



HiPath optiPoint 410 S / 420 S Using with Asterisk

Configuration Guide

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1 Overview	3
2 System Requirements	3
3 Configuration of Asterisk	3
3.1 SIP Registration	4
3.2 Dial Plan	7
4 Basic Configuration of optiPoint Telephones	8
4.1 Web Interface	8
4.2 Network IP Address and Routing	12
4.3 Quality of Service	14
4.4 SIP Environment	15
5 Multi Line	17
6 Voicemail	18
6.1 Settings in voicemail.conf	18
6.2 Settings in extensions.conf	19
6.3 Settings in the file sip.conf	20
6.4 Setting up the Message Key on the optiPoint Telephone	21
7 Call Pickup	22
7.1 Group Definition	22
7.2 Execution of Call Pickup	22
8 Music on Hold (MOH)	24
9 Reference List	26
10 Sample Configuration	27

1 Overview

Using the products from the Siemens HiPath portfolio you can design communication solutions for a wide variety of requirements (i.e. scalability and features). Apart from IP communication systems, optiPoint telephones with SIP protocol are an integral part of these solutions. These telephones can - due to the standards used - easily be integrated into open SIP-based environments.

This documentation helps you to start up a Siemens optiPoint SIP telephone with only a few steps. It also shows the preparation of the SIP system, using an Asterisk open source PBX environment as an example.

2 System Requirements

Asterisk runs on the following operating systems: Linux, BSD, Mac OS X.

As long as you use Asterisk for VoIP only, no specific hardware requirements have to be met. If a larger number of codec transcodings is necessary, you should use a more powerful PC hardware.

In case of compatibility questions regarding protocols or hardware please refer to <http://www.asterisk.org> (see Section 9, "Reference List").

3 Configuration of Asterisk

We assume that you are familiar with the installation and basic usage of an Asterisk environment. If you need further help, there is a wide variety of literature available as well as comprehensive information in the world wide web (see Section 9, "Reference List").

We will explain the basic statements in the configuration files *sip.conf*, *extensions.conf* and *voicemail.conf* using an running sample configuration for two optiPoint telephones.

Configuration of Asterisk

SIP Registration

3.1 SIP Registration

The file *sip.conf* has to have an entry for each user agent, i.e. for each telephone.

3.1.1 Syntax

The syntax of the configuration file must correspond to the following pattern:

```
[general]
parameter1=value
parameter2=value
[User Agent 1]
parameter1=value
parameter2=value
```

3.1.2 Sample configuration

The global section `[general]` contains statements that are valid for all user agents.

Example:

```
[general]
context=default
realm=usergruppe@asterisk
bindport=5060
bindaddr=0.0.0.0
allow=alaw
qualify=yes
dtmfmode=info
```

Explanations:

- `context`
The `context` groups certain extensions that are defined in *extensions.conf*.
- `realm`

Name of the `realm`. Within a `realm` certain combinations of user & password are valid. During registration the Asterisk server assigns the `realm` name to the telephone. The presetting is `asterisk`.

- `bindport`
UDP port the asterisk server uses for receiving data. Presetting: 5060.
- `bindaddr`
IP address used to receive incoming SIP connections. Using the presetting `0.0.0.0` Asterisk receives SIP connections on all network interfaces and aliases.
- `allow`
Permitted codecs. Only the codecs listed here are permitted. The order of the `allow` entries determines the preference of the codecs.
- `qualify`
If the value `yes` is used, you can monitor latency and availability of the end point. As per presetting, an end point is considered to be available if the latency is below 2,000 ms. Instead of `yes/no` you can also specify a maximum latency in milliseconds.
- `dtmfmode`
Using DTMF (Dual Tone Multiple Frequency) tone signals to control the connection are sent via the keypad. These are used in menu-driven info services (Attendant Control), e.g. the voicemail features of Asterisk. There are 3 options to transmit the signals 1) inband: the tone signal is faded into the voice signal; 2) outband: another data stream is started for the transmission of the tone signal; 3) instead of a tone signal an INFO field is sent. optiPoint telephones support both inband as well as INFO fields. We recommend to use INFO fields as in case of using inband and a high degree of compression there might be errors in the data interpretation. Set `dtmfmode` to `info`.

The following section explains the entries for the individual user agents.

Configuration of Asterisk

SIP Registration

Example First User/Telephone:

```
[10]
type=friend
context=myphones
username=10
secret=123456
host=dynamic
```

Example Second User/Telephone:

```
[11]
type=friend
context=myphones
username=11
secret=123456
host=dynamic
```

Explanations:

- `type`
Three types are available: `peer`, `user` and `friend`. For telephones (i.e. user agents) you have to enter `friend`.
- `context`
The section in *extensions.conf* that is valid for the particular user, has to be labelled with the value used here; in the example: `[myphones]`.
- `username`
This parameter is only necessary if the user name contacted by Asterisk in case of an incoming call is different from the registered user name or if the user is not registered. The user is registered under the name or the number heading the section.
- `secret`
This is the password the user needs to register at the SIP server. If you do not want to save the password in plain text, you can use `md5secret` instead of `secret` and use the MD5 hash of a string consisting of the user name, realm and password for the value. In this case, the following steps have to be completed:

1. To generate the MD5 hash for the password, enter the following command into the Linux command line:

```
echo -n "username:realm:secret" | md5sum
```

(in the example: `echo -n "10:usergruppe@asterix:123456" | md5sum`)

2. Enter the MD5 hash. In the example:

```
md5secret=9ef347178a279fffea649ac16c6b7510
```

- host

Possible values are `dynamic`, an IP address or a host name. If the telephone is to register itself, enter `dynamic`.

