be dialled is not required.

    - "For external numbers": Default value. All public network numbers will be prefixed with the access code that allows a number a call to be routed outside the enterprise network. However, international numbers that use the + prefix will not be given access code.

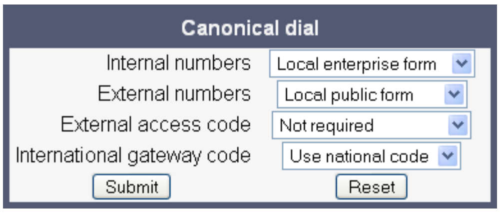
- **International gateway code**:
    - "Use national code": Default value. All international formatted numbers will be dialled explicitly by using the access code for the international gateway to replace the "+" prefix.
    - "Leave as +": All international formatted numbers will be prefixed with "+".

Administration via WBM

Local functions > Locality > Canonical dial settings



Local functions > Locality > Canonical dial



Administration via Local Phone

- Administration
  - Local Functions
    - Locality
      - Canonical dial settings
        - **Local country code**
        - **National prefix digit**
        - **Local national code**
        - **Minimum local number length**
        - **Local enterprise node**
        - **PSTN access code**
        - **International access code**
        - **Operator code**
        - **Emergency number**

- Administration
  - Local Functions
    - Locality
      - Canonical dial
        - **Internal numbers**
        - **External numbers**
        - **External access code**
        - **International gateway**

3.11.2 Canonical Dial Lookup

The parameters given here are important for establishing outgoing calls and for recognizing incoming calls.

In the local phonebook, and, mostly, in LDAP directories, numbers are stored in canonical format. In order to generate an appropriate dial string according to the settings in **Internal numbers** and **External numbers** (-> Section 3.11.1), internal numbers must be discerned from external numbers. The canonical lookup table provides patterns which allow for operation.

Furthermore, these patterns enable the phone to identify callers from different local or international telephone networks by looking up the caller's number in the phone book. As incoming numbers are not always in canonical format, their composition must be analyzed first. For this purpose, an incoming number is matched against one or more patterns consisting of country codes, national codes, and enterprise nodes. Then, the result of this operation is matched against the entries in the local phone book.

To make sure that canonical dial lookup works properly, at least the following parameters of the phone must be provided:

- **Local country code** (-> Section 3.11.1)
- **Local area code** (-> Section 3.11.1)
- **Local enterprise code** (-> Section 3.11.1)

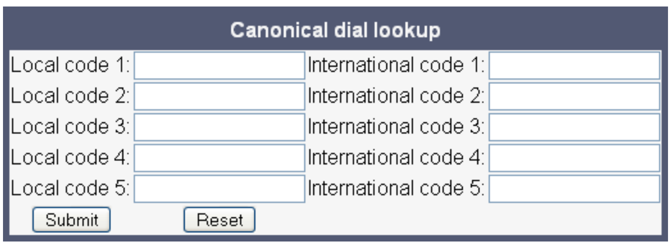
Up to 5 patterns can be defined. The **Local code 1 ... 5** parameters define up to 5 different local enterprise nodes, whilst **International code 1... 5** define up to 5 international codes, that is, fully qualified E.164 call numbers for use in a PSTN.

Data required

- **Local code 1 ... 5**: Local enterprise code for the node/PBX the phone is connected to. Example: "722" for Siemens Munich.
- **International code 1 ... 5**: Sequence of "+", local country code, local area code, and local enterprise node corresponding to to one or more phone book entries.
  Example: "+4989722" for Siemens Munich.

Administration via WBM

Locality > Canonical dial lookup



Administration via Local Phone

```
|--- Administration
     |--- Local Functions
          |--- Locality
               |--- Canonical Dial Lookup
                    |— Local code 1
                    |— International code 1
                    |— Local code 2
                    |— International code 2
                    |— Local code 3
                    |— International code 3
                    |— Local code 4
                    |— International code 4
                    |— Local code 5
                    |— International code 5
```

3.11.3 Dial Plan (V2)

With firmware version V2, OpenStage phones may optionally use a dial plan residing on the phone. By means of the dial plan, the phone can infer from the digits entered by the user that a complete call number has been entered, or that a particular prefix has been entered. Thus, the dialing process can start without the need to confirm after the last digit has been entered, without delay or with a configurable delay. The standard timer, which is found on the WBM under User menu > Configuration > Outgoing calls > Autodial delay (seconds), is overridden if a dial plan rule is matched.

A dial plan consists of rules defining patterns, timeouts and actions to be performed when a pattern is matched and/or a timeout has expired. The phone can store one dialplan, which can contain up to 48 different rules.

It is very important that the phone's dial plan does not interfere with the dial plan in the SIP server, PBX, or public network.

The dial plan can be created and uploaded to the phone using the DLS (please refer to the Deployment Service Administration Manual). The DLS can also export and import dial plans in .csv format. For details about the composition of a dial plan, please refer to Section 5.5, "Dial Plan (V2)".

The current dial plan, along with its status (enabled/disabled) and error status can be displayed on the WBM via Diagnostics > Fault trace configuration > Download dial plan file.

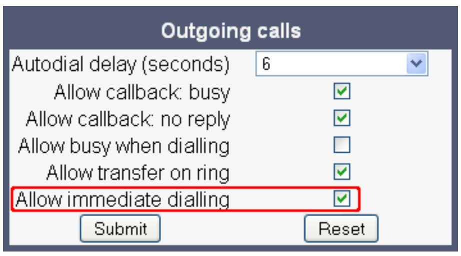
With software version V2R2, the **Dial plan ID** and the **Dial plan status** is displayed in the local menu.

To make use of the dial plan facility, the following requirements must be met:

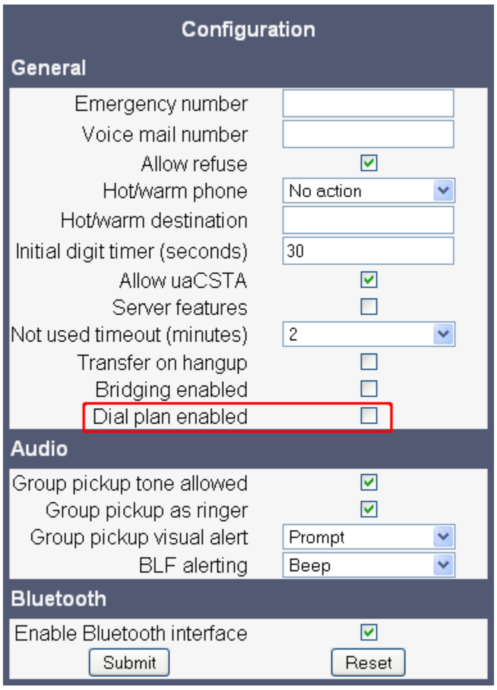
- A correct dial plan is loaded to the phone.
- In the user menu, **Allow immediate dialing** is enabled.
- **Dial plan enabled** is checked.

Administration via WBM

User menu > Configuration > Outgoing calls > Allow immediate dialing



System > Features > Configuration > Dial plan enabled



Administration via Local Phone

```
|--- User
       |--- Configuration
              |--- Outgoing calls
                     |--- Immediate dialing

|--- Administration
|      |--- System
              |--- Features
                     |--- Configuration
                            |--- General
                                   |--- Dial plan

|--- Administration
       |--- General Information
       |--- |--- Dial plan ID
              |--- Dial plan status
```

3.12 Distinctive Ringing (V2)

The SIP server may provide information indicating a specific type of call within an incoming call. With firmware V2, the phone can use this information to choose a ring tone according to the call type.

The relevant information is carried as a string in the SIP Alert-Info header. This string is configured in the OpenScape Voice system; please refer to the relevant OpenScape Voice documentation. When the string sent via Alert-Info matches the string specified in the **Name** parameter, the corresponding ringer is triggered. For instance, the OpenScape Voice system may send the string `Bellcore-dr1` to indicate that a call is from within the same business group, and the **Name** parameter is set to "Bellcore-dr1". To select a specific ring tone for calls from the same business group, the other parameters corresponding to that **Name** must be set accordingly.

The **Ringer sound** parameter determines whether a pattern, i. e. melody, or a specific sound file shall be used as ringer.

Pattern Melody selects the melody pattern that will be used if **Ringer sound** is set to "Pattern".

Pattern sequence determines the length for the melody pattern, and the interval between the repetitions of the pattern. There are 3 variants:

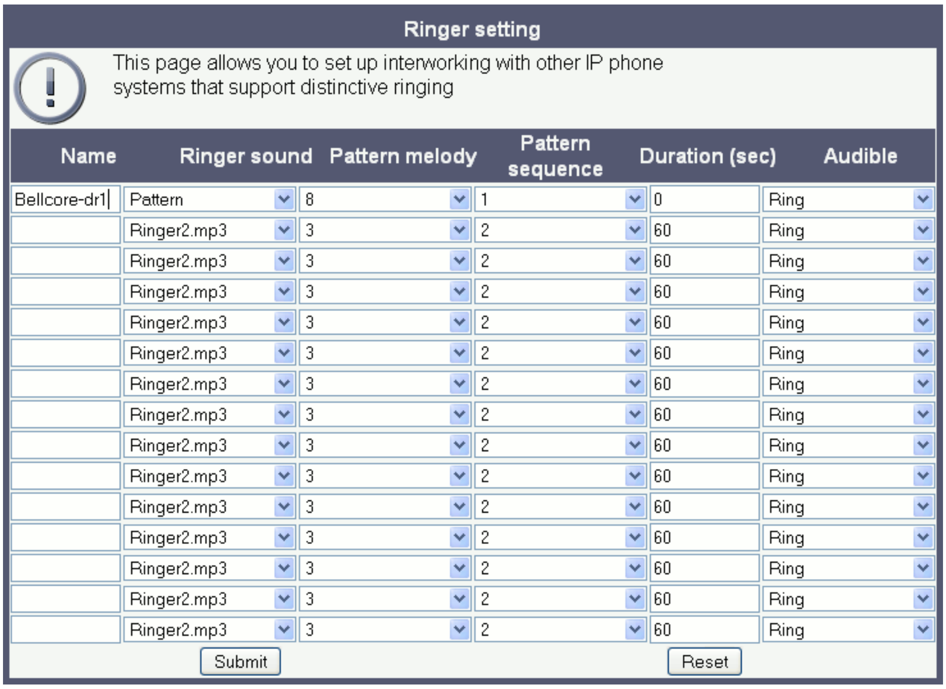
- "1": 1 sec ON, 4 sec OFF
- "2": 1 sec ON, 2 sec OFF
- "3": 0.7 sec ON, 0.7 sec OFF, 0.7 sec ON, 3 sec OFF

The **Duration** parameter determines how long the phone will ring on an incoming call. The range is 0-300 sec.

With the **Audible** parameter, the ringer can be muted. In this case, an incoming call will be indicated only visually.

Administration via WBM

Ringer setting



Ringer setting

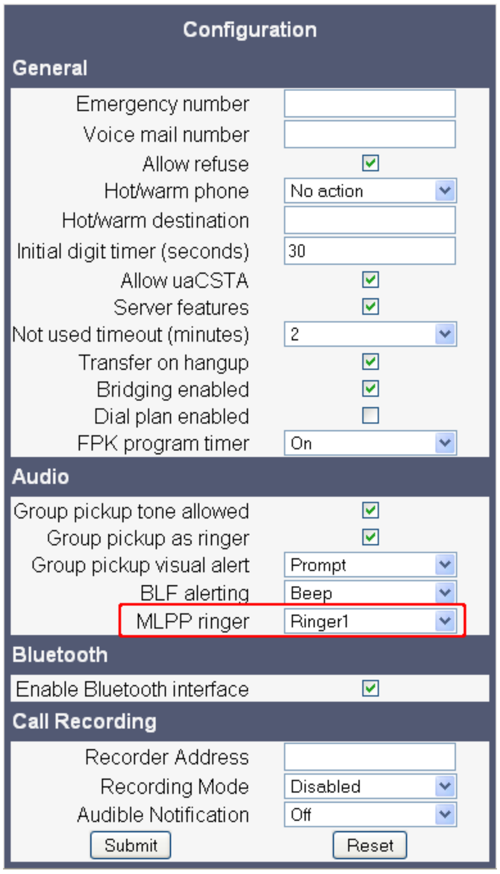
This page allows you to set up interworking with other IP phone systems that support distinctive ringing

Name	Ringer sound	Pattern melody	Pattern sequence	Duration (sec)	Audible
Bellcore-dr1	Pattern	8	1	0	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring

Submit Reset

Administration via Local Phone

- Administration
  - Ringer setting
    - **<1 .... 15>**
      - **Name**
      - **Ringer sound**
      - **Pattern melody**
      - **Pattern sequence**
      - **Duration**
      - **Audible**


Configuration
General
Emergency number
Voice mail number
Allow refuse
Hot/warm phone
No action
Hot/warm destination
Initial digit timer (seconds)
30
Allow uaCSTA
Server features
Not used timeout (minutes)
2
Transfer on hangup
Bridging enabled
Dial plan enabled
FPK program timer
On
Audio
Group pickup tone allowed
Group pickup as ringer
Group pickup visual alert
Prompt
BLF alerting
Beep
MLPP ringer
Ringer1
Bluetooth
Enable Bluetooth interface
Call Recording
Recorder Address
Recording Mode
Disabled
Audible Notification
Off
Submit
Reset

3.13 Mobility

The Mobility feature requires the HiPath Deployment Severice (DLS). If the phone is mobility enabled by the DLS, a mobile user can log on to the phone and thereby have his own user settings transferred to the phone. These user data are stored in the DLS database and include, for instance, SIP registration settings, dialing properties, key layouts, as well as the user's phonebook.

If the mobile user changes some settings, the changed data is sent to the DLS server. This ensures that his user profile is updated if necessary.

If **Unauthorized logoff trap** is set to "Yes", a message is sent to the SNMP server if an unauthorized attempt is made to log off the mobile user.

Logoff trap delay defines the time span in seconds between the unauthorized logoff attempt and the trap message to the SNMP server.

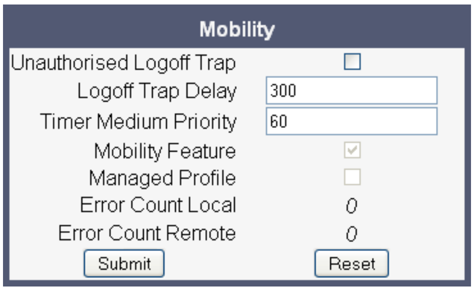
Timer med priority determines the time span in seconds between a change of user data in the phone and the transfer of the changes to the DLS server.

The **Mobility feature** parameter indicates whether the mobility feature is enabled by the DNS or not.

Data required

- **Unauthorized logoff trap**: An SNMP trap is sent on an unauthorized logoff attempt.
  Value range: "Yes", "No"
  Default: "No"
- **Logoff trap delay**: Time span in seconds between the unauthorized logoff attempt and the SNMP trap.
  Default: 300
- **Timer med priority**: Time span in seconds between a data change in the phone and its transfer to the DLS server.
  Default: 60
- **Mobility feature**: Indicates whether the mobility feature is enabled.

Administration via WBM



Administration via Local Phone

|— Administration
    |— Mobility
        |— **Unauthorized logoff trap**
        |— **Logoff trap delay**
        |— **Timer med priority**
        |— **Mobility feature**

3.14 Transferring Phone Software, Application and Media Files

New software images, hold music, picture clips for phonebook entries, LDAP templates, company logos, screensaver images, and ring tones can be uploaded to the phone via DLS (Deployment Service) or WBM (Web Based Management).

> For all user data, which includes files as well as phonebook content, the following amounts of storage place are available:
> - OpenStage 15/20/40: 4 MB
> - OpenStage 60/80: 8 MB

3.14.1 FTP/HTTPS Server

There are no specific requirements regarding the FTP server for transferring files to the OpenStage phone. Any FTP server providing standard functionality will do.

3.14.2 Common FTP/HTTPS Settings

For each one of the various file types, e.g. phone software, hold music, and picture clips, specific FTP/HTTPS access data can be defined. If some or all file types have the parameters **Download method**, **FTP Server**, **FTP Server port**, **FTP account**, **FTP username**, **FTP path**, and **HTTPS base URL** in common, they can be specified here. These settings will be used for a specific file type if its **Use defaults** parameter is set to "Yes".

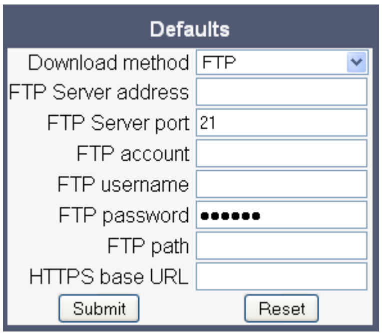
> If **Use defaults** is activated for a specific file type, any specific settings for this file type are overridden by the defaults.

Data required

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP server in use.
- **FTP Server port**: Port number of the FTP server in use. For HTTPS, port 443 is assumed, unless a different port is specified in the HTTPS base URL.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use. If no port number is specified here, port 443 is used. Only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Defaults



Administration via Local Phone

```
|--- Administration
    |--- File Transfer
        |--- Defaults
            |--- Download method
            |--- FTP Server
            |--- FTP Port
            |--- FTP Account
            |--- FTP Username
            |--- FTP Password
            |--- FTP path
            |--- HTTPS base URL
```

3.14.3 Phone Software

The firmware for the phone can be updated by downloading a new software file to the phone.

> ⚠ Do not disconnect the phone from the LAN or power unit during software update. An active update process is indicated by blinking LEDs and/or in the display.

3.14.3.1 FTP/HTTPS Access Data

If the default FTP/HTTPS Access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

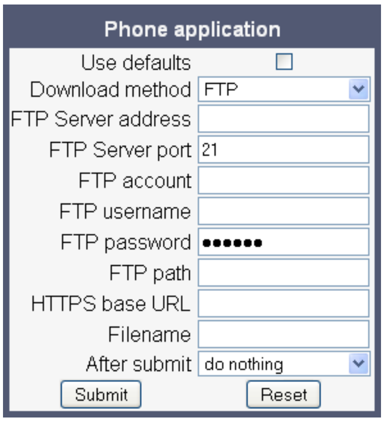
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used. Value range: "Yes", "No".
  Default: "No".
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Phone application



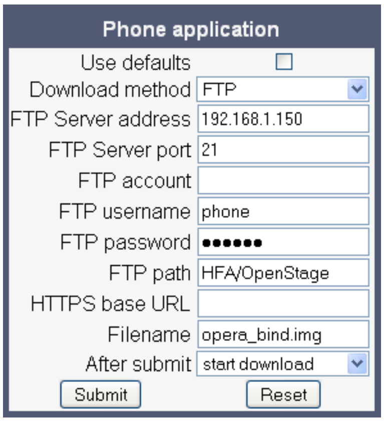
Administration via Local Phone

```
|--- Administration
     |--- File Transfer
          |--- Phone app
               |--- Use default
               |--- Download method
               |--- FTP Server
               |--- FTP Port
               |--- FTP Account
               |--- FTP Username
               |--- FTP Password
               |--- FTP path
               |--- HTTPS base URL
               |--- Filename
```

3.14.3.2 Download/Update Phone Software

If applicable, phone software should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu. When the download has been successful, the phone will restart and boot up using the new software.

Start Download via WBM



In the **File transfer** > Phone application dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Phone app**.

|— Administration
    |— File Transfer
        |— **Phone app**

2. Press the ➔ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.4 Music on Hold

If enabled by the user, the Music on Hold (MoH) sound file is played when a call is put on hold.

> The file size for a Music on Hold file is limited to 1MB. If the file is too large or the contents of the file are not valid, the file will not be stored in the phone.

The following formats for Music on Hold are supported:

- Proprietary Music on Hold format for optiPoint 410/420 phones
- WAV format. The recommended specifications are:
  - Audio format: PCM
  - Bitrate: 16 kB/sec
  - Sampling rate: 8 kHz
  - Quantization level: 16 bit
- MIDI format
- MP3 format (OpenStage 60/80 only). A bitrate of 48 kB/sec is recommended.

3.14.4.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use Default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used. Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

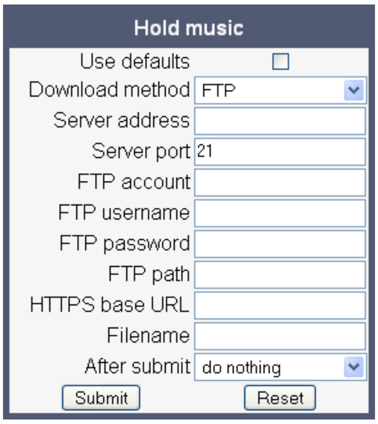
Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.

- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Hold music



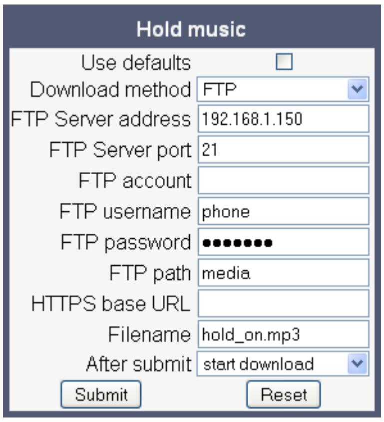
Administration via Local Phone

```
|--- Administration
     |--- File Transfer
          |--- Hold Music
               |--- Use default
               |--- Download method
               |--- FTP Server
               |--- FTP Port
               |--- FTP Account
               |--- FTP Username
               |--- FTP Password
               |--- FTP path
               |--- HTTPS base URL
               |--- Filename
```

3.14.4.2 Download Music on Hold

If applicable, Music on Hold should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

Start Download via WBM



In the **File transfer** > Hold music dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Hold Music**.

|--- Administration
    |--- File Transfer
        |--- **Hold Music**

2. Press the ➔ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.5 Picture Clips

> Picture clips are available only on OpenStage 60/80 phones.

> The file size for a picture clip is limited to 300 KB. If the file is too large or the contents of the file are not valid, the file will not be stored in the phone.

Picture Clips are small images used for displaying a picture of a person that is calling on a line. The supported file formats for picture clips are JPEG and PNG (recommended).

3.14.5.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

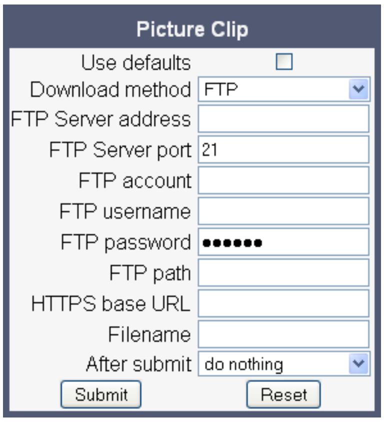
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
  Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Picture clip



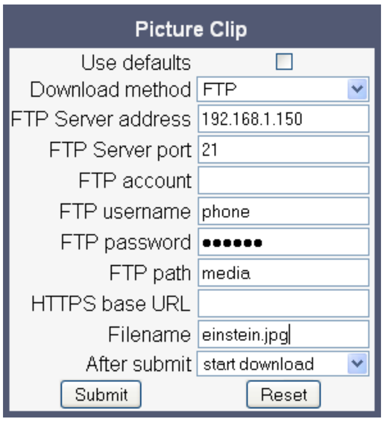
Administration via Local Phone

|— Administration
|— File Transfer
|— Picture Clip
|— **Use default**
|— **Download method**
|— **FTP Server**
|— **FTP Port**
|— **FTP Account**
|— **FTP Username**
|— **FTP Password**
|— **FTP path**
|— **HTTPS base URL**
|— **Filename**

3.14.5.2 Download Picture Clip

The download can be triggered from the web interface or from the local phone menu.

Start Download via WBM



In the **File transfer** > Picture clip dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Picture clip**.

|--- Administration
    |--- File Transfer
        |--- **Picture clip**

2. Press the ➔ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.6 LDAP Template

> LDAP is available only on OpenStage 60/80 phones and on OpenStage 40 phones with firmware version V2R1 onwards.

The LDAP template is an ASCII text file that uses an allocation list to assign directory server attributes to input and output fields on an LDAP client. The LDAP template must be modified correctly for successful communication between the directory server and the LDAP client.

> The OpenStage phone supports LDAPv3.

3.14.6.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, "Common FTP/HTTPS Settings") are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

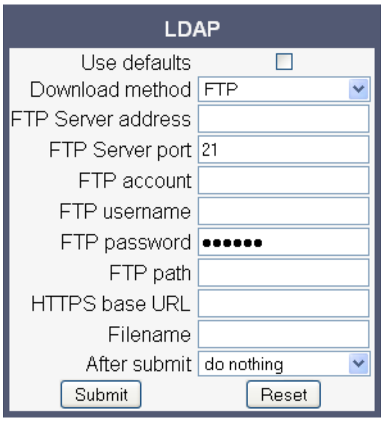
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
  Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTPServer address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > LDAP



Administration via Local Phone

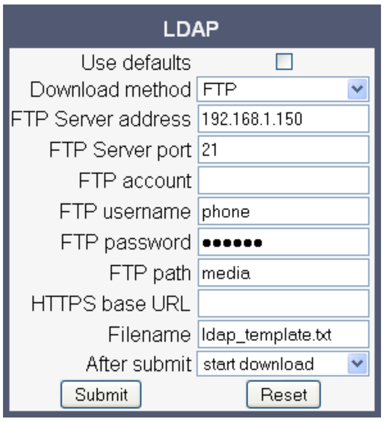
|--- Administration
    |--- File Transfer
        |--- LDAP
            |--- **Use default**
            |--- **Download method**
            |--- **FTP Server**
            |--- **FTP Port**
            |--- **FTP Account**
            |--- **FTP Username**
            |--- **FTP Password**
            |--- **FTP path**
            |--- **HTTPS base URL**
            |--- **Filename**

3.14.6.2 Download LDAP Template

If applicable, LDAP templates should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

> The OpenStage phone supports LDAPv3.

Start Download via WBM



In the **File transfer** > LDAP dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **LDAP**.

|— Administration
    |— File Transfer
        |— **LDAP**

2. Press the → key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.7 Logo

On OpenStage 40/60/80, a custom background image for the telephony interface can be supplied. In most cases, this will be the company logo.

On OpenStage 40, monochrome bitmap files (BMP) are supported. The ideal size is as follows:
- Width: 144 px
- Height: 32 px

On OpenStage 60/80, the supported file formats are JPEG and PNG. The ideal size values are is as follows:

OpenStage 60:
- Width: 240 px
- Height: 70 px

OpenStage 80:
- Width: 480 px
- Height: 142 px

If the size should deviate from these values, the image will appear skewed.

For guidance on creating a logo file for OpenStage 40/60/80, see Section 5.2, “How to Create Logo Files for OpenStage Phones”.

3.14.7.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

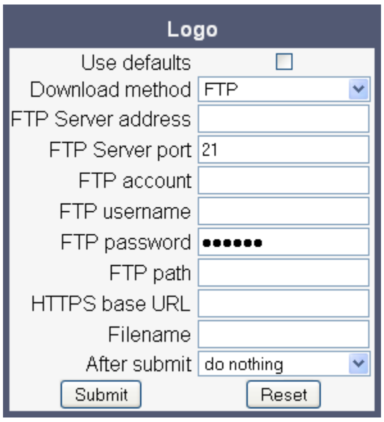
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
- Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Logo



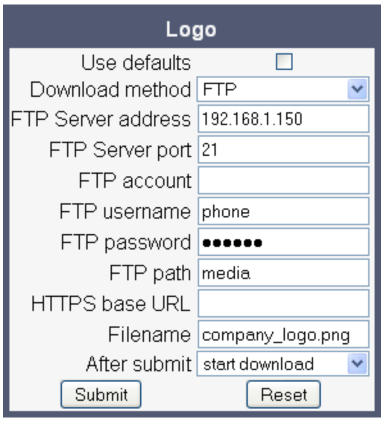
Administration via Local Phone

- Administration
  - File Transfer
    - Logo
      - **Use default**
      - **Download method**
      - **FTP Server**
      - **FTP Port**
      - **FTP Account**
      - **FTP Username**
      - **FTP Password**
      - **FTP path**
      - **HTTPS base URL**
      - **Filename**

3.14.7.2 Download Logo

If applicable, logos should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

Start Download via WBM



In the **File transfer** > Logo dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Logo**.

|— Administration
    |— File Transfer
        |— **Logo**

2. Press the ➔ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.8 Screensaver

The screensaver is displayed when the phone is in idle mode. It performs a slide show consisting of images which can be uploaded using the web interface.

> Screensavers are available only on OpenStage 60/80 phones.

> The file size for a screensaver image is limited to 300 KB. If the file is too large or the contents of the file are not valid, the file will not be stored in the phone.

For screensaver images, the following specifications are valid:

- Data format: JPG or PNG. JPG is recommended.
- Screen format: 4:3. The images are resized to fit in the screen, so that images with a width/height ratio differing from 4:3 will appear with deviant proportions.
- Resolution: The phone’s screen resolution is the best choice for image resolution:
  - OpenStage 60: 320x240
  - OpenStage 80: 640x480

3.14.8.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
  Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

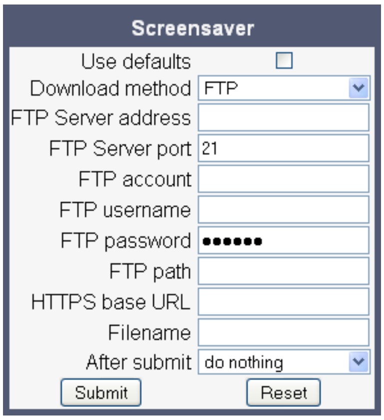
Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).

- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Screensaver



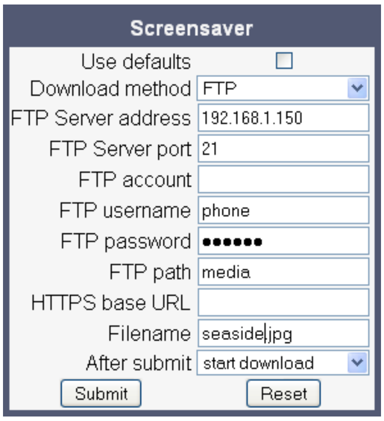
Administration via Local Phone

```
|--- Administration
     |--- File Transfer
          |--- Screensaver
               |--- Use default
               |--- Download method
               |--- FTP Server
               |--- FTP Port
               |--- FTP Account
               |--- FTP Username
               |--- FTP Password
               |--- FTP path
               |--- HTTPS base URL
               |--- Filename
```

3.14.8.2 Download Screensaver

If applicable, screensavers should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

Start Download via WBM



In the **File transfer** > Screensaver dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Screensaver**.

|— Administration
    |— File Transfer
        |— **Screensaver**

2. Press the → key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.9 Ringer File

Custom ring tones can be uploaded to the phone.

> The file size for a ringer file is limited to 1 MB. If the file is too large or the contents of the file are not valid, the file will not be stored in the phone.

The following file formats are supported:

- WAV format. The recommended specifications are:
  - Audio format: PCM
  - Bitrate: 16 kB/sec
  - Sampling rate: 8 kHz
  - Quantization level: 16 bit
- MIDI format.
- MP3 format (OpenStage 60/80 only). The OpenStage 60/80 phones are able to play MP3 files from 32 kbit/s up to 320 kbit/s. As the memory for user data is limited to 8 MB, a constant bitrate of 48 kbit/sec to 112 kbit/s and a length of max. 1 minute is recommended. Although the phone software can play stereo files, mono files are recommended, as the phone has only 1 loudspeaker.
  See the following table for estimated file size (mono files):

Length	**64 kbit/s**	**80 kbit/s**	**96 kbit/s**	**112 kbit/s**
0:15 min	0,12 MB	0,15 MB	0,18 MB	0,21 MB
0:30 min	0,23 MB	0,29 MB	0,35 MB	0,41 MB
0:45 min	0,35 MB	0,44 MB	0,53 MB	0,62 MB
1:00 min	0,47 MB	0,59 MB	0,70 MB	0,82 MB

3.14.9.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to "Yes", and only the **Filename** must be specified.

Data required (in every case)

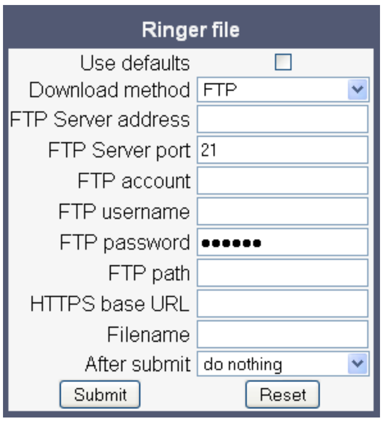
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
  Value range: "Yes", "No"
  Default: "No"
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: "FTP", "HTTPS"
  Default: "FTP"
- **FTP Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **FTP Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to "HTTPS".

Administration via WBM

File transfer > Ringer file



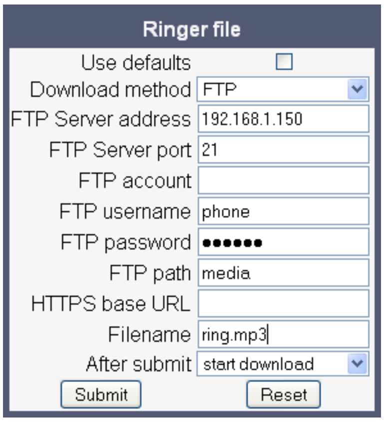
Administration via Local Phone

```
|--- Administration
     |--- File Transfer
          |--- Ringer
               |--- Use default
               |--- Download method
               |--- FTP Server
               |--- FTP Port
               |--- FTP Account
               |--- FTP Username
               |--- FTP Password
               |--- FTP path
               |--- HTTPS base URL
               |--- Filename
```

3.14.9.2 Download Ringer File

If applicable, ring tone files should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

Start Download via WBM



In the File transfer > Ringer dialog, set **After submit** to "start download" and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Ringer**.

|— Administration
    |— File Transfer
        |— **Ringer**

2. Press the ➜ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.14.10 Dongle Key

The HPT dongle key is a special file that contains a secret hash number which is required to connect the HPT tool to the phone. This testing tool is used exclusively by the service staff.

3.14.10.1 FTP/HTTPS Access Data

If the default FTP/HTTPS access settings (see Section 3.14.2, “Common FTP/HTTPS Settings”) are to be used, **Use default** must be set to „Yes“, and only the **Filename** must be specified.

Data required (in every case)

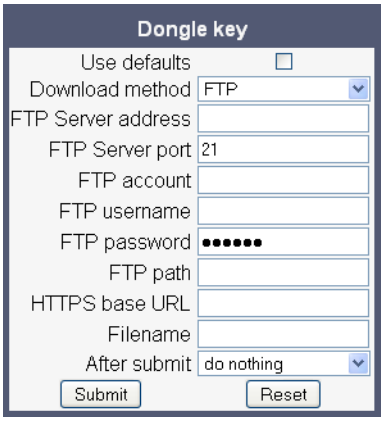
- **Use default**: Specifies whether the default FTP/HTTPS access settings shall be used.
  Value range: „Yes“, „No“
  Default: „No“
- **Filename**: Specifies the file name of the phone software.

Data required (if not derived from Defaults)

- **Download method**: Selects the protocol to be used.
  Value range: „FTP“, „HTTPS“
  Default: „FTP“
- **Server address**: IP address or hostname of the FTP/HTTPS server in use.
- **Server port**: Port number of the FTP/HTTPS server in use.
  Default: 21
- **FTP account**: Account at the server (if applicable).
- **FTP username**: User name for accessing the server.
- **FTP password**: Password corresponding to the user name.
- **FTP path**: Path of the directory containing the files.
- **HTTPS base URL**: IP address or hostname of the HTTPS server in use; only applicable if **Download method** is switched to „HTTPS“.

Administration via WBM

File transfer > Dongle key



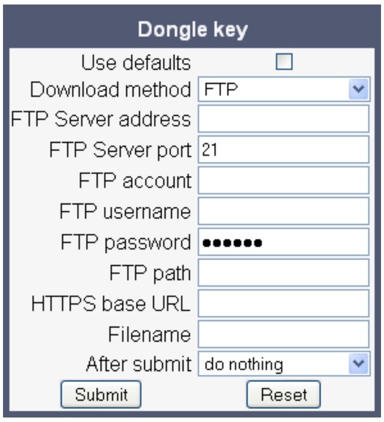
Administration via Local Phone

|--- Administration
    |--- File Transfer
        |--- Dongle key
            |--- **Use default**
            |--- **Download method**
            |--- **Server**
            |--- **Port**
            |--- **Account**
            |--- **Username**
            |--- **Password**
            |--- **FTP path**
            |--- **HTTPS base URL**
            |--- **Filename**

3.14.10.2 Download Dongle Key File

If applicable, dongle key files should be deployed using the DLS (Deployment Service). Alternatively, the download can be triggered from the web interface or from the Local phone menu.

Start Download via WBM



In the **File transfer** > Dongle key dialog, set **After submit** to „start download“ and press the **Submit** button.

Start Download via Local Phone

1. In the administration menu, set the focus to **Dongle key**.

|--- Administration
    |--- File Transfer
        |--- **Dongle key**

2. Press the ➔ key. A context menu opens. In the context menu, select **Download**. The download will start immediately.

3.15 Corporate Phonebook: Directory Settings

3.15.1 LDAP

> LDAP is available only on OpenStage 60/80 phones and on OpenStage 40 phones with firmware version V2R1 onwards.

The Lightweight Directory Access Protocol enables access to a directory server via an LDAP client. Various personal information is stored there, e.g. the name, organisation and contact data of persons working in an organisation. When the LDAP client has found a person's data, e. g. by looking up the surname, the user can call this person directly using the displayed number.

> The OpenStage phone supports LDAPv3.

For connecting the phone's LDAP client to a LDAP server, the required access data must be configured. The parameters **Server address** and **Server port** specify the IP address and hostname as well as the port used by the LDAP server. If the **Authentication** is not set to "Anonymous", the user must authenticate himself with the server by providing a **User name** and a corresponding **Password**. The user name is the string in the LDAP bind request, e. g. "C=GB,O=SIEMENS COMM,OU=COM,L=NTH,CN=BAYLIS MICHAEL". The internal structure will depend on the specific corporate directory.

For a quick guide on setting up LDAP on an OpenStage phone, please refer to Section 5.3, "How to Set Up the Corporate Phonebook (LDAP)".

With firmware V2, the OpenStage 60/80 GUI features a new search field for LDAP requests. The search string is submitted to the LDAP server as soon as the ⓞ key is pressed, or when the **Search trigger timeout** expires.

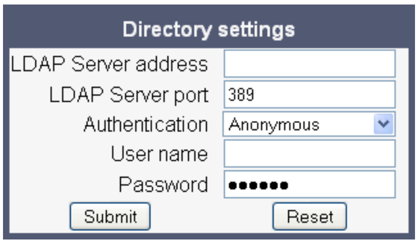
Data required

- **Server address**: IP address or hostname of the LDAP server.
- **Server port**: Port on which the LDAP server is listening for requests.
  Default: 389
- **Authentication**: Authentication method used for connecting to the LDAP server. value range: "Anonymous", "Simple"
  Default: "Anonymous"
- **User name**: User name used for authentication with the LDAP server in the LDAP bind request.
- **Password**: Password used for authentication with the LDAP server.

- **Search trigger timeout** (V2): Timespan between entering the last character and search string submission to the LDAP server.

Administration via WBM

Local Functions > Directory settings

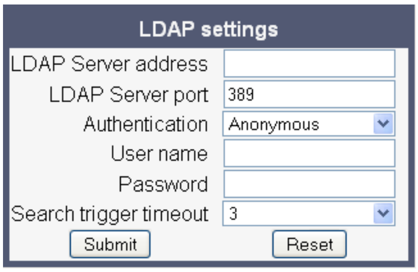


Administration via Local Phone

|--- Administration
    |--- Local Functions
        |--- Directory Settings
            |--- **LDAP server address**
            |--- **LDAP server port**
            |--- **LDAP authentication**
            |--- **LDAP user name**
            |--- **LDAP password**

Administration via WBM (V2)

Local Functions > LDAP settings



Administration via Local Phone (V2)

```
|--- Administration
     |--- Local Functions
          |--- LDAP Settings
               |--- server address
               |--- server port
               |--- Timeout (sec)
               |--- authentication
               |--- user name
               |--- password
```

3.16 Speech

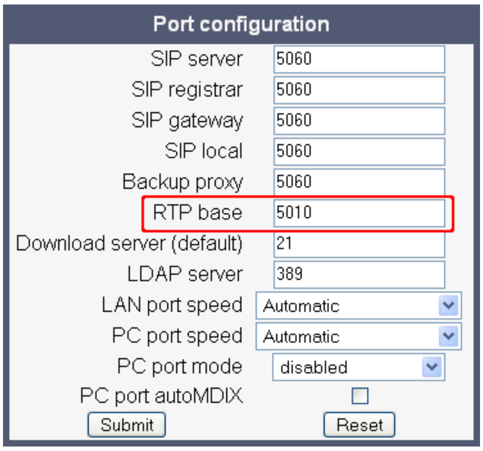
3.16.1 RTP Base Port

The port used for RTP is negotiated during the establishment of a SIP connection. The RTP base port number defines the starting point from which the phone will count up when negotiating. The default value is 5010.

