



HiPath 8000

optiPoint 410 S V7.0
optiPoint 420 S V7.0

Administrator Manual

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Safety Precautions

Important 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.



Attention

If the optiPoint 410/420 S V7.0 is supplied with power over the LAN interface → page 232, the power source must be a limited power source PowerHub compliant with IEC 60950.

- 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 commercial environment.
- 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. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his own expense.
- The IP telephone optiPoint 410/420 S V7.0 complies with the European standard EN 60 950.
- The earpiece in this telephone handset contains a magnet. To prevent injury, before each use ensure objects such as pins or staples are not stuck to the earpiece.
- There is always the danger of small objects being swallowed by young children. In the case of the optiPoint 410/420 S V7.0, this applies in particular to the connecting cord clip. Please make sure that such items are not accessible to children.
- Never allow the telephone to come into contact with staining or corrosive liquids, such as coffee, tea, juice or soft drinks.

The information provided in this document contains merely general descriptions or characteristics of performance features which in case of actual use do not always apply as described or which may change as a result of further development of the products.

An obligation to provide the respective performance features only exists if expressly agreed in the terms of contract.



Note! (for U.S.A 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.

Location of the Telephone

- The telephone should be operated in a controlled environment with an ambient temperature between 5 °C and 40 °C (41 °F and 104 °F).
- To ensure good handsfree talking quality, the area in front of the microphone (front right) should be kept clear. The optimum handsfree distance is 20 inches (50cm).
- Do not install the telephone in a room where large quantities of dust accumulate; this can considerably reduce the service life of the telephone.
- Do not expose the telephone to direct sunlight or any other source of heat, as this is liable to damage the electronic equipment and the plastic casing.
- Do not operate the telephone in damp environments such as bathrooms.

Telephone Maintenance

- Always use a damp or antistatic cloth to clean the telephone. Never use a dry cloth.
- If the telephone is very dirty, clean it with a diluted neutral cleaner containing some form of surfactant, such as a dish detergent. Afterwards, remove all traces of the cleaner with a damp cloth (using water only).
- Never use cleaners containing alcohol, cleaners that corrode plastic, or abrasive powders.

Labels



The device conforms to the EU guideline 1999/5/EG, as attested by the CE mark.



This device has been manufactured in accordance with our certified environmental management system (ISO 14001). This process ensures that energy consumption and the use of primary raw materials are kept to a minimum, thus reducing waste production.



All electrical and electronic products should be disposed of separately from the municipal waste stream via designated collection facilities appointed by the government or the local authorities.

The correct disposal and separate collection of your old appliance will help prevent potential negative consequences for the environment and human health. It is a precondition for reuse and recycling of used electrical and electronic equipment.

For more detailed information about disposal of your old appliance, please contact your city office, waste disposal service, the shop where you purchased the product or your sales representative.

The statements quoted above are only fully valid for equipment which is installed and sold in the countries of the European Union and is covered by the directive 2002/96/EC. Countries outside the European Union may have other regulations regarding the disposal of electrical and electronic equipment.

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General Information

About the Manual

The instructions within this manual will help you in administering and maintaining the **entry/economy/economy plus/standard/advance**. The instructions contain important information for safe and proper operation of the advance. 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 advance and who have a fundamental understanding of SIP. The tasks described in this guide are not intended for end users of the phones. 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.



For your own protection, please read the section dealing with safety. Follow the safety instructions carefully in order to avoid endangering yourself or other persons and to prevent damage to the unit.

These instructions are laid out in a user-oriented manner, which means that you are led through the functions of the advance step by step – from the setup, through descriptions of tools and extensions discussions of special administrative and service tasks at the end of the manual. For the users, a separate manual is provided.

Step by Step

Symbols in the Manual



Attention

This symbol indicates a hazard. Failure to follow the instructions given may result in injury or in damage to the unit.



Key information important for the proper use of the unit is marked with this symbol.



Shows administration tasks with menu paths at the advance and on the Web Interface.



Shows additional information about each parameter in the Alphabetical Reference.



Shows the related web pages.



Means that you are in the administration menu and you have already entered the correct administrator password. (Access: Phone → page 48, Web Interface → page 110).



Means that you are in the diagnostics menu and you have already entered the correct administrator password. (Access: Phone → page 49, Web Interface → page 110).



Means that you are in the **setup** menu and you have already entered the correct user password, if required (→ page 148).

Step by Step

Operating the telephone



Lift the handset (off-hook).



Replace the handset (on-hook).



Conduct a call.



Enter a telephone number or code.



Increase or reduce the value depending on the current operating mode.

optiPoint 410/420 economy/economy plus/standard/advance

Continue? >



The option appears on the screen.
Press the  key to confirm your selection.



02=System? >



Look for the select option.
Press the   keys,
until the option appears on the screen.
Then press the  key to confirm your selection.

entry

Changing and viewing the configuration data in the entry is done by entering different reference numbers.

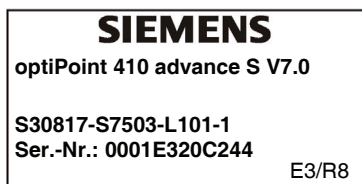
For description of viewing data values on the LEDs of the entry see → Seite 160.

Intended Use

The advance telephone is a desktop unit designed for voice transmission and for connection to LAN.

Product Identification

The identification details of your telephone are given on the nameplate containing the exact product label and serial number on the bottom of the base unit → page 20. Please have these ready whenever you call our service department in case of trouble with or defects on the unit itself.



Application Version

To find out the current application version of your advance see → page 94.

Service



The Siemens service department can only help you with problems or defects on the telephone unit itself.

Should you have any questions regarding the operation, your specialist retailer or network administrator will gladly help you.

For any questions regarding the telephone connection, please contact your network provider.

In the case of any trouble or defects on the telephone unit itself, please dial the service number of your local distributor or your local Siemens Branch office.

Installation

Prerequisites

The advance 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 (required).
- A Dynamic Host Configuration Protocol (DHCP) server (optional).
- Either a Call Control System
 - Proxy server — There must be a device running RFC 3261 SIP-compliant software.
 - Voice packet gateway (optional) — Required if your VOIP Network is connected to the Public Switched Telephone Network (PSTN).
- or a voice packet gateway if the phone is used in gateway routing mode → page 216.

Connecting to the Network



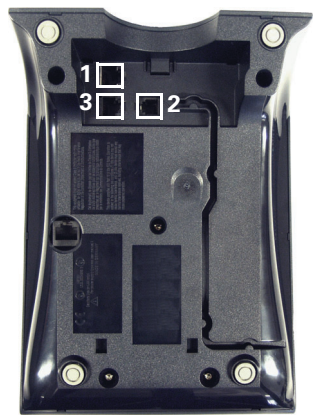
You have to connect the advance first to the LAN and then to the power supply.

The advance has two RJ-45 ports labelled "10/100 LAN" and "10/100 PC." Each port supports 10/100 Mbps half- or full-duplex connections.

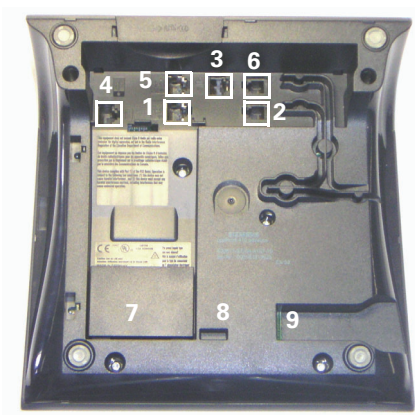
We recommend that you use the port setting "Auto" → page 194 on all ports for auto detection of transferring speed and type of connected cable (either straight-through or crossed).

Installing the Phone

Connectors on the bottom of the telephone



optiPoint 410 entry



optiPoint 410 advance

	entry S	economy S	economy plus S	standard S	advance S
1	Ethernet port for LAN connection (optional with PoL [*])				
2	Handset connector				
3	Connector for a local power supply unit (optional*)				
4	-	-	-	Module connector	
5	-	-	Ethernet port for PC		
6	-	-	Headset connector		
7	-	-	-	Adapter 1	
8	-	-	-	Adapter 2	-
9	-	-	-	-	USB-Master

^{*} Power over LAN:
If power is supplied over the LAN cable, no local power supply is required.

Starting up the optiPoint 410/420 S V7.0



The optiPoint 410/420 phone is to connect to a Switch. (The phone is working also on a Hub, but without a guarantee of quality)



The Western plugs of all cable connections must audibly snap into place.

- Plug the short end of the handset cable into the handset and the other end into the connector **2** at the bottom of the telephone and feed the cable through the guide channel in the base unit.
- Using the headset connector:
Plug the jack of the headset cable into connector **6** at the bottom of the telephone and feed the cable through the guide channel in the base unit.
- Using optiPoint modules (**4**):
Mount this device following the instructions in the installation guide (A31003-H8400-B934-*-6ZD1).
- Using adapter (**7, 8**):
Mount this device following the instructions in the installation guide (A31003-H8400-B934-*-6ZD1).
- Using an external keyboard:
Plug the keyboard cable into the USB connector **9** at the bottom of the telephone.
- Using a LAN connection to PC:
Plug the jack of the connection cable into the connector **5** at the bottom of the telephone.

Only if power **not** supported by LAN:



Use only the plug-in power supply unit fitting the optiPoint 410/420:

- GER/IM: AUL:06D1284
- GBR: AUL:06D1287
- USA: AUL:51A4827

- Plug the plug-in power supply unit into the mains.
- Plug the connector **3** at the bottom of the telephone into the plug-in power supply unit.

- Plug the jack of the LAN cable into the connector **1** at the bottom of the telephone and connect the cable with LAN.
- Feed the cables through the relief on the back of the housing and fix them by means of the cable clip.

Mini Switch

The default operation for the mini switch will be to auto negotiate transfer rate (10/100 Mb/s) and duplex method (full or half duplex) with whatever equipment is connected to the mini switch.

The software provides options to prevent auto negotiation and specify the required transfer rate and duplex mode for the LAN and PC ports
→ page 194.

In the default configuration for the mini switch 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. However if the phone is set up to manually configure the switch port settings then the cable detection mechanism is disabled, in this case care must be taken to use the correct cable type.

Depending on what has been implemented in the software IEEE802.1x (Port based network access control) packets generated by equipment connected to the PC port will be passed through the mini switch and out of the LAN port.

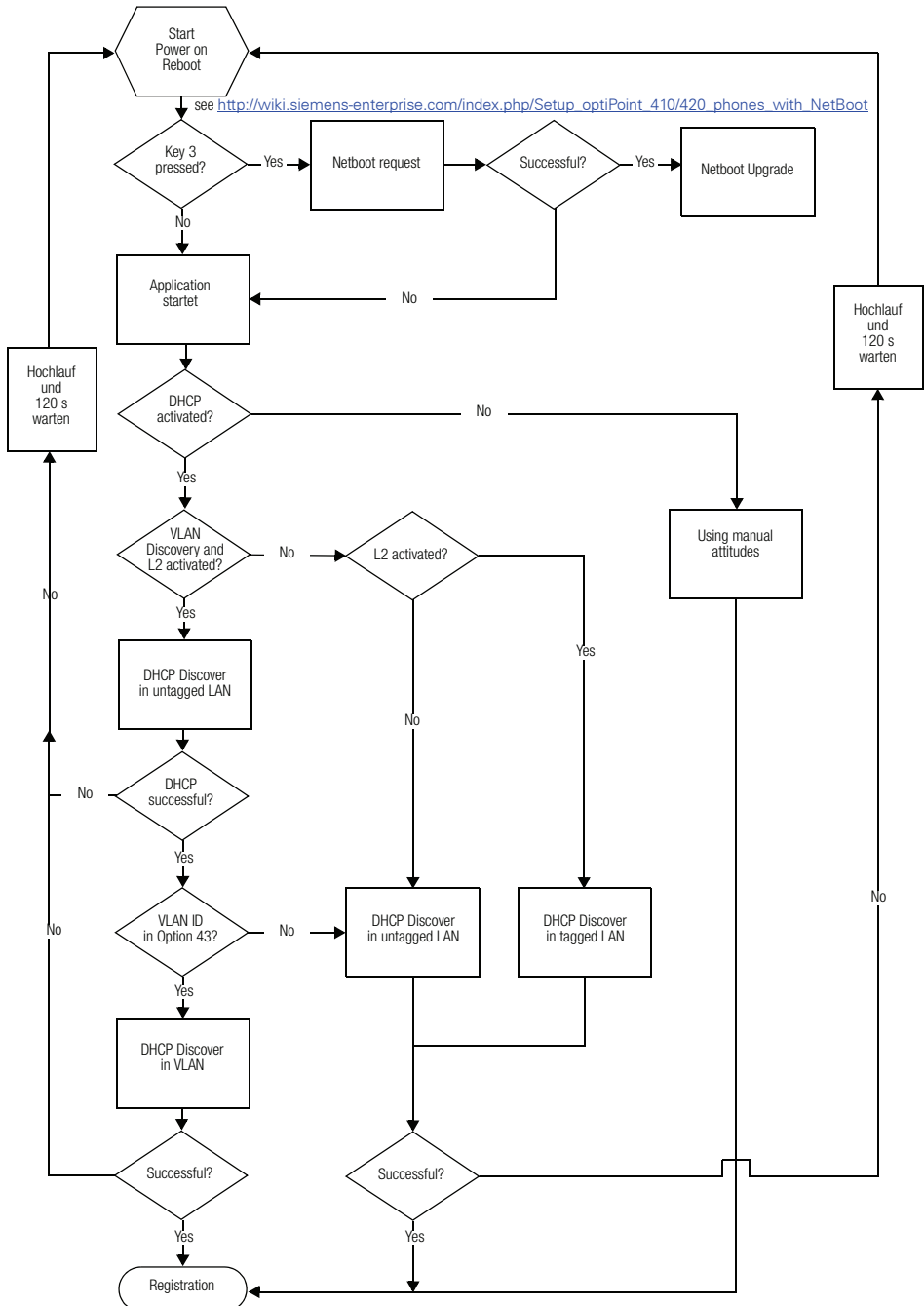
Removing the power from the phone, or a phone reset/reboot will result in the temporary loss of the network connection to the PC port. In the case of a reset/reboot this is about 5 seconds.

The Line Monitor diagnostic routine provides information on the configuration of the mini switch → page 198.

Power over LAN information

Power over LAN support is provided on the LAN port and complies with the IEEE802.3af standard. 8 wire Ethernet cables are required to use it.

Startup Procedure



Using the optiPoint 410/420 family S V7.0

Properties of the optiPoint 410/420 Telephone Models

This chapter gives you an overview of the optiPoint 410/420 telephone models and their properties

Telephone Model	Function keys	Connection of Side Car Unit	Display Lines x Characters	Headset connection	USB-Master	Mini-Switch e. g. PC-connection	Speakerphone mode	Self labeling keys
optiPoint 410 entry	8	No	-	No	No	No	No	No
optiPoint 410 economy	4/8	No	2x24	No	No	No	No	No
optiPoint 410 economy plus	4/8	No	2x24	Yes	No	Yes	No	No
optiPoint 410 standard	4/8	Yes	2x24	Yes	No	Yes	Yes	No
optiPoint 410 advance	4/15	Yes	4x24	Yes	Yes	Yes	Yes	No
optiPoint 420 economy	5/7	No	2x24	No	No	No	No	Yes
optiPoint 420 economy plus	5/7	No	2x24	Yes	No	Yes	No	Yes
optiPoint 420 standard	5/7	Yes	2x24	Yes	No	Yes	Yes	Yes
optiPoint 420 advance	5/13	Yes	4x24	Yes	Yes	Yes	Yes	Yes

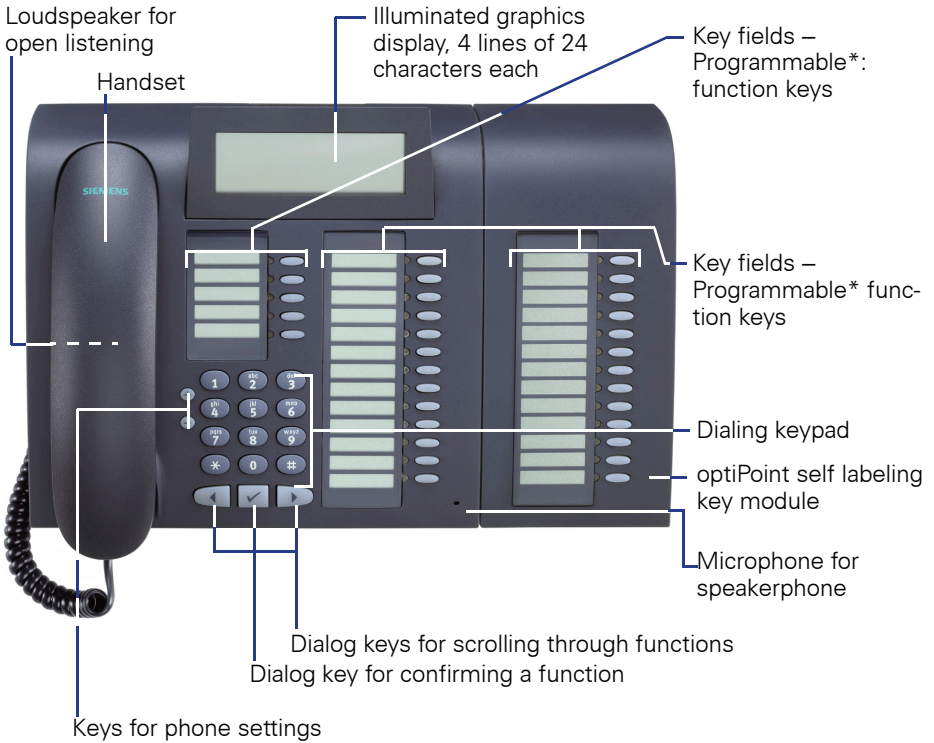
Telephone Modes

Your administrator can configure the optiPoint 410/420 economy/economy plus/standard advance S V7.0 for use as:

- A SingleLine phone with one line.
- A MultiLine phone with up to 10 lines (in relation with the SIP server).

Control panel

Example: optiPoint 420 advance



* With automatic key labelling

Example: optiPoint 410 entry



Display and Dialog Keys

The optiPoint 410/420 is provided with a four-line (advance)/two-line (economy/economy plus/standard) display. In the normal operating mode, it displays the basic menu where you make or receive telephone calls.

optiPoint 410/420 advance

4:15PM	05.03.04	Time and Date
1228		Terminal number or name
Username		Programmable Username
Menu	>	Menu

optiPoint 410/420 economy/economy plus/standard

4:15PM	05.03.04	Time and Date
1228	Username>	Terminal number or name Programmable Username

The basic menu shows in its first line the time and date, and the terminal number or name in the second line. The arrow symbol ">" on the right hand side of the last display line points to additional functions offered. The third line of the optiPoint 410/420 advance and the right corner of the second line of the optiPoint 410/420 economy/economy plus/standard is used for a free programmable Name e. g. name of the user.

If you want to navigate in menus or make settings, use the three dialog keys , ,  and function key "Stop/Escape" to navigate through the hierarchically built up menu structure. Within this structure, the third (first) line shows the currently selected menu, the last line a menu item of that menu.

optiPoint 410/420 advance

4:15PM	05.03.04	Time and Date
Aministration:		Selected Menu
01=Network?	>	Menu item

optiPoint 410/420 economy/economy plus/standard

Aministration:		Selected Menu
01=Network?	>	Menu item



You can put the advance in idle mode rapidly by lifting off and replacing the handset.

Dialling Keypad

The dialling keypad of the advance is labeled with digits, letters and some special characters. You can key in letters and special characters in the corresponding input mode by pressing the corresponding key as often as is necessary until the required letter or the required special character appears on the display.

For example, if you want to enter the letter "R", press the key "7" three times as "R" is at the third position. For the letter "U", press the key "8" twice.



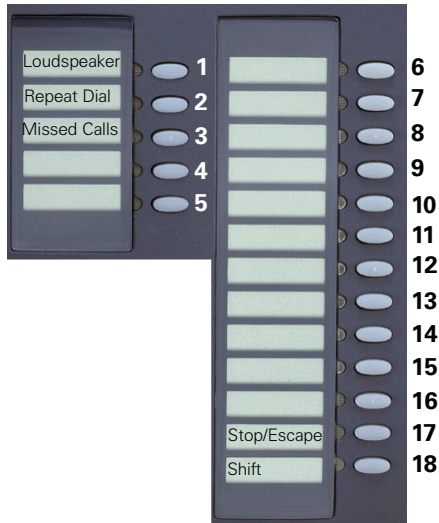
More information about text editors see → page 155.

Programmable Keys

The optiPoint 410/420 family is equipped with function keys which are user-programmable in two levels (see User Manual). The function key "Stop/Escape" should not be programmed. Five of these keys come already preassigned in the first level

The types of the optiPoint 410/420 family have various numbers of function keys:

Example optiPoint 420 advance:



Function Key	Function
1	Switches loudspeaker of the base unit on/off
2	Shows the last 20 dialled numbers, and allows selection and redial actions
3	Shows the last 20 missed calls and allows selection, edit and redial actions
17	Cancels the current action
18	Toggles between first and second key levels

Control Keys

The two control keys ⊖ and ⊕ are located on the left side of the dialling keypad. Depending on the operating mode, you can vary the following settings:

Operating mode	⊖ key	⊕ key
Receiving a call	Reduce the volume of the ringer tone	Increase the volume of the ringer tone
Handsfree talking	Reduce the volume of the loudspeaker in the base unit	Increase the volume of the loudspeaker in the base unit
Using the handset of the telephone	Reduce the volume of the handset loudspeaker	Increase the volume of the handset loudspeaker
Setting in the menu "Setup" - "Audio settings"	Adjust loudspeaker volume, ringer volume, sequence and melody of the ringer tone, handset volume and key click volume (confirmed by ☒)	
Restart and factory setting	Starts these functions	

Phone Features

Protocol support

The economy/economy plus/standard/advance supports the following protocols:

- → SIP (RFC 3261 compliance)
- → SDP
- → TCP/→ UDP
- → FTP
- → SNMP
- → SNTP
- → HTTP
- → RTP/→ RTCP
- → DNS
- → DHCP
- → EAP (802.1X)
- Quality of service in accordance with DiffServe and IEEE 802.1p/q.

Capabilities

The advance supports the following capabilities:

Network

- Power over LAN
- DHCP for automatic IP address assignment or static IP configuration
- SNTP for automatic time synchronization
- Support for VLANs
- Support for configurable Layer 2 and 3 Quality of Service

Configuration

- Language definition (5 languages)
- Country definition allowing flexible tone generation
- Feature enable/disable
- User and administrator levels (password protected)
- Upload and download of configuration files (INI file format)

Management

- Deployment service (DLS) for configuring phones
- Web interface for configuring individual phones
- BroadSoft Centralized Management for automatic configuration
- SNMP
- Flexible phone menu system

Speech

- Support for G711 (U and A Law), G723 and G729
- Configurable Jitter buffer support
- High Quality speaker phone functionality
- G711 Silence Suppression
- Audio codec G.722 offers a wider audio bandwidth resulting in major improvement in the represented speech quality.

Call Features

- Call Reject
- Call deflection
- Call forwarding (Unconditional, On Busy, On no Reply)
- Call waiting¹
- Consultation
- Unattended Transfer
- Attended Transfer (Join)
- Local Conference (3 party, G711 only)
- Do not Disturb
- Hold
- Message Waiting
- MultiLine
- Call Park

DTMF

The phone provides 2 mechanisms for transmitting → DTMF information, inband and DTMF in RTP (see RFC 2833). The phone does not support out-band DTMF through SIP messaging. There are no configuration parameters on the phone which control the use of DTMF.

A process of negotiation is used during call-setup to determine which form of DTMF signaling will be used. The phone supports send DTMF information in response to the user pressing the keys 0-9 and * and # when in a call connected state.

When a call is made from a phone it will "Offer" the remote endpoint support for DTMF in RTP (this is carried in the SDP protocol). If the far end does not "answer" that it can support DTMF in RTP then DTMF in-band will be used otherwise DTMF in RTP will be used.

When DTMF in RTP is negotiated the phone will always "offer" payload 100 to carry the DTMF events. The far end may accept and confirm this payload or it may suggest a different payload value. In this case the phone will follow that payload preferred by the far end. On an incoming call the phone will follow the payload value suggested by the far end.

The phone is not capable of retrieving or understanding DTMF in-band or DTMF in RTP information it may receive. This information is normally used by application or media servers to control feature access. If the user presses keys when in a call connected state and in-band DTMF has been negotiated he will hear the tones being sent in the speech path (handset only). If DTMF in RTP has been negotiated he will here clicks as speech packets are removed and replaced with DTMF in RTP key events.

Technical Overview

Session Initiation Protocol (SIP)

Overview

The Session Initiation Protocol (SIP) is a ASCII-based signalling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session.

Like other VoIP protocols, SIP provides signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management controls the attributes of an end-to-end call.

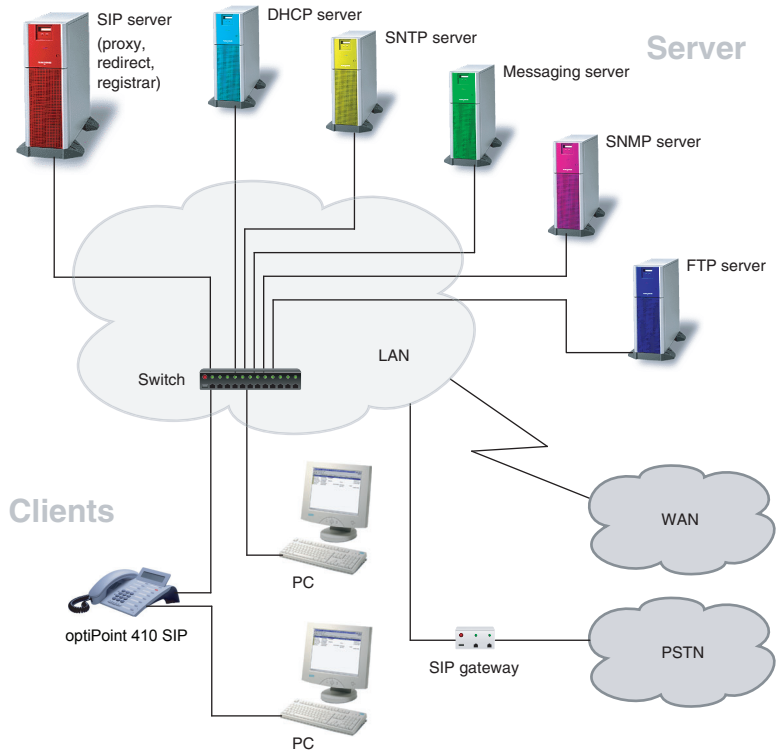
SIP was originally developed in the MMUSIC group within the IETF (Internet Engineering Task Force), it has been published since February 1999 as RFC 2543. The SIP working group is continuing to enhance the protocol and published version 2 as RFC 3261 in 2002.

SIP Functions

Systems which use SIP are able to provide the following:

- The location of the target endpoint — SIP supports address resolution, name mapping, and call redirection.
- The media capabilities of the target endpoint — Via Session Description Protocol (SDP), SIP determines the lowest level of common services between endpoints. Conferences are established using only the media capabilities that can be supported by all endpoints.
- A session between the originating and target endpoint — If the call can be completed, SIP establishes a session between the endpoints. SIP also supports mid-call changes such as adding another endpoint to the conference and changing media characteristic or codec.

Components in a SIP system



Configuration example with additional components → page 36

SIP Components

SIP is a peer-to-peer protocol. The peers in a session are called user agents (UAs).

SIP Clients

SIP clients include the following:

- Telephones — Act as UAS and UAC. The advance can initiate SIP requests and respond to requests.
- Gateways — Provide call control. Gateways provide many services, the most common being translation between SIP conferencing endpoints of transmission format, communications procedures, and codecs. Other functions include call setup and clearing on both the LAN side and the switched-circuit network side.

SIP Servers

SIP servers include the following:

- Proxy servers — Receive SIP requests from a client and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
- Registrar servers — Process requests from UACs for registration of their current location. Registrar servers are often colocated with redirect or proxy servers.

Additional Components

- DHCP server
Distributes IP data and further information in a network automatically (list of distributed information → page 175).
- SNTP server
Provides time, date, daylight saving and timezone information.
- Messaging server
For recording and reading messages.
- SNMP server
Logging and maintenance of network components.
- FTP server
For up- and download of files from and to the phone. These include configuration files and music files.
- PC with internet browser
Enables the administration of the phone by using a Web Client such as Internet Explorer.

Registration

 Note that registration only occurs when the SIP Routing mode → page 216 is set to "Server".

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 unauthenticated registration the following parameters must be set on the phone:

- Terminal number → page 224 or Terminal name → page 223.
- SIP Routing → page 216 set to "Server".
- SIP Server/Registrar address → page 214 configured (IP address or host name).

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

The phone supports the Digest authentication scheme and requires the following parameters to be configured in addition to those for unauthenticated registration:

- SIP user ID → page 218.
- SIP Password → page 215.
- SIP Realm (optional) → page 215.

For authentication to work the server must have created an account for the user with matching user ID, password and Realm parameters.

 Note 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 eg INVITE.

Below are some specific details relating to SIP registration configuration parameters found on the phone:

- Terminal Number → page 224
- Terminal Name → page 223
- Register by Name → page 210
- SIP Routing → page 216
- SIP Registrar (SIP Addresses) → page 214
- SIP Realm → page 215
- Registration Timer → page 211

If registration has not succeeded at startup or registration fails after having been previously successfully registered the phone will try to re-register every 60 seconds. This is not configurable.

IP Network Protocols

DHCP

The Phone contains a Dynamic Host Configuration Protocol (DHCP) client that supports automatic configuration of various parameters.

If DHCP is enabled in the phone, the phone will try to obtain the following options that are essential for the configuration of its Ethernet interface automatically from a DHCP server:

- Terminal IP Address
- Terminal Mask (Network Mask)

When the telephone requests its IP address, it sends – apart from other information – its default host name to the DHCP server. The default host name consists of telephone model + type + MAC address (e.g. OP410STD0001e325a845).

The available telephone models + type are listed below:

- OP410ENT
- OP410ECO
- OP410STD
- OP410ADV
- OP420ECO
- OP420STD
- OP420ADV
- OP410ECP
- OP420ECP

The DHCP server forwards this name to the DNS server, together with the IP address assigned.

If the phone fails to configure its Ethernet interface from a DHCP server it will eventually time-out indicating no DHCP server found and imminent restart.

Other configuration options that the phone attempts to retrieve from the DHCP server include:

- 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 DNS 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 → page 39)

These parameters are not essential to basic network configuration the operation of the phone and if not obtained will not cause a reboot. The phone assumes these parameters are not provided by DHCP until they are returned from the DHCP server. If these parameters are returned from the DHCP server they are used and not editable in the various phone menus. If these parameters cannot be obtained from the DHCP server the manually configured settings for these options are used.



SIP Server option 120:

Because the phone only reads the first name/IP address supplied in option 120, the maximum length of the contents has been limited to 50 octets. Please be aware of this when you are using it.

VLAN discovery per DHCP

An additional use for DHCP in the phone is the → VLAN discovery per DHCP feature. This allows the phone to discover its VLAN from a DHCP server in the untagged LAN. After discovering its VLAN the phone starts its standard DHCP process within that discovered VLAN to configure itself from the DHCP within that VLAN.

DHCP Support Explanation of Option 43

As no DHCP option exists for the exchange of VLAN information over DHCP, the Vendor Specific Information option (43) shall be used to encapsulate VLAN and download configuration. The following diagram illustrates the format of the Vendor Specific Option.

Byte #	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25
Description	Code	Length	Code	Length	Option 1 data - "Siemens" identifier										Code	Length	Option 2 data - to be allocated VID					Padding	Padding	Padding	Padding	Code
Value	43	20	1	10	'S'	'i'	'e'	'm'	'e'	'n'	's'	0	0	0	2	4	0	0	VID HI	VID LO	0	0	0	0	0	255

Byte 1 contains the tag "43", option 1 data contains the "Siemens" identifier and the VLAN ID is contained in option 2. Five **Padding Bytes** starting in Byte 20 and the **terminate option 255** in byte 25 complete the option frame.

If you have to specify a configuration download server for configuration update (see → page 88) so you have to add the values of the server. You can place the new option before or after the VLAN information but the five **Padding Bytes** (Value 0) and the **terminate option 255** have to close the option frame.

Example:

Code: 3

length: 28

data: sdlp://dls.siemens.com:18443

	20	21	22	23	24	25	26	27	28	29	30	31	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47	48	49	50	51	52	53	54	55
	Code	Length	Option 1 data - "dls server address"																											Padding	Padding	Padding	Padding	Code		
	3	28	's'	'd'	'l'	'p'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	'/'	255	

The five **Padding Bytes** and the **terminate option 255** now completes the option frame in byte 48.

