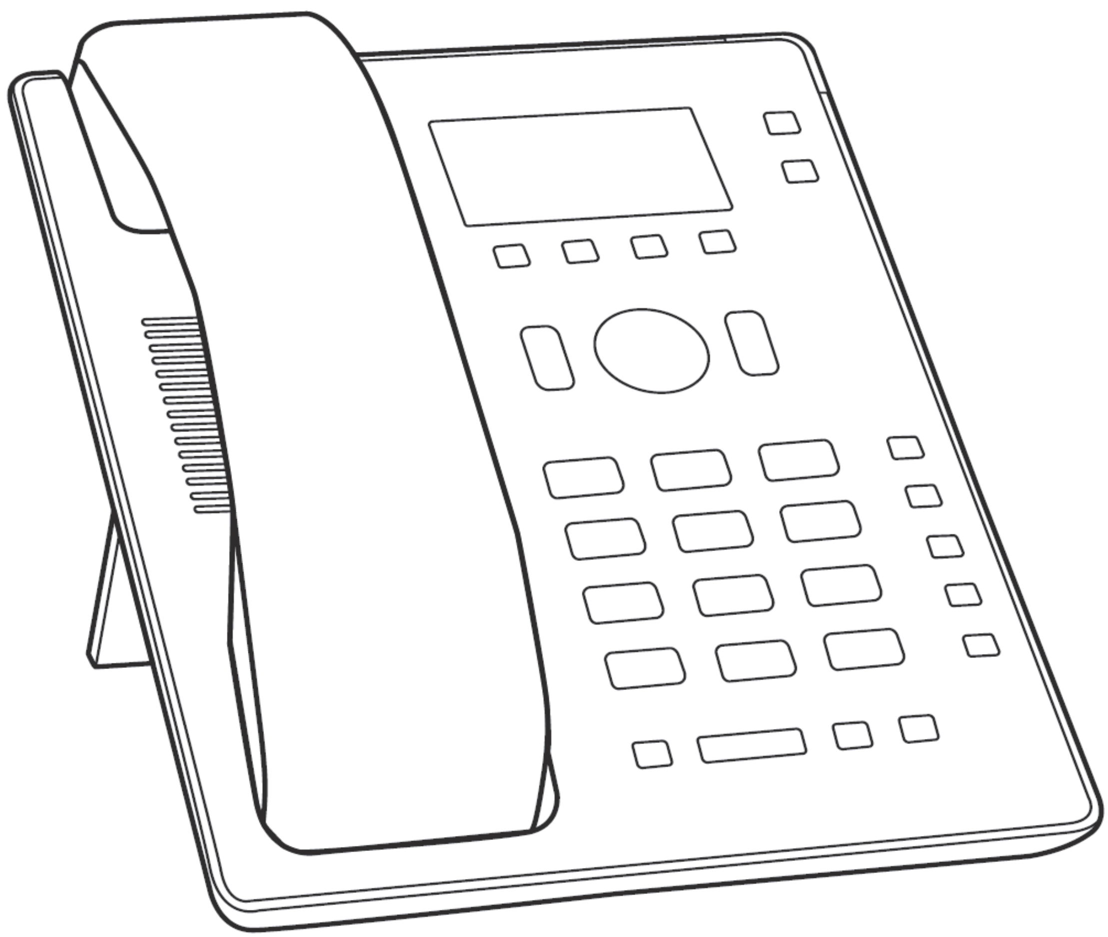
EriesTerminal® SIP Deskset

ET605

Administrator and Provisioning Manual



Rev. 1 07/18

CONTENTS

PREFACE

Congratulations on your purchase of this VTech product. Please thoroughly read this manual for all the feature operations and troubleshooting information necessary to install and operate your new VTech product. You can also visit our website at *businessphones.vtech.com* or call **1 (888) 370-2006**.

This administrator and provisioning manual contains detailed instructions for installing and configuring your ET605 SIP Deskset with software version 8.10.1.x. See *"System Info" on page 73* for instructions on checking the software version on the ET605. Please read this manual before installing the product.

Please print this page and record the following information regarding your product:

Model number: ET605

Type: Small to medium business SIP-endpoint deskset

Serial number: ______________________________

Purchase date: ______________________________

Place of purchase: ___________________________

Both the model and serial numbers of your VTech product can be found on the bottom of the console.

Save your sales receipt and original packaging in case it is necessary to return your telephone for warranty service.

Text Conventions

Table 1 lists text formats and describes how they are used in this guide.

Table 1. Description of Text Conventions

Text Format	Description
Screen	Identifies text that appears on a device screen or a WebUI page in a title, menu, or prompt.
HARD KEY or **DIAL-PAD KEY**	Identifies a hard key, including the dial-pad keys.
CallFwd	Identifies a soft key.
NOTE Notes provide important information about a feature or procedure.	Example of a Note.
CAUTION *A caution means that loss of data or unintended circumstances may result.*	Example of a Caution.

Audience

This guide is written for installers and system administrators. It assumes that you are familiar with networks and VoIP, both in theory and in practice. This guide also assumes that you have ordered your IP PBX equipment or service and selected which PBX features you want to implement. This guide references specific IP PBX equipment or services only for features or settings that have been designed for a specific service. Please consult your equipment supplier or service provider for recommended switches, routers, and firewall and NAT traversal settings, and so on.

As the ET605 SIP Deskset becomes certified for IP PBX equipment or services, VTech may publish interop guides for those specific services. The interop guides will recommend second-party devices and settings, along with ET605-specific configurations for optimal performance with those services. For the latest updates, visit our website at *businessphones.vtech.com*.

Related Documents

The ***ET605 Quick Start Guide*** contains a quick reference guide to the ET605 external features and brief instructions on connecting the ET605 to a working IP PBX system.

The ***ET605 User Guide*** contains a quick reference guide, full installation instructions, instructions for making and receiving calls, and a guide to all user-configurable settings.

The documents are available from our website at *businessphones.vtech.com*.

CHAPTER 1

INTRODUCING THE ET605

This administrator and provisioning guide contains detailed instructions for configuring the ET605 SIP Deskset. Please read this guide before attempting to configure the ET605.

This chapter covers:

About the ET605 Deskset

The VTech ET605 SIP Deskset is a business phone designed to work with popular SIP telephone (IP PBX) equipment and services. Once you have ordered and configured your SIP equipment or service, the ET605 enables you to make and receive calls as you would with any other business phone.

The ET605 Deskset features include:

- Support for 2 SIP lines/accounts
- Dual Ethernet ports, 10/100 Mbps
- Power over Ethernet (PoE) support (AC adapter optional)
- 132 x 64 pixel (w x h) mono LCD display, providing five lines of information
- 4 configurable soft keys
- 2 programmable feature keys with green LEDs
- 2-way navigational pad
- Zero touch provisioning
- RJ9 headset port
- Sensor hook switch
- Full-duplex base speakerphone
- Message waiting LED indicator
- Local phonebook up to 100 entries
- Call history up to 100 entries

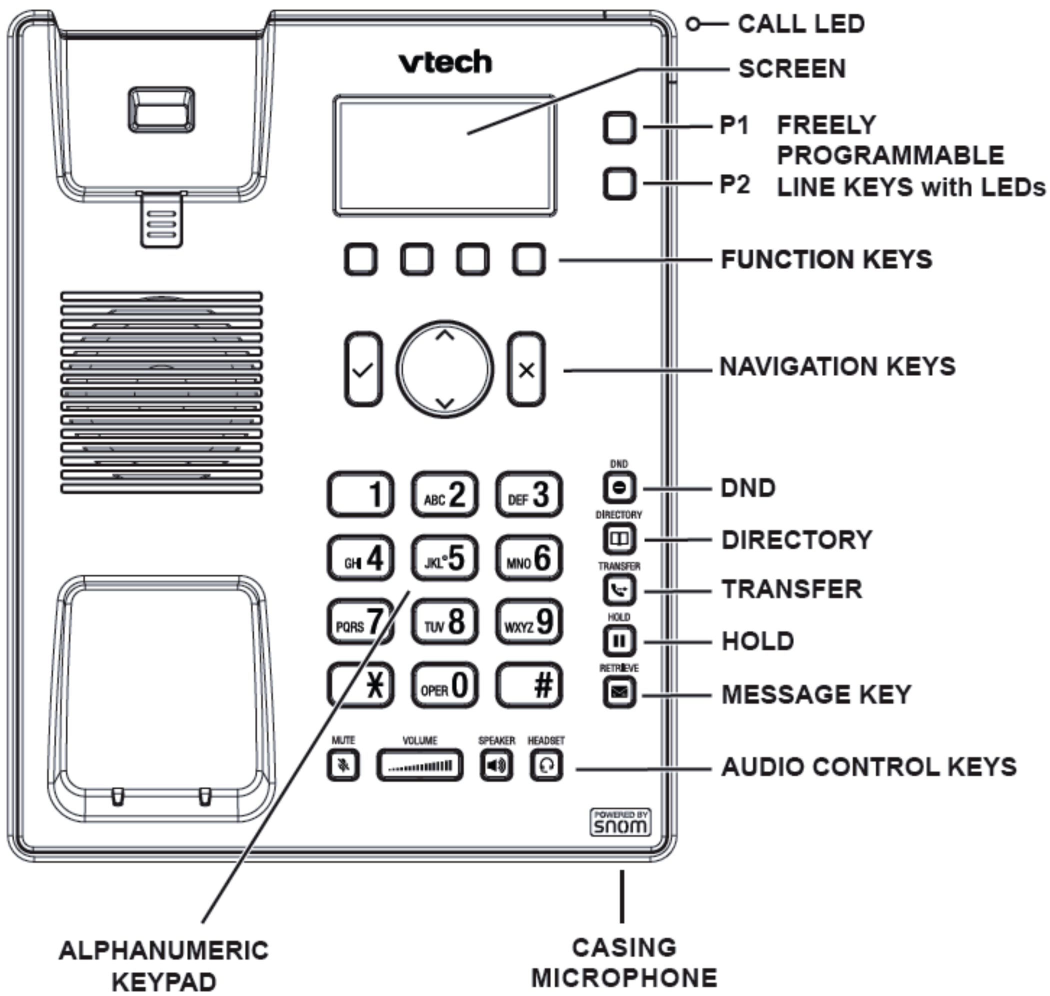
There are two network ports, known as the Ethernet port and PC port, at the back of the ET605. The Ethernet port allows the ET605 Deskset to connect to the IP PBX. The PC port is for another device such as a personal computer to connect to the Ethernet network through the ET605.