The number of the port used for RTCP will be the RTP port number increased by 1.

Administration via WBM

Network > Port Configuration



Administration via Local Phone

|--- Administration
    |--- Network
        |--- Port Configuration
            |--- **RTP base**

3.16.2 Codec Preferences

If **Silence suppression** is activated, the transmission of data packets is suppressed on no conversation, that is, if the user doesn't speak.

The OpenStage phone provides the codecs **G.711**, **G.722**, and **G.729**. When a SIP connection is established between two endpoints, the phones negotiate the codec to be used. The result of the negotiation is based on the general availability and ranking assigned to each codec. The administrator can allow or disallow a codec as well as assign a ranking number to it.

The **Packet size**, i. e. length in milliseconds, of the RTP packets for speech data, can be set to 10ms, 20ms, 30ms or to automatic detection.

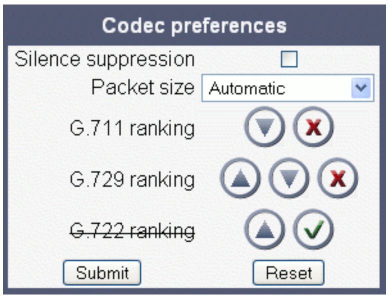
> The fixed packet sizes are used only in DMC (Direct Media Connection) connections only.

Data required

- **Silence suppression**: Suppression of data transmission on no conversation.
  Value range: "On", "Off"
  Default: "Off"
- **Packet size**: Size of RTP packets in milliseconds.
  Value range: "10 ms", "20ms", "30ms", "Automatic"
  Default: "Automatic"
- **G.711**: Parameters for the G. 711 codec.
  Value Range: "Choice 1", "Choice 2", "Choice 3", "Disabled", "Enabled"
  Default: "Choice 1"
- **G.729**: Parameters for the G. 729 codec.
  Value Range: "Choice 1", "Choice 2", "Choice 3", "Disabled", "Enabled"
  Default: "Choice 2"
- **G.722**: Parameters for the G. 722 codec.
  Value Range: "Choice 1", "Choice 2", "Choice 3", "Disabled", "Enabled"
  Default: "Disabled"

Administration via WBM

Speech > Codec preferences



Administration via Local Phone

- Administration
  - Speech
    - Codec Preferences
      - **Silence suppression**
      - **Packet size**
      - **G.711**
      - **G.729**
      - **G.722**

3.16.3 Audio Settings

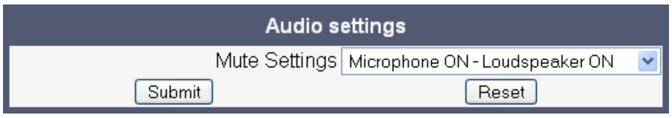
The usage of microphone and speaker for speakerphone mode can be controlled by the administrator.

Both microphone and loudspeaker can be switched on or off separately. By default, both microphone and loudspeaker are switched on.

> The microphone control is not valid for OpenStage 20E, as this model has no built-in microphone.

Administration via WBM

Speech > Audio Settings



Administration via Local Phone

```
|--- Administration
    |--- Speech
        |--- Audio Settings
            |--- Disable microphone
            |--- Disable loudspeech
```

3.17 Applications

3.17.1 XML Applications/Xpressions (OpenStage 60/80)

3.17.1.1 Setup/Configuration

The XML interface enables server-based applications with a set of GUI elements. The technologies commonly used in web applications can be used: Java Servlets, JSP, PHP, CGI etc., delivered by servers such as Tomcat, Apache, Microsoft IIS.

> A maximum number of 20 XML applications can be configured on OpenStage 60/80 phones.

There are several types of XML applications, which mainly differ in the way they are started and stopped:

- Regular XML applications are started by navigating to the applications menu using the key, or by pressing a programmable key (see Section 3.7.28, "Start Application"). They can be stopped via the applications menu. Regular XML applications are configured via **Applications** > **XML applications** > **Add application**.
- Xpressions is a special Unified Communications application which also uses the XML interface. Thus, the configuration is just the same as with other XML applications, except a few parameters, which are pre-configured. For details, please refer to the relevant Xpressions documentation. When configured on the phone, a press on the messages mode key will invoke this application. Xpressions is configured via **Applications** > **XML applications** > **Xpressions**.
- A messages application is configured like a regular application. It is started and stopped via the messages mode key, thus enabling the deployment of an alternative voicemail server. From firmware version V2R1 onwards, the XML application can control the white LED which indicates new messages. A messages application is configured via **Applications** > **XML applications** > **Add messages application**.
- A phonebook application is configured like a regular application. It is started and stopped via the phonebook mode key, thus enabling the deployment of a remote phonebook in place of the personal (local) or corporate (LDAP) phonebook. A messages application is configured via **Applications** > **XML applications** > **Add phonebook application**.
- A call log application is configured like a regular application. It is is started and stopped via the call log mode key, thus enabling the deployment of a remote application that handles call history. From firmware version V2R1 onwards, the XML application can control the white LED which indicates missed calls. A call log application is configured via **Applications** > **XML applications** > **Add call log application**.

- A help application (firmware version V2R1 onwards) is configured like a regular application. It is is started and stopped via the help mode key ?, thus enabling the deployment of a remote help. A help application is configured via **Applications** > **XML applications** > **Add help application**.

For detailed information about the OpenStage XML application interface, please see the OpenStage 60/80 XML Applications Developer's Guide. You can find the current version under http://wiki.siemens-enterprise.com/index.php/OpenStage_XML_Applications

To set up a new XML application, enter the access data for the application on the server, which is described in the following.

The **Display name** can be defined freely. This name will appear in the applications tab once the application is configured, and it will appear in a newy created tab when the application is running. With Xpressions, this value is predefined as "Xpressions".

The **Application name** is used by the phone software to identify the XML application running on the phone. With Xpressions, this value is predefined as "Xpressions".

The **Protocol** for exchanging XML data with the server-side program can be set to "HTTP" or "HTTPS".

The **HTTP Server address** is the IP address or domain name of the server which hosts the remote program. **Server port number** specifies the corresponding port.

Program name/Program name on server specifies the relative path to the servlet or to the first XML page of the application on the server. The relative path refers to the root directory for documents on the web server. The program name cannot be longer than 100 characters.

Auto start (V2R1 onwards) determines whether the application is started automatically on phone startup. If activated, the application will be ready without delay as soon as the user presses the corresponding start key resp. navigates to the application in the application menu.

XML trace enabled determines whether debugging information is sent to a special debugging program on the remote server. The relative path for the debugging program is given by the **Debug program name** parameter.

XML applications can have internal tabs, if desired. The number of these tabs is specified in **Number of tabs**.

> For an XML application with a number of tabs > 0, one of the entries between **Tab 1 Application Name** and **Tab 3 Application Name** must be set to the same value as the **Application name** that it is associated with. When the XML application is started, the tab which has the same name as the XML application is the tab that initially gets focus.

All tabs start (V2R1 onwards) determines whether all tabs of the application are started automatically when the application is started.

Tab 1...3 Display Name provides the label text for the corresponding tab.

Tab 1...3 Application Name is required if the application has internal tabs. This is a unique name for the specified tab. The remote program will use this name to provide the tab with specific content.

Auto restart / **Restart after change** (V2): If checked, a running XML application is automatically restarted after it has been modified. This might be especially useful for special XML applications, like messages applications, or phonebook applications, as these cannot be stopped or restarted by the user. Please note that a restart will take place even if no changes have been made for the application selected in the **Modify/Delete application** mask, and **Submit** has been pressed. After the XML application has restarted, this option is automatically unchecked. If the option is checked whilst the XML application is not running, there will be no restart, and the option is automatically unchecked.

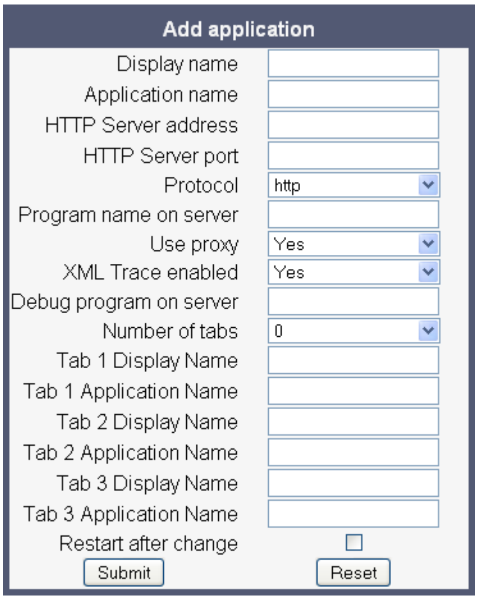
Data required

- **Display name**: Program name to be displayed on the phone.
  Value specifications:
  - It must be unique on the phone.
  - It cannot contain the '^' character.
  - It cannot not be empty.
  - Its length cannot not exceed 20 characters.
- **Application name**: Used internally to identify the XML application running on the phone.
  Value specifications:
  - It must be unique on the phone.
  - It cannot contain non-alphanumeric characters, spaces for instance.
  - The first character must be a letter.
  - It must not be empty.
  - Its length must not exceed 20 characters.
- **Protocol**: Communication protocol for the data exchange with the server.
  Value range: "HTTP", "HTTPS"
  Default: "HTTPS"
- **HTTP Server address**: IP address or domain/host name of the server that provides the application or the XML document.
- **Server port number**: Number of the port that the server uses to provide the application or XML document.
- **Program name**: Relative path to the servlet or to the first XML page of the application on the server.

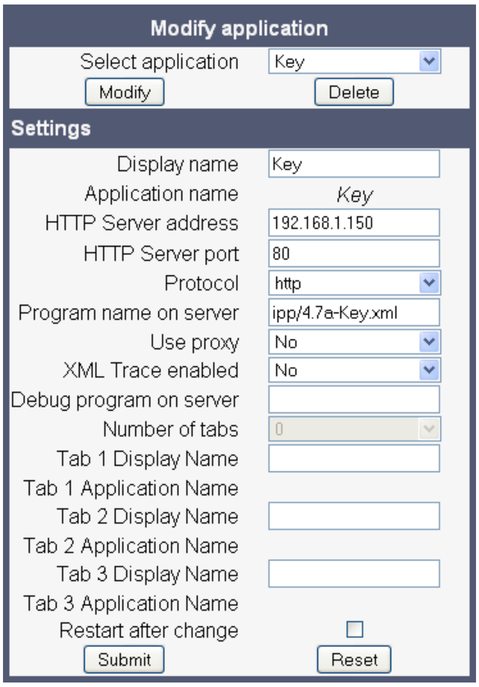
- **XML trace enabled**: Enables or Disables the debugging of the XML application.
  Value range: "Yes", "No"
  Default: "No"
- **Debug program name**: The relative path to a special servlet that receives the debug information.

Administration via WBM (up to V2R0)

Applications > XML Applications > Add application



Applications > XML Applications > Modify application



Administration via WBM (V2R1 onwards)

With firmware version V2R1, a fixed function key can be defined as a start key for an XML application, in addition to the previously available start methods. Since the parameters are the same for those types of application, only the screenshot for a regular XML application is shown underneath.

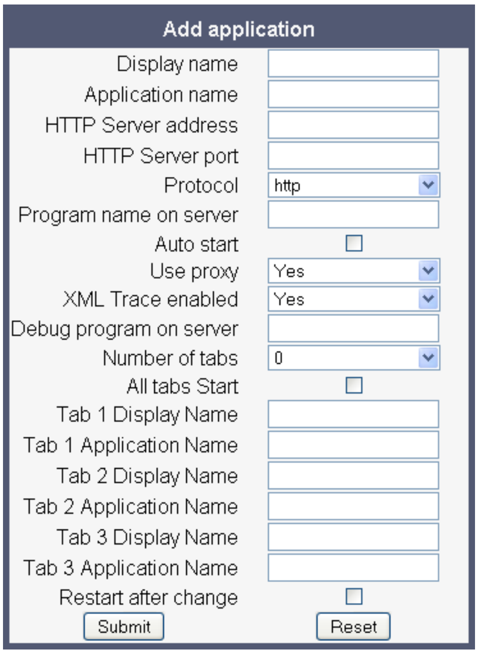
Applications > XML Applications > Add application

Applications > XML Applications > Add messages application

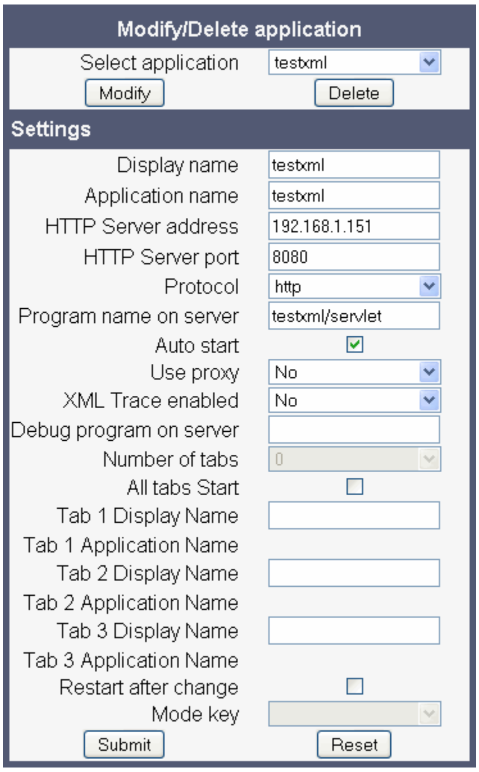
Applications > XML Applications > Add phonebook application

Applications > XML Applications > Add call log application

Applications > XML Applications > Add help application



Applications > XML Applications > Modify/Delete application



Administration via Local Phone (up to V2R0)

```
|— Administration
    |— Applications
        |— XML
            |— Add application
                |— Display name
                |— Application name
                |— Server address
                |— Server port
                |— Protocol
                |— Program name
                |— Use proxy
                |— XML trace enabled
                |— Debug program name
                |— Number of tabs
                |— Tab 1 display name
                |— Tab 1 application name
                |— Tab 2 display name
                |— Tab 2 application name
                |— Tab 3 display name
                |— Tab 3 application name
                |— Restart after change
```

Administration via Local Phone (V2R1 onwards)

```
|--- Administration
     |--- Applications
          |--- XML
               |--- Add application
                    |--- Display name
                    |--- Application name
                    |--- Server address
                    |--- Server port
                    |--- Protocol
                    |--- Program name
                    |--- Auto start
                    |--- Use proxy
                    |--- XML trace enabled
                    |--- All tabs start
                    |--- Debug program name
                    |--- Number of tabs
                    |--- Tab 1 display name
                    |--- Tab 1 application name
                    |--- Tab 2 display name
                    |--- Tab 2 application name
                    |--- Tab 3 display name
                    |--- Tab 3 application name
                    |--- Restart after change
```

3.17.1.2 HTTP Proxy

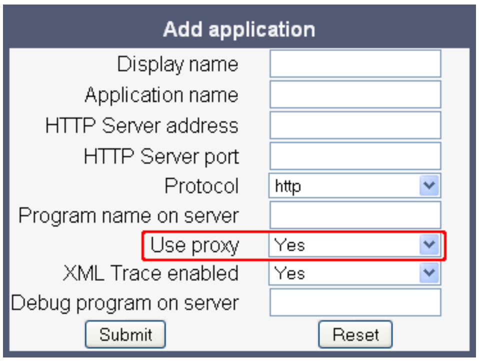
For the HTTP data transfer between the phone and the server hosting the remote program, an HTTP proxy can be used.

First, the proxy itself must be configured. Enter the IP address of the proxy it in the Network > IP configuration > HTTP proxy parameter, and the corresponding port in the Network > Port configuration > HTTP proxy parameter.

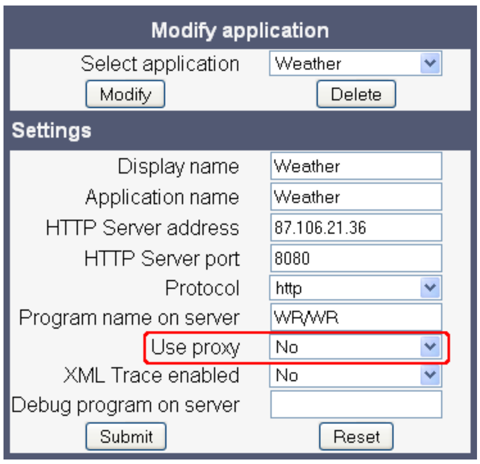
Use proxy enables or disables the use of the proxy. If disabled, the phones connects directly to the server. By default, the use of a proxy is disabled.

Administration via WBM

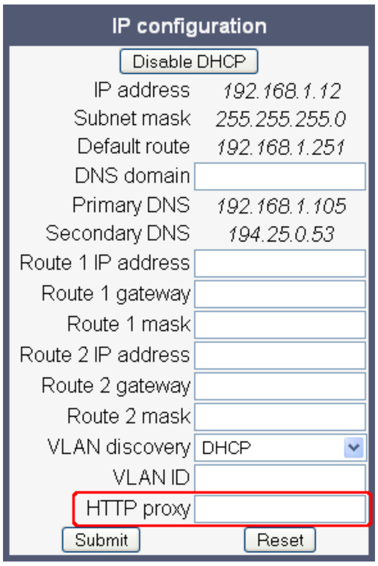
Applications > XML Applications > Add application



Applications > XML Applications > Modify application



Network > IP Configuration



Administration via Local Phone

```
|--- Administration
     |--- Applications
          |--- XML
               |--- Add application
               |    |--- Use proxy
               |--- Add Xpressions
                    |--- Use proxy
```

```
|--- Administration
     |--- Network
          |--- IP Configuration
               |--- HTTP proxy
```

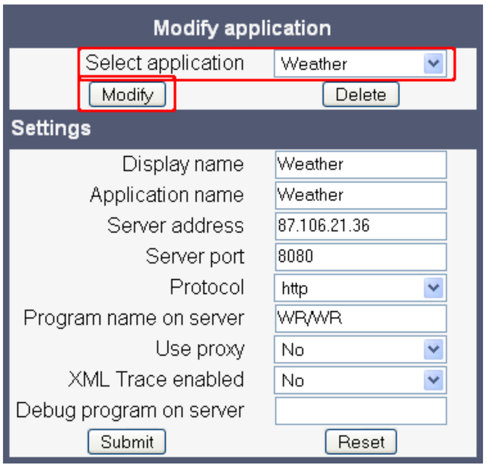
```
|--- Administration
     |--- Network
          |--- Port configuration
               |--- HTTP proxy
```

3.17.1.3 Modify an Existing Application

An existing application can be modified by changing its parameters. Please ensure to select the desired application before changing the parameters.

Administration via WBM

Applications > XML applications > Modify application



Administration via Local Phone

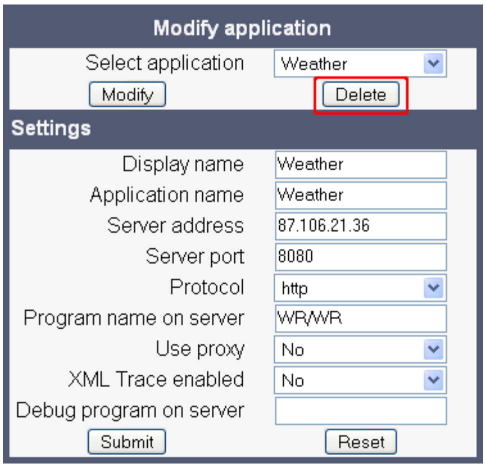
|--- Administration
    |--- Applications
        |--- XML
            |--- <Application to be modified>
                |— **Display name**
                |— **Application name**
                |— **Server address**
                |— **Server port**
                |— **Protocol**
                |— **Program name**
                |— **XML trace enabled**
                |— **Debug program name**

3.17.1.4 Remove an Existing Application

An existing application can be removed. Please ensure to select the desired application before changing the parameters.

Administration via WBM

Applications > XML applications > Modify application



Administration via Local Phone

Select the application to be deleted, and, in the context menu, select **Remove & exit**.

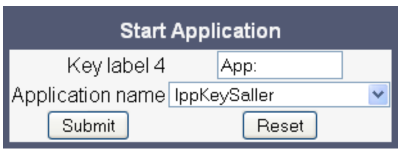
|--- Administration
    |--- Applications
        |--- XML
            |--- <Application to be deleted>

3.17.1.5 Application Start by Programmable Key

To offer more convenience to the user, a previously configured application can be started by a free programmable key. For this purpose, the appropriate **Application name** and a **Key label** must be entered.

Administration via WBM

System > Features > Program Keys

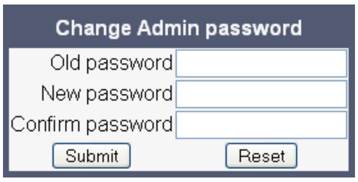


3.18 Password

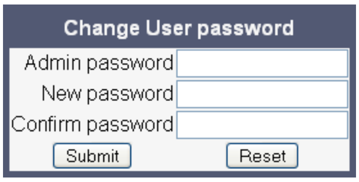
The passwords for user and administrator can be set here. They have to be confirmed after entering. The factory setting is "123456"; it should be changed after the first login.

Administration via WBM

Authentication > Change Admin password



Authentication > Change User password



Administration via Local Phone

|--- Administration
    |--- Password
        |--- **Admin**
        |--- **Confirmation**
        |--- **User**
        |--- **Confirmation**

Administration via Local Phone (V2R2 onwards)

|--- Administration

3.19 Troubleshooting: Lost Password

If the administration and/or user password is lost, and there is no DLS available, new passwords must be provided. For this purpose, a factory reset is necessary. Take the following steps to initiate a factory reset:

1. On the phone, press the key to activate the administration menu (the key toggles between the user's configuration menu and the administration menu).
2. Press the number keys 2-8-9 simultaneously. The factory reset menu opens.
3. In the input field, enter the special password for factory reset: "124816".
4. Confirm by pressing .

3.20 Restart Phone

If necessary, the phone can be restarted from the administration menu.

Administration via WBM

3.21 Factory Reset

This function resets all parameters to their factory settings. A special reset password is required for this operation: "124816".

Administration via WBM

Maintenance > Factory reset

Administration via Local Phone

|--- Administration
    |--- Maintenance
        |— **Factory reset**

3.22 SSH - Secure Shell Access (V2)

With firmware V2, the phone’s operating system can be accessed via SSH for special trouble-shooting tasks. Hereby, the administrator is enabled to use the built-in Linux commands. As soon as SSH access has been enabled using the WBM, the system can be accessed by the user "admin" for a specified timespan. When this timespan has expired, no connection is possible any more. The user "admin" has the following permissions:

- Log folder and files: read only
- User data folder and files: read/write access
- Opera deploy folders and files: read only
- Version folder: read/write access; version files: read only

> It is not possible to logon as root via SSH.

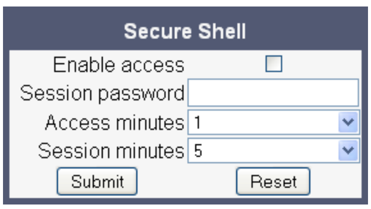
When **Enable access** is enabled, and the parameters described underneath are specified, SSH access is activated. By default, SSH access is disabled.

With the **Session password** parameter, a password for the "admin" user is created. This password is required. It will be valid for the timespan specified in the parameters described underneath.

Access minutes defines the timespan in minuts within which the SSH connection must be established. After it has expired, a logon via SSH is not possible. The possible values are 1, 3, 5, 10, 15.

Session minutes defines the maximum length in minutes for an SSH connection. After it has expired, the "admin" user is logged out. The possible values are 5, 10, 20, 30, 60.

Administration via WBM



3.23 Display License Information

The license information for the OpenStage phone software currently loaded can be viewed via the local menu.

|— Administration
    |— **Licence information**

3.24 Diagnostics

> Some of the diagnostic tools and functions may reveal personal data of the user, such as caller lists. Thus, with regards to data privacy, it is recommended to inform the user when diagnostic functions are to be executed.

3.24.1 Display General Phone Information

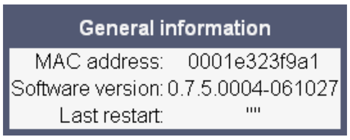
General information about the status of the phone can be displayed if desired.

Displayed Data

- **MAC address**: Shows the phone's MAC address.
- **Software version**: Displays the version of the phone's firmware.
- **Last restart**: Shows date and time of the last reboot.
- **Backlight type** (V2R2): Indicates whether the phone has a backlight, and, if applicable, the type of backlight.
  Value range: 0 (no backlight); 1 (cathode tube backlight); 2 (LED backlight)

Display on the WBM

General information



Display on the Local Phone

```
|--- Administration
     |--- General Information
          |--- MAC address
          |--- Software version
          |--- Last restart
```

3.24.2 LAN Monitoring

The LAN port mirror facility allows for monitoring all network traffic at the phone's LAN port. For further information, see Section 3.2.1, "LAN Port Settings".

Additionally, there is a possibility to monitor LAN traffic and port settings in the Local user menu:

```
|— User
    |— Network information
        |— IP address
        |— WBM URL
        |— DNS domain
        |— LAN RX
        |— LAN TX
        |— PC RX
        |— PC TX
        |— LAN autonegotiated
        |— LAN information
        |— PC autonegotiated
        |— PC information
```

3.24.3 LLDP-MED

When the phone is connected to a switch with LLDP-MED capabilities, it can receive a VLAN ID and QoS parameters and advertise its own network-related properties. The data is exchanged in TLV (Type-Length-Value) format.

Both sent and received LLDP-MED data can be monitored at the administrator interface.

> For details on LLDP-MED, please refer to the ANSI/TIA-1057 standard.

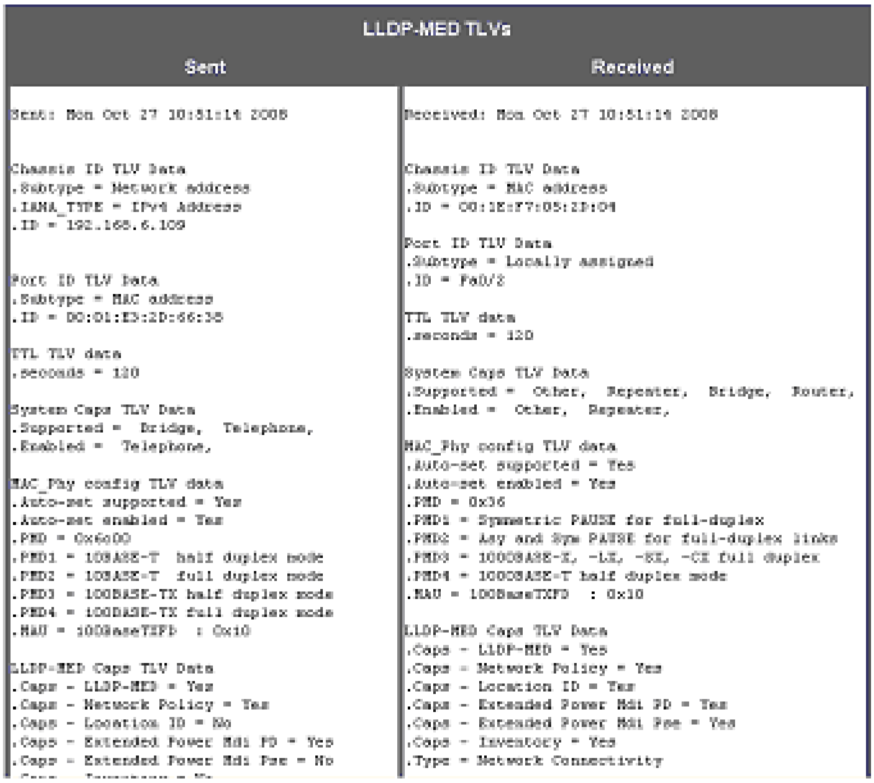
For a network configuration example that shows LLDP-MED in operation, please refer to Section 5.4, “An LLDP-Med Example”.

Displayed Data

- **Extended Power**: Power Consumption; relevant for PoE.
- **Network policy (voice)**: VLAN ID and QoS (Quality of Service) parameters for voice transport.
- **Network policy (signalling)**: VLAN ID and QoS (Quality of Service) parameters for signalling.
- **LLDEP-MED capabilities**: The LLDP-MED TLVs supported by the phone and the switch as well as the specific device class they belong to.
- **MAC_Phy configuration**: Identifies the possible duplex and bit-rate capability of the sending device, its current duplex and bit-rate capability, and whether theses settings are the result of auto-negotiation during the initialization of the link, or of manual set override actions.
- **System capabilities**: The devices advertise their potential and currently enabled functions, e. g. "Bridge", "Telephone".
- **TTL**: **T**ime **T**o **L**ive. This parameter determines how long the TLVs are valid. When expired, the device will send a new set of TLVs.

View Data From WBM

Diagnostics > LLDEP-MED TLVs



View Data From Local Menu

If both sent and received values are concordant, **OK** is appended to the parameter. If not, an error message is displayed.

```
|— Administration
    |— Network
        |— LLDP-MED operation
            |— Extended Power
            |— Network policy (voice)
            |— LLDEP-MED cap's
            |— MAC_Phy config
            |— System cap's
            |— TTL
```

3.24.4 IP Tests

For network diagnostics, the OpenStage phone can ping any host or network device to determine whether it is reachable. Additionally, the IP route to a host or network device can be traced using the traceroute tool contained in the phone software.

The **Pre Defined Ping tests** provide pinging for a pre-defined selection of servers: DLS, SIP server, and SIP registrar.

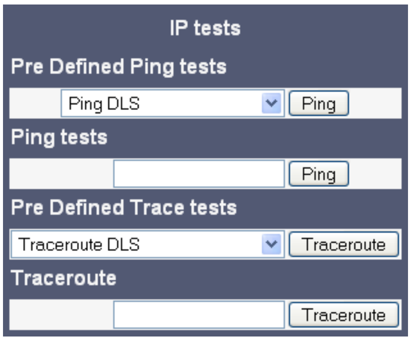
Ping tests enables the pinging of a random IP address.

The **Pre Defined Trace tests** provide traceroute tests for a pre-defined selection of servers: DLS, SIP server, and SIP registrar.

Traceroute enables traceroute tests for a random IP address.

Administration via WBM

Diagnostics > Miscellaneous > IP tests



3.24.5 Process and Memory Information

The processes currently running on the phone's operating system as well as their CPU and memory usage can be monitored here. 100 processes are monitored on the web page. For further information, please refer to the manual of the "top" command for Unix/Linux systems, or to related documentation.

With firmware version V2, the amount of free memory is checked on a regular basis in order to prevent problems caused by low memory. This check determines whether a recovery is necessary.

When **Disable reboot** is checked, no reboot will take place when a memory problem has been found. However, recovery requires a reboot.

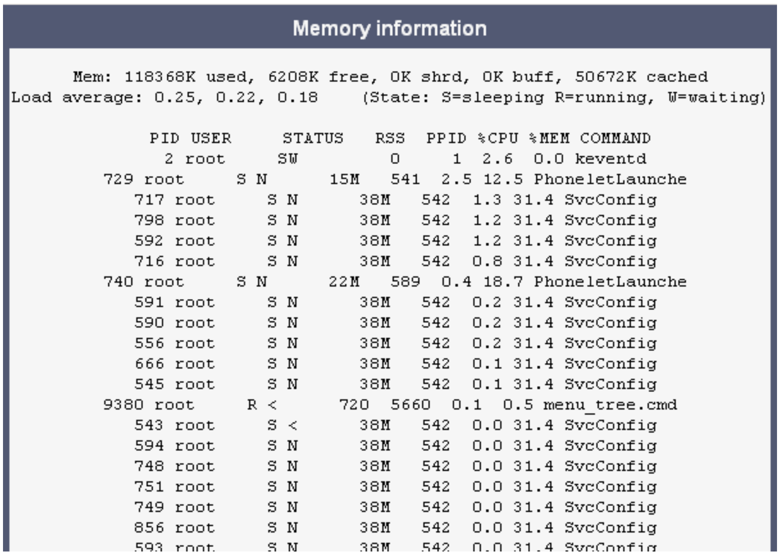
The recovery process will be triggered when the available main memory (RAM) falls below a given threshold value. As memory consumption is assumed to be higher during working hours, two thresholds are configurable. The **High Threshold (MBs)** parameter defines the threshold for off-time. For OpenStage 15/20/40, the default value is 10 MB, and for OpenStage 60/80, it is 30 MB. With **Low Threshold (MBs)**, the threshold for off-time is defined. For OpenStage 15/20/40, the default value is 8 MB, and for OpenStage 60/80, it is 20 MB.

The beginning and end of the working hours are defined in 24 hours format with **Working Hour Start** (Default: 5) and **Working Hour End** (Default: 24).

When memory shortage has occured, information about the incident is written to a log file which can be viewed via the **Download memory info file** link. If there has been a previous case of memory shortage, the corresponding log file can be viewed via **Download memory info file**.

Administration via WBM (V1R5)

Diagnostics > **Miscellaneous** > Memory information



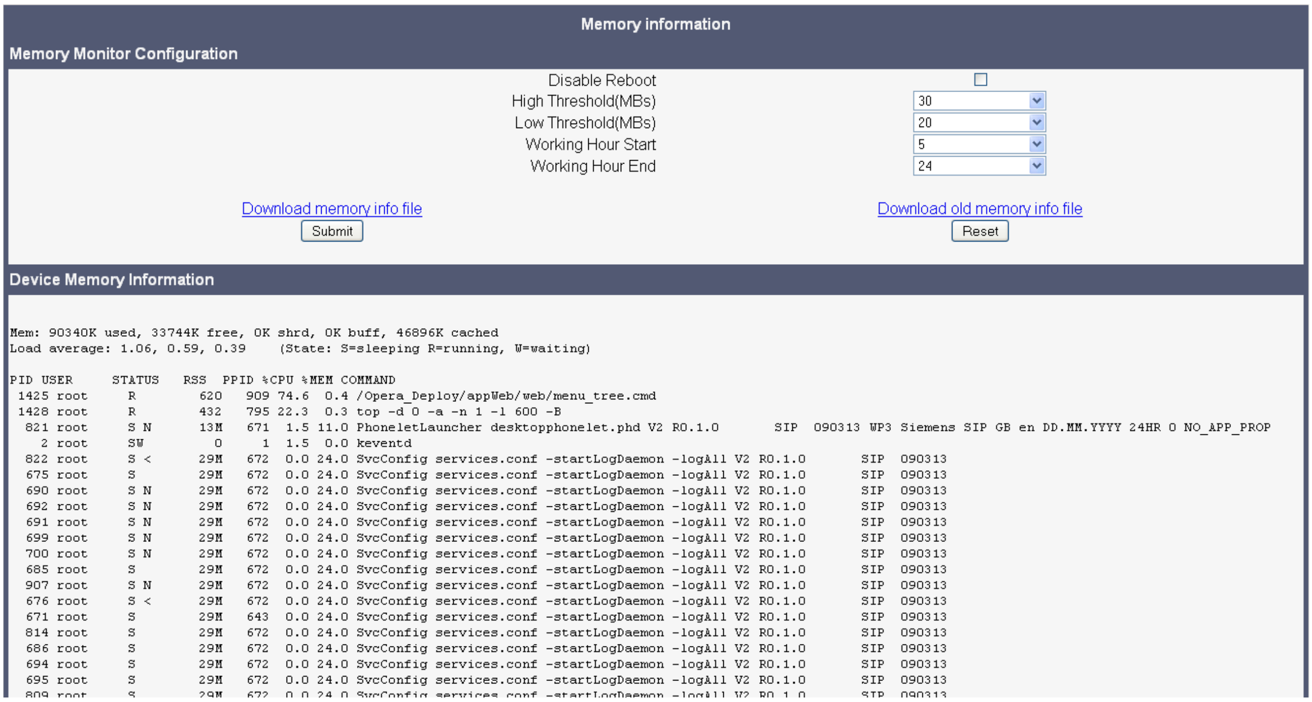
Memory information

```
Mem: 118368K used, 6208K free, 0K shrd, 0K buff, 50672K cached
Load average: 0.25, 0.22, 0.18    (State: S=sleeping R=running, W=waiting)

  PID USER     STATUS   RSS  PPID %CPU %MEM COMMAND
    2 root     SW         0     1  2.6  0.0 keventd
  729 root     S N      15M   541  2.5 12.5 PhoneletLaunche
  717 root     S N      38M   542  1.3 31.4 SvcConfig
  798 root     S N      38M   542  1.2 31.4 SvcConfig
  592 root     S N      38M   542  1.2 31.4 SvcConfig
  716 root     S N      38M   542  0.8 31.4 SvcConfig
  740 root     S N      22M   589  0.4 18.7 PhoneletLaunche
  591 root     S N      38M   542  0.2 31.4 SvcConfig
  590 root     S N      38M   542  0.2 31.4 SvcConfig
  556 root     S N      38M   542  0.2 31.4 SvcConfig
  666 root     S N      38M   542  0.1 31.4 SvcConfig
  545 root     S N      38M   542  0.1 31.4 SvcConfig
 9380 root     R <      720  5660  0.1  0.5 menu_tree.cmd
  543 root     S <      38M   542  0.0 31.4 SvcConfig
  594 root     S N      38M   542  0.0 31.4 SvcConfig
  748 root     S N      38M   542  0.0 31.4 SvcConfig
  751 root     S N      38M   542  0.0 31.4 SvcConfig
  749 root     S N      38M   542  0.0 31.4 SvcConfig
  856 root     S N      38M   542  0.0 31.4 SvcConfig
  593 root     S N      38M   542  0.0 31.4 SvcConfig
```

Administration via WBM (V2)

Diagnostics > **Miscellaneous** > Memory information



3.24.6 Fault Trace Configuration

Error tracing and logging can be configured separately for all components, i. e. the services and applications running on the OpenStage phone. The resulting files can be viewed in the WBM web pages over the **Download** links.

The **File size (bytes)** parameter sets the maximum file size. When it is reached, the data is saved as old file, and a new file is generated. From then on, the trace data is written to the new file. When the maximum file size is reached again, the data is saved as old file once more, thereby overwriting the previous old file. The default value is 65536.

> The absolute maximum file size is 6 290 000 bytes. However, on OpenStage 15/20/40 phones, a maximum size no greater than 500 000 bytes is recommended due to the amount of available memory.

The **Trace timeout (minutes)** determines when to stop tracing. When the timeout is reached, the trace settings for all components are set to OFF, but ERROR and STATUS messages are still written to the trace file ad infinitum. When the trace file has reached its maximum size, the data is saved, and a new file is created (for more information, see **File size (bytes)** above). If the value is 0, the trace data will be written without time limit.

If **Automatic clear before start** is checked, the existing trace file will be deleted on pressing the **Submit** button, and a new, empty trace file will be generated. By default, it is unchecked.

You can read the log files by clicking on the appropriate hyperlinks (the hyperlinks work only if the file in question has been created). The following logs can be viewed:

- **Download trace file**
  The trace data according to the settings specified for the services.
- **Download boot file** (not present with V2)
  The system messages of the booting process. With firmware version V2, these messages will be incorparated in the syslog file (see **Download syslog file** underneath).
- **Download saved trace file**
  Normally, the trace file is saved only in the phone RAM. When the phone restarts in a controlled manner, the trace file will be saved in permanent memory.
- **Download saved boot file** (not present with V2)
  Normally, the boot file is saved only in the phone RAM. When the phone restarts in a controlled manner, the boot file will be saved in permanent memory. With firmware version V2, these messages will be incorparated in the syslog file (see **Download syslog file** underneath).
- **Download upgrade trace file**
  The trace log created during a software upgrade.
- **Download upgrade error file**
  The error messages created during a software upgrade. With firmware version V2, these messages will be incorparated in the syslog file (see **Download syslog file** underneath).