Using Vendor Classes

A "Vendor Class" is used to make sure that vendor-related information is only sent to the telephones (instead of sending it to all other terminal devices as well). By using a vendor class, vendor information elements for each vendor class can be sent to all devices of this vendor class. The vendor class name is either "OptiPoint" and/or "OptilpPhone". optiPoint telephones send their vendor class name using the option 60 to the DHCP server whenever they request data from the DHCP server.

If the option "VLAN Discovery" is set to DHCP on the telephone, the telephone registers using the vendor class name "optiPoint" during the initial boot process, and then using the vendor class name "OptilpPhone". On the DHCP server, you can therefore use the vendor class name "optiPoint" to assign the VLAN, and use the vendor class name "OptilpPhone" to assign the DLS server address or the name. This enables a more specific configuration of the DHCP server.



The Authentication is done via digital Certificates. For detailed information refer to the IEEE 802.1x Configuration Management Administration Guide.

DNS

The phone uses → DNS services provided the phones operating system to perform the following:

- Resolve the IP address of servers that have been configured as names (DNS A,AAAA records).
- Resolve the IP address of the domain part of users called by URL (DNS A,AAAA records).
- Identify the location of servers and provide for failover and load balancing (DNS A, AAAA and DNS SRV).

For DNS services to be used on the phone the following must be configured either manually or provided by DHCP:

- DNS Domain Name → page 182
- Prim DNS IP addr → page 207
- Sec DNS IP addr → page 212



If a DNS Domain Name and one or both of the DNS IP addresses have been configured, then additionally the following host names can be entered alternatively to the corresponding IP addresses:

- SIP addresses → page 214
(server, gateway, registrar)
- Download server IP address or DNS name → page 183
- SNMP Trap IP address or DNS name → page 219
- SNMP server address or DNS name → page 219
- Message Waiting IP address → page 203
- Ping → page 207

The Primary DNS server IP address must be configured if DNS is to be used however the secondary server is optional. The purpose of the secondary server is to allow a backup DNS server to be used in the system environment to increase system availability and reliability.

The phone can contact several types of server for different reasons (SNTP, SNMP, SIP server etc). If a server has been configured by name and not IP address the phone will provide a DNS lookup to resolve the name to an IP address when the server needs to be contacted. To optimize network traffic performance the phone caches the result of the normal A and AAAA record lookups and will not re-issue a request to the DNS server to resolve that address again until the Time To Live value from the previous lookup has expired. This mechanism does not currently apply to DNS SRV lookups.

If a secondary DNS server has been configured and the primary fails to respond to a request, that request will be re-issued to the secondary DNS server.

DNS SRV

The phone supports the use of DNS SRV record lookups to allow SIP servers to be located. This mechanism is described in detail in RFC 3263 - Locating SIP Servers.

If the location to which a SIP message is to be sent is defined as a name as opposed to an IP address a DNS SRV lookup will be performed. An example query being:

`_sip._tcp.example.com.`

This indicates a query for a SIP server supporting the TCP transport protocol. The transport used in this query is determined by the SIP transport menu setting → page 217. The DNS server may return an IP address for the requested SIP server or may return a single name or list of names which require further A or AAAA record lookups to determine an IP address. The response to a DNS SRV query will also contain information regarding the Time To Live for the information returned, the port address to which requests should be sent and weighting information relating to load balancing of requests.

DNS SRV and failover

Lists of candidate SIP server names are often returned in response to DNS SRV queries to allow failover mechanisms to be implemented which increase overall system availability.

If the phone sends a request to the first address in the list but fails to receive a response (currently the failover time is 6 seconds, this is not configurable) the address is placed in a "penalty box" which means that it will not be tried again until a specific time interval has past (currently pre-set to 1 minute). The request is sent to the next SIP server in the list and the process continues. The penalty box mechanism ensures that the responsiveness of the phone is maintained by not continually retrying SIP servers that are failing to respond.

For example the request to the first SIP server in the list fails, a call to another user hosted on the SIP server will result in the user experiencing a 6 second delay before the failover to the secondary occurs. This means the "Calling.." status will remain on the display until the second SIP server is contacted. All subsequent messages for this call will go to the second SIP server until the first SIP server is removed from the penalty box and it can be tried once again to see if it is back in service. Note this mechanism is independent of call setup. The first SIP server will not be retried necessarily when the next call is established, only when it is removed from the penalty box.

SNTP

The phone support → SNTP. The SNTP server address can be supplied via DHCP or manually configured → page 219. If the SNTP server address is available the server will be queried for the time. If a server address is not configured the phone will look for SNTP broadcasts and setup the time accordingly, if these are received any manually configured time and date information would be over written.

SNMP

The phone provides → SNMP which allows network related information to be browsed (MIB II support). Standard SNMP browsers are sufficient for this purpose.

IP Network Configuration

Routing

The phone allows a default route to be configured to allow access to Servers on a different subnet to the one in which the phone resides. In addition it is possible to configure 2 additional routes. Each route consists of a IP address, gateway and mask.

Virtual LAN (VLAN)

VLAN or virtual LAN 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 of the data and voice networks being partitioned 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.

VLAN is a layer 2 (Physical Layer) protocol. In the case of Ethernet the physical header is extended allowing endpoints to be not only be addressed via MAC address, but also VLAN ID → page 203. Ethernet VLANs support the partitioning of a physical LAN into up to 4095 virtual LANs.

To implement a voice network based on VLANs requires the network infrastructure (the switch fabric) to support VLANs at layer 2. Dependant on the overall architecture it may or may not be necessary for the endpoint (phone) to support layer 2 VLAN.

The ports of the network switches in the switch fabric can be logically grouped as ports belonging to particular VLAN. The switch only forwards traffic to a particular port if that port is a member of the VLAN that the traffic is allocated to. In this way an endpoint connected to a particular port on the switch is automatically a member of that VLAN without being a VLAN aware device; the switch ensures the endpoint only receives traffic for that VLAN and ensures traffic from the endpoint is only forwarded to ports that are configured to be in the same VLAN. This is known as port based VLAN in the switch world.

When multiple endpoints are connected to a single network switch port and these endpoints belong to multiple VLANs then a different approach is needed to implement a VLAN based topology. A typical example of this is a phone with a PC connected behind the phone. The phone would be a member of the Voice VLAN and the PC a member of the data VLAN. In this case the Phone must be configured as a VLAN aware endpoint.

VLAN support

The phone can be configured as a → VLAN aware endpoint by enabling VLAN support via the configuration menus. The following VLAN related configurations can be achieved:

- Manually configured L2 VLAN only.
- Manually configured L2 VLAN and QoS.
- Automatically discovered VLAN and manual QoS.

To configure manual L2 VLAN only the phone must be configured at manual VLAN ID between 1 and 4095. Vlan discovery mode must be set to manual.

To configure manual L2 VLAN and QoS the phone must be configured as QoS layer 2 on and a manual VLAN id between 1 and 4095. Vlan discovery mode should be set to manual and QoS layer 2 and 3 values should be configured as described in the QoS section below.

If you mis-configure a phone to an incorrect VLAN the phone will behave as though it is not configured for and possibly not connected 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.

To automatically discover a VLAN ID using DHCP the phone must be configured as DHCP enabled and VLAN discovery mode set to DHCP. If QoS is required this can be turned on and QoS layer 2 and 3 values should be configured as described in the QoS section below.

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. In these circumstances network routing will probably not be correct; the SIP server may or may not be reached. The default setting for the phone is to try and perform VLAN discovery.

DLS

The Deployment Service (DLS) is a HiPath Management application for administering workpoints (optiPoint telephones and optiClient installations) in HiPath 8000 networks. It has a Java-supported, web-based unuser interface, which runs on an internet browser. Amongst the most important features are: security (e.g. PSS generation and distribution within an → SRTP security domain), mobility for optiPoint Sip phones, software deployment, plug&play support, as well as error and activity logging.



For detailed informationen about DLS refer to the HiPath Deployment Service Administration Manual.



The Authentication is done via digital Certificates. For detailed informationen refer to the IEEE 802.1x Configuration Management Administration Manual.

Quality of Service (QoS)

Modern networks can be used to provide various Qualities of Service to network endpoints based upon the importance of the endpoint and its generated traffic. Quality of Service is a term used to describe this categorisation of network traffic in networks based on the importance of the data and the treatment of that prioritized traffic.

A typical example of use of QoS in a network is that of an IP Phone. Telephone Voice traffic is more important than for HTTP related traffic for a PC. Prioritization of voice traffic over that of the HTTP traffic means that during periods of heavy network load that voice service is maintained, whereas the response times for a user's Web Browser will degrade and possibly stop working.

Quality of Service can be supported in networks at both Layer 2 and Layer 3. At Ethernet layer 2 the MAC header is extended to provide VLAN information and Quality of Service priorities. Ethernet layer 2 allows for prioritisation of traffic from 0 (lowest) to 7 (highest). At the layer 3, the IP layer traffic can be prioritized using information embedded in the IP Type of service (DiffServ) field that allows for 64 levels of prioritization.

To utilize Quality of Service features the network infrastructure (switch fabric) must support prioritized delivery of traffic based on layer 3 and/or layer 2.

Secure Payload

optiPoint 410/420 advance S V7.0 telephones enable you to establish a secure telephone connection, provided that the recipient's telephone is also capable of this. Voice transmission is encrypted and subsequently decrypted by the called party's telephone and vice versa. Even the signaling for the call-setup and the exchange of the encryption data is carried out via a secure connection. The telephones have to have a valid registration at an SIP server via.

Administration Interfaces

You can configure the optiPoint 410/420 by using any of the following methods.

Phone Menu

For direct configuration of an optiPoint 410/420 entry economy/economy plus/standard/advance S V7.0. Access to the phone is required.

The dialling-keypad, the two-line display and the handset of the phone supports you in

configuring the optiPoint 410/420.

Menu overview see → page 148.



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

Web Interface

For remote configuration of individual IP phones in your network. Direct access to the phone is not required.

Menu overview see → page 110.



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

The remote configuration is not applicable while the phone is not in idle mode.

Basic Administration

The phone is factory preconfigured to allow for a minimum of configuration activities required on the unit itself. A number of parameters can be configured centrally by using a DHCP server. When the phone is connected to the network it will react as follows:

- If your network **use a DHCP server**, the telephone will try to get its IP Address, IP Address Mask, SIP Addresses (server, gateway, registrar), SNTP Server Address Configuration Download Server Address and Time Offset from the DHCP server (completely list see → page 38). In this case the telephone will boot with the IP address and will get the exact time from the configured SNTP server. You only have to configure the Terminal number, SIP user ID and password and the language → page 50.
- If the **DHCP server is not available** or configured to provide these parameters, the telephone will become idle and has to be manually configured → page 51.

Access to the Phone Administrator and Diagnostics Menu

Administrator menu

The following steps describe the access to the optiPoint 410/420 economy/economy plus/standard/advance administration menu, starting from idle state.

1. Select the menu „05=Setup“ by pressing the key  several times.

Menu:
05=Setup? >

Confirm with the key .

2. Select the administrator menu by pressing the key .

Enter admin password

3. Enter the administrator password (default: „123456“). The entry is masked by asterisks „*“.

Enter admin password

Confirm with the key .

4. Now the first menu item in the administrator menu is shown:

Administration
01=Network>

You can change the sub-menu by pressing the keys   or you can open the menu „01=Network“ by pressing the key .

Diagnostics menu

The following steps describe the access to the optiPoint 410/420 economy/economy plus/standard/advance diagnostics menu, starting from idle state.

1. Select the menu „05=Setup“ by pressing the key  several times.

Menu:
05=Setup? 

Confirm with the key .

2. Select the diagnostics menu by pressing the key .

Enter admin password

3. Enter the administrator password (default: „123456“).
The entry is masked by asterisks „*“.

Enter admin password

Confirm with the key .

4. Now the first menu item in the diagnostics menu is shown:

Diagnostics
01=Display test?>

You can change the sub-menu by pressing the keys   or you can open the menu „01=Display test“ by pressing the key .

Basic Configuration

optiPoint 410/420 economy/economy plus/standard/advance

Configuration using DHCP Server

The optiPoint 410/420 is factory-configured to have an IP address automatically assigned to it by the DHCP server as soon as it's connected to LAN.



The Authentication is done via digital Certificates. For detailed information refer to the IEEE 802.1x Configuration Management Administration Manual.

If your network use a DHCP server, the telephone will try to get its IP Address, IP Address Mask, SIP Addresses (server, gateway, registrar), SNTP Server Address Configuration Download Server Address and Time Offset from the DHCP server (completely list see → page 38).

In this case the following parameters have to be configured:

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Terminal Number  → page 224	 - 02=System - 01=Terminal Number (change terminal number)	 - SIP environment - Terminal number (enter terminal number)
SIP User ID  → page 218	 - 02=System - 10=SIP User Id (change ID)	 - SIP environment - SIP user ID (enter ID)
SIP Password  → page 215	 - 02=System - 11=SIP Password (change password)	 - SIP environment - New/Confirm SIP password (enter/re-enter password)
Language	 - 2=Configuration - 02=Language (select language)	(WBM user interface)

Manual Configuration

If your network does not use a DHCP server, you must disable the DHCP IP assignment manually and specify the phone's IP address and subnet mask and the network gateway IP address (default route) for the phone.

In this case the following parameters have to be configured:

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
DHCP IP Assignment  → page 175	 - 01=Network - 01=DHCP IP assign (switch off)	(not applicable ¹)
Terminal IP Address  → page 223	 - 01=Network - 02=Terminal IP addr. (change IP address)	(not applicable [*])
Terminal Mask  → page 223	 - 01=Network - 03=Terminal mask (change terminal mask)	(not applicable [*])
Default Route  → page 174	 - 01=Network - 04=Default route (change default route)	(not applicable [*])

¹ as long as no IP connection exists

If you want to use the Web Interface for further administration (phone configuration by PC) you have to reboot the phone now.

After reboot

After reboot the following parameters have to be configured:

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Terminal Number  → page 224	 - 02=System - 01=Terminal Number (change terminal number)	 - SIP environment - Terminal number (enter terminal number)
SIP User ID  → page 218	 - 02=System - 10=SIP User Id (change ID)	 - SIP environment - SIP user ID (enter ID)
SIP Password  → page 215	 - 02=System - 11=SIP Password (change password)	 - SIP environment - New/Confirm SIP password (enter/re-enter password)
SIP Server Address  → page 214	 - 02=System - 07=SIP Addresses - 1=SIP Server (change IP address ¹)	 - SIP environment - Server IP address (enter IP address*)
SIP Registrar Address  → page 214	 - 02=System - 07=SIP Addresses - 2=SIP Registrar (change IP address*)	 - SIP environment - Registrar IP address (enter IP address*)
SIP Routing  → page 216	 - 02=System - 04=SIP Routing (select routing)	 - SIP environment - SIP routing (select routing)
Language	 - 2=Configuration - 02=Language (select language)	(not applicable)

¹ or host name (if DNS is applicable → page 41)

Step by Step

optiPoint 410 entry

Configuration using DHCP Server

The optiPoint 410/420 is factory-configured to have an IP address automatically assigned to it by the DHCP server as soon as it's connected to LAN.

 The Authentication is done via digital Certificates. For detailed informationen refer to the IEEE 802.1x Configuration Management Administration Manual.

If your network use a DHCP server, the telephone will try to get its IP Address, IP Address Mask, SIP Address-es (server, gateway, registrar), SNTP Server Address Configuration Download Server Address and Time Off-set from the DHCP server (completely list see → page 38).

In this case the following parameters have to be config-ured using the **Web Interface Administrator Menu**.

Parameter	Web Interface path (Menu → page 110)
Terminal Number  → page 224	 - SIP environment - Terminal number (enter terminal number)
SIP User ID  → page 218	 - SIP environment - SIP user ID (enter ID)
SIP Password  → page 215	 - SIP environment - New/Confirm SIP password (enter/re-enter password)
Language	WBM user interface

Step by Step

Manual Configuration

If your network does not use a DHCP server, you must disable the DHCP IP assignment manually and specify the phone's IP address and subnet mask and the network gateway IP address (default route) for the phone.

Entering the administration area



Press the keys **simultaneously**

or



Press the keys **successively**.



Enter admin password
(default: **123456**).



Terminate the operation.

You are now in the **Administration Area** (all LEDs flash).

The top two function keys take over the following functions in this area:

1st Function key



Press 1st function key to make settings.

2nd Function key



Press 2nd function key to view settings. (ASCII results codes see → Seite 160)

Configure basics



Press 1st function key.



Enter code



Switch the DHCP IP assign **off**.



Terminate the operation.



Press 1st function key.



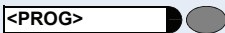
Enter code.



Enter **IP address** of the **optiPoint 410 entry**
(to edit see → Seite 160).



Terminate the operation.



Press 1st function key.

Step by Step

0 2

Enter code.



Enter **terminal mask** of the **optiPoint 410 entry** (to edit → Seite 160).



Terminate the operation.

<PROG>



Press 1st function key.

0 3

Enter code.



Enter the **default Route** of the **optiPoint 410 entry** (to edit see → Seite 160).



Terminate the operation.

Only if you are working in a Virtual LAN (VLAN):

<PROG>



Press 1st function key.

2 5 0

Input the code to define the **manual configuration** of the VLAN Discovery Mode.



Terminate the operation.

<PROG>



Press 1st function key.

2 4

Enter code.



Enter (**0 - 4095**) for the **Virtual LAN ID**.



Terminate the operation.

Confirm your entries and start the telephone:

<PROG>



Press key.

9 9

Enter the code.



Confirm the entry.

After the start, the telephone is silent and you can make the other settings.

Other settings

All other settings of your optiPoint 410 entry must be made through the "**Web-based Management Tool**" → page 109.

After **reboot** the following parameters have to be configured:

Parameter	Web Interface path (Menu → page 110)
Terminal Number  → page 224	 - SIP environment - Terminal number (enter terminal number)
SIP User ID  → page 218	 - SIP environment - SIP user ID (enter ID)
SIP Password  → page 215	 - SIP environment - New/Confirm SIP password (enter/re-enter password)
SIP Server Address  → page 214	 - SIP environment - Server IP address (enter IP address*)
SIP Registrar Address  → page 214	 - SIP environment - Registrar IP address (enter IP address*)
SIP Routing  → page 216	 - SIP environment - SIP routing (select routing)

Extended Administration

Configure Network Parameters

To access a SIP server as an IP client some network related information have to be configured.



Depending on the SIP network environment different changes are necessary → page 48.

Network Addresses

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
DHCP IP Assignment → page 175	 - 01=Network - 01=DHCP IP assign (switch on/off)	 - Network IP and routing - DHCP (activate/deactivate checkbox)
Terminal IP Address → page 223	 - 01=Network - 02=Terminal IP addr. (change IP address)	 - Network IP and routing - Terminal IP address (enter IP address)
Terminal Mask → page 223	 - 01=Network - 03=Terminal mask (change terminal mask)	 - Network IP and routing - Terminal mask (enter terminal mask)
Default Route (Gateway) → page 174	 - 01=Network - 04=Default route (change default route)	 - Network IP and routing - Default gateway (enter gateway address)
IP Route 1/2 (IP address/ Gateway/Mask) → page 191	 - 01=Network - 05=IP routing - 1=Route 1 or - 2=Route 2 - 1=IP address 1/2 (change IP address) - 2=Gateway 1/2 (change IP address) - 3=Mask 1/2 (change subnet mask)	 - Network IP and routing - IP routing (Route 1/2) - Route (enter IP address) - Gateway (enter IP address) - Mask (enter subnet mask)
DNS Domain Name → page 182	 - 01=Network - 07=DNS Domain name (change domain name)	 - Network IP and routing - Domain name (enter domain name)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Primary DNS IP Address  → page 207	 - 01=Network - 08=Prim DNS IP addr (change IP address)	 - Network IP and routing - Primary DNS IP address (enter IP address)
Secondary DNS IP Address  → page 212	 - 01=Network - 09=Sec DNS IP addr (change IP address)	 - Network IP and routing - Secondary DNS IP address (enter IP address)
User specific web ad	 - 01=Network - 12=User specific web ad (switch on/off)	(not applicable)
Terminal hostname  → page 221	 - 01=Network - 13=Terminal Hostname (change name)	 - Network IP and routing - Terminal hostname (change name)
Use dynamic hostname  → page 227	 - 01=Network - 14=Use dynamic hostname (mark to enable/disable))	 - Network IP and routing - Use dynamic hostname concept (mark to enable/disable))

 Changing either the DHCP IP assignment or the Terminal IP address will take effect as soon as the optiPoint 410/420 is restarted.

Quality of Service (QoS)

By changing the Quality of Service parameter you can affect the speech quality results. Further speech quality parameters see → page 93.

QoS Configuration

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
QoS Mode  → page 208	 - 01=Network - 06=QoS - 2=QoS L2/L3 - 1=L2Off/L3Off or - 2=L2Off/L3On or - 3=L2On/L3On	 - Quality of Service - Required (select QoS mode)
Layer 3 Voice (only if L3On)  → page 193	 - 01=Network - 06=QoS - 3=L3 Voice (select value)	 - Quality of Service - Layer 3 Voice (select Layer 3 Voice value)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Layer 3 Signalling (only if L3On)  → page 193	 - 01=Network - 06=QoS - 4=L3 Signalling (select value)	 - Quality of Service - Layer 3 Signalling (select Layer 3 Signalling value)
Layer 2 Voice (only if L2On)  → page 194	 - 01=Network - 06=QoS - 5=L2 Voice (enter Value)	 - Quality of Service - Layer 2 Voice (enter Layer 2 Voice value)
Layer 2 Signalling (only if L2On)  → page 193	 - 01=Network - 06=QoS - 6=L2 Signalling (enter Value)	 - Quality of Service - Layer 2 Signalling (enter Layer 2 Signalling value)
Layer 2 Default (only if L2On)  → page 193	 - 01=Network - 06=QoS - 7=L2 Default	 - Quality of Service - Default (enter Default value)

 Changing any QoS settings will take effect as soon as the optiPoint 410/420 is restarted.

VLAN Settings

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Manual VLAN Identifier  → page 203	 - 01=Network - 06=QoS - 8=VLAN ID (enter VLAN ID)	 - Quality of Service - Manual vLAN identifier (enter VLAN ID)
VLAN Discovery Method  → page 228	 - 01=Network - 06=QoS - 9=VLAN discovery (select VLAN discovery)	 - Quality of Service - vLAN discovery method (select VLAN discovery)

 Changing the VLAN Discovery Method will take effect as soon as the optiPoint 410/420 is restarted.

Configure LAN Ports

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
LAN/PC Port Setting  → page 194	 - 01=Network - 10=LAN port settings or - 11=PC port settings (select port setting)	 - LAN port settings - LAN port 1 (LAN) or - LAN port 2 (PC) (select port setting)

 Changing any LAN port setting will cause the optiPoint 410/420 to restart.

Configure System Information

To be granted access to a SIP Server some terminal and SIP related information have to be configured.

 First of this the SIP server has to be configured.

Terminal Details

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Terminal Number  → page 224	 - 02=System - 01=Terminal Number (change terminal number)	 - SIP environment - Phone number (enter terminal number)
Terminal Name  → page 223	 - 02=System - 02=Terminal Name (change terminal name)	 - SIP environment - Phone name (enter terminal name)
Register by Name  → page 210	 - 02=System - 03=Register by Name (switch on/off)	 - SIP environment - Register by name (activate/deactivate checkbox)
User Display ID  → page 182	 - 02=System - 44=Display ID - 1=New/Change ID (enter display name)	 - SIP environment - Display ID (enter display name)
Use Display ID  → page 182	 - 02=System - 44=Display ID - 2=Use Terminal ID - 3=Use Display ID (enable/disable)	 - SIP environment - Use Display ID (enable/disable)

SIP Specific Configuration

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
SIP Routing  → page 216	 - 02=System - 04=SIP Routing (select routing)	 - SIP environment - SIP routing (select routing)
Outbound Proxy  → page 205	 - 02=System - 05=Outbound Proxy (switch on/off)	 - SIP environment - Outbound Proxy (activate/deactivate checkbox)
Default OBP Domain Name  → page 174	 - 02=System - 06=Default OBP Domain (change domain name)	 - SIP environment - Default domain name (enter domain name)
SIP Server Address  → page 214	 - 02=System - 07=SIP Addresses - 1=SIP Server (change IP address ¹ and port)	 - SIP environment - Server IP address or DNS name (enter IP address*)
SIP Registrar Address  → page 214	 - 02=System - 07=SIP Addresses - 2=SIP Registrar (change IP address* and port)	 - SIP environment - Registrar IP address or DNS name (enter IP address*)
SIP Gateway Address  → page 214	 - 02=System - 07=SIP Addresses - 3=SIP Gateway - 1=SIP Register Address - 2=SIP Gateway Port (change IP address* and port)	 - SIP environment - Gateway IP address or DNS name (enter IP address*)
SIP Phone Port  → page 214	 - 02=System - 07=SIP Addresses - 5=SIP Phone - 1=SIP Port - 2=RTP Base Port (change port address)	 - SIP environment - SIP Port - RTP Base Port (enter Port addresses)
SIP Transport  → page 217	 - 02=System - 08=SIP Transport - 1=UDP - 2=TCP (select transport)	 - SIP environment - SIP transport (select transport)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
SIP server type  → page 216	 - 02=System - 31=SIP Server type (select type) - 1=HiQ/HiPath 8000 - 2=Broadsoft - 3=Sylantro - 4=Other	 - SIP environment - SIP server type (select type)
SIP Realm  → page 215	 - 02=System - 09=SIP Realm (change realm name)	 - SIP environment - SIP realm (enter realm name)
SIP User ID  → page 218	 - 02=System - 10=SIP User Id (change ID)	 - SIP environment - SIP user ID (enter ID)
SIP Password  → page 215	 - 02=System - 11=SIP Password (change password)	 - SIP environment - New/Confirm SIP password (enter/re-enter password)
SIP Session Timer  → page 217	 - 02=System - 12=SIP Session Timer (switch on/off)	 - SIP environment - SIP session timer enabled (activate/deactivate checkbox)
SIP Session Time  → page 216	 - 02=System - 13=SIP Session Time (enter time)	 - SIP environment - SIP session timer value (enter time)
Registration Timer  → page 211	 - 02=System - 17=Registration Timer (enter time)	 - SIP environment - Registration timer value (enter time)
Transaction timer  → page 225	 - 02=System - 46=Transaction timer (enter time)	 - SIP environment - Transaction Timer (enter time)

1 or host name (if DNS is applicable → page 41)



Changing either the Terminal Number or the SIP Routing setting will take effect as soon as the optiPoint 410/420 is restarted.

SIP Features

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Call handling options		
Auto answer  → page 214	 - 2=Configuration - 32=Auto answer (mark to be enabled)	 - SIP features - Auto answer (mark to be enabled)
Beep on Auto answer  → page 215	 - 2=Configuration - 33=Autoanswer beep (mark to be enabled)	 - SIP features - Beep on Auto answer (mark to be enabled)
Auto reconnect  → page 215	 - 2=Configuration - 34=Auto reconnect (mark to be enabled)	 - SIP features - Auto reconnect (mark to be enabled)
Beep on Auto reconnect  → page 215	 - 2=Configuration - 35=Autoreconnect beep (mark to be enabled)	 - SIP features - Beep on Auto reconnect (mark to be enabled)
Allow Refuse  → page 164	 - 02=System - 32=Allow Refuse (enable or disable)	 - SIP features - Refuse call feature enabled (mark to be enabled)
Group pickup		
Group pickup URI  → page 189	 - 02=System - 30=Group pickup URI (enter Group pickup URI)	 - SIP features - Group pickup URI (enter URI)
HotWarm Phone		
Phone type  → page 190	 - 02=System - 33=Hot/Warm Phone - 1=Normal line Action - 2=Warm line - 3=Hot line (select type)	 - SIP features - Phone type (select type)
Default dial string  → page 190	and default dial string	- Default dial string (enter dial string)
Centralized-controlled conference (only with Sylanro Servers)		
Conference factory URI  → page 170	 - 02=System - 27=Conf Server Addr. (enter URI)	 - SIP features - Conference factory URI (enter address)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Call park (only with Sylantrio Servers)		
Call park URI  → page 169	 - 02=System - 38=Call Park URI (enter URI)	 - SIP features - Call park URI (enter URI)
Call pickup URI  → page 169	 - 02=System - 39=Call Park Pickup URI (enter URI)	 - SIP features - Call pickup URI (enter URI)
Transfer on Ringing  → page 226	 - 02=System - 40=Transfer on Ringing (enable or disable)	 - SIP features - Allow transfer on ringing (mark to allow)
Join allowed in conference  → page 192	 - 02=System - 41=Join allowed in conf (enable or disable)	 - SIP features - Allow join in conference (mark to allow)
Callback URIs  → page 169	 - 02=System - 42=Callback URIs - 1=Callback-busy - 2=Callback-no reply ¹ - 3=Cancel callbacks select and enter access code	 - SIP features - Callback (enter related access code)
Initial Digit Timer  → page 191	 - 02=System - 43=Initial Digit Timer (set timer 1 to 120)	 - SIP features - Initial digit timer (set timer 1 to 120)
Call Recorder  → page 168	 - 02=System - 45=Call recorder - 1=Dial string - 2=Audible indication (enter string and set indication to on/off)	 - SIP features - Call Recorder - Number - Audible indication (enter number and mark indication as on)
Transfer on hangup  → page 226	 - 02=System - 48=Transfer on hangup (switch on/off)	 - SIP features - Allow transfer on hangup (mark to enable)

¹ For future use – not supported with HiPath 8000

Miscellaneous

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Emergency Number ? → page 185	 - 02=System - 14=Emergency No. (change emergency number)	 - SIP environment - Emergency number (enter emergency number)
Voicemail Number ? → page 228	 - 02=System - 15=Voice mail No. (change voicemail number)	 - SIP environment - Voicemail number (enter voicemail number)
Message Waiting Address ? → page 203	 - 02=System - 16=MWI Server Addr. (change IP address ¹)	 - SIP environment - Message Waiting IP Address or DNS name (enter IP address [*])
Banding Identity name ? → page 168	 - 02=System - 26=System Name (change description)	 - SIP environment - Banding/Identity name (enter short description)

1 or host name (if DNS is applicable → page 41)

Configuring Date and Time

If the DHCP server in your network provides information about the SNTP server access, the date and time is automatically shown on the phone.

If the DHCP server in your network does not provide a SNTP address you have to set the SNTP address manually.

If no SNTP server is in your network you have to configure the date and time manually.



If SNTP is being used, any user specified value for Time and Date will be overwritten when the next SNTP update occurs!

SNTP is available, but no automatic access by DHCP server

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
SNTP Address ?/? → page 219	- 04=Date/Time - 1=SNTP IP addr (change IP address¹)	- Time and date - SNTP server IP address or DNS name (enter IP address*)
Timezone Offset ?/? → page 225	- 04=Date/Time - 2=Timezone offset (change timezone offset)	- Time and date - Time zone offset (enter timezone offset)

1 or host name (if DNS is applicable → page 41)

No SNTP server available

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Date/Time ?/? → page 173	- 2=Configuration - 03=Date/time (select a date format and change date and time)	- Time and date - Local time/Date (enter Local time, enter/select Date)
Daylight Saving Time ?/? → page 174	- 2=Configuration - 14=Daylight Saving (switch on/off)	- Time and date - Daylight saving (activate/deactivate checkbox)

Multiline

Line key configuration

In line overview menu you can configure lines and an assign lines to keys. It suffices to assign one line to a key for to go in multiline operation.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Line  → page 196	 (not applicable)	 - Funcktion keys - Phone or - Key module - Select a key with EDIT a configuration dialog appears - Select:line (A key with line is ready for configuring)
Key Label  → page 192	(not applicable)	 Line key configuration dialog - Key Label (enter a name for the line key – only with optiPoint 420)
Address of record  → page 199	(not applicable)	 Line key configuration dialog - Address of record (enter e.g. phone number)
Realm  → page 201	(not applicable)	 Line key configuration dialog - Realm (enter IP address)
Primary line  → page 201	(not applicable)	 Line key configuration dialog - Primary line (mark as primary if required)
Ring  → page 201	 - 02=System - 28=Keyset - 01=Line select a line - 2=Ringer setting (Read only)	 Line key configuration dialog - Ring (enable/disable)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Hunt ranking  → page 199	 - 02=System - 28=Keyset - 1=Line select a line - 1=Rank (enter Rank 1 to 10)	 Line key configuration dialog - Hunt ranking (select order)
User ID  → page 201	(not applicable)	 Line key configuration dialog - User ID (enter ID)
Password  → page 200	(not applicable)	 Line key configuration dialog - Password (enter password)
Shared type  → page 200	(not applicable)	 - Line key configuration dialog - Shared type (select type)

Configure Multiline Operation

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Registration LEDs  → page 210	 - 02=System - 28=Keyset - 2=Registration LEDs (switch on/off)	 - Multiline operation - Registration LEDs (activate/deactivate checkbox)
Rollover type  → page 212	 - 02=System - 28=Keyset - 03=Rollover type (select rollover type)	 - Multiline operation - Rollover type (select rollover type)
Rollover Volume  → page 212	 - 02=System - 28=Keyset - 04=Rollover Vol (change volume)	(not applicable)
Originating line preference  → page 204	 - 02=System - 28=Keyset - 05=Orig line pref (select line preference)	 - Multiline operation - Originating line preference (select line preference)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Terminating line preference  → page 224	 - 02=System - 28=Keyset - 06=Term line pref (select line preference)	 - Multiline operation - Terminating line preference (select line preference)
Line action mode  → page 197	 - 02=System - 28=Keyset - 07=Line action mode (select line action mode)	 - Multiline operation - Line key operation mode (select operation mode)
Show focus  → page 213	 - 02=System - 28=Keyset - 09=Show focus (switch on or off)	 - Multiline operation - Show focus (mark to enable)
Forwarding Indication  → page 186	 - 02=System - 28=Keyset - 10=Forwarding Ind (switch on or off)	 - Multiline operation - Use LED to indicate Remote Forwarding (mark to enable)
Reservation Timer  → page 211	 - 02=System - 28=Keyset - 11=Reservation Timer (set time)	 - Multiline operation - Reservation timer (set time)

Dial Plan Configuration and Status

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Dial Plan  → page 176	 - 02=System - 29=Dial Plan (enable/disable)	 - System - Dial Plan - Action (enable/disable)
Dial Plan Info  → page 176	 - 07=General info - 4=Dial Plan - 1=ID - 2=Stauts	 - General information - Dial plan - Name and Status

Dialling Properties

Only used with optiPoint 410/420 advance/standard

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Dialling properties  → page 181	 - 02=System - 35=LDAP Select one of the options for to change its value - 3=External code - 4=International prefix - 5=Country code - 6=National prefix - 7=Area code - 8=District code	 - Dialling properties (enter values for the properties)

Direct Station Select (DSS – HiPath 8000 only)

Each DSS key will be a special variant of a line key. The configuration specifies whether a line key will be a DSS key or a normal multiline key. The system operation and protocol of the DSS key will be the same as for a line key and the HiPath 8000 will not be required to know if a line appearance is associated with a DSS key or a multiline key.