You can configure the ET605 using the menus on the phone, a browser-based interface called the WebUI, or an automatic provisioning process (see *"Auto Provisioning" on page 14*). The WebUI enables you to configure the ET605 using a computer that is connected to the same Local Area Network. The WebUI resides on the ET605, and may get updated with firmware updates.

The ET605 SIP Deskset supports intercom and call transfers between system extensions and can connect you and two other parties on the same conference call. The ET605 has four programmable soft keys and 2 programmable line keys. You can program these keys for quick dial, busy lamp field, line access or any of the functions described in *"Function Keys page" on page 91*.

Quick Reference

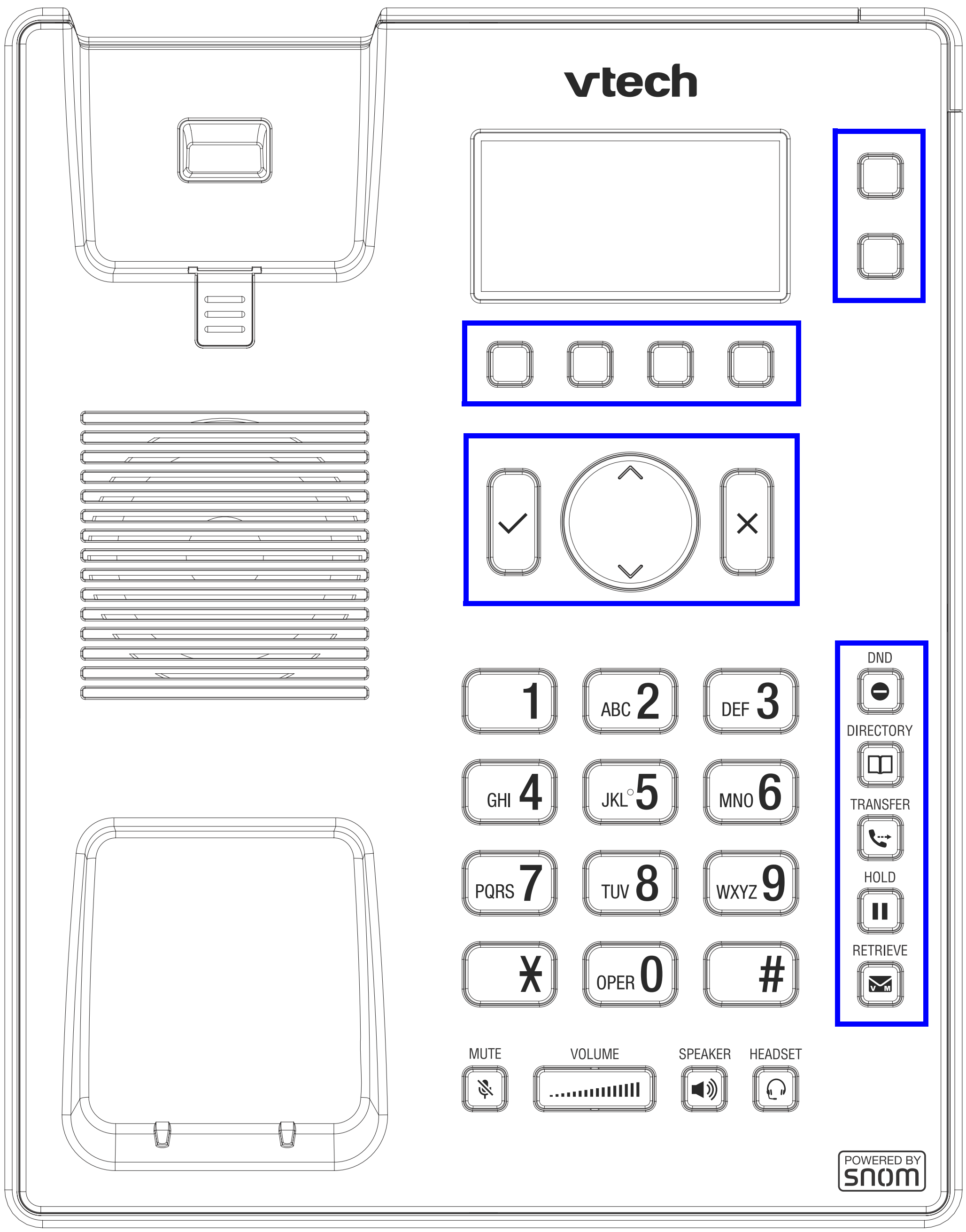
The following diagram shows the ET605 external features and controls.



Programmable Keys

You can use the WebUI to change the function of the four soft keys below the display, two programmable LED line keys to the right of the display, navigation keys, and the customizable function keys.

For more information, see *"Function Keys page" on page 91*.



Configuration Methods

You can use any of the following methods to configure your ET605 SIP Deskset:

- **Provisioning** – see *"Provisioning" on page 13*.
- **Phone User Interface** – see *"Phone Menu Reference" on page 53*.
- **Web User Interface (WebUI)** – see *"Web User Interface (WebUI) Reference" on page 75*.

CHAPTER 2

PROVISIONING

Provisioning refers to the process of acquiring and applying new settings for the ET605 using configuration files retrieved from a remote computer. After a ET605 is deployed, subsequent provisioning can update the ET605 with new settings; for example, if your service provider releases new features.

With automatic provisioning, you enable the ET605 to get its settings automatically—the process occurs in the background as part of routine system operation. Automatic provisioning can apply to multiple devices simultaneously.

With Manual Software Update on the WebUI, you update the ET605 settings (configuration and/or firmware) yourself via **Setup** > **Software Update**.

This chapter covers

- *"Auto Provisioning" on page 14*
- *"Manual Software Update" on page 51*

Auto Provisioning

Auto Provisioning (Mass deployment) enables remote administration (configuration and maintenance) of the ET605 deskset.

Auto Provisioning is particularly useful for out-of-the-box scenarios in larger phone installations.

Auto Provisioning can be used to provide general and specific configuration parameters (Settings) to the phones and to manage firmware actualization.

Requirements

Auto Provisioning requires a central setting (or provisioning) server.The Auto Provisioning Server stores the Auto Provisioning Configuration Files and provides them on request to the phones. Firmware images may also be stored here.

The following setting server types/protocols can be used for provisioning of configuration parameters and firmware images: TFTP Server, HTTP Server, and HTTPS Server.

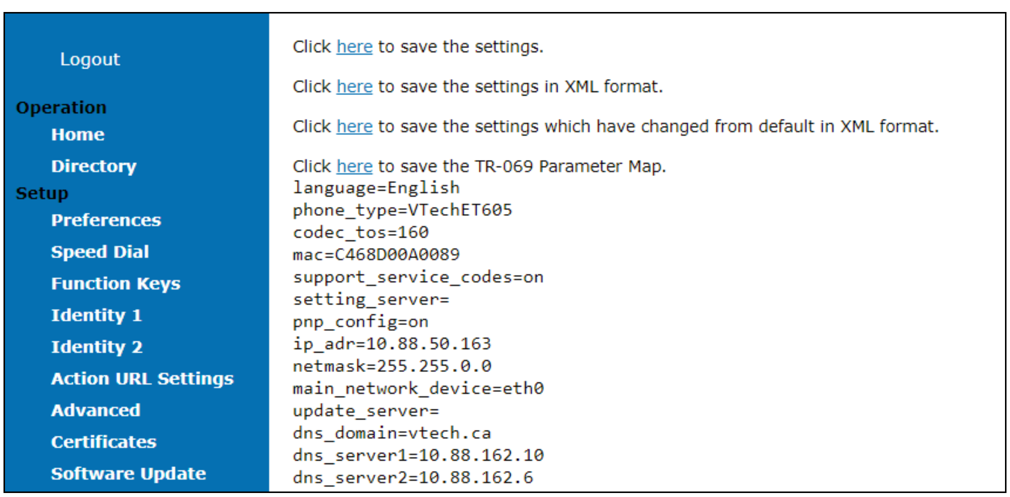
Selected Configuration parameters can be stored in configuration files (phone type/MAC address based) or can be created on request by means of script files (MAC address based). See *"Configuration File Types" on page 23*. The location of these files is defined in the parameter **setting_server**.

Please check the Bootup Process in order to select the appropriate auto provisioning method. See *"Bootup Process" on page 22*.

Saving Configuration Files

You can save a sample configuration file from your phone using the WebUI interface.

1. Open the ET605 WebUI interface, and open the **Settings** page.



2. To save the settings, click the link for the file format you want.
   The first link will save the settings in ASCII format.

NOTE: VTech recommends that you only work with XML format when saving configuration files.

You can now make copies of the settings file, and edit them as required for auto provisioning.