- **Download exception file** (not present with V2)
  If an exceptions occurs in a process running on the phone, a message is written to this file. With firmware version V2, these messages will be incorparated in the syslog file (see **Download syslog file** underneath).
- **Download old exception file** (not present with V2)
  The exception file is stored permanent memory. When the file has reached its size limit, it will be saved as old exception file, and the current exception file is emptied for future messages. The old exception file can be viewed here.
- **Download old trace file**
  The trace file is stored in permanent memory. When the file has reached its size limit, it will be saved as old trace file, and the current exception file is emptied for future messages. The old trace file can be viewed here.
- **Download error file** (not present with V2)
  All error messages the phone has created, according to the settings for the individual services.
- **Download syslog file**
  Messages from the phone's operating system, including error and exception messages.
- **Download old syslog file** (V2)
  Old messages from the phone's operating system.
- **Download saved syslog file** (V2)
  Saved messages from the phone's operating system.
- **Download Database file** (V2)
  Configuration parameters of the phone in SQLite format.
- **Download HPT remote service log file** (V2)
  Log data from the HPT service.
- **Download dial plan file**
  If a dial plan has been uploaded to the phone, it is displayed here, along with its status (enabled/disabled) and error status. For details, please refer to Section 3.11.3, "Dial Plan (V2)" and Section 5.5, "Dial Plan (V2)".

By pressing **Submit**, the trace settings are submitted to the phone. With **Reset**, the recent changes can be canceled.

The following trace levels can be selected:
- **OFF**: Default value. Only error messages are stored.
- **ERROR**: Error messages are stored.
- **TRACE**: Trace messages are stored. These contain detailed information about the processes taking place in the phone.
- **DEBUG**: All types of messages are stored.

Brief Descriptions of the Components/Services

- **Administration**
  Deals with the changing and setting of parameters within the phone database, from both the User and Admin menus.
- **Application framework**
  All applications within the phone, e.g. Call view, Call log or Phonebook, are run within the application framework. It is responsible for the switching between different applications and bringing them into and out of focus as appropriate.
- **Application menu**
  This is where applications to be run on the phone can be started and stopped.
- **Bluetooth service**
  Handles the Bluetooth interactions between external Bluetooth devices and the phone.
  Bluetooth is available only on OpenStage 60/80 phones.
- **Call log**
  The Call log application displays the call history of the phone.
- **Call view**
  Handles the representation of telephony calls on the phone screen.
- **Certificate management**
  Handles the verification and exchange of certificates for security and verification purposes.
- **Communications**
  Involved in the passing of call related information and signaling to and from the CSTA service.
- **Component registrar**
  Handles data relating to the type of phone, e.g. OpenStage 20/40 HFA/SIP, OpenStage 60/80 HFA/SIP.
- **CSTA service**
  Any CSTA messages are handled by this service. CSTA messages are used within the phone by all services as a common call progression and control protocol.
- **Data Access service**
  Allows other services to access the data held within the phone database.
- **Desktop**
  Responsible for the shared parts of the phone display. Primarily these are the status bar at the top of the screen and the FPK labels.
- **Digit analysis service**
  Analyses and modifies digit streams which are sent to and received by the phone, e.g. canonical conversion.
- **Directory service**
  Performs a look up for data in the phonebook, trying to match incoming and outgoing numbers with entries in the phonebook.
- **DLS client management**

Handles interactions with the DLS (Deployment Service).

- **Health service**
  Monitors other components of the phone for diagnostic purposes and provides a logging interface for the services in the phone.
- **Help**
  Handles the help function.
- **Instrumentation service**
  Used by the Husim phone tester to exchange data with the phone for remote control, testing and monitoring purposes.
- **Java**
  Any Java applications running on the phone will be run in the Java sandbox controlled by the Java service.
- **Journal service**
  Responsible for saving and retrieving call history information, which is used by the Call log application.
- **Media control service**
  Provides the control of media streams (voice, tones, ringing etc.) within the phone.
- **Media processing service**
  This is a layer of software between the media control service, the tone generation, and voice engine services. It is also involved in the switching of audio devices such as the handset and loudspeaker.
- **Media recording service (V2R2 onwards)**
  Logs the data flow generated with call recording.
- **Mobility service**
  Handles the mobility feature whereby users can log onto different phones and have them configured to their own profile.
- **OBEX service**
  Involved with Bluetooth accesses to the phone.
  Bluetooth is available only on OpenStage 60/80 phones.
- **Openstage client management**
  Provides a means by which other services within the phone can interact with the database.
- **Phonebook**
  Responsible for the phonebook application.
- **POT service** (not present with V2)
  Takes over control of basic telephony if the callview application fails.
- **Performance Marks**
  Aid for measuring the performance of the phone. For events triggered by the user, a performance mark is written to the trace file, together with a timestamp in the format hh:mm:ss yyyy.milliseconds, and information about the event. The timespan between two performance marks is an indicator for the performance of the phone.

> The trace level must be set to "TRACE" or "DEBUG".

- **Password management service**
  Verifies passwords used in the phone.
- **Physical interface service**
  Handles any interactions with the phone via the keypad, mode keys, fixed feature buttons, clickwheel and slider.
- **Service framework**
  This is the environment within which other phone services operate. It is involved in the starting and stopping of services.
- **Service registry**
  Keeps a record of all services currently running inside the phone.
- **SIP call control**
  Contains the call model for the phone and is associated with telephony and call handling.
- **SIP messages**
  Traces the SIP messages exchanged by the phone.

> After changing the level for the tracing of SIP messages, the phone must be rebooted. Otherwise the changes would have no effect.

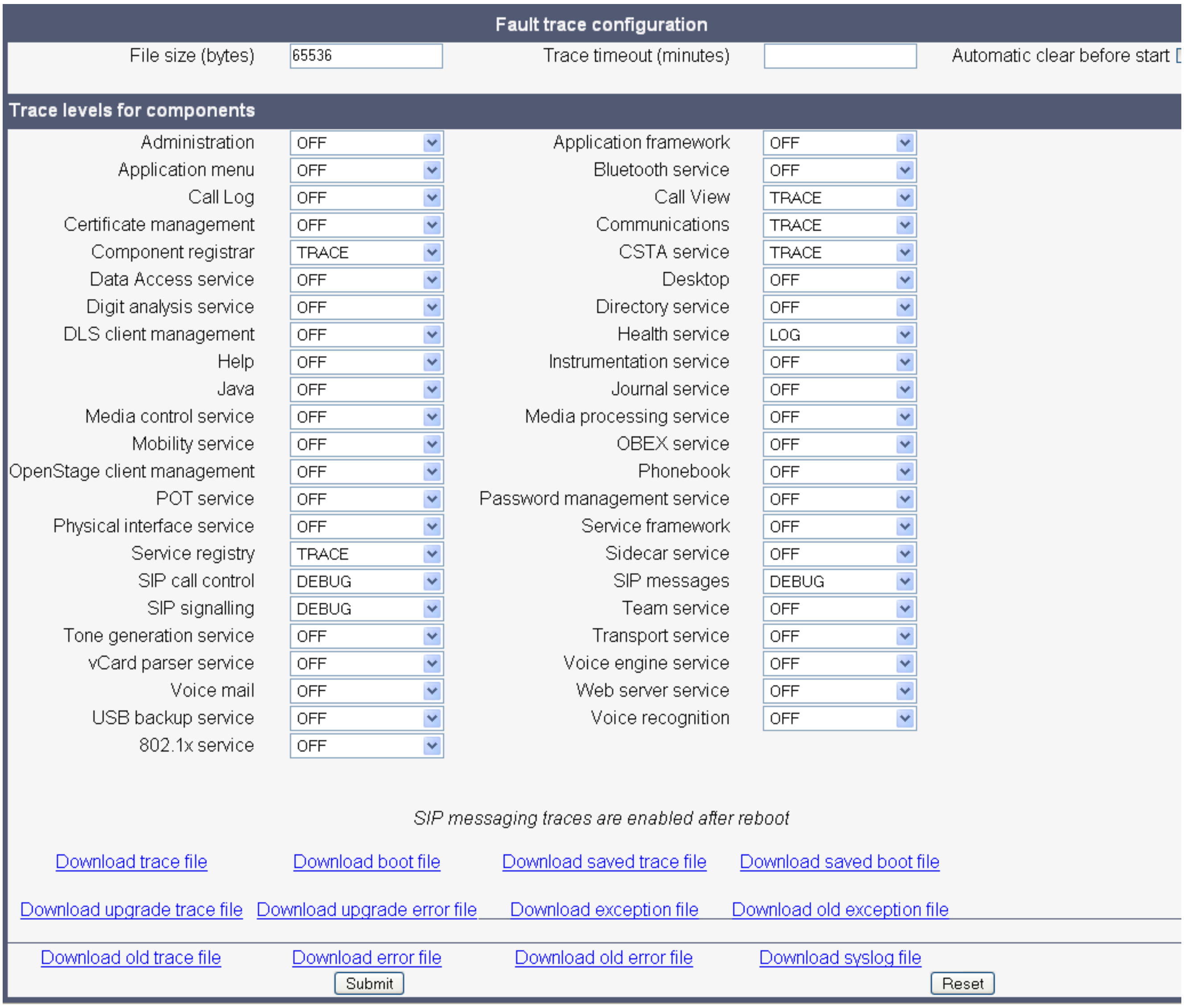
- **SIP signalling**
  Involved in the creation and parsing of SIP messages. This service communicates directly with the SIP stack.
- **Sidecar service**
  Handles interactions between the phone and any attached sidecars.
- **Team Service**
  Primarily concerned with keyset operation.
- **Tone generation service**
  Handles the generation of the tones and ringers on the phone.
- **Transport service**
  Provides the IP (LAN) interface between the phone and the outside world.
- **USB backup service**
  Used to make backup/restore to/from USB stick by using password. This item is available in the phone GUI.
- **vCard parser service**
  Handles parsing and identification of VCard information while sending or getting VCards via Bluetooth.
- **Voice engine**

Provides a switching mechanism for voice streams within the phone. This component is also involved in QDC, Music on hold and voice instrumentation.

- **Voice mail**
  Handles the voice mail functionality.
- **Voice recognition**
  Used by the voice dial facility for recognizing spoken dialing commands.
- **Web server service**
  Provides access to the phone via web browser.
- **802.1x service**
  Provides authentication to devices attached to a LAN port, establishing a point-to-point connection or preventing access from that port if authentication fails. The service is used for certain closed wireless access points.
- **Clock service**
  Handles the phone’s time and date, including daylight saving and NTP functionality.

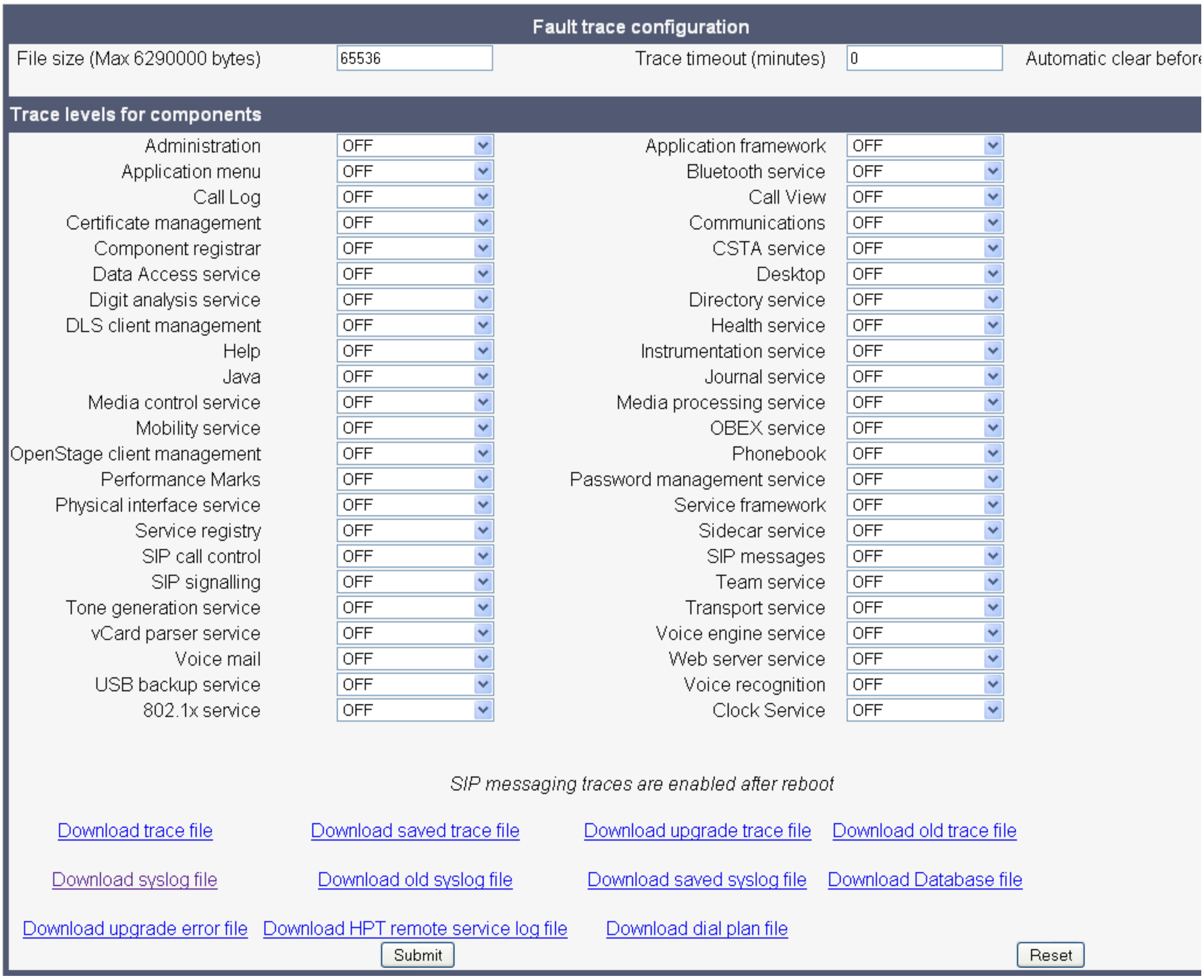
Administration via WBM (V1R5)

Diagnostics > Fault Trace Configuration



Administration via WBM (V2)

Diagnostics > Fault Trace Configuration


Fault trace configuration

File size (Max 6290000 bytes)	65536	Trace timeout (minutes)	0	Automatic clear before

Trace levels for components

Administration	OFF	Application framework	OFF
Application menu	OFF	Bluetooth service	OFF
Call Log	OFF	Call View	OFF
Certificate management	OFF	Communications	OFF
Component registrar	OFF	CSTA service	OFF
Data Access service	OFF	Desktop	OFF
Digit analysis service	OFF	Directory service	OFF
DLS client management	OFF	Health service	OFF
Help	OFF	Instrumentation service	OFF
Java	OFF	Journal service	OFF
Media control service	OFF	Media processing service	OFF
Mobility service	OFF	OBEX service	OFF
OpenStage client management	OFF	Phonebook	OFF
Performance Marks	OFF	Password management service	OFF
Physical interface service	OFF	Service framework	OFF
Service registry	OFF	Sidecar service	OFF
SIP call control	OFF	SIP messages	OFF
SIP signalling	OFF	Team service	OFF
Tone generation service	OFF	Transport service	OFF
vCard parser service	OFF	Voice engine service	OFF
Voice mail	OFF	Web server service	OFF
USB backup service	OFF	Voice recognition	OFF
802.1x service	OFF	Clock Service	OFF

SIP messaging traces are enabled after reboot

Download trace file | Download saved trace file | Download upgrade trace file | Download old trace file

Download syslog file | Download old syslog file | Download saved syslog file | Download Database file

Download upgrade error file | Download HPT remote service log file | Download dial plan file

Submit | Reset

3.24.7 Easy Trace Profiles

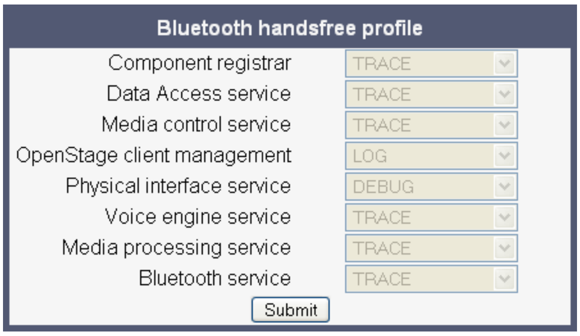
In order to simplify tracing for a specific problem, the tracing levels can be adjusted using pre-defined settings. The Easy Trace profiles provide settings for a specific area, e. g. call connection. On pressing **Submit**, those pre-defined settings are sent to the phone. If desired, the settings can be modified anytime using the general mask for trace configuration under **Diagnostics** > Fault Trace Configuration (see Section 3.24.6, “Fault Trace Configuration”).

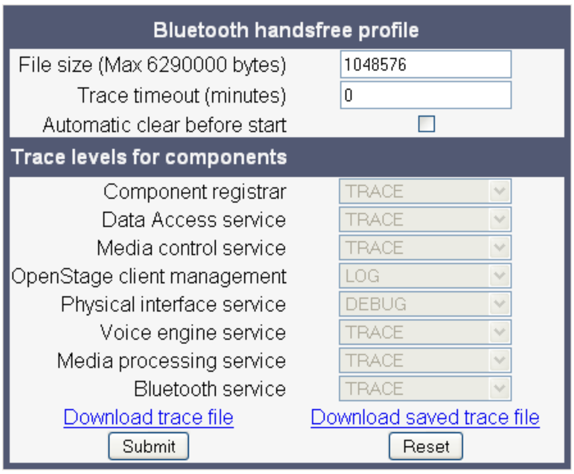
If desired, the tracing for all services can be disabled (see Section 3.24.7.23, “No Tracing for All Services”).

The following sections describe the Easy Trace profiles available for the phone.

3.24.7.1 Bluetooth Handsfree

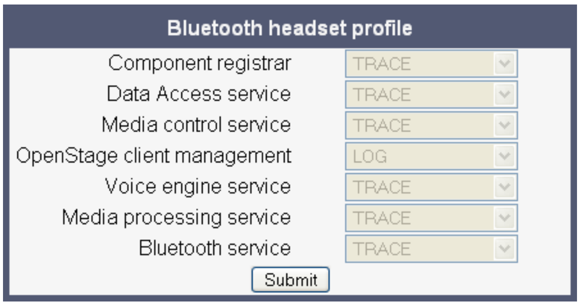
Diagnostics > Easy Trace Profiles > Bluetooth handsfree profile

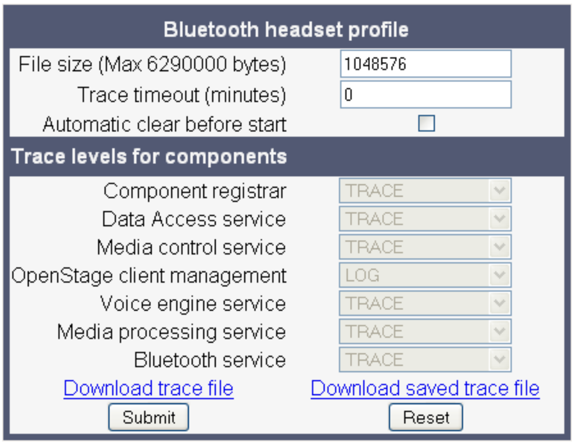




3.24.7.2 Bluetooth Headset

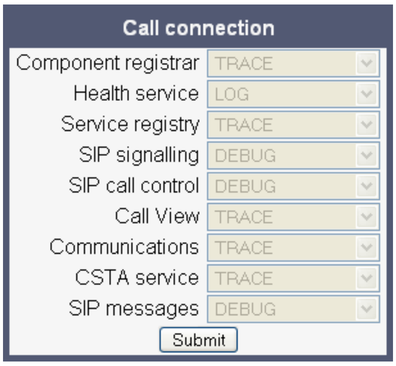
Diagnostics > Easy Trace Profiles > Bluetooth headset profile

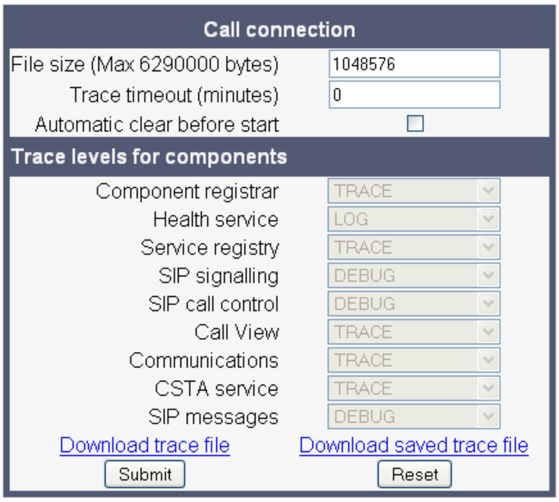




3.24.7.3 Call Connection

Diagnostics > Easy Trace Profiles > Call connection

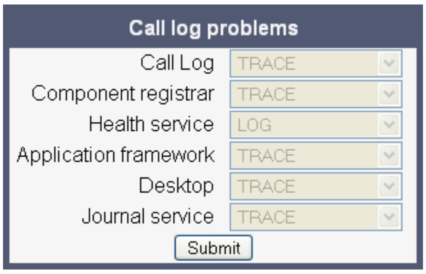


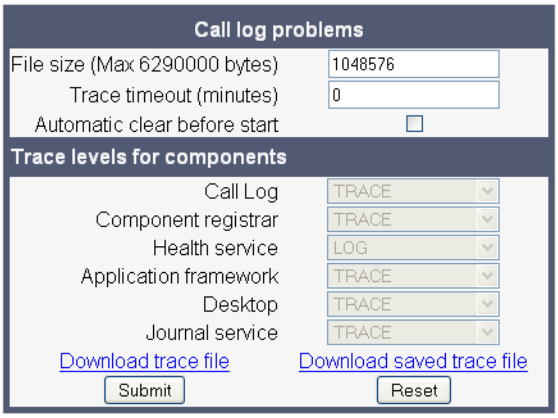


This Easy Trace profile contains the tracing of SIP messages. Please note that after changing the level for the tracing of SIP messages, the phone must be rebooted.

3.24.7.4 Call Log

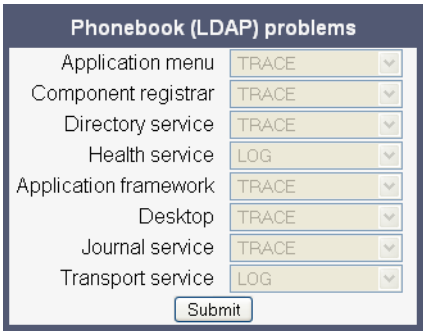
Diagnostics > Easy Trace Profiles > Call log problems

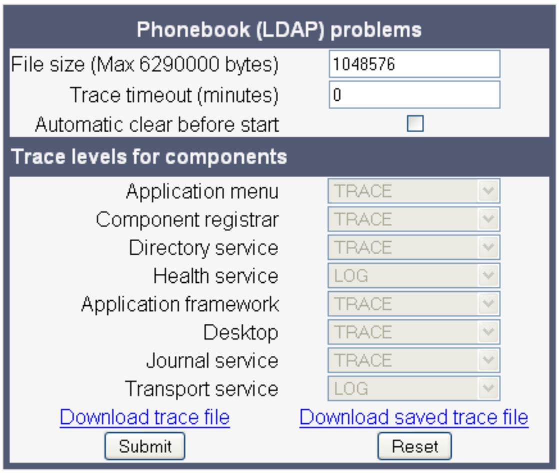

Call log problems
Call Log
TRACE
Component registrar
TRACE
Health service
LOG
Application framework
TRACE
Desktop
TRACE
Journal service
TRACE
Submit


Call log problems
File size (Max 6290000 bytes)
1048576
Trace timeout (minutes)
0
Automatic clear before start
Trace levels for components
Call Log
TRACE
Component registrar
TRACE
Health service
LOG
Application framework
TRACE
Desktop
TRACE
Journal service
TRACE
Download trace file
Download saved trace file
Submit
Reset

3.24.7.5 LDAP Phonebook

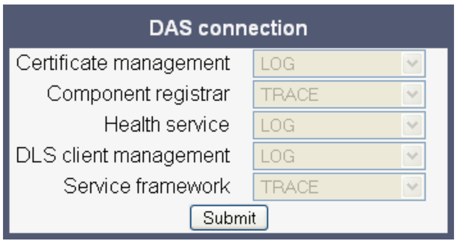
Diagnostics > Easy Trace Profiles > Phonebook (LDAP) problems

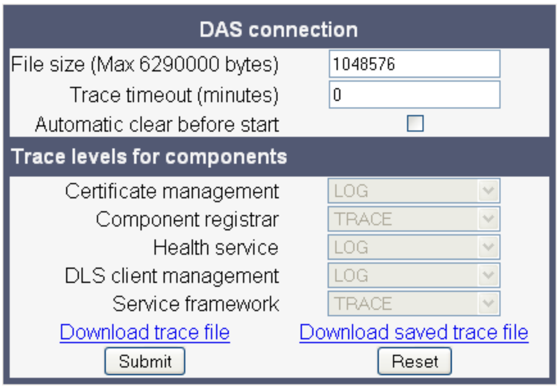




3.24.7.6 DAS Connection

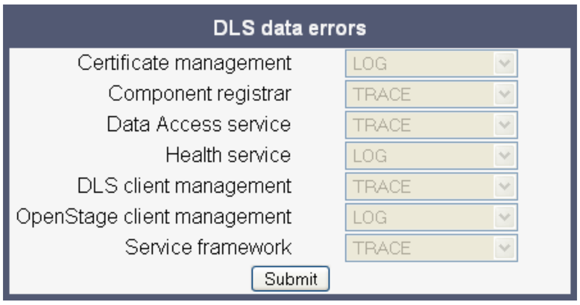
Diagnostics > Easy Trace Profiles > DAS connection

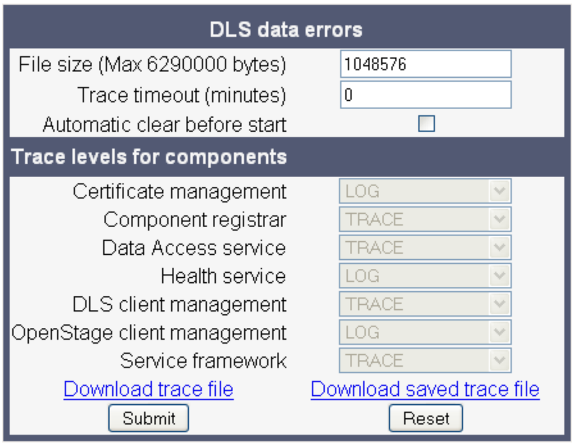




3.24.7.7 DLS Data Errors

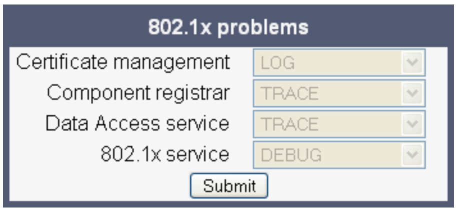
Diagnostics > Easy Trace Profiles > DLS data errors

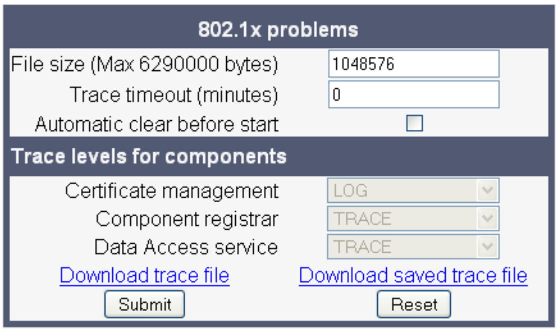




3.24.7.8 802.1x

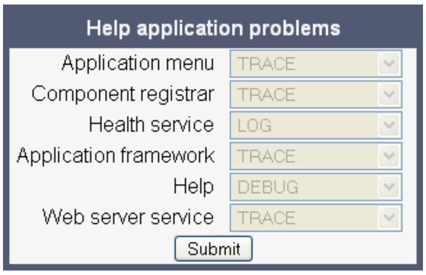
Diagnostics > Easy Trace Profiles > 802.1x

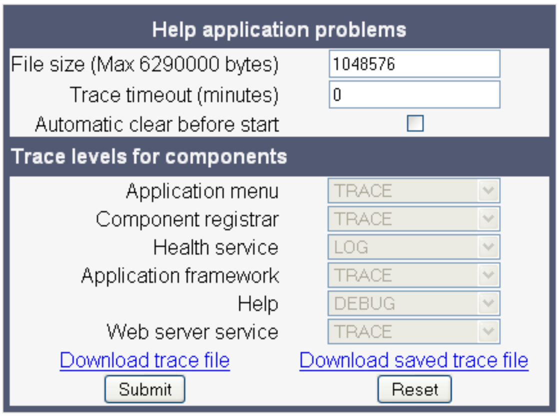




3.24.7.9 Help Application

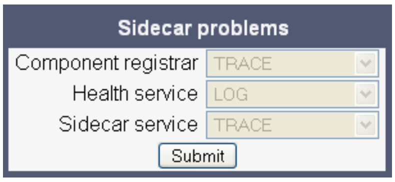
Diagnostics > Easy Trace Profiles > Help application problems

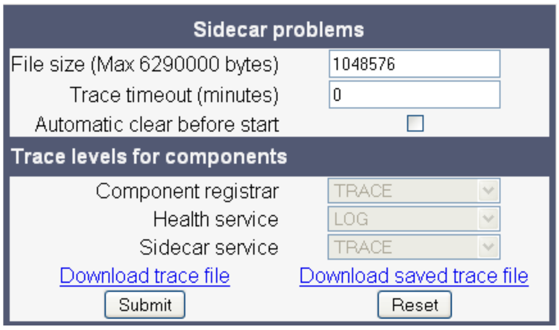




3.24.7.10 Sidecar

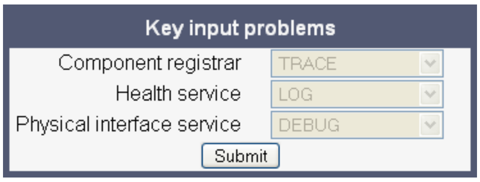
Diagnostics > Easy Trace Profiles > Sidecar problems

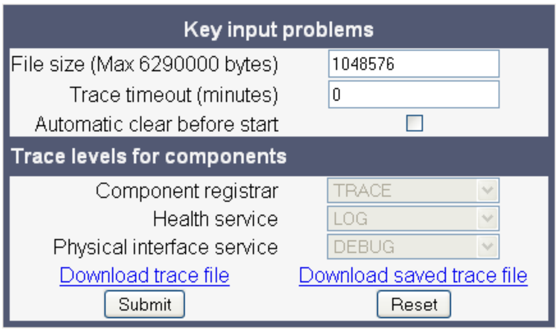




3.24.7.11 Key Input

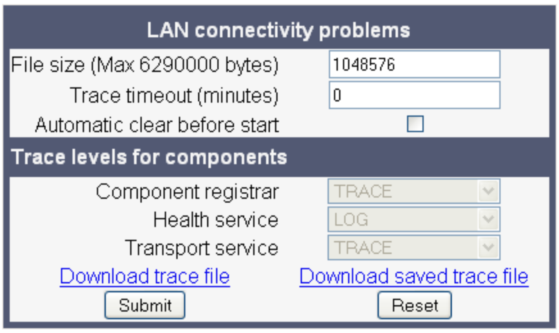
Diagnostics > Easy Trace Profiles > Key input problems





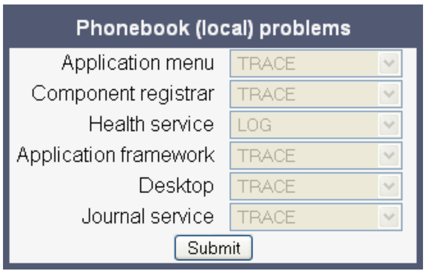
3.24.7.12 LAN Connectivity

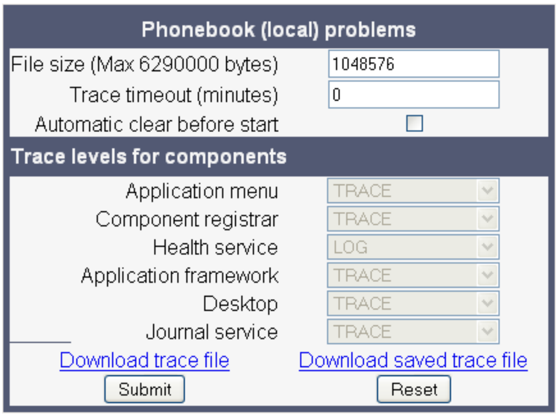
Diagnostics > Easy Trace Profiles > LAN connectivity problems



3.24.7.13 Local Phonebook

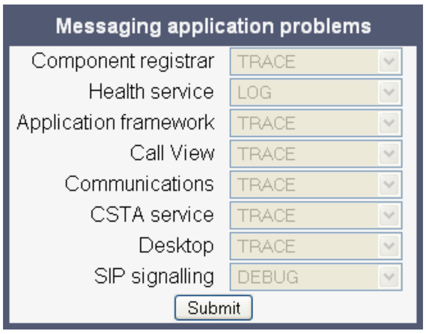
Diagnostics > Easy Trace Profiles > Phonebook (local) problems

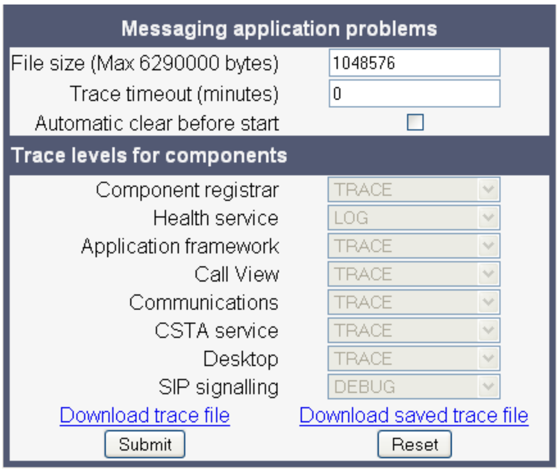




3.24.7.14 Messaging

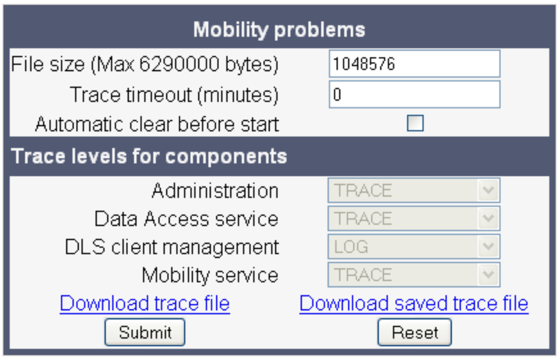
Diagnostics > Easy Trace Profiles > Messaging application problems





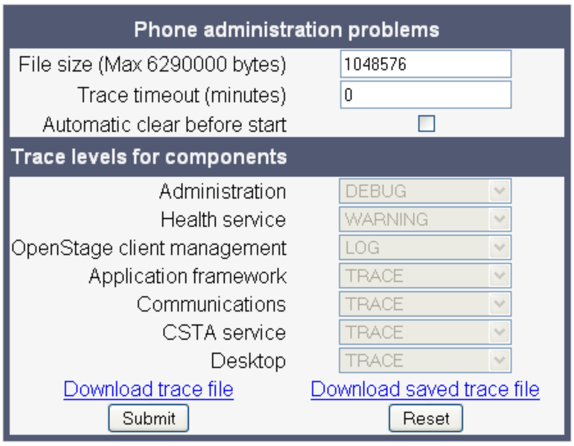
3.24.7.15 Mobility

Diagnostics > Easy Trace Profiles > Mobility problems



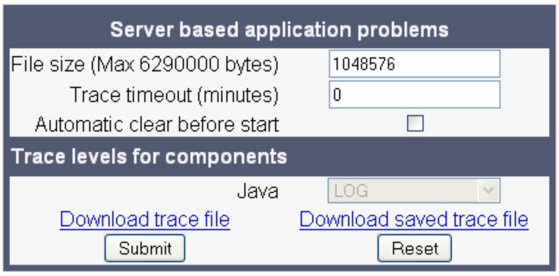
3.24.7.16 Phone administration

Diagnostics > Easy Trace Profiles > Phone administration problems



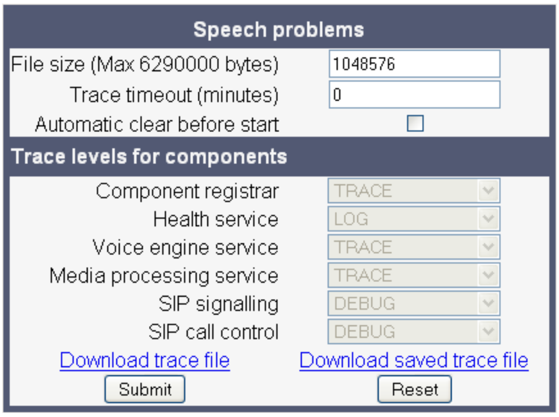
3.24.7.17 Server based applications

Diagnostics > Easy Trace Profiles > Server based application problems



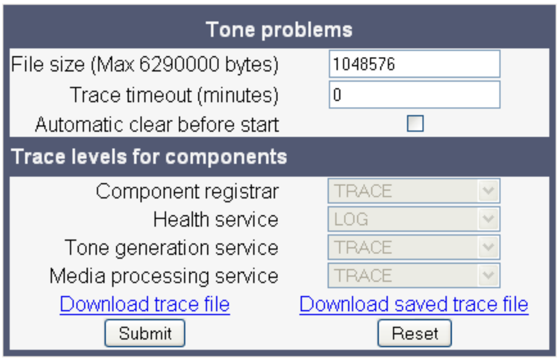
3.24.7.18 Speech

Diagnostics > Easy Trace Profiles > Speech problems



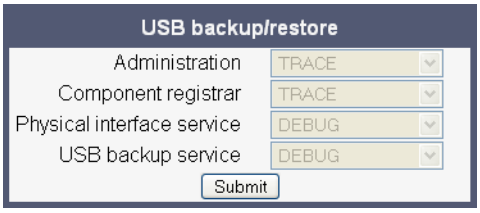
3.24.7.19 Tone

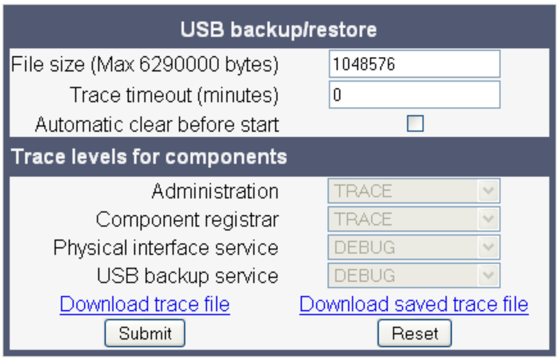
Diagnostics > Easy Trace Profiles > Tone problems



3.24.7.20 USB Backup/Restore

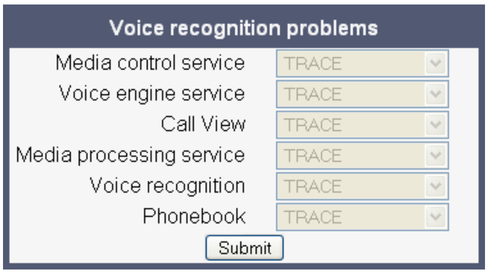
Diagnostics > Easy Trace Profiles > USB backup/restore

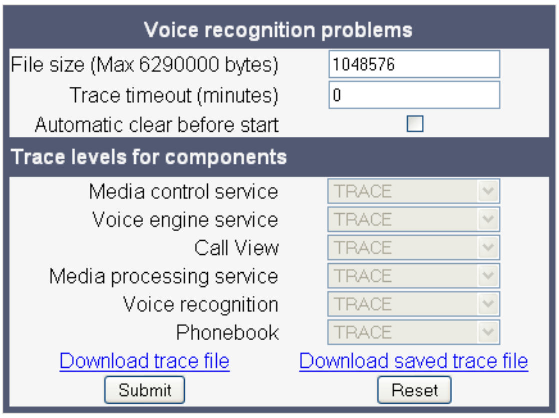




3.24.7.21 Voice Dialling

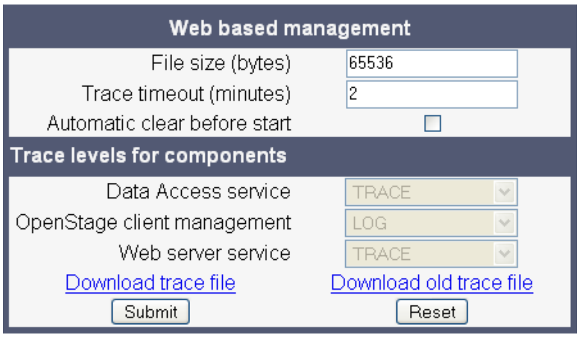
Diagnostics > Easy Trace Profiles > Voice recognition problems

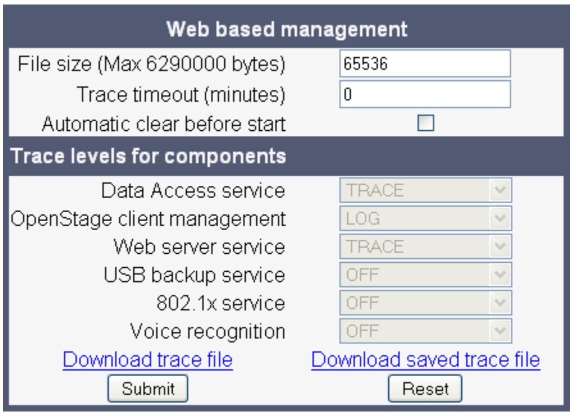




3.24.7.22 Web Based Management

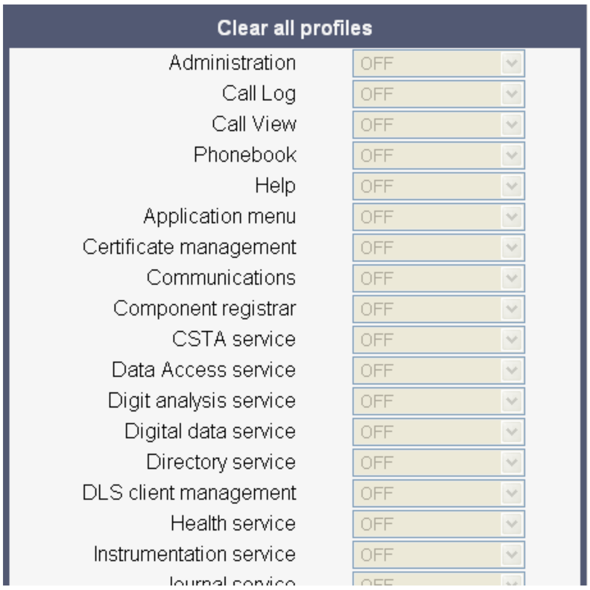
Diagnostics > Easy Trace Profiles > Web based management

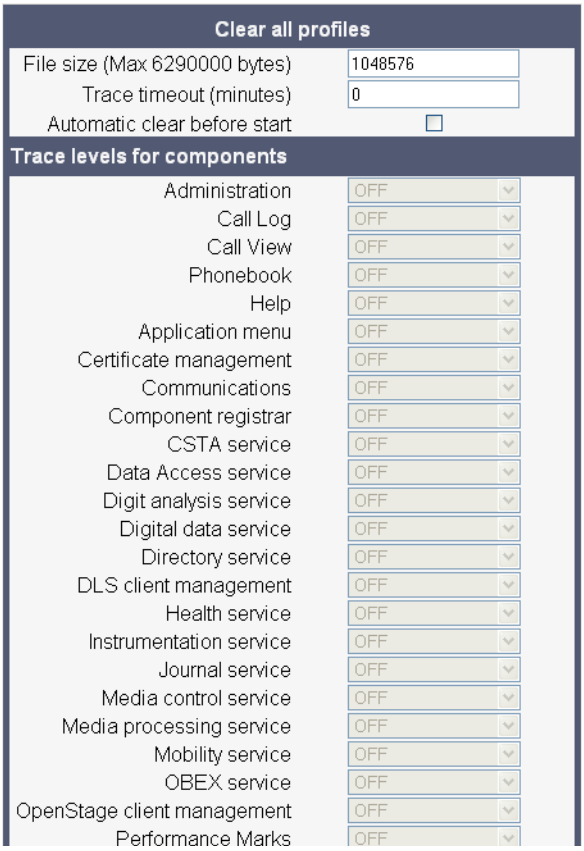

Web based management
File size (bytes)
65536
Trace timeout (minutes)
2
Automatic clear before start
Trace levels for components
Data Access service
TRACE
OpenStage client management
LOG
Web server service
TRACE
Download trace file
Download old trace file
Submit
Reset


Web based management
File size (Max 6290000 bytes)
65536
Trace timeout (minutes)
0
Automatic clear before start
Trace levels for components
Data Access service
TRACE
OpenStage client management
LOG
Web server service
TRACE
USB backup service
OFF
802.1x service
OFF
Voice recognition
OFF
Download trace file
Download saved trace file
Submit
Reset

3.24.7.23 No Tracing for All Services

Diagnostics > Easy Trace Profiles > Clear all profiles




Clear all profiles
File size (Max 6290000 bytes)
1048576
Trace timeout (minutes)
0
Automatic clear before start
Trace levels for components
Administration
Call Log
Call View
Phonebook
Help
Application menu
Certificate management
Communications
Component registrar
CSTA service
Data Access service
Digit analysis service
Digital data service
Directory service
DLS client management
Health service
Instrumentation service
Journal service
Media control service
Media processing service
Mobility service
OBEX service
OpenStage client management
Performance Marks
OFF

3.24.8 Bluetooth Advanced Traces (V2)

For OpenStage 60/80 phones with firmware V2, low level Bluetooth traces can be controlled and viewed via web interface, in addition to the tracing facilities available in previous firmware versions (see Section 3.24.6, “Fault Trace Configuration”). Internally, the phone uses the hc-dump utility for creating the traces. It is also possible to run the trace from the shell via SSH (for information about the SSH access, please refer to Section 3.22, “SSH - Secure Shell Access (V2)”).