A DSS key will use the line key mechanism to display the line state via the LED associated with the key. However, the DSS key will only present a subset of the line states to the user; i.e. Idle, Alerting and Busy. All other states that a Keyset line key could present will be forced into one of the valid DSS states.

A major departure from Keyset line key operation is the action taken when a DSS key is pressed. The DSS action falls into two basic camps

1. Pickup a call alerting the DSS target
2. Make/complete a call using the DSS target as the destination.

Completion of a call applies to cases where the user has performed an operation at the phone which results in them being prompted for destination digits

DSS key configuration

Each DSS key will be configured similarly to a Keyset line key and will require the following to be specified for the line:

- SIP URI of the primary line at the DSS target
- SIP Realm
- SIP User ID
- SIP Password

The remaining line configuration items will be forced to specific values for a DSS key line

In addition a DSS key being configured will set the phone to not use the optiPoint display module by default, and a DSS key will not appear as a line key on the optiPoint display module.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
DSS  → page 196	 (read only if configured)	 - Funcktion keys - Phone or - Key module - Select a key with EDIT a configuration dialog appears - Select:DSS (A key with DSS is ready for configuring)
DSS Address of Record  → page 184	 (read only if configured)	 DSS key configuration dialog - Address of record (enter e.g. phone number)
DSS Realm  → page 184	 (read only if configured)	 DSS key configuration dialog - Realm (enter IP address)
DSS User ID  → page 185	 (read only if configured)	 DSS key configuration dialog - User ID (enter ID)
DSS Password  → page 185	 (read only if configured)	 DSS key configuration dialog - Password (enter SIP-password)

Feature Access

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Feature Access  → page 185	 - 02=System - 37=Feature Access Select one of the options for to enable oder disable - 01=Call Hold Explicit - 02=Call Deflection - 03=Call Forwarding - 04=Log Forwarded Calls - 05=Call Duration - 06=Call Waiting - 07=Call Transfer - 08=Call Join - 09=Call Display Name - 10=Call Display Number - 11=Music On Hold - 12=Do Not Disturb - 13=Message Waiting - 14=Local Conference - 15=Auto Answer CTI - 16=Auto Reconnect CTI - 17=Call Park - 18=Call Park Pickup - 19=Wap Browser On DSM - 20=Address Book - 21=DSM Call Control - 22=Voice Recognition - 23=Speed Dial On DSM - 24=Contacts - 25=Hot Keypad Dialing - 26=Callback-busy - 27=Callback-no reply - 28=Call recording - 29= GPU New Call Beep	 - Feature Access (mark as enabled) - Auto answer - CTI - Auto reconnect - CTI¹ - Call deflect - Call display by name - Call display by number - Call duration - Call forwarding - Call hold (explicit) - Call join - Call park² - Call recording - Call pickup² - Call transfer - Call waiting - Do not disturb - DSM - ENB³ - DSM - LDAP³ - DSM - Speed dial³ - DSM - Telephony use³ - DSM - Voice recognition³ - DSM - WAP browser³ - GPU New Call Beep - Local conference⁴ - Log forwarded calls² - Message waiting - Music on hold - Hot keypad dialing - Callback - busy - Callback - no reply

¹ Not used with optiPoint 410 entry

² Not supported with HiPath 8000

³ Not used with optiPoint 410/420 economy economy plus and optiPoint 410 entry

⁴ Not used with optiPoint 410/420 economy and optiPoint 410 entry

Function Key assignments

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Function key  → page 188	assignments:see operating manual lines can not be configured in the phone menu.	 - Funcktion keys - Phone or - Key module - Select a key with EDIT a configuration dialog appears - Select:a function in the list (configure parameters in the dialog if required)

Software Update/Transferring Files

The optiPoint 410/420 is capable of transferring files using the → FTP protocol. This feature can be used to update the phone software, download a music file and **up- or download** the phone's configuration file.

The phone acts as a FTP client and requires a FTP server in the IP network where the files are located or can be placed.

Application Software Update

If it is necessary to change or upgrade the application software of your optiPoint 410/420, perform the following.

- Find out the current application version of your optiPoint 410/420 → page 91.
- Decide whether an update is useful and necessary.



Be careful! Consider that the software must be compatible with the telephone.

- If useful download it from internet and install the application software via FTP → page 77.

FTP Server Requirements

There are no specific requirements on the functionality of the FTP server to be used. Any FTP server providing standard functionality will do. There is a variety of servers available including freeware on the internet.

Please read the documentation for the FTP software for details of how to install and configure the FTP server.

Common FTP Server Access Configuration

The FTP client on the phone will open a session and therefore requires:

- Account name
- Username
- Password
- Path (optional)

Please note that Account name and Username might be the same on the FTP server used. In this case use the name for setting both parameters.

The parameter path allows you to specify a directory path on the FTP server where the files you want to up- or download are located. This path is relative to the path set for the user on the FTP server. The combination of both settings will make up the full path.

Example: If the user's path on the FTP server is "C:\temp" and the path set on the phone is "op400\files" the directory where you will need to put the files is "C:\temp\op400\files".

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Download Server Address  → page 183	 - 03=File transfer - 06=DL server IP addr. (change IP address ¹)	 - File transfer - Download server IP address or DNS name (enter IP address*)
FTP Account Name  → page 187	 - 03=File transfer - 07=FTP account name (change account name)	 - File transfer - FTP account name (enter account name)
FTP Username  → page 188	 - 03=File transfer - 08=FTP username (change username)	 - File transfer - FTP username (enter username)
FTP Password  → page 187	 - 03=File transfer - 09=FTP password (change password)	 - File transfer - New/Confirm FTP password (enter/re-enter password)
FTP Path  → page 188	 - 03=File transfer - 10=FTP path (change path)	 - File transfer - FTP path (enter path)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
FTP Passive Mode  → page 187	 - 03=File transfer - 25=Use Passive Mode FTP (select switch on/off)	 - File transfer - Use Passive Mode FTP (mark to use)

1 or host name (if DNS is applicable → page 41)

Upload Configuration File

The optiPoint 410/420 allows you to upload the phones configuration file. Uploading will be done in the binary format to the **FTP download server** with its common settings. The fixed name of file which will be uploaded is **opti400c** without an extension.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Upload Configuration  → page 227	 - 03=File transfer - 03=UL config	 - File transfer (submit with "upload configuration") - Upload configuration

Downloading Files

If you want to download files (application, configuration, hold music) you can either browse through the menu structure on the FTP server (phone menu only) or specify this filename before download.

Download files using the phone menu without setting a filename

The FTP-Client on the optiPoint 410/420 does support a user dialog for selecting a file for download. The mechanism can be triggered by omitting the download file name and starting the download.

For example if the application download file name is not set and download application is selected ( > 03=File transfer > [required type of download file]), the phone will establish connection to the FTP-Server.

Example of information provided on the display:



Upper left corner: current folder or file,
upper right corner: number of sub-folders and files,
lower left corner: selected folder or file (e.g. "op400a"),
lower right corner: number of current folder or file.

A dot (".") means the current folder, double dot ("..") means the superordinately folder.



Press this keys to browse in the current folder.



Press this keys to change the folder level (plus: subordinately folder, minus: superordinately folder).



Press this key to start download.



After the download is completed it will cause the optiPoint 410/420 to restart.

Specify and download application file

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Application Download filename ?/? → page 165	- 03=File transfer - 11=Applic. DL filename (change filename)	- File transfer - Application download filename (enter filename)
Download Application ?/? → page 183	- 03=File transfer - 01=DL application	- Action on submit (select Download type) or - Download Application



After the download is completed it will cause the optiPoint 410/420 to restart.

Specify and download configuration file

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Configuration Download Filename ?/? → page 170	- 03=File transfer - 12=Cfg DL filename (change filename)	- File transfer - Configuration download filename (enter filename)
Download Configuration ?/? → page 183	- 03=File transfer - 02=DL config	- Action on submit (select Download configuration) or - Upload configuration

Specify and download hold music file

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Hold Music Filename ?/? → page 189	- 03=File transfer - 14=Hold music filename (change filename)	- File transfer - Hold music download filename (enter filename)
Download Hold Music ?/? → page 183	- 03=File transfer - 05=DL hold music	- Action on submit (select Download hold music)

Specify files using with DSM for download

Download server address see → page 75

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
LDAP Template  → page 195	 - 03=File transfer - 17=LDAP Template filename (change file name)	 - File transfer - LDAP template filename (enter filename) - Action on submit (select LDSP template)
	 - 03=File transfer - 16=DL LDAP Template (start download)	
Download Java midlet  → page 192	 - 03=File transfer - 19=JavaApplet Filename (change file name)	 - File transfer - Java Program filename (enter filename) - Action on submit (select Java midlet)
	 - 03=File transfer - 18=DL JavaApplet (start download)	
Download DSM Logo  → page 184	 - 03=File transfer - 13=Logo filename (change file name)	 - File transfer - Logo filename (enter filename) - Action on submit (select Download logo)
	 - 03=File transfer - 04=DL DSM Logo (start download)	
Download DSM Firmware  → page 184	 - 03=File transfer - 24=DSM f'ware filename (change file name)	 - File transfer - DSM firmware filename (enter filename) - Action on submit (select Download DSM firm-ware))
	 - 03=File transfer - 23=DL DSM f'ware (start download)	

Applications

LDAP Server Settings

Only applicable with optiPoint display module.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
LDAP server  → page 194 LDAP port  → page 194	 - 02=System - 35=LDAP - 1=LDAP Server address (change IP address ¹) - 2=LDAP Server port (change port)	 - Applications - Directory (change IP address * and port)

¹ or host name (if DNS is applicable → page 41)

Java Proxy

Only applicable with optiPoint display module.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Java Proxy server  → page 192 Java Proxy port  → page 192	 - 02=System - 36=Java Proxy - 1=Java Proxy Address (change IP address ¹) - 2=Java Proxy port (change port)	(not applicable)

¹ or host name (if DNS is applicable → page 41)

Address Book Settings

Only applicable with optiPoint display module.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Address Book server address  → page 162	(not applicable – to enter data, use optiPoint display module Contacts)	 - Applications - Address Book - Address Book server address (enter IP address)
FTP account name  → page 162	(not applicable – to enter data, use optiPoint display module Contacts)	 - Applications - Address Book - FTP account name (enter account name)
FTP user name  → page 162	(not applicable – to enter data, use optiPoint display module)	 - Applications - Address Book - FTP user name (enter user name)
FTP password  → page 162	(not applicable – to enter data, use optiPoint display module Contacts)	 - Applications - Address Book - New FTP password (enter password and cornfirm))
FTP path  → page 162	(not applicable – to enter data, use optiPoint display module Contacts)	 - Applications - Address Book - FTP path (enter path)
FTP filename  → page 162	(not applicable – to enter data, use optiPoint display module Contacts)	 - Applications - Address Book - FTP filenameek (enter name with extension csv)
Download Address Book	(not applicable – to download data, use optiPoint display module Contacts)	 - Applications - Address Book (submit with download Address Book)
Upload Address Book	(not applicable – to upload data, use optiPoint display module Contacts)	 - Applications - Address Book (submit with Upload Address Book)
Delete Address Book Entries	(not applicable – to delete data, use optiPoint display module Contacts)	 - Applications - Address Book (submit with Delete ALL entries)

WAP Settings

Only applicable with optiPoint display module.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
WAP mode  → page 229	 - 02=System - 34=WAP - 1=WAP mode select - 1=WSP or - 2=HTTP	 - Applications - WAP Settings - WAP mode (select WAP mode)
WAP server  → page 229 WAP port  → page 229	 - 02=System - 34=WAP - 2=WAP Server address (change IP address) - 3=WAP Server port (change port)	 - Applications - WAP Settings - WAP Server address - Port (enter IP- and Port address)
WAP proxy Username  → page 189	not applicable	 - Applications - WAP Settings - WAP proxy Username (enter Username)
WAP proxy password  → page 189	not applicable	 - Applications - WAP Settings - New WAP proxy password (enter password and confirm)
Help internet URL  → page 189	not applicable	 - Applications - WAP Settings - Help internet ULRL (enter URL)
Home page  → page 189	not applicable	 - Applications - WAP Settings - Home page (enter home page)

Port Numbering

The phone will provide the ability to configure any TCP or UDP port number that is currently fixed, with the exception of:

- SNMP
- SNTP
- DNS
- WAP Push
- WAP Push secure.

The table below gives the port addresses that will be configurable. The table also shows the default values (see also [RFC 1700](#) – Assigned numbers).

In some cases a port base number is used. The port base number will be the configurable item, not the port numbers derived from the port number base.

Function	Default Value UDP	Default Value TCP	Comment
RTP port range (local)	5004 to 5006		RTP port number is even. RTCP port number is (RTP port number)+1. 5004 to 5006 reserved by IANA but can be used by the phone.
RTP port range (remote)	any		
RTCP port range (local)	5005 to 5007		RTP port number is even. RTCP port number is (RTP port number)+1. 5005 to 5007 reserved by IANA but can be used by the phone.
RTCP port range (remote)	any		
HTTP - Hypertext Transfer Protocol		8085	
HTTPS – Secure Hypertext Transfer Protocol		443	
SNMP	161		Not configurable
SNMP Traps	162		Not configurable
SNTP	123		Not configurable
SNTP Heart-beat	580		Not configurable
DNS	53		
DHCP Server port	67		Default BOOTP port numbers. Not configurable

Function	Default Value UDP	Default Value TCP	Comment
DHCP Client port	68		Default BOOTP port numbers. Not configurable
FTP		21	
TFTP	69		Not configurable
LDAP	389	389	
WAP	2948	2948	WAP push. Not configurable
	9200	9200	WAP connectionless session service
Service Agent Request Port	5100		SA port base Default value = 5100
Auto-discovery	5100		SA port base + 0
Config Service	5130	5130	SA port base + 30
Resource Sharing	5135	5135	SA port base + 35
QDC server	12010	12010	
DLS	18443	18443	
Sip server	5060	5060	

The port numbers and port base numbers will be configurable via the Web pages and by the DLS.

Note that changing the value of a port number may require the phone to restart.

Configuration Management

During "Initialising" the phone will attempt to contact a configuration server to determine if a configuration update is required.

The phone automatically contacts an FTP config server and downloads 2 XML configuration files in which phone-specific parameters may have been automatically inserted by a customer management system for the environment concerned.

- The phone can also perform this download
- on a signal from the customer management system or
- periodically as determined by a configurable time, or by an administration level optiGuide menu option.

If the downloaded configuration files show that a **different version of software** is required, the phone will automatically initiate software download from the FTP software server.

Two configuration files are used:

- a system-wide configuration file, which does not contain phone-specific parameters; and
- a device configuration file, which includes phone-specific parameters.

There is provision for downloading only a device configuration file so that phone-specific parameters can be changed without impacting those system-wide parameters that have been changed locally at the phone.

The configuration download mechanism is intended only for configuring those phone parameters that are under administration control. It is not intended for configuring parameters that are under user control. The changes to the configuration will be reflected in the configuration INI file.

If the configuration download mechanism is not available, e.g. no FTP configUD server, the phone will not perform configuration download and will then rely on other means for configuration updates.

As a result of carrying out a configuration download, the phone will detect whether a software update is required and if so automatically download new software.

File Formats

Configuration files

The configuration files will be stored on the FTP server. There are two types of configuration files:

System Wide File

The system-wide configuration file will have a well-known file name (e.g., "op410adv_conf.xml"), which does not contain phone-specific parameters.

Example System-wide configuration file

```
<?xml version = "1.0" encoding = "utf-8" ?>
- <Message>
  <Action>WriteItems</Action>
  <ItemList>
    <Item name = "system-config-version">2005.07.19 05:39 GMT</Item>
    <Item name = "admin-pwd">424242</Item>
    <Item name = "system-description">Siemens Communications</Item>
    <Item name = "dns-domain-name">ust.local</Item>
    <Item name = "dns-server-addr">172.16.127.2</Item>
    <Item name = "dns-server-addr2">172.16.127.3</Item>
    <Item name = "port1">0</Item>
    <Item name = "port2">0</Item>
    <Item name = "reg-ttl">3600</Item>
    <Item name = "registrar-addr">as.broadworks.net</Item>
    <Item name = "reg-addr">as.broadworks.net</Item>
    <Item name = "function-key-def" index = "6">31</Item>
    <Item name = "function-key-def" index = "14">10</Item>
  </ItemList>
</Message>
```

Device Specific File

Each device-specific configuration file will have a file name that includes the phone's MAC address (e. g. "op410adv_conf001E300A9F3.xml"), which includes phone-specific parameters.

Example Device Specific File

```
<?xml version = "1.0" encoding = "utf-8" ?>
- <Message>
  <Action>WriteItems</Action>
- <ItemList>
  <Item name = "device-config-version">2005.07.19 06:09 GMT</Item>
  <Item name = "e164">3015553005</Item>
  <Item name = "sip-name">John Doe</Item>
  <Item name = "register-by-name">false</Item>
  <Item name = "line-sip-uri" index = "6">3015553009</Item>
  <Item name = "line-primary" index = "6">true</Item>
  <Item name = "line-hunt-sequence" index = "6">1</Item>
  <Item name = "line-shared-type" index = "6">0</Item>
  <Item name = "line-sip-realm" index = "6">as.broadworks.net</Item>
  <Item name = "line-sip-user-id" index = "6">3015553009</Item>
  <Item name = "line-sip-pwd" index = "6">0dskmwefd</Item>
</ItemList>
</Message>
```



Detailed example configuration files and informations you will find in the "XML Configuration Management" Administrator Manual .

Download

Configuration file download

By default, the telephone checks the FTP server for an account called "config" using the password "config" (for details on the plug & play settings, please refer to the "Configuration Management Settings" dialog of the telephone's web interface → page 135).

In certain circumstances the phone will access the FTP server and check in the system-wide configuration file for a `<write item="system-config-file">` element and in its own device configuration file for a `<write item="device-config-file">` element. If the system-wide file version has changed compared with the most recent download, the phone will execute the write instructions in both the system-wide file and its device configuration file. If the system-wide file version has not changed but the device configuration file version has changed, the phone will execute only the write instructions in the device configuration file. Otherwise the phone will not execute write instructions from either configuration file. This preserves any local settings if there is no change to the configuration files.

When the phone downloads configuration files, it will download the system-wide configuration file first and then its own device configuration file. Any parameters values present in the system-wide configuration file will overwrite previous values. Any parameter values present in the device configuration file will overwrite previous values or values in the system-wide configuration file.

How to configure the download server see → page 88.

Period check of configuration files

While registered with the SIP registrar, the phone will, subject to configuration, perform FTP server address discovery and configuration file download periodically (e.g., once a day) to ensure that any changed configurable parameters are obtained. This is useful for two reasons:

- When the phone has started up using previous parameters values because it has been unable to download the configuration files (e.g., because of congestion at the FTP server), this gives a second chance to obtain any updated parameters.
- In environments that do not support the notification mechanism of the following requirements:
 - there is a telephone call in progress,
 - the phone's admin user interface is being used,
 - the phone's admin web pages are being used,

The interval between attempts will be configurable.

Automatic software download

A configuration file can contain a `<write item="ftp-app-file">` element, which contains the file name of the software version to be used. The phone will treat this differently from other `<write>` elements. If the file name is the same as the file name of the software version that is currently running, the phone will take no action and will continue processing other `<write>` elements in the configuration files. If it is different, the phone will stop processing further `<write>` elements and initiate software download using the new file name and the configured FTP server address, path, user name and password. If software download is successful, the phone will install the new software in flash memory and start it up. This will result in downloading the configuration files again. On this occasion the phone will find that it is already running the correct software version and will process the remaining `<write>` elements. If software download fails, the phone will log an error and continue with its existing software version.



This method allows a configuration file to include `<write>` elements for configurable parameters that exist only in the new version of software. These `<write>` elements should appear later in the configuration files than the `<write item="ftp-app-file">` element.



The configured FTP server address, path, user name and password can differ from those used for configuration file download. Any `<write>` element that sets these parameters should be placed in the configuration files prior to the `<write item="ftp-app-file">` element.

Each version of software will have its own built-in file name for the comparison above. This will comprise a fixed string (e.g., `optiPoint400SIP`) followed by the version number followed by the extension (e.g., `.bin`). This same file name should be used when posting a software file on the FTP server.

This mechanism can be used to revert to an older version of software as well as for advancing to a later version.



After the download is completed it will cause the optiPoint 410/420 to restart.

Specify configuration update file

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 109)
Configuration update check for update  → page 169	 - 12=Config Update - 01=Check for update (confirm to start the check manually)	 - Configuration management - Check for updates (confirm to start the check manually)
Error Log  → page 136	(not applicable)	 - Configuration management - Error Log (confirm to open the error log file if exists)
Configuration update FTP IP  → page 172	 - 12=Config Update - 02=Ftp DL Params - 01=Config FTP Addr (change/enter IP address)	 - Configuration management - Settings - Non-secure configuration download (FTP) - FTP server IP address or DNS name (enter IP address)
Configuration update FTP Port  → page 172	 - 12=Config Update - 02=Ftp DL Params - 02=Config FTP Port (enter Port address)	 - Configuration management - Settings - Non-secure configuration download (FTP) - FTP server Port (enter Port)
Configuration update Pathname  → page 172	 - 12=Config Update - 02=Ftp DL Params - 03=Config UD Pathname (enter path name)	 - Configuration management - Settings - Non-secure configuration download (FTP) - FTP Path (enter Path name)
Configuration update Account name  → page 171	 - 12=Config Update - 02=Ftp DL Params - 04=Account ID (enter Account ID)	 - Configuration management - Settings - Non-secure configuration download (FTP) - FTP Account name (enter Account name)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 109)
Configuration update username  → page 173	 - 12=Config Update - 02=Ftp DL Params - 05=Username (enter user name)	 - Configuration management Settings - Non-secure configuration download (FTP) - FTP username (enter user name)
Configuration update Password  → page 172	 - 12=Config Update - 02=Ftp DL Params - 06=Password (enter new FTP password and confirm password)	 - Configuration management Settings - Non-secure configuration download (FTP) - New FTP password (enter new password and confirm password)
Configuration update Dls DL Params  → page 171  → page 170	 - 12=Config Update - 03=Dls DL Params - 1=Config UD Dls - 2= Config UD Dls Port (change IP-Address or Port)	 - Configuration management Settings - Deployment Service (DLS) - IP address or DNS name/Port
Configuration update Https DL Params  → page 135	 - 12=Config Update - 04=Https DL Params - 1=Config UD Https Addr - 2= Config UD Https Port - 3= Config UD Https Path	 - Configuration management Settings - Secure configuration download (Https) - IP address) - IP address/Port - File path for URL
Configuration update File name  → page 171	 - 12=Config Update - 05=Config UD Filename (enter filename – optipoint410config)	 - Configuration management Settings - Common settings (HTTPS and FTP) - Configuration file name (enter filename – optipoint410config)
Configuration update file type  → page 171	 - 12=Config Update - 06=Config UD FileType (enter File type –xml)	 - Configuration management Settings - Common settings (HTTPS and FTP) - Configuration file type (enter type – xml)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 109)
Configuration update Periodic Timer  → page 172	 - 12=Config Update - 07=Periodic Timer (enter time in seconds)	 - Configuration management - Settings - Common settings (HTTPS and FTP) - After registration check for updates every n seconds (enter time in seconds)
Configuration update Unregistration Timer  → page 173	 - 12=Config Update - 08=Unreg Timer (enter time in seconds)	 - Configuration management - Settings - Common settings (HTTPS and FTP) - If registration fails, check for updates every n seconds (enter time in seconds)
Configuration update Management Type  → page 227	 - 12=Config Update - 09=Config Mgmt Type (read only)	 - Configuration management - Settings - Deployment service (DLS) Use deployment service enable/disable
Configuration update Management Type  → page 228	 - 12=Config Update - 09=Config Mgmt Type (read only)	 - Configuration management - Settings - Secure configuration download (HTTPS) Use secure configuration download enable/disable
Configuration update Management Type  → page 228	 - 12=Config Update - 09=Config Mgmt Type (read only)	 - Configuration management - Settings - Non-secure configuration download (FTP) Use non-secure configuration download enable/disable
Configuration update Authentication  → page 171	 - 12=Config Update - 10=Authentication (switch on/off)	 - Configuration management - Settings - Common settings (HTTPS and FTP) - Authentication enabled (mark enabled/disabled)

Display Upload/Download Status

Before you transfer a file, it could be useful to have a look at the current status of transferred files.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Application download  → page 227	 - 09=UL/DL status - 1=Application download	 - General Information - File Transfer status
Configuration download  → page 227	 - 09=UL/DL status - 2=Config download	 - General Information - File Transfer status
Config upload  → page 227	 - 09=UL/DL status - 3=Config upload	 - General Information - File Transfer status
Logo download  → page 227	 - 09=UL/DL status - 4=Logo download	 - General Information - File Transfer status
MoH download  → page 227	 - 09=UL/DL status - 5=MoH download	 - General Information - File Transfer status
System configuration download  → page 227	(not applicable)	 - General Information - File Transfer status
Phone configuration download  → page 227	(not applicable)	 - General Information - File Transfer status
Personal directory import/export  → page 227	(not applicable)	 - General Information - File Transfer status

Display Application Version

If you want to update the optiPoint 410/420 → page 74 you should find out the current version of the application software → page 94.

Reset Music on Hold to default music

Set the customized (downloaded) hold music to default.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Default music  → page 174	 - 03=File transfer - 15=Default music (select Reset or Cancel)	(not applicable)

Use SNMP

SNMP Server Configuration

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
SNMP Trap Address  → page 219	 - 05=SNMP - 01=SNMP trap IP addr (change IP address ¹)	 - SNMP Settings - Management Center Address (enter IP address*)
SNMP Password  → page 218	 - 05=SNMP - 02=SNMP password (change password)	 - SNMP settings - New/Confirm Query password (enter/re-enter password)
Management Center Port  → page 202	Not applicable	 - SNMP settings - Management Center Port (enter port address)
Send Generic Taps  → page 213	Not applicable	 - SNMP settings - Send Generic Taps to Management Center (enable/disable)
Send QDC Taps  → page 213	Not applicable	 - SNMP settings - Send QDC Taps to Management Center (enable/disable)
Queries Allowed  → page 218	Not applicable	 - SNMP settings - Queries Allowed (enable/disable)

¹ or host name (if DNS is applicable → page 41)

View SNMP Errors

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
MIB2 Discards  → page 182	 - 10=MIB2-Discards	 - SNMP - SNMP MIB2 errors
MIB2 Err Count  → page 191	 - 11=MIB2 err count	 - SNMP - SNMP MIB2 errors

Change Speech Parameters

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Audio Mode  → page 165	 - 06=Speech - 2=Audio mode (select audio mode)	 - Speech - Audio mode (select audio mode)
Compression Encoding  → page 170	 - 06=Speech - 3=Compression (select compression)	 - Speech - Compression encoding (select compression)
G.711 Silence  → page 188	 - 06=Speech - 4=G711 Silence (switch on/off)	 - Speech - G.711 Silence Suppression (activate/deactivate checkbox)
RTP Packet Size  → page 212	 - 06=Speech - 5=RTP Packet Size (select packet size)	 - Speech - G.711 RTP packet size (select packet size)
Microphone  → page 203	 - 06=Speech - 6=Microphone - 1=Normal - 2=Disabled (select option)	 - Speech - Microphone Disable (Mark as disabled or not)

Configure Ringer Settings

Audio/Visual Indications

This setting is used to setup Alert Indications that can be used to differentiate between call types.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Alert Indications  → page 163	 - 11=Ringer Settings - 01=Alert indications - 01=Indication 1 up to - 15=Indication 15 (change identifier, select alert type and enter ringer melody and tone)	 - Ringer Settings (enter alert indication string and enter melody, tone and duration)

 This feature is only supported in specific system environments.

Display static Phone Information

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
MAC Address  → page 202	 - 07=General info - 1=MAC address	 - General information (MAC address)
Application Version  → page 228	 - 07=General info - 2=Version Info - Application	 - General information - Versions (Application)
SIP Stack Version  → page 217	 - 07=General info - 2=Version Info - SIP stack	 - General information - Versions (SIP stack)
SIP Signalling Version  → page 217	 - 07=General info - 2=Version Info - SIP signalling	 - General information - Versions (SIP signalling)
RTP Version  → page 212	 - 07=General info - 2=Version Info - RTP	 - General information - Versions (RTP)

Perform Diagnostic Tests

Non user-assisted diagnostic tests

These types of diagnostic tests do not require assistance from a local user.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
PING Test  → page 207	 - 07=PING (change or select IP address ¹)	 - Non user-assisted diag. tests - Ping name - IP address or DNS (activate checkbox and enter IP address*)
RAM Test  → page 210	 - 05=RAM test	 - Non user-assisted diag. tests - RAM test (activate checkbox)
ROM Test  → page 212	 - 06=ROM test	 - Non user-assisted diag. tests - ROM test (activate checkbox)

1 or host name (if DNS is applicable → page 41)

User-assisted diagnostic tests

These types of diagnostic tests require a local user to confirm the result at the optiPoint 410/420.

 Once a particular test has been started, the local user will assume full control of the test (using the keypad) until the test is terminated.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Display test  → page 182	 - 01=Display test	 - User-assisted diagnostic tests - Display test
LED test  → page 196	 - 02=LED test	 - User-assisted diagnostic tests - LED test
Key test  → page 193	 - 03=Key test	 - User-assisted diagnostic tests - Key test
Audio loop test  → page 165	 - 04=Audio loop	 - User-assisted diagnostic tests - Audio loop test
Line monitor  → page 198	 - 08=Line monitor	 - User-assisted diagnostic tests - Line monitor
Key Label Test  → page 213	 - 13=Key Label Test	 - User-assisted diagnostic tests - Self Labelling keys test

Security settings

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Payload Security allowed  → page 206	 - 02=System - 47=Payload Security All (Switch off/on)	 - Security Settings - Payload Security allowed (mark as allowed)
Connectivity check interval  → page 173	 - 02=System - 48=Conn.Check Interval (enter value)	 - Security Settings - Connectivity check interval (enter value)
SIP Server Validate  → page 216	 - 02=System - 49=SIP Server Validate (Switch off/on)	 - Security Settings - SIP server validation (mark as enabled)

Restart the optiPoint 410/420

Perform a Restart to optiPoint 410/420 S V7.0 display phones

Restart by Hardware

Remove the network connection → page 20 for about 5 seconds and re-plug the connection.

Restart by software

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Restart Terminal	press     successively confirm with ☒	 - Restart terminal (the connection to the phone will be lost temporarily)



After changing administrative settings an automatic restart of the phone may be necessary, e.g. switching DHCP IP assignment on/off → page 51.