3.2 Dial Plan

Syntax:

```
exten => name, priority, application()
```

Example:

```
[myphones]
```

```
exten => 10,1,Dial(SIP/10,5,r)
```

```
exten => 11,1,Dial(SIP/11,5,r)
```

Explanations (for lines 1 and 2 in the example):

- [myphones]

Context grouping several extensions. Other sections refer to this context, e.g. in the file *sip.conf*.

- exten =>

This indicates that one step of an extension is defined. Extensions define how Asterisk is to act in case of an incoming call or if a certain number is entered in a channel.

- 10

Extension number. If this number is called, the actions defined in the corresponding extension are executed.

- 1

Priority of this step of the extension. An extension may consist of several steps executed in the order of their priorities.

Basic Configuration of optiPoint Telephones

Web Interface

- `Dial(SIP/10,5,r)`

The application defined in this step is `Dial()`. In the example, the system tries to call the SIP participant number 10 for a duration of 5 seconds. The option `r` specifies that the caller hears a ring tone while the telephone called is ringing.

4 Basic Configuration of optiPoint Telephones

Currently, the portfolio of the SIP-based optiPoint devices 410 S and 420 S comprises 9 models. To a large extent, the instructions and steps in the following section are independent of the model. For additional information about the individual optiPoint models, please refer to the corresponding data sheets, user guides and administrator guides, see Section 9, "Reference List").

There are 4 options to configure an optiPoint telephone:

- **Web Interface:** The telephone can be configured and controlled via a standard web browser (recommended).
- **Database-driven configuration tool DLS (Deployment Service):** This Siemens IP telephone manager (available at no charge) is a powerful tool for configuring larger telephone systems.
- **optiGuide:** User interface of the telephone. For additional information, please refer to the corresponding User Guide or Administration Guide, see Section 9, "Reference List".
- **XML interface of the telephone.** For more information, please refer to the XML Administrator Guide, see Section 9, "Reference List".

The following section describes the configuration via the web interface as this is the most convenient and viable method for most installations.

4.1 Web Interface

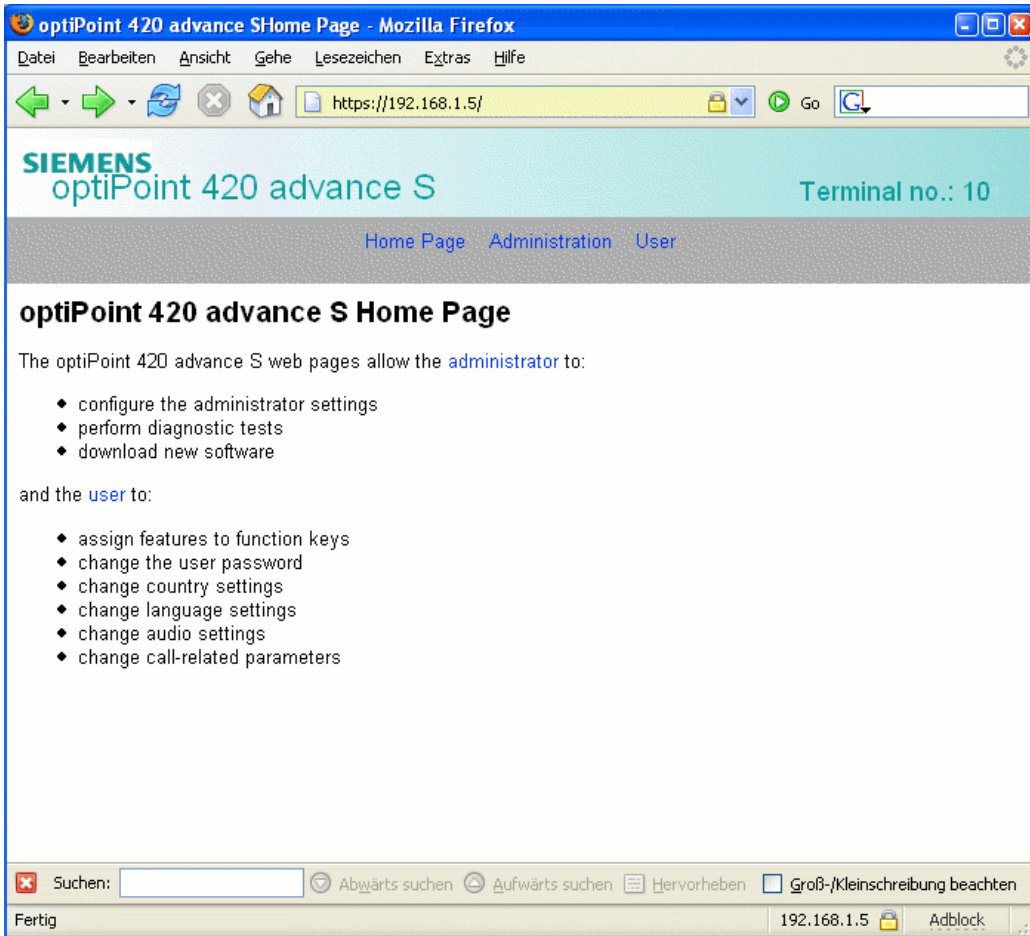
Open your browser and enter "https://" followed by the IP address of the optiPoint telephone, e.g.:

"https://192.168.1.5".