Scenarios

Depending on the installation environment, the following scenarios can be applied to provide the setting (provisioning) URL to the phones:

1. **DHCP Option 66/67** - see *page 15*.

   The DHCP Server in the LAN may send the provisioning URL via Option 66/67.

2. **Plug & Play** - see *page 18*.

   Any SIP Server in the LAN may send the provisioning URL by replying to SIP SUBSCRIBE Broadcast messages.

3. **Automatic Redirection Service** - see page *page 19*.

   VTech's public provisioning server will be contacted automatically and may redirect MAC address based provisioning requests to any other server.

4. **TR-069 Provisioning** - see page *page 20*.

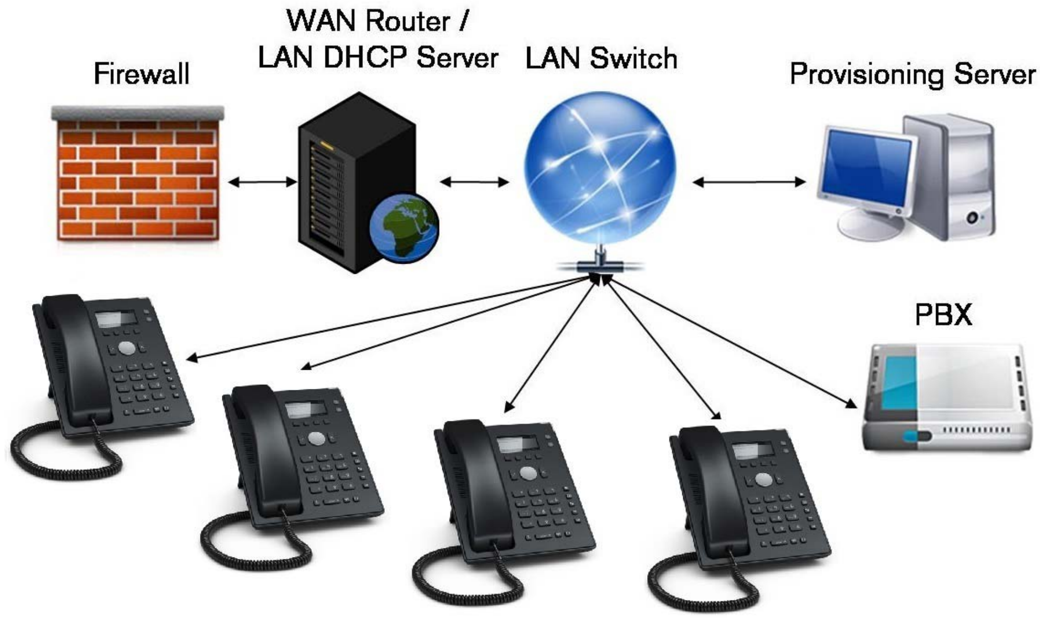
   Either scenario 1/2/3 can be used to enable the phone for TR-69 Provisioning.

DHCP Option 66/67

This configuration method requires the following components:

- DHCP Server
  ONE DHCP Server per LAN supporting DHCP Option(s) 66 or/and 67.
  See *"DHCP Options" on page 16*.
- Auto Provisioning Server
- Configuration files
  See *"Configuration File Types" on page 23*.
- VTech VOIP Phone Configuration

The DHCP Server must be configured with additional DHCP Options containing the URL of the Auto Provisioning Server to the VTech VoIP phones on boot-up. The phones will then request their configuration parameters from the Auto Provisioning Server which will result in a ready-to-use phone setup without manual configuration.



DHCP Options

Option 66 (TFTP server name)

This option is used to identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options. The code for this option is 66, and its minimum length is 1.

VALIDVALUE

<protocol> : // <IP address> or <domain> e.g. http://10.0.0.2, https://provisioning.company.com

<IP address> or <domain> e.g. 10.0.0.2, provisioning.company.com

where <protocol> = server type/protocol

where <IP address> = server IP address

where <domain> = server domain name

NOTE: Without specifying the <protocol> the firmware will attempt all supported server protocol types consecutively:

1. tftp://...
2. http://...
3. https://...

Configuration Parameter: update_server

Option 67 (Bootfile name)

This option is used to identify a bootfile when the 'file' field in the DHCP header has been used for DHCP options. The code for this option is 67, and its minimum length is 1.

VALIDVALUE

<path> e.g. settingfiles/vtech/VTechET605.cfg, settingfiles/vtech/ VTechET605.htm, settingfiles/vtech/VTechET605.xml

<empty> or <not used>

where <path> = path to the location of the setting file/script file

NOTE: If this option is empty or not specified at all the firmware automatically requests the following setting files, except the whole URL is encoded in option 66:

all ET605 phones request --> http://<domain>/VTechET605.htm

Configuration Parameter: update_filename

Option 43 (vendor-encapsulated-options)

Encapsulated Option 66, Option 67, Option 132, and Option 133 are supported.

Encapsulated DHCP options, for encoding see RFC 2132 Section 2. DHCP Option Field Format; One can tunnel vendor specific DHCP options depending on the vendor-id (option 60) send before from the phone to the DHCP server. Vendor specific DHCP options may be provided encapsulated in option 43, see RFC 2132 Section 8.4. Vendor Specific Information. Values of options like 66/67/132/133, which are tunneled via option 43, take precedence over direct options 66/67/132/133.

VALIDVALUE (Examples)

linux dhcpd3 syntax:

```
option vendor-encapsulated-options
42:0c:68:74:74:70:3a:2f:2f:74:65:73:74:00:43:12:73:6e:6f:6d:2f:73:65:74:74:6
9:6e:67:73:2e:70:68:70:00;
```

Which means tunnel opt 66 http://test and opt 67 vtech/settings.php via opt 43.

```
option vendor-encapsulated-options
84:02:33:00;
```

Which means tunnel opt 132 value 3 via opt 43.

```
option vendor-encapsulated-options
84:04:31:31:34:00:85:02:35:00;
```

Which means tunnel opt 132 value 114 and opt 133 value 5 via opt 43.

Option 60 (Vendor class identifier)

This option is used by DHCP clients to optionally identify the vendor type and configuration of a DHCP client. The information is a string of n octets, interpreted by servers. Vendors may choose to define specific vendor class identifiers to convey particular configuration or other identification information about a client. For example, the identifier may encode the client's hardware configuration. Servers not equipped to interpret the class-specific information sent by a client MUST ignore it (although it may be reported). Servers that respond SHOULD only use option 43 to return the vendor-specific information to the client. The code for this option is 60, and its minimum length is 1.

The phone sends its type (i.e. VTechET605) via this option to the DHCP server.

NOTE: Vendor class identifier for VTech ET605: VTechET605

Plug & Play

Plug & Play (PnP) provides a proprietory method to enable Auto Provisioning on all VTech VoIP phones. By default (Parameter pnp config = on) the phones send SIP SUBSCRIBES messages to a multicast address. Any SIP server understanding that message may reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration from.

SIP Flow

ET605 phones send on boot-up a SIP SUBSCRIBE message to a multicast address:

```
Sent to udp:224.0.1.75:5060 at 24/12/2001 00:00:19:248 (448 bytes):

SUBSCRIBE sip:MAC%3a00135E874B49@intern.vtech.ca SIP/2.0
Via: SIP/2.0/UDP 192.168.10.67:5060;rport
From: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=658512961
To: <sip:MAC%3a00135E874B49@intern.vtech.ca>
Call-ID: 1930770594@192.168.10.67
CSeq: 1 SUBSCRIBE
Event: ua-profile;profile-type=device;vendor=OEM;model=OEM;version=7.1.19
Expires: 0
Accept: application/url
Contact: <sip:192.168.10.67:5060>
Content-Length: 0
```

If any SIP application within one hop range understands this message a confirmation is sent:

```
Received from udp:192.168.100.10:5060 at 24/12/2001 00:00:19:287 (480 bytes):

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.10.67:5060;rport=5060
Record-Route: <sip:127.0.0.1;lr;transport=tcp;route-id=fb4fb92b7775c2a7>
Record-Route:
<sip:192.168.100.10;lr;transport=UDP;route-id=fb4fb92b7775c2a7>
Contact: <sip:192.168.100.10;transport=TCP;handler=dum>
To: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=91955270
From: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=658512961
Call-ID: 1930770594@192.168.10.67
CSeq: 1 SUBSCRIBE
Expires: 0
Content-Length: 0
```

followed by a SIP NOTIFY message containing the Auto Provisioning URL

http://192.168.100.10/sipphone/sipphoneconfig.xml?mac={mac}:

```
Received from udp:192.168.100.10:5060 at 24/12/2001 00:00:19:293 (868 bytes):
```

```
NOTIFY sip:192.168.10.67:5060 SIP/2.0
Via: SIP/2.0/UDP
192.168.100.10:5060;branch=z9hG4bK-d8754z-c3ea5f0e74462613-1---d8754z-;rport
Via: SIP/2.0/TCP
127.0.0.1:5060;branch=z9hG4bK-d8754z-7ca96c30144f3e04-1---d8754z-;rport=4091
6
Max-Forwards: 20
Record-Route: <sip:192.168.100.10;lr;route-id=e3470eb400e9c0a4>
Record-Route: <sip:127.0.0.1;lr;transport=TCP;route-id=e3470eb400e9c0a4>
Contact: <sip:192.168.100.10;transport=TCP;handler=dum>
To: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=658512961
From: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=91955270
Call-ID: 1930770594@192.168.10.67
CSeq: 3 NOTIFY
Content-Type: application/url
Subscription-State: terminated;reason=timeout
Event: ua-profile;profile-type=device;vendor=OEM;model=OEM;version=7.1.19
Content-Length: 59
```