View Date and Time of Last Restart

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Last Restart  → page 194	 - 07=General info - 3=Last Restart	(not applicable)

Perform a Restart to optiPoint 410 /entry

Restart by Hardware

Remove the network connection → page 20 for about 5 seconds and re-plug the connection.

Restart by software

Precondition: A confirm key is established (→ page 127)

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Restart Terminal	press     successively confirm with confirm key	 - Restart terminal (the connection to the phone will be lost temporarily)

Restore Factory Settings

optiPoint 410/420 S V7.0 display phones

The following procedure can be invoked in order to reset the optiPoint 410/420 /economy/economy plus/standard/advance back to its default factory settings.



Attention

A factory reset deletes all administration data, passwords (except reset password) and user configurations. IP and SIP connections will be lost.

To avoid the necessity of re-entering the phone configuration manually after restoring factory settings use the Upload Configuration function → page 76 to save the configuration. After factory reset the FTP password is replaced with the string "123abc".

To perform a factory reset:

- Remove the LAN connection () → page 20.



When Power over LAN do not disconnect the LAN.

- Take the handset off hook.
- Make sure the phone is in idle state ("No network" is flashing).
- Press the following keys in a row: .
- Enter reset password: "124816".
- Press the key .
- Reconnect the LAN connection → page 20.
- Replace the handset.

You also can use the **Web Interface** to reset factory settings → page 145.

After reset you will be prompted for the terminal number. Enter the requested terminal number. If you do not enter a valid number within a pre-defined time, the IP number of the phone appears on the display. You can enter the number later, using the phone menu or the Web Interface. If you use the "downloaded configuration", the number will be inserted automatically.

optiPoint 410 entry

To perform a factory reset:

- Remove the LAN connection → page 20.



When Power over LAN do not disconnect the LAN.

- Press hard keys '2', '8' and '9' simultaneously,
- Press the '*' hard key
- Enter the standard factory reset password "124816",
- Terminate by pressing the '#' hard key.

You also can use the **Web Interface** to reset factory settings → page 145

Change Administrator Password

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Admin Password  → page 163	 - 08=Admin password (change password)	 - Change administrator password (enter current and new password and confirm)

Reset User Password

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Reset User Password  → page 143	 - 3=Local functions - 02=User password (select change and enter new password)	 - Reset user password (enter new password and confirm)

Clear ALL user data

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Clear ALL user data  → page 169	 - 3=Local functions - 04=Memory - 02=Delivery status (confirm clear)	 - Clear ALL user data (confirm with OK)

Port Control

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Port Control  → page 207	 - 15=Port Configuration - 1=Service Agent - 2=Test Interface - 3=Resource sharing - 4=SNMP Port - 5=Web Server (enable or disable)	 - Port control - Service Agent - Test Interface SNMP Interface Resource Sharing (mark as enabled)

Survivability

The survivability feature will allow the SIP User Agent to register with a backup SIP Proxy which will be used to make and receive calls when the primary SIP Proxy fails or is not reachable due to a network failure.

The prime reason for this feature is to maintain basic call functionality when network failures occur and it is therefore expected that some features and functionality will not be available when working in a survivability mode.

Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Backup Address  → page 219	 -16=Survivability - 1=Backup address (change IP address¹)	 - Survivability - Backup IP address or DNS name (enter IP address¹)
Backup Port  → page 219	 -16=Survivability - 2=Backup port (change port address)	 - Survivability - Port (enter port address)
Backup Registration  → page 220	 - 16=Survivability - 3=Backup registration (switch on/off)	 - Survivability - Backup Registration (mark to enable)
Backup Reg Timer  → page 220	 - 16=Survivability - 4=Backup reg timer (enter time)	 - SIP environment - Backup Registration timer value (enter time)
Backup OBP  → page 220	 - 16=Survivability - 5=Backup OBP (switch on/off)	 - Survivability - Backup Outbound proxy (mark to enable)
Backup Transport  → page 220	 - 16=Survivability - 6=Backup transport (TCP or UDP; UDP is prepared)	 - Survivability - Backup SIP transport (TCP or UDP;UDP is prepared)

1 or host name (if DNS is applicable → page 41)

Behaviour regarding the Survivability settings

Please make sure all parameters are set completely and accurately.

Backup Server not entered and activated

- Backup IP address or DNS name:
 - **No** IP address was entered (→ page 219, → page 147)
- Backup registration:
 - The Checkbox for the "Backup registration" feature is **not** marked (→ page 147) and/or the feature was **not** activated in the telephone menu (→ page 220)

The telephone only registers at the server (Hipath8000). In case the server fails or is not available, the message "Server not found" is displayed. No calls can be made.

Backup Server is entered, but not activated

- Backup IP address or DNS name:
 - IP address is entered (e.g. 192.168.1.1 → page 219, → page 147)
- Backup registration:
 - The Checkbox for the "Backup registration" feature is **not** marked (→ page 147) and/or the feature was **not** activated in the telephone menu (→ page 220).

The telephone only registers at the server (Hipath8000). In case the server fails or is not available, no message is displayed. Even after restarting the telephone no message is displayed, and the display looks as if the telephone had registered at the HiPath8000. However, outbound calls are possible via the backup server, while inbound calls to this telephone are not possible as it is not registered at the backup server.

Backup Server is entered and activated

- Backup IP address or DNS name:
 - IP address is entered (e.g. 192.168.1. → page 219, → page 147)
- Backup registration:
 - The Checkbox for the "Backup registration" feature is marked (→ page 147) and the feature is activated in the telephone menu (→ page 220).

The telephone registers both at the server (Hipath8000) and the backup server. In case the Hipath8000 fails or is not available, the message "Temporary limited mode" is displayed. Both outbound and inbound calls are possible via the backup server..

User Mobility (Hipath 8000)

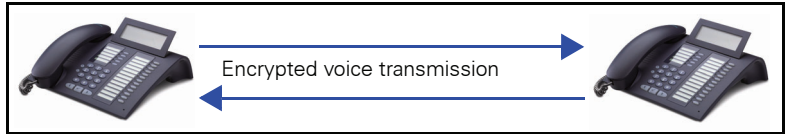
Parameter	Phone path (Menu → page 148)	Web Interface path (Menu → page 110)
Unauth Logoff Trap  → page 226	 -17=User mobility - 02=Unauth Logoff trap (enable/disable)	 - User Mobility - Send trap on unauthorised logoff attempt (enable/disable)
Logoff Trap Delay  → page 202	 -17=User mobility - 03=Logoff trap delay (enter delay time))	 - User Mobility - Trap delay (0 - 600 seconds) (enter delay time)
Logoff Error Count  → page 202	 -17=User mobility - 04=Logoff error count (dlear to set to 0)	 - User Mobility - Number of local/remote failed logoffs (read only)
Timer High Priority  → page 225	 -17=User mobility - 05=Timer HighPriority (enter delay time)	 - User Mobility - Timeout before saving high priority user data (0 - 999 seconds (enter delay time)
Count Medium Priority  → page 173	 -17=User mobility - 06=Count MedPriority (enter number)	 - User Mobility - Medium Priority peg count threshold (1 - 999) (enter number)
Timer Medium Priority  → page 225	 -17=User mobility - 07=Timer MedPriority (enter delay time))	 - User Mobility - Timeout before saving medium priority user data (enter delay time)
Mobility feature  → page 203	 -17=User mobility - 08=Mobility feature (read-only)	 - User Mobility - User Mobility feature status: (read only)
Managed Profile  → page 202	 -17=User mobility - 09=Managed profile (enable/disable)	 - User Mobility - DLS Managed profile (enable/disable)
International Mob ID  → page 204	 -17=User mobility - 10=International Mob ID (enable/disable)	 - User Mobility - International Mobility ID (enable/disable)
Force logoff to basic user  → page 186	not applicable	 - User Mobility - Force logoff to basic user (enable/disable)

SIP Security Configuration

Overview

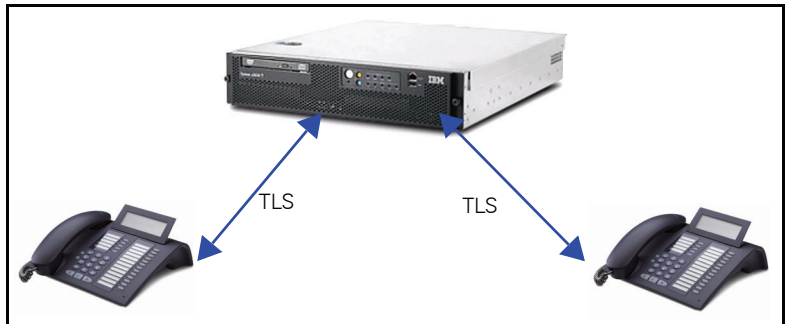
Aim: secure voice transmission

The aim of SIP security configuration is secure voice transmission between telephones.



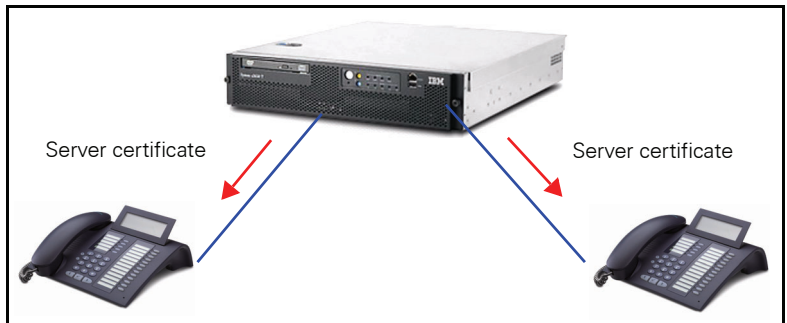
Prerequisite: secure signal transmission

Secure signalling must first be provided for, in order to facilitate secure voice transmission.



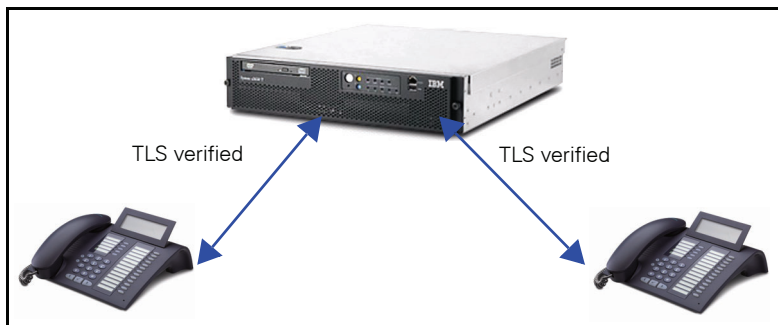
Additional security: server authentication

The server (HiPath 8000) must authenticate with the telephone.



Additional feature: continuous connection verification

This mechanism is used to check whether there is still a TLS connection, also then when the phone is not in use. If it determines that the original server is not reachable, the phone can switch to a redundant server (if e.g. DNS-SRV is used). This reduces longer waiting times for the user if the phone would have to first register with the redundant server for the next call and the call is then set up.



Implementation

optiPoint 410/420 SIP telephone settings

The following settings must be performed on the telephones for which voice encryption is to be enabled:

- Set payload security to **allowed** (→ page 144)
- Set connectivity check interval as required (→ page 144)
 - (0, when deactivated; 10-3600 = duration of verification in seconds; 1-9 = 10; >3600 = 3600)
- Set SIP server validation as required (→ page 144)
- Set SIP transport to TLS (→ page 115)
- Set SIP port to 5061 (→ page 115)

The same settings can also be performed using the telephone menu. In the case of web settings, references to the corresponding locations are provided.

Settings via DLS

If you prefer DLS configuration, settings are performed in the **SIP Settings** tab in the **IP Phone Configuration/VOIP Security** area, and

Deployment Service V2

VoIP Security Job

Object Edit View Action Help

Views: ☐ Search ☒ Object ☐ Table ☐ Template

IP Address:	10.23.15.12	SW Version:	V7 R0.11.0
Device ID:	00:01:E3:23:70:11	SW Type:	Siemens SIP
Device Type:	optiPoint 420 economy	Reg-Address:	10.23.22.102
E.164:	498972115132	Last Registration:	2007-07-30 11:26:5
Basic E.164:	498972115132		

Remark:

HFA Key Distribution | TLS Server CA Certificates | **SIP Settings** | HFA Settings

SIP Transport Protocol:

SIP Backup Transport Protocol:

☒ Payload security allowed

Connectivity check interval:

☒ TLS server validation

☐ TLS backup server validation

in the **SIP Registration 1** tab in the **IP Phone Configuration/Gateway/Server** area:

Deployment Service V2

Gateway / Server

Object Edit View Action Help

Views: Search Object Table Template

Deployment Service

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Gateway / Server

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Keysets / Keylayo

WLAN Settings

VoIP Security

IEEE 802.1x

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IP Client Configuration

IP Gateway Configura

IP Device Interaction

IP Device Managemen

IP Address: 10.23.15.12

SW Version: V7 R0.11.0

Device ID: 00:01:E3:23:70:11

SW Type: Siemens SIP

Device Type: optiPoint 420 economy

Last Registration: 2007-07-30 11:26:5

E.164: 498972115132

Basic E.164: 498972115132

Remark:

Gateway Gateway (Standby) SIP Terminal Settings SIP Registering 1 SIP Registering 2 SIP Survivabilit

SIP Routing: Server

SIP Gateway Addr: 0.0.0.0 SIP Gateway Port:

SIP Registrar Addr: 10.23.22.102 SIP Registrar Port:

SIP Phone Port: 5061

RTP Base Port: 5004

HiPath 8000 settings :

Via Assistant Common Management portal: if the Registration Type is set to Dynamic, only the Transport Type must be set to TLS:

Subscriber: MCHH - 498972115112

Subscriber Description

General User SIP Advanced Hunt Groups

Phone

The name for this SIP phone is used during the registration of this phone.

Name: 498972115112

Type: Dynamic

Transport Protocol: TLS

IP Address: 10.23.15.10 Port: 45678

View Phone

SIP Phone:

Executive Secretary Configuration

Associated Proxy Device:

Remote Phone

Handling server certificates

These certificates are used for server authentication.

Using the standard HiPath 8000 certificate

The root certificate root.pem is exported via FTP from the directory /usr/local/ssl/certs.

The certificate is then imported to DLS, activated, and distributed to the telephones.

SIP Server CA Certificate 1 tab in the **IP Phone Configuration/VOIP Security** area:

Deployment Service V2

Attention

VoIP Security

Job ID:

Object Edit View Action Help

Views: Search Object Table Template

IP Address: 10.23.15.12

SW Version: V7 R0.11.0

Device ID: 00:01:E3:23:70:11

SW Type: Siemens SIP

Device Type: optiPoint 420 economy

Reg-Address: 10.23.22.102

E.164: 498972115132

Last Registration: 2007-07-30 11:26:50

Basic E.164: 498972115132

Remark:

HFA Key Distribution TLS Server CA Certificates SIP Settings HFA Settings

Table Selected entry 1 / 1

Index: 0

Status Active/Import: no active certificate ☒ Activate certificate

Serial Number: 2

Owner: CN=hiQ8000,O=SEN,L=Munich,ST=Germ

Issuer: CN=Server CA,O=SEN,L=Munich,ST=Germ

Valid from: 2007-02-07 17:42:26

Valid to: 2017-02-04 17:42:26

Fingerprint (SHA1): 2EBB6B681797B146608451CFBE1E20F

Expires in ... [days]:

Alarm Status:

Save Discard Read

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 **Attention** There are two ways to delete the certificates. Either a factory reset of the phone will delete the certificate or the certificate must be deleted in the DLS, then the empty certificate must be activated and finally saved so that the change is send to the phone.

Web Interface

Establishing the Connection to the Phone

You can display and configure device and network information for the optiPoint 410/420 /economy/economy plus/standard/advance S V7.0 and the optiPoint 410 entry S V7.0 through the Web Interface.

You can access the Web Interface using one of the following web browsers:

- Microsoft Internet Explorer (recent version recommended)

To access the Web Interface, perform the following steps.

Open a web browser and enter the URL of the web page for the phone as follows:

https://[address]

where [address] is the IP address or host name of the Phone.

You can access the web interface in the browser using the host name assigned to your telephone. The presetting for the host name is the current E164 number. An example for the browser call is:

https://hostname.domainname

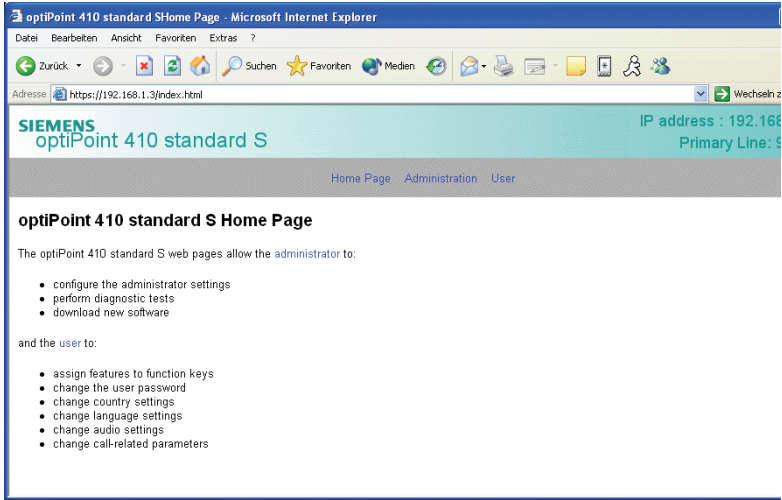
For configuring the phone's IP address see chapter "Basic Administration" on page 48.

For example, the configuration page for the Phone with the IP address 192.168.1.137 is: "https://192.168.1.137".

If applicable, confirm the following advisory message with "Yes".



A screen like the following home page appears:



 Click on the required field in the dialogs to see a description for each parameter.

Access to the Web Interface Administrator Menu

The following steps describe the access to the administrator menu, starting from the home page of the optiPoint 410/420.

1. Click on the link "Administration". The following login dialog appears:

A small dialog box titled "Local administrator login" with a red header bar. It contains a "Password:" label followed by a text input field. At the bottom, there are two buttons: "Login" and "Cancel".

2. Enter the administrator password (default: "123456"; max length 24 digits) and confirm.

Administrations Menu (optiPoint 410/420 advance standard)

Administrator menu

- General Information
- Network IP and Routing
- System
 - SIP environment
 - SIP features
- Quality of service
- File transfer and phone download settings
- Time and date
- SNMP
- Speech
- Ring settings
- LAN Port settings
- Multiline operation
- Function keys...
 - Phone
 - Key module 1
 - Key module 2
 - Key module SLK 1
 - Key module SLK 2
- Dial plan
- Dialing properties
- Feature Access
- User Mobility
- Configuration Management...
 - Settings
 - Check for updates
 - Error log
- Applications...
 - Directory
 - Address Book
 - WAP
- Upload/Download...
 - Upload configuration
 - Download application
 - Download configuration
 - Download hold music
- Diagnostics and statistics...
 - Non user-assisted tests
 - User-assisted tests
 - RTP Statistics
 - QoS Data Collection
 - Fault investigation
 - Simplified trace page
- Security
- Restart terminal
- Reset user password
- Change admin password
- Clear all user data
- Restore factory setting
- Port Control
- FPN Port Settings
- Survivability

Fault Investigation

- Trace Configuration
- Trace View
- Simplified trace page
- FTP Client
- Exception Data
- Windview Configuration
- SIP UDP Trace

[Home](#)

Administrations Menu (optiPoint 410 entry/economy)

Administrator menu

- General Information
- Network IP and Routing
- System
 - SIP environment
 - SIP features
- Quality of service
- File transfer and phone download settings
- Time and date
- SNMP
- Speech
- Ringer settings
- LAN Port settings
- Multiline operation
- Function keys...
 - Phone
- Dial plan
- Dialing properties
- Feature Access
- User Mobility (not used with optPoint entry)
- Configuration Management...
 - Settings
 - Check for updates
 - Error log
- Upload/Download...
 - Upload configuration
 - Download application
 - Download configuration
 - Download hold music
- Diagnostics and statistics...
 - Non user-assisted tests
 - User-assisted tests
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Fault Investigation

- Trace Configuration
- Trace View
- Simplified trace page
- FTP Client
- Exception Data
- Windview Configuration
- SIP UDP Trace

Web Pages



Click on the required field to navigate to the description of a parameter (e.g. move cursor over "Application: 2.3.7" and press the left mouse button to get to the description in the alphabetical reference chapter).

The links after the symbol  lead to the administration tasks with menu paths.

If DNS is applicable → page 41, the fields for entering the IP addresses on the following web pages have the addition " or DNS name":

- → SIP environment
- → File transfer
- → Time and date
- → SNMP
- → Non user-assisted tests

General information

 Display static Phone Information → page 94, Display Application Version → page 91, Display Upload/Download Status → page 91, Dial Plan Configuration and Status → page 70.

General Information

MAC address: 00:01:e3:25:66:aa

Versions:

Application:	6.0.53.1
SIP stack:	3.6.4.8
SIP signaling:	0.0.1
Web content:	6.0.53.1

Dial plan name & status:

None

File Transfer status:

	Last transfer	Status
Application file download:	20:38, 14 Jan 2004	OK
Configuration file download:	03:26, 8 Jan 2004	failed
Configuration file upload:	10:50, 27 Jul 2006	OK
Hold music file download:	never	-
System configuration download:	never	-
Phone configuration download:	never	-
Personal directory import:	never	-
Personal directory export:	never	-

Network IP and routing

 Network Addresses → page 57

Network IP Address and Routing



WARNING
If you make changes to the fields marked with an asterisk (*) you will have to restart the terminal manually before they take effect.

DHCP ☒ *

Terminal IP address: 192.168.1.14 *

Terminal mask: 255.255.255.0 *

Primary DNS IP address: 0.0.0.0

Secondary DNS IP address: 0.0.0.0

Default Route: 192.168.1.251

Domain Name:

Use dynamic hostname concept ☒

Terminal hostname: 1234 *

IP Routing:

Route 1: 0.0.0.0

Route 2: 0.0.0.0

Gateway 1: 0.0.0.0

Gateway 2: 0.0.0.0

Mask 1: 0.0.0.0

Mask 2: 0.0.0.0

Submit

Reset

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SIP environment

 Configure System Information → page 61

SIP Environment


WARNING
 If you make changes to the fields marked with an asterisk (*) you will have to restart the terminal manually before they take effect.

Terminal details:

Phone number:

Phone name:

Register by name: ☐

Display ID:

Use Display ID: ☐

SIP details:

SIP routing:

Registrar IP address or DNS name: Port:

Server IP address or DNS name: Port:

Gateway IP address or DNS name: Port:

SIP port:

RTP Base port: *

Outbound proxy: ☐

Default OBP domain name:

SIP transport: *

SIP server type: *

SIP session timer enabled: ☐

SIP session timer value: seconds

Registration timer value: seconds

SIP realm:

SIP user ID:

New SIP password:

Confirm SIP password:

Transaction timer: milliseconds, default=32000

Miscellaneous:

Message Waiting IP address or DNS name:

Emergency number:

Voicemail number:

Branding/identity name:

SIP features

 SIP Features → page 64 Transfer on Ringing

SIP Features

Call handling options

Auto answer: ☐ Off

Beep on auto answer: ☐ Off

Refuse call feature enabled: ☒ On

Allow Join in conference: ☒ On

Auto reconnect: ☐ Off

Beep on auto reconnect: ☐ Off

Allow transfer on ringing: ☒ On

Allow transfer on hangup: ☒ On

Group pickup

Group pickup URI:

Hot line/warm line working

Phone type:

Deactivated

Default dial string:

Initial digit timer

Initial digit timer value: seconds (1 to 120)

Station-controlled conference

Conference server URI:

Call park URI:

Call pickup URI:

Callback

Callback - busy URI:

Callback - no reply URI:

Cancel callbacks URI:

Call Recorder

Number:

Audible indication: ☒ On

Auto start: ☐ Off

Submit

Reset

Quality of Service

 Quality of Service (QoS) → page 58

Quality of Service (QoS)

**WARNING**
If you make changes to any of the fields on this page you will have to restart the terminal manually before they take effect.

Layer 2	Layer 3
Required: ☒	☒
Voice: 5	EF 
Signaling: 3	AF31 
Default: 0	EF AF11 AF12 AF13 AF21 AF22 AF23 AF31 AF32 AF33 AF41 AF42 AF43
VLAN Discovery: Manual 	
Manual VLAN id: 0	
Submit	Reset

File transfer

optiPoint 410/420 standard/advance

 Software Update/Transferring Files → page 74

FTP Settings and Phone Downloads



WARNING
Note that file transfers to and from the download server will take place over an insecure link.

FTP Server Address: 192.168.1.148

FTP account name:

FTP username: sipuser

New FTP password:

Confirm new FTP password:

FTP path: .

Use Passive Mode FTP ☒

Application filename: 410std_SIP_V6_R1_65_3.app

Configuration filename: opti410c

Hold music filename: opti410.moh

Logo filename: logo.bmp

Java Program filename:

LDAP template filename: template.txt

DSM firmware filename:

Action on submit: No transfer

Submit

Reset

optiPoint 410/420 economy/economy plus and optiPoint 410 entry

FTP Settings and Phone Downloads

**WARNING**
Note that file transfers to and from the download server will take place over an insecure link.

FTP Server Address:

FTP account name:

FTP username:

New FTP password:

Confirm new FTP password:

FTP path:

Use Passive Mode FTP ☐

Application filename:

Configuration filename:

Hold music filename:

Action on submit:

 Software Update/Transferring Files → page 74

Time and date

 Configuring Date and Time → page 67

Time & date

SNTP server IP address: 0.0.0.0

Entering a blank (or invalid) address will cause the SNTP client to listen for broadcasts.

Timezone offset (hours): 0

This is the number of hours by which time sourced via SNTP (UTC) must be adjusted to give local time. Valid range is -12 to +12.

☐ Daylight saving

Local Time (hh:mm): 18 : 41

Date (day, month, year): 1 January 2004

Date Format: DD.MM.YY

Note - If SNTP is being used, a user-specified value for Time & Date will be overwritten when an SNTP update next occurs.

Submit

Reset

SNMP

[A](#) Use SNMP → page 92

SNMP Settings

Send trap on unauthorised logoff attempt: ☐

Trap delay (0 - 600 seconds):

Management Center Address:

Management Center Port:

Send Generic Traps to Management Center: ☐

Send QDC Traps to Management Center: ☐

The password fields must be left blank unless you want to change the password.

New Query password:

Confirm new Query password:

Queries Allowed: ☒ [Modifiable via Port Control](#)

SNMP MIB2 errors:

Discarded inbound packets:	0
Invalid inbound packets:	0
Discarded outbound packets:	0
Invalid outbound packets:	0

Speech

[A](#) Change Speech Parameters → page 93

Speech

Codec:

Audio mode:

RTP packet size:

Silence Suppression: ☐

Microphone Disable: ☒

Ringer settings

 Configure Ringer Settings → page 94

Ringer Settings

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

Alert indication	Melody (0 = off, or 1 to 8)	Tone (1 to 3)	Duration (sec)
alert external	2	1	60
belcore-dr2	5	1	10
alert-internal	1	3	5
alert-priority	1	2	30

Submit

Reset

LAN port settings

 Configure LAN Ports → page 60

WARNING
 If you make changes to any of the fields on this page you will have to restart the terminal manually before they take effect.

Select the settings for LAN ports 1 and 2 on the terminal:

LAN port 1 (LAN) ☒ Auto ☐ 10 Mb/s Full duplex ☐ 10 Mb/s Half duplex ☐ 100 Mb/s Full duplex ☐ 100 Mb/s Half duplex	LAN port 2 (PC)¹ ☒ Auto ☐ 10 Mb/s Full duplex ☐ 10 Mb/s Half duplex ☐ 100 Mb/s Full duplex ☐ 100 Mb/s Half duplex
--	---

¹ Only used with optiPoint 410/420 standard/advance

Multiline operation

 Multiline → page 68

Multiline Operation

Originating line preference:

Terminating line preference:

Line key operation mode:

Rollover type:

Rollover volume:

Registration LEDs: ☒

Show focus: ☒

Use LED to indicate Remote Forwarding: ☐

Reservation timer: seconds

Note: set reservation timer to 0 to disable line reservation

Function keys

Function Key assignments → page 74

Phone function key assignments

Key	Normal function	Shifted function
1:	Loudspeaker	
2:	Repeat dialing	
3:	Missed calls	
4:	Swap screens	
5:	Line	
6:	Selected dialing	
7:	Repertory dial	
8:	DSS	
9:	State Key	
10:		
11:		
12:		
13:		
14:		
15:		
16:		
17:		
18:		
19:	Shift	Shift

Unsaved changes

WARNING There may be unsaved changes to the function key assignments. If you want to save them, click on the **Save changes** button.

Parameters

Parameters for function 'Line' assigned to Key 5

Name	Value
Primary Line	Yes
Ring	Yes
Hunt ranking	1
Address of record	123456789
Realm	customserver
User ID	123456789
Shared type	Shared
Hot line	No
Hot line dial string	
Intrusion allowed	No
Short description	
Overview position on DSM	
Hide on DSM	No

Function key "Select a function" optiPoint 410/420 advance/standard

Assign a function to key 5

Select a function: Abbreviated dialing

Locked by administrator: ☐

Apply Cancel

- Abbreviated dialing
- Alternate
- Blind Transfer
- Callback
- Call park
- Call pickup
- Call Recording
- Cancel callbacks
- Cancel/Release
- Clear definition
- Conference
- Confirm
- Consult/Transfer
- Deflect
- Do not disturb
- DSS
- Feature toggle
- Forwarding
- Group pickup
- Headset
- Hold
- Join
- Line
- Local conference
- Loudspeaker
- Missed calls
- Mute
- Notebook
- Phone lock
- Repeat dialing
- Repertory dial
- Ringer off
- Room echoing
- Room muffled
- Selected dialing
- Settings
- Setup menu
- Shift
- State Key
- Swap screens
- Voice Messages

Keys can be locked by Administrator for the user can't change the contents.

Example:

Assign a function to key 5

Select a function: Deflect

Locked by administrator: ☒ On

Apply Cancel

[A](#) Function Key assignments → page 74

Function key "Select a function" optiPoint 410/420/economy/economy plus

Assign a function to key 5

Select a function:

Abbreviated dialing

Abbreviated dialing

Alternate

Blind Transfer

Callback

Call park

Call pickup

Cancel callbacks

Cancel/Release

Clear definition

Conference

Confirm

Consult/Transfer

Deflect

Do not disturb

DSS

Feature toggle

Forwarding

Group pickup

Headset

Hold

Join

Line

Loudspeaker

Missed calls

Mute

Notebook

Phone lock

Repeat dialing

Repertory dial

Ringer off

Selected dialing

Settings

Setup menu

Shift

State Key

Voice Messages

Locked by administrator:

Apply

Cancel

Function key "Select a function" optiPoint 410 entry

Assign a function to key 4

Select a function: Callback

Locked by administrator: ☐

Apply Cancel

- Callback
- Cancel callbacks
- Cancel/Release
- Clear definition
- Confirm
- Do not disturb
- DSS
- Feature toggle
- Group pickup
- Line
- Loudspeaker
- Mute
- Phone lock
- Repeat dialing
- Repertory dial
- Ringer off
- Selected dialing
- Shift
- State Key
- Voice Messages

Function key "Select a function" optiPoint 420

See also → Function key "Select a function" optiPoint 420

A Function Key assignments → page 74

Assign a function to key 14

Select a function: Loudspeaker

Locked by administrator: ☐ Off

Key Label: Loudspeaker

Apply Cancel

Function key "Line key" optiPoint 410/420 advance/standard

 Multiline → page 68

Assign a function to key 7

Select a function: Line

Parameters for function 'Line'

Primary line: ☒ On

Ring: ☒ On

Hunt ranking: 1

Address of record: (max 48 char)

Realm: (max 48 char)

User ID: (max 48 char)

Password: (max 48 char)

Confirm password:

Shared type: Shared

Hot line: ☐ Off

Hot line dial string:

Intrusion allowed: ☐ Off

Short description:

Overview position on DSM:

Hide on DSM: ☐ Off

Apply

Cancel

Function key "Line key" optiPoint 410/420 economy/economy plus and optiPoint 410 entry

Assign a function to key 5

Select a function: Line

Parameters for function 'Line'

Primary line: ☒ On

Ring: ☒ On

Hunt ranking: 1

Address of record: * (max 48 char)

Realm: * (max 48 char)

User ID: * (max 48 char)

Password: * (max 48 char)

Confirm password: *

Shared type: Shared

Hot line: ☐ Off

Hot line dial string:

Intrusion allowed: ☐ Off

Apply Cancel

Function key "Selected dialing"

A Function Key assignments → page 74

Assign a function to key 6

Select a function: Selected dialing

Parameter for function 'Selected dialing'

Locked by administrator: ☐ Off

Dial string: 28 chars left

Apply Cancel

Function key "Repertory dial"

 Function Key assignments → page 74

Assign a function to key 6

Select a function: Repertory dial

Parameter for function 'Repertory dial'

Locked by administrator: Off

Dial string: <PA>5618239954invalid string

Append codes: OKClearConsultPause

ApplyCancel

Function key "Feature Toggle"

 Function Key assignments → page 74

Assign a function to key 6

Select a function: Feature toggle

Parameters for function 'Feature toggle'

Locked by administrator: Off

Feature Code: 123456789038 chars left

Description: Make line busy - Stop hunt

ApplyCancel

Function key "DSS"

Precondition: A line key has to be configured before.