Prior to loading the page the browser displays a warning indicating that the certificate sent by the optiPoint device is not verified or installed. In this context, you can ignore this warning.

The website of the telephone is opened. If it is already configured, the terminal number (i.e. the phone number) of the device is displayed in the upper right corner.



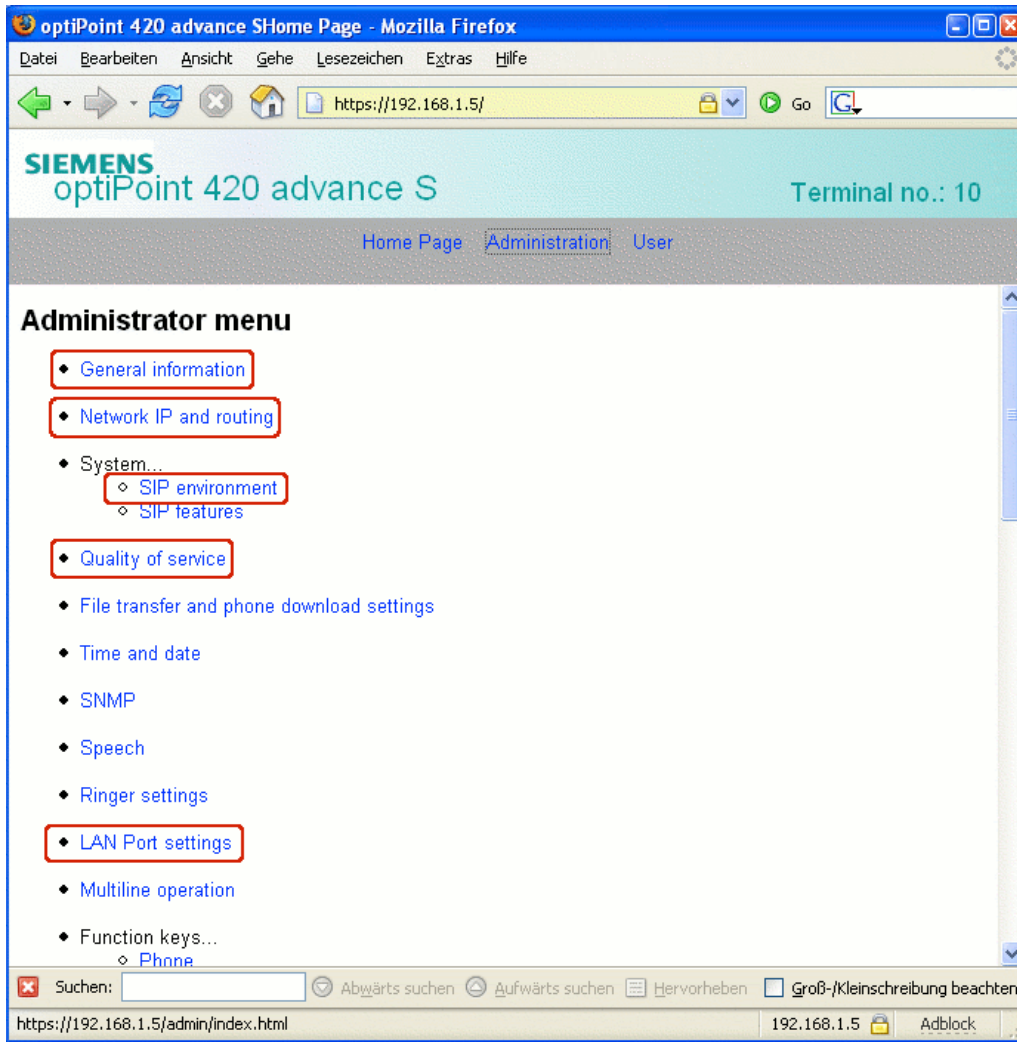
Select the **Administration** section and enter the administrator password. The initial password is "123456". You should change the password after the first login.

 The user of the telephone can also modify device settings. To do so, he has to log in into the **User** section.

The Administrator Menu is displayed. In the following screenshot all sub-menus that are or may be relevant for operation with Asterisk are highlighted.

Basic Configuration of optiPoint Telephones

Web Interface



General Information

This section contains general information about the telephone, e.g. software release or MAC address.

Network IP and routing

In this section the IP settings are entered. If DHCP is supported within the network, no changes should be necessary here. During the first setup of the optiPoint telephone without DHCP these settings have to be entered via the local Administrator Menu in the telephone (optiGuide) as follows:

1. Press the right arrow key and enter the numbers 1,3 in idle mode.
2. Enter the administrator password.

3. Select the menu options using the arrow keys. The following options are available: **DHCP IP assign** (activate/deactivate DHCP usage), **Terminal IP address** (IP address of the optiPoint), **Terminal mask** and **Default route** (IP address of the router).