`http://192.168.100.10/sipphone/sipphoneconfig.xml?mac={mac}`

The phone accepts this message and confirms:

```
Sent to udp:192.168.100.10:5060 at 24/12/2001 00:00:19:315 (542 bytes):
```

```
SIP/2.0 200 Ok
Via: SIP/2.0/UDP
192.168.100.10:5060;branch=z9hG4bK-d8754z-c3ea5f0e74462613-1---d8754z-;rport
=5060
Via: SIP/2.0/TCP
127.0.0.1:5060;branch=z9hG4bK-d8754z-7ca96c30144f3e04-1---d8754z-;rport=4091
6
Record-Route: <sip:192.168.100.10;lr;route-id=e3470eb400e9c0a4>
Record-Route: <sip:127.0.0.1;lr;transport=TCP;route-id=e3470eb400e9c0a4>
From: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=91955270
To: <sip:MAC%3a00135E874B49@intern.vtech.ca>;tag=658512961
Call-ID: 1930770594@192.168.10.67
CSeq: 3 NOTIFY
Content-Length: 0
```

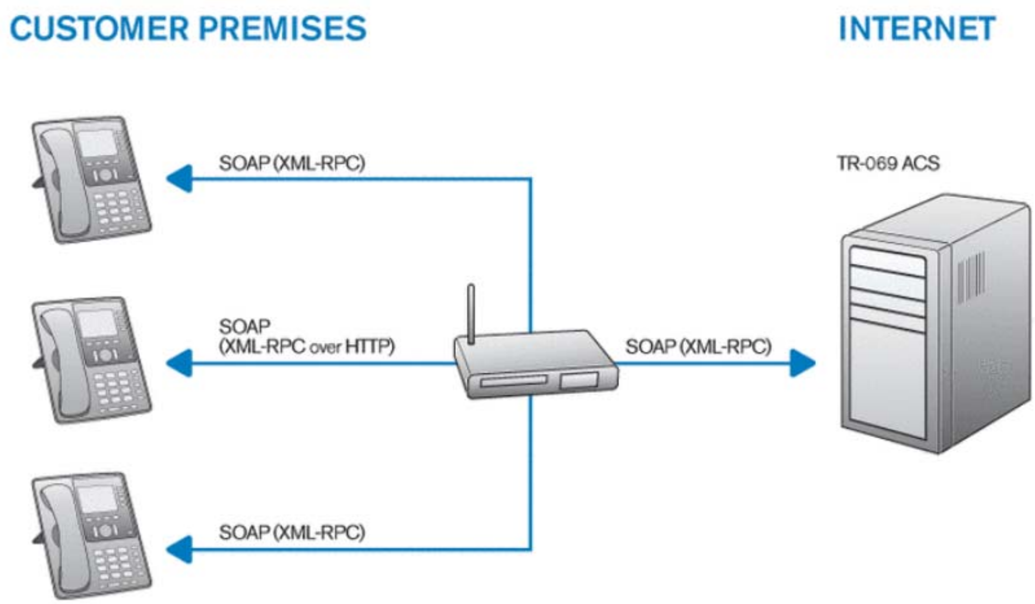
Automatic Redirection Service

This redirection service enables customers to register/list/unregister the MAC addresses of their VTech VoIP phones on VTech's Redirect Server and assign a redirection URL pointing to their own Auto Provisioning Server. Any ET605 updated to the latest firmware release will have the Redirection Server URL available as the default Provisioning Server URL

NOTE: Using the Redirection Service requires contacting the VTech support team for an account.

TR-069 Provisioning

TR-069 is a standard for remote management of CPE (Customer Premises Equipment) defined by the DSL Broadband Forum. TR-069 uses common transport mechanisms (HTTP and HTTPS) for communication with CPE. The HTTP(S) messages contain XML-RPC methods defined in the standard for configuration and management of the device.



Typically, one router on customer premises provides Internet connectivity to many phones as indicated in the above diagram. The ACS can now manage the router and all phones located behind it remotely.

What does remote management mean?

Where provisioning was used to provide configuration information to many phones at once, remote management takes this one step further. Of course, it is still possible to configure the phone remotely when it boots up, but with TR-069 the phone can actually be managed remotely.

In addition to the configuration you can also, for example:

- Reboot the phone
- Customize the phone look and feel
- Push XML-Minibrowser pages to the phone
- Update the firmware of the device

Another difference is the way the actions are triggered. Unlike provisioning, where the phone triggers the provisioning process according to a fixed schedule, TR-069 allows the administrator to initiate provisioning via ACS at anytime. Another major difference is that in case of TR-069 the server can be notified whenever a user changes a setting. This enables the administrator to correct possible mistakes right away.

TR-069 specific phone settings

ACS settings are the settings specific to the ACS connection and need to be adjusted to the specific environment. The following table describes the ACS settings with their data types and default values.

Setting name	Valid Values	Default	Description
tr69_acs_url	URLs (STRING)	empty	URL of the TR-069 ACS. This is the URL the phone will send TR-069 messages to. Please contact your ACS vendor to find out about this URL.
tr69_acs_url	URLs (STRING)	empty	URL of the TR-069 ACS. This is the URL the phone will send TR-069 messages to. Please contact your ACS vendor to find out about this URL.
tr69_acs_user	STRING	empty	Username for HTTP authentication against the ACS
tr69_acs_passwd	STRING	empty	Password for HTTP authentication against the ACS
tr69_use_acs	BOOLEAN (off, 0, on, 1)	off	Turn TR-069 management on and off.
tr69_bootstrap	BOOLEAN (off, 0, on, 1)	on	Send BOOTSTRAP event in the Inform Message. Needs to be set to on when a new ACS is contacted.
tr69_cnr_user	STRING	empt	Username to authenticate incoming connection requests.

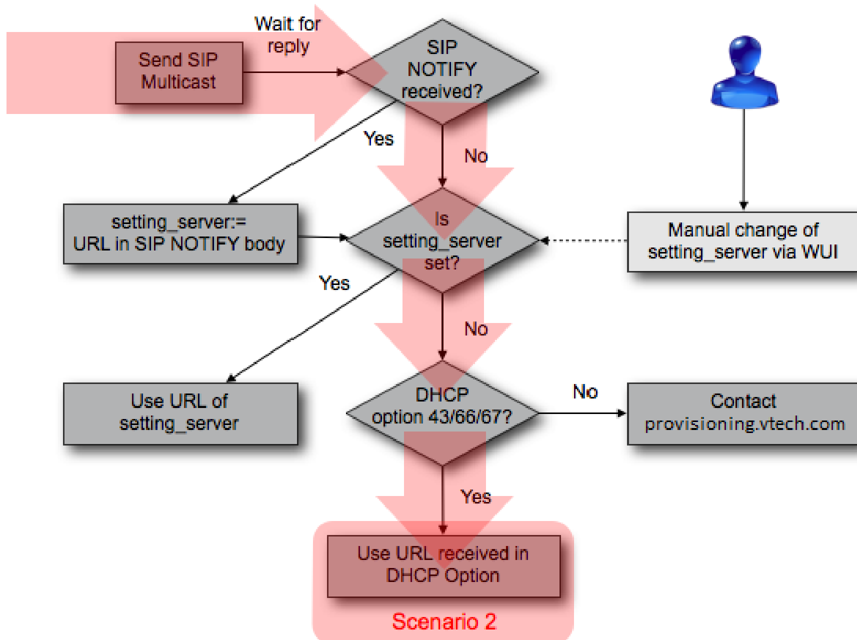
Internal settings (tr69_events, tr69_params, download_status) are used internally to control the TR-069 stack and should not be modified manually.

WARNING: modifying the internal settings manually may result in unexpected phone behavior.

Bootup Process

1. The firmware loads the configuration parameters (Settings) stored on the phone's flash memory (either factory defaults or previously changed).
2. The firmware performs a check if the Provisioning URL (parameter setting_server) has been changed manually.
   - YES: The given Provisioning URL (parameter setting_server) will be requested.
   - NO: see next step.
3. If the DHCP parameter is enabled the firmware performs a check whether the supported DHCP options have been received in the DHCP offer:
   - YES:
     - The value found in Option 66 will be stored in parameter update_server, e.g. http://server
     - The value found in Option 67 will be stored in parameter update_filename, e.g. vtech/vtech.xml
       - Initially the Provisioning URL will be composed using update_server and update_filename and will be requested,
         e.g. http://server/vtech/vtech.xml. If Option 67 is absent, the Provisioning URL is composed using update_server and {phoneType}.htm, e.g. http://server/VTechET605.htm
       - In a second attempt the MAC address, i.e.-{mac}, will be concatenated and the resulting Provisioning URL will be requested,
         e.g. http://server/vtech/vtech-0011A0YXXXX.xml or http://server/VTechET605-0011A0YXXXX.htm respectively.
   - NO: see next step.
4. Since the pnp_config parameter is enabled by default, the phone will send a SIP SUBSCRIBE message to the multicast address 224.0.1.75:5060. The firmware waits for a limited time whether a SIP NOTIFY reply is received with the Provisioning URL in the body, e.g. http://server/vtech/vtech.xml
   - YES:
     - Initially the Provisioning URL found in the body will be requested, e.g. http://server/vtech/vtech.xml
     - In a second attempt the MAC address, i.e.-{mac}, will be concatenated and the resulting Provisioning URL will be requested,
       e.g. http://server/vtech/vtech-0011A0YXXXX.xml
   - NO: see next step.
5. If none of the above steps could be applied the firmware requests the factory default Provisioning URL: http://provisioning.vtech.com/vtechXXX/vtechXXX.php?mac={mac}

6. **IMPORTANT NOTE:** If the parameter **tr69_use_acs** is enabled and will be delivered back by any of the provisioning methods, the URL of the TR-069 ACS will be requested immediately.