 Function Key assignments → page 74

Assign a function to key 6

Select a function: DSS

Parameters for function 'DSS'

Address of record: 9952* (max 48 char)

Realm: 192.168.1.2* (max 48 char)

User ID: 9952* (max 48 char)

Password: ****** (max 48 char)

Confirm password: ******

ApplyCancel

Function key „HotDesking“

The key can only be configured if the telephone has been preset in the Hi-Path 8000 as guest telephone by the administrator. For more Information see user manual.

Assign a function to key 6

Select a function: State Key

Parameters for function 'State Key'

Locked by administrator: ☐ Off

Feature Code: #60 45 chars left

Feature URI: HDLogonToggle

Description: HotDesking

Apply Cancel

Dial plan

[A](#) Dial Plan Configuration and Status → page 70

Dial Plan

Name:

Status: None

Displaying: File

Action: Enable

Digit string C/S Min Max Tmr Trm

Display file Display memory

Dialling Properties

optiPoint 410/420 advance/standard

 Dialling Properties → page 71

Dialling Properties

External Access Code:

9

International Dial Prefix:

00

Local Country Code:

49

National Dial Prefix:

0

Local Area Code:

89

Local District Code:

722

Submit

Reset

Feature Access

optiPoint 410/420 standard/advance (example)

Feature access

This page allows you to control which features are available to the User on this phone.

Checking the Status box allows the user to access the feature, but the feature may also need enabling at the phone for it to become active.

Feature	Status	Feature	Status
Auto answer - CTI	☒	Auto reconnect - CTI	☒
Callback - busy	☒	Callback - no reply	☒
Call deflect	☒	Call display by name	☒
Call display by number	☒	Call duration	☒
Call forwarding	☒	Call hold (explicit)	☒
Call join	☒	Call park	☒
Call pickup	☒	Call recording	☒
Call transfer	☒	Call waiting	☒
Do not disturb	☒	DSM - Contacts	☒
DSM - Address book	☒	DSM - Speed dial	☒
DSM - Call control	☒	DSM - Voice recognition	☒
DSM - WAP browser	☒	GPU New Call Beep	☒
Hot keypad dialing	☒	Local conference	☒
Log forwarded calls	☒	Message waiting	☒
Music on hold	☒		

optiPoint 410/420 economy/economy plus (example)

Feature access

This page allows you to control which features are available to the User on this phone.

Checking the Status box allows the user to access the feature, but the feature may also need enabling at the phone for it to become active.

Feature	Status	Feature	Status
Auto answer - CTI	☒	Auto reconnect - CTI	☒
Callback - busy	☒	Callback - no reply	☐
Call deflect	☒	Call display by name	☒
Call display by number	☒	Call duration	☒
Call forwarding	☒	Call hold (explicit)	☒
Call join	☒	Call park	☒
Call pickup	☒	Call recording	☒
Call transfer	☒	Call waiting	☒
Do not disturb	☒	DSM - Contacts	☐
DSM - Address book	☐	DSM - Speed dial	☐
DSM - Call control	☐	DSM - Voice recognition	☐
DSM - WAP browser	☐	Hot keypad dialing	☒
Local conference	☐	Log forwarded calls	☒
Message waiting	☒	Music on hold	☒

¹ Only used with optiPoint 410/420 economy plus

optiPoint 410/420 entry (example)

Feature access

This page allows you to control which features are available to the User on this phone.

Checking the Status box allows the user to access the feature, but the feature may also need enabling at the phone for it to become active.

Feature	Status
Auto answer - CTI	☒
Callback - busy	☒
Call deflect	☐
Call display by number	☐
Call forwarding	☐
Call join	☐
Call pickup	☐
Call transfer	☐
Do not disturb	☒
DSM - Address book	☐
DSM - Call control	☐
DSM - WAP browser	☐
Hot keypad dialing	☒
Log forwarded calls	☐
Music on hold	☒

Feature	Status
Auto reconnect - CTI	☐
Callback - no reply	☐
Call display by name	☐
Call duration	☐
Call hold (explicit)	☐
Call park	☐
Call recording	☐
Call waiting	☐
DSM - Contacts	☐
DSM - Speed dial	☐
DSM - Voice recognition	☐
GPU New Call Beep	☒
Local conference	☐
Message waiting	☒

Submit

Reset

[A](#) Feature Access → page 73

User Mobility

[A](#) User Mobility (Hipath 8000) → page 102

User Mobility

Send trap on unauthorised logoff attempt:

☐

International Mobility ID:

☒

Trap delay (0 - 600 seconds):

300

Number of local failed logoffs:

0

Number of remote failed logoffs:

0

Medium Priority peg count threshold (1 - 999):

5

Timeout before saving medium priority user data (1 - 999 seconds):

60

Timeout before saving high priority user data (0 - 999 seconds):

5

User Mobility feature status:

This phone is mobility-enabled

DLS Managed profile:

☐

Force logoff to basic user:

☐

Submit

Reset

Configuration Management

Configuration management settings

A Specify configuration update file → page 88

Configuration management settings



DHCP has been turned off or has failed, so the download mechanism needs to be configured manually. You will have to enter values into the fields marked with an asterisk (*) for the required mechanism(s), click on the corresponding check box and click the Submit button.

Deployment service (DLS) (In use)

IP address or DNS name: *

Port: *

Use deployment service ☒

Secure configuration download (HTTPS) (Not in use)

IP address or DNS name: *

Port: *

File path for URL:

Use secure configuration download ☐

Non-secure configuration download (FTP) (Not in use)

IP address or DNS name: *

Port: *

Use non-secure configuration download ☐

FTP account name:

FTP username:

New FTP password:

Confirm new FTP password:

FTP path:

Common settings (HTTPS and FTP)

Configuration file name:

Configuration file type:

After registration, check for updates every: seconds (0 = don't check)

If registration fails, check for updates every: seconds

Authentication enabled: ☐

Check for updates

 Specify configuration update file → page 88



WARNING

Clicking on **Check for updates** will check that the terminal's configuration files are up to date. If they are out of date, newer versions will be downloaded.

Note that the terminal will be restarted even if the files are up to date.

Check for updates

Configuration Management Log file

 Specify configuration update file → page 88

This log file shows errors if occur while using XML for configuration.

Configuration Management Log file

This page displays the contents of the configuration management error log

Empty log file

There is no log file.

Applications (optiPoint 410/420 standard/advance)

Directory

 Applications → page 79

Directory

LDAP server address: 0.0.0.0

Port: 389

SubmitReset

Address Book

[A](#) Address Book → page 162

Address Book

Address Book server address:

192.168.1.148

FTP account name:

FTP username:

sipuser

New FTP password:

Confirm new FTP password:

FTP path:

.

FTP filename:

420buch.csv

Action on submit:

No transfer

Submit

Reset

WAP

[A](#) WAP Settings → page 81

WAP Settings

WAP Server address:

0.0.0.0

Port:

9200

WAP mode:

HTTP

WAP proxy Username:

New WAP proxy password:

Confirm new WAP proxy password:

Help internet URL:

Home page:

Submit

Reset

Upload configuration

 Software Update/Transferring Files → page 74

This operation will upload (save) the terminal's current configuration data to file **opti410c** on the fileserver using the settings in the [File transfer](#) page. The upload will take place over an insecure link.

Upload

Download application

 Software Update/Transferring Files → page 74

This operation will download application file **410adv_Sip_4_1_9.app** from the fileserver to the terminal using the settings in the [File transfer](#) page. The download will take place over an insecure link.

When you click on **Download**, the application will be downloaded and validated. If it is valid, the new file will replace the existing file and the terminal will restart automatically.



WARNING

This operation will take approximately 2 to 3 minutes during which the terminal will be restarted twice. You will not be able to reconnect to the terminal's web pages until this sequence is complete.

Download

Download configuration

 Software Update/Transferring Files → page 74

This operation will download configuration file **opti410c** from the fileserver to the terminal using the settings in the [File transfer](#) page. The download will take place over an insecure link.



WARNING

After the configuration data is downloaded, it will replace the existing data and **the terminal will be restarted automatically.**

Download

Download hold music

A Software Update/Transferring Files → page 74

This operation will download hold music file **opti410.moh** from the fileserver to the terminal using the settings in the [File transfer](#) page. The download will take place over an insecure link.

Download

Non user-assisted tests

A Non user-assisted diagnostic tests → page 95

Non User-Assisted Tests

Select which tests to invoke:

- ☒ Ping -- IP Address:
- ☒ ROM test
- ☒ RAM test

This page will be reloaded with the updated test results after the tests have been invoked.

Invoke

Reset

Test results

	Last run at	Result
Ping:	19:41, 1 Jan 2004	passed (192.168.1.151, T=10ms)
RAM test:	19:41, 1 Jan 2004	passed
ROM test:	19:41, 1 Jan 2004	passed

User-assisted tests

 User-assisted diagnostic tests → page 96

optiPoint 410/420 standard/advance

User-Assisted Tests

The diagnostic tests on this page require a local user to confirm the outcome at the terminal.

Note that once a particular test has been started, the local user will assume full control of the test (using the keypad) until the test is terminated.

Run the required tests:

Display test:

LED test:

Key test:

Audio loop test:

Line monitor:

Self Labelling Keys test::

optiPoint 410/420 economy/economy plus and optiPoint 410 entry

User-Assisted Tests

The diagnostic tests on this page require a local user to confirm the outcome at the terminal.

Note that once a particular test has been started, the local user will assume full control of the test (using the keypad) until the test is terminated.

Run the required tests:

Display test:

LED test:

Key test:

Audio loop test:

Line monitor:

Not used with
optiPoint 410 entry

RTP Statistics

These parameters are used for development only.

RTP Statistics

Address-of-record:

From:

To:

Call Id:

Request URI:

Transport addr - local:

Transport addr - remote:

Session begin:

Session end:

Codec:

Frame size:

Jitter:

Av. round trip delay:

Packets received:

Packets lost:

Packets discarded due to:

out-of-sequence/delay:

buffer overrun:

other:

<< < records > >>

QoS Data Collection

These parameters are used for development only.

QoS Data Collection

Send QoS Reports to QCU:

☐

QCU home address:

QCU port number:

Report mode:

OFF

Report interval:

60

seconds

Observation interval:

10

seconds

Minimal session length:

20

* 100 milliseconds

Codec Independent Threshold Values

Max jitter threshold:

15

milliseconds

Avg round trip delay threshold:

100

milliseconds

Non-compressing Codec Threshold Values

Lost packet thresholds:

10

per 1000 packets

Consecutive lost packets:

2

Consecutive good packets:

8

Compressing Codec Threshold Values

Lost packet thresholds:

10

per 1000 packets

Consecutive lost packets:

2

Consecutive good packets:

8

Session data:

[Click here](#)

Submit

Reset

Submit & Resend Last Reports

142

Session data

These parameters are used for development only.

 → page 142

Select a report to view: None available

Session data
Start of report period:
End of report period:
IP address (local):
Port number (local):
IP address (remote):
Port number (remote):
SSRC (receiving):
SSRC (sending):
Codec:
Maximum packet size:
Silence suppression:
Count of good packets:
Maximum jitter:
Maximum inter-arrival jitter:
Periods jitter threshold exceeded:
Round trip delay:
Round trip delay threshold exceeded:
Count of lost packets:
Count of discarded packets:
Periods of lost packets:
Consecutive Packet Loss (CPL):
Periods of consecutive lost packets:
Consecutive good packets (CGP):
Periods of consecutive good packets:
Count of Jitter buffer overruns :
Count of Jitter buffer under-runs :
Codec Change on the fly:
Periods with at least one threshold exceeded:
HiPath Switch ID:
LTU Number:
Slot Number:
Endpoint Type :
Version:
Subscriber Number Type:
Subscriber Number:
Call ID:
MAC Address:

Reset user password

 Reset User Password → page 99

Reset User Password

This operation will reset the user password to the default value

Change admin password

 Change Administrator Password → page 99

Change administrator password

Current password:

New password:

Confirm new password:

Note: The password must be numeric and be at least 6 digits long.

Submit

Reset

Security

 Security settings → page 96

Security Settings



WARNING

If you enable or disable the fields marked with an asterisk (*) you will have to restart the terminal manually before they take effect.

Payload security allowed:

☒

Connectivity check interval (T):

0

* seconds (10 to 3600)

SIP server validation:

☐

Submit

Reset settings

Restart terminal

 Restart the optiPoint 410/420 → page 97

Restart terminal

This operation will restart the terminal immediately.
(You will lose your web connection to the terminal.)

Restart

Clear ALL user data

 Clear ALL user data → page 99

Clear ALL user data

This operation will clear out all personal data relating to the user (including Personal Directory entries and structure definition), ready for the phone to be given to a different user. It does not clear out Phone settings (such as contrast or touch screen calibration settings), nor Administration settings (such as network details).

Restore factory settings

[A](#) Restore Factory Settings → page 98

Restore factory settings

Please enter reset password

Port Control

[A](#) Port Control → page 99

Port control

This page allows you to disable selected services on the terminal (and close their associated port) when they are not required.

Checking the Status box enables the service; unchecking it disables the service.

Service	Status
Service Agent	☐
Test Interface	☐
SNMP Interface	☒
Resource Sharing	☒

FPN Port Settings

Only available in Web Interface

FPN Port Settings

**WARNING**
If you make changes to the fields marked with an asterisk (*) you will have to restart the terminal manually before they take effect.

SIP Server: 5060 *

SIP Registrar: 5060 *

SIP Gateway: 5060 *

SIP Dialog: 5060 *

SIP Phone Port: 5060 *

RTP Base Port: 5004 *

Service Agent: 5100 *

Web Server HTTP: 8085 *

Web Server HTTPS: 443 *

Config Update by FTP

FTP Server: 20

Config Update by HTTPS

HTTPS Server: *

Config Update by DLS

DLS Server: 18443

QDC: 12010

LDAP: 389

WAP: 9200

Java: 9200

Submit

Reset

146

Fault Investigation

Available tracing menus:

- Trace Configuration
- Trace View
- Simplified trace page
- FTP Client
- Exception Data
- Windview Configuration
- SIP UDP Trace

For information how to trace the optiPoint 410/420 please refer to the Tracing guide.

 → page 111 or → page 112

Survivability

 Survivability → page 100


WARNING
 If you make changes to any of the fields below you will have to restart the terminal manually before they take effect.

Backup details:

Backup IP address or DNS name: Port:

Backup Registration: ☒

Backup Outbound proxy: ☐

Backup SIP transport:

Backup Registration timer value: seconds

Phone Menu

➡ The Administration and Diagnostics options in the Setup menu are permanently not shown by using the ◀ and ▶ keys to navigate. Entry to these menus is gained by using its index number ("6" for Administration, "7" for Diagnostics).

Further information about accessing the administrator menu see → page 48.

Phone Menu Structure Overview

optiPoint 410/420 economy/economy plus/standard/advance

➡ Some user related menu items were omitted. For this items see User Manual.

Menu	Default	Editor
... 05=Setup?		
... 2=Configuration?		
... 02=Language?	English	
03=Date/time?	DD.MM.YY	
... 14=Daylight Saving?	Off	
... 32=Auto answer	Off	
33=Beep on auto answer	On	
34=Auto reconnect	Off	
35=Beep on auto reconnect	On	
... 6=Administration?		Password required
01=Network?		
01=DHCP IP assign?	Off	→ Switch Editor
02=Terminal IP addr.?	0.0.0.0	→ IP Number Editor
03=Terminal mask?	0.0.0.0	→ IP Number Editor
04=Default route?	0.0.0.0	→ IP Number Editor
05=IP routing?		
1=Route 1?		
1=IP address 1?	0.0.0.0	→ IP Number Editor
2=Gateway 1?	0.0.0.0	→ IP Number Editor
3=Mask 1?	0.0.0.0	→ IP Number Editor
0=End?		
2=Route 2?		
1=IP address 2?	0.0.0.0	→ IP Number Editor
2=Gateway 2?	0.0.0.0	→ IP Number Editor
3=Mask 2?	0.0.0.0	→ IP Number Editor
0=End?		
0=End?		
06=QoS?		
2=QoS L2/L3?		
1=L2Off/L3Off?	✓	
2=L2Off/L3On?		
3=L2On/L3On?		
0=End?		

Menu

Default Editor

---	3=L3 Voice?			L3 have to be activated
---	4=L3 Signalling?			L3 have to be activated
---	5=L2 Voice?			L2 have to be activated
---	6=L2 Signalling?			L2 have to be activated
---	7=L2 Default?			L2 have to be activated
---	8=Vlan ID?	0		
---	9=VLAN discovery?			
---	1=Manual?	✓		
---	2=DHCP?			
---	0=End?			
---	0=End?			
---	07=DNS Domain name?			→ Text Editor
---	08=Prim DNS IP addr?	0.0.0.0		→ IP Number Editor
---	09=Sec DNS IP addr?	0.0.0.0		→ IP Number Editor
---	10=LAN port settings?			
---	1=Auto?	✓		
---	2=10Mbit/Full?			
---	3=10Mbit/Half?			
---	4=100Mbit/Full?			
---	5=100Mbit/Half?			
---	0=End?			
---	11=PC port settings?			
---	1=Auto?	✓		
---	2=10Mbit/Full?			
---	3=10Mbit/Half?			
---	4=100Mbit/Full?			
---	5=100Mbit/Half?			
---	0=End?			
---	12=User specific web ad?			
---	00=End?			
---	02=System?			
---	01=Terminal Number?			→ Number Editor
---	02=Terminal Name?			→ Text Editor
---	03=Register by Name?	Off		→ Switch Editor
---	04=SIP Routing?			
---	1=Server?	✓		
---	2=Gateway?			
---	3=Direct?			
---	0=End?			
---	05=Outbound Proxy?	Off		→ Switch Editor
---	06=Default OBP Domain?			→ Text Editor
---	07=SIP Addresses?			
---	1=SIP Server?			
---	1=SIP Server Address?	0.0.0.0		→ IP Number Editor
---	2=SIP Server Port?	5060		→ Number Editor
---	0=End?			
---	2=SIP Registrar?			
---	1=SIP Registrar Address?	0.0.0.0		→ IP Number Editor
---	2=SIP Registrar Port?	5060		→ Number Editor
---	0=End?			
---	3=SIP Gateway?			
---	1=SIP Gateway Address?	0.0.0.0		→ IP Number Editor
---	2=SIP Gateway Port?	5060		→ Number Editor
---	0=End?			
---	5=SIP Phone?			
---	1=SIP Port?	5060		→ Number Editor
---	2=RTP Base Port?	5004		→ Number Editor
---	0=End?			
---	08=SIP Transport?			
---	1=UDP?			
---	2=TCP?			
---	0=End?			
---	09=SIP Realm?			→ Text Editor

Menu		Default	Editor
— 10=SIP User Id?			→ Number Editor
— 11=SIP Password?			→ Password Editor
— 12=SIP Session Timer?			→ Switch Editor
— 13=SIP Session Time?		Off	→ Number Editor
— 14=Emergency No.?		3600	→ Number Editor
— 15=Voice mail No.?			→ Number Editor
— 16=MWI Server Addr.?		0.0.0.0	→ IP Number Editor
— 17=Registration Timer?		3600	→ Number Editor
— 26=System Name?			→ Text Editor
— 27=Conf Server Addr.?		0.0.0.0	→ IP Number Editor
— 28=Keyset?			
— 01=Line			Select a line
— 1=Rank			
— 2=Ringer setting			
— 3=Hot Line			→ Number Editor
— 0=End			
— 02=Registration LEDs?		On	→ Switch Editor
— 03=Rollover type?			
— 1=No ring?		✓	
— 2=Alert ring?			
— 3=Standard ring?			
— 4=Alert beep?			
— 0=End?			
— 04=Rollover Vol?		2	→ Gradient Editor
— 05=Orig line pref?		✓	
— 1=Idle?			
— 2=Primary?			
— 3=Last?			
— 4=None?			
— 0=End?			
— 06=Term line pref?		✓	
— 1=Ringin?			
— 2=Ringin PLP?			
— 3=Incoming?			
— 4=Incoming PLP?			
— 5=None?			
— 0=End?			
— 07=Line action mode?		✓	
— 1=Hold ?			
— 2=Release?			
— 0=End?			
— 09=Show focus?		on	
— 10=Forwarding Ind?		off	
— 11=Reservation Timer?		60	
— 00=End?			
— 29=Dial Plan?		Off	→ Switch Editor
— 30=Group Pickup URI?		Notamemb.	
— 31=SIP Server type?			
— 1=HiQ/HiPath 8000?			
— 2=BroadSoft?			
— 3=Sylantro?			
— 4=Auto?			
— 0=End?			
— 32=Allow Refuse?		On	
— 33=Hot/Warm Phone?		Off	
— 1=Normal line Action?			
— 2=Warm line?			
— 3=Hot line?			
— 0=End?			
— 34=Wap?			Not economy/economy plus
— 1=Wap Mode?			
— 2=Wap Server Address?			
— 3=Wap Server Port?			
— 0=End			

Menu

Default Editor

---	35=LDAP?		Not economy/economy plus
---	1=LDAP Server Address?		
---	2=LDAP Server Port?		
---	3=External Code?		
---	4=International Prefix?		
---	5=Country Code?		
---	6=National Prefix?		
---	7=Area Code?		
---	8=District Code?		
---	0=End		
---	36=Java Proxy?		Not economy/economy plus
---	1=Java Proxy Address?		
---	2=Java Proxy Port?		
---	00=End?		
---	37=Feature Access?		
---	01=Call Hold Explicit?	off	
---	02=Call Deflection?		
---	03=Call Forwarding?		
---	04=Log Forwarded Calls?		
---	05=Call Duration?		
---	06=Call Waiting?		
---	07=Call Transfer?		
---	08=Call Join?		
---	09=Call Display Name?		
---	10=Call Display Number?		
---	11=Music On Hold?		
---	12=Do Not Disturb?		
---	13=Message Waiting?		
---	14=Local Conference?		Not economy
---	15=Auto Answer CTI?		
---	16=Auto Reconnect CTI?		
---	17=Call Park?		
---	18=Call Park Pickup?		
---	19=Wap browser On DSM?		Not economy/economy plus
---	20=Address Book?		Not economy/economy plus
---	21=DSM Call Control?		Not economy/economy plus
---	22=Voice Recognition?		
---	23=Speed Dial On DSM?		Not economy/economy plus
---	24=Contacts?		Not economy/economy plus
---	25=Hot Keypad Dialing?	on	
---	26=Callback-busy?	on	
---	27=Callback-no reply?	on	
---	28=Call recording?	on	
---	00=End?		
---	38=Call park URI?		
---	39=Call pickup URI?		
---	40=Transfer on Ringing?	on	
---	41=Join allowed in conf?	on	
---	42=Callback URIs?		
---	1=Callback-busy?		
---	2=Callback-no reply?		
---	3=Cancel callbacks?		
---	0=End		
---	43=Initial Digit Timer?	20	
---	44=Display ID?	20	
---	1=New/Change ID?		
---	2=Use Terminal ID?		
---	3=Use Display ID?		
---	0=End		
---	00=End?		
---	45=Call Recorder?		
---	46=Transaction timer?		
---	47=Payload Security All?		
---	48=Conn.Check Interval?		

Menu	Default	Editor
49= SIP Server Validate?		
03= File transfer?		
01= DL application?		
02= DL config?		
03= UL config?		
04= DL DSM Logo?		
05= DL hold music?		
06= DL server IP addr.?	0.0.0.0	→ IP Number Editor
07= FTP account name?	guest	→ Text Editor
08= FTP username?	guest	→ Text Editor
09= FTP password?		→ Password Editor
10= FTP path?		→ Text Editor
11= Applic. DL filename?	opti420a	→ Text Editor
12= Cfg DL filename?	opti420c	→ Text Editor
13= Logo filename?		
14= Hold music filename?	Not economy/economy plus	
15= Default Music?	opti400.moh	→ Text Editor
16= DL LDAP Template?		
17= LDAP Template filename?	Not economy/economy plus	
18= DL JavaApplet?	Not economy/economy plus	
19= JavaApplet Filename?	Not economy/economy plus	
00= End?		
04= Date/Time?		
01= SNTP IP addr?	0.0.0.0	→ IP Number Editor
02= Timezone offset?	+0	→ Offset Editor
0= End?		
05= SNMP?		
01= SNMP trap IP addr?		→ IP Number Editor
02= SNMP password?		→ Password Editor
0= End?		
06= Speech?		
02= Audio mode?		
1= HiQual preferred?	✓	
2= LoBand preferred?		
3= LoBand always?		
4= G711 preferred?		
0= End?		
03= Compression?		
1= G729?	✓	
2= G723?		
0= End?		
04= G711 Silence?	Off	→ Switch Editor
05= RTP Packet Size?		
01= Auto?		
02= 10ms?		
03= 20ms?	✓	
0= End?		
06= Microphone?		
1= Normal?	✓	
2= Disabled?		
0= End?		
0= End?		
07= General info?		
01= MAC address?		
02= Version Info?		
03= Last Restart?		
04= Dial Plan?		
01= ID?		
02= Status?		
0= End?		
0= End?		
08= Admin password?		→ Password Editor
11= Ringer Settings?		
01= Alert indications?		

Menu	Default	Editor
— 01=Indication 1? — ... — 15=Indication 15? — 1=Identifier? — 0=End? — 00=End? — 0=End?		→ Text Editor
— 12=Config Update? — 01=Check UD Config.? — 02=Ftp DL Params? — 01=Config FTP Addr? — 02=Config FTP Port? — 03=Config UD Pathname? — 04=Acount ID? — 05=Username? — 06=Password? — 00=End?		→ Text Editor → Text Editor → Text Editor → Text Editor → Password Editor
— 03=Dis DL Params? — 1=Config UD Dls — 2=Config UD Dls Port? — 0=End?		→ IP Number Editor → Number Editor
— 04=Https DL Params? — 1=Config UD Https Addr — 2=Config UD Https Port? — 3=Config UD Https Path? — 0=End?		→ IP Number Editor → Number Editor → Text Editor
— 05=Config UD Filename? — 06=Config UD File Type? — 07=Periodic Timer? — 08=Unreg Timer? — 09=Config Mgmt Type? — 10=Authentication? — 00=End?		→ Text Editor → Text Editor → Text Editor → Text Editor → Switch Editor
— 15=Port Configuration? — 01=Service Agent? — 02=Test Interface? — 03=Resource Sharing? — 04=SNMP Port? — 05=Web Server? — 00=End?		
— 16=Survivability? — 1=Backup address? — 2=Backup port? — 3=Backup registration? — 4=Backup reg timer? — 5=Backup OBP? — 6=Backup transport? — 0=End?		
— 17=User mobility? — 02=Unauth Logoff trap? — 03=Logoff trap delay? — 04=Logoff error count? — 05=Timer HighPriority? — 06=Count MedPriority? — 07=Timer MedPriority? — 08=Mobility feature? — 09=Managed profile? — 00=End?		
— 00=End? — 7=Diagnostics? — 01=Display test? — 02=LED test? — 03=Key test? — 04=Audio loop test?	Password required Off	→ Switch Editor

Menu		Default	Editor
	--- 05=RAM test?		
	--- 06=ROM test?		
	--- 07=PING?		
	--- 01=User specified IP?		→ IP Number Editor
	--- 02=SIP Server IP?		
	--- 03=SIP Registrar IP?		
	--- 04=SIP Gateway IP?		
	--- 05=DL server IP addr.?		
	--- 06=SIP Dialog ID		
	--- 07=SNTP IP addr.?		
	--- 08=Default route IP?		
	--- 09=SNMP trap IP addr.?		
	--- 10=Terminal IP addr.?		
	--- 11=DNS Primary IP?		
	--- 12=DNS Secondary IP?		
	--- 00=End?		
	--- 08=Line monitor?	Off	→ Switch Editor
	--- 09=UL/DL status?		
	--- 1=Application download?		
	--- 2=Config download?		
	--- 3=Config upload?		
	--- 4=Logo download?		
	--- 5=MoH download?		
	--- 0=End?		
	--- 10=MIB2-Discards?		
	--- 11=MIB2 err count?		
	--- 13=Key Label Test?		Only with optiPoint 420
	--- 00=End?		
	--- 0=End?		
	--- 00=End		

Editors

Text Editor

Display example of the text editor:

Siem_
d→ef3DEL

Key Functionality

Key	Use
...	Text entry
	Delete or move insertion point
	Mode control → page 156
	Accept entry
	Abort entry

Characters Sets

Character entry is performed by multiple presses of the keypad keys according to the following tables (table shows lower case characters).

Example: "+" = press the key 5x times.

Key	1x	2x	3x	4x	5x	6x	7x	8x	9x	10x
	\$	(1)	1							
	a	b	c	2						
	d	e	f	3						
	g	h	i	4						
	j	k	l	5						
	m	n	o	6						
	p	q	r	s	7					
	t	u	v	8						
	w	x	y	z	9					
	/	(	)	[	]	*	<	=	>	%
	.	(1)	,	?	0	!	:	;	'	"
	#	@	-	&	+	-	^			

(1) Space

This character sets also applies to entry of alphanumeric passwords
→ page 158.

Text Editor Modes

Depending on the function key "Shift" the behavior of the editor is different.

Display/Key	Shift (not shifted)	Shift (shifted)
Lower right corner of the display shows	DEL	INS
 ...  * 	Insert lower case characters	Insert upper case characters
	Delete backwards	Move insertion point left
	Delete forwards	Move insertion point right

Use of the Editor

This editor is used in the following administration scenarios:

- Alert indication → page 163
- Application download filename → page 165
- Branding/Identity name → page 168
- Configuration download filename → page 170
- Default domain name → page 174
- Domain Name → page 182
- FTP account name → page 187
- FTP path → page 188
- FTP username → page 188
- Hold music download filename → page 189
- Ping → page 207
- SIP realm → page 215
- SIP user ID → page 218
- Terminal name → page 223

If DNS is applicable → page 41, this editor can also be used in the following administration scenarios:

- Download server IP address or DNS name → page 183
- Message Waiting IP address → page 203
- SIP addresses → page 214
- SNMP Trap IP address or DNS name → page 219
- SNTP server address or DNS name → page 219

Switch Editor

Display example of the switch editor:

```
DHCP IP assign:On
Switch off?>
```

Key Functionality

-  Press this key to change the value ("Switch on", "Switch off").
-  Press this key to exit without changes ("Continue").

Use of the Editor

This editor is used in the following administration scenarios:

- DHCP IP assignment → page 175
- G.711 Silence Suppression → page 188
- Outbound proxy → page 205
- Register by terminal name → page 210
- SIP session timer enabled → page 217

Number Editor

Display example of the number editor:

```
Terminal Number
23554_
```

Permitted: Numeric values.

Key Functionality



Enter the numbers.



Press this key to delete one character at a time to the left.



Press this key to confirm the entry.

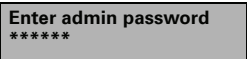
Use of the Editor

This editor is used in the following administration scenarios:

- Alert indication → page 163
- Emergency number → page 185
- Layer 2 signalling → page 193
- Layer 2 voice → page 194
- Registration timer value → page 211
- SIP session timer value → page 216
- Terminal number → page 224
- Manual VLAN identifier → page 203
- Voicemail number → page 228

Password Editor

Display example of the password editor:



The password editor is used in authorisation and storage mode.



Enter the password. If alphanumeric passwords are permitted, the behaviour is generally like editing texts → page 155.

Depending on the function key "Shift" the behavior of the editor is different.

Display/Key	Shift (not shifted)	Shift (shifted)
Lower right corner of the display shows		CAPS
 ...  * 	Insert lower case characters	Insert upper case characters



Press this key to delete characters/numbers to the left.

Use of the Editor

This editor is used in the following administration scenarios:

- Alphanumeric passwords:
 - FTP password → page 187
 - SIP password → page 215
 - SNMP password → page 218
- Numeric passwords only:
 - Administrator password → page 163

Gradient Editor

The Gradient Editor is used to change settings such as audio and visual settings, e.g. volume and contrast settings. On entry to this editor the configurable option name and its current status will be shown on the third line of the display, whilst the bottom line will display a graphical representation of the current setting and the available range of values.

Display example of the gradient editor:



Key Functionality



Enter gradient value.



Press this key to decrease the current gradient value.



Press this keys to increase the current gradient value.

IP Number Editor

optiPoint 410/420 economy/economy plus/standard/advance

Display example of the IP number editor:

SIP Server
192.168.001.012

Permitted: Integer values from 0 to 255 per octet and dots (".") as separator.