If **Automatic clear before start** is enabled, the log file will be emptied before the **Start** button is pressed, so that the log file will only containd newly created entries. By default, this parameter is enabled.

The **File size (Max 6290000 bytes)** parameter determines the maximum size of the log file. If this value is exceeded, no more data will be written to the file. The default value is 265536.

If **Extended dump** is enabled, all hexadecimal and ASCII data is displayed for each packet. If disabled, only the packet type is displayed. By default, this parameter is enabled.

If **Verbose decoding** is enabled, the packets are decoded in a more verbose way. By default, this parameter is enabled.

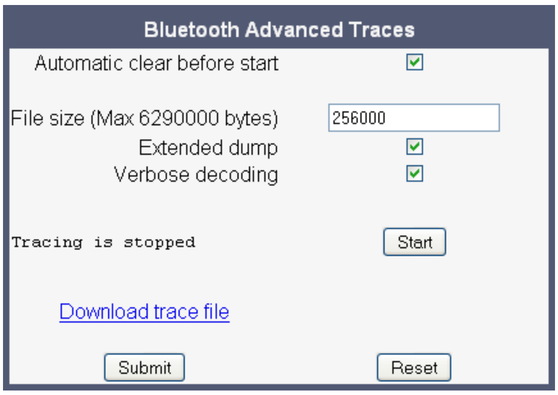
With the **Start**/**Stop** button, tracing is started or halted. The label depends on whether tracing is active or not.

On clicking the **Download trace file** link, the trace file is displayed.

With **Submit**, the changes on the parameters described above are sent to the phone.

With **Reset**, parameter changes that have been made in the form, but not yet sent to the phone, are cancelled.

Administration via WBM



3.24.9 QoS Reports

3.24.9.1 Conditions and Thresholds for Report Generation

> For details about the functionality, please refer to the release notes.

The generation of QoS (Quality of Service) reports which are sent to a QCU server (see Section 3.3.8, “SNMP”) is configured here.

Data required

- **Report mode**: Sets the conditions for generating a QoS report.
  Value range:
  - "OFF": No reports are generated.
  - "EOS Threshold exceeded": Default value. A report is created if a) a telephone conversation longer than the **Minimum session length** has just ended, and b) a threshold value has been exceeded during the conversation.
  - "EOR Threshold exceeded": A report is created if a) the report interval has just passed, and b) a threshold value has been exceeded during the observation interval.
  - "EOS (End of Session)": A report is created if a telephone conversation longer than the **Minimum session length** has just ended.
  - "EOR (End of Report Interval)": A report is created if the report interval has just passed.
- **Report interval (seconds)**: Time interval between the periodical observations.
  Default: 60
- **Observation interval (seconds)**: During this time interval, the traffic is observed.
  Value: 10
- **Minimum session length (100 millisecond units)**: When the Report mode is set to "EOS Threshold exceeded" or "EOS (End of Session)", a report can be created only if the duration of the conversation exceeds this value.
  Default: 20
- **Maximum jitter (milliseconds)**: When the jitter exceeds this value, a report is generated.
  Default: 20
- **Average round trip delay (milliseconds)**: When the average round trip time exceeds this value, a report is generated.
  Default: 100

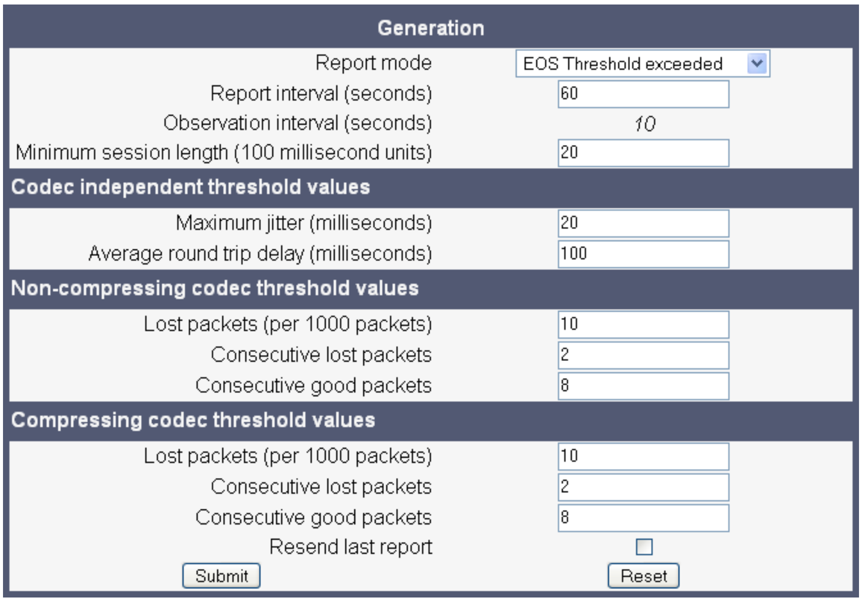
Non-compressing codecs / Compressing codes:

- **Lost packets (per 1000 packets)**: When the number of lost packets exceeds this maximum value during the observation interval, a report is created.
  Default: 10
- **Consecutive lost packets**: When the number of lost packets following one another exceeds this maximum value during the observation interval, a report is created.
  Default: 2
- **Consecutive good packets**: When the number of good packets following one another falls below this minimum value, a report is created.
  Default: 8
- **Resend last report**: If checked, the previous report is sent once again on pressing **Submit**.
  Value range: "Yes", "No"
  Default: "No"

The transmission of report data can be triggered manually by pressing **Send now** in the local menu.

Administration via WBM

Diagnostics > QoS Reports > Generation



Administration via Local Phone

```
|— Administration
    |— Network
        |— QoS
            |— Reports
                |— Generation
                |   |— Mode
                |   |— Report interval
                |   |— Observe interval
                |   |— Minimum session length
                |— Send now
                |— Thresholds
                    |— Maximum jitter
                    |— Round-trip delay
                    |— Non-compressing:
                    |— ...Lost packets (K)
                    |— ...Lost consecutive
                    |— ...Good consecutive
                    |— Compressing:
                    |— ...Lost packets (K)
                    |— ...Lost consecutive
                    |— ...Good consecutive
```

3.24.9.2 View Report

OpenStage phones generate QoS reports using a HiPath specific format, QDC (**Q**oS **D**ata **C**ollection). The reports created for the last 6 sessions, i. e. conversations, can be viewed on the WBM.

To enable the generation of reports, please ensure that:

- the switch **QoS traps to QCU** (System > SNMP) is activated (see Section 3.3.8, “SNMP”);
- the conditions for the generation of reports are set adequately (see Section 3.24.9.1, “Conditions and Thresholds for Report Generation”).

For details about QoS reports on HiPath devices, see the HiPath QoS Data Collection V 1.0 Service Manual.

A QoS report contains the following data:

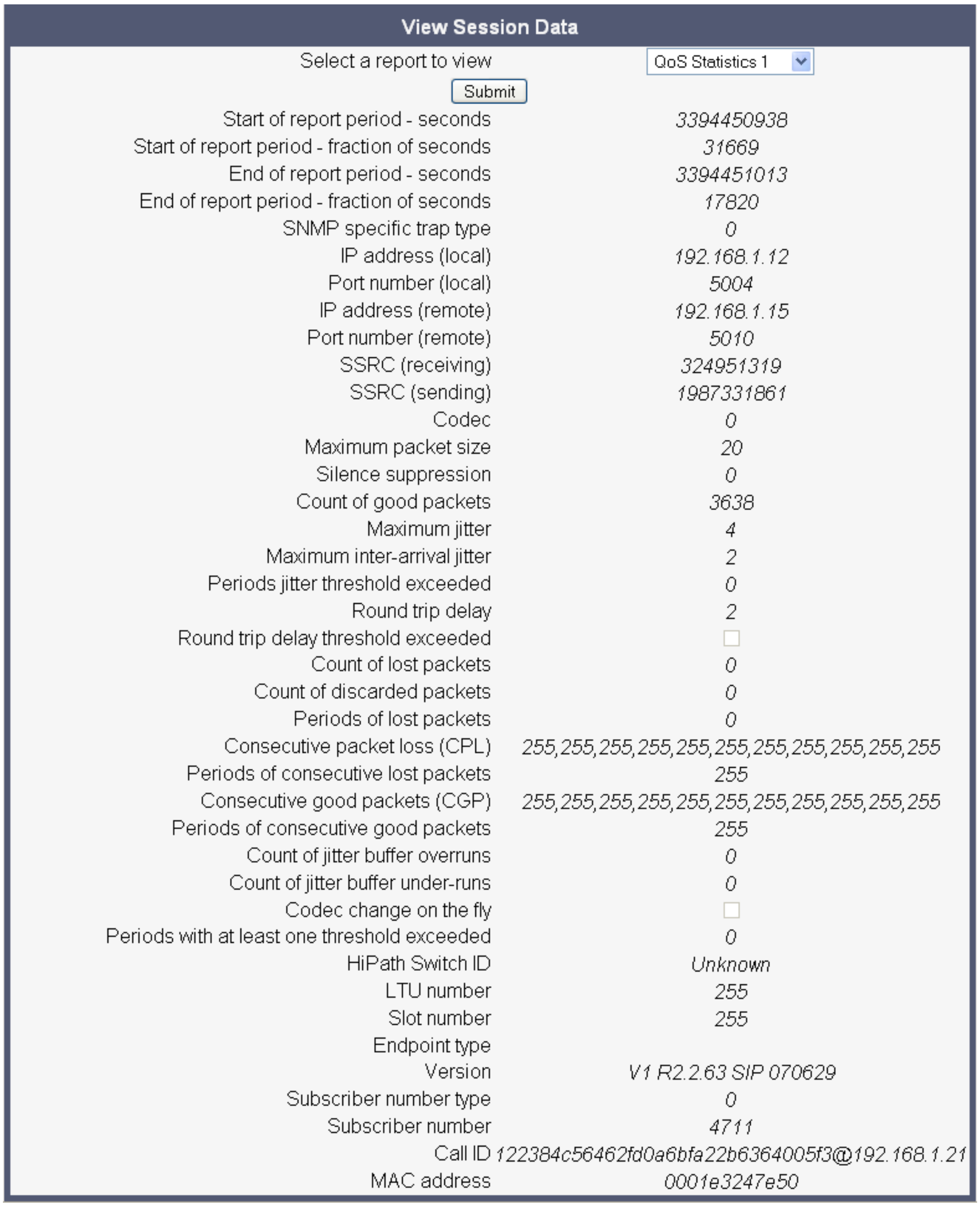
- **Start of report period - seconds**: NTP time in seconds for the start of the report period.
- **Start of report period - fraction of seconds**: Additional split seconds to be added to the seconds for an exact start time.
- **End of report period - seconds**: NTP time in seconds for the end of the report period.
- **End of report period - fraction of seconds**: Additional split seconds to be added to the seconds for an exact end time.
- **SNMP specific trap type**: The trap type is a 5 bit value calculated from a list of threshold-exceeding bits. Every time a threshold is exceeded, the associated bit is set, otherwise it is cleared.
  The trace type bits are defined as follows:
  - Bit 0: Jitter threshold was exceeded.
  - Bit 1: Delay threshold was exceeded.
  - Bit 2: Threshold for lost packets was exceeded.
  - Bit 3: Threshold for consecutive lost packets was exceeded.
  - Bit 4: Threshold for consecutive good packets was exceeded.
- **IP address (local)**: IP address of the local phone.
- **Port number (local)**: RTP receiving port of the local phone.
- **IP address (remote)**: IP address of the remote phone that took part in the session.
- **Port number (remote)**: RTP sending port of the local phone.
- **SSRC (receiving)**: RTP Source Synchronization Identifier of the local phone.
- **SSRC (sending)**: RTP Source Synchronization Identifier of the remote phone.
- **Codec**: Number of the Payload Type applied in the session; see RFC 3551 (Table 4 and 5).
- **Maximum packet size**: Maximum size (in ms) of packets received during the report interval.

- **Silence suppression**: Number of silence suppression activation objects found in the RTP stream received. A silence suppression activation object is defined as a period of silence when no encoded voice signals were transmitted by the sender.
- **Count of good packets**: Total amount of good packets.
- **Maximum jitter**: Maximum jitter (in ms) found during the report interval.
- **Maximum inter-arrival jitter**: Maximum of the interarrival jitter values (in ms). The interarrival jitter is the smoothed absolute value of the jitter measurements. It is calculated continuously. For details about the calulation, see RFC 3550.
- **Periods jitter threshold exceeded**: Number of observation intervals in which the threshold for maximum jitter was exceeded.
- **Round trip delay**: Average value of delay calculated for each RTCP packet. The first value is available after about 15 sec.
- **Round trip delay threshold exceeded**: Set to "true" if the average round trip delay threshold value was exceeded in the report interval.
- **Count of lost packets**: Number of packets lost in the course of speech decoding.
- **Count of discarded packets**: Number of the packets discarded without transferring the contents.
- **Periods of lost packets**: Number of observation intervals in which the threshold for lost packets was exceeded.
- **Consecutive packet loss (CPL)**: List of sequences consecutive packets that were all lost, grouped according to the amount of packets per sequence. The first number in the list counts single lost packets, the second number counts sequences of two lost packets, and so on. The last number counts sequences of more than 10 lost packets.
- **Periods of consecutive lost packets**: Number of observation intervals in which the threshold for consecutive lost packets was exceeded.
- **Consecutive good packets (CGP)**: List of sequences consecutive packets that were all processed, grouped according to the amount of packets per sequence. The first number in the list counts single good packets, the second number counts sequences of two good packets, and so on. The last number counts sequences of more than 10 good packets. All values are reset to 0 after an interval without packet loss.
- **Periods of consecutive good packets**: Number of intervals in which the count of lost packets went below the threshold.
- **Count of jitter buffer overruns**: Number of packets rejected because the jitter buffer was full.
- **Count of jitter buffer under-runs**: Increased by one whenever the decoder requests new information on decoding and finds an empty jitter buffer.
- **Codec change on the fly**: The value is 1, if there has been a codec or SSRC change during the observation period, and 0, if there has been no change.
- **Periods with at least one threshold exceeded**: Number of observation intervals with at least one threshold exceedance. If there is no data, the value is 255. The threshold values included are:

  - maximum jitter;
  - lost packets;
  - consecutive lost packets;
  - consecutive good packets.
- **HiPath Switch ID**: Unique number identifying the HiPath switch to which the endpoints are assigned.
- **LTU number**: In HiPath 4000 only, the shelf identification is taken from the shelf containing a gateway.
- **Slot number**: The slot number where the phoneis connected in the shelf.
- **Endpoint type**: Type of the local phone.
- **Version**: Software version of the local phone.
- **Subscriber number type**: Type of subscriber number assigned to the local phone. The possible types are:
  - 1: local number, extension only
  - 2: called number, network call
  - 3: E.164 number of the local phone
- **Subscriber number**: Subscriber number of the local phone.
- **Call ID**: SIP call id.
- **MAC address**: MAC address of the local phone.

Data viewing via WBM

Diagnostics > QoS reports > View Session Data



View Session Data	
Select a report to view	QoS Statistics 1
Submit	
Start of report period - seconds	3394450938
Start of report period - fraction of seconds	31669
End of report period - seconds	3394451013
End of report period - fraction of seconds	17820
SNMP specific trap type	0
IP address (local)	192.168.1.12
Port number (local)	5004
IP address (remote)	192.168.1.15
Port number (remote)	5010
SSRC (receiving)	324951319
SSRC (sending)	1987331861
Codec	0
Maximum packet size	20
Silence suppression	0
Count of good packets	3638
Maximum jitter	4
Maximum inter-arrival jitter	2
Periods jitter threshold exceeded	0
Round trip delay	2
Round trip delay threshold exceeded	
Count of lost packets	0
Count of discarded packets	0
Periods of lost packets	0
Consecutive packet loss (CPL)	255,255,255,255,255,255,255,255,255,255,255
Periods of consecutive lost packets	255
Consecutive good packets (CGP)	255,255,255,255,255,255,255,255,255,255,255
Periods of consecutive good packets	255
Count of jitter buffer overruns	0
Count of jitter buffer under-runs	0
Codec change on the fly	
Periods with at least one threshold exceeded	0
HiPath Switch ID	Unknown
LTU number	255
Slot number	255
Endpoint type	
Version	V1 R2.2.63 SIP 070629
Subscriber number type	0
Subscriber number	4711
Call ID	122384c56462fd0a6bfa22b6364005f3@192.168.1.21
MAC address	0001e3247e50

3.24.10 Core dump

If **Enable core dump** is checked, a core dump will be initiated in case of a severe error. The core dump will be saved to a file. By default, this function is activated.

When **File size unlimited** is checked, there is no size limit for the core dump file. By default, it is not checked.

The maximum size for core dump files in MBytes can be chosen in the **Limited file size (MBs)** field. The possible values are 1, 5, 10, 25, 50, 75, and 100. The default value is 100.

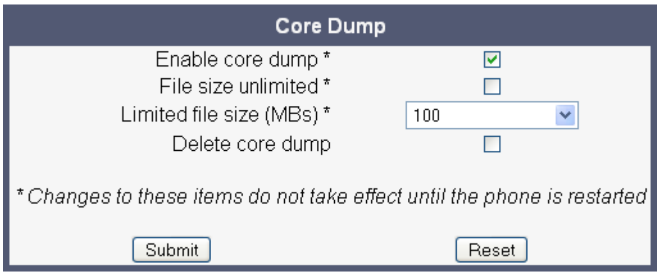
> With firmware V2R1, unlimited file size is preset, and the parameters **File size unlimited** as well as **Limited file size (MBs)** are not available.

If **Delete core dump** is activated, the current core dump file is deleted on **Submit**. By default, this is not activated.

If one or more core dump file exist, hyperlinks for downloading will be created automatically.

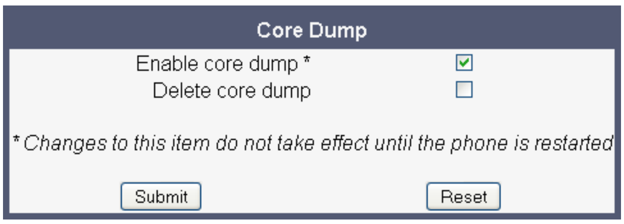
Administration via WBM (up to V2R0)

Diagnostics > Miscellaneous > Core dump



Administration via WBM (V2R1)

Diagnostics > Miscellaneous > Core dump



3.24.11 Remote Tracing - Syslog

All trace messages created by the components of the phone software can be sent to a remote server using the syslog protocol. This is helpful especially for long-term observations with a greater number of phones.

To enable remote tracing, **Remote trace status** must be set to "Enabled". Furthermore, the IP address of the server receiving the syslog messages must be entered in **Remote ip**, and the corresponding server port must be given in **Remote port**.

With version V2, the **User notification** parameter controls whether the user is notified about the remote tracing or not. If user notification is enabled, a blinking symbol (on OpenStage 60/80; on OpenStage 15/20/40) will inform the user when remote tracing is active, that is, when **Remote trace status** is set to "Enabled".

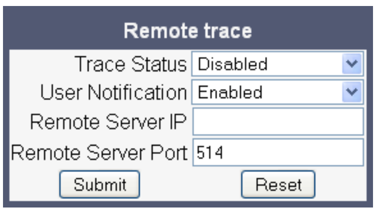
Administration via Local Phone

|--- Administration
    |--- Maintenance
        |--- Remote trace
            |--- **Remote trace status**
            |--- **Remote ip**
            |--- **Remote port**

Administration via Local Phone (V2)

|--- Administration
    |--- Maintenance
        |--- Remote trace
            |--- **Remote trace status**
            |--- **User notification**
            |--- **Remote ip**
            |--- **Remote port**

Administration via WBM (V2)



3.24.12 HPT Interface (For Service Staff)

For special diagnosis and maintenance tasks, the service staff may employ the HPT tool, which is able to control and observe an OpenStage phone remotely. For security reasons, this tool can only be used when a dongle key file is uploaded to the phone (see Section 3.14.10, “Dongle Key”). This key is accessable to the service staff only. It is specific for a particular SIP firmware version, but it will also be valid for previous versions.

There are 2 types of HPT sessions, control session and observation session.

A control session allows for activating phone functions remotely. When a control session is established, the following changes will occur:

- The display shows a message indicating that remote service is active.
- Handset, microphone, speaker, headset, and microphone are disabled.

An observation session allows for supervising events on the phone, like, for instance, pressing a key, incoming calls or navigating in the menus. Before an observation session is started, the user is prompted for allowing the observation. During an observation session, the phone operates normally, including loudspeaker, microphone and ringer. Thus, the local user can demonstrate an error towards the service staff that is connected via HPT.

The HPT interface is enabled by downloading the dongle key file to the phone (see Section 3.14.10, “Dongle Key”). It can be disabled via local menu or WBM. Thereby, the dongle key file is deleted. To enable the HPT interface again, the file must be downloaded anew.

The session data is written to a log file on the phone. It can be downloaded from the Diagnostics > Fault trace configuration menu (see Section 3.24.6, “Fault Trace Configuration”).

Administration via WBM (Disable)

Maintenance > HPT interface

3.25 Bluetooth

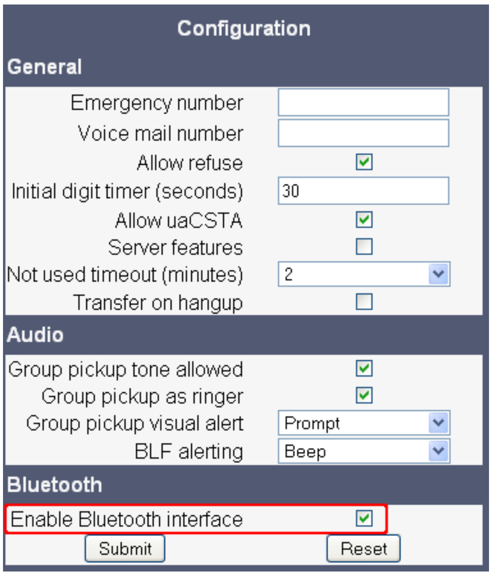
The Bluetooth interface can be enabled or disabled in the admin menu. By default, it is enabled. If Bluetooth is enabled, the user has the possibility to activate or deactivate it.

Additionally, the Bluetooth address is displayed.

> Bluetooth is available only on OpenStage 60/80 phones.

Administration via WBM

System > Features > Configuration



Administration via Local Phone

Bluetooth can be enabled or disabled, and the device address can be viewed via the local admin menu:

|— Administration
  |— System
    |— Features
      |— Configuration
        |— Bluetooth
          |— **Local device address**
          |— **Enable**

4 Technical Reference

4.1 Menus

This section describes the structure of the administration menus of the OpenStage phone. For information on user menus, please refer to the user manual.

4.1.1 Web Interface Menu

4.1.1.1 Menu Structure

Admin Login

Applications (OpenStage 60/80)
- **XML applications**
  - Add application
  - Modify application (up to V2R0) / Modify/Delete application (V2R1)
  - Xpressions
  - Add messages application (V2R1)
  - XML Phonebook (up to V2R0) / Add phonebook application (V2R1)
  - Add call log application (V2R1)
  - Add help application (V2R1)

Bluetooth

Network
- IP configuration (up to V2R0) / IP configuration (V2R1)
- Update Service (DLS)
- QoS
- Port configuration
- LLDP-MED operation

System
- System Identity / System Identity (V2)
- SIP interface
- Registration
- SNMP
- **Features**
  - Configuration (V1R5) / Configuration (V2) / Configuration (V2R2)
  - DSS settings / DSS settings (V2)

Program keys > Line (V1R5 on OpenStage 15/40/60/80)
Program keys > Line (V2 on OpenStage 15/40/60/80)
Key Module 1
Key Module 2
Fixed keys (V2R0) / Fixed keys (V2R1)
Keyset operation (V1R5) / Keyset operation (V2)
Services

Security (up to V2R2)

File transfer
Defaults
Phone application
Hold music
Picture Clip (OpenStage 60/80)
LDAP (OpenStage 60/80)
Logo (OpenStage 40/60/80)
Screensaver (OpenStage 60/80)
Ringer file
Dongle key

Local functions
Directory settings (OpenStage 60/80, OpenStage 40 V2R1)
Messages settings (V2)
Locality
Canonical dial settings
Canonical dial lookup
Canonical dial
Energy saving

Date and time

Speech
Codec preferences
Audio settings

General information

Authentication
Change Admin password
Change User password

Ringer setting (V2)

Mobility

Diagnostics

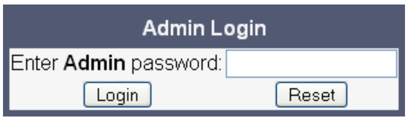
- LLDP-MED TLVs
- Fault trace configuration / Fault trace configuration (V1R5) / Fault trace configuration (V2) Fault trace configuration (V2R2)
- **EasyTrace Profiles**
  - Bluetooth handsfree profile (OpenStage 60/80)
  - Bluetooth headset profile (OpenStage 60/80)
  - Call connection
  - Call log problems
  - DAS connection
  - DLS data errors
  - Help application problems (OpenStage 60/80)
  - Key input problems
  - LAN connectivity problems
  - Messaging application problems
  - Mobility problems
  - Phone administration problems
  - Phonebook (LDAP) problems (OpenStage 60/80)
  - Phonebook (local) problems (OpenStage 60/80)
  - Server based application problems (OpenStage 60/80)
  - Sidecar problems
  - Speech problems
  - Tone problems
  - USB backup/restore
  - Voice recognition problems (OpenStage 60/80)
  - Web based management (V1R5) / Web based management (V2)
  - 802.1x problems
  - Clear all profiles
- Bluetooth Advanced Traces (V2)
- **QoS Reports**
  - Generation
  - View Session Data
- **Miscellaneous**
  - IP tests
  - Memory information (V1R5) / Memory information (V2)
  - Core dump / Core dump (V2R1)

Maintenance

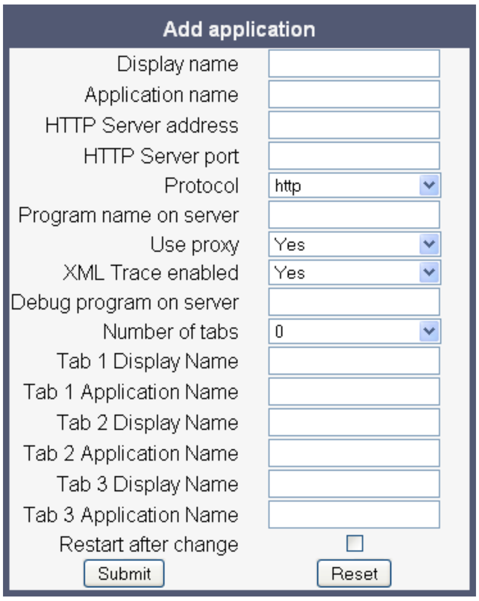
Remote trace (V1R5) / Remote trace (V2)
Restart phone
Factory reset
HPT interface
Secure shell (V2)

4.1.1.2 Web Pages

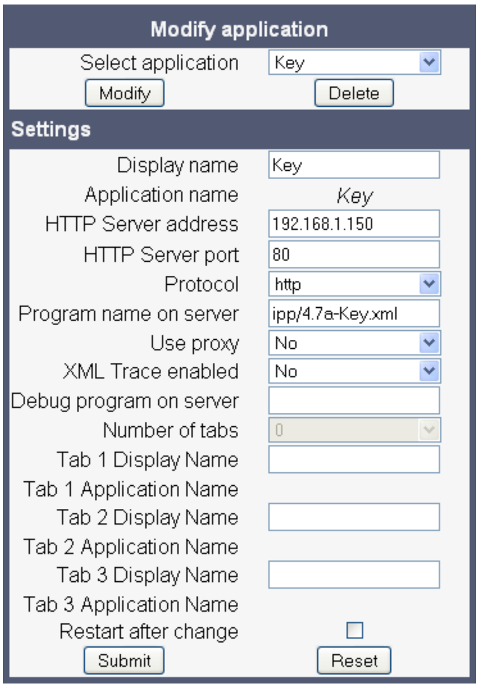
Admin Login



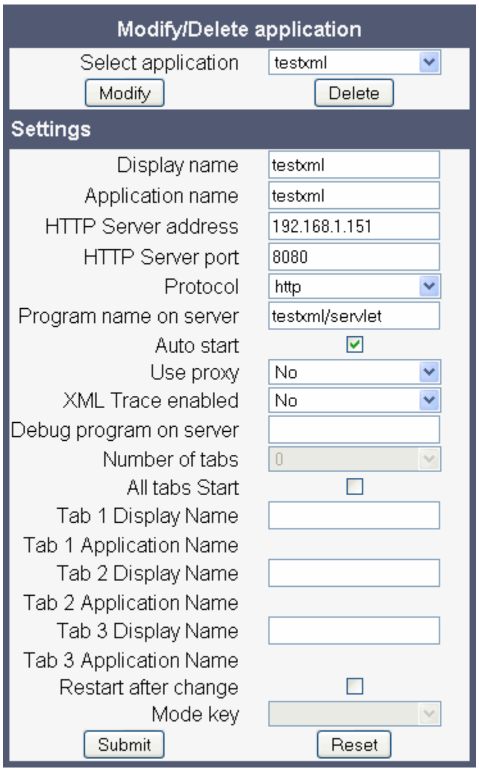
Add application



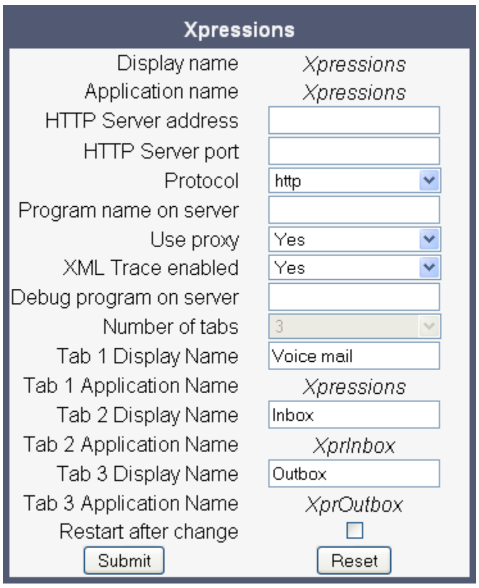
Modify application (up to V2R0)



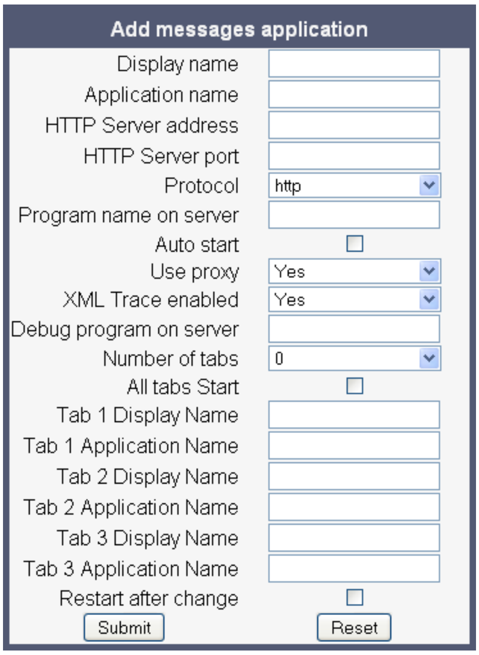
Modify/Delete application (V2R1)



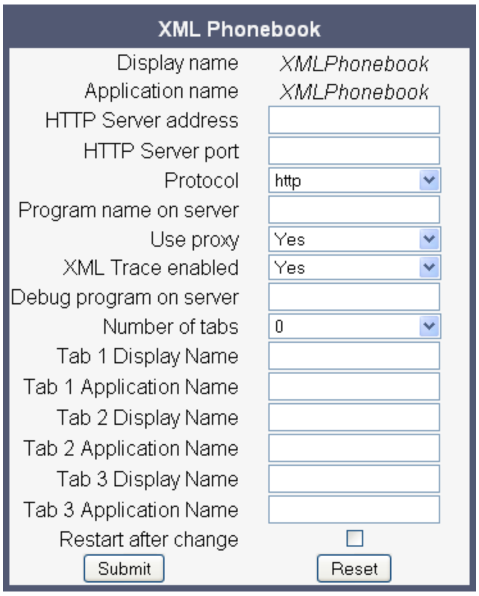
Xpressions



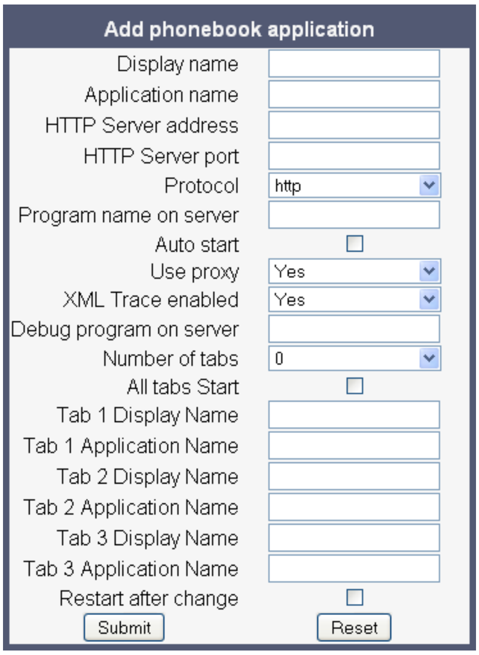
Add messages application (V2R1)



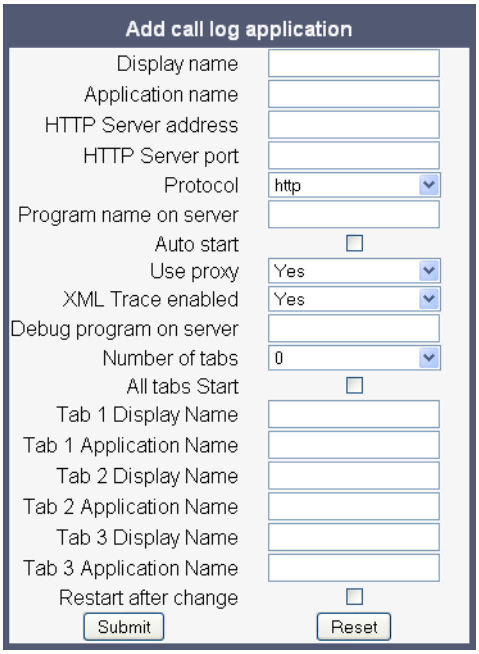
XML Phonebook (up to V2R0)