SIP environment

This section is used to enter the SIP-related settings.

Quality of Service

The QoS (Quality of Service) settings have to be synchronized with the network.

Lan Port setting

You can specify the transfer rates of the two LAN ports here.

For other settings, please refer to the Administrator Guide, see Section 9, "Reference List".

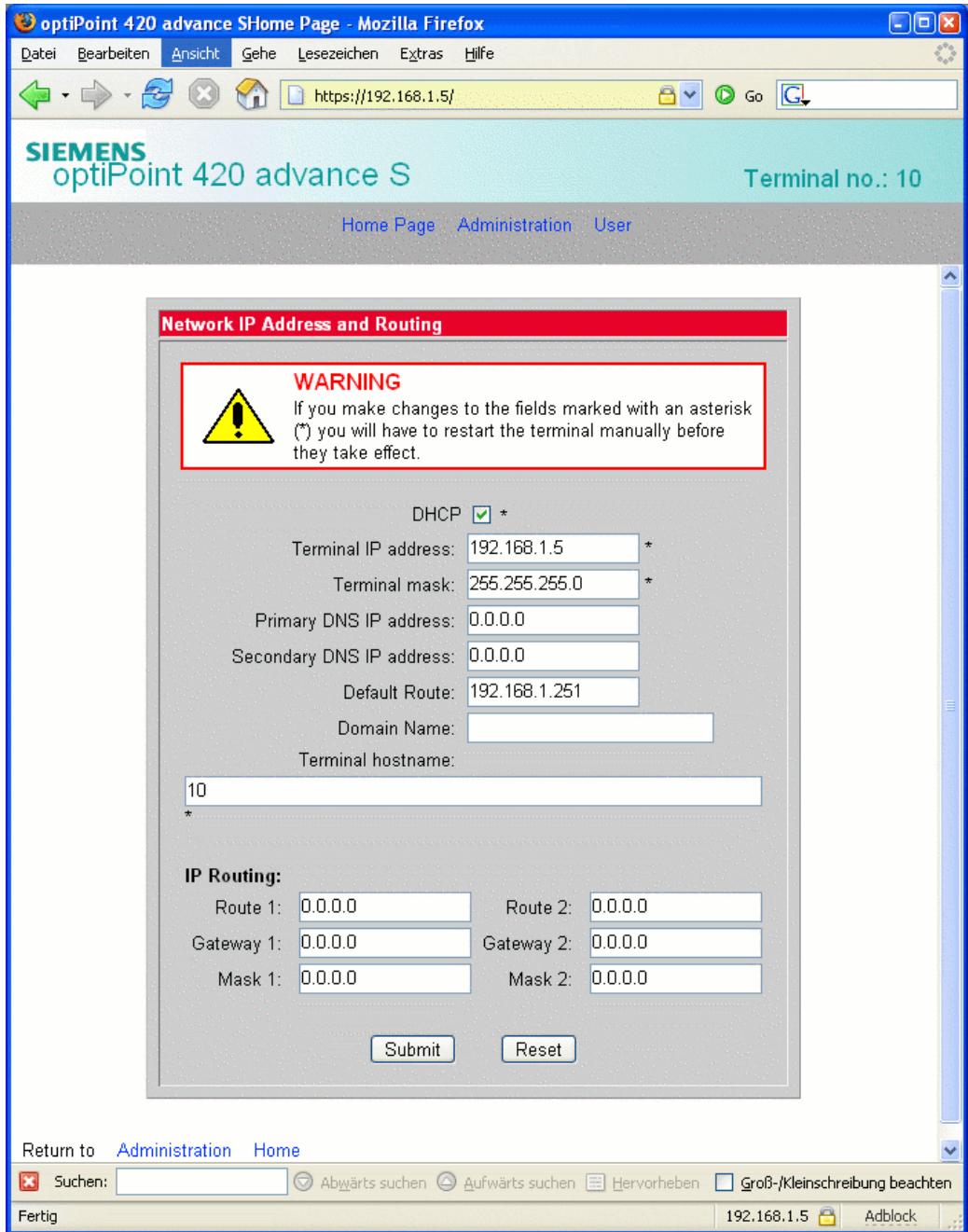
Basic Configuration of optiPoint Telephones

Network IP Address and Routing

4.2 Network IP Address and Routing

Use this section to configure network settings such as DHCP usage, IP address of the telephone, terminal mask, DNS server and default route.

For additional information about the configuration options, please refer to the Administrator Guide, see Section 9, "Reference List".