Leading zeros will be ignored.

Key Functionality



Enter the numbers.



Press this key to step from octet to octet to the right.



Press this key to delete numbers within a octet or step from octet to octet to the left.

Use of the Editor

This editor is used in the following administration scenarios:

- Default Route → page 174
- Download server IP address or DNS name → page 183
- IP routing → page 191
- Message Waiting IP address → page 203
- Ping → page 207
- Primary DNS IP address → page 207
- Secondary DNS IP address → page 212
- SIP addresses → page 214
- SNMP Trap IP address or DNS name → page 219
- Sntp server address or DNS name → page 219
- Terminal IP address → page 223
- Terminal mask → page 223

If DNS is applicable → page 41, the text editor → page 155 can also be used to enter host names.

Offset Editor

Display example of the offset editor:

Timezone offset: +2
+1

Permitted: Integer values from -12 to +12.

Key Functionality

-   Press this keys to increase/decrease the current value by one.
-   Press this keys to change the sign of the value (plus/minus).
-  Press this key to confirm the entry.

Use of the Editor

This editor is used in the following administration scenario:

- Time zone offset → page 225

Melody/Tone Editor

Display example of the melody/tone editor:

Indication 1 Melody: 2/2
Melody 2 Tone sequ. 2

Permitted: Numeric values.

Key Functionality

-  Enter the numbers.
-   Press this keys to increase/decrease the current value by one.
-   Press this keys to toggle between the two options.
-  Press this key to confirm the entry.

Use of the Editor

This editor is used in the following administration scenario:

Alert indication → page 163

Alphabetical Reference

This reference offers basic information that can be used by the administrator to carry out administration- and diagnostics-related jobs in the optiPoint 410/420 family S V7.0.

- The Chapter explains alphabetically sorted terms that, for instance, you will encounter in the menus.
Used symbols:
 -  Shows administration tasks with menu paths at the optiPoint 410/420 family S V7.0 and on the Web Interface.
 -  Shows the related Web Interface surfaces.
 -  Refers to the User Manual.
- This is followed by the Chapter “Abbreviations and Specialized Terms”.

Description of Functions

Action on submit

- Select the download type:
 - No transfer
 - Upload configuration
 - Download configuration
 - Download application
 - Download hold music
 - Download logo
 - Download Java program (only if DSM-Module is available)
 - Download LDAP template (only if DSM-Module is available)
 - Download DSM firmware (only if DSM-Module is available)

 → Seite 77  → page 118

Address Book

For information see description of

- Download server IP address or DNS name → page 183
- FTP account name → page 187
- FTP path → page 188
- FTP password → page 187
- FTP username → page 188

For further information of the Address book see "optiPoint display module User Manual".

 → page 80  → page 137

Administrator password

- Use this function to change the password that is necessary for accessing the administrator area.
- Value range table:

Permitted values	numeric
Length min.	Is predefined by DLS
Length max.	Is predefined by DLS (up to 24 digits)
Default value	123456

- Editing by phone: Password Editor → page 158.

 → page 99  → page 144

Alert indication

- Use this function to specify different ring tones for distinctive alert info URLs (identifiers).
 - Melody
 - Tone
 - Duration

Identifier

- Value range table:

Permitted values	alphanumeric
Length max.	50 digits
Example Strings (see system documentation for identifier string)	bellcore-dr1, bellcore-dr2, bellcore-dr3, bellcore-dr4, alert-group, alert-external , alert-internal, alert-visual, alert-emergency, alert-autoanswer, alert-priority, alert-acd, alert-community-1, alert-community-2, alert-community-3, alert-community-4

- Editing by phone: Text Editor → page 155.

Alert Type

- List of options:

Type	Description
Ring	In case of this type the phone rings.
Silence	In case of this type the phone does not ring.

Ringler

- Value range table:

Permitted values	numeric
Range	Melody: 1 ... 8 Tone sequence: • 1 (standard melody), • 2 (single-shot tone), • 3 (silence)
Default value	Melody: 2 , Tone sequence: 2

- To hear the configured melody and tone sequence you have to proceed the according call.
- Editing by phone: Melody/Tone Editor → page 161.

Duration

- Value range table:

Permitted values	numeric
Range	0 ... 300 seconds
Default value	60 seconds

- Editing by phone: Number Editor → page 157.

 → page 94  → page 122

Allow Refuse

This function allows the administrator to control whether the Refuse call feature is available on the Phone. The following options are always available.

- Switch off/Schwitch on
- Continue

 → page 64  → page 116

Append codes

- The following buttons represent the possibility of the append codes:
 - OK
 - Clear
 - Consult
 - Pause

 User Manual  → page 130

Application download filename

- Specify the name of the file containing the software of the optiPoint 410/420 family S V7.0.
- The file must exist in a defined directory on the → FTP server (→ Download server IP address or DNS name, → FTP path).
- Value range table:

Permitted values	alphanumeric
Length max.	92 digit
Default value	opti410a

- Editing by phone: Text Editor → page 155.

 → page 77  → page 118

Audio loop test

- The test activates the microphone and the loudspeaker in the handset.
- You can check these components by speaking and listening.

 → page 96  → page 140

Audio mode

- Use this function to select the audio transfer codec.

Audio Mode	
High Quality Preferred	Uncompressed audio transmission.
Low Bandwith Preferred	Use preferred compressed audio transmission (narrow band).
Low Bandwith only	Use compressed audio transmission only (narrow band)..
G711 Preferred	Uncompressed audio transmission. (narrow band)
G729 Preferred	Compressed transmission. (band of about 8 kbit/s.

- The value of compression encoding is depending on the selected compression codec, see Compression encoding → page 170.
- Default value: G711 preferred.

 **Audio codec G.711:**
If the country code is set to US the audio codec G.711 ulaw is preferred. All other country codes causes the audio codec G.711 alaw to be preferred.

Audio codec G.722:
This speech codec offers a wider audio bandwidth resulting in major improvement in the represented speech quality.

Codec Negotiation

The following table indicates which codec will be selected depending on which codec is selected on each phone, a G711 always option is included because some non-Siemens phones may include this option.

Phone B Phone A	High Quality Preferred (Codec G729)	High Quality Preferred (Codec G723)	Low Band Preferred (over 729)	Low Band Preferred (over 723)	Low Band Only (over 729)	Low Band Only (over 723)	G711 Preferred (over 729)	G711 Preferred (over 723)	G729 Preferred (over 723)	G729 Preferred (over 729)
High Quality Preferred (G729)	G722	G722	G722	G722	G729*	G729*	G722	G722	G722	G722
High Quality Preferred (G723)	G722	G722	G722	G722	G723*	G723*	G722	G722	G722	G722
Low Band Preferred (over 729)	G729	G729	G729	G729	G729	G729	G729	G729	G729	G729
Low Band Preferred (over 723)	G729	G729	G723	G723	G723	G723	G723	G723	G723	G723
Low Band Only (over 729)	G729	G729	G729	G729	G729	G729	G729	G729	G729	G729
Low Band Only (over 723)	G723	G723	G723	G723	G723	G723	G723	G723	G723	G723
G711 Preferred (Codec 729)	G711	G711	G711	G711	G729*	G729*	G711	G711	G711	G711
G711 Preferred (Codec 723)	G711	G711	G711	G711	G723*	G723*	G711	G711	G711	G711
G729 Preferred (Codec 723)	G729	G729	G729	G729	G729	G729	G729	G729	G729	G729
G729 Preferred (Codec 729)	G729	G729	G729	G729	G729	G729	G729	G729	G729	G729

* Note that the B party uses a codec in the preferred order of the A party and this regarding RFC 3264 section 6.1.

Note that the table is read with Phone A calling Phone B. The table is not symmetrical so does not describe the situation if B calls A. If a call is cleared because the codecs are not compatible, the phone will display "Terminal Incompatible".

Bandwidth calculation

Calculation into dependence of:

- Codec
- RTP packet size
- Transmission medium

1. LAN

Maximum LAN-Overhead (Ethernet-Full Duplex per direction)

Protocol	Bytes
RTP	12
UDP	8
IP	20
802.1Q VLAN Tagging	4
MAC (incl. Preamble, FCS)	26
Total	70

Bandwith calculation for Ethernet
(Values in brackets – security enabled)

Voice Codec	Packet size [ms]	Payload Bytes	Ethernet Packet size [Bytes]	Used Ether- net-Band- width incl. Preamble [kBit/s]
G.711 [*]	10	80 (90)	150 (160)	120,0 (128,0)
G.711 ^{**}	20	160 (170)	230 (240)	92,0 (96,0)
G723 ^{**}	30	24 (34)	1014 (114)	25,1 (30,4)
G.729 ^{***}	10	10 (20)	80 (90)	64,0 (72,0)
G.729 ^{***}	20	20 (30)	90 (100)	36,0 (40,0)
G.722 ^{****}	10	80 (90)	150 (160)	120,0 (128,0)
G.722 ^{****}	20	160 (170)	230 (240)	92,0 (96,0)

* In the SIP environment it is possible to activate silence supression. In this case the band-
width calculation values are max values

** For G.723 it is not possible to activate silence suppression and the only available sample
size is 30 ms

*** In the SIP environment silence suppression is always active currently. In this case the band-
width calculation values are max values.

****For G.722 it is not possible to activate silence suppression.

2. WAN (e.g. ATM)

WAN-Overhead

Protocol	Bytes
RTP	12
UDP	8
IP	20
ATM-Overhead	5
Total	45

Bandwidth calculation for WAN (e.g. ATM)
(Values in brackets – security enabled)

Voice Codec	Packet size [ms]	Payload Bytes	WAN Packet size [Bytes]	Used WAN- Bandwidth [kBit/s]
G.711*	10	80 (90)	125 (125)	100,0 (108,0)
G.711**	20	160 (170)	205 (215)	82,0 (86,0)
G723**	30	24 (34)	69 (79)	18,4 (21,1)
G.729***	10	10 (20)	55 (65)	44,0 (52,0)
G.729***	20	20(30)	65 (75)	22,0 (26,0)
G.722****	10	80 (90)	125 (135)	100,0 (108,0)
G.722****	20	160 (170)	205 (215)	82,0 (86,0)

- * In the SIP environment it is possible to activate silence suppression. In this case the bandwidth calculation values are max values
- ** For G.723 it is not possible to activate silence suppression and the only available sample size is 30 ms
- *** In the SIP environment silence suppression is always active currently. In this case the bandwidth calculation values are max values.
- ****For G.722 it is not possible to activate silence suppression.

 → page 93  → page 121

Branding/Identity name

- Enter the name (e.g. a company name), which is to be displayed in the third line of the display in idle mode of the optiPoint 410/420 family S V7.0.
- Range: Alphanumeric – 10 digits.
- Editing by phone: Text Editor → page 155.

 → page 66  → page 115

Call Recorder (HiPath 8000)

Enter the dial string for example for ASC Call Recording (EVOip). Activate- or deactivate the "Audible indication".

You can search for calls and replay them e.g. using the standard ASC applications POWERplay, WEBplay, or INSTANT WEBplay.

Call Recording has to be enabled in the access list (see → page 73) and in the configuration menu of the phone (see user manual).

 → page 64  → page 116

Callback URIs

For "Callback – Busy line", and "Delete callbacks" enter the access code of the HiPath 8000. "Callback – No reply" is for future use – not supported with HiPath 8000.

 → page 64  → page 116

Call park URI

Allows the administrator to configure the URI of the call park server <call-park@<SIP Server IP> (e. g. callpark@172.16.127.95 or Domain Name).

This option is displayed with its current setting. If the URI is empty or invalid the user will not be able to use the **Call park** feature. This URIs is only effective with **Sylantro** server.

 → page 64  → page 116

Call pickup URI

Allows the administrator to configure the URI of the call pickup server <pickup@<SIP Server IP> (e. g. pickup@172.16.127.95 or Domain Name). This option is displayed with its current setting. If the URI is empty or invalid the user will not be able to use the Call pickup feature. This URIs is only effective with **Sylantro** server.

 → page 64  → page 116

Check for update

This option is used to initiate a check for a configuration change at the configuration server and, if found, hence an update to the phone's configuration. This option is always displayed and will also be automatically displayed as a prompt to confirm a periodic check for configuration updates. Selecting this option will initiate the configuration download process.

 → page 88  → page 136

Clear ALL user data

This operation will clear out all personal data relating to the user (including Personal Directory entries and structure definition), ready for the phone to be given to a different user. It does not clear out Phone settings (such as contrast or touch screen calibration settings), nor Administration settings (such as network details).

 → page 99  → page 144

Compression encoding

- Use this function to select one of the two compression encodings that should be used if the compressed audio mode was selected, see Audio mode → page 165.
- Selectable values: G729 and G723.

 → page 93  → page 121

Conference factory URI

This field identifies the server used for the system based conference. Enter the URI of the conference server <conf@<SIP Server IP> (e. g. conf@172.16.127.95 or Domain Name). Reboot is required. This URIs is only effective with **Sylantro** server.

 → page 64  → page 116

Config DLS Port

If Deployment service is used, enter the port address of the server.

 → page 88  → page 135

Configuration download filename

- Specify the name of the file containing the configuration data of the optiPoint 410/420 family S V7.0.
- The file must exist in a defined directory on the → FTP server (→ Download server IP address or DNS name, → FTP path).
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits
Default value	opti410c

- Editing by phone: Text Editor → page 155.

 → page 77  → page 118 or

Config Server address

Specify the address of the Server where to **upload** the phones configuration file.

 The configuration file is in Unix text format. Editing the document might change the formatting.

 → page 76  → page 118

Config Update Account ID

The name of the FTP account that holds the configuration files. The following options are always available.

- Change
- Continue

 → page 88  → page 135

Config Update Authentication

Mark if user should have authentication. The following options are always available.

- On
- Off

 → page 88  → page 135

Config Update DLS IP

If Deployment service is used, enter the → IP address of the server.

 → page 88  → page 135

Config Update Filename

The name part of the system wide configuration file (also used as a template to the name of the terminal specific configuration file). The following options are always available.

- Change
- Continue

 → page 88  → page 135

Config Update File Type

The extension type part of the system wide configuration file (also used as a template to the name of the terminal specific configuration file). The following options are always available.

- Change
- Continue

 → page 88  → page 135

Config Update FTP IP

The IP address of the FTP server used for the Configuration download. The address and port are normally obtained at startup from DHCP (as a vendor specific extension), or failing that from DNS (as a text record associated with the phone's domain name). The address obtained automatically updates this menu item. If the DHCP or DNS methods do not yield a valid IP address the menu item value will be used if it holds a non-null IP address.

If the IP address had been obtained by DHCP then it cannot be altered. In other cases selecting the OK button will invoke the IP Number Editor to edit the address. The updated address will be used the next time that a config download is performed. The IP address will be set during startup if a valid IP address is obtained from DHCP or DNS.

 → page 88  → page 135

Config Update FTP Port

The IP port of the FTP server used for the Configuration download. Further information see → page 172.

 → page 88  → page 135

Config Update Pathname

The path from the FTP root to the directory on the FTP server that holds the configuration files. The following options are always available.

- Change
- Continue

 → page 88  → page 135

Config Update Password

The FTP password that is used to access the configuration files. The following options are always available.

- Change
- Continue

 → page 88  → page 135

Config Update Periodic Timer

The interval between automatic checking for a configuration download which occurs when the SIP phone is registered (at least on the primary line of a Keyset). A value of 0 prevents periodic checking.

 → page 88  → page 135

Config Update Unregistration Timer

The interval between automatic checking for a configuration download which occurs when the SIP phone is unregistered (on any line of a Keyset). A value of 0 prevents periodic checking.

 → page 88  → page 135

Config Update User Name

The name of the FTP user that is used to access the configuration files. The following options are always available.

- Change
- Continue

 → page 88  → page 135

Connectivity check

To check the TLS connectivity, the value in this field has to be greater than 0. The default value is 120. To deactivate the check function you set the value to 0. Any value greater than 0 activates the function. After deactivation/activation the phone has to be restarted. If you only changed the value (greater than 0), you do not have to restart the phone.

- 0 = off
- 10 - 3600 = normal range (1-9 is equivalent to 10)
- >3600 = 3600

 → page 96  → page 144

Count Medium Priority

The number of changes to medium prioity items allowed before triggering a save to the DLS.

- Medium Priority peg count threshold
- The value has to be a number in the range 1 to 999

 → page 102  → page 134

Date/Time

- Manual definition is necessary only if this information is not transmitted automatically by a → SNTP server.
- Select the date format and enter the date and time information.

 → page 67  → page 120

Daylight saving

- If your country uses daylight saving time, you have to switch this feature on and off manually twice a year (independently whether SNTP is used or not → page 219).
- "On" means an offset of +1, "Off" means no offset (default).

 → page 67  → page 120

Default domain name

- If you use an Outbound Proxy server you can define a valid domain name of this server. To use this setting you have to activate the Outbound Proxy option → page 205.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 62  → page 115

Default Music

If a hold music file was downloaded → page 74 you can use this function to reset the hold music to default.

 → page 92

Default Route

- Enter the → IP address that was assigned to the router of your IP network if not provided by → DHCP dynamically (→ DHCP IP assignment).
- If the value was assigned dynamically, it can only be read.
- The change will only have effect if you restart the phone.
- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

- Editing by phone: IP Number Editor → page 159.

 → page 57  → page 114

Description

Enter a description for the relevant HiPath 8000 feature here.

 → page 131 or → page 130

DHCP IP assignment

- At power up starting endpoints search for a → DHCP server and try to obtain configuration parameters from that central server. The protocol is based on broadcasts and hence the endpoints do not need to know the location of the DHCP server.
- DHCP is an extension of the BOOTP protocol. The original BOOTP protocol only allowed for the automatic configuration of IP related parameters and for the detection of a server to boot and endpoint from. DHCP is a more generic in that it allows for the request of a set configuration options and these options are not constrained to the basic IP related parameters.
- Activate this option if the required IP data of the telephone should be assigned dynamically by a → DHCP server.
- If no DHCP server is available, please deactivate this option. In this case, the data corresponding to the → Terminal IP address, → Terminal mask and → Default Route must be defined manually.
- List of information obtained by DHCP:

Basic informations	• Terminal IP Address • Terminal Mask (Network Mask)
Optional informations	• 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)

- The change will only have effect if you restart the phone.
- Default value: On.
- Editing by phone: Switch Editor → page 157.

 → page 57  → page 114

Dial Plan

The dial plan of the phone should not be confused with the dial plan of the server. Phone documentation uses the term “number plan” for the server’s “dial plan.” The phone’s terminology usage is supported by standards – RFC.

The purpose of the dial plan is to reduce or remove the post dialling delay caused by other methods of determining end of dialling – either use of a timer or pressing of a “done” key. Both alternatives are supported by the phone: the timer defaults to 6 seconds and the done key is either the tick or lifting the handset.

The dial plan makes a critical assumption about the number plan – it is possible to determine the total number of digits required from the leading digits. It does support a variable range of lengths for a given set of initial digits, but in this scenario it is not as efficient and brings some restrictions.

It is possible to create a SIP number plan which prevents effective use of the dial plan. If it is desired to use a dial plan, it has to be considered when the number plan is created. Dial plans are supported by other manufacturer’s SIP phones.

A well planned dial plan can significantly impact on a user’s perception of the system. Absence of a dial plan gives the appearance of a slow system. A bad dial plan gives a view more akin to “sometimes it’s very slow” – the user sees both fast and slow responses.

The Make-up of a Dial Plan

Currently a dial plan is restricted to 48 entries. This figure was believed to be “enough” while avoiding excessive searching.

Each entry consists of a number of fields:

- Leading digits – a string to match the dialled digits against. Both * & # can be included. There is also the wildcard X to represent any single digit. There is no “any sequence” value.
- Action(s) – Originally, there were two, but these have been joined by a third.
 - **S** – Send the digits when the maximum digits have been received or if the timer expires after the minimum digits have been received or on receipt of the terminator after the minimum digits.
 - **C** – Check for other actions. Minimum and maximum must match the length of the string. The timer is run at this point only; sending will occur on expiry. If more digits are received, further entries will be checked. It is possible to use C in combination with D and have no timer.
 - **D** – Give secondary dial tone when the leading digits match. This action does not depend on later parameters. In theory, there is a choice of 9 tones, but only digit 1 is currently valid. D can be used in combination with C or S.
- Minimum length – Automatic sending will not occur until at least this many digits have been dialled.

- Maximum length – Automatic sending will occur when this many digits have been dialled.
- Timer – A shorter than normal interdigit timer to be used once the minimum number of digits have been dialled or when the “check” function occurs. The timer can be zero if the two lengths are equal or with CD1. Actually, the timer can be set to values larger than the phone’s interdigit timer (it can be as high as 9, while the default internal timer is 6).
- Terminator – A digit used to indicate that dialling is complete before reaching the maximum number of digits. The terminator can only be * or #. The terminator is sent as part of the digit string.
- Option – A special function to be applied when the digits are sent. Currently two options are supported:
 - **B** – Lock by-pass. Strings with this attribute can be dialled when the phone is locked (other strings are barred by the phone).
 - **E** – Emergency; implies B. Dialling these numbers will cancel both forwarding and DND to allow the emergency service to return calls to this caller. There is nothing to prevent the user re-invoking these features later.
- Comment – although this can be left blank, it is useful to explain why this entry is present.

Field separators depend on exactly how the dial plan is put on the phone. Raw database entries use the separator “^”, while an external document uses “.”

Dial plan entries are in priority order. The phone will lock onto a matching entry and not check later ones, except as described for the C action. It is bad practice to have conflicting or duplicate entries as these may confuse other investigations.

Examples of Dial Plans

Combinations of these examples can be used according to need. These examples are deliberately short to explain specific principles.

A SIP server can be used in one of three ways. The nature of the numbers used will vary according to the type of use – a PABX replacement, a complete private network or a public exchange.

A PABX replacement in the UK might have only a few entries:

- 2,S,,4,,,B, internal numbers, can be dialled when locked
- 3,S,,4,,,B, internal numbers, can be dialled when locked
- 9,S,4,20,3,,,external numbers

Notice that minimum lengths need not be specified at input time, however the maximum length will be substituted internally (compare the results of viewing “memory” and “file” on the web page with any of these examples).

A few refinements can be added, according to need:

- There could be an internal emergency number (3333). This has to be placed first. It does not conflict with the use of 3 as a first digit. Once the user deviates from a sequence of 3's, the search will find the other entry. This entry is only required if it is desired to cancel forwarding and DND to allow calls to be returned to the caller.
3333,S,,4,,,E, internal emergencies
- Refinements could be made to the public network entries as having critical timing over such a range of lengths might be hard on a caller. Also, only certain lengths actually exist. Nottingham's local numbers are typical of large UK cities; smaller cities would have 7 digits, while London needs 9. Some of these entries are order sensitive.
 - 9999,S,,4,,,E, traditional public emergency number
 - 99,S,,8,,, Nottingham local numbers
 - 98,S,,8,,, Nottingham local numbers
 - 9112,S,,4,,,E, European standard emergency number
 - 9118,S,,6,,, Directory enquiries, although this would often be barred by a PABX.
 - 91XX,C,4,4,3,,, Special services, in theory, longer numbers can be used, but there are none at present. Use of "C" will allow these with changes to this plan.
 - 900,S,13,20,3,,, International numbers
 - 90,S, 11,12,3,,, National numbers
- Secondary dial tone might be wanted after the initial 9. In the initial case, this can be done by changing the action from "S" to "SD1." When the more refined set of numbers are used, they would need to be preceded by:
9,CD1,1,1,,, give dial tone after 9
- Feature codes starting with * and # might be useful. Studies similar to that for the public numbers would be needed, but standard values might be possible for a given server.

A private network replacement would need longer local numbers, possibly with the leading digits being used to nominally identify the site. Examples would add nothing to those above.

British public network usage would need entries similar to the ones used to access the public network above, but removing the leading "9" (the access code) and having one less digit. More discrimination on the international numbers might be used.

The American public network uses a different technique for discriminating between local and long distance numbers, so it might include:

- *,S,3,15,3,#, feature codes, 1
- #,S,3,15,3,#, feature codes, 2
- 911,S,,3,,,E, police
- X11,C,3,3,3,#, special service numbers
- X1,S,,10,,#, long distance, type 1
- X0,S,,10,,#, long distance, type 2
- X,S,,7,,#, local

This sequence is order critical, except that 911 could be moved higher.

What it doesn't do well

Variable length local numbers involving subsets give a poor performance, needing the critical timer to dial the shorter numbers. If you have both 2 digit and 4 digit numbers beginning with 2, you need:

2,S,2,4,3,,

Another drawback of this method is that the phone will send 3 digit numbers and these are not valid. You could make extensive use of the check function:

- 21,C,2,2,,
- 22,C,2,2,,
- 23,C,2,2,,
- 24,C,2,2,,
- 25,C,2,2,,
- 26,C,2,2,,
- 27,C,2,2,,
- 28,C,2,2,,
- 29,C,2,2,,
- 20,C,2,2,,
- 2,S,,4,,,

This is cumbersome and would cause the table to be too big if other digits followed this practice or the lengths were 3 and 5 (you'd need 100 entries to handle the three-digit values).

The dial plan can not be edited on the phone or via the Web Interface. The dial plan is a CSV file. To load the dial plan to the phone, the deployment service (DLS) has to be used.

Another example for an US Dialplan below:

Dialplan_us.csv					
!UK_Pri Vpp.pp					
555	S				Emergency
999	S				Emergency
0	S				Operator
1	S				Illegal
2	S		5		Extensions
3	S		4		Extensions
4	S		4		Extensions
5	S		4		Extensions
6	S		2		Communication Group
7	S	6		5	Private Network Trunks
900	S	10	18	5	International
901	S	11	12	3	National
902	S		12	3	National
903	S	11	12	3	National
9118	S		7		Directory Enquiries
99	S		8		Public Local Area
98	S		8		Public Local Area
9x	S		2		Illegal
*	S	6		5	# Feature Codes
#	S	6		5	# Feature Codes

The dial plan should begin with a line of up to 14 characters providing a unique identification of the Dial Plan.

What it can't do

You can't have different entries which are used when the phone is locked. All entries are processed all of the time. Lock bypass is an option added to the basic functionality. You cannot bar the sending of digit strings, except when the phone is locked.

There is no mechanism to activate and deactivate individual entries, although it is possible to turn the whole plan off.

 → page 70  → page 113 and → page 131

Dial string

 User Manual  → page 130

Dialling properties

The dialling properties menu provides access to the parameters that are required to convert a number expressed in canonical format to a digit string suitable for dialling. This conversion may be required for numbers taken from a directory application. The value of the following option can be changed:

- External code
- International prefix
- Country code
- National prefix
- Area code
- District code

The fields

- Country code
- Area code
- District code

are also used by the telephone to complete the Mobility ID for the registration at the DLS server, if the Mobility ID is smaller than or equals 6 digits. In order to trigger the completion, the field "International Mobility ID" has to be activated (see  → page 134). Make sure the field entries are consistent with the entries in the DLS as otherwise the registration of the mobility user might fail.

Example:

Country Code: 49

Area code: 89

District code: 722

 → page 71  → page 132

Discarded in-/outbound packets

- Displays the number of discard messages according to → MIB.
- The used MIB objects are:

MIB-Objects	Explanation
ifInDiscards	Discarded ingoing packets
ifOutDiscards	Discarded outgoing packets

 → page 93  → page 121

Display ID

Enter an individual ID (Name or Number) for to display instead of terminal number or name.

 → page 61  → page 115

Display ID Use

Enable/disable to display the Display ID instead of terminal number or name

 → page 61  → page 115

Display test

- Conduct this test to check the function of the → LCD display in the Display telephone.
- Different display contents are displayed with the  and  keys. The  key terminates the test.

 → page 96  → page 140

Domain Name

- Is the name of the local domain the phone belongs to.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- If DHCP is enabled, this field is provided automatically. It is not writeable
- Editing by phone: Text Editor → page 155.

 → page 57  → page 114

Download Application

- Use this function to download an updated software version for the optiPoint 410/420 family S V7.0 from the → FTP server.
- The following parameters must be set before undertaking the download operation:
 - → Download server IP address or DNS name, → FTP path, → Application download filename
 - → FTP account name
 - → FTP username, → FTP password
 - Detailed description: → page 74.

 → page 76  → page 118  → page 138

Download Configuration

- Use this function to download a configuration for the optiPoint 410/420 family S V7.0 stored on the → FTP server.
- The following parameters must be set before undertaking the download operation:
 - → Download server IP address or DNS name, → FTP path, → Configuration download filename
 - → FTP account name
 - → FTP username, → FTP password

 → page 76  → page 118  → page 138

Download Hold Music

- Normally used to download a hold music file for the optiPoint 410/420 family S V7.0 from the → FTP server. This function is not supported for customer usage.
- The following parameters must be set before undertaking the download operation:
 - → Download server IP address or DNS name, → FTP path, → Hold music download filename
 - → FTP account name
 - → FTP username, → FTP password

 → page 76  → page 118  → page 139

Download server IP address or DNS name

- Enter the → IP address or host name of the → FTP server to upload and download files from and to the optiPoint 410/420 family S V7.0.
- Value range table:

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and — if DNS is applicable → page 41 — also Text Editor → page 155.

 → page 75  → page 118

DSM Firmware

Allows the administrator to specify the filename of the optiPoint display module firmware to be downloaded to the optiPoint display module. This also allows the Administrator to configure the filename in anticipation of a optiPoint display module being installed.

 → page 78  → page 118

DSM Logo

An option to download a bitmap image for display on the idle optiPoint display module screen. The bitmap image may be customer specific, for example a logo. A default image is provided with the phone. Enter the filename of the logo to be downloaded to the optiPoint display module.

The logo graphic in the start screen, which is user-defined on installation, is a Windows bitmap graphic with the following properties:

- max. size (h x w): 65 x 190 pixels ca. 15 x 45 mm
- colour: 4-bit greyscale
- file name e.g. Logo.bmp

 → page 78  → page 118

DSS Address of Record

Each DSS (Direct Station Select) will have the SIP **Address Of Record (AoR)** of DSS destination and will have an unshifted function key and LED (DSS key) assigned to it. The assignment of key to DSS is determined by administration.

- Value range table:

Permitted values	numeric
Length min.	1 digit
Length max.	20 digits

 → page 72  → page 130

DSS Realm

- This field displays the realm of the DSS destination.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

 → page 72  → page 130

DSS user ID

- Enter the according SIP User ID of DSS destination.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

 → page 72  → page 130

DSS password

- Enter the according SIP Password.
- Value range table:

Permitted values	alphanumeric
Length min.	6 digits
Length max.	24 digits

 → page 72  → page 130

Emergency number

- Enter a valid emergency number.
- Value range table:

Permitted values	numeric
Length max.	20 digits

- Editing by phone: Number Editor → page 157.

 → page 66  → page 115

Feature Access

This option allows the Administrator to view the feature access settings and potentially activate or deactivate individual features on the phone. When a feature is deactivated it is no longer available at user level. If a feature is activated (so it is available at User level), its status is shown as "On". If the feature status is shown as "Off", the feature is not available at User level and is not displayed as an option in the Configuration menu.

 → page 73  → page 133

Feature Code

For Feature Toggle: Enter the code for the required HiPath 8000 feature, which you would like to assign to the function key. HiPath 8000 supports the following features for example:

- Make line busy: the phone number is treated as busy for the hunt group.
- Stop hunt: the phone number is removed from the hunt group.

 → page 130

For State Key: Enter the code for the required HiPath 8000 feature that you want to assign to this function key. HiPath 8000 supports the following functions for this key:

- HotDesking On: The number of your main telephone is assigned to this guest telephone. HotDesking is activated.
- HotDesking Off: The number of your main telephone is assigned back to the main telephone. HotDesking is deactivated.

 → page 131

Feature toggle

 User Manual  → page 130

Feature URI

Enter the code "HDLogonToggle" for the HotDesking feature.

 → page 131

Force logoff to basic user

If this field is marked and a submit command is issued, the current mobility user is logged off and the telephone is assigned the status "MOB logon".

 → page 134

Forwarding Indication

Only for the forwarding function of the HiPath 8000. If this function is activated, a blinking line key indicates the forwarding on all phones where a forwarded primary or secondary line is active.

 → page 69  → page 123

FTP account name

- Refer to the documentation of your FTP server for information about the FTP account.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 75  → page 118

FTP passive mode

If the passive FTP (also "passive mode") is activated, the telephone transmits a PASV command, the server opens a port and transmits this together with the IP address to the telephone. Usually both sides use a port higher than 1023.

This method is used if the server cannot reach the telephone. This is for example the case if the telephone is switched by a router which translates the telephone address using NAT, or if a firewall shields the network of the telephone from external attacks.

 → page 75  → page 118

FTP password

- Enter the password defined in the → FTP server as password for accessing this server.
- The password must correspond to the → FTP username and match the password on the server.
- Value range table:

Permitted values	alphanumeric
Length max.	24 digits
Default value	123abc

- Editing by phone: Password Editor → page 158.
- After factory reset the FTP password is replaced with the string "123abc".