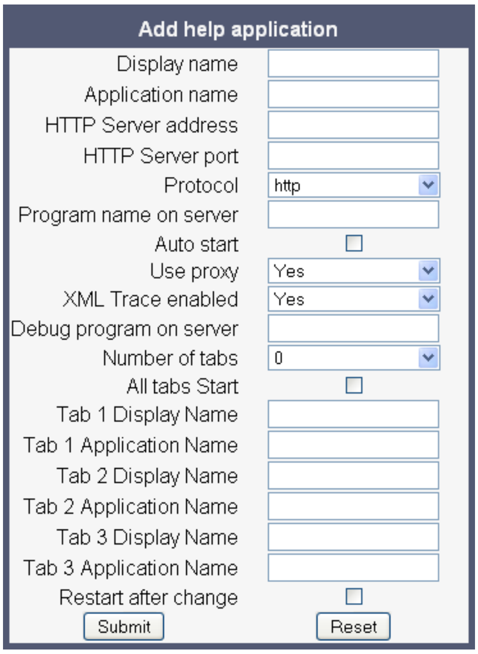
Add phonebook application (V2R1)



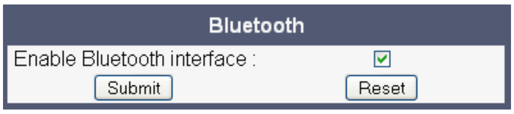
Add call log application (V2R1)



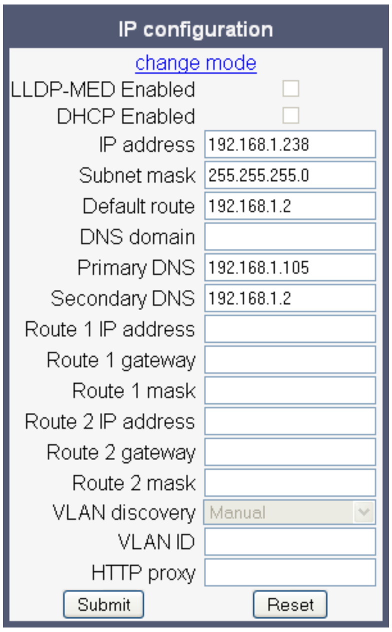
Add help application (V2R1)



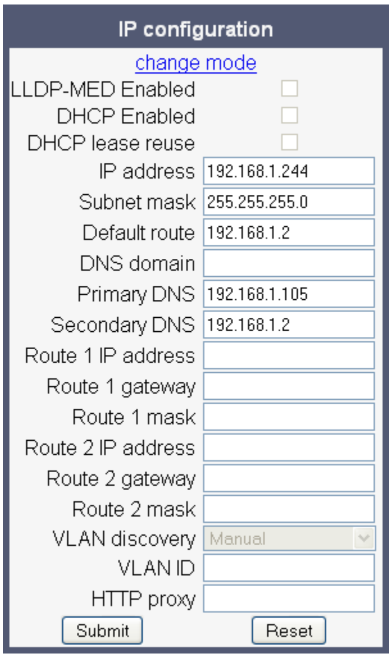
Bluetooth



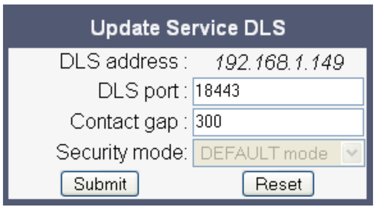
IP configuration (up to V2R0)



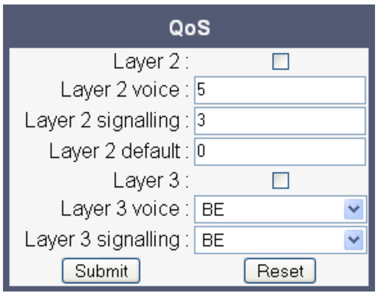
IP configuration (V2R1)



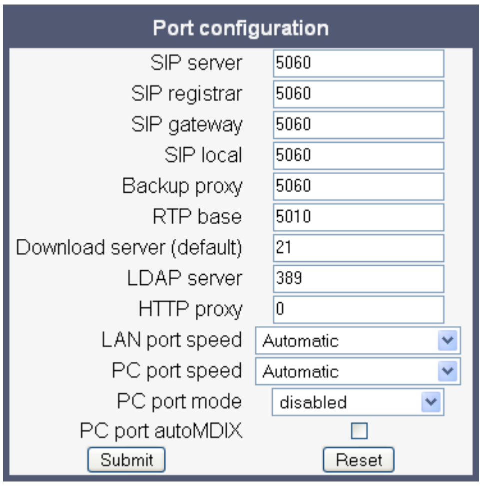
Update Service (DLS)



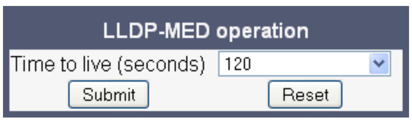
QoS



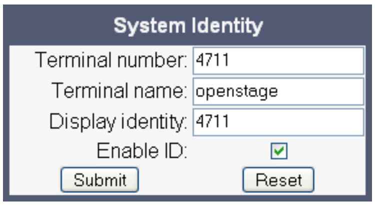
Port configuration



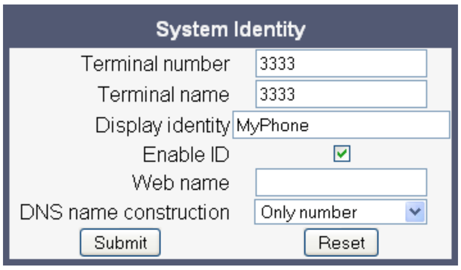
LLDP-MED operation



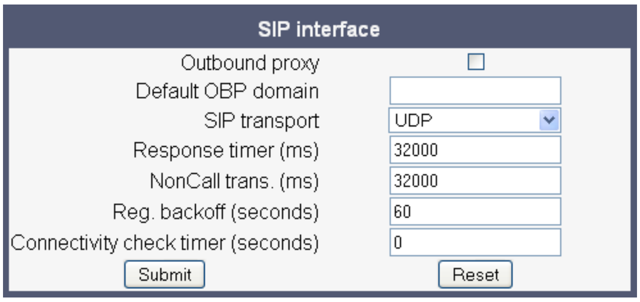
System Identity



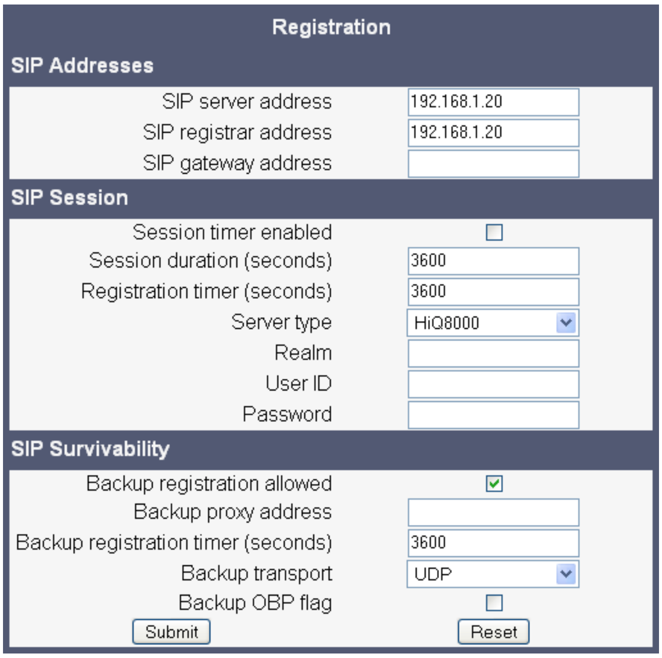
System Identity (V2)



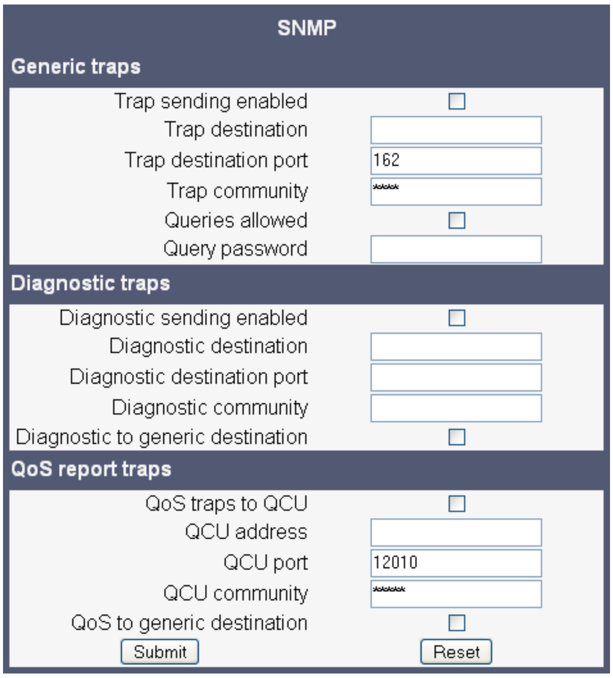
SIP interface



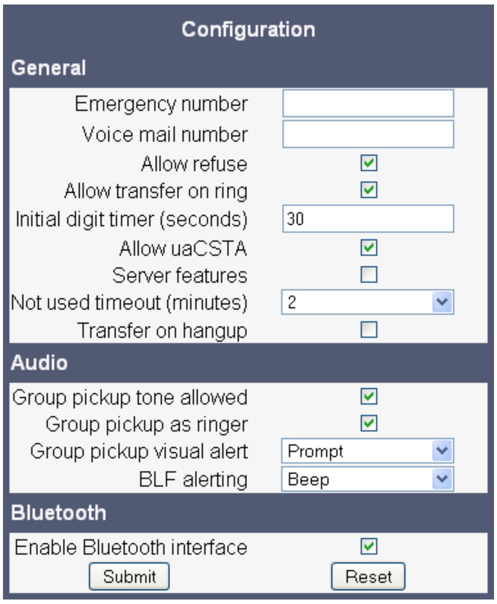
Registration



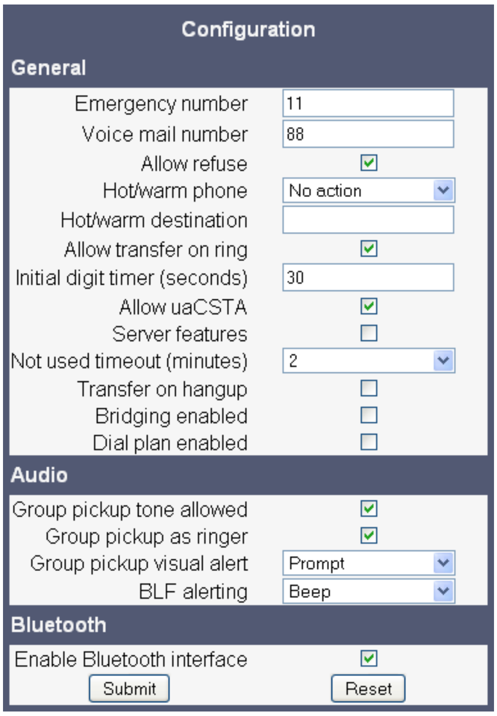
SNMP



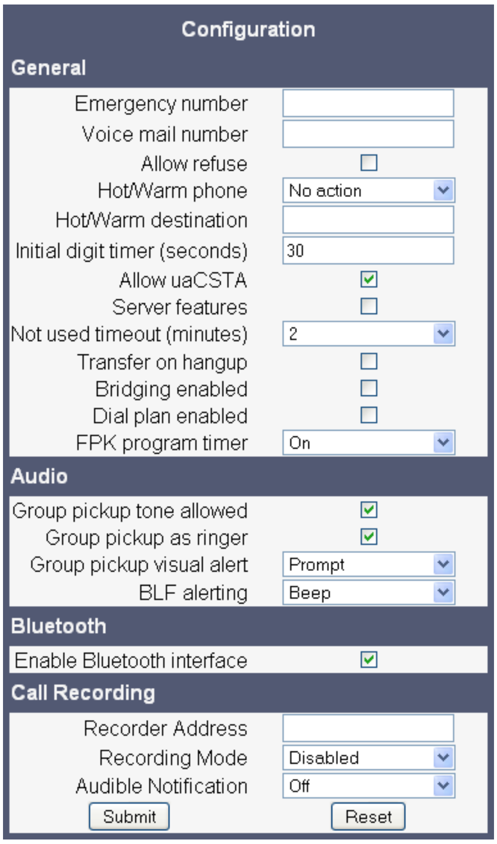
Configuration (V1R5)



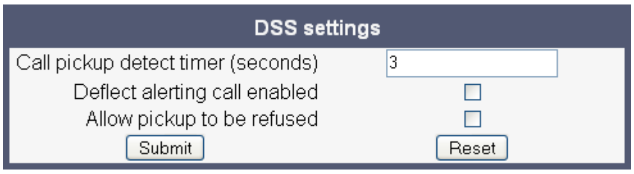
Configuration (V2)



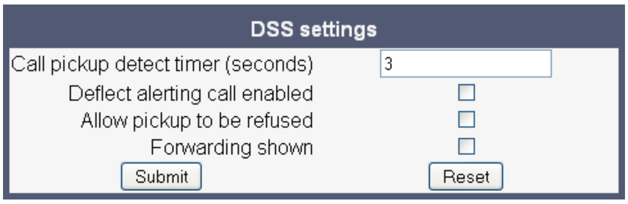
Configuration (V2R2)



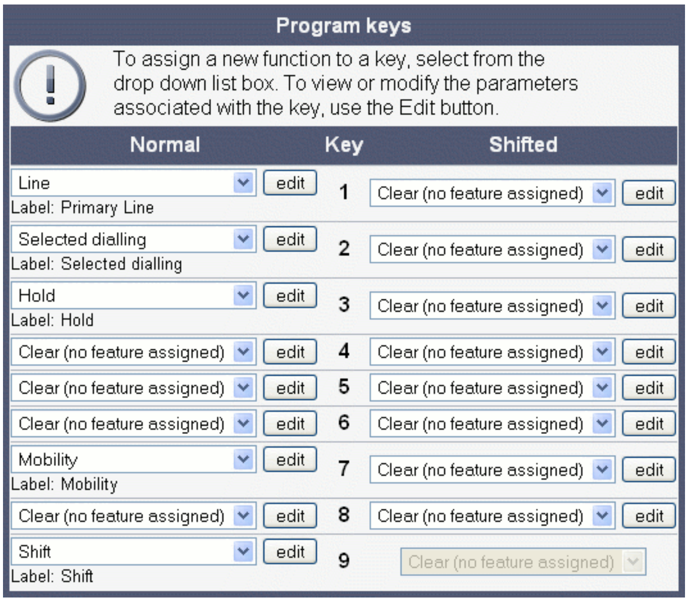
DSS settings



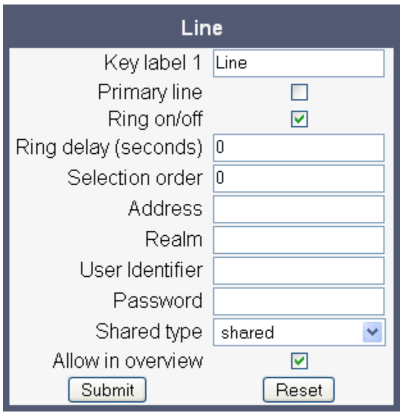
DSS settings (V2)



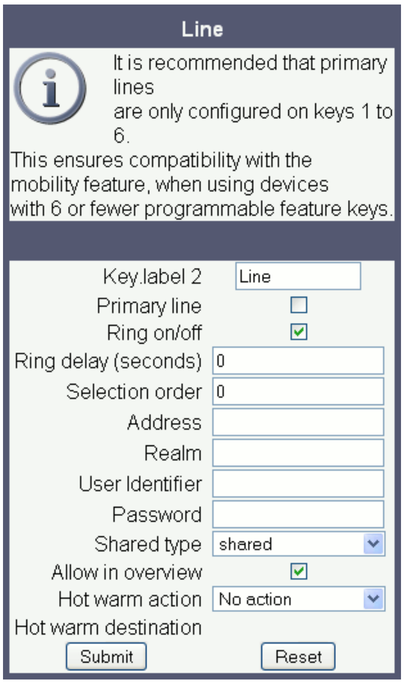
Program keys



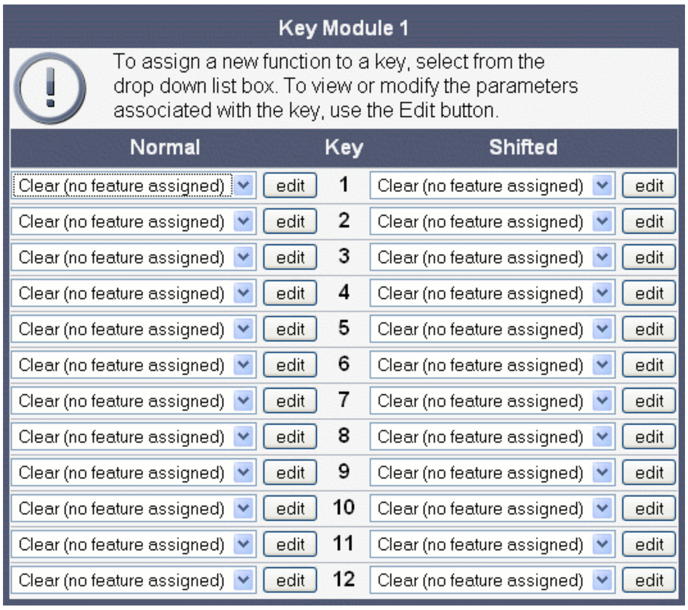
Line (V1R5 on OpenStage 15/40/60/80)



Line (V2 on OpenStage 15/40/60/80)



Key Module 1



Key Module 2

Key Module 2				
To assign a new function to a key, select from the drop down list box. To view or modify the parameters associated with the key, use the Edit button.				
Normal		**Key**	**Shifted**	
Clear (no feature assigned)	edit	1	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	2	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	3	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	4	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	5	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	6	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	7	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	8	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	9	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	10	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	11	Clear (no feature assigned)	edit
Clear (no feature assigned)	edit	12	Clear (no feature assigned)	edit

Fixed keys (V2R0)

Fixed Keys

To assign a new function to a key, select from the drop down list box. To view or modify the parameters associated with the key, use the Edit button.

Function		Key
Server feature	edit	**Forwarding**

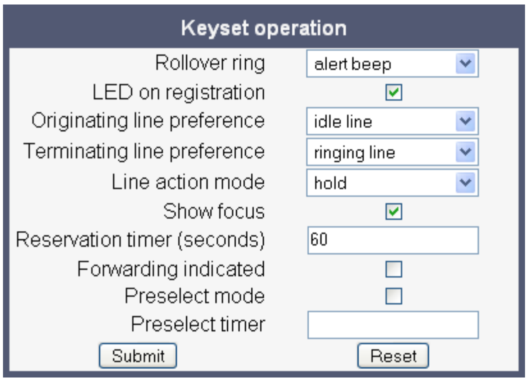
Fixed keys (V2R1)

Fixed Keys

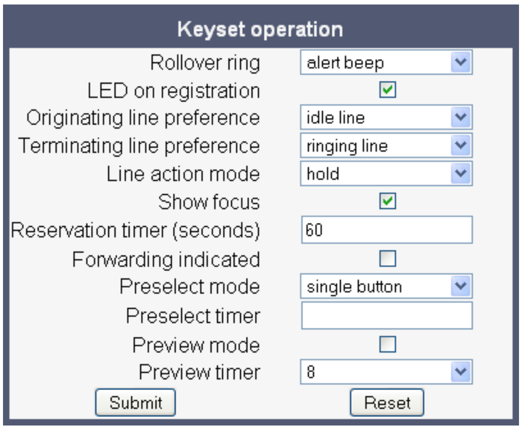
To assign a new function to a key, select from the drop down list box. To view or modify the parameters associated with the key, use the Edit button.

Forwarding key	Built-in forwarding	Edit
Release key	Built-in release	Edit
Voice recognition key	Built-in voice recognition	Edit

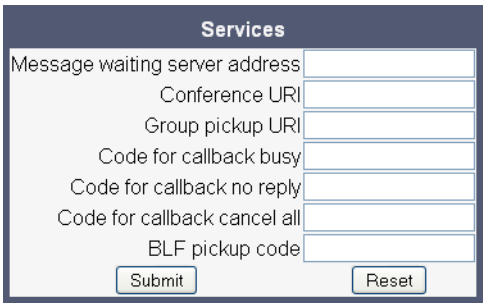
Keyset operation (V1R5)



Keyset operation (V2)

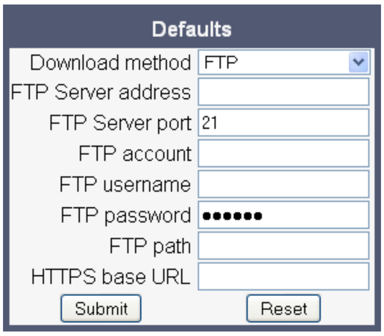


Services

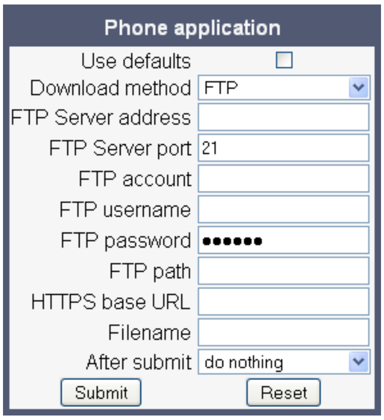


Security

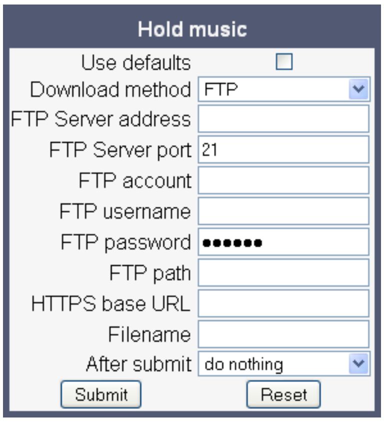
Defaults



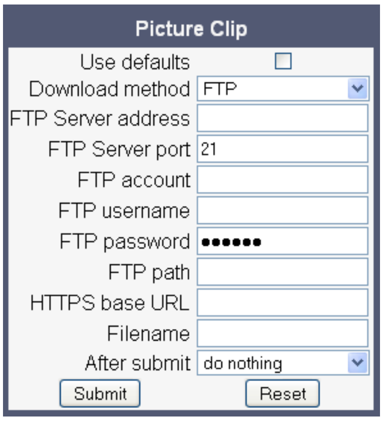
Phone application



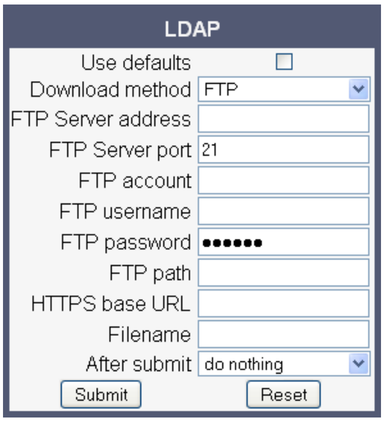
Hold music



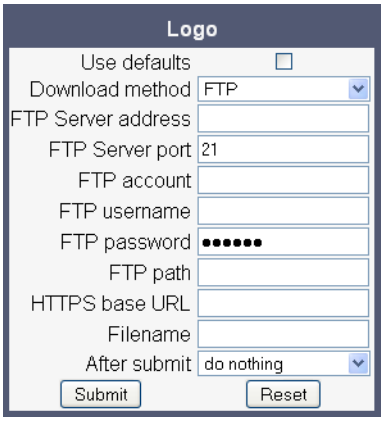
Picture Clip



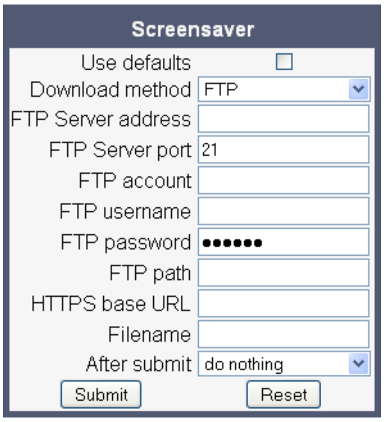
LDAP



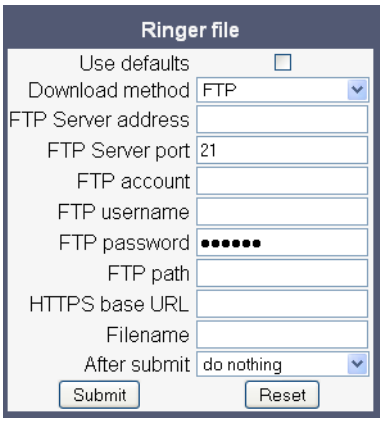
Logo



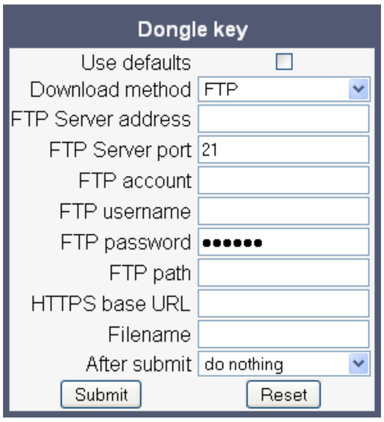
Screensaver



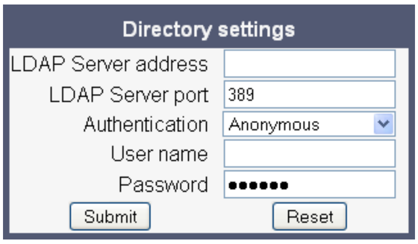
Ringer file



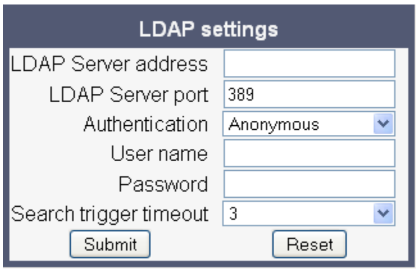
Dongle key



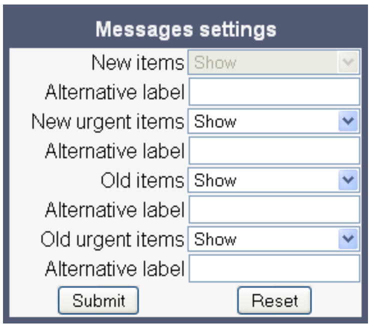
Directory settings (OpenStage 60/80, OpenStage 40 V2R1)



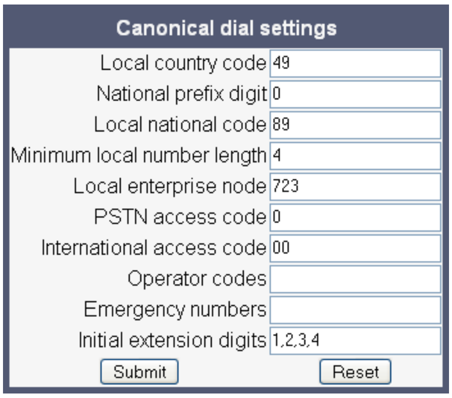
LDAP settings (V2)



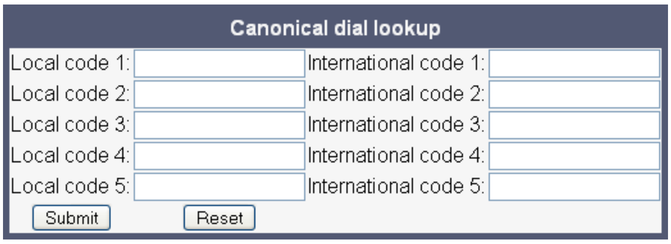
Messages settings (V2)



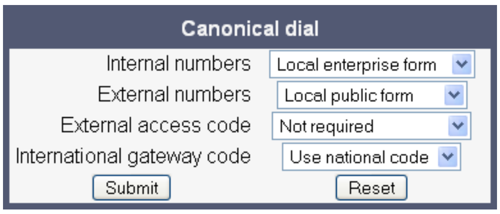
Canonical dial settings



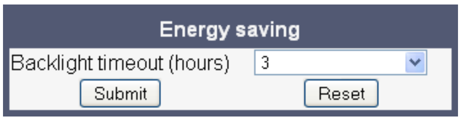
Canonical dial lookup



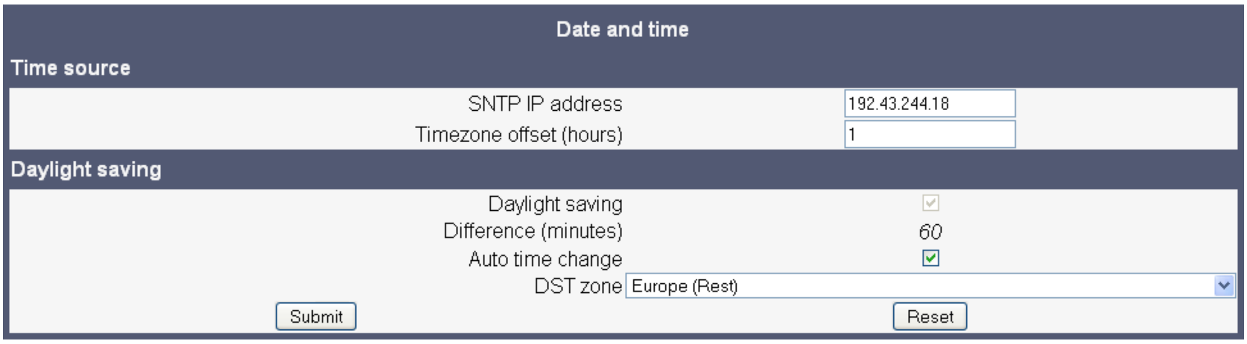
Canonical dial



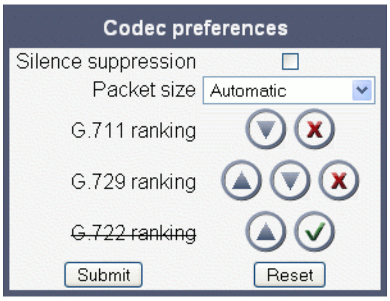
Energy saving



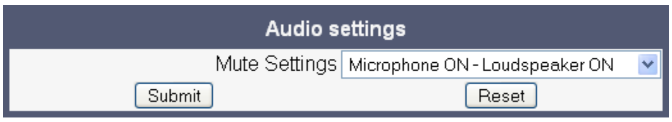
Date and time



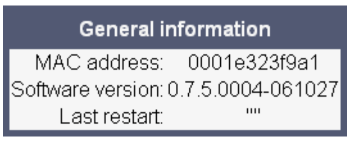
Codec preferences



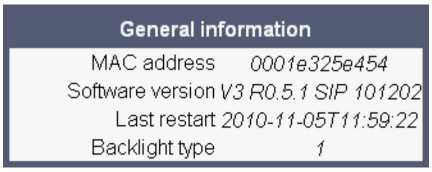
Audio settings



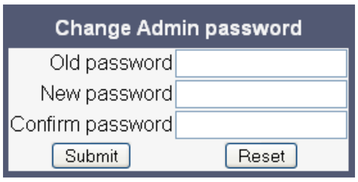
General information



General information (V2R2)



Change Admin password



Change User password

Ringer setting (V2)

Ringer setting

This page allows you to set up interworking with other IP phone systems that support distinctive ringing

Name	Ringer sound	Pattern melody	Pattern sequence	Duration (sec)	Audible
Bellcore-dr1	Pattern	8	1	0	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring
	Ringer2.mp3	3	2	60	Ring

Submit Reset

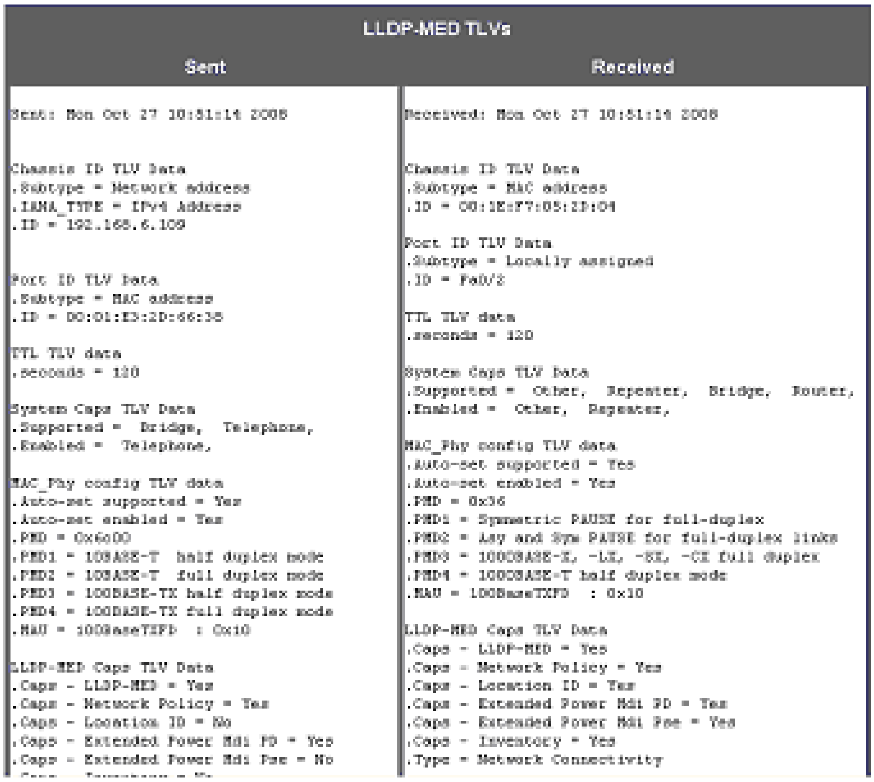
Mobility

Mobility

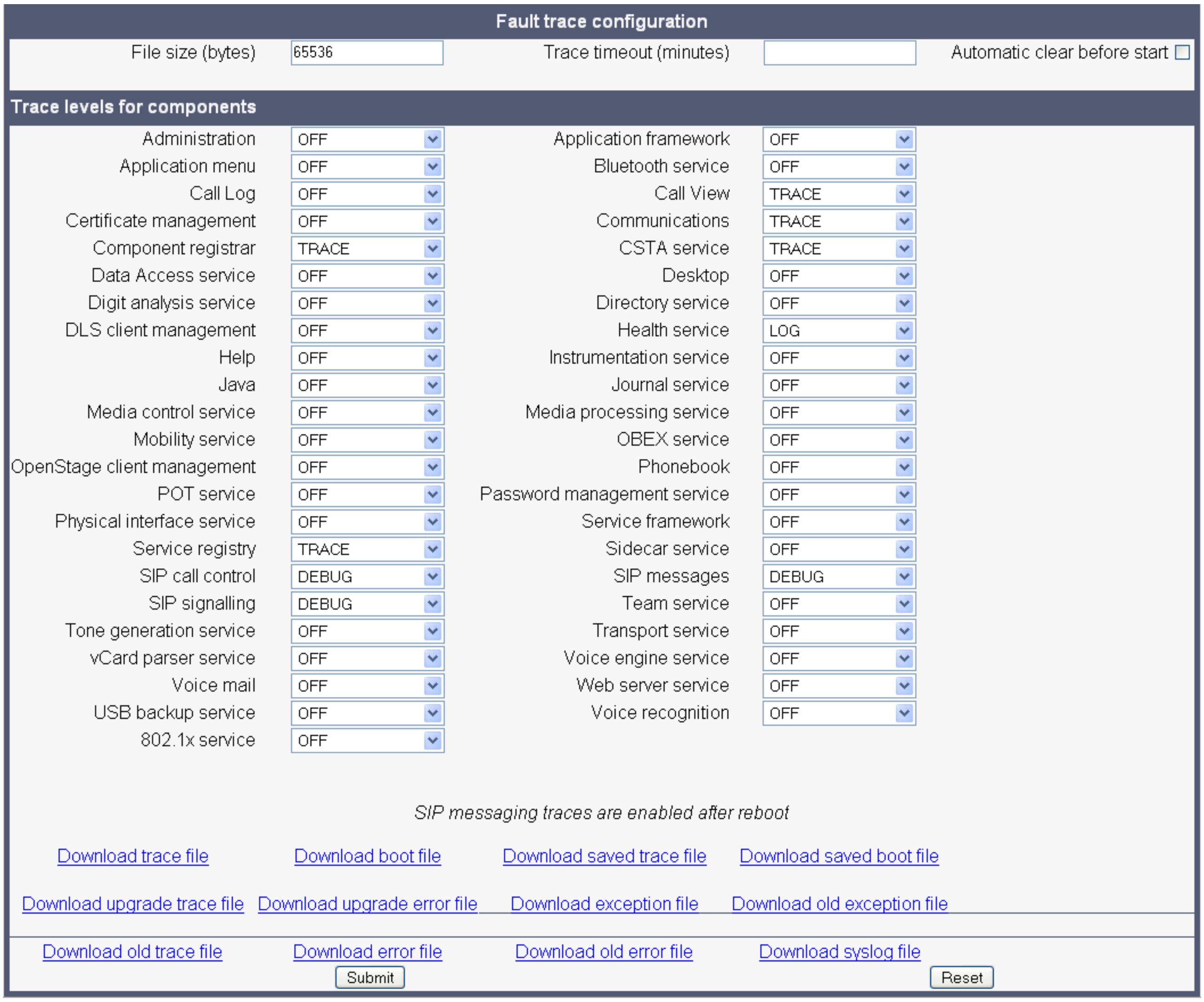
Unauthorised Logoff Trap	☐
Logoff Trap Delay	300
Timer Medium Priority	60
Mobility Feature	☑
Managed Profile	☐
Error Count Local	0
Error Count Remote	0

Submit Reset

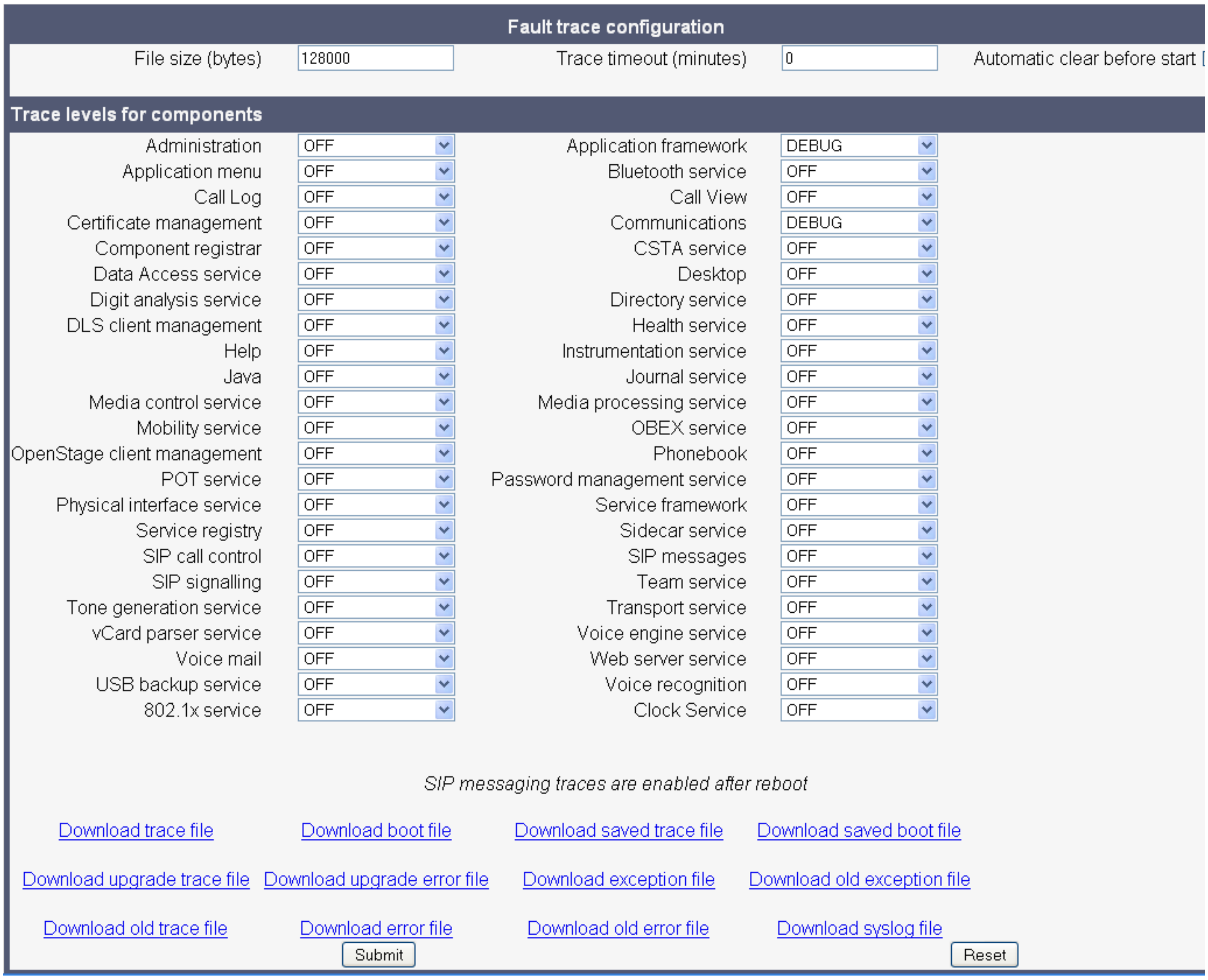
LLDP-MED TLVs



Fault trace configuration



Fault trace configuration (V1R5)