Configuration File Types

Setting files are container for a subset of configuration parameters needed to customize and maintain the ET605 phone remotely.

Depending on the firmware version currently installed on the ET605 phone, two formats can be distinguished:

- ASCII text format (restrictions apply)
- XML format

The following hints apply to both ASCII Text Format and XML Format.

Hints	ASCII Text Format AND XML Format
Start	1. Start with a factory reset phone ■ Apply the desired modifications in your **working (live) phone environment** first. ■ Observe the **stability and performance** of the applied changes. 2. Do **NOT** use the complete parameter list as starting point, instead: ■ Delete or uncomment **unused** configuration parameters from the complete parameter list. ■ Specify only those parameters you really want to change --> Check the meaning of each parameter before usage. ■ Finally your setting file may contain only a **few parameters**.
Flags	1. Do **NOT** use read-only flags at the beginning. They can be added at the end in order to protect certain parameters to be notified by the user! 2. Inside firmware setting files do **NOT** use any flags at all.
Network/System Settings	1. Do **NOT** provide network settings when using DHCP. 2. Do **NOT** specify setting_server unless a redirection to a different setting server is desired. ■ Remember the phone has already obtained the setting file correctly - **repeated usage of the same setting server can have unpredictable side effects and is NOT recommended.**
Firmware Setting Files	1. Do NOT specify neither **bootloader** nor **firmware** inside setting files: ■ In order to perform automated firmware updates specify a firmware setting file URL inside firmware status which points to the firmware setting file containing the firmware image URL. 2. Inside **firmware setting files** use **ONLY** the configuration parameters **bootloader** or **firmware.**

ASCII Format

ASCII format provides limited provisioning support:

- NO multiple language support. Only english phone user/web user interface languages are pre-installed.
- NO script dialplan
- NO support of formerly used internal directory entries

  [Name (tn), Number (tu), Contact Type (tc), Outgoing Identity (to)]

Structure

1. One general setting file per phone type, i.e. ET605, containing general configuration parameters

2. One specific setting file per phone, i.e. (**MAC address based**), containing phone specific configuration parameters.

3. One firmware setting file per phone type OR phone containing firmware related configuration parameters in order to perform automated firmware updates.

Hints

- Lines may end with **newline** or **carriage return/newline** pairs
- Comments start with **#** or **<**
- The **<** and **>** characters allow easy integration of **HTML tags**
- Names may consist of the characters **a-z, A-Z, 0-9** and **_**.

Flags

Parameter names can be followed by one specific character called **flag**:

- A parameter followed by **!** can be changed by the user. However the parameter value will only be stored if that parameter has not been configured yet. Only parameters followed by **$** can be overwritten, DO NOT use ! in that case.
- A parameter followed by **&**
  (or no flag) becomes write-protected (read only)
- A parameter followed by **$** can be changed but will be overwritten on reboot.
  $ will appear on the Settings page as **!**

General Setting File

General (phone type specific) setting files are requested from the setting server at first

example naming scheme: http://provisioning.mycompany.com/VTechET605.htm

in this case the general setting file was placed in the HTTP server root and will be requested automatically by any ET605 --> necessary in mixed phone type environments

```
<html>
<pre>
```

```
# example VTech general setting file
# After each setting (before the colon) you can set a flag
# General language and time configuration parameter
language$: English
web_language$: English
timezone$: USA-5
date_us_format&: on
time_24_format&: off
</pre>
</html>
```

Specific Setting File

The Phone specific setting file is requested from the setting server right after the general setting file by appending

"-MAC address" (dash+phone's MAC address)

to the general setting filename:

http://provisioning.mycompany.com/VTechET605.htm) -->
http://provisioning.mycompany.com/VTechnABLE 2.8.1 User Guide/VTechET605-000413241111.htm

```
<html>
<pre>
# example VTech specific setting file
# After each setting (before the colon) you can set a flag

user_pname1$: AUTHUSER1
user_pass1$: AUTHPASSWORD1
user_name1$: LINEPORT1
user_realname1$: User1
user_host1$: proxy.net
user_srtp1$: off
user_dp_str1$: !([^#]%2b)#!sip:\1@\d!d

user_pname2$: AUTHUSER2
user_pass2$: AUTHPASSWORD2
user_name2$: LINEPORT2
user_realname2$: User2
user_host2$: proxy.net
user_srtp2$: off
user_dp_str2$: !([^#]%2b)#!sip:\1@\d!d

# You may add up to 2 ET605 accounts
```

```
# set 1st account to active outgoing identity
active_line$: 1


# the following parameters are only required to provide automated firmware
updates
# IMPORTANT: define the URL of the --> firmware setting file
firmware_status: http://provisioning.mycompany.com/VTechET605/firmware.htm
# additionally the --> update policy may be defined
update_policy: auto_update
# additionally the --> firmware update interval may be defined
firmware_interval: 2880


</pre>
</html>
```

Firmware Setting File

The firmware setting file is requested if the firmware_status URL has been defined either in the general or --> specific setting file

example name: http://provisioning.mycompany.com/VTechET605/firmware.htm

```
<html>
<pre>

# example VTech firmware setting file

# Firmware setting specifies the URL of the firmware/root fs/linux image file
firmware:
http://provisioning.mycompany.com/firmware/VTechET605/VTechET605-X.X.bin

# Bootloader setting MUST NOT be used from Version 5.0 onwards
# bootloader:

</pre>
</html>
```

The firmware compares the URL (string) obtained from the firmware parameter with the last firmware image URL successfully loaded by the phone --> if both strings are different the provided firmware image URL is requested from the setting server otherwise no firmware will be loaded.

XML Format

XML Format provides Full provisioning support

- Default phone configuration support
- Automatic firmware update support
- Multiple language support
- Extended dial plan support
- Directory provisioning support

Structure

1. One general setting file container <setting-files> per phone type, i.e. ET605, etc., providing a list of setting file URLs linked to:

- **One settings container (<settings>) per phone type** containing general configuration parameters grouped in XML tags (<phone-settings>, <functionKeys>, <tbook>, <dialplan>) OR/AND individual XML Settings Files per phone type containing general configuration parameters:(Phone settings setting file, Function key setting file, Directory setting file, Dial plan setting file).
- **One Phone user interface language file container per phone type** with a list of phone user interface language file URLs.
- **One Web user interface language file container per phone type** with a list of web user user interface language file URLs.

2. One specific setting file container <setting-files> per phone, i.e. **MAC address** based, providing a list of setting file URLs linked to:

- One settings container (<settings>) per phone containing phone specific configuration parameters grouped in XML tags (<phone-settings>, <functionKeys>, <tbook>, <dialplan>) AND/OR **individual XML Settings Files one per phone** containing **phone specific** configuration parameters:(Phone settings setting file, Function key setting file, Directory setting file, Dial plan setting file).

3. Firmware setting files containing a subset of firmware related configuration parameters allowing **automated firmware updates**.

Containers are XML structures allowing to specify a list of setting file URLs/tags which will be consecutively requested by the phone. There are currently two container types supported:

- Setting Files Container
- Setting Container

Setting Files Container <setting-files>

Setting files container are XML files using the <setting-files> tag

They should be the first XML file provisioned.

They allow to specify a list of setting file URLs:

1. XML phone settings files
2. XML function key setting files
3. XML directory setting files
4. XML dial plan setting files
5. XML uploads setting files
6. XML certificate setting files
7. XML Language setting files
   - phone user interface language
   - web user interface language

The URLs are requested in the defined order.

tree:openlevels=3|root=Setting Files Container <setting-files>

Element: File

Attributes: url

Atribute values:

- XML <phone-settings> file
- XML <functionKeys> file
- XML <ReplacementPlan> file
- XML <tbook> file
- XML <dialplan> file
- XML <uploads> file
- XML <certificates> file
- XML <gui-languages> file
- XML <web-languages> file

Settings Container <settings>

Setting container are XML files using the <settings> tag.

They allow to specify the following setting file tags in one file, e.g:

- <phone-settings> tag

- <functionKeys> tag
- <tbook> tag
- <dialplan> tag
- <uploads> tag
- <certificates> tag

tree:openlevels=2|root=Settings Container <settings>

Supported Container Tags and Sub Tags

<phone-settings> XML tag

The phone settings XML tag (<phone-settings>) contains the main part of the available settings (configuration parameters).

This XML tag can be used either:

- inside the <settings> tag:

```
<phone-settings e="2">
  <parameter(1)> idx="<index>" perm="<permission flag>"
<value></<parameter>
  ...
  <parameter(n)> idx="<index>" perm="<permission flag>"
<value></<parameter>>
</phone-settings>
```

- or as an individual XML file whose URL is listed inside <setting-files> tag:

```
<?xml version="1.0" encoding="utf-8"?>
 <phone-settings e="2">
  <parameter(1)> idx="<index>" perm="<permission flag>"
<value></<parameter>
  ...
  <parameter(n)> idx="<index>" perm="<permission flag>"
<value></<parameter>>
 </phone-settings>
```

Level 1

Element: phone-settings

Attributes: e

- e="2" defines that unicode-values inside xml-escapes (e.g. & # 6 4 ;) may be greater than 255.

Level 2

Element: <phone-settings-parameter>

Attributes:

- **idx** representing a valid account index.
- **perm** representing a valid permission flag.
- **value** representing the parameter value. For a detailed list of parameter values, see Chapter 5, *Configuration File Parameter Guide*

<functionKeys> XML tag

The function key settings XML (<functionKeys> or <function-keys>) tag contains the free programmable function key configuration parameters.