DHCP

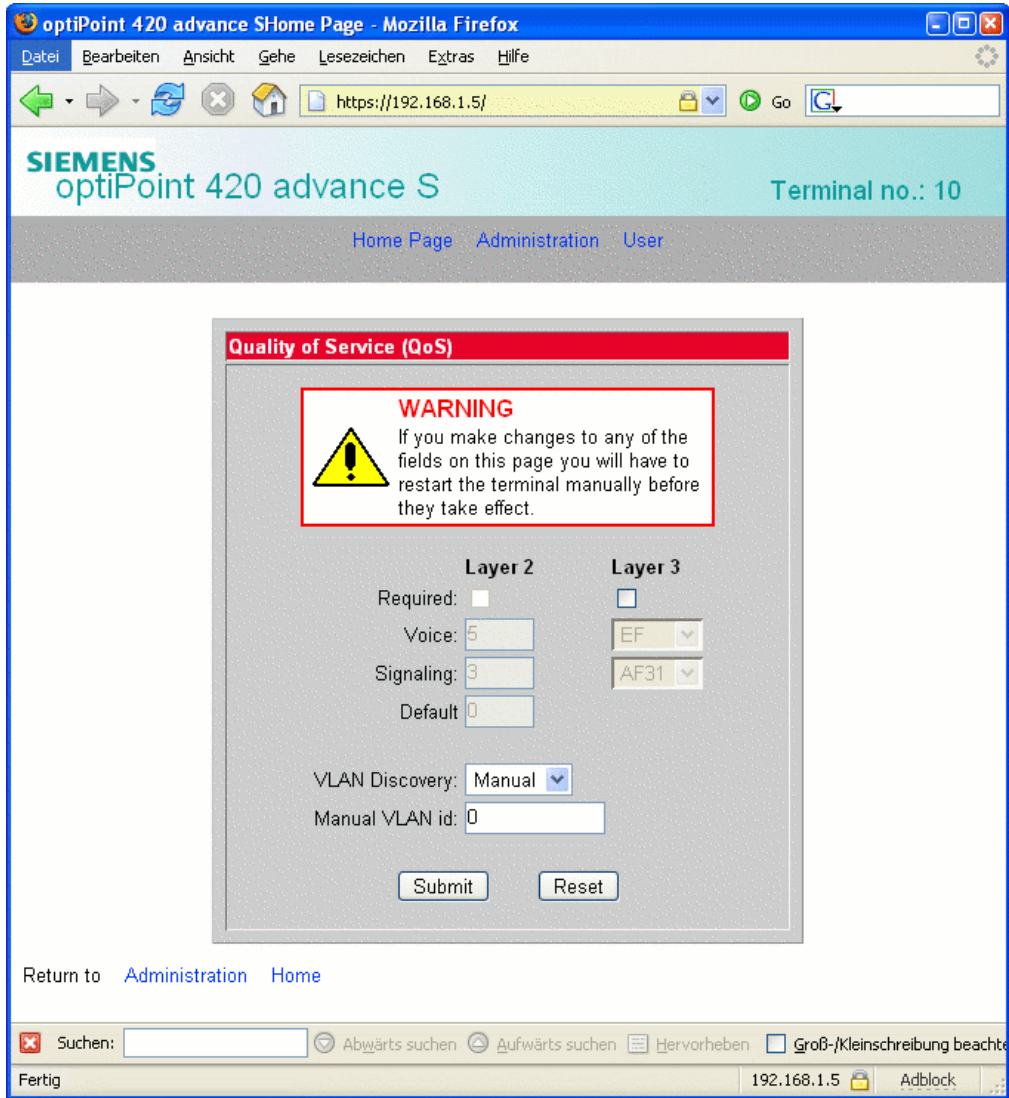
If DHCP is to be used, mark the checkbox **DHCP** (default setting).

Terminal hostname


The contents of the field **Terminal hostname** is sent to the DNS server. Thus, the web interface is also available for users know knowing the IP address of the telephone.

4.3 Quality of Service

The QoS settings ensure packet-optimized transmissions within the network.



Usually, QoS is only supported in large networks. Please deactivate **Layer 2** und **Layer 3** in low-end network that do not support QoS / Vlan.

 Older network components may not be able to read packets with Vlan tag. In these cases, **Layer 2** und **Layer 3** have to be set to "Off" using the user interface of the telephone.

The QoS settings have to be synchronized with the network administrator.

4.4 SIP Environment

Use this dialog to enter basic SIP parameters.

The screenshot shows the configuration page for a Siemens optiPoint 420 advance S terminal. The browser window title is "optiPoint 420 advance S Home Page - Mozilla Firefox". The address bar shows "https://192.168.1.5/". The page header includes "SIEMENS optiPoint 420 advance S" and "Terminal no.: 10". Navigation links for "Home Page", "Administration", and "User" are present.

Terminal details:

- Phone number: 10
- Phone name: Verleihnisk
- Register by name:
- Display ID: Verleihnisk
- Use Display ID:

SIP details:

- SIP routing: Server
- Registrar IP address or DNS name: 192.168.1.17 Port: 5060
- Server IP address or DNS name: 192.168.1.17 Port: 5060
- Gateway IP address or DNS name: 192.168.1.149 Port: 5060
- SIP port: 5060
- RTP Base port: 5004 *
- Outbound proxy:
- Default OBP domain name:
- SIP transport: UDP *
- SIP server type: Other *
- SIP session timer enabled:
- SIP session timer value: 3600 seconds
- Registration timer value: 3600 seconds
- SIP realm: usergruppe@asterisk
- SIP user ID: 10
- New SIP password:
- Confirm SIP password:

Miscellaneous:

- Message Waiting IP address or DNS name: 0.0.0.0
- Emergency number:
- Voicemail number: 99
- Branding/identity name: optiPoint

The status bar at the bottom shows "Fertig", "192.168.1.5", and "Adblock".