Fault trace configuration

File size (bytes)	128000	Trace timeout (minutes)	0	Automatic clear before start	

Trace levels for components

Administration	OFF	Application framework	DEBUG
Application menu	OFF	Bluetooth service	OFF
Call Log	OFF	Call View	OFF
Certificate management	OFF	Communications	DEBUG
Component registrar	OFF	CSTA service	OFF
Data Access service	OFF	Desktop	OFF
Digit analysis service	OFF	Directory service	OFF
DLS client management	OFF	Health service	OFF
Help	OFF	Instrumentation service	OFF
Java	OFF	Journal service	OFF
Media control service	OFF	Media processing service	OFF
Mobility service	OFF	OBEX service	OFF
OpenStage client management	OFF	Phonebook	OFF
POT service	OFF	Password management service	OFF
Physical interface service	OFF	Service framework	OFF
Service registry	OFF	Sidecar service	OFF
SIP call control	OFF	SIP messages	OFF
SIP signalling	OFF	Team service	OFF
Tone generation service	OFF	Transport service	OFF
vCard parser service	OFF	Voice engine service	OFF
Voice mail	OFF	Web server service	OFF
USB backup service	OFF	Voice recognition	OFF
802.1x service	OFF	Clock Service	OFF

SIP messaging traces are enabled after reboot

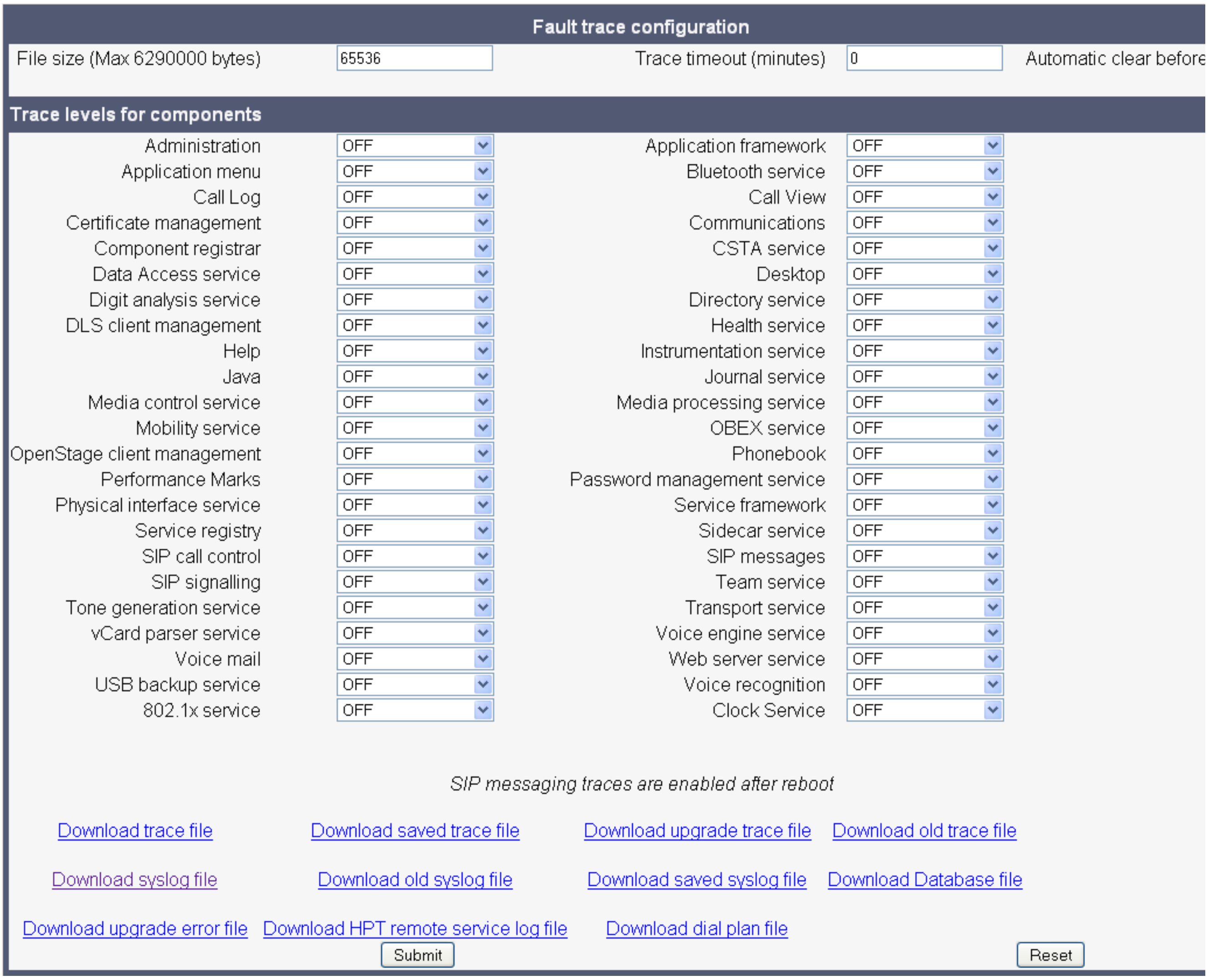
Download trace file | Download boot file | Download saved trace file | Download saved boot file

Download upgrade trace file | Download upgrade error file | Download exception file | Download old exception file

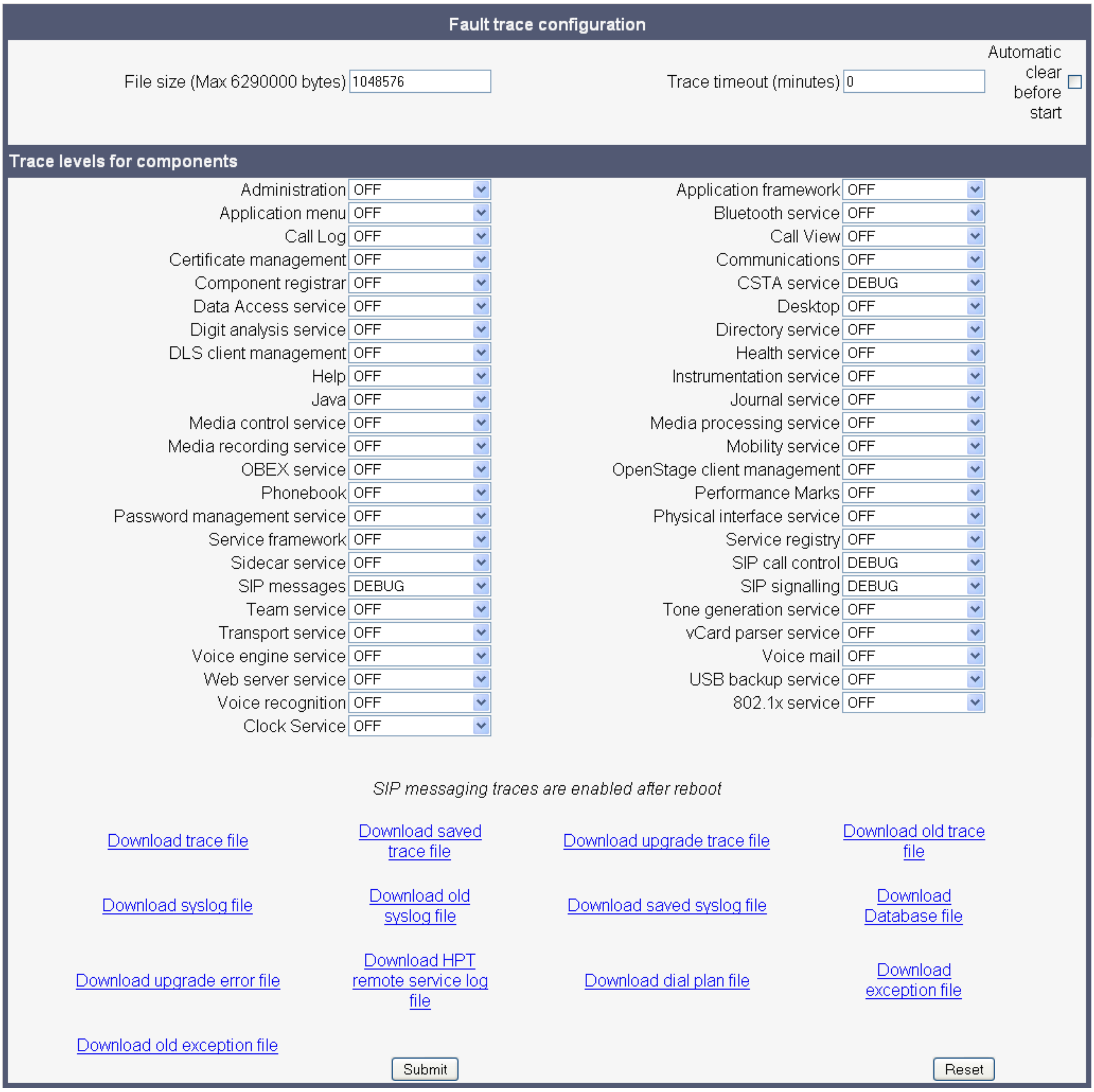
Download old trace file | Download error file | Download old error file | Download syslog file

Submit | Reset

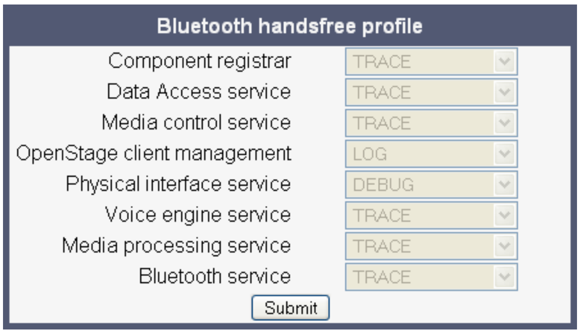
Fault trace configuration (V2)



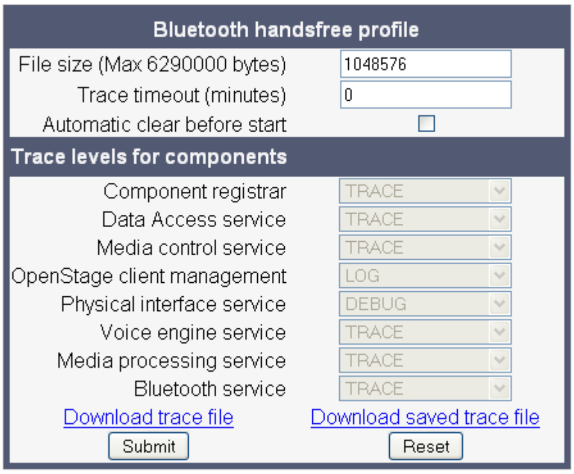
Fault trace configuration (V2R2)



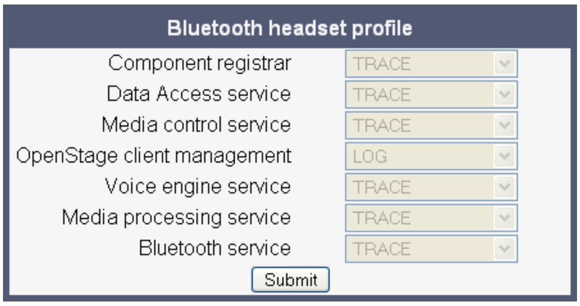
Bluetooth handsfree profile



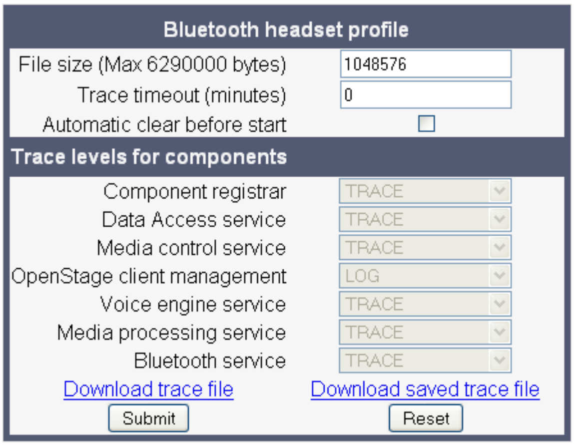
Bluetooth handsfree profile (V2R2)



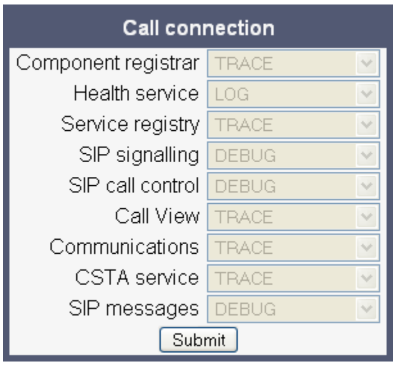
Bluetooth headset profile



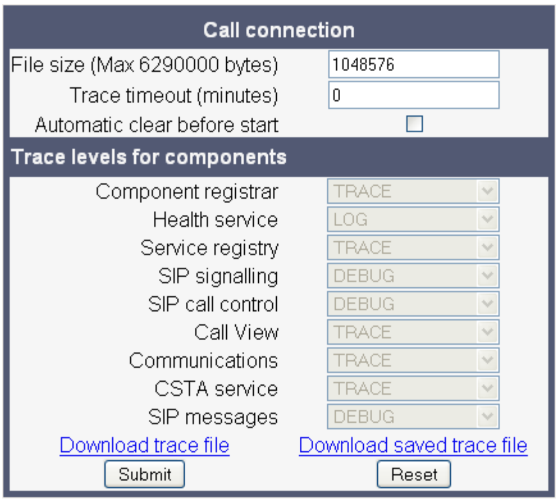
Bluetooth headset profile (V2R2)



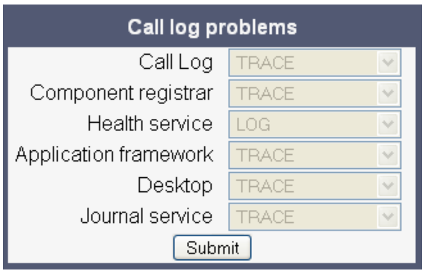
Call connection



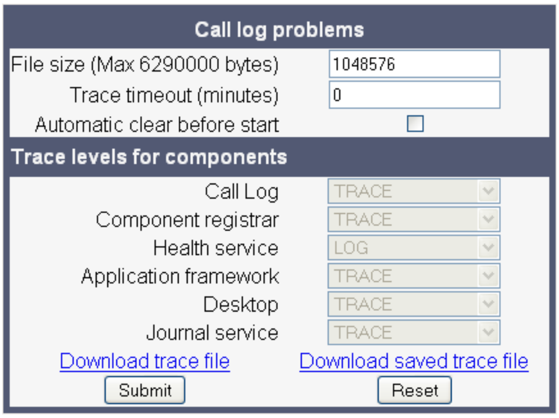
Call connection (V2R2)



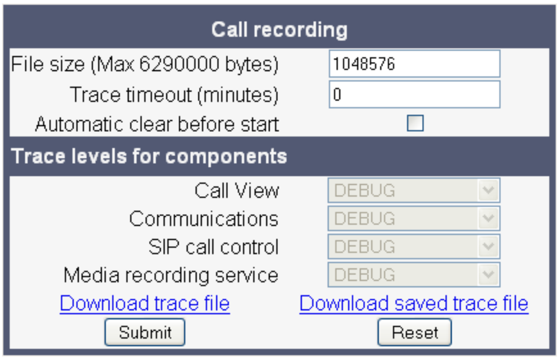
Call log problems



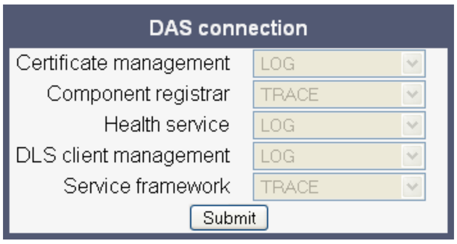
Call log problems (V2R2)



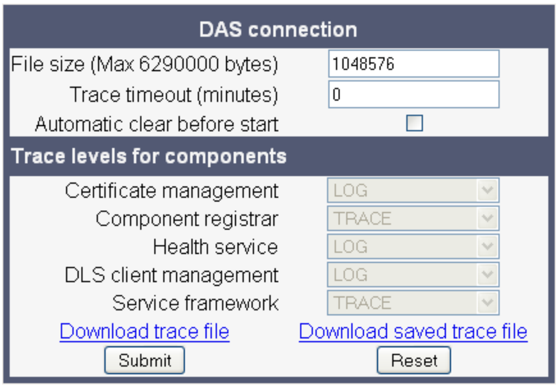
Call Recording (V2R2)



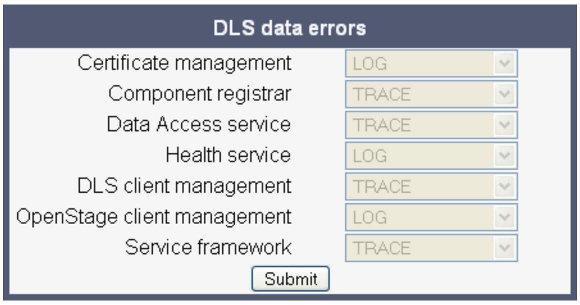
DAS connection



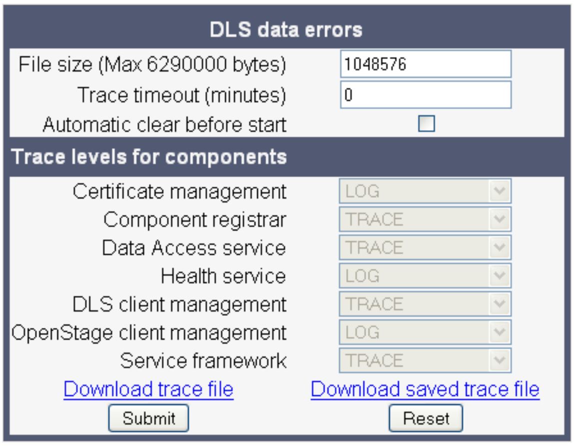
DAS connection (V2R2)



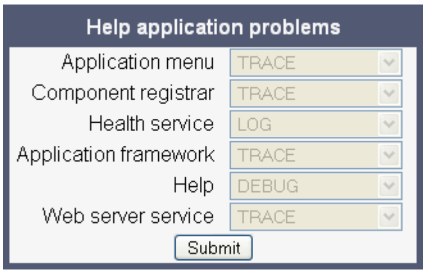
DLS data errors



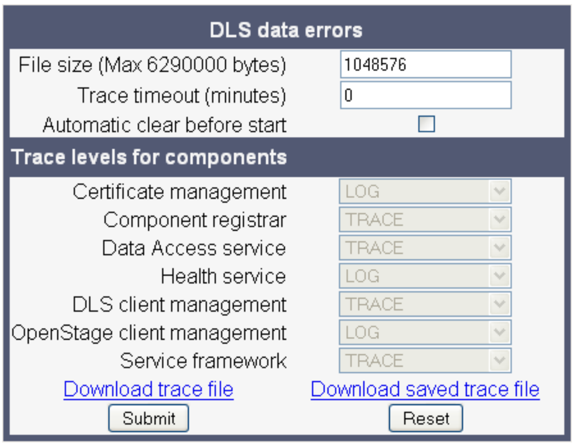
DLS data errors (V2R2)



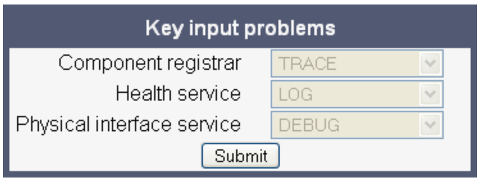
Help application problems



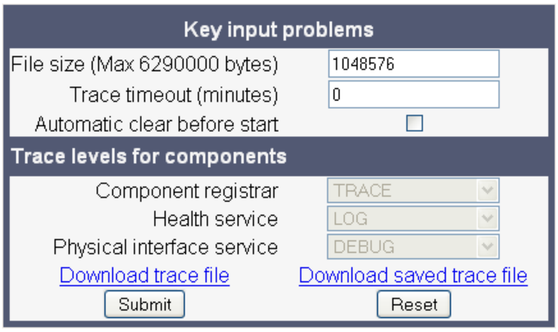
Help application problems (V2R2)



Key input problems

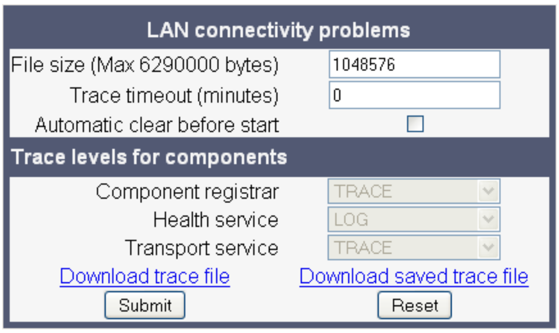


Key input problems (V2R2)

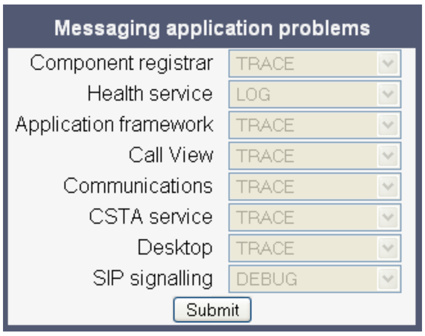


LAN connectivity problems

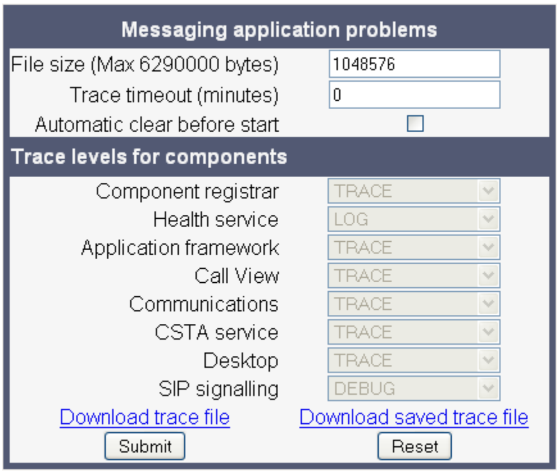
LAN connectivity problems (V2R2)



Messaging application problems

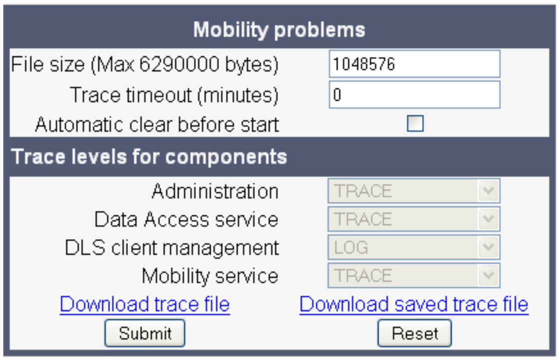


Messaging application problems (V2R2)



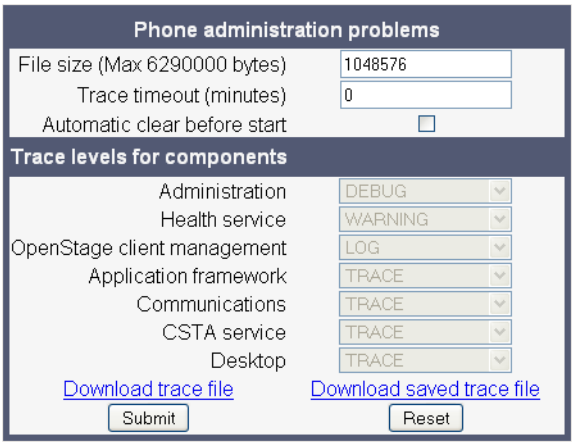
Mobility problems

Mobility problems (V2R2)

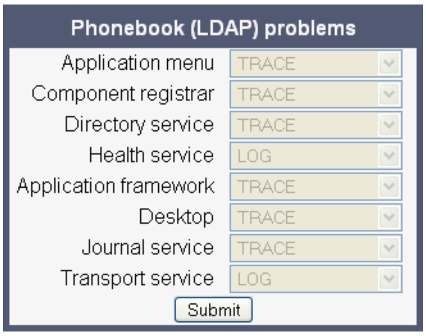


Phone administration problems

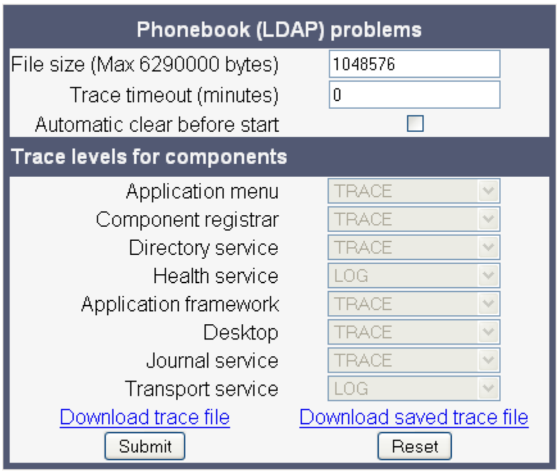
Phone administration problems (V2R2)



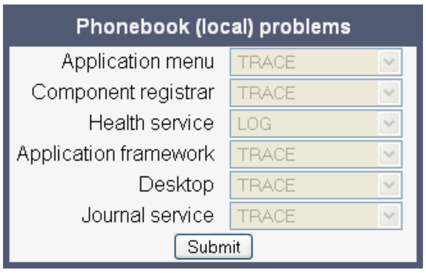
Phonebook (LDAP) problems



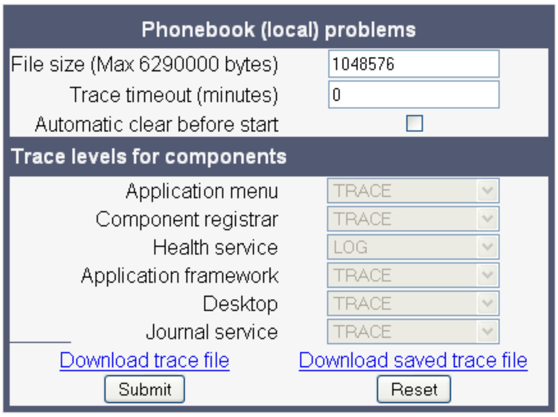
Phonebook (LDAP) problems (V2R2)



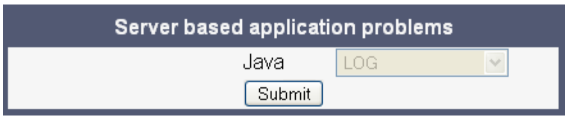
Phonebook (local) problems



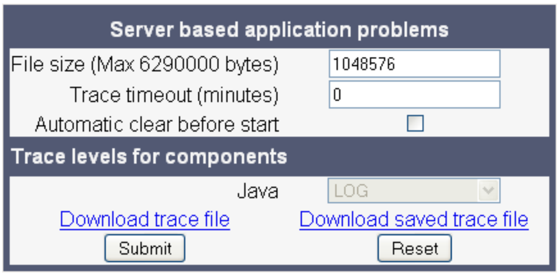
Phonebook (local) problems (V2R2)



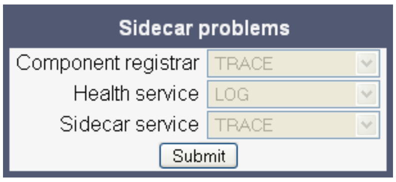
Server based application problems



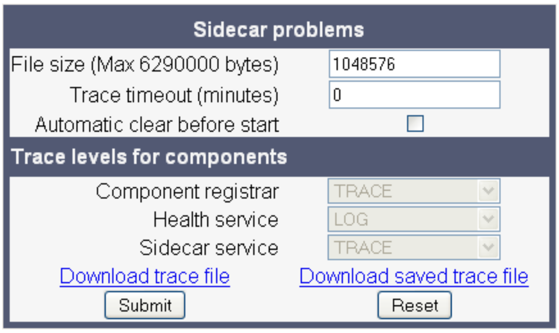
Server based application problems (V2R2)



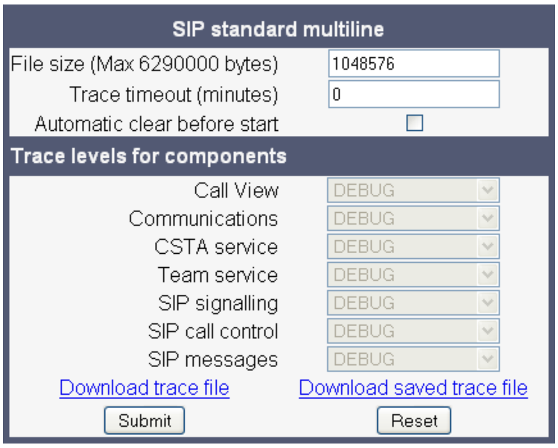
Sidecar problems



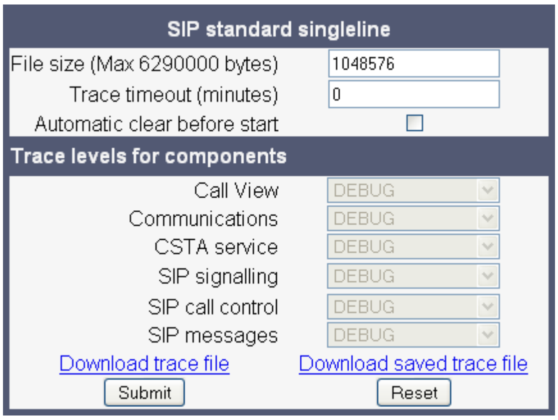
Sidecar problems (V2R2)



SIP standard multiline (V3)

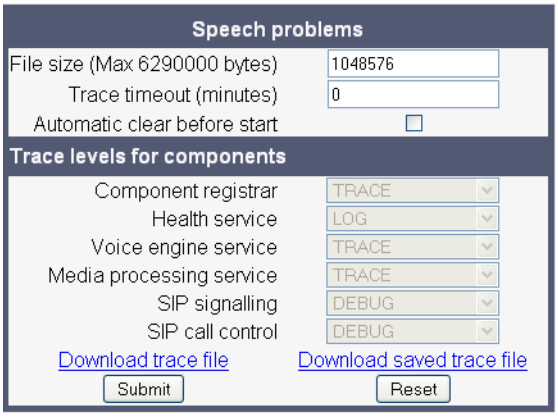


SIP standard singleline (V3)

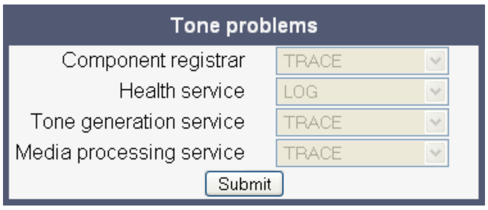


Speech problems

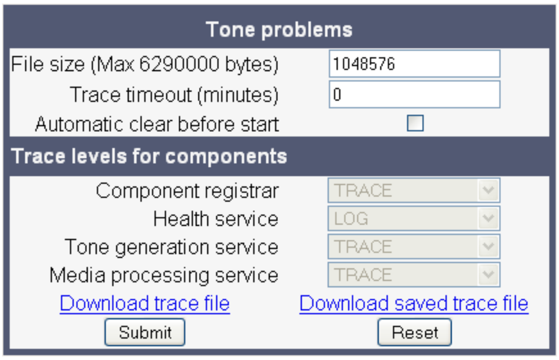
Speech problems



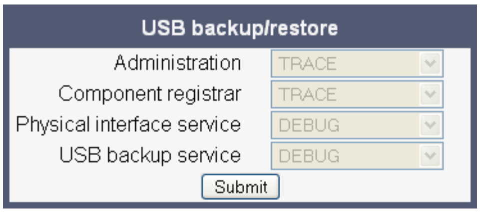
Tone problems



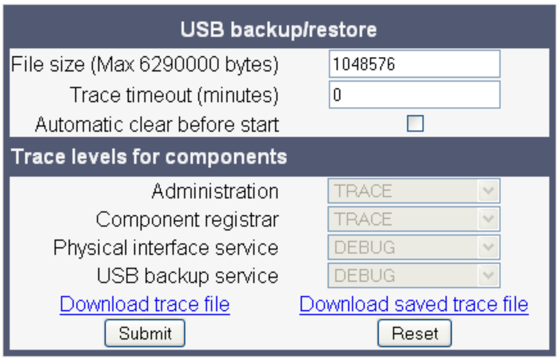
Tone problems (V2R2)



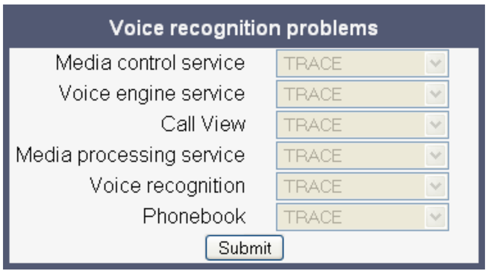
USB backup/restore



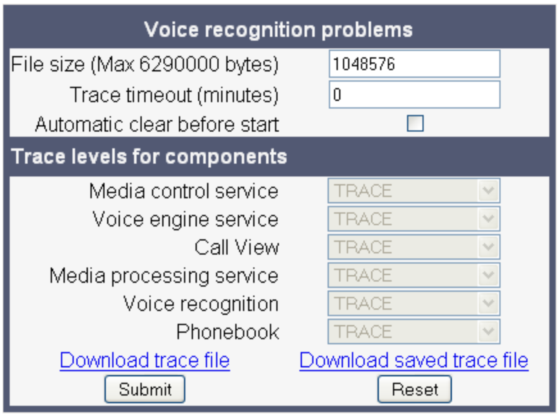
USB backup/restore (V2R2)



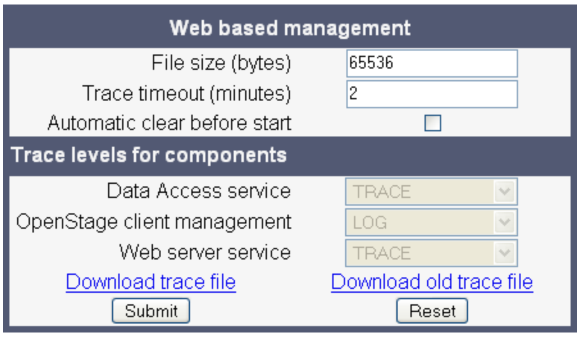
Voice recognition problems



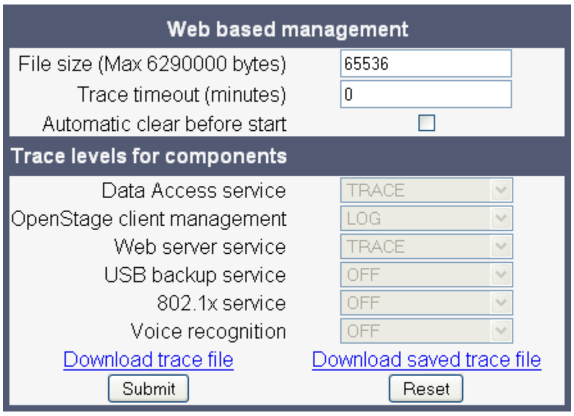
Voice recognition problems (V2R2)



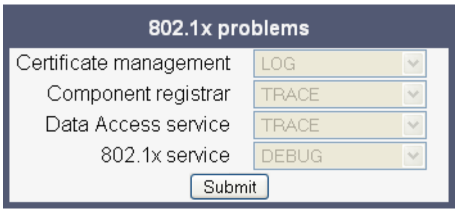
Web based management (V1R5)



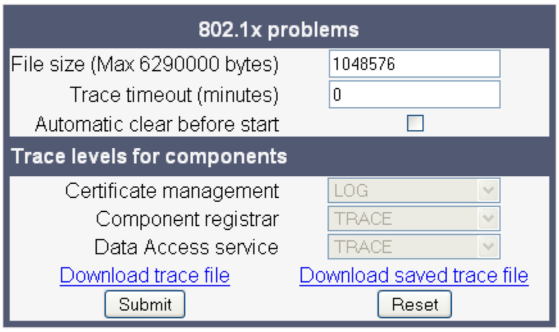
Web based management (V2)



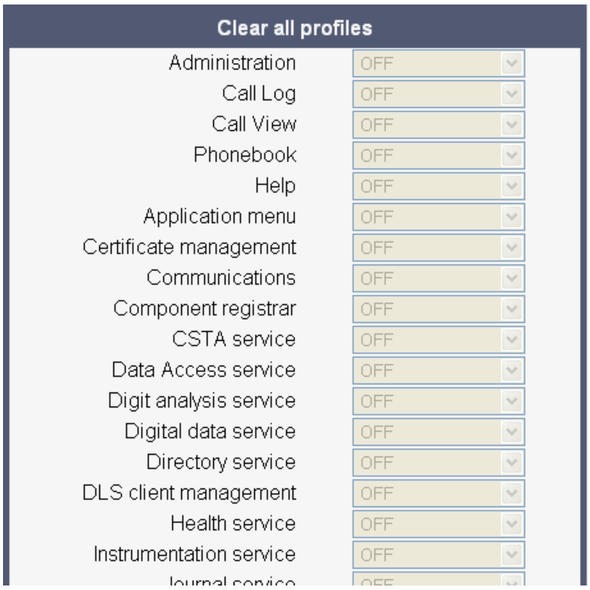
802.1x problems



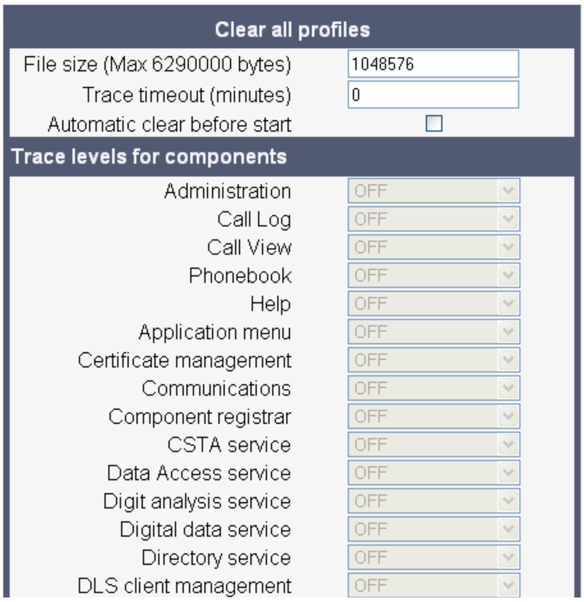
802.1x problems (V2R2)