 → page 75  → page 118

FTP path

- Enter the path of the directory defined in the → FTP server for uploading and downloading files.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits
Default value	.

- Editing by phone: Text Editor → page 155.

 → page 75  → page 118

FTP username

- Enter the name defined in the → FTP server as user for accessing the server.
- The password must correspond to the → FTP password and match the username on the server.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 75  → page 118

Function key

- A key on the optiPoint 410/420 family S V7.0 that may be logically associated with a supported function. A function key can support a second function:
 - Normal function
 - Shifted function
- For optiPoint 420 advance and optiPoint self labeling key module there is an additional field available to enter a name or number for this key.
- If required, enter additional parameters in the dialog.
- For detailed information see operating manual.

 → page 74  → page 124

G.711 Silence Suppression

- It suppresses transmission of packets on no conversation.
- Effects the following codecs: G.711, G.723 und G.729.
- Editing by phone: Switch Editor → page 157.

 → page 93  → page 121

Group pickup URI

- To be a member of a Call Pickup group for a notification service or the user initiated service the phone must be configured by administration with the URI of the call Pickup group service which is provided by the server. Enter the URI of the Call Pickup group <groupcallpickup@<SIP Server IP> (e. g. groupcallpickup@172.16.127.95 or Domain Name) .

 → page 64  → page 116

Help internet URL

Selecting this option allows the current setting to be changed. The following options are always available

- Change
- Continue

 → page 81  → page 137

Hide on DSM

Mark Checkbox **on** for to hide the line or **off** to show the line in overview.

 User Manual  → page 128

Hold music download filename

- Enter the name of the file containing the hold music.
- This function is not supported for customer usage.
- The file must exist in a defined directory on the → FTP server (→ Download server IP address or DNS name, → FTP path).
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits
Default value	opti410.moh

- Editing by phone: Text Editor → page 155.

 → page 77  → page 118

Home page

Allows the Home page to be configured.

 → page 81  → page 137

Hot line for selected line

Mark Checkbox if a hotline is defined for this line.

 → page 128

Hot/Warm line default dial string

This control allows the administrator to enter a default dial string associated with the phone to be used in connection with Hot Line or Warm Line working.

 → page 64  → page 116

Hot line dial string for selected line

This is the hotline target for this line. Can only be set by administrator in web interface.

 → page 128

Hot/Warm Phone

This dropdown allows the user to specify the Hot Line/Warm Line operation of the phone. There are three options:

- **Normal line Action:** Any Hot Line/Warm Line parameters associated with the phone will be ignored, regardless of whether or not the phone is in a keysystem group.
- **Warm line:** If there is a default dial string set against the phone then it will be automatically dialled (after the delay set with “Initial Digit Timer”) when the user lifts the handset. This occurs regardless of whether or not the phone is in a keysystem group.
- **Hot line:** There are two cases to consider, depending on whether the phone is part of a keysystem group.
 1. If the phone is part of a keysystem group, there may be a Hot Line dial string associated with each of the Line keys on the phone
 - If there is a Hot Line dial string associated with the line then, it will be automatically dialled immediately when the line is manually selected.
 - If there is not a Hot Line dial string associated with the Line, but there is a Default dial string set against the phone, then the Default dial string will be automatically dialled immediately when the line is manually selected.
 - If there is not a Hot Line dial string associated with the Line, and there is not a Default dial string set against the phone, then the user will receive dial tone when the line is manually selected.
 2. If the phone is not part of a keysystem group.
 - If there is a Default dial string set against the phone, then it will be automatically dialled immediately when the line is manually selected.
 - If there is not a Default dial string set against the phone, then the user will receive dial tone when the line is manually selected.

 → page 64  → page 116

Initial Digit Timer

This timer determines the delay after which the phone goes back to idle mode or calls a specified number (if "Warm line" is active) when the handset was lifted or the speaker key was pressed and no number was dialled.

 → page 64  → page 116

Intrusion allowed

Allow that an alerting or ringing line is shown in an intrusion bar on DSM call view. Mark the Checkbox to enable the feature.

 User Manual  → page 128

Invalid in-/outbound packets

- Displays the number of error messages according to → MIB.
- The used MIB objects are:

MIB-Objects	Explanation
ifInErrors	Non-valid ingoing packets
ifOutErrors	Non-valid outgoing packets

 → page 93  → page 121

IP routing

- To have constant access to network subscribers of other domains, you can enter a total of two more network destinations.
- An → IP address of the domain and gateway, and a → Subnet Mask must be entered for each further domain you wish to use.
- Use this function to define the following IP addresses for Route 1/2.

IP address	Use
IP address	IP address of the selected route
Gateway	IP address of the gateway for this route
Mask	Network mask for this route.

- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

- Editing by phone: IP Number Editor → page 159.

 → page 57  → page 114

Java Program download filename

- Enter the name of the file containing the Java program (Java midlet).
- The file must exist in a defined directory on the → FTP server (→ Download server IP address or DNS name, → FTP path).
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

If a DSM-Modult is connected you can download the file with **submit** and the option **download java program**.

 → page 78  → page 118

Java Proxy Server IP Address

- If an Java Proxy server is being used, enter the → IP address of this server here.
- IP editor in optiPoint 410/420 family S V7.0 → page 159.

 → page 79  → page 136

Java Proxy Server Port

- If an Java Proxy server is being used, enter the → Port number here for communication with this server.
- Valid values: 1 ... 65535.
- Number editor in optiPoint 410/420 family S V7.0 → page 157.

 → page 79  → page 136

Join allowed in conference

Use this option to specify whether the option "Join" is available in the menu during a conference and whether the partners are connected when you hang up.

 → page 64  → page 116

Key label

If you have an **optiPoint 420** or an additional optiPoint self labeling key module you can enter a name for the desired key which then will be displayed automatically.

 → page 68  → page 128 or → page 131

Key test

- Test to check the functions of the telephone keys.
- If you press a key on the optiPoint 410/420 family S V7.0 (except the  key), the associated LED lights up and the corresponding name is displayed in the → LCD display. The  key terminates the test.

 → page 96  → page 140

Layer 2/3

The QoS technology based on layer 2 and the two QoS technologies Diff-serv and TOS/IP Precedence based on layer 3 are allowing the VoIP application to request and receive predictable service levels in terms of:

- data through put capacity (bandwidth)
- latency variations (jitter)
- and delay.

For further information see → Quality of Service (QoS)

 → page 58  → page 117

Layer 3 Voice

Can be defined only if the → Layer 3 support is activated . Select the desired value – see → Quality of Service (QoS)

 → page 58  → page 117

Layer 3 Signalling

Can be defined only if the → Layer 3 support is activated . Select the desired value – see → Quality of Service (QoS).

 → page 58  → page 117

Layer 2 Default

- Can be defined only if the → Layer 2 support is activated (→ Quality of Service (QoS)).
- Value range table:

Permitted values	numeric
Range	0 ... 7 (for each 64 positions)
Default values	pos. 12: 6, pos. 18: 3

 → page 58  → page 117

Layer 2 signalling

Can be defined only if the → Layer 2 support is activated (→ Quality of Service (QoS)).

 → page 58  → page 117

Layer 2 voice

Can be defined only if the → Layer 2 support is activated (→ Quality of Service (QoS)).

 → page 58  → page 117

LAN port settings

- Use this function to define the bandwidth at which the optiPoint 410/420 family S V7.0 should be run. The required value depends on the bandwidth that the switch or router supports in the network.

Bandwith	Use
Automatic	in standard case (automatic detection)
10 Mbit/s full	in 10 Mbit networks in full duplex* process
10 Mbit/s half	in 10 Mbit networks with half duplex** process
100 Mbit/s full	in 100 Mbit networks in full duplex ¹ process
100 Mbit/s half	in 100 Mbit networks with half duplex ² process

* The data can be transmitted **and** received simultaneously.
** The data can only be transmitted **or** received.

- The PC port can setup in the same way as the LAN port.
- The change will only have effect if you restart the phone.

 → page 60  → page 123

Last Restart

- Use this function to view the date and time of the last restart
→ page 97 of the optiPoint 410/420 family S V7.0.

 → page 97

LDAP Server IP Address

- If an → LDAP server is being used, enter the → IP address of this server here.
- IP editor in optiPoint 410/420 family S V7.0 → page 159.

 → page 79  → page 136

LDAP server Port

- If an → LDAP server is being used, enter the → Port number here for communication with this server.
- Valid values: 1 ... 65535.
- Default port is 389
- Number editor in optiPoint 410/420 family S V7.0 → page 157.

 → page 79  → page 136

LDAP Template

- If an → LDAP server is used, enter the name of the LDAP template file that is used in connection with this server.
- The file must exist in a defined directory on the → FTP server (→ Download server IP address or DNS name, → FTP path).
- Standard name → page 243.
- Text editor in optiPoint 410/420 family S V7.0 → page 155.

If a DSM-Modult is connected you can download the file with **submit** and the option **download LDAP Template**.

LDAP template example

Template-String	declaration
SEARCHBASIC "O=SIEMENS COMM, C=GB"	organisation and specification of country
SEARCHFILTER1 = "surnameNational,Surname"	search criterion (fast and advanced)
SEARCHFILTER2 = "givenNameNational,First-name"	search criterion (fast and advanced)
SEARCHFILTER3 = "department,Department"	search criterion (advanced)
SEARCHFILTER4 = "localityShortCode,Location"	search criterion (advanced)
SEARCHFILTER5 = "ou,Org. Unit"	search criterion (advanced)
SEARCHATTRIB1 = "commonNameNational,Name,0"	cannot be selected
SEARCHATTRIB2 = "telephoneNumber,Telephone,1"	selectable field
SEARCHATTRIB3 = "alternatePhone,Telephone 2,1"	selectable field
SEARCHATTRIB4 = "mobileTelephoneNumber,Moblie,1"	selectable field
SEARCHATTRIB5 = "ou,Organisational Unit,0"	non-selectable field
SEARCHATTRIB6 = "localityNational,Location,0"	non-selectable field
SEARCHATTRIB7 = "department,Department,0"	non-selectable field
SEARCHATTRIB8 = "mainFunction,Function,0"	non-selectable field
EOF	end of file

[A](#) → page 78  → page 118

LED test

- Run this test to check the function of the → LEDs at the optiPoint 410/420 family S V7.0.
- During the test all LEDs are flashing. The  key terminates the test.

 → page 96  → page 140

Line key

Basic terminology

Call Appearance	The standard optiPoint 410/420 family S V7.0 single line device supports 2 call instances per line. When the line is idle there will be no call instances. In contrast an analogue wired line only supports a single Call Appearance at a time.
Multi-Call Appearance	[Future] Support of multiple line keys associated with the same Address of Record (AoR) on a single optiPoint 410/420 family S V7.0. Example: AoR of the primary line appears on multiple line keys on a single device.
Function key	A key on the optiPoint 410/420 family S V7.0 phone that may be logically associated with a supported function → page 188.
Keypad	A phone where all lines are represented by a line key plus an LED. Every keypad will have a primary line and may have secondary or phantom lines. 10 Line keys can be configured for a keypad.
Line	A representation of a valid SIP AoR (Address of Record). A line is the context for connecting SIP calls. A line may support one or more calls.
Line Appearance	A line (Directory Number) that appears on one or more Keypad devices as a primary line (currently only one device per primary line), secondary Line or phantom line.
Line key	A function key that is used to represent a line appearance (or, in the future, call appearance on a line).
Consultation hold	A form of hold which is private to the holding keypad.
Manual hold	A form of hold which is accessible to any keypad on the same shared line.
Private line type	A line that only is accessed by one SIP endpoint, i.e. it is exclusively owned.

Shared line type	A line that may be accessed by multiple SIP endpoints INVITES to a shared line are FORKED to all SIP endpoints sharing the line.
Primary line	The line that characterises the oP410/420 phone (user) . Every keyset will have a primary line. This line can be expected to use the public DN of the optiPoint 410/420 family S V7.0 phone (user). There is only one Primary line instance per optiPoint 410/420 family S V7.0 phone. A primary line on a different phone may appear as a secondary line on the optiPoint 410/420 family S V7.0 phone. The line type may be: • Private line type • Shared line type (i.e. secondary line on other SIP endpoints)
Secondary line	Any line on the Keyset that is a shared appearance of a Primary line (on another Keyset) but not the Primary line for the Keyset.
Phantom line	A line that is not characteristic of any specific SIP endpoint (i.e. not a primary line on any SIP endpoint). May be shared or private.

 → page 68  → page 128

Line key operation mode

Select the according line key operation mode:

- **Hold:** If a call is initiated to a phone, and speech path is established and then the primary or secondary line function key is pressed then the other phone is set to hold and the phone now is in idle state with two options:
 - To press the primary or secondary line key that now flashing and re-connect to the other phone or
 - To initiate another call.
- **Release:** If a call is initiated to a phone, and speech path is established and then the primary or secondary line function key is pressed then the other phone is cleared.

 → page 69  → page 123

Line monitor

- Activate the display to view the functions and values used to run the → LAN connections at the phone (layout → page 25).
- The status monitor remains active even during the normal operation of the optiPoint 410/420 family S V7.0. However, it does not affect the operation of the function keys.

Line Monitor information optiPoint 410

LAN Port 1 (LAN)

10 Mb/s Full duplex	LED:	9, 11
10 Mb/s Half duplex	LED:	9
100 Mb/s Full duplex	LED:	9, 10, 11
100 Mb/s Half duplex	LED:	9, 10

LAN Port 2 (PC)

10 Mb/s Full duplex	LED:	5, 7
10 Mb/s Half duplex	LED:	5
100 Mb/s Full duplex	LED:	5, 6, 7
100 Mb/s Half duplex	LED:	5, 6

Explanation report:

LED 9 or 5 'on'	Connection established
LED 9 or 5 'off'	Connection not established
LED 6 or 10 'on'	100 Mb/s connection
LED 6 or 10 'off'	10 Mb/s connection
LED 7 or 11 'on'	Full duplex connection
LED 7 or 11 'off'	Half duplex connection

Line Monitor information optiPoint 420

LAN Port 1 (LAN)

10 Mb/s Full duplex	LED:	10, 12
10 Mb/s Half duplex	LED:	10
100 Mb/s Full duplex	LED:	10, 11, 12
100 Mb/s Half duplex	LED:	10, 11

LAN Port 2 (PC)

10 Mb/s Full duplex	LED:	6, 8
10 Mb/s Half duplex	LED:	6
100 Mb/s Full duplex	LED:	6, 7, 8
100 Mb/s Half duplex	LED:	6, 7

Explanation report:

LED 10 or 6 'on'	Connection established
LED 10 or 6 'off'	Connection not established
LED 7 or 11 'on'	100 Mb/s connection
LED 7 or 11 'off'	10 Mb/s connection
LED 8 or 12 'on'	Full duplex connection
LED 8 or 12 'off'	Half duplex connection

 → page 96  → page 140

Line Address of Record

Each line will have a unique SIP **Address Of Record (AoR)** and will have an unshifted function key and LED (line key) assigned to it. The assignment of key to line is determined by administration.

- Value range table:

Permitted values	numeric
Length min.	1 digit
Length max.	20 digits

 → page 68  → page 128

Line Hunt Ranking

The ordered rank is used to search for a line that is suitable for making a call when making outgoing calls. Multiple lines may be given the same rank. Lines that are in rank 1 are the first lines to be considered for use. Lines of the same rank are considered for use in key number order.

Select the according rank (1 bis 10).

 → page 68  → page 128

Line Shared type

Each line on a keyset may be:

- Private; only allocated to that Keyset
 - Would be normally be used for calls made to the Keyset user.
- Shared; accessible by several keysets
 - A shared line is an **Address of Record** which appears on multiple Keysets. The SIP server is responsible to coordinating basic call control between the Keyset that have an appearance of the shared line.

 Each line is treated as if the Keyset Operation is Line based, the server may change call information if the line is Device based.

Only one call may be active at a time (connected to the handset or speaker/ mic.) but there may be several other calls connected to keyset that have been held or are alerting the Keyset. The line keys may be used to select different lines.

A keyset LED for a shared (or private) line reflects the status of a call on the line, whether the call is connected to the keyset or to another keyset with an appearance of the shared line.

Select the desired Type.

- Shared
- Private

 → page 68  → page 128

Line password

- Enter the according SIP Password.
- Value range table:

Permitted values	alphanumeric
Length min.	6 digits
Length max.	24 digits

 → page 68  → page 128

Line Primary line

- Every keyset has a primary line with an **Address of Record** that typically represents the user of the keyset (line owner). Only one line on a keyset can be designated as the primary line by the downloaded configuration, all other lines are secondary or and referred to as non-primary lines.



A non-Keyset phone only has a single line which is considered to be the same as the primary line of a Keypset.

- Both primary and non-primary lines can be shared or private. For example a shared primary line appears as a secondary line on another keyset which is then able to monitor the calls to the owner of the primary line.
- Some features can only be applied to the primary line to ensure that conflicting feature settings between different keysets on the same shared line cannot occur.
- Mark as primary if correct.

 → page 68  → page 128

Line Realm

- This field displays the realm that the phone is registered in.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

 → page 68  → page 128

Line Ring

If the Keypset is not currently in use, and the alerting line is allowed to ring, the alerting line will get the focus and the optiGuide display will persist until the call is no longer alerting. The audible ringing will be the standard ring as used on a non-Keypset phone.

Mark ringer on/off.

 → page 68  → page 128

Line user ID

- Enter the according SIP User ID.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

 → page 68  → page 128

Logoff Error Count

The number of **local** and **remote** unauthorised failed logout attempts detected and displayed.

Select "clear" (only on the phone)

- Stop the error message "Logoff failed:" on the phones display
- Set the error counters to 0

 → page 102  → page 134

Logoff Trap Delay

Delay before sending an → SNMP trap for an unauthorised logoff.

- Trap delay in seconds
- The value has to be a number in the range 0 to 600

 → page 102  → page 134 or → page 121

MAC address

- Displays the world-wide unique → MAC address of your optiPoint 410/420 family S V7.0.
- The MAC address is also visible on the label at the bottom of the optiPoint 410/420 family S V7.0 → page 18.

 → page 94  → page 113

Managed Profile

The DLS controlled setting which determines whether or not the DLS sets a default profile in the phone on startup. This field is read-only.

 → page 102  → page 134

Management Center Port

Allows the user to specify the network management port to receive → SNMP traps sent by the phone's SNMP agent.

 → page 121

Manual VLAN identifier

- Can be defined only if the → Layer 2 support is activated (→ Quality of Service (QoS)).
- This value describes the association with a certain VLAN, if a → VLAN is used.
- Value range table:

Permitted values	numeric
Range	0 ... 4095
Default value	0

- Editing by phone: Number Editor → page 157.

 → page 59  → page 117

Message Waiting IP address

- Use this function to configure the IP address or host name of the message waiting server.
- Value range table:

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and — if DNS is applicable → page 41 — also Text Editor → page 155

 → page 66  → page 115

Microphone Disable

Switch off microphone, i.e. hands-free speaking (e.g. in public places).

 → page 93  → page 121

Mobility feature

Displays the status of the User Mobility feature. The status is determined by the phone from other settings and is read-only (not used with optiPoint 410 entry)

Fixed text indicating the mobility-enabled status of the phone. The text is either:

- "This phone ist NOT mobility-enabled" or
- "This phone ist mobility-enabled"

 → page 102  → page 134

Mobility International ID

If this function is activated, the Mobility ID entered is completed using values from the Dialing Properties Table for the registration. This is only done if the ID entered does not exceed 6 digits. See also → page 181

 → page 102  → page 134

Originating line preference

Originating Line selection – provides connection of the phone to one of the lines appearing on it, on an automatic basis when the user goes off-hook or starts on-hook dialing. There are four originating options.

A Keyset is assigned one of the following preferences.

- **Prime Line Preference** – The designated Prime Line is always selected for originating calls.
- **Idle Line Preference** – Any idle line is selected for originating calls with the lines selected based on line selection table for the device (e.g., prime line first). See also line rank on → page 199 and Terminating line preference on → page 224.
- **Last Line Preference** – The line selected for originating calls is the line selected for the last call (originating or terminating).
- **No (Originating) Line Preference** – A line key must be pre-selected or post-selected each time the user goes off-hook.

Select the according line preference:

- Idle line
- Primary
- Last
- None

 → page 69  → page 123

Outbound proxy

- The optiPoint 410/420 family S V7.0 implements outbound proxy routing according to RFC 3261. If set, the phone routes any request outside the context of an existing dialog to the configured proxy, regardless of the contents of the Request-URI. The phone does not apply this rule to requests sent within the context of an existing dialog. These requests will always be sent to the address indicated in the received Contact header (the remote target) or, if present, the Record-Route header. See also [RFC 3261](#).
- If the user dials a URI "p.kelly@dom1.com" and the Outbound Proxy flag is "Off" and the Server/Registrar is in domain "dom2.com" the phone will attempt to resolve the domain part of the URI "dom1.com" the result will be form the request URI that is sent to the server in the "dom2.com" domain.
- If the user dials a URI "p.kelly@dom1.com" and the Outbound Proxy flag is on and the Server/Registrar is in domain "dom2.com" the phone will not attempt to resolve the domain part of the URI "dom1.com" the request URI will contain "p.kelly@dom1.com" but will be sent to the Server/Registrar in the "dom2.com" domain. It is then upto the Server/Registrar to determine the location of the "dom1.com" domain and forward the request there.
- With the Outbound Proxy flag "Off" if the user types just a phone number or name the domain part is automatically per pended and is based on the configured Server/Registrar domain name or IP address eg. If the Server/Registrar is in "dom1.com" domain and the user dials "123456" the request URI will be "123456@dom1.com". The phone will resolve the "dom1.com" part via DNS and forward the request there.
- With the Outbound Proxy flag "On" if the user types just a phone number or name the domain part can come from 1 of 2 possible sources. If the menu item "Default OBP domain" → page 174 is configured then this will be per pended to the name or number. If it is not configured then the Server/Registrar domain will be per pended and the request sent.
- If the Outbound Proxy flag is "On" and the Default OBP domain is set and the user types a URI, not just a name or number, then the entered domain will be used not the Default OBP domain thus providing an override mechanism.
- If this flag is "Off" but a Default OBP domain is configured it will be ignored.
- If DHCP delivers the address of an SIP Server according to draft-ietf-sip-dhcp-06.txt the server must be treated as an Outbound Proxy server. In this case, if the Outbound Proxy Flag was "Off" but the server address was delivered by DHCP this flag would be automatically enabled and both the flag setting and the Server/Registrar address would be read-only
- Configuration examples and their behaviors
(Server/Registrar → page 214 is in "dom1.com" and resolves to an IP address "w.x.y.z", "dom2.com" resolves to "a.b.c.d"):

User input	OBP option	OBP domain	User	Message routed to
p.kelly@dom2.com	Enabled	(Not set)	p.kelly@dom2.com	w.x.y.z
p.kelly@dom2.com	Disabled	(Not set)	p.kelly@dom2.com	a.b.c.d
p.kelly	Disabled	(Not set)	p.kelly@dom1.com	w.x.y.z
p.kelly	Enabled	dom2.com	p.kelly@dom2.com	w.x.y.z
p.kelly@dom1.com	Disabled	dom2.com	p.kelly@dom1.com	w.x.y.z

- If you use an Outbound Proxy, you have also to configure the domain name of the Outbound Proxy server → page 174.
- The default setting for the Outbound Proxy flag is Off.
- Editing by phone: Switch Editor → page 157.

 → page 62  → page 115

Overview position on DSM

Enter the Position of the line in the MultiLine overview on the optiPoint display module.

 User Manual  → page 128

Payload security allowed

Mark "Payload security allowed" as on to make sure you set up a secure connection whenever possible.

 → page 96  → page 144

PC Port Settings

- "LAN port settings" → page 194

Ping

- Run this → PING test to check whether a server or another terminal device (e.g. the optiPoint 410/420 family S V7.0 or servers) can be reached by → IP or domain name.
- For this, enter or select an → IP address or domain name as a test target (the connection to which you wish to test).
- Value range table (for user specified IP):

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and — if DNS is applicable → page 41 — also Text Editor → page 155

 → page 95  → page 139

Port Control

This parameters are needed for development only. The following options are available:

- Service Agent
 -
- TestInterface
- Resource Sharing
- SNMP Port
- Web Server
 - Enables/disables the Web interface for this phone

 → page 99  → page 145

Primary DNS IP address

- Enter the → IP address of the → DNS server if not provided by → DHCP dynamically (→ DHCP IP assignment).
- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

- Editing by phone: IP Number Editor → page 159.

 → page 57  → page 114

QDC Address

Enter IP address of the Quality Data Collection Server. For more information refer to HiPath QoS Data Collection V1.0 Interface Description.

 → page 142

QDC Port

Enter port address of the Quality Data Collection Server. For more information refer to HiPath QoS Data Collection V1.0 Interface Description.

 → page 142

Quality of Service (QoS)

The QoS technology based on layer 2 and the two QoS technologies Diff-serv and TOS/IP Precedence based on layer 3 are allowing the VoIP application to request and receive predictable service levels in terms of:

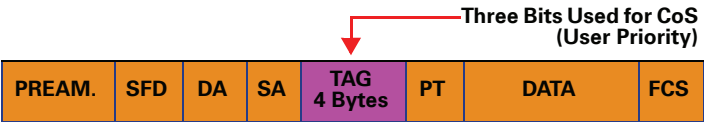
- data through put capacity (bandwidth)
- latency variations (jitter)
- and delay.

Please note that all these technologies are just marking packets which allow the network to classify and prioritize the packets accordingly. This means that the network decides which QoS configuration (marking) will be used and should be set in the endpoints. The default values are well known recommendations.

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. The value 7 is describing the highest priority and is reserved for network management. Value 5 is used for voice (RTP-streams) by default. Value 3 is used for signaling by default.



Layer 3 / Diffserv

Diffserv is classifying traffic flows, like voice (RTP-streams) or signaling messages into 2 per-Hop-Behavior groups:

1. Expedited Forwarded (EF referred to RFC 2598)
Expedited forwarded is used for voice (RTP-streams) by default.
 - High priority traffic to be handled at the arrival rate
 - DSCP value: 1 0 1 1 1 0 (DSCP = Diffserv Codepoint)
 - Effectively creates a special low-latency path in the network
2. Assured Forwarding (AF referred to RFC 2597)

Assured forwarding is used for signaling messages by default (AF31) and less stringent than EF in a multiple dropping system.

The AF values are containing two digits X and Y (AFX_Y), where X is describing the priority class and Y the drop level.

Four classes X are reserved for AFX_Y: AF1_Y (High Priority), AF2_Y, AF3_Y and AF4_Y (Low Priority).

Three drop levels Y are reserved for AFX_Y: AFX1 (low drop level), AFX2 and AFX3 (High drop level). 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.

Standard IPV4: Three MSB Called IP Precedence
(DiffServ Uses Six MSB Bits)

Version Length	ToS 1 Byte	Len	ID	Offset	TTL	Proto	FCS	IP-SA	IP-DA	Data
----------------	------------	-----	----	--------	-----	-------	-----	-------	-------	------

Layer 3 / IP Precedence (old recommendation, replaced by Diffserv)

IP Precedence is classifying traffic flows into 8 different precedence levels. These 3 Bits are the same as the priority Bits from Diffserv. Value 5 (BIN: 101) is used for voice by default. And value 3 (BIN: 011) is used for signaling. These values are already configured by configuring the DiffServ values.

If a routing partner can only work with one of the two standards (DiffServ or IP precedence, for example an older router that only works with IP precedence), then the router can translate the ToS field accordingly. This can be set for each PSTN peer or LAN interface.

QoS												
IP header											Ethernet-Header	
DiffServ											IEEE802.1p	
Diffserv-Codepoint	Binary	Decimal		Hex (8Bit)	Priority	Drop level			versus	IP Precedence	Binary	Decimal
		6-Bit	8-Bit			high	med	low				
EF	101110	46	184	B8	very high	x			<->	101	101	5
AF 43	100110	38	152	98	high	x			<->	100	100	4
AF 42	100100	36	144	90			x		<->	100	100	4
AF 41	100010	34	136	88				x	<->	100	100	4
AF 33	011110	30	120	78	medium	x			<->	011	011	3
AF 32	011100	28	112	70			x		<->	011	011	3
AF 31	011010	26	104	68				x	<->	011	011	3
AF 23	010110	22	88	58	low	x			<->	010	010	2
AF 22	010100	20	80	50			x		<->	010	010	2
AF 21	010010	18	72	48				x	<->	010	010	2
AF 13	001110	16	64	38	very low	x			<->	001	001	1
AF 12	001100	12	48	30			x		<->	001	001	1
AF 11	001010	10	40	28				x	<->	001	001	1

→ page 58 and → page 46 → page 117

RAM test

- Use this function to test the → RAM memory of your optiPoint 410/420 family S V7.0.
- The results are displayed after the test.

 → page 95  → page 139

Register by terminal name

- If set to "On", the phone will send REGISTER messages which contain the contents of the Terminal Name field → page 223. If set to "Off", the phone will send REGISTER messages which contain the contents of the Terminal Number field → page 224.
- Editing by phone: Switch Editor → page 157.

 → page 61  → page 115

Registration LEDs

- This option determines whether the line LEDs will be lit to show if they have been registered successfully when the phone starts up. If set to be On then as each line is successfully registered its LED will be set ON.

 → page 69  → page 123

Registration timer value

- This field determines whether the phone sends an expires header in the REGISTER messages that it sends, and if so, to what value it sets it. The expires header in a REGISTER is a suggestion to the Registrar server of how long it should be before the phones registration expires. To stop its registration from expiring, the phone has to send another REGISTER to the Registrar before its current one has expired. The expires value which the phone sends is only a suggestion - the actual value to be used will be supplied to the phone by the Registrar in the OK message that it sends in response to the REGISTER. Normally this will be the same as the value that the phone has suggested, but if the suggested value is outside the Registrars range of acceptable values, then it could be different.
- The phone actually adds 60 seconds to the value that is puts into the expires header, so that if the REGISTERs that it sends get delayed because of a congested network, they will still arrive at the Registrar before the registration expires.
- As an example, if 10 seconds is entered as the Registration Timer value, the REGISTER messages that the phone sends will have an expires header of 70 seconds (10 + 60). If 70 seconds is lower than the Registrars lowest acceptable value, it will ignore the suggested value and send back its lower limit value, e.g. 90 seconds. The phone will subtract 60 seconds from this, and use that as the amount of time to wait before sending its next REGISTER, so 30 seconds (90 - 60) later it will send out its next REGISTER.
- If the Registration Timer is set to 0, the phone will not put an Expires header into the REGISTER messages that it sends - i.e. it will not make any suggestion to the Registrar of how long it would like the registration to remain valid.
- Value range table:

Permitted values	alphanumeric
Range	0, 10 ... 4320 seconds
Default value	3600 seconds

- Editing by phone: Number Editor → page 157.

 → page 62  → page 115

Reservation Timer

Determines the timeframe for which a line remains reserved for a user who is dialling; after this timeframe another user who's phone is using the same line can access the line.

 → page 69  → page 123

Ringer Settings

- See Alert indication → page 163.

Rollover type

The Rollover ring setting will be used if the Keypad is currently active in a call when an incoming call arrives on a different line.

Selectable rollover types are:

- 1=No ring
- 2=Alert ring
- 3=Standard ring
- 4=Alert beep

 → page 69  → page 123

Rollover Volume

While you are active on one line of a keypad telephone, the rollover ringing feature signals additional incoming calls on other lines. The volume can be set from 1 to 5. For more information see operating manual.

 → page 69  → page 123

ROM test

- Use this function to test the → ROM memory of your optiPoint 410/420 family S V7.0.
- The results are displayed after the test.

 → page 95  → page 139

RTP packet size

Use this function to define the RTP G711/G729 packet size. Options are:

- Auto recognition
- 10 milliseconds
- 20 milliseconds

 → page 93  → page 121

RTP Version

Shows the version of RTP of the optiPoint 410/420 family S V7.0.

 → page 94

Secondary DNS IP address

See “Primary DNS IP address”.

Self Labelling keys test

Every dummy in the label is tested, whether he is writeable (only for opti-Point 420).

 → page 96  → page 140

Send Generic Traps to Management Center

Allows the user to control whether or not the phone sends generic (standardised) traps to the management center (see → page 219).

 → page 121

Send QDC Traps to Management Center

Allows the user to control whether or not the phone sends QCD (Quality Data Collection) traps to the management center (see → page 219).

 → page 121

Short description

Is the line name in DSM line overview and call view. Can only be set by administrator in web interface.

 → page 128

Show focus

The Show focus option allows a Keyset to be set so that the LED of the line that is currently being shown in the display flutters to identify it.

 → page 69  → page 123

SIP addresses

- Use this function to define the following IP addresses or host names.