Basic Configuration of optiPoint Telephones

SIP Environment

Phone number

Phone number of the user agent or telephone. This number must refer to the corresponding user data section in the file *sip.conf* (see Section 3.1, "SIP Registration"). Alternatively, you can enter a name under **Phone Name**. To ensure that the telephone is registered with this name, mark the checkbox **Register by name**.

Registrar IP address or DNS name / Server IP address or DNS name

Here you have to enter one of the IP addresses or hostnames that entered as `bindaddr` in the section `[general]` of the file *sip.conf*.

SIP Port

Specifies the destination port of the server, i.e. the port the server uses to receive the telephone's SIP packets. The default value is 5060.

SIP server type

Enter the type of the SIP server. If you use Asterisk, enter "Other".

SIP realm

If the parameter `realm` is set in the section `[general]` of the file *sip.conf*, Asterisk transmits this parameter to the telephone during the first registration. Therefore, no value has to be entered here.

SIP user ID

This value complies with the term in the corresponding user-specific section in the file *sip.conf* (see Section 3.1, "SIP Registration").

New SIP password / Confirm SIP password

The SIP password has to comply with the one set using the parameter `secret` in the corresponding user-specific section (client definition) in the file *sip.conf*.



Both optiPoint telephones as well as Asterisk support the MD5 password encryption. The password can also be saved on the Asterisk server as a MD5 hash (see Section 3.1.2, "secret").

Message Waiting IP address or DNS name

IP address or hostname of the server sending the MWI notifications (Message Waiting Indicator). The presetting 0.0.0.0 should work in most cases.

Voicemail number

Specifies the number to reach the voicemail feature of Asterisk. This complies with the extension specified for the feature `VoiceMailMain()` in the file *extensions.conf*.

5 Multi Line

During setup, a phone number - or alternatively: a name - is assigned to the telephone; this number or name can be used to call this telephone and is used at the called party's side to identify the caller.

Very often this is referred to as a (telephone) "line" the telephone is assigned to - in analogy to conventional (TDM) telephones using phone numbers.

In addition, powerful telephones support the so-called "Multi Line" feature. This feature enables one end device to have access to several lines. Apart from that, one line can be assigned to several end devices. If a call comes in on such a shared line, this triggers a ringing tone on all end devices assigned to that line. Depending on the model, optiPoint 410 S / 420 S can attend to up to 10 lines.

If you want to use the Multi Line feature with Asterisk, the parameter `pedantic=yes` has to be set in the file *sip.conf*; this ensures that Asterisk can identify the Multi Line Header in the SIP messages (presetting is `no`).

The lines are not configured in the SIP environment section but using the key programming of the optiPoint telephone. For each line one function key has to be configured as "line" key. User data such as E.164 number, user name and password have to be entered. The LED belonging to the line key indicates the status of that line. For more information about functionality and setup, please refer to the User Guide and the Administrator Guide (see Section 9, "Reference List").

For each line on the end device there has to be a corresponding entry in the file *sip.conf*. The server settings have to be entered in the section SIP environment.



As Asterisk Version 1.2.7 has some problems with forking, Multi Line with several user agents (shared line) are not yet possible. So far, only one user agent can be connected per line. However, Digium is already working on improving forking support.

Voicemail

Settings in *voicemail.conf*

6 Voicemail

Voicemail can be considered an answering machine or voice-based info service integrated into Asterisk. There are two ways to notify the user of a new voice message: he receives an email with the voice message as an attachment, and the VoiceMail Message LED on his optiPoint starts blinking. The following section briefly describes the voicemail configuration of Asterisk.

6.1 Settings in *voicemail.conf*

The global section `[general]` contains - amongst other values - the audio formats for recording voice messages.

Example:

```
[general]
format=wav49|gsm|wav
```

Explanation:

- `format`

The parameter can contain a number of audio formats (of course you can also select only one file format). The voice messages are recorded in all formats listed.

To give the user access to his mailbox, his access data have to be entered into the file *voicemail.conf*.

Example:

```
[myphones]
10 => 123456,10,user1@mail.com
11 => 123456,11,user2@mail.com
```

Erläuterungen:

- 10

Number of the mailbox. In the example, this number is the same as the phone number of the user.

- 123456

The user password for access to the mailbox.

- 10
SIP user name or phone number.
- user1@mail.com
Email address of the user. If the Asterisk server is set up accordingly, the voice message can be sent to the user as an email attachment.

6.2 Settings in *extensions.conf*

In the user-specific section of the dial plan, an overflow option for voicemail has to be set up. You can differentiate between two available options: a) the called party is not available, and b) the called party's line is busy. The following example shows the configuration for the user with phone number 10; the lines have to be entered into the corresponding section of the file *extensions.conf*.