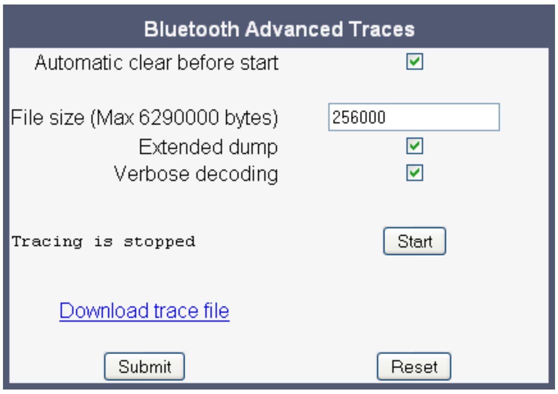
Clear all profiles



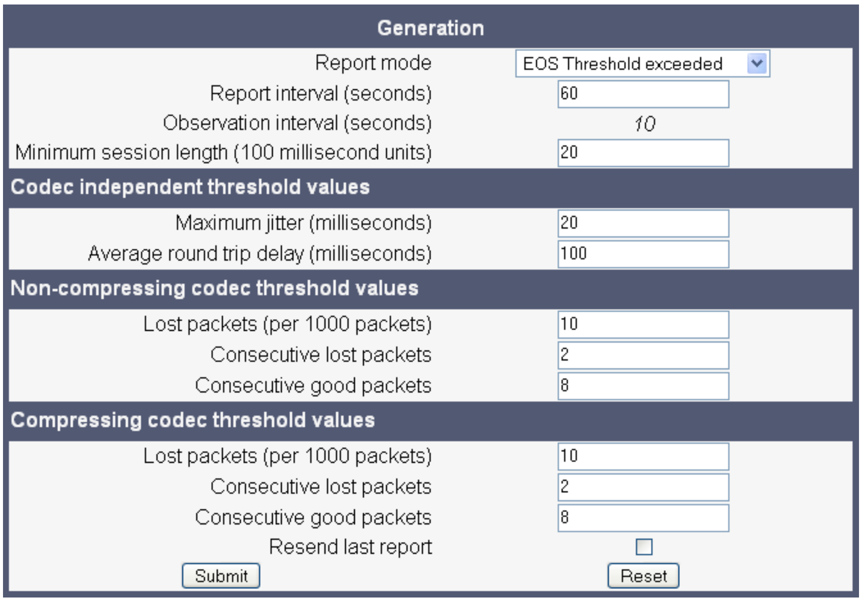
Clear all profiles (V2R2)



Bluetooth Advanced Traces (V2)



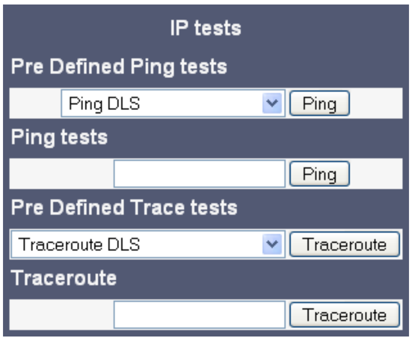
Generation


Generation
Report mode
EOS Threshold exceeded
Report interval (seconds)
60
Observation interval (seconds)
10
Minimum session length (100 millisecond units)
20
Codec independent threshold values
Maximum jitter (milliseconds)
20
Average round trip delay (milliseconds)
100
Non-compressing codec threshold values
Lost packets (per 1000 packets)
10
Consecutive lost packets
2
Consecutive good packets
8
Compressing codec threshold values
Lost packets (per 1000 packets)
10
Consecutive lost packets
2
Consecutive good packets
8
Resend last report
Submit
Reset

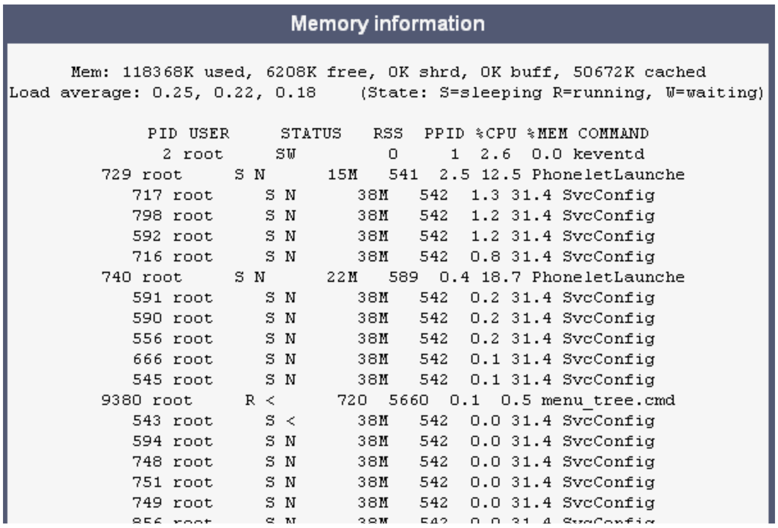
View Session Data

View Session Data	
Select a report to view	QoS Statistics 1
Submit	
Start of report period - seconds	3394450938
Start of report period - fraction of seconds	31669
End of report period - seconds	3394451013
End of report period - fraction of seconds	17820
SNMP specific trap type	0
IP address (local)	192.168.1.12
Port number (local)	5004
IP address (remote)	192.168.1.15
Port number (remote)	5010
SSRC (receiving)	324951319
SSRC (sending)	1987331861
Codec	0
Maximum packet size	20
Silence suppression	0
Count of good packets	3638
Maximum jitter	4
Maximum inter-arrival jitter	2
Periods jitter threshold exceeded	0
Round trip delay	2
Round trip delay threshold exceeded	☐
Count of lost packets	0
Count of discarded packets	0
Periods of lost packets	0
Consecutive packet loss (CPL)	255,255,255,255,255,255,255,255,255,255,255
Periods of consecutive lost packets	255
Consecutive good packets (CGP)	255,255,255,255,255,255,255,255,255,255,255
Periods of consecutive good packets	255
Count of jitter buffer overruns	0
Count of jitter buffer under-runs	0
Codec change on the fly	☐
Periods with at least one threshold exceeded	0
HiPath Switch ID	Unknown
LTU number	255
Slot number	255
Endpoint type	
Version	V1 R2.2.63 SIP 070629
Subscriber number type	0
Subscriber number	4711
Call ID	122384c56462fd0a6bfa22b6364005f3@192.168.1.21
MAC address	0001e3247e50

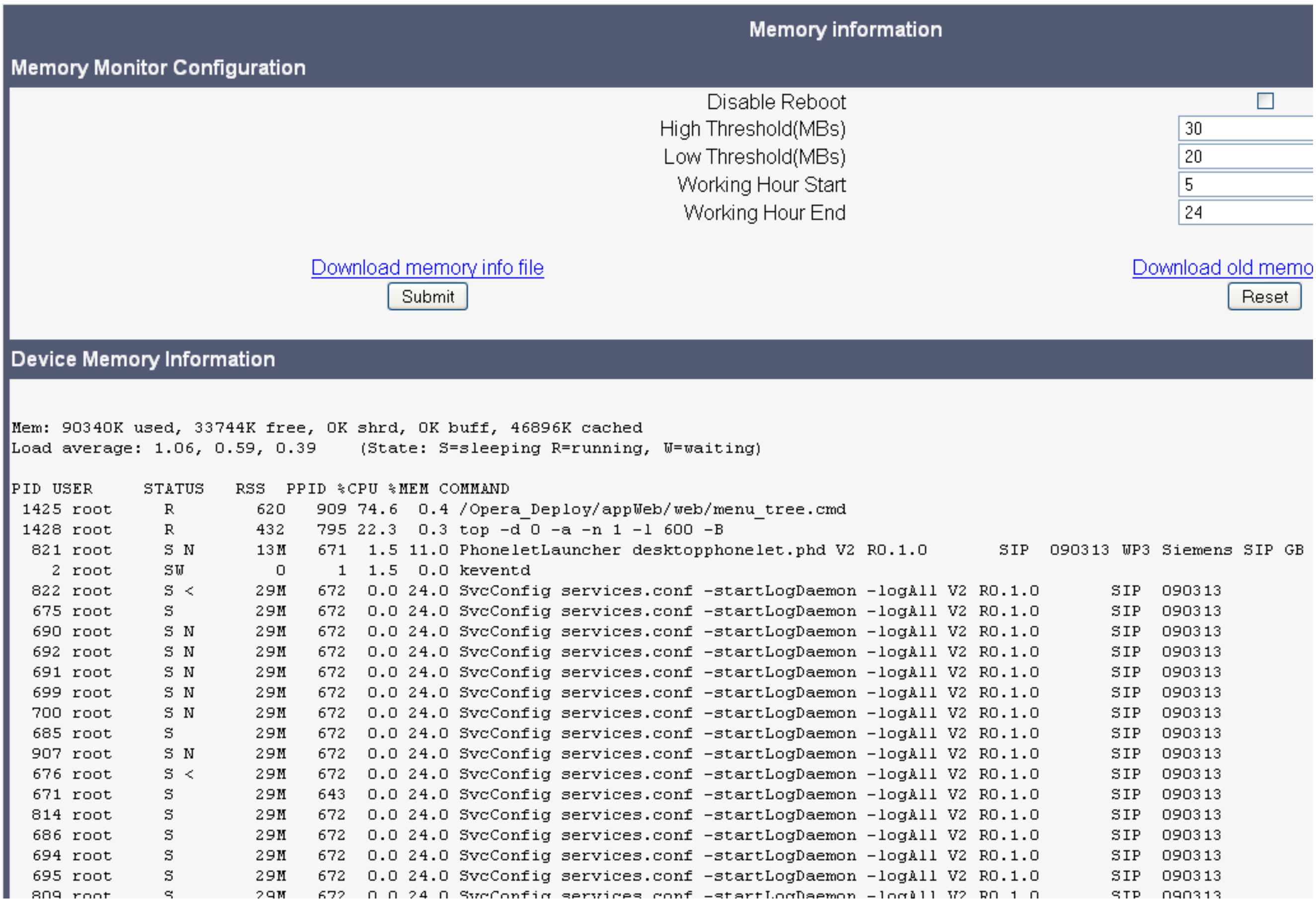
IP tests



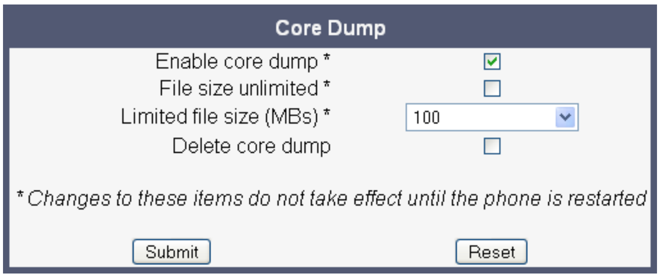
Memory information (V1R5)



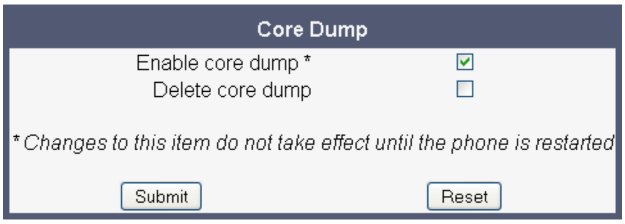
Memory information (V2)



Core dump

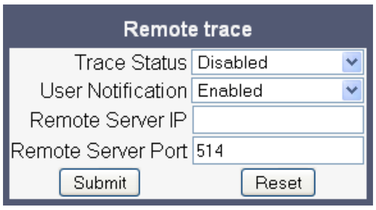


Core dump (V2R1)

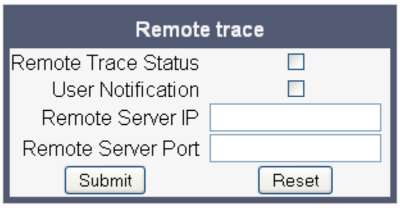


Remote trace (V1R5)

Remote trace (V2)



Remote trace (V2R2)

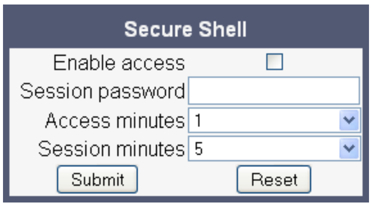


Restart phone

Factory reset

HPT interface

Secure shell (V2)



4.1.2 Local Phone Menu

Menu — **Further information ...**

- Administration
  - Applications[1]
    - CPP
    - Java
    - XML
      - Add application
        - Display name -> Section 3.17.1.1
        - Application name -> Section 3.17.1.1
        - Server address -> Section 3.17.1.1
        - Server port -> Section 3.17.1.1
        - Protocol -> Section 3.17.1.1
        - Program name -> Section 3.17.1.1
        - Auto start[2] -> Section 3.17.1.1
        - Use proxy -> Section 3.17.1.1
        - XML trace enabled -> Section 3.17.1.1
        - Debug program name -> Section 3.17.1.1
        - Number of tabs -> Section 3.17.1.1
        - All tabs start[2] -> Section 3.17.1.1
        - Tab 1 display name -> Section 3.17.1.1
        - Tab 1 application name -> Section 3.17.1.1
        - Tab 2 display name -> Section 3.17.1.1
        - Tab 2 application name -> Section 3.17.1.1
        - Tab 3 display name -> Section 3.17.1.1
        - Tab 3 application name -> Section 3.17.1.1
        - Auto restart / Restart after change -> Section 3.17.1.1
      - Add Xpressions
        - Display name -> Section 3.17.1
        - Application name -> Section 3.17.1
        - Server address -> Section 3.17.1
        - Server port -> Section 3.17.1
        - Protocol -> Section 3.17.1
        - Program name -> Section 3.17.1
        - Auto start[2] -> Section 3.17.1
        - Use proxy -> Section 3.17.1
        - XML trace enabled -> Section 3.17.1
        - Debug program name -> Section 3.17.1
        - Number of tabs -> Section 3.17.1.1
        - All tabs start[2] -> Section 3.17.1.1
        - Tab 1 display name -> Section 3.17.1.1
        - Tab 1 application name -> Section 3.17.1.1
        - Tab 2 display name -> Section 3.17.1.1
        - Tab 2 application name -> Section 3.17.1.1
        - Tab 3 display name -> Section 3.17.1.1
        - Tab 3 application name -> Section 3.17.1.1
        - Auto restart / Restart after change -> Section 3.17.1.1
      - Add phonebook
        - Display name -> Section 3.17.1
        - Application name -> Section 3.17.1
        - Server address -> Section 3.17.1

Menu	Further information ...
— Server port	-> Section 3.17.1
— Protocol	-> Section 3.17.1
— Program name	-> Section 3.17.1
— Auto start[2]	-> Section 3.17.1
— Use proxy	-> Section 3.17.1
— XML trace enabled	-> Section 3.17.1
— Debug program name	-> Section 3.17.1
— Number of tabs	-> Section 3.17.1.1
— All tabs start[2]	-> Section 3.17.1.1
— Tab 1 display name	-> Section 3.17.1.1
— Tab 1 application name	-> Section 3.17.1.1
— Tab 2 display name	-> Section 3.17.1.1
— Tab 2 application name	-> Section 3.17.1.1
— Tab 3 display name	-> Section 3.17.1.1
— Tab 3 application name	-> Section 3.17.1.1
— Auto restart	-> Section 3.17.1.1
— Add application[2]	
— Display name	-> Section 3.17.1
— Application name	-> Section 3.17.1
— Server address	-> Section 3.17.1
— Server port	-> Section 3.17.1
— Protocol	-> Section 3.17.1
— Program name	-> Section 3.17.1
— Use proxy	-> Section 3.17.1
— XML trace enabled	-> Section 3.17.1
— Debug program name	-> Section 3.17.1
— Number of tabs	-> Section 3.17.1.1
— Tab 1 display name	-> Section 3.17.1.1
— Tab 1 application name	-> Section 3.17.1.1
— Tab 2 display name	-> Section 3.17.1.1
— Tab 2 application name	-> Section 3.17.1.1
— Tab 3 display name	-> Section 3.17.1.1
— Tab 3 application name	-> Section 3.17.1.1
— Auto restart	-> Section 3.17.1.1
— Add application[2]	
— Display name	-> Section 3.17.1
— Application name	-> Section 3.17.1
— Server address	-> Section 3.17.1
— Server port	-> Section 3.17.1
— Protocol	-> Section 3.17.1
— Program name	-> Section 3.17.1
— Use proxy	-> Section 3.17.1
— XML trace enabled	-> Section 3.17.1
— Debug program name	-> Section 3.17.1
— Number of tabs	-> Section 3.17.1.1
— Tab 1 display name	-> Section 3.17.1.1
— Tab 1 application name	-> Section 3.17.1.1
— Tab 2 display name	-> Section 3.17.1.1
— Tab 2 application name	-> Section 3.17.1.1
— Tab 3 display name	-> Section 3.17.1.1
— Tab 3 application name	-> Section 3.17.1.1

Menu	Further information ...
└— Auto restart	-> Section 3.17.1.1
└— ? Add application[2]	
— Display name	-> Section 3.17.1
— Application name	-> Section 3.17.1
— Server address	-> Section 3.17.1
— Server port	-> Section 3.17.1
— Protocol	-> Section 3.17.1
— Program name	-> Section 3.17.1
— Use proxy	-> Section 3.17.1
— XML trace enabled	-> Section 3.17.1
— Debug program name	-> Section 3.17.1
— Number of tabs	-> Section 3.17.1.1
— Tab 1 display name	-> Section 3.17.1.1
— Tab 1 application name	-> Section 3.17.1.1
— Tab 2 display name	-> Section 3.17.1.1
— Tab 2 application name	-> Section 3.17.1.1
— Tab 3 display name	-> Section 3.17.1.1
— Tab 3 application name	-> Section 3.17.1.1
└— Auto restart	-> Section 3.17.1.1
— Network	
— IP configuration / IPv4 configuration	
— Discovery mode	-> Section 3.2.2
— Use LLDP-Med	-> Section 3.2.2
— Use DHCP	-> Section 3.2.2
— DHCP reuse	-> Section 2.3.4
— IP address	-> Section 3.3.3
— Subnet mask	-> Section 3.3.3
— Route (default)	-> Section 3.3.4
— DNS domain	-> Section 3.3.6.1
— Primary DNS	-> Section 3.3.6.2
— Secondary DNS	-> Section 3.3.6.2
— Route 1 IP	-> Section 3.3.6
— Route 1 gateway	-> Section 3.3.6
— Route 1 mask	-> Section 3.3.6
— Route 2 IP	-> Section 3.3.6
— Route 2 gateway	-> Section 3.3.6
— Route 2 mask	-> Section 3.3.6
— VLAN discovery	-> Section 3.2.2.1
— VLAN ID	-> Section 3.2.2.3
└— HTTP proxy[1]	-> Section 3.17.1.2
— Update Service (DLS)	
— DLS address	-> Section 3.3.7
— DLS port	-> Section 3.3.7
— Contact gap	-> Section 3.3.7
└— Security status	-> Section 3.3.7
— QoS	
— Service	
— Layer 2	-> Section 3.3.1.1
— Layer 2 voice	-> Section 3.3.1.1
— Layer 2 signalling	-> Section 3.3.1.1
— Layer 2 default	-> Section 3.3.1.1

Menu	Further information ...
|— Layer 3	-> Section 3.3.1.2
|— Layer 3 voice	-> Section 3.3.1.2
└— Layer 3 signalling	-> Section 3.3.1.2
└— Reports	
|— Generation	
|— Mode	-> Section 3.24.9
|— Report interval	-> Section 3.24.9
|— Observe interval	-> Section 3.24.9
└— Minimum session length	-> Section 3.24.9
|— Send now	-> Section 3.24.9.1
└— Thresholds	
|— Maximum jitter	-> Section 3.24.9
|— Round-trip delay	-> Section 3.24.9
|— Non-compressing:	
|— ...Lost packets (K)	-> Section 3.24.9
|— ...Lost consecutive	-> Section 3.24.9
|— ...Good consecutive	-> Section 3.24.9
|— Compressing:	
|— ...Lost packets (K)	-> Section 3.24.9
|— ...Lost consecutive	-> Section 3.24.9
└— ...Good consecutive	-> Section 3.24.9
|— Port configuration	
|— SIP server	-> Section 3.5.5.2
|— SIP registrar	-> Section 3.5.5.2
|— SIP gateway	-> Section 3.5.5.2
|— SIP local	-> Section 3.5.5.2
|— Backup proxy	-> Section 3.5.9.5
|— RTP base	-> Section 3.16.1
|— LDAP server port	-> Section 3.15.1
|— LAN port type	-> Section 3.2.1
|— PC port status	-> Section 3.2.1
|— PC port type	-> Section 3.2.1
|— PC port autoMDIX	-> Section 3.2.1
└— HTTP proxy	-> Section 3.17.1.2
└— LLDP-MED operation	
|— Extended Power	-> Section 3.24.3
|— Network policy (voice)	-> Section 3.24.3
|— LLDEP-MED cap's	-> Section 3.24.3
|— MAC_Phy config	-> Section 3.24.3
|— System cap's	-> Section 3.24.3
└— TTL	-> Section 3.2.3
|— System	
|— Identity	
|— Terminal number	-> Section 3.5.1.1
|— Terminal name	-> Section 3.5.1.1
|— Display identity	-> Section 3.5.1.2
|— Enable ID	-> Section 3.5.1.2
|— Web name[3]	-> Section 3.3.6.3
└— DDNS hostname[3]	-> Section 3.3.6.3
|— SIP Interface	
|— Outbound proxy	-> Section 3.5.7.1

Menu	Further information ...
— Default OBP domain	-> Section 3.5.7.1
— SIP transport	-> Section 3.5.7.2
— Call trans (ms) / Response timer (ms)	-> Section 3.5.9.2
— NonCall trans (ms)	-> Section 3.5.9.3
— Registration backoff	-> Section 3.5.9.4
— Connectivity timer (ms)	-> Section 3.5.9.1
— Registration	
— SIP addresses	
— SIP server	-> Section 3.5.5.1
— SIP registrar	-> Section 3.5.5.1
— SIP gateway	-> Section 3.5.5.1
— SIP session	
— Session timer	-> Section 3.5.8
— Session duration (s)	-> Section 3.5.8
— Registration timer (s)	-> Section 3.5.6
— Server type	-> Section 3.5.6
— Realm	-> Section 3.5.6
— User ID	-> Section 3.5.6
— Password	-> Section 3.5.6
— SIP survivability	
— Backup registration flag	-> Section 3.5.9.5
— Backup proxy address	-> Section 3.5.9.5
— Backup registration timer (s)	-> Section 3.5.9.5
— Backup transport	-> Section 3.5.9.5
— OBP flag	-> Section 3.5.9.5
— SNMP	
— Queries allowed	-> Section 3.3.8
— Query password	-> Section 3.3.8
— Trap sending enabled	-> Section 3.3.8
— Trap destination	-> Section 3.3.8
— Trap destination port	-> Section 3.3.8
— Trap community	-> Section 3.3.8
— Diagnostic sending enabled	-> Section 3.3.8
— Diagnostic destination	-> Section 3.3.8
— Diagnostic destination port	-> Section 3.3.8
— Diagnostic community	-> Section 3.3.8
— QoS traps to QCU	-> Section 3.3.8
— QCU address	-> Section 3.3.8
— QCU port	-> Section 3.3.8
— QCU community	-> Section 3.3.8
— QoS to generic destination	-> Section 3.3.8
— Features	
— Configuration	
— General	
— Emergency number	-> Section 3.5.2
— Voicemail number	-> Section 3.5.2
— Allow refuse	-> Section 3.6.1
— Hot / warm phone[3]	-> Section 3.6.2
— Hot / warm destination[3]	-> Section 3.6.2
— Initial digit timer	-> Section 3.6.3
— Allow uaCSTA	-> Section 3.6.13

Menu	Further information ...
— Server features	-> Section 3.6.11
— Transfer on hangup	-> Section 3.6.5.2
— Not used timeout	-> Section 3.6.14
— DSS Pickup timer	-> Section 3.9.5.1
— Bridging enabled	-> Section 3.9.2
— Dial plan[3]	-> Section 3.11.3
— FPK prog. timer[2]	-> Section 3.7
— Audio	
— Group pickup tone allowed[4]	-> Section 3.6.4.2
— Group pickup as ringer	-> Section 3.6.4.2
— Group pickup visual alert	-> Section 3.6.4.2
— BLF alerting	-> Section 3.6.4.2
— Keyset Lines	
— Details For Keyset Line <n>	
— Address	-> Section 3.9.1
— Ring on/off	-> Section 3.9.1
— Selection order	-> Section 3.9.1
— Hot/warm action[3]	-> Section 3.9.1
— Bluetooth	
— Local device address[2]	-> Section 3.25
— Enable[2]	-> Section 3.25
— Call recording	
— Recorder number	
— Recording mode	
— Audible notification	
— Keyset operation[4]	
— Rollover ring	-> Section 3.9.2
— LED on registration	-> Section 3.9.2
— Originating line preference	-> Section 3.9.2
— Terminating line preference	-> Section 3.9.2
— Line action mode	-> Section 3.9.2
— Show focus	-> Section 3.9.2
— Reservation timer	-> Section 3.9.2
— Forwarding indicated / Forwarding shown	-> Section 3.9.2
— Preselect mode	-> Section 3.9.2
— Preselect timer	-> Section 3.9.2
— Preview mode[3]	-> Section 3.9.3
— Preview timer[3]	-> Section 3.9.3
— DSS operation[4]	
— Deflect to DSS	-> Section 3.9.5.1
— Refuse DSS pickup	-> Section 3.9.5.1
— Forwarding shown	-> Section 3.9.5.1
— Group pickup	
— Group pickup tone / Pickup tone allowed	-> Section 3.6.4.2
— Group pickup as ringer / Pickup as ringer	-> Section 3.6.4.2
— Group pickup visual / Pickup visual alert	-> Section 3.6.4.2
— BLF alerting	
— Addressing	
— MWI server URI	-> Section 3.6.7
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— Group pickup URI	-> Section 3.6.4

Menu	Further information ...
— Callback: busy	-> Section 3.6.6
— Callback: no reply	-> Section 3.6.6
— Callback: cancel all	-> Section 3.6.6
— BLF pickup code	-> Section 3.6.4
— Feature Access	
— Call establish[4]	
— Deflect to DSS	-> Section 3.9.5.1
— Refuse DSS pickup	-> Section 3.9.5.1
— Security	
— Server cerfificate	-> Section 3.4
— Backup certificate	-> Section 3.4
— Use secure calls	-> Section 3.4
— File Transfer	
— Defaults	-> Section 3.14.2
— Download method	-> Section 3.14.2
— FTP Server	-> Section 3.14.2
— FTP Port	-> Section 3.14.2
— FTP Account	-> Section 3.14.2
— FTP Username	-> Section 3.14.2
— FTP Password	-> Section 3.14.2
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— Phone app	-> Section 3.14.3
— Use default	-> Section 3.14.3.1
— Download method	-> Section 3.14.3.1
— FTP Server	-> Section 3.14.3.1
— FTP Port	-> Section 3.14.3.1
— FTP Account	-> Section 3.14.3.1
— FTP Username	-> Section 3.14.3.1
— FTP Password	-> Section 3.14.3.1
— FTP path	-> Section 3.14.3.1
— HTTPS base URL	-> Section 3.14.3.1
— Filename	-> Section 3.14.3.1
— Hold Music	
— FTP Use default	-> Section 3.14.4.1
— FTP Download method	-> Section 3.14.4.1
— FTP Server	-> Section 3.14.4.1
— FTP Port	-> Section 3.14.4.1
— FTP Account	-> Section 3.14.4.1
— FTP Username	-> Section 3.14.4.1
— FTP Password	-> Section 3.14.4.1
— FTP path	-> Section 3.14.4.1
— HTTPS base URL	-> Section 3.14.4.1
— Filename	-> Section 3.14.4.1
— Ringer	
— Use default	-> Section 3.14.6.1
— Download method	-> Section 3.14.6.1
— FTP Server	-> Section 3.14.6.1
— FTP Port	-> Section 3.14.6.1
— FTP Account	-> Section 3.14.6.1
— FTP Username	-> Section 3.14.6.1

Menu	Further information ...
├─ FTP Password	-> Section 3.14.6.1
├─ FTP path	-> Section 3.14.6.1
├─ HTTPS base URL	-> Section 3.14.6.1
└─ Filename	-> Section 3.14.6.1
─ Picture clip[2]	
├─ Use default	-> Section 3.14.5.1
├─ Download method	-> Section 3.14.5.1
├─ FTP Server	-> Section 3.14.5.1
├─ FTP Port	-> Section 3.14.5.1
├─ FTP Account	-> Section 3.14.5.1
├─ FTP Username	-> Section 3.14.5.1
├─ FTP Password	-> Section 3.14.5.1
├─ FTP path	-> Section 3.14.5.1
├─ HTTPS base URL	-> Section 3.14.5.1
└─ Filename	-> Section 3.14.5.1
─ LDAP[2]	
├─ Use default	-> Section 3.14.6.1
├─ Download method	-> Section 3.14.6.1
├─ FTP Server	-> Section 3.14.6.1
├─ FTP Port	-> Section 3.14.6.1
├─ FTP Account	-> Section 3.14.6.1
├─ FTP Username	-> Section 3.14.6.1
├─ FTP Password	-> Section 3.14.6.1
├─ FTP path	-> Section 3.14.6.1
├─ HTTPS base URL	-> Section 3.14.6.1
└─ Filename	-> Section 3.14.6.1
─ Logo[5]	
├─ Use default	-> Section 3.14.7.1
├─ Download method	-> Section 3.14.7.1
├─ FTP Server	-> Section 3.14.7.1
├─ FTP Port	-> Section 3.14.7.1
├─ FTP Account	-> Section 3.14.7.1
├─ FTP Username	-> Section 3.14.7.1
├─ FTP Password	-> Section 3.14.7.1
├─ FTP path	-> Section 3.14.7.1
├─ HTTPS base URL	-> Section 3.14.7.1
└─ Filename	-> Section 3.14.7.1
─ Screensaver[2]	
├─ Use default	-> Section 3.14.8.1
├─ Download method	-> Section 3.14.8.1
├─ FTP Server	-> Section 3.14.8.1
├─ FTP Port	-> Section 3.14.8.1
├─ FTP Account	-> Section 3.14.8.1
├─ FTP Username	-> Section 3.14.8.1
├─ FTP Password	-> Section 3.14.8.1
├─ FTP path	-> Section 3.14.8.1
├─ HTTPS base URL	-> Section 3.14.8.1
└─ Filename	-> Section 3.14.8.1
└─ Java midlets[2]	
├─ Use default	
├─ Download method	

Menu	Further information ...
├─ FTP Server	
├─ FTP Port	
├─ FTP Account	
├─ FTP Username	
├─ FTP Password	
├─ FTP path	
├─ HTTPS base URL	
└─ Filename	
─ Local Functions	
├─ Directory Settings / LDAP[2]	
├─ (LDAP) server address	-> Section 3.15.1
├─ (LDAP) server port	-> Section 3.15.1
├─ Timeout (sec) for / Search Trigger (s)	
├─ (LDAP) authenticate / Authentication	-> Section 3.15.1
├─ (LDAP) user name	-> Section 3.15.1
└─ (LDAP) password	-> Section 3.15.1
├─ Locality	
├─ Canonical settings	
├─ Local country code	-> Section 3.11.1
├─ National prefix digit	-> Section 3.11.1
├─ Local national code	-> Section 3.11.1
├─ Minimum local number length	-> Section 3.11.1
├─ Local enterprise node	-> Section 3.11.1
├─ PSTN access code	-> Section 3.11.1
├─ International access code	-> Section 3.11.1
├─ Operator code	-> Section 3.11.1
├─ Emergency number	-> Section 3.11.1
└─ Initial digits	-> Section 3.11.1
├─ Canonical lookup	
├─ Local code 1	-> Section 3.11.2
├─ International code 1	-> Section 3.11.2
├─ Local code 2	-> Section 3.11.2
├─ International code 2	-> Section 3.11.2
├─ Local code 3	-> Section 3.11.2
├─ International code 3	-> Section 3.11.2
├─ Local code 4	-> Section 3.11.2
├─ International code 4	-> Section 3.11.2
├─ Local code 5	-> Section 3.11.2
└─ International code 5	-> Section 3.11.2
└─ Canonical dial	
├─ Internal numbers	-> Section 3.11.1
├─ External numbers	-> Section 3.11.1
├─ External access code	-> Section 3.11.1
└─ International gateway / International access	-> Section 3.11.1
├─ Energy saving[5]	
└─ Backlight timeout	-> Section 3.5.3
└─ Messages settings	
├─ New items	
├─ Alternative label	
├─ New urgent items	
├─ Alternative label	

```
Menu                                              Further information ...
|     |          |— Old items
|     |          |— Alternative label
|     |          |— Old urgent items
|     |          |— Alternative label
|     |— Date and Time
|     |     |— Time source
|     |     |     |— SNTP IP address              -> Section 3.5.4.1
|     |     |     |— Timezone offset              -> Section 3.5.4.1
|     |     |— Daylight saving                    -> Section 3.5.4.1
|     |     |     |— Daylight saving              -> Section 3.5.4.1
|     |     |     |— Difference (mins)            -> Section 3.5.4.1
|     |     |     |— Auto DST                     -> Section 3.5.4.1
|     |     |     |— DST zone                     -> Section 3.5.4.1
|     |— Speech
|     |     |— Codec Preferences
|     |     |     |— Silence suppression          -> Section 3.16.2
|     |     |     |— Packet size                  -> Section 3.16.2
|     |     |     |— G.711                        -> Section 3.16.2
|     |     |     |— G.729                        -> Section 3.16.2
|     |     |     |— G.722                        -> Section 3.16.2
|     |     |— Audio Settings
|     |           |— Disable microphone           -> Section 3.16.3
|     |           |— Disable loudspeech           -> Section 3.16.3
|     |— General Information
|     |     |— MAC address                        -> Section 3.24.1
|     |     |— Software version                   -> Section 3.24.1
|     |     |— Last restart                       -> Section 3.24.1
|     |     |— Dial plan ID[6]                    -> Section 3.11.3
|     |     |— Dial plan status[6]                -> Section 3.11.3
|     |— Licence information                      -> Section 3.23
|     |— Password
|     |     |— Admin                              -> Section 3.18
|     |     |— Confirm admin                      -> Section 3.18
|     |     |— User                               -> Section 3.18
|     |     |— Confirm user                       -> Section 3.18
|     |— Security & policies[6]
|     |     |— Password
|     |     |     |— Change admin password
|     |     |     |     |— Current password
|     |     |     |     |— New password
|     |     |     |     |— Confirm password
|     |     |     |— Change user password
|     |     |           |— Current password
|     |     |           |— New password
|     |     |           |— Confirm password
|     |     |— Certificates
|     |     |     |— Authentication policy
|     |     |           |— Secure file transfer
|     |     |           |— Secure send URL
|     |— Ringer setting
|     |     |— <1 .... 15>
```

Menu	Further information ...
│ │ ├── Name	-> Section 3.12
│ │ ├── Ringer sound	-> Section 3.12
│ │ ├── Pattern melody	-> Section 3.12
│ │ ├── Pattern sequence	-> Section 3.12
│ │ ├── Duration	-> Section 3.12
│ │ └── Audible	-> Section 3.12
│ ├── Mobility	
│ │ ├── Unauthorized logoff trap	-> Section 3.13
│ │ ├── Logoff trap delay	-> Section 3.13
│ │ ├── Timer med priority	-> Section 3.13
│ │ ├── Mobility feature	-> Section 3.13
│ │ ├── Managed profile	
│ │ ├── Error count local	
│ │ └── Error count remote	
│ └── Maintenance	
│ ├── Factory reset	-> Section 3.21
│ ├── Disable HPT	-> Section 3.24.12
│ ├── Remote trace	
│ │ ├── Remote trace status	-> Section 3.24.11
│ │ ├── User notification[3]	-> Section 3.24.11
│ │ ├── Remote IP	-> Section 3.24.11
│ │ └── Remote port	-> Section 3.24.11
│ └── Memory monitor	
│ ├── Disable reboot	-> Section 3.24.5
│ ├── High threshold	-> Section 3.24.5
│ ├── Low threshold	-> Section 3.24.5
│ ├── Working Hour start	-> Section 3.24.5
│ └── Working Hour end	-> Section 3.24.5

1 OpenStage 60/80 only.
2 V2R1 onwards only.
3 V2 only.
4 OpenStage 15/40/60/80 only.
5 OpenStage 40/60/80 only.
6 V2R2 onwards only.

4.2 Default Port List

The following table contains all default ports, resp. port ranges, and protocols used by the services running on OpenStage SIP phones.

Service	Server Default Port	Client Default Port	Protocol Stack
Payload transport (for 30 lines)	5004 - 5065	5004 - 5065	RTP - RTCP / UDP
SIP subscriber; TCP is used	5060	1024 - 65535	SIP / TCP
SIP subscriber; TLS is used	5061	1024 - 65535	SIP / TLS
SIP subscriber; UDP is used	5060	5060	SIP / UDP
XML applications in phone, connecting to an application server	---	1024 - 65535	HTTP / TCP
Directory access via LDAP (Only relevant for OpenStage 60/80)	---	1024 - 65535	LDAP / TCP
DHCP Client	---	68	DHCP / UDP
DNS Client	---	1024 - 65535	DNS / TCP_UDP
DLS contact me service - workpoint side	8085	---	HTTP / TCP
Communication with the DLS workpoint interface, default mode	---	18443	HTTPS / TCP - SSL / TLS
Communication with the DLS workpoint interface, secure mode	---	18444	HTTPS / TCP - SSL / TLS
Connection to the control port of FTP server	21	1024 - 65535	FTP / TCP
FTP client; uses the FTP server in active mode	1024 - 65535	20	FTP / TCP
HTTPS file download server	---	443	HTTPS / TCP - SSL/TLS
Client application which sends QDC data to the QCU	---	1024 - 65535	SNMP / UDP
Sender part of SNMP agent	---	1024 - 65535	SNMP / UDP
Receiver part of SNMP agent; receives Set/Get commands	161	---	SNMP / UDP
SNTP client; queries time information in unicast operation	---	123	SNTP / UDP

Service	Server Default Port	Client Default Port	Protocol Stack
SNTP client; receives time information in broadcast operation	123	---	SNTP / UDP
Web server for unencrypted WBM access (up to firmware version V1.4; in higher versions, only encrypted-connections are possible)	8085	---	HTTP / TCP
Secure web Server for encrypted WBM access	443	---	HTTPS / TCP - SSL / TLS
OpenStage Phone Manager	65530	---	HTTP / UDP
OpenStage Phone Manager	65531	---	HTTP / TCP

4.3 Troubleshooting: Error Codes

For a set of error cases, specific error codes are defined. These error codes are shown in brackets on the display, following a general error note. Example: „No Telephony possible (LP1)“.

Problem	Description	Error code
Network Problem	No network connection	LI1
Not Initialised	Waiting for data	I1
Unable to use LAN	802.1x error	LX1
Unable to use LAN	Physical connection missing	LP1
Unable to Register	Server timeout	RT2
Unable to Register	Server failed	RF2
Unable to Register	Authentication failed	RA2
Unable to Register	No number configured	RN2
Unable to Register	No server configured	RS2
Unable to Register	No registrar configured	RG2
Unable to Register	No DNS domain configured	RD2
Unable to Register	Rejected by server	RR2
Unable to Register	No phone IP address set	RI2
Survivability	Backup route active	B8
Survivability	Backup not configured	RS8
Survivability	Backup timeout	RT8
Survivability	Backup authentication failed	RA8

Tabelle 4-1

5 Examples and HowTos

5.1 Canonical Dialing

5.1.1 Canonical Dialing Settings

The following example shows settings suitable for the conversion of given dial strings to canonical format. The example phone is located in Nottingham, UK.

Parameter	Example value	Explanation
Local country code	44	International country code for the UK.
National prefix digit	0	Used in front of national codes when dialled without international prefix.
Local national code	115	Area code within the UK (here: Nottingham).
Minimum local number length	7	Minimum number of digits in a local PSTN number (e. g. 3335333 = 7 digits).
Local enterprise node	780	Prefix to access Nottingham numbers from within the Siemens network.
PSTN access code	9	Prefix to make an international call in the UK.
Operator codes	0, 7800	Set of numbers to access the local operators.
Emergency numbers	999, 555	Set of numbers to access emergency services.
Initial extension digits	2, 3, 4, 5, 6, 8	1^{st} digits of numbers that are used for extension numbers on the local node.

5.1.2 Canonical Dial Lookup

The following example shows settings suitable for recognizing incoming numbers and assigning them to entries in the local phone book, and for generating correct dial strings from phone book entries, depending on whether the number is internal or external.

Parameter	Example value	Explanation
Local code <1>	780	Enterprise node prefix (here: Nottingham).
International code <1>	+44115943	Equivalent prefix to access numbers on this node from the PSTN. Here, the prefix used by the PSTN (DID/DDI: direct inward dialing) is 943, which differs from the enterprise node prefix used within the enterprise network.
Local code <2>	722	Enterprise node prefix (here: Munich).
International code <2>	+4989722	Equivalent prefix to access numbers on this node from the PSTN. Here, the prefix used by the PSTN for direct inward dialing is identical to the enterprise node prefix.