The tags <functionKeys> and <function-keys> are equivalent. These XML tags can be used either

- inside the <settings> tag:

```
 <functionKeys>
  <fkey idx="<function_key_index)>" context="<function_key_context>"
label="<function_key_label>" [default_text="<label_default_text>"]
perm="<permission flag>"><value></fkey>
  ...
  <fkey idx="<function_key_index>" context="<function_key_context>"
label="<function_key_label>" [default_text="<label_default_text>"]
perm="<permission flag>"><value></fkey>
 </functionKeys>
```

- or as an individual XML file whose URL is listed inside <setting-files> tag:

```
<?xml version="1.0" encoding="utf-8"?>
 <functionKeys>
  <fkey idx="<function_key_index)>" context="<function_key_context>"
label="<function_key_label>" [default_text="<label_default_text>"]
perm="<permission flag>"><value></fkey>
  ...
  <fkey idx="<function_key_index>" context="<function_key_context>"
label="<function_key_label>" [default_text="<label_default_text>"]
perm="<permission flag>"><value></fkey>
 </functionKeys>
```

Level 1

Element: functionKeys

Level 2

Element: fkey

Attributes:

- **idx** string defines the free function key index n.

This is the function key index (fkey idx) range on ET605 phones:

- Freely Programmable Line Keys (P1-P2):
  - Line keys: 0-1

- **context** string assigns the function key to a SIP Identity (1 to 2) registered on the phone. “Active” assigns the current active identity to that function key.
- **label** string defines the short label to be used to describe the fkey.
- **lp** string defines if long press of the fkey on the phone can be used to display the fkey’s configuration menu. Default value is “on”. NOTE: Value must be set to “off” for the functions Push2Talk (p2t) and Line Info Layer. When setting a value that is not the default value for this setting you have to also add clp="1".
- **default_text** string defines what to show as decription for a key that has neither its fkey_label setting set nor an XML-description that provides a label.

  This attribute is optional and applicable only to self-labeling keys. It has no effect when the key is not self-labeling. When omitted on a self-labeling key, label_default_text remains unchanged.

  You may define any arbitrary fixed text, but note that there are three key words that allow to insert dynamic information related to the key:

  - **$name :**
    - on a (shared) line key:
      - when there is an active call on the key:
        the remote name (or number if no name is available) is inserted
      - when there is no active call:
        - when context is 'active' and $type is not also included:
          the key type is inserted
        - when context is a specific identity:
          the local name or number is inserted
    - on other keys: the destination configured on the key is inserted
  - **$state** will insert the key state, when applicable (not all keys have states)

  Setting with index 0 describes the format of the upper left key on the first ET6 attached on phones without self-labeling keys. On phones with self-labeling keys, 0 describes the format of the first key on page 1.

- **perm** string defines the permission flag. See *“Flags” on page 50*.
- **value** string defines the function key value, optionally followed by a space and a value-specific argument.

List of valid values of the value string

The following table lists the available values for the value string.

value string	Description
`auto_answer`	Enables you to switch Auto Answer functionality on/off for the first outgoing identity. If you don't provide the identity, the auto answer functionality is switched for all identities.
`blf`	Busy Lamp Field (BLF). Enables users to monitor the dislog state of another phone/user extension. This is indicated by the LEDs adjacent to the particular key.
`button`	This is a button that is connected to your PBX.
`BW-ACD`	BroadWorks Automated Call Distribution (ACD) configuration.
`BW-Anywhere`	BroadWorks Anywhere configuration.
`BW-RemoteOffice`	BroadWorks Remote Office configuration.
`BW-ServerBLF`	Broadworks Busy Lamp Field (BLF) configuration.
`call_agent`	The phone can be used as a Call Agent that distinguishes five states: ■ AgentLoggedOnEvent (Sign-In) ■ AgentLoggedOffEvent (Sign-Out) ■ AgentNotReadyEvent (Unavailable) ■ AgentReadyEvent (Available) ■ AgentWorkingAfterCallEvent (Wrap-Up) These states are governed by the function key ACD, which is configured in the Function Keys section of the webinterface.
`conference`	Press the key to set up a conference call and select desired participants.
`Contact List Buddy`	Let the key reflect one of the buddies from a resource-list-subscription.

value string	Description
`dest`	Extension/destination. This key type is used for: ■ Extension Monitoring (Busy Lamp Field (BLF)) & Call Pickup: This allows showing the status (idle, ringing, held call, busy) of a distinct phone extension on your phone ■ Speed Dial: Pressing this key during idle state will dial the programmed extension ("number"). ■ Call Deflection: Pressing this key during an incoming call will deflect the incoming call to the programmed extension ("number").
`dtmf`	This option allows the specification of arbitrary key sequences (allowed digits: "0-9", "*", "#", "A-D" and flash: "!"), which will be sent via DTMF when this button is pressed. This can only be done during an active call.
`icom`	Pressing the key bound to “Intercom” enables the intercom mode: the phone will be directly connected to the VTech phone if authentication is set up properly. This feature is useful in an office environment as a quick access key to connect to the operator or the secretary.
`ivr`	The argument is a number that is dialed on key press i.e. sending out an INVITE. Once the call has been established, pressing the same IVR key would send out dtmf digits comprising that number. This can be used to control IVR applications by one key only.
`keyevent`	Key events than can be mapped onto the predefined or the usual function keys. Use the text `keyevent` followed by a space, and one of the key events in *“List of valid key events” on page 36*. Example: `keyevent F_ADR_BOOK`
`line`	“Line” key can behave as a private or line shared line key, according to the setting **user_shared_line**. ■ **Private Line:** Assigns local SIP identities (lines) to programmable keys. ■ **Shared Line:** Assigns local SIP identities (lines) to programmable keys. Enables subscribers to share SIP lines and also provides status monitoring of the shared line. See also *“Line” on page 95*.

value string	Description
`multicast`	With this function key the phone can start a multicast RTP stream. You must insert the multicast destination address and a port, e.g.: 239.255.255.245:5555
`none`	If you like to map a key to no functionality at all, use this type.
`orbit`	Park Orbit. This feature is useful for call center environments and all places where there is a great inflow of calls and some kind of queuing is required to manage them. Some PBX solutions provide its customers with the opportunity to set up parking orbits, where calls can be parked and picked up. The option “Park Orbit” enables the phone to provide this feature.
`p2t`	Push2Talk feature enables users to make Intercom calls to a programmed destination via the function keys. **lp** string (long press) must be turned “off” as it blocks the Push2Talk (PTT) functionality. See also *“Push2Talk” on page 96*.
`presence`	The phone will subscribe to the presence state of the destination url with event type presence. The associated led will reflect the presence state of the destination e.g. ringing, available etc. Hitting the programmable key (usually when the destination is available and can receive a call) shall dial that number.
`recorder`	Voice recorder. This feature can be used to record a conversation during an active call or short messages or memos for personal use. Another possible usage is the recording of a debate or discussion, to keep audio minutes of a meeting, or to record a conference. This option can be set up with a valid voice recording account.
`redirect`	Forward To. This option can be used to create a shortcut for setting up call forwarding for the phone. If you are using a programmable function key with LED, the LED will indicate the current state of the call forwarding.
`SendSipInfo`	Send SIP INFO while call is connected. Message contains a "generic_value" header field with custom content.
`speed`	Enables the key to speed dial a preset number. See also *“Speed Dial” on page 96*.
`Starcod`	For making SIP calls without audiovisual indication on the phone user interface (PUI).

value string	Description
transfer	Transfers the current incoming/active call.
url	Action URLs are basically HTTP GET Requests. They can be used to send various data from the phone to a web server. See also "Action URL" on page 92.
UserInputAndSendSipInfo	Send SIP INFO while call is connected. Message contains a "generic_value" header field with custom string. User will be prompted to input the custom string when key is pressed.
xml	XML Definition/Customizable via XML.
XMPP-ContactPres	Enables you to publish a presence state to indicate your current communication status in order to inform your contacts of your availability and willingness to communicate.

List of valid key events

This tables lists the valid key events for **value** strings defined as keyevent. See "keyevent" on page 34.

keyevent	Description
F_ADR_BOOK	Provides access to the internal phone directory.
F_ACCEPTED_LIST	Provides access to the ACCEPTED call history list.
F_CALL_LIST	Provides access to the call history list (missed, received, dialed calls).
F_CONFERENCE	Enables the user to press the key to set up a conference call and select desired participants.
F_CONTACTS	Provides access to the Contact List, where the Presence State of selected users can be seen (online, busy, offline).
F_DELETE_MSG	Deletes a text message.
F_DENYALL	This key event will deny the incoming call and add the number to the deny list. All phones with call screen settings can alternatively do this by long-pressing the cancel key.
F_DIALOG	Shows the list of monitored extensions and allows call pickup. Will auto hide when not applicable, i.e. when the list would be empty.
F_DIRECTORY_SEARCH	Enables the user to lookup remote directory while dialing a number. Once set, this pressed key will open up the Direcory Search window.

keyevent	Description
F_DND	Toggles the Do Not Disturb (DND) status on the phone. When mapped to a function key with a LED, it will indicate the current DND state. Permanent light is 'DND on' and no light means 'DND off'.
F_FAVORITES	Opens the Favorites Address Book.
F_HOLD	Places an active call on "Hold".
F_HOLD_PRIVATE	Places an active call on "Private Hold".
F_HOTELING	Hoteling feature enables users (guests) within an office to use any cubicle phone (hosts) in the office by logging in to the host phone and having the host phone provisioned with guest's device profile settings.
F_LABEL_PAGE_NEXT	Opens the next label page in a round-robin fashion on phones with self-labeling keys.
F_LABEL_PAGE_PREV	Opens the previous label page in a round-robin fashion on phones with self-labeling keys.
F_LOGOFF_ALL	Caution: This option will delete all account settings!! Usage: Mainly useful for call centers with frequently changing users.
F_MISSED_LIST	Provides access to the MISSED call history list.
F_MUTE	Mutes/Unmutes during an active call. Please note that on some phones the mute key can work as a DND when Idle. You can manage this feature through the mute_is_dnd_in_idle setting.
F_NEXT_ID	Shows the next outgoing ID.
F_NONE	If you like to map a key to no functionality at all, use this type.
F_OCIP	Access the Broadsoft directory via the Open Client Interface-Provisioning (OCI-P) that allows third-party applications to peform all business functions peformed by BroadWorks.
F_PRESENCE	Provides access to the Presence State list, where the Presence State of each SIP Identity can be defined e.g. online, offline, busy, invisible).
F_PREV_ID	Shows the previous outgoing ID.
F_REBOOT	Displays a screen on the phone asking if you want to reboot.
F_REC	Toggle recording on/off during an active call.

keyevent	Description
F_REDIAL	Provides access to the DIALED call history list.
F_REDIRECT	Can be used to create a shortcut for setting up call forwarding for the phone. If you are using a programmable function key with LED, the LED will indicate the current state of the call forwarding.
F_RETRIEVE	Retrieves the mailbox messages. This key becomes active after the phone has received a message waiting indication (MWI) with a valid mailbox URI.
F_RINGER_SILENT	Turns the ringer off/on.
F_SERVER_AB	Provides access to an external phone directory.
F_SETTINGS	Shows the current MENU of the phone.
F_STATUS	Shows a list of status messages.
F_SUPPORT	Displays the Help screen as seen in *"Help" on page 74*.
F_TRANSFER	Transfers the current incoming/active call.
F_ZONES	Multicast paging zones.
HEADSET	Turn Headset mode on/off.

<ReplacementPlan> XML tag

The xml replacement plans (<ReplacementPlan> tag) contain XMLs that get inserted into the settings when certain conditions are met. The <ReplacementPlan> tag can be used either:

- inside the <settings> tag or
- as an individual XML file whose URL is listed inside <setting-files> tag

Example:

```
 <ReplacementPlan>
  <key id="ResourceListBuddy"
wui_translation_key="fkeys_ssi_buddy_from_server_list">...</key>
  <setting_replacement id="user_event_list_uri">...</setting_replacement>
 </ReplacementPlan>
```

Level 1

Element: ReplacementPlan

- <ReplacementPlan> knows two sorts of subtrees: <key> and <setting_replacement> (described below).

- You may delete plans already on the phone by providing the <key> or <setting_replacement> with the correct id-attribute set but without any subtree-content.

Level 2

Element: **key** defines a key-type that will get listed in fkey-WUI-page as type for a line-key.

Attributes:

- **id** attribute is mandatory and used to define the key type, so it can be deleted or altered in later provisions.
- **wui_translation_key** attribute is mandatory and used to define the key type, so it can be deleted or altered in later provisions.

  If the wui_translation_key is not part of the translation-map, it will be used directly to describe the key in the WUI. Note: renamed and moved to general tag since firmware version 8.9.3.66).
- The subtrees will get additional variables in the beginning of the init-section:
  - The variable "ui_argument" will hold whatever is entered in the "Number"-text-field next to the type in the fkey-WUI-page.
  - The variable "ui_label" will hold whatever is entered in the "Short Text"-text-field next to the number in the fkey-WUI-page.

Element: **setting_replacement** defines a an XML that will be used should the named setting get set up with non-XMLcontent.

Attributes: **id** attribute names the setting, currently ONLY user_event_list_uri is valid.

- The subtrees will get additional variables in the beginning of the init-section:
  - The variable "setting_value" contains the exact non-XML setting value that was used for set up.
  - The variable "setting_index" contains the index of the setting.

<tbook> XML tag

The directory settings XML tag (<tbook> or <phone-book) contains a list of contact entries to be provisioned into the internal phone directory.

The tags <tbook> and <phone-book> are equivalent: These XML tags can be used either

- inside the <settings> tag:

```
 <tbook complete="true">
  <item context="<outgoing_SIP_identity>" type="<contact_category>"
index="<contact_index(0)>">
   <name><contact_name</name>
   <number><contact_name></number>
  </item>
  ...
```

```
  <item context="<outgoing_SIP_identity>" type="<contact_category>"
index="<contact_index(n)>">
   <name><contact_name</name>
   <number><contact_name></number>
  </item>
 </tbook>
```

- or as an individual XML file whose URL is listed inside <setting-files> tag

```
<?xml version="1.0" encoding="utf-8"?>
 <tbook complete="true">
  <item context="<outgoing_SIP_identity>" type="<contact_category>"
index="<contact_index(0)>">
   <name><contact_name</name>
   <number><contact_name></number>
  </item>
  ...
  <item context="<outgoing_SIP_identity>" type="<contact_category>"
index="<contact_index(n)>">
   <name><contact_name</name>
   <number><contact_name></number>
  </item>
 </tbook>
```

Level 1

Element: tbook

Attributes: e

e="2" defines that unicode-values inside xml-escapes (e.g. & # 6 4 ;) may be greater than 255.

- **complete**

  When **complete**="true" is provisioned, the phones know that the provided tbook is complete and thus the previous one can be deleted (this is the only way to delete entries from the internal tbook via provisioning).

Level 2

Element: Item

Each Item tag defines one directory contact entry and requires the following attributes:

Attributes:

- **context** string defines the SIP identity (line/account) this contact should be called with
- **type** string defines the contact's category. Only provides either one of these contact types: ""/"VIP"/"DENY"

- **fav** marks a person as favorite
- **index** provided is used to change the specific entry at that index. Previously, the tbook tried to match the entries provided to the internal entries via the given number string (and still does so when no index is provided), which allowed the provisioner to change everything but this phone number. Now, with the help of the index, even that can be done.

Elements:

- **name** string defines the contact's name
- **number** string defines the contact's number
- **number_type** defines either one of ""/"sip"/"mobile"/"fixed"/"home"/"business"
- **first_name** string defines a person’s first name
- **last_name** string defines a person’s first name
- **title** string defines a person’s company title like "Head of Finances"
- **organization** string defines the organization/company the person works for
- **email** string defines the person’s email address
- **note** string defines a note.
- **action_url** string defines the action URL to request when the phone receives or places a call with this directory entry.
- **group** defines either one of ""/"work"/"colleague"/"family"/"friend"
- **birthday** defines the birthday in either dd.mm.yyyy or mm/dd/yyyy format.

Multiple numbers per person are achieved by defining a Master-entry, which sets up certain attributes that hold true for all its telephone numbers (like first_name and last_name) and 2 or more Member-entries.

- The Master-entry is defined through:
  - **type**="MASTER"
  - **number**=AnyUniqueNumber - must be one of the telephone **number**s of one of the members
  - Masters cannot define a context.
- The Member-entries are defined by:
  - **first_name**=Member_Alias
  - **last_name**=UniqueNumberOfMaster
  - Members cannot define neither **birthday** nor **fav** attribute.

<dialplan> XML tag

The dial plan settings (<dialplan> or <dial-plan> tag) contains the global dial plan parameters.

XML Dial plan can be placed either:

- inside the <settings> tag

```
<?xml version="1.0" encoding="utf-8" ?>
 <settings>
  <phone-settings></phone-settings>
  <functionKeys></functionKeys>
  <tbook></tbook>
  <uploads></uploads>
  <certificates></certificates>
  <dialplan e="2">
   <!--Example North American Dialplan-->
   <TEMPLATE MATCH="0" Timeout="1" User="Phone"/>
   <TEMPLATE MATCH="9,011*" Timeout="6" User="Phone"/>
   <TEMPLATE MATCH="9,0" Timeout="1" User="Phone"/>
   <TEMPLATE MATCH="9,11" Timeout="0" User="Phone" Rewrite="9911"/>
   <TEMPLATE MATCH="9,.11" Timeout="0" User="Phone"/>
   <TEMPLATE MATCH="9,101.............." Timeout="0" User="Phone"/>
   <TEMPLATE MATCH="9,10............." Timeout="0" User="Phone"/>
   <TEMPLATE MATCH="9,10*" Timeout="6" User="Phone"/>
   <TEMPLATE MATCH="9,1.........." Timeout="0" User="Phone"/>
   <TEMPLATE MATCH="9,......." Timeout="0" User="Phone"/>
   <TEMPLATE MATCH="*" Timeout="15"/>
  </dialplan>
 </settings>
```

- or as an individual XML file whose URL is listed inside <setting-files> tag

```
<?xml version="1.0" encoding="utf-8"?>
<dialplan e="2">
 <!--Example North American Dialplan-->
 <TEMPLATE MATCH="0" Timeout="1" User="Phone"/>
 <TEMPLATE MATCH="9,011*" Timeout="6" User="Phone"/>
 <TEMPLATE MATCH="9,0" Timeout="1" User="Phone"/>
 <TEMPLATE MATCH="9,11" Timeout="0" User="Phone" Rewrite="9911"/>
 <TEMPLATE MATCH="9,.11" Timeout="0" User="Phone"/>
 <TEMPLATE MATCH="9,101.............." Timeout="0" User="Phone"/>
 <TEMPLATE MATCH="9,10............." Timeout="0" User="Phone"/>
 <TEMPLATE MATCH="9,10*" Timeout="6" User="Phone"/>
 <TEMPLATE MATCH="9,1.........." Timeout="0" User="Phone"/>
 <TEMPLATE MATCH="9,......." Timeout="0" User="Phone"/>
 <TEMPLATE MATCH="*" Timeout="15"/>
</dialplan>
```

Level 1

<u>Element:</u> dialplan

<u>Attributes:</u> e

e="2" defines that unicode-values inside xml-escapes (e.g. & # 6 4 ;) may be greater than 255.