Example:

```
exten => 10,2,VoiceMail(u10@myphones)
exten => 10,102,VoiceMail(b10@myphones)
```

Explanations:

- 10
SIP user name or phone number.
- 2 / 102
Priority for forwarding to the mailbox. The value 2 in the example determines that - if the called party is unavailable - the caller is forwarded after the attempted call (priority 1 in the example) is forwarded to the mailbox `10@myphones` and hears a corresponding message. If the line is busy, Asterisk jumps to a priority number that is 101 higher than the priority of the attempted call, in the example: $1 + 101 = 102$.
- `u10@myphones` / `b10@myphones`
Further specification of the mailbox. The prefix `u` makes sure that the message designated for unavailability is played. If you use the prefix `b`, the message for "line busy" is played. `10` is the number of the mailbox. `@myphones` specifies the context for this entry. This is necessary if the user entry in the file *voicemail.conf* was entered in a specific context instead of the context `[default]`.

Weiterhin muss für den jeweiligen Kontext (im Beispiel: `myphones`) die zu wählende Nummer der Voicemail-Funktion eingetragen werden, damit der Benutzer seine Sprachnachrichten abhören kann. Auch dies geschieht in Form einer Extension, also eines Eintrags im Wählplan.

Voicemail

Settings in the file sip.conf

Example:

```
exten => 99,1,VoiceMailMain()
```

Explanations:

- 99
Phone number for the mailbox.
- 1
Priority of the VoiceMailMain feature in the dial plan.
- VoiceMailMain()
The user is forwarded to the Voicemail main menu.

6.3 Settings in the file *sip.conf*

As soon as a new voice message is available, a MWI signal (Message Waiting Indicator) is sent to the telephone which displays a corresponding message. On the optiPoint 410 S / 420 S you can easily retrieve messages using a previously configured key. A blinking LED indicates that new messages are available (see Section 6.4, "Setting up the Message Key on the optiPoint Telephone"). In the section [general] the MIME type has to be set as follows in order to ensure the MWI signal is transmitted correctly to the telephone:

```
notifymimetype=text/plain
```

Finally, the mailbox to be used has to be specified in the user-specific section of the file *sip.conf*.

Example:

```
mailbox=10@myphones
```

Explanation:

The number of the mailbox together with the context is added to the key word `mailbox`.

6.4 Setting up the Message Key on the optiPoint Telephone

In the web interface, go to Administration > Function keys... Phone to get to the function keys assignment feature.

The screenshot shows the Siemens optiPoint 420 advance S web interface. The main window displays the 'Phone function key assignments' section, which is divided into 'Normal function' and 'Shifted function' columns. Each key (1-18) has an 'Edit' button. Key 18, 'Voice Messages', is highlighted with a red box. Below this section is a 'Parameters for function 'Voice Messages' assigned to Key 18' table.

Name	Value
Key Label	Nachrichten

On the right side, a dialog box titled 'Assign a function to xx' is open. It features a dropdown menu labeled 'Select a function:' with 'Voice Messages' selected. Below the dropdown, there is a checkbox for 'Locked by administrator:' which is currently unchecked and labeled 'Off'. There is also a text input field for 'Key Label:'. At the bottom of the dialog are 'Apply' and 'Cancel' buttons.

Click the **Edit** button of the key you want to assign the message function to.

On the right side of the web interface another input mask is displayed. Select "Voice Message" from the pull-down menu **Select a function**. Using the option **Locked by administrator** you can define whether this key is to be locked. In case of an optiPoint 420 S telephone you can enter the key labeling into the field **Key Label**. Once you have entered all the data you are asked to save them. As soon as you have saved the data, the new function key is available.

Call Pickup

Group Definition

7 Call Pickup

Asterisk provides the possibility of creating Pickup Groups. Calls to a member of such a group can be picked up by all group members.

7.1 Group Definition

Groups are defined in the file *sip.conf*. Using `callgroup=X` within the section for user A, this user A is assigned to a group called X. This group pools all call targets that can be picked up by a pre-defined pickup group. To make sure a user B can pick up a call for user A, you have to enter `pickupgroup=X` into the section for user B. X can be set to values from 0 through 63.

Example:

```
[10]
callgroup=1
pickupgroup=1
```

Explanations:

- `[10]`
Indicates the user for which the following parameters are valid.
- `callgroup`
Using this parameter the user is assigned to a group within which calls can be picked up by members of a pickup group with the same number (in the example: 1).
- `pickupgroup`
Using this parameter the user is assigned to a pickup group; the members of this pickup group can pick up calls to any telephone within the group of call targets that have the same number (in the example: 1).

7.2 Execution of Call Pickup

To execute a call pickup, i.e. to pick up a call to the group, the user has to select the sequence `*8`.

The `Pickup()` command has to be defined in the file *extension.conf*.

Example:

```
exten => *_8.,1,Pickup(10&11)
```

Explanations:

- `_*8.`

Notation for the dial sequence `"*8"`.

- `1`

Priority of this step in the extension.

- `Pickup(10&11)`

If a call arrives for one of the telephones specified in the parameters, this call is picked up as soon as the sequence detailed above is dialed. The telephones listed are checked for their status in their order of appearance in the parameter list.



The optiPoint 410 S / optiPoint 420 S telephones feature a key function Call Pickup signalling a call within a pickup group via an LED. However, in an Asterisk environment this feature is inoperable as Asterisk does not send the necessary SIP notification.