5.1.2.1 Conversion examples

In the following examples, numbers entered into the local phonebook by the user are converted according to the settings given above.

Example 1: Internal number, same node as the local phone

User entry		2345
External numbers		Local public form
External access code		Not required
International gate-way code		Use national code
Number stored in the phone book		+441159432345
Dial string sent when dialing from the phone book	Internal numbers = Local enterprise form	1234
	Internal numbers = Always add node	7802345
	Internal numbers = Use external numbers	9432345

Example 2: Internal number, different node

User entry		7222345
External numbers		Local public form
External access code		Not required
International gate-way code		Use national code
Number stored in the phone book		+49897222345
Dial string sent when dialing from the phone book	Internal numbers = Local enterprise form	2345
	Internal numbers = Always add node	7802345
	Internal numbers = Use external numbers	9432345

Example 3: External number, same local national code as the local phone

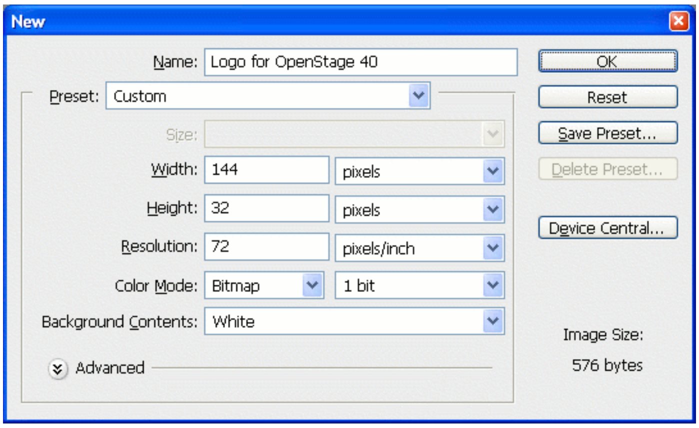
User entry		011511234567
External numbers		Local public form
External access code		Not required
International gate-way code		Use national code
Number stored in the phone book		+4411511234567
Dial string sent when dialing from the phone book	External numbers = Local public form	234567
	External numbers = National public form	011511234567
	External numbers = International form	004411511234567

5.2 How to Create Logo Files for OpenStage Phones

5.2.1 For OpenStage 40

1. **Create a New Image**
   Create an image with the following specifications:
   - Width: 144 px
   - Height: 32 px
   - Color Mode: 1 bit (monochrome)

   Adobe Photoshop:



2. **Insert the Logo**
   Place the logo image on the background, e.g. by copying it from a source file. Due to the size and color specifications, some adaptations may be necessary.

   Adobe Photoshop Example:

3. **Save the Image**
   Finally, save the image in BMP format. You can now upload the logo file to the phone as described in Section 3.14.7, “Logo”.

5.2.2 For OpenStage 60/80

In the following, the creation of a transparent image suitable for use as a logo in OpenStage 60/80 is described. This description is based on Adobe Photoshop, but any similar graphics software can be used as well.

> Because of performance issues, half transparency in the alpha channel of the PNG files is not allowed on OpenStage phones. Therefore only 100% transparency or no transparency is used in the phone’s UI elements.

1. **Select the Background Color**
   For production purposes, we set the background color to the background color of the skin currently selected on the phone. Later, the background color will be replaced by transparency, which facilitates placing a logo on a gradient background. The following table lists the hexadecimal values, as used in HTML:

Phone Type	Skin	Color Code
OpenStage 60	Crystal Sea	#BDBDBD
OpenStage 60	Warm Grey	#424242[1]
OpenStage 80	Crystal Sea	#E6EBEF
OpenStage 80	Warm Grey	#3A3D3A

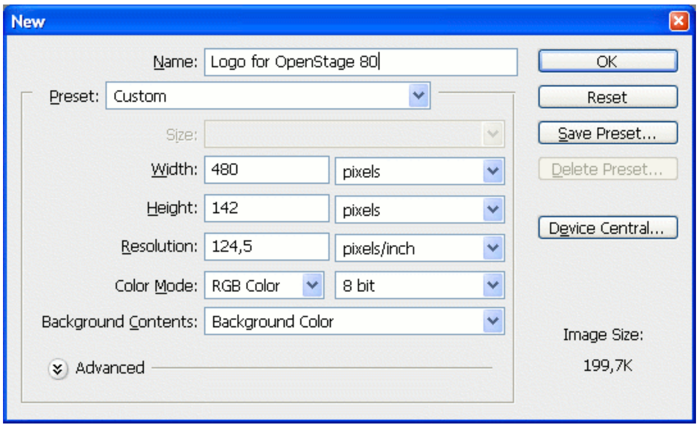
1 The background color on WP4 - skin 1 is a gradient; the colour listed here is an average value.

Adobe Photoshop:
Click on the Background Color icon on the Color palette group, then type the color code without leading "#" into the **#** field)

2. **Create a New Image**
Create an image with the size according to the phone type:

Phone Type	Size (px)
OpenStage 60	240 x 70
OpenStage 80	480 x 142

Adobe Photoshop:



3. **Insert the Logo**
Place the logo image on the background, e.g. by copying it from a source file.
Adobe Photoshop Example:

4. **Merge Layers**
Merge the two layers to one.
Adobe Photoshop:
In the Panel, select both the background layer and the new layer containing the inserted logo. Afterwards, go to **Layer** in the Menu bar, and select **Merge Layers**.

5. **Background Transparency**

   Delete the background colour so that only the exact former background colour is 100% transparent.

   Adobe Photoshop:

   Make sure that the background color is selected by clicking on the Background Color icon. In the Tool palette, click on the Eraser symbol with the right Mouse button and select the **Magic Eraser Tool**. After this, got to the Menu bar and set the **Tolerance** field to "0".

6. **Save the Image**

   Finally, save the image in PNG format. You can now upload the logo file to the phone as described in Section 3.14.7, “Logo”.

5.3 How to Set Up the Corporate Phonebook (LDAP)

The Corporate Phonebook function is based on an LDAP client that can be connected to the company's LDAP service. A variety of LDAP servers can be used, for instance Microsoft Active Directory, OpenLDAP, or Apache Directory Server.

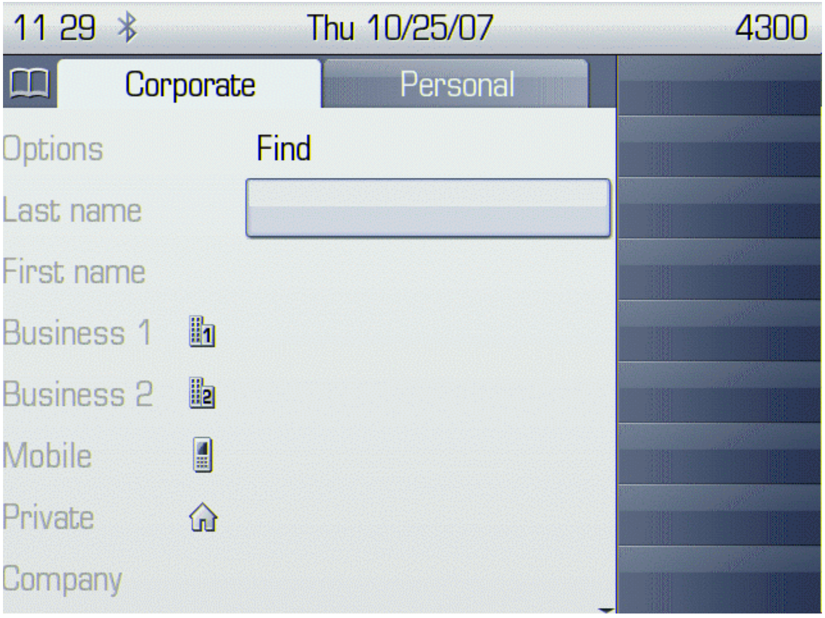
> The Corporate Phonebook is available only on OpenStage 60/80 and on OpenStage 40 phones with firmware version V2R1 onwards.

5.3.1 Prerequisites:

1. An LDAP server is present and accessible to the phone's network. The standard port for LDAP is **389**.
2. Query access to the LDAP server must be provided. Unless anonymous access is used, a user name and passwort must be provided. It might be feasible to use a single login/password for all OpenStage phones.
3. To enable dialing internal numbers from the corporate phonebook, an LDAP entry must be provided that contains the proper number format required by OpenScape Voice.
   In Microsoft Active Directory, the standard LDAP attribute `telephoneNumber` is typically populated as follows: **+1<area code><call number>**. However, in a standard configuration, OpenScape Voice will not handle this dial string correctly, due to the **+1** prefix. Therefore, it is recommended to use the **ipPhone** field, which is typically unused in Active Directory. It can be found in the **Telephones** tab of the Active Directory User Manager.

5.3.2 Create an LDAP Template

The user interface of the corporate phonebook application provides a form which is used both for search and retrieval.



The task of an LDAP template is to map the phone's search and display fields to LDAP attributes that can be delivered by the server. In the LDAP template, the fields are represented by hard-coded names: `ATTRIB01`, `ATTRIB02`, and so on. These field names are assigned to LDAP attributes, as appropriate.

The following examples show the relations between GUI field names, the attribute labels used in the template, and exemplary mappings to LDAP attributes.

> In an LDAP template for OpenStage 40, the entries must be sorted according to the sequential number of the template labels, as shown in the example underneath. For OpenStage 60/80 phones, it is also recommended to use pre-sorted entries, which will reduce the use of resources.

Generic Example (Standard Attributes)

OpenStage Field	LDAP Template Lables	LDAP Attribute	Example Value
Last name	`ATTRIB01`	`sn`	`Doe`
First name	`ATTRIB02`	`givenName`	`John`
Business 1	`ATTRIB03`	`telephoneNumber`	`9991234`
Business 2	`ATTRIB04`	`facsimileTelephoneNumber`	`9992345`
Mobile	`ATTRIB05`	`mobile`	`017711223344`
Private	`ATTRIB06`	`homePhone`	`441274333444`
Company	`ATTRIB07`	`o`	`Example Inc.`
Address 1	`ATTRIB08`	`departmentNumber`	`0815`
Address 2	`ATTRIB09`		
Job function	`ATTRIB10`	`title`	`Product Manager`
Email	`ATTRIB11`	`mail`	`doe@example.com`

Given "example.com" as the LDAP subtree to be searched, the LDAP template file would look like this:

```
OpenStage LDAP TEMPLATE (v.1)
SEARCHBASE="dc=example,dc=com"
ATTRIB01="sn"
ATTRIB02="givenname"
ATTRIB03="telephoneNumber"
ATTRIB04="facsimileTelephoneNumber"
ATTRIB05="mobile"
ATTRIB06="homePhone"
ATTRIB07="o"
ATTRIB08="departmentNumber"
ATTRIB09=""
```

```
ATTRIB10="title"
ATTRIB11="mail"
EOF
```

Microsoft Active Directory Specific Example

OpenStage Field	LDAP Template Attribute	LDAP Attribute	Example Value
Last name	ATTRIB01	sn	Doe
First name	ATTRIB02	givenName	John
Business 1	ATTRIB03	ipPhone	9991234
Business 2	ATTRIB04	otherTelephone	9992345
Mobile	ATTRIB05	mobile	017711223344
Private	ATTRIB06	homePhone	441274333444
Company	ATTRIB07	company	Example Inc.
Address 1	ATTRIB08	department	Administration
Address 2	ATTRIB09		
Job function	ATTRIB10	title	Product Manager
Email	ATTRIB11	mail	doe@example.com

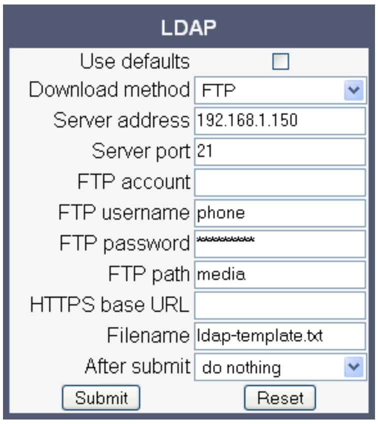
Given "example.com" as the LDAP subtree to be searched, the LDAP template file would look like this:

```
OpenStage LDAP TEMPLATE (v.1)
SEARCHBASE="dc=example,dc=com"
ATTRIB01="sn"
ATTRIB02="givenname"
ATTRIB03="ipPhone"
ATTRIB04="otherTelephone"
ATTRIB05="mobile"
ATTRIB06="homePhone"
ATTRIB07="company"
ATTRIB08="department"
ATTRIB09=""
ATTRIB10="title"
ATTRIB11="mail"
EOF
```

5.3.3 Load the LDAP Template into the Phone

When you have configured the LDAP template, you can upload it to the phone:

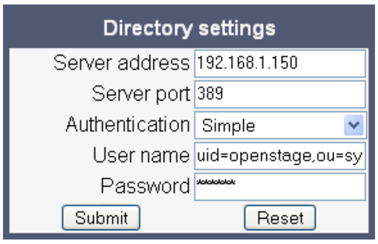
1. Save the template under a suitable name, for example, `ldap-template.txt`.
2. Copy the template file to the FTP server designated for deploying LDAP templates.
3. Upload the file using the WBM (see Section 3.14.6, “LDAP Template”), or, alternatively, the Local menu, or the DLS (see the Deployment Service Administration Manual). For an example configuration, see the following WBM screenshot (path: **File transfer** > LDAP):



5.3.4 Configure LDAP Access

To enter the access data using the WBM, take the following steps:

1. Navigate to **Local Functions** > Directory Settings.
2. Enter the following parameters:
   - **Server address** (IP address or hostname of the LDAP server)
   - **Server port** (port used by the LDAP, typically 389)
   - **Authentication** (authentication method for the connection to the LDAP server)
   - **User name** (only required if simple authentication is selected); **Password** (relating to the user name).



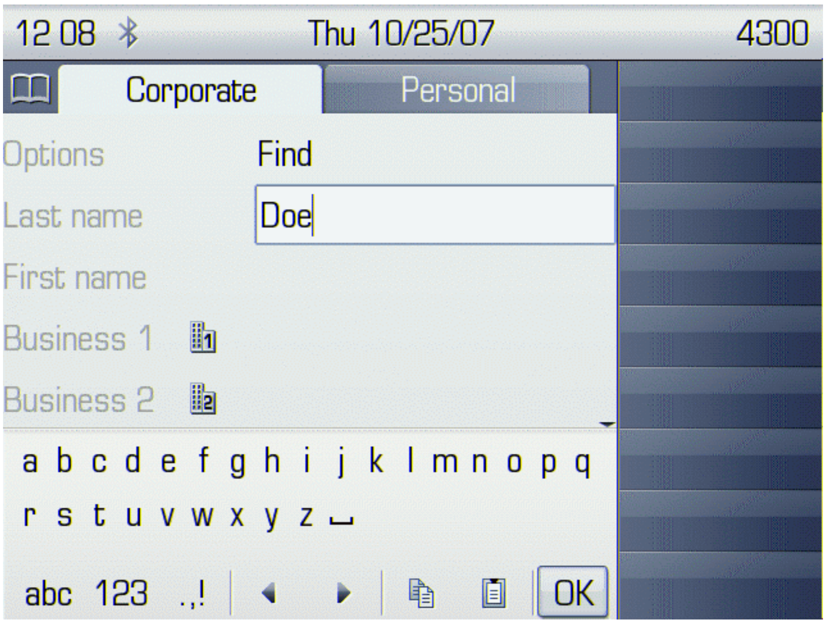
3. Press **Submit**.

5.3.5 Test

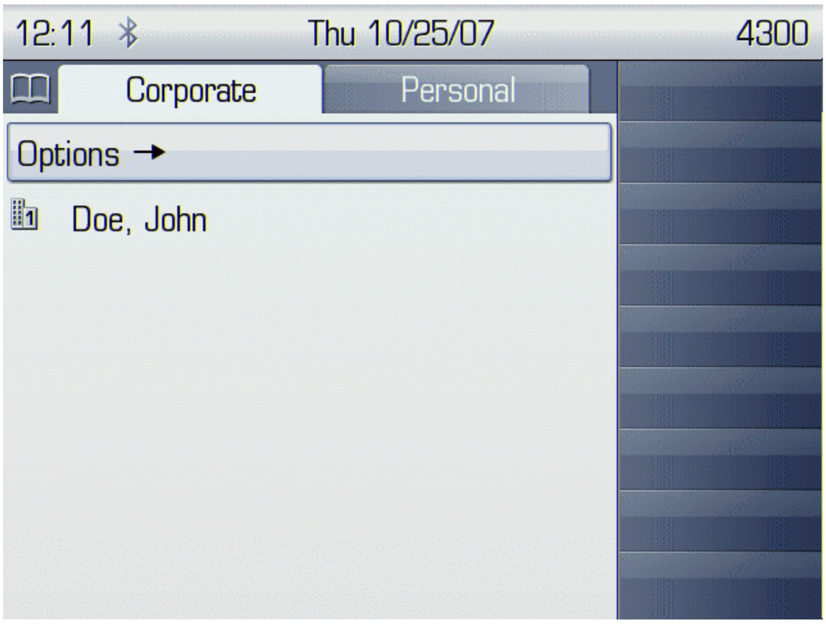
If everything went well, you can run a test query on your OpenStage phone.

1. To navigate to the phone's corporate phonebook, press the button twice.
2. Press ➔ on the TouchGuide. In the context menu, select Find by pressing .
3. In the query mask, select the entry to be searched, for instance **Last Name**. Press to open the onscreen keypad for text input.

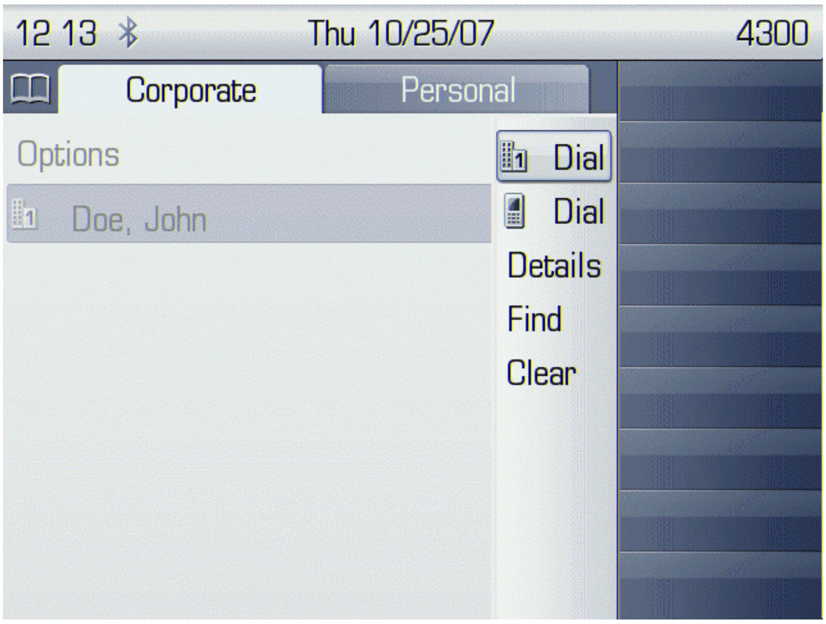
4. Enter the text to be searched. For information on using the onscreen keypad, see Section 3.1, “Access via Local Phone”, step 5.



5. Navigate to the Find option and press ⊛. If the query was successful, at least one entry will be listed in the following manner:



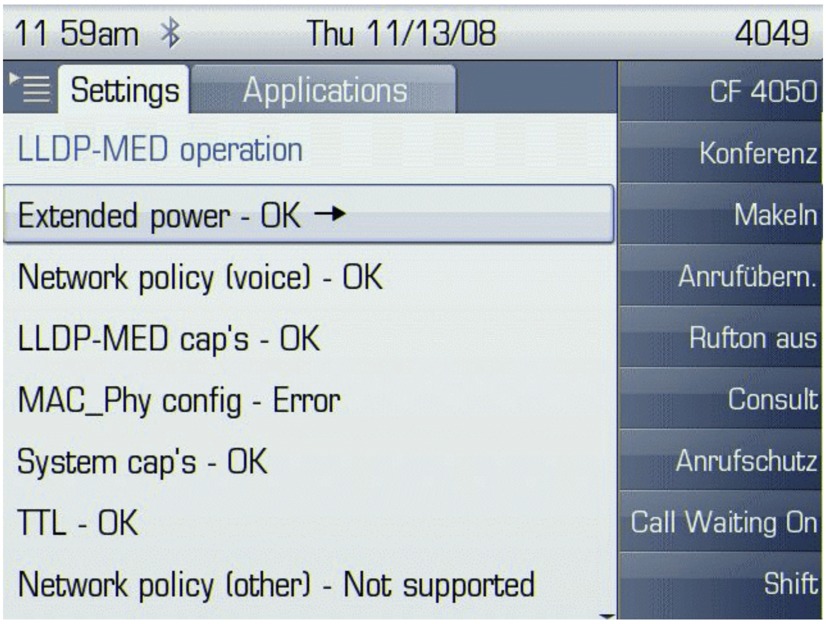
6. Navigate to the desired entry and press ➔ on the TouchGuide to open the context menu. You can select one of the following options:
   - Dial the **Business 1** number.
   - Dial the **Mobile** number.
   - Have the entry’s details, that is, all attributes displayed.
   - Start a new search.
   - Clear the list of search results.



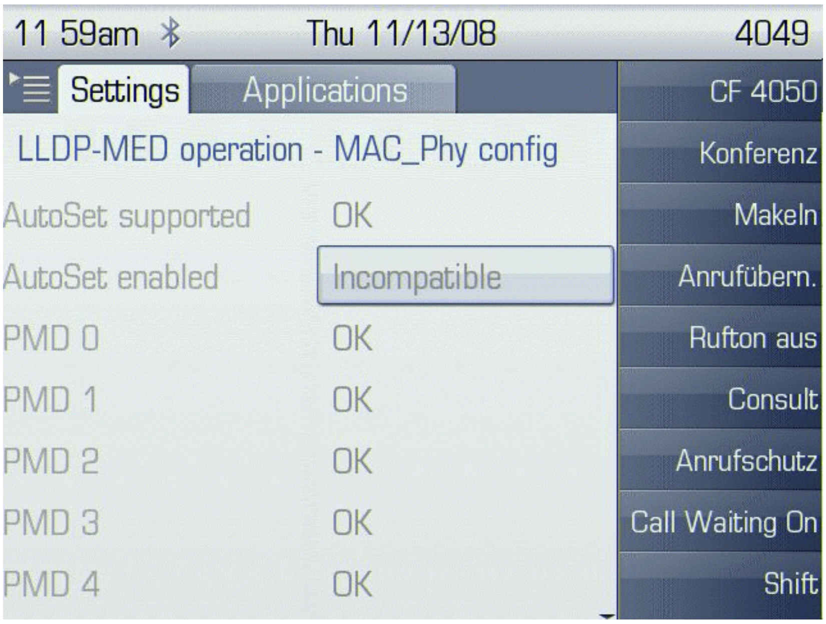
5.4 An LLDP-Med Example

The following example illustrates the mode of operation of LLDP-MED. In order to evoke a reaction from LLDP-MED, the LAN switch has been set to auto-negotiation, whereas the phone's LAN port (see Section 3.2.1, "LAN Port Settings") is set to 100Mbit/s, hence a fixed value. This configuration error is discovered by LLDP-MED. The following sceenshots from the phone's local menu will show the error messages.

This screenshot shows the LLDP-MED operation submenu (see Section 3.2.3, "LLDP-MED Operation"). Please note the status of **MAC_Phy config**.



When **MAC_Phy config is** selected, the details are displayed.


11 59am
Thu 11/13/08
4049
Settings
Applications
LLDP-MED operation - MAC_Phy config
AutoSet supported
OK
AutoSet enabled
Incompatible
PMD 0
OK
PMD 1
OK
PMD 2
OK
PMD 3
OK
PMD 4
OK
CF 4050
Konferenz
Makeln
Anrufübern.
Rufton aus
Consult
Anrufschutz
Call Waiting On
Shift

5.5 Dial Plan (V2)

5.5.1 Introduction

A dial plan is a set of rules that determine the phone's behaviour on digit entry by the user. Up to 48 rules are possible. With OpenStage phones, a dial plan rule is constructed from 9 parameters. In the following, the setup of a dial plan is explained.

The dial plan entries are preceded by a title line. This is a free format string, e. g. a descriptive name or version number, which can be used by the administrator for version control purposes.

5.5.2 Dial Plan Syntax

> The phone will not perform any checking on the title; ensuring that different dial plans are given different titles is part of the administration process.

A dial plan rule is built from the parameters described underneath.

- **Digit string**: A pattern of digits or "*", "#", or "x" characters that is to be matched for starting an action. The maximum length is 24 characters. The "x" character is a wildcard character that represents any of the other digits (it may be upper or lower case).
- **Action** : The action to be taken when the criteria are met. The following options are available:
  - "S" (Send digits): The digits entered are sent to the server when one of the following three conditions is satisfied:

    a) the maximum digits have been received, or

    b) the timer expires after the minimum digits have been received, or

    c) on receipt of the terminator after the minimum digits.
  - "C" (Check for other actions): If the the digit sequence entered by the user matches **Digit string**, **Maximum length**, and **Minimum length**, the timer starts. On timer expiry, the digit string will be sent to the server. If further digits are received before timer expiry, further entries will be checked.
    If the timer is set to 0, the dial string will be sent immediately.
    This option is used when there are more than one rules which start with the same digits.

- **Minimum length**: The dial plan rule will not initiate the sending of digits until at least this number of digits have been entered. However, the digits will be sent after the delay configured in User menu > Configuration > Outgoing calls > Autodial delay (seconds).

- **Maximum length**: Automatic sending will occur when this number of digits have been dialed. If not specified, then the digits will be sent when the timer expires, or a terminating character is entered.
- **Timer**: This indicates the timeout to be used for subsequent digit handling. If not specified, the default timer value is used (User menu > Configuration > Outgoing calls > Autodial delay (seconds)).
- **Terminating character**: A "*" or "#" character which indicates that the preceding digits should be considered complete, even though the maximum length may not be reached. However, the reach the minimum length must be reached by the string built from the digits entered and the terminating characters.
- **Special indication**:
  - "E" (Emergency): If this character is entered here, the digits matching this rule will be sent even if the phone is locked. The number will be dialed immediately even when immediate dialing is disabled, and the phone is on-hook.
  - "b" (bypass): The phone lock is bypassed. The number will be dialed immediately even when immediate dialing is disabled, if the phone is off-hook.
- **Comment**: A remark on this dial plan entry.
- **Terminator sent**: If set to true, the terminating character is sent to the server along with the dial string proper. If set to false, the dial string is sent without the terminating character.

5.5.3 How To Set Up And Deploy A Dial Plan

For creating and deploying a dial plan to an OpenStage phone, a working installation of the DLS (version V2R4 onwards) is required. This HowTo describes the creation of a simple dial plan for OpenStage phones by example. Unless otherwise stated, the actions described underneath are made in the DLS.

1. Log on to the DLS with an account that has suitable rights for deploying a dial plan. For details, please refer to the Deployment Service Administration Manual.
2. Navigate to IP Devices > IP Phone Configuration > Features > "Dialplan" tab.
3. Check **Dialplan**, if not checked already.
4. Enter a suitable **Dialplan ID**.
5. Click on to create the first dial plan rule.
6. Enter the following data:

Parameter	**Value**	**Description/Remarks**
Digit string	`3`	This rule matches numbers beginning with 3. For instance, theses might be internal numbers.
Action	`S`	When all criteria are met, the number is sent to the server.
Minimum length	4	This rule matches numbers with a length of 4 digits.
Maximum length	4	
Timer	0	The specified **Action** will take place without delay when all other criteria are met.

Summary: This rule determines that digit strings which begin with 3 and have a length of 4 digits are sent to the server without delay after the last digit has been entered.

7. Click on to create the second dial plan rule.
8. Enter the following data:

Parameter	**Value**	**Description/Remarks**
Digit string	`0`	This rule matches numbers beginning with 0. In the USA, this number calls the operator.
Action	`C`	When **Minimum length**, **Maximum length**, and the length of the digit string entered by the user match, the **Timer** is started. When it expires, the digits are sent to the server. When another digit is entered before expiry, the next dial plan entry will come into operation.
Minimum length	`1`	This rule matches numbers with a length of 1 digits.
Maximum length	`1`	
Timer	`1`	The phone waits 1 second for further digits. If the user does not enter any further digits, the action specified in **Action** is initiated.

Summary: When 0 is entered as first digit, the phone will wait 1 second. After this, 0 will be sent to the server, which might result in a call to an operator, for instance. When further digits are entered during the 1 second timespan, the next dial plan rule will take control.

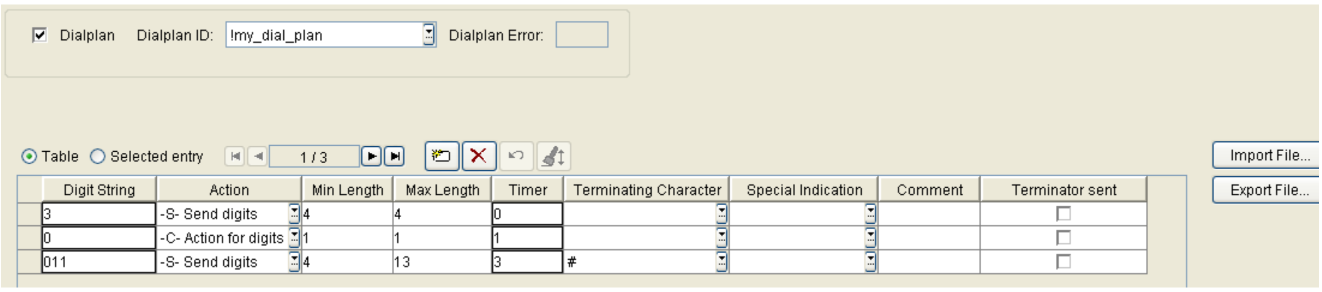
9. Click on to create the third dial plan rule.

10. Enter the following data:

Parameter	Value	Description/Remarks
Digit string	`011`	This rule matches numbers beginning with 011. In the USA, this digit string is the prefix international calls.
Action	`S`	When the entered digit string reaches the **Minimum length**, the **Timer** is started. On expiry, the digit string is sent.
Minimum length	`4`	When the length of the digit sequence entered by the user reaches this value, the **Timer** is started.
Maximum length	`13`	When the length of the digit sequence entered by the user reaches this value, the digits are sent to the server immediately. The **Timer** is overridden.
Timer	`3`	When the length of the digit sequence entered by the user reaches the **Minimum length**, the phone waits 3 seconds for further digits. If the user does not enter any further digits, the **Action** is triggered.
Terminating Character	`#`	When this character is entered, the digits are sent to the server immediately, regardless of the criteria contained in this rule.

Summary: Any numbers that start with 011 and have a length of 13 digits are sent to the server immediately. Shorter numbers with a length from 4 digits onwards are sent after a 3 seconds delay.

11. The example dial plan is completed; it should look like this:



12. You can check the dial plan using the phone's web interface; navigate to Diagnostics > Fault trace configuration > Download dial plan file.

Glossary

A

Address of Record (AoR)

A ->SIP ->URI that represents the "public address" of a SIP user resp. a phone or line. The format is similar to an E-mail address: "username@hostname". (for a definition, see RFC 3261)

ADPCM

Adaptive **D**ifferential **P**ulse **C**ode **M**odulation. A compressed encoding method for audio signals which are to be transmitted by a low bandwidth. As opposed to regular ->PCM, a sample is coded as the difference between its predicted value and its real value. As this difference is usually smaller than the real, absolute value itself, a lesser number of bits can be used to encode it.

C

CSTA

Computer **S**upported **T**elecommunications **A**pplications. An abstraction layer for telecommunications applications allowing for the interaction of ->CTI computer applications with telephony devices and networks.

CTI

Computer **T**elephony **I**ntegration. This term denotes the interaction of computer applications with telephony devices and networks.

D

DFT

Digital **F**eature **T**elephone. A phone with no line keys.

DHCP

Dynamic **H**ost **C**onfiguration **P**rotocol. Allows for the automatic configuration of network endpoints, like IP Phones and IP Clients.

DiffServ

Differentiated **Serv**ices. Specifies a layer 3 mechanism for classifying and managing network traffic and providing quality of service (->QoS) guarantees on ->IP networks. DiffServ can be used to provide low-latency, guaranteed service for e. g. voice or video communication.

DLS
The Deployment Service (DLS) is a HiPath management application for the administration of workpoints, i. e. IP Phones and IP Clients, in both HiPath- and non-HiPath networks.

DNS
Domain **N**ame **S**ystem. Performs the translation of network domain names and computer hostnames to ->IP addresses.

DTMF
Dual **T**one **M**ulti **F**requency. A means of signaling between a phone and e. g. a voicemail facility. The signals can be transmitted either in-band, i. e. within the speech band, or out-band, i. e. in a separate signaling channel.

E

EAP
Extensible **A**uthentication **P**rotocol. An authentication framework that is frequently used in WLAN networks. It is defined in RFC 3748.

F

FTP
File **T**ransfer **P**rotocol. Used for transferring files in networks, e. g., to update telephone software.

G

G.711
ITU-T standard for audio encoding, used in ISDN and ->VoIP. It requires a 64 kBit/s bandwidth.

G.722
ITU-T standard for audio encoding using split band ->ADPCM. The audio bandwidth is 7 kHz at a sampling rate of 16 kHz. There are several transfer rates ranging from 32 to 64 kBit/s, which correspond to different compression degrees. The voice quality is very good.

G.729
ITU-T standard for audio encoding with low bandwidth requirements, mostly used in VoIP. The standard bitrate is 8 kBit/s. Music or tones such as ->DTMF or fax tones cannot be transported reliably with this codec.

Gateway
Mediation components between two different network types, e. g., ->IP network and ISDN network.

GUI
: **G**raphical **U**ser **I**nterface.

H

HTTP
: **H**yper**t**ext **T**ransfer **P**rotocol. A standard protocol for data transfer in ->IP networks.

I

IP
: **I**nternet **P**rotocol. A data-oriented network layer protocol used for transferring data across a packet-switched internetwork. Within this network layer, reliability is not guaranteed.

IP address
: The unique address of a terminal device in the network. It consists of four number blocks of 0 to 255 each, separated by a point.

J

Jitter
: Latency fluctuations in the data transmission resulting in distorted sound.

L

LAN
: **L**ocal **A**rea **N**etwork. A computer network covering a local area, like an office, or group of buildings.

Layer 2
: 2nd layer (Data Link Layer) of the 7-layer OSI model for describing data transmission interfaces.

Layer 3
: 3rd layer (Network Layer) of the 7-layer OSI model for describing the data transmission interfaces.

LCD
: **L**iquid **C**rystal **D**isplay. Display of numbers, text or graphics with the help of liquid crystal technology.

LDAP
: **L**ightweight **D**irectory **A**ccess **P**rotocol. Simplified protocol for accessing standardized directory systems, e.g., a company telephone directory.

LED
Light **E**mitting **D**iode. Cold light illumination in different colours at low power consumption.

LLDP
Link **L**ayer **D**iscovery **P**rotocol (IEEE Standard 802.1AB). Provides a solution for the discovery of elements on a data network and how they are connected to each other.

M

MAC Address
Media **A**ccess **C**ontrol address. Unique 48-bit identifier attached to network adapters.

MDI-X
Media **D**ependent **I**nterface crossover (**X**). The send and receive pins are inverted. This MDI allows the connection of two endpoints without using a crossover cable. When Auto MDI-X is available, the MDI can switch between regular MDI and MDI-X automatically, depending on the connected device.

MIB
Management **I**nformation **B**ase. A type of database used to manage the devices in a communications network.

MWI
Message **W**aiting **I**ndicator. A signal, typically a LED, to notify the user that new mailbox messages have arrived.

P

PBX
Private **B**ranch E**x**change. Private telephone system that connects the internal devices to each other and to the ISDN network.

PCM
Pulse **C**ode **M**odulation. A digital representation of an analog signal, e. g. audio data, which consists of quantized samples taken in regular time intervals.

PING
Packet **In**ternet **G**ro(u)per. A program to test whether a connection can be made to a defined IP target. Data is sent to the target and returned from there during the test.

PoE
Power **o**ver **E**thernet. The IEEE 802.3af standard specifies how to supply power to compliant devices over Ethernet cabling (10/100Base-T).

Port
Ports are used in ->IP networks to permit several communication connections simultaneously. Different services often have different port numbers.

PSTN
Public **S**witched **T**elephone **N**etwork. The network of the world's public circuit-switched telephone networks.

Q

QoS
Quality **o**f **S**ervice. The term refers to control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from the application program. The OpenStage phone allows for the setting of QoS parameters on layer 2 and layer 3 (DiffServ).

QDC
QoS **D**ata **C**ollection. A HiPath IP service that is used to collect data from HiPath products in order to analyze their voice and network quality.

QCU
Quality of Service Data **C**ollection **U**nit. A service tool that collects QoS report data from IP endpoints.

QoS
Quality **o**f **S**ervice. Provides different priority to different users or data flows, or guarantee a certain level of performance to a data flow.

R

RAM
Random **A**ccess **M**emory. Memory with read / write access.

ROM
Read **O**nly **M**emory. Memory with read only access.

RTCP
Realtime **T**ransport **C**ontrol **P**rotocol. Controls the ->RTP stream and provides information about the status of the transmission, like QoS parameters.

RTP
Realtime **T**ransport **P**rotocol. This application layer protocol has been designed for audio and video communication. Typically, the underlying protocol is ->UDP.

S

SDP
Session **D**escription **P**rotocol. Describes and initiates multimedia sessions, like web conferences. The informations provided by SDP can be processed by ->SIP.

SIP

Session **I**nitiation **P**rotocol. Signaling protocol for initialising and controlling sessions, used e. g. for ->VoIP calls.

SNMP

Simple **N**etwork **M**anagement **P**rotocol. Used for monitoring, controlling, and administration of network and network devices.

SNTP

Simple **N**etwork **T**ime **P**rotocol. Used to synchronize the time of a terminal device with a timeserver.

Subnet Mask

To discern the network part from the host part of an ->IP address, a device performs an AND operation on the IP address and the network mask. The network classes A, B, and C each have a subnet mask that demasks the relevant bits: 255.0.0.0 for Class A, 255.255.0.0 for Class B and 255.255.255.0 for Class C. In a Class C network, for instance, 254 IP addresses are available.

Switch

Network device that connects multiple network segments and terminal devices. The forwarding of data packets is based on ->MAC Addresses: data targeted to a specific device is directed to the switch port that device is attached to.

T

TCP

Transfer **C**ontrol **P**rotocol. The protocol belongs to the transport layer and establishes a connection between two entities on the application layer. It guarantees reliable and in-order delivery of data from sender to receiver, as opposed to ->UDP.

TLS

Transport **L**ayer **S**ecurity. Ensures privacy between communicating applications. Typically, the server is authenticated, but mutual authentication is also possible.

U

UDP

User **D**atagram **P**rotocol. A minimal message-oriented transport layer protocol used especially in streaming media applications such as ->VoIP. Reliability and order of packet delivery are not guaranteed, as opposed to ->TCP, but ->UDP is faster and more efficient.

URI

Uniform **R**esource **I**dentifier. A compact string of characters used to identify or name a resource.

URL

Uniform **R**esource **L**ocator. A special type of ->URI which provides means of acting upon or obtaining a representation of the resource by describing its primary access mechanism or network location.

V

VLAN

Virtual **L**ocal **A**rea **N**etwork. A method of creating several independent logical networks within a physical network. For example, an existing network can be separated into a data and a voice VLAN.

VoIP

Voice **o**ver **IP**. A term for the protocols and technologies enabling the routing of voice conversations over the internet or through any other ->IP-based network

W

WAP

Wireless **A**pplication **P**rotocol. A collection of protocols and technologies aiming at enabling access to internet applications for wireless devices. WAP can also be used by the OpenStage phone.

WBM

Web **B**ased **M**anagement. A web interface which enables configuration of the device using a standard web browser.

WML

Wireless **M**arkup **L**anguage. An XML-based markup language which supports text, graphics, hyperlinks and forms on a ->WAP-browser.

WSP

Wireless **S**ession **P**rotocol. The protocol is a part of the ->WAP specification. Its task is to establish a session between the terminal device and the WAP gateway.

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