Level 2

Element: TEMPLATE

Attributes:

- **MATCH="pattern"** is the dial pattern to match. While entering the pattern: numbers 0-9, * and # represent the keys on the phone that are entered. Use a period (.) to match any key. An asterisk (*) at the very end of the pattern matches one or more characters. Matching just the * key without interference with the wildcard character is done by escaping it with a backslash "*". To have the phone generate a secondary dial tone when the part of the template matches, use a comma (,).
- **Timeout="sec"** is the number of seconds before a timeout will occur and the number will be dialed as entered by the user. To have the number dial immediately, specify 0.
- **User="type"** is the either IP or Phone. Enter User=phone or User=IP to have the tag automatically added to the dialed number. Currently User=phone is supported.
- **Rewrite="altstrng"** is the alternate string to be dialed instead of what the user enters. This field can be left empty.
- **identity="number"** is the identity that is used to establish the call. If no identity is given, the active identity is used.

If desired, specify at the end of each string where comment defines the type of plan (for example, Long Distance or Corporate Dial Plan).

Special note on dialplan nomenclature:

1. The special characters supported in 'match' include '.' for any digit between 0-9.
2. '*' as a wildcard for all charcaters and digits.
3. '[' & ']' to specify a range for single digit input e.g. match="[4-7].." would mean any three digit number where the first digit is either 4, 5, 6 or 7 i.e. 4-7 inclusive of both limits.
4. ',' is used to indicate secondary local dialtone. It often follows a digit usually 9 or 0.
5. The closest logical match through all the dialplans would be selected for ay given input match. Ascending or descending order does not over rule this feature.
6. If one doesn't want to specify a timeout, rewrite or user; either leave them empty or do not include them at all. In this case the default for all would be used.
7. The dialplan attributes can be saved either in capital or small letters. The phone would internally store them in lower case.

<uploads> XML tag

The <uploads> tag contains a list of the URLs for uploading new designs onto the phone.

This XML tag can be used either

- inside the <settings> tag

```
<uploads>
 <file url=URL type=TYPE />
</uploads>
```

- or as an individual XML file whose URL is listed inside <setting-files> tag

```
<?xml version="1.0" encoding="utf-8" ?>
<uploads>
 <file url=URL type=TYPE />
</uploads>
```

Level 2

Element: file

Attributes:

- **url** = The URL of the customization tarball file (*.tar) to be uploaded onto the phone.
- **type** =

  the following miscellaneous customization options:

  - **gui** allows replacing the default Phone User Interface background images and icons by customized ones.
  - **web** allows replacing the default Web User Interface images and stylesheets by customized ones.
  - **font** allows replacing the default Phone User Interface font by customized ones.
  - **defaults** allows replacing the default configuration parameter values by customized values.
  - **license** allows replacing the current phone license with a new license to enable additional features. The license will be ignored if it's not valid (e.g. not matching the mac address of the phone).
  - **moh** allows uploading a local music on hold file (RAW PCMU 20ms).
  - **qml** allows replacing the default QML description.

  the following allow replacing the default behaviour of the respective PUI state which is specified via XML:

  - **gui_xml_state_settings** allows replacing the default Phone User Interface Menu by a customized menu, see PUI Menu
  - gui_xml_addperson
  - gui_xml_contactlist
  - gui_xml_state_conference state conference
  - gui_xml_state_details state details

- gui_xml_state_holding state holding
- gui_xml_state_multicast state multicast part 1
- gui_xml_state_multicast_file state multicast part 2
- gui_xml_state_status_message_file
- gui_xml_call_lists_file
- gui_xml_call_lists_list_file
- gui_xml_contact_pool
- gui_xml_message_file
- gui_xml_call_lists_details
- gui_xml_edit_user
- gui_xml_templates
- gui_xml_broadsoft_acd_state_chooser
- gui_xml_decision
- gui_xml_login_wizard
- gui_xml_pkeys
- gui_xml_ucmenu

<certificates> XML tag

The certificates settings (<certificates> tag) contains the trusted server certificates. This XML tag can be used either

- inside the <settings> tag or
- as an individual XML file whose URL is listed inside <setting-files> tag

The tag contains an attribute with the URL of the certificate file to fetch:

```
<certificate url="http://some.url/certificate.der" />
```

Please note that the download of the certificate is delayed after all provisioning xml files have been loaded and processed.

A second variant of this tag is supported, where the content of the certificate file is included as a base64 encoded string:

```
<certificate type="base64">...</certificate>
```

The benefit of this variant is, that the certificate is immediately available after processing the line in the provisioning XML

Level 1

Element: certificates

Attribute: url, type

Language File Container

Language file container may consist of a list of language file URLs each one representing a different language. The following language file containers are currently supported:

- Phone User Interface language file container (**<gui-languages> tag**)
- Web User Interface language file container (**<web-languages> tag**)

<gui-languages> XML tag

Syntax:

```
<?xml version="1.0" encoding="utf-8" ?>
 <gui-languages>
  <language url="<Phone User Interface Language file URL(1)>"
name="<language_name(1)>" />
  ...
  <language url="<Phone User Interface Language file URL(n)>"
name="<language_name(n)>" />
 </gui-languages>
```

Level 1

Element: gui-languages

Level 2

Element: language

Attributes:

- **url** string contains phone user interface language file URLs (1)..(n)
- **name** string determines the language's name in the phone user interface language list.

<web-languages> XML tag

Syntax:

```
<?xml version="1.0" encoding="utf-8" ?>
 <web-languages>
  <language url="<Web User Interface Language file URL(1)>"
name="<language_name(1)>" />
  ...
  <language url="<Web User Interface Language file URL(n)>"
name="<language_name(n)>" />
 </web-languages>
```

Level 1

Element: web-languages

Level 2

Element: language

Attributes:

- **url** string contains Web User Interface language file URLs (1)..(n)
- **name** string determines the language's name in the web user interface language list.

Language files

Language files contain the language phrases. When selecting a new language from the phone or web user interface language list the content of the associated file will be stored in the phone´s RAM. The following language files are currently supported:

- Phone User Interface language files (**<phrases> tag**)
- Web User Interface language files (**<w_phrases> tag**)

Language files depend on the firmware version, i.e. each file is unique per firmware version. However the language files of the latest release are always backwards compatible.

<phrases> XML tag

Syntax:

```
<?xml version="1.0" encoding="utf-8"?>
 <phrases>
  <phrase i="<index>" n="<name>" t="<translation>"/>
  ...
  <phrase i="<index>" n="<name>" t="<translation>"/>
  <language i="<index>" t="<language name>"/>
 </phrases>
```

Level 1

Element: phrases

Level 2

Element: **phrase** tag defines one Phone User Interface phrase.

Attributes:

- **i** string represents the running <index> of the phrases
- **n** string represents the internally used (english) variable <name> used for the translation
- **t** string represents the <translation>

Element: **language** tag defines the language name

Attributes:

- **i** string represents the <index> of the language name, usually equal 0
- **t** string represents the <language name>, should match the name string used in (<gui-languages> tag)

<w-phrases> XML tag

Syntax:

```
<?xml version="1.0" encoding="utf-8"?>
 <w_phrases>
  <w_phrase i="<index>" n="<name>" t="<translation>"/>
  ...
  <w_phrase i="<index>" n="<name>" t="<translation>"/>
  <language i="<index>" t="<language name>"/>
 </w_phrases>
```

Level 1

Element: w-phrases

Level 2

Element: **w_phrase** tag defines one Web User Interface phrase

Attributes:

- **i** string represents the running <index> of the phrases
- **n** string represents the internally used (english) variable <name> used for the translation
- **t** string represents the translation

Element: **language** tag defines the language name

Attributes:

- **i** string represents the index of the language name, usually equal 0
- **t** string represents the <language name>, should match the name string used in (<web-languages> tag)

<firmware-settings> XML tag (Firmware File)

The Firmware Configuration File (<firmware-settings> tag) contains the "firmware image" URL. The Firmware Configuration File will only be requested if its URL had been specified by the configuration parameter firmware_status before. firmware_status should only be defined in the phone settings file (<phone-settings> tag).

NOTE: The firmware configuration file URL must not be specified in any container setting file.

Phone firmware syntax

```
<?xml version="1.0" encoding="utf-8" ?>
 <firmware-settings>
  <firmware perm="<permission flag>"><value></firmware>
 </firmware-settings>
```

Level 1

Element: firmware-settings

Level 2

Element: **firmware** tag represents the only allowed configuration parameter.

Attributes:

- **perm** string represents the <permission flag> (see *"XML Syntax" on page 50*).
- **value** string represents the phone firmware image file URL.

Expansion module firmware syntax

you can also update the expansion module via provisioning defining the firmware_uxm parameter.

```
<?xml version="1.0" encoding="utf-8" ?>
 <firmware-settings>
```

```
 <firmware_uxm perm="<permission flag>"><value></firmware_uxm>
</firmware-settings>
```

Level 1

Element: firmware-settings

Level 2

Element: **firmware_uxm** tag represents the only allowed configuration parameter.

Attributes:

- **perm** string represents the <permission flag> (see *“XML Syntax” on page 50*).
- **value** string represents the expansion module firmware image file URL.