8 Music on Hold (MOH)

The RFC 3261 offers several options for indicating "On Hold". In case of an optiPoint telephone, a method "INACTIVE" is added to the session description if the other party is put on hold. Asterisk, however, only recognizes "SENDONLY".

There is a workaround to make sure that Asterisk plays the MOH (Music on Hold) in spite of that. You have to modify the file *chan_sip.c* which by default is stored in the folder */usr/src/asterisk/channels*.

In line 3552 the RTP MAP is assigned.

This is the original source code:

```
sdpLineNum_iterator_init(&iterator);
while ((a = get_sdp_iterate(&iterator, req, "a"))[0] != '\0') {
    char* mimeSubtype = ast_strdupa(a); /* ensures we have enough
space */
    if (!strcasecmp(a, "sendonly")) {
        sendonly=1;
        continue;
    }
    if (!strcasecmp(a, "sendrecv")) {
        sendonly=0;
    }
    if (sscanf(a, "rtpmap: %u %[^/]/", &codec, mimeSubtype) != 2)
        continue;
    if (debug)
        ast_verbose("Found description format %s\n",
mimeSubtype);
    /* Note: should really look at the 'freq' and '#chans' params
too */
    ast_rtp_set_rtpmap_type(p->rtp, codec, "audio",
mimeSubtype);
    if (p->vrtp)
        ast_rtp_set_rtpmap_type(p->vrtp, codec, "video",
mimeSubtype);
}
```

For evaluating the INACTIVE of the optiPoint telephones, the If loop has to be adjusted in a certain way so that the INACTIVE is interpreted as SENDONLY.

This is the modified source code:

```
sdpLineNum_iterator_init(&iterator);
while ((a = get_sdp_iterate(&iterator, req, "a"))[0] != '\0') {
    char* mimeType = ast_strdupa(a); /* ensures we have enough
space */
    if (!strcasecmp(a, "sendonly")) {
        sendonly=1;
        continue;
    }

    /* condition added by VGarroy to test MOH */
    if (!strcasecmp(a, "inactive")) {
        sendonly=1;
        continue;
    }
    if (!strcasecmp(a, "sendrecv")) {
        sendonly=0;
    }
    if (sscanf(a, "rtpmap: %u %[^/]/", &codec, mimeType) != 2)
        continue;
    if (debug)
        ast_verbose("Found description format %s\n",
mimeType);
    /* Note: should really look at the 'freq' and '#chans' params
too */
    ast_rtp_set_rtpmap_type(p->rtp, codec, "audio",
mimeType);
    if (p->vrtp)
        ast_rtp_set_rtpmap_type(p->vrtp, codec, "video",
mimeType);
}
```

To compile the changes into the system, complete the following steps:

1. Go to the folder containing the source files (default: `/usr/src/asterisk`).
2. Execute the commands `make` and `make install`.

Reference List

Execution of Call Pickup

9 Reference List

This list comprises references to documentation or other sources for additional information. This list is not exhaustive.

Official Asterisk Website

News, general information, supported hardware, support and forums, bug tracking.

URL: <http://www.asterisk.org/>

Asterisk Wiki on voip-info.org

Up-to-date and comprehensive collection of information about Asterisk, e.g. articles, how-to's references.

URL: <http://www.voip-info.org/wiki/view/Asterisk>

Digium

Digium Inc., co-founded by Mark Spencer, the creator of Asterisk, is the most important developer of Asterisk software.

URL: <http://www.digium.com>

User Guides for the optiPoint 410 / 420 family of products

Description of local and system-based operating processes, explanation of user interface, etc.

URLs:

<http://www.siemens.de/hipath/> > Products, Solutions & Services > Clients & Devices > SIP Telephones > User Guides

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Administrator Guides for the optiPoint 410 / 420 family of products

Details on administrative tasks for setup and configuration.

URL: <http://www.siemens.de/hipath/> > Downloads > Administration Guides

Select "Englisch" and the product category "Phones".

10 Sample Configuration

The following section is a summary of all the sample configurations used in this Configuration Guide.

sip.conf

```
[general]
context=default
realm=usergruppe@asterisk
bindport=5060
bindaddr=0.0.0.0
allow=alaw
qualify=yes
dtmfmode=info
pedantic=yes ; fuer Multi Line
notifymime-type=text/plain
```

```
[10]
type=friend
context=myphones
username=10
secret=123456
host=dynamic
mailbox=10@myphones
callgroup=1
pickupgroup=1
```

```
[11]
type=friend
context=myphones
username=11
secret=123456
host=dynamic
mailbox=11@myphones
callgroup=1
pickupgroup=1
```

Sample Configuration

Execution of Call Pickup

extensions.conf

```
[myphones]
exten => 10,1,Dial(SIP/10,5,r)
exten => 10,2,VoiceMail(u10@myphones)
exten => 10,102,VoiceMail(b10@myphones)

exten => 11,1,Dial(SIP/11,5,r)
exten => 11,2,VoiceMail(u11@myphones)
exten => 11,102,VoiceMail(b11@myphones)

exten => 99,1,VoiceMailMain()

exten => *_8.,1,Pickup(10&11)
```

voicemail.conf

```
[general]
format=wav49|gsm|wav
```

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

An obligation to provide the respective characteristics shall only exist if expressly agreed in the terms of contract.

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