IP address	Use
SIP Server	IP address or host name and port of the SIP proxy server (Hipath 8000).
SIP Registrar	This field is only used when the phone is in "Server" routing mode → page 216. It contains the IP address or host name and port of the registration server, to which the phone will send REGISTER messages. Either an IP address or a host name may be entered. With an address entered in the SIP Registrar field, the phone will register and be able to receive incoming calls, but in order to make outgoing calls it is also necessary to enter an address into the SIP Server field (see above).
SIP Gateway and Port	IP address or host name and port of the SIP gateway. E.g. for a hardware box to phone directly into the public network (conversion of SIP to TDM). SIP routing has to be set to "Gateway" for this function.
SIP Port	Access base IP port for receiving & sending SIP messages.
RTP Base Port	Access base IP port for RTP transport.

- Value range table:

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and if DNS is applicable → page 41 also Text Editor → page 155.

 → page 62  → page 115

SIP Auto answer

- This setting controls whether or not alerting calls can obey any auto-answer request signalled for the call. Automatic answering will only apply to the primary line of a Keyset. Mark this option enabled/disabled.
- When you dial a number using the CTI application while "Auto Answer" is activated, the telephone automatically switches to handsfree mode. If "Auto Answer" is deactivated, the telephone will ring first and you must then press the loudspeaker key or lift the handset to dial the number and set up the connection to the other station.

 → page 64  → page 116

SIP Auto reconnect

- This setting controls whether or not a toggle (or alternate) between the active call and held call can be signalled to the phone and be automatically applied. Automatic reconnect will only apply to the primary line of a Keyset. Mark this option enabled/disabled.
- Use this option for placing a call on hold and for retrieving it again using both a CTI application and the telephone.

 → page 64  → page 116

SIP Beep on auto answer

- This setting controls whether or not beeping is heard (via the current audio path) when an alerting call is auto-answered. Mark this option enabled/disabled.

 → page 64  → page 116

SIP Beep on auto reconnect

- This setting controls whether or not beeping is heard (via the current audio path) when a toggle (or alternate) between the active call and held call occurs automatically. Mark this option enabled/disabled.

 → page 64  → page 116

SIP password

- Value range table:

Permitted values	alphanumeric
Length min.	6 digits
Length max.	24 digits

- Editing by phone: Password Editor → page 158.

 → page 62  → page 115

SIP realm

- This field displays the realm that the phone is registered in.
- Authentication
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 62  → page 115

SIP routing

- When "Server" is selected, the phone will send REGISTER messages to the Registrar server, whose address is in the SIP Registrar field.
- If "Gateway" is selected, the gateway address of e.g. a hardware box is used to phone into the public network (direct conversion of SIP to TDM). When The phone will not send REGISTER messages.

Routing type	Use
Server	If a SIP proxy server is used.
Gateway	If a gateway is used.

 → page 62  → page 115

SIP server type

- Select the according server type:
 - HiQ8000
 - Broadsoft
 - Sylantro
 - Other

 → page 62  → page 115

SIP server validation

If this function is activated, a server certificate is requested and validated during the HiPath 8000 registration.

 → page 96  → page 144

SIP session timer value

- The expiration time for the session is set via the system/SIP Session Time value. This value is ignored if the SIP Session Timer is not enabled. The RFC standard defines a Minimum expiry time which is 90 sec. For detailed information see [RFC 4028](#) paragraph 4.
- Value range table:

Permitted values	numeric
Range	90 ... 3600 seconds (recommended: 1800)
Dafault value	3600 seconds

- Editing by phone: Number Editor → page 157.

 → page 62  → page 115

SIP session timer enabled

- The phone supports the SIP draft ietf-sip-session-timer-08. For detailed information relating to this draft please see <http://www.ietf.org>. 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. The interval for the re-INVITES is determined through a negotiation mechanism defined in the above draft. If a re-INVITE is not 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 need understand the draft and implement the SIP extensions for it to work.
- Default value: Off.
This is because some server environments support their own mechanism for auditing the health of a session (e.g. Broadsoft).
- Editing by phone: Switch Editor → page 157.

 → page 62  → page 115

SIP Signalling Version

- Shows the version of SIP signalling of the optiPoint 410/420 family S V7.0.

 → page 94

SIP Stack Version

- Shows the version of the SIP stack of the optiPoint 410/420 family S V7.0.

 → page 94

SIP Transport

- Use this function to define the transport protocol.

Protocol	Use
UDP	Use → UDP for SIP messages.
TCP	Use → TCP for SIP messages.
TLS	→ TLS requires the optiPoint 410/420 family S V7.0 to have a Root certificate.

 → page 62  → page 115

SIP user ID

- User name
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 62  → page 115

SNMP MIB2 errors

Lists the following packets:

- Discarded inbound packets
- Invalid inbound packets
- Discarded outbound packets
- Invalid outbound packets

 → page 121

SNMP password

- Specify the password that was defined in the → SNMP server as the password for accessing this server.
- Value range table:

Permitted values	alphanumeric
Length max.	24 digits
Default value	public

- Editing by phone: Password Editor → page 158.

 → page 92  → page 121

SNMP Queries Allowed

Allows the user to control whether or not the phone responds to → SNMP queries received from an SNMP manager.

 → page 121

SNMP Trap IP address or DNS name

- If an → SNMP server exists in the network, enter the → IP address or host name of this server (also called "Management Center").
- Value range table:

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and — if DNS is applicable → page 41 — also Text Editor → page 155.

 → page 92  → page 121

SNTP server address or DNS name

- If an → SNTP server exists in the network, enter the → IP address or host name of this server.
- Value range table:

Permitted values	numeric (with DNS also alphanumeric)
Length max.	15 digits (incl. dots) (with DNS also 92 digits)

- Editing by phone: IP Number Editor → page 159 and — if DNS is applicable → page 41 — also Text Editor → page 155.

 → page 67  → page 120

Survivability Backup Address

IP address of the backup server. Has to be entered manually as it cannot be retrieved via DHCP.

- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

Editing by phone: IP Number Editor → page 159.

 → page 100  → page 147

Survivability Backup Port

The "Backup port" option displays the port number used for SIP communication with the backup proxy server port.

 → page 100  → page 147

Survivability Backup Registration

The "Backup registration" option indicates whether or not the phone treats the Backup proxy server as a SIP .Registrar. If the setting is 'On', the phone tries to register its SIP address with the Backup proxy server.

 → page 100  → page 147

Survivability Backup Registration Timer

The "Backup Reg Timer" option displays the duration of the SIP registration requested by the phone when it registers with the backup proxy server. Note. The phone only registers with the backup proxy if the "Backup registration" setting is 'On'.

 → page 100  → page 147

Survivability Backup OBP

The "Backup OBP flag" indicates whether or not the Backup Proxy Server is used as an outbound proxy.

 → page 100  → page 147

Survivability Backup Transport

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

Options:

- TCP
- UDP

UDP is prepared.

 → page 100  → page 147

System Name

- See "Branding/Identity name".

Terminal Hostname

- This field is provided with the E164 number but you can change it. Enter a new hostname for the telephone. The hostname is transmitted to the DHCP server together with the MAC address while the telephone registers at the DHCP server. The DHCP server sends an IP address to the telephone; at the same time, it transmits this IP address together with the host name to the DNS server, where this association is registered. Within the DNS server's range, the telephone can now be addressed using its host name.
- If Mobility is using, the hostname is overwritten by the current E164 number of the Mobility user if the option "Use E164 as hostname" is enabled. The new hostname can be used to open the WBM of the phone in the browser.
- DHCP has to be activated to enable this feature.

Scenarios for "Terminal Hostname" handling:

- In the default state, the option "Use E164 as hostname" is activated and the default host name set. If the E164 entry is input or modified, this value is transferred to the "Terminal Hostname" field.
- The terminal host name can be modified by the administrator, even if the option "Use E164 as hostname" is activated. This modification is possible both via the telephone menu and via the web interface.
- Modification for a mobile user is only possible via DLS if the option "Use E164 as hostname" is deactivated.
- Once a value has been entered to the "Terminal Hostname" field, it is no longer possible to set this field to ZERO, not even via DLS.
- If a terminal host name is to be modified via DLS, the option "Use E164 as hostname" must be deactivated for this purpose.
- If the option "Use E164 as hostname" is activated for a mobile user, the E164 number of the mobile user is entered to the "Terminal Hostname" field when the data is downloaded.
- If the option "Use E164 as hostname" is deactivated for a mobile user, the value in the "Terminal Hostname" field is not modified when the data is downloaded. This can be useful when a device name is assigned to the IP address in the DNS, for example.
- If the option "Use E164 as hostname" was deactivated for a basic user, the value stored in the "Terminal Hostname" field is used when the data is downloaded (e.g. device name).

- If the option "Use E164 as hostname" was activated for a basic user, the E164 number of the basic user is entered to the "Terminal Hostname" field when data is downloaded.

Overview of the HostName handling on normal operation mode							
action	FLAG E-164 SET	Host Name: Change FROM	Host Name TO	E-164 Change FROM	E-164 TO	Host- Name Result	Remark
Change Hostame	N	A	B	1	1	B	
Change E164 #	N	A	A	1	2	A	
Change Hostame	Y	A	B	1	1	B	
Change E164 #	Y	A	A	1	2	2	
Overview of the HostName handling on Mobility operation mode							
action	FLAG E-164 SET	Host Name MOB	E-164 MOB	Host Name Basic	E-164 Basic	HN during logon Result	Remark
Log on MOB user	N	NULL	3	A	2	A	No change of host- name, Phone will send a HN to DLS for Mobile user
Log on MOB user	N	C	3	A	2	A	No change of hostname
Log on MOB user	Y	NULL	3	2	2	3	After mob logoff MOB- HN = MOB-E164
Log on MOB user	Y	C	3	2	2	C	

 → page 57
  → page 114

Terminal IP address

- Enter the → IP address for the optiPoint 410/420 family S V7.0 if not provided by → DHCP dynamically (→ DHCP IP assignment).
- If the value was assigned dynamically, it can only be read.
- The change will only have effect if you restart the phone.
- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

- Editing by phone: IP Number Editor → page 159.

 → page 57  → page 114

Terminal mask

- Enter the → Subnet Mask for the optiPoint 410/420 family S V7.0 if not provided by → DHCP dynamically (→ DHCP IP assignment).
- If the value was assigned dynamically, it can only be read.
- The change will only have effect if you restart the phone.
- Value range table:

Permitted values	numeric
Length max.	15 digits (incl. dots)

- Editing by phone: IP Number Editor → page 159.

 → page 57  → page 114

Terminal name

- The phone will send REGISTER messages containing the contents of this parameter, but only when the Register by Name field → page 210 is set to "On" and the SIP Routing field → page 216 is set to "Server".
- Whether this parameter is used, depends on the configuration of the registrar server.
- Value range table:

Permitted values	alphanumeric
Length max.	92 digits

- Editing by phone: Text Editor → page 155.

 → page 61  → page 115

Terminal number

- The phone will send REGISTER messages containing the contents of this parameter, but only when the Register by Name field → page 210 is set to "Off" and the SIP Routing field → page 216 is set to "Server".
- Whether this parameter is used, depends on the configuration of the registrar server.
- Value range table:

Permitted values	numeric
Length min.	1 digit
Length max.	20 digits

- Editing by phone: Number Editor → page 157.

 → page 61  → page 115

Terminating line preference

Terminating Line selection – provides connection of the phone to one of the lines appearing on it, on an automatic basis when calls are alerting or ringing (audible) and the user goes off-hook. There are five terminating options.

A Keyset is assigned one of the following preferences.

- **Ringing Line Preference** – A line in the ringing state is selected for terminating calls. In the case of multiple lines, lines shall be selected on a first-in, first-out basis.
- **Ringing Line Preference with prime line preferred** – Same as Ringing Line Preference, but if the prime line is ringing at any time it is signaled and selected before calls on secondary lines.
- **Incoming Line Preference** – In the case of multiple lines alerting or ringing on a device, an incoming line is selected based on its ringing assignment. Ringing lines are selected on a first-in, first-out basis first, then alerting (visual only) lines are selected on a first-in, first-out basis.
- **Incoming Line Preference with prime line preferred** – Same as Incoming Line Preference, but if the prime line is ringing at any time it is signaled and selected before calls on secondary lines. Prime line must have ringing arrangement.
- **No (Terminating) Line Preference** – A line key must be pre-selected or post-selected each time the user elects to answer a call.

Select the according terminating line preference:

- Ringing
- Incoming
- Incoming PLP
- Ringing PLP
- None

See also Originating line preference on → page 204.

 → page 69  → page 123

Time zone offset

- The specification describes the shift in hours corresponding to the time zone information of the SNTP server.
- Make an entry only if an → SNTP server provides time zone information.
- Value range table:

Permitted values	numeric
min.	-12
max.	+12
Default value	0

- Editing by phone: Offset Editor → page 161.

 → page 67  → page 120

Timer High Priority

Set the delay in seconds after the last high priority item change before saving the changes to the DLS.

- Causes time out before saving high priority user data.
- Value range: 0 to 999

 → page 102  → page 134

Timer Medium Priority

Set the delay in seconds after the last medium priority item change before saving the changes to the DLS. Used in conjunction with the Medium Priority peg count threshold.

- Timeout before saving medium priority user data
- The value has to be a number in the range 1 to 999

 → page 102  → page 134

Transaction timer

The Transaction Timer sets the amount of time that the phone will wait for a response to any SIP request that it sends. The time value is in milliseconds and the user can enter any value between 3000 and 32000. The default value is 32000 milliseconds.

 → page 66  → page 115

Transfer on hangup

1. If "Transfer on hangup = **no**," the action is depending on the direction of the involved calls:

1.st call	2nd. call	action
Incoming	Incoming	the active call is released, the held call is re-presented.
outgoing	incoming	the active call is released, the held call is re-presented
Incoming	outgoing	the calls are joined
outgoing	outgoing	the calls are joined

2. If "Transfer on hangup = **yes**," two calls can be connected by releasing the handset
- The "Transfer on hangup" option does not control the availability of the transfer (blind, ringing, attended)
 - The "Transfer on hangup" option does not control the availability of the "Join" option provided by the local phone menu. For example, if "Transfer on hangup = no," then it is possible to connect one active and one held call by selecting the "join" via optiGuide. Note that "Join" is only offerered for two incoming calls – otherwise it is called "Complete Transfer"
 - The default for the administration option and for the user option shall be "on."

The adminstration option is controlled via DLS

 → page 64  → page 116

Transfer on Ringing

If this function is active, a consultation can be transferred after you have dialled the third participant's number, but before the third party has answered the call.

 → page 64  → page 116

Unauthorised Logoff Trap

Determines wheter or not the phone sends an → SNMP trap whenever the DLS notifies the Phone of an unauthorised logoff attempt.

 → page 102 or  → page 134 or → page 121

Upload Configuration

- Use this function to save (back up) an optiPoint 410/420 family S V7.0 configuration on the → FTP server.
- The following parameters must be set before the upload operation:
 - → Download server IP address or DNS name, → FTP path,
 - Configuration download filename
 - → FTP account name
 - → FTP username, → FTP password

 → page 76  → page 138

Upload/Download Status

Shows the status of the following downloads with the date of the last transfer:

- Application file download
- Configuration file download
- Configuration file upload
- Hold music file download
- System configuration download
- Phone configuration download
- Personal directory import
- Personal directory export

 → page 91  → page 113

Use deployment service (DLS)

Allows the administrator to manually set the download mechanism to be the deployment service (**DLS**).

 → page 88  → page 135

Use dynamic hostname concept

This option is to be considered in combination with the entry in the "Terminal Hostname" field. Further information is provided on → page 221.

 → page 57  → page 114

Use secure/non-secure configuration download

Allows the administrator to manually set the configuration download mechanism to be **secure** (HTTPS) or **non-secure** (FTP).

 → page 88  → page 135

Versions Info

- Displays some telephone versions like:
 - Application version
 - SIP stack version
 - SIP signalling version
 - RTP version
 - Web content
- The application version identifies the release level of the loaded software. The others relate to versions of internal software components.

 → page 91  → page 113

VLAN discovery method

- Can be defined only if the → Layer 2 support is activated (→ Quality of Service (QoS)).
- Use this function to define the location from where the → Manual VLAN identifier should be fetched, if → VLAN is used.

Manual	The ID entered in → Manual VLAN identifier is used.
DHCP	If a → DHCP server is used, then the ID delivered by this server is applied.

 → page 59  → page 117

Voicemail number

- The number of where your voice mail server is located.
- Value range table:

Permitted values	numeric
Length max.	20 digits

- Editing by phone: Number Editor → page 157.

 → page 66  → page 115

WAP Connection Type/mode

- Choose the protocol over which the data transfer should take place for
→ WAP applications: → HTTP or → WSP.

 → page 81  → page 137

WAP proxy Password

Allows the user password on the WAP server to be changed.

 → page 81  → page 137

WAP Server Address

- If a → WAP server exists, enter the → IP address of this server here.
- Standard address → page 243.
- IP editor in optiPoint 410/420 → page 159.

 → page 81  → page 137

WAP Server Port Number

- If a → WAP server exists, enter the → Port number for communication with this server here.
- Standard value → page 243.
- Number editor in optiPoint 410/420 → page 157.

 → page 81  → page 137

WAP proxy Username

Allows the username on the WAP Server to be specified

 → page 81  → page 137

Abbreviations and Specialized Terms

You will find more information in the relevant literature on the Network Technology and → VoIP.

DHCP

Abbreviation for "**D**ynamic **H**ost **C**onfiguration **P**rotocol".

The DHCP is an Ethernet protocol that allows for the automatic configuration of IP based endpoints. Additional information see → page 38.

DNS

Abbreviation for "**D**omain **N**ame **S**ystem".

Additional information see → page 41.

DTMF

Abbreviation for "**D**ual **T**one **M**ulti **F**requency".

DLS

The Deployment and Licensing Server (DLS) is a HiPath Management application that provides an integrated solution for the customers and the service personal to administer workpoints (that are optiClients and opti-Point devices) in HiPath- and non-HiPath networks

EAP

Extensible Authentication Protocol

FTP

Abbreviation for "**F**ile **T**ransfer **P**rotocol".

Is used for transferring files in networks, e.g., to update telephone software (→ Download Application).

G.711

Audio protocol for uncompressed voice transmission. Requires a bandwidth of 64 kbit/s.

G.722

The G.722 recommendation describes ADPCM coding with a sub-band. The bandwidth for the sub-band is 7 kHz at a sampling rate of 16 kHz. The transfer rate is 64 kbps, voice quality has a MOS rating of 4.5 which is quite high.

G.723

Audio protocol for compressed voice transmission. The quality is worse than in → G.711 and → G.729. Requires a bandwidth of about 6 kbit/s.

G.729

Audio protocol for compressed voice transmission. The quality is worse than in → G.711 and better than in → G.723. Uses a bandwidth of about 8 kbit/s.

Gateway

Mediation components between two different network types, e.g., → IP network and ISDN network.

HTTP

Abbreviation for "**H**ypertext **T**ransfer **P**rotocol".
Protocol for the transfer of data in → IP networks.

IP

Abbreviation for "**I**nternet **P**rotokoll".

IP address

Also called "→ IP" in short. 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. To simplify the notation, voice names can be released from a → DNS into the IP addresses.

Jitter

Runtime fluctuations in data transmission in → IP networks.

LAN

Abbreviation for "**L**ocal **A**rea **N**etwork".

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

Abbreviation for "**L**iquid **C**rystal **D**isplay".
Display of numbers, text or graphics with the help of liquid crystal technology.

LDAP

Abbreviation for "**L**ightweight **D**irectory **A**ccess **P**rotocol".
Simplified protocol for accessing standardized directory systems, e.g., a company telephone directory.

LED

Abbreviation for "**L**ight **E**mitting **D**iode".
Cold light illumination in different colours at low power consumption.

MAC

Abbreviation for "**M**edium **A**ccess **C**ontrol **A**ddress".
A 48-bit address with the help of which a terminal device (e.g., → IP telephone or Network card) identifies itself uniquely in a network all over the world.

MIB

Abbreviation for "**M**anagement **I**nformation **B**ase".
Database containing descriptions of error messages of the devices and functions in a network.

PBX

Abbreviation for "**P**rivate **B**ranch **eX**change".
Private telephone system that connects the different internal devices to the ISDN network

PING

Abbreviation for "**P**acket **I**nternet **G**roper".
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. The result of the test displays the success / failure of the transmission and possible additional information such as the transmission time.

PoL

Abbreviation for "**P**ower **o**ver **L**AN".

Port

Ports are used in → IP networks to permit several communication connections simultaneously. Different services often have different port numbers.

QoS

Abbreviation für „**Q**uality **o**f **S**ervice".
Additional information see → page 46.

RTCP

Abbreviation for "**R**ealtime **T**ransport **C**ontrol **P**rotocol".

RTP

Abbreviation for "**R**ealtime **T**ransport **P**rotocol".

RAM

Abbreviation for "**R**andom **A**ccess **M**emory".
Memory with read / write access.

ROM

Abbreviation for "**R**ead **O**nly **M**emory".
Memory with read only access.

SDP

Abbreviation for "**S**ession **D**escription **P**rotocol".

SIP

Abbreviation for "**S**ession **I**nitiation **P**rotocol".
Protocol standard for initialising calls in → IP networks. Additional information see → page 34.

SNMP

Abbreviation for "**S**imple **N**etwork **M**anagement **P**rotocol".
The protocol is used for communication with servers that takeover network management functions. This includes for example, protocolling errors that occur in network components (SNMPTrap). Additional information see → page 43.

SNTP

Abbreviation for "**S**imple **N**etwork **T**ime **P**rotocol".
The protocol is used between timeservers and terminal devices of a network to synchronize the time of the terminal device. Additional information see → page 43.

SRTP

The Secure Real-time Transport Protocol is a profile of the Real-time Transport Protocol (RTP), which can provide confidentiality, message authentication, and replay protection to the RTP traffic and to the control traffic for RTP, the Real-time Transport Control Protocol (RTCP). More Information see [RFC 3711](#)..

Subnet Mask

Classifies networks in A-, B- and C networks. Each class has 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, there are 254 → IP addresses.

Switch

Network device that selects a path or circuit for sending data to its next destination. A switch may also include a router function.

TCP

Abbreviation for "**T**ransmission **C**ontrol **P**rotocol".

TLS

Abbreviation for "**T**ransport **L**ayer **S**ecurity".
This protocol ensures privacy between communicating applications.

UDP

Abbreviation for "**U**ser **D**atagram **P**rotocol".

VLAN

Abbreviation for "**V**irtual **L**ocal **A**rea **N**etwork".
Additional information see → page 44.

VoIP

Abbreviation for "**V**oice **o**ver **I**P".
E.g., voice transmission through → IP technology.

WAP

Abbreviation for "**W**ireless **A**pplication **P**rotocol".
Synonym for graphical applications on mobile telephones, organizers and other suitable terminal devices, transferred in accordance with the protocol by the same name.

WSP

Abbreviation for "**W**ireless **S**ession **P**rotocol".
Protocol for transferring data to → WAP-enabled terminal devices.

Troubleshooting

General Troubleshooting Tips

The suggestions in this section are general troubleshooting tips.

- If using a → DHCP server, make sure that it is operating correctly. The DHCP server should show an incoming request from the → MAC address listed on the product label.
- If you do not see the idle menu after you connect the Ethernet cable, make sure the power cord and the Ethernet connection are secure.
- If you do not hear a dial tone, make sure that the telephone-handset line cord is plugged into the IP Phone port. Also make sure that the appropriate Ethernet cable (crossover or straight-through) is used, and that all cable connections are secure.
- A fast-busy tone indicates that the number you called is not valid or that external circuits are busy. Verify the number or try your call again later.
- If you place a call to another IP telephone, hear ringback, and the called party answers but you cannot hear the speaker's voice ("Incompatible terminal"), verify that the optiPoint 410/420 family S V7.0 and the other IP telephone support at least one common audio codec → page 165.
- In some cases it can be useful if you perform a update of the phone's software → page 74.

Fault Finding

This section provides guidance of identifying the source of a problem which is affecting the phones ability to function correctly. The user display can provide some basic information which helps this process. A description of the various faults that are displayed and possible actions are described below.

Display reported faults

No Network

- Ensure that the network hub or switch that the phone's is connected to is turned on and that the Hub/Switch port the phone is connected to is active and correctly configured.
- Try using a different network cable.

No DHCP Server

- Ensure that the phone's DHCP setting is correctly configured → page 175.
- Check the phone's VLAN ID is correct → page 203. If the phone's VLAN is provided by DHCP check that the phone successfully discovers a VLAN on start-up.
- Ensure that the DHCP server is on the same VLAN as the phone.
- Confirm the L2 QoS settings are correctly configured → page 193 and match those configured in network components.
- Check that the DHCP Server is active and correctly configured and has enough free leases.
- Check the configuration of the network switch's port which the phone is connected to. If the network is using VLANs ensure that the switch's port is configured for the same VLAN as the phone.

No Server Set

- Ensure that the both the SIP Server and SIP Registrar addresses are correctly configured → page 214.

No Registrar Set

- Ensure that the SIP Registrar address is correctly configured → page 214.

No Sip Gateway Set

- Ensure that the SIP Gateway address is correctly configured → page 214.

No Name Set

- Ensure that the Terminal Name is correctly configured → page 223.
- Ensure that the "Register By Name" setting → page 210 is correct. This should be "On" if the phone is using the Terminal name and "Off" if the phone is using the Terminal Number.

No Terminal address set

- Ensure that the Terminal Number is correctly configured.
- Ensure that the Register By Name setting is correct. This should be "On" if the phone is using the Terminal Name and "Off" if the phone is using the Terminal Number.

No SIP Password

- Ensure that the SIP Password is correctly configured → page 215.

No Server...

- Ensure that the Server/ Registrar addresses are correctly configured → page 214 and can be pinged → page 207 – do not attempt to ping domain names used when configuring the phone for DNS-SRV. If the SIP Server or SIP Registrar can not be pinged and are configured as host names see → page 242.
- Ensure that the phone is displaying the same time as the SIP Server. If not see → page 242.
- Ensure that the "Register By Name" setting → page 210 is correct. This should be "On" if the phone is using the Terminal Name and "Off" if the phone is using the Terminal Number.
- Ensure that the Terminal Name → page 223 or Number → page 224 (depending on the "Register By Name" setting) is the same as that held by the SIP Server.
- Ensure that the SIP Transport is correctly configured → page 217.
- Ensure that the Outbound Proxy setting is correctly configured → page 205. If the Outbound Proxy setting is On, the Default OBP Domain should be correctly configured → page 174.

Status Reporting Error

- The most likely cause of this problem is that the phone can not contact the SIP Server. Ensure that the SIP Server is active and can be pinged by the phone → page 207.

Authentication Failure or Password Error

- Check SIP Username → page 218, SIP Password → page 215 and SIP Realm → page 215 against information held in SIP Server.
- Ensure the correct time and date is displayed by the phone.

Terminal Incompatible" whenever call is attempted

To avoid compatibility problems try the following:

- Ensure that all IP endpoints in your system, including SIP Gateways, are using compatible codec → page 165 and/or compression → page 170 settings. If problems persist configure all endpoints to G.711 speech with no Silence Suppression → page 188.
- Use packet size "Auto" for RTP → page 212.

No memory after a Software Update

In some cases "No memory" is displayed after an application download → page 138.

- Restart the phone → page 144 to free up memory.
- Perform the software update again.

If "No memory" is still displayed:

- Save the current configuration → page 76.
- Restore the factory settings → page 98.
- Re-enter the minimal set of IP → page 48 and FTP → page 75 parameters.
- Perform the software update again.
- Restore the saved configuration → page 76.

Security

Payload not Encrypted

The payload between two optiPoints is not encrypted (no lock displayed on the screen).

1. Check whether the phones have the current time. The use of a time server is recommended for this.
2. Check whether the phones use TLS as transport protocol.

TLS Authentication Failed

"TLS Authentication failed" is displayed on the screen. A possible cause is that "SIP server validation" is activated but no certificate is installed on the phone.

No Registration with Line Keys

A line key does not register when digest authentication is activated. The reason for this is that realm ID is required to be registered with the line key from V7.

Error Messages optiPoint 410 entry

LEDs on the phone:

The LEDs flash till you switch to the Administration Mode → page 54.

No IP address

The DHCP server cannot assign a terminal IP address → page 53 (Code 1).

Possible solution

Check the DHCP server.

Terminal Mask not assigned

The DHCP server has failed to assign a Terminal Mask → page 53 (Code 2).

Possible solution

Check the DHCP server.

No Default Route

The DHCP server cannot assign a default route → page 53 (Code 3).

Possible solution

Check the DHCP server.

No IP Address is set

The DHCP IP assignment → page 54 is switched off and no terminal IP address is configured (Code 4).

Possible solution

Activate the DHCP IP assignment → page 54.

Enter the terminal IP address → page 54.

No terminal Mask is set

- ▶ The DHCP IP assignment → page 54 is switched off and no terminal Mask is configured (Code 5).
- ▶ **Possible solution**
- ▶ Activate the DHCP IP assignment → page 54.
- ▶ Enter the terminal Mask → page 54.

No Default Route is set

- ▶ The DHCP IP assignment → page 54 is switched off and no default route is configured (Code 6).
- ▶ **Possible solution**
- ▶ Activate the DHCP IP assignment → page 54.
- ▶ Enter the default route → page 54.

Subscriber identity not set

- ▶ The subscriber identity (number or name) is not configured (Code 7).
- ▶ **Possible solution**
- ▶ Enter number or/and name → page 115.

No network

- ▶ The telephone cannot find the network (Code 8).
- ▶ **Possible solution**
- ▶ Check the network cable.

SIP Server address not set

- ▶ The DHCP setting is switched off an a SIP server has not been configured (Code 9).
- ▶ **Possible solution**
- ▶ Configure SIP server address → page 115.

SIP Server not responding

- ▶ The SIP Server Address has been set but the SIP server is not responding (Code 10).

Possible solution

- ▶ Check SIP realm (→ page 115).

SIP registration error

- ▶ The SIP server/proxy has rejected registration of the phone (Code 11).

Possible solution

- ▶ Check SIP user ID and password (→ page 115).

Common problems

Phone Fails to Display Correct time and Date

- Ensure that the optiPoint 410/420 family S V7.0 phone's time zone is correct → page 225. If the time zone is incorrect and can not be edited it has been configured by the DHCP server on start-up so check the configuration of the DHCP server which is supplying the phone's IP address.
- Check that the phone's Daylight Savings configuration is correct → page 174.
- SNTP Address is correct and can be pinged → page 207. If the SNTP Address can not be pinged and is a host name see → page 242.
- Ensure that the SNTP Server the phone is configured to is displaying the correct time.

Phone Can Not Contact Host Names

- Ensure that the "DNS Domain Name" is correctly configured → page 182.
- Ensure that the "Prim DNS IP addr" is correctly configured → page 207 and can be pinged → page 207.

Poor Speech Quality

- Poor speech quality can be the result of an overloaded network. Consider the implementing VLANs → page 203 and QoS → page 208.
- Ensure that all IP endpoints in your system, including SIP Gateways, are using compatible Speech Codec settings to avoid Duplex Mismatch → page 165.
- Ensure that the Room Character is correctly configured for the type of room the phone is located in (see User Manual, chapter "Room Character").

Display sticks at "Calling.." whenever call is attempted

- The most likely cause of this problem is that the phone can not contact the SIP Server. Ensure that the SIP Server is active and can be pinged by the phone → page 207.

Phone Configurations

This section identifies the configuration settings to allow the phone to operate in various system environments these include for example:

- HiQ8000
- Broadsoft
- Sylantro

This configuration is a common one dealing with settings generic to all systems. System specific ones follow.

Common Configuration (Factory Defaults)

Function	standard value
Administration password Password Editor → page 158	123456
Application DL filename*	ipphone.app
Audio mode	G711 Preference
Configuration DL filename	
Default route	0.0.0.0
DHCP IP assign	on
LAN Port Setting	Auto
PC Port Setting	Auto
VLAN Discovery	DHCP
DL server IP address	0.0.0.0
Domain name	
FTP account name	guest
FTP password Password Editor → page 158	123abc
FTP path	null
FTP username	guest
Gatekeepers	0.0.0.0
Gateway	0.0.0.0
Hold music filename	MoHFile
IP routing	0.0.0.0 (all)
Jitter buffer	normal
Compression	G729
G711 Silence	Off
Language	English
LDAP server address	0.0.0.0

Function	standard value
LDAP server port	389
National Dial Prefix	0
QoS L2/L3	Off/Off
SNMP password Password Editor → page 158	public
SNMP trap ID address	0.0.0.0
SNTP IP address	0.0.0.0
Terminal IP address	0.0.0.0
Terminal mask	0.0.0.0
Time to live	3
Time zone offset	+0
User password Password Editor → page 158	null
WAP mode	HTTP
WAP server address	0.0.0.0
WAP server port	9200

* Filename is postfixed by a letter a, b, c or d for the various language packages.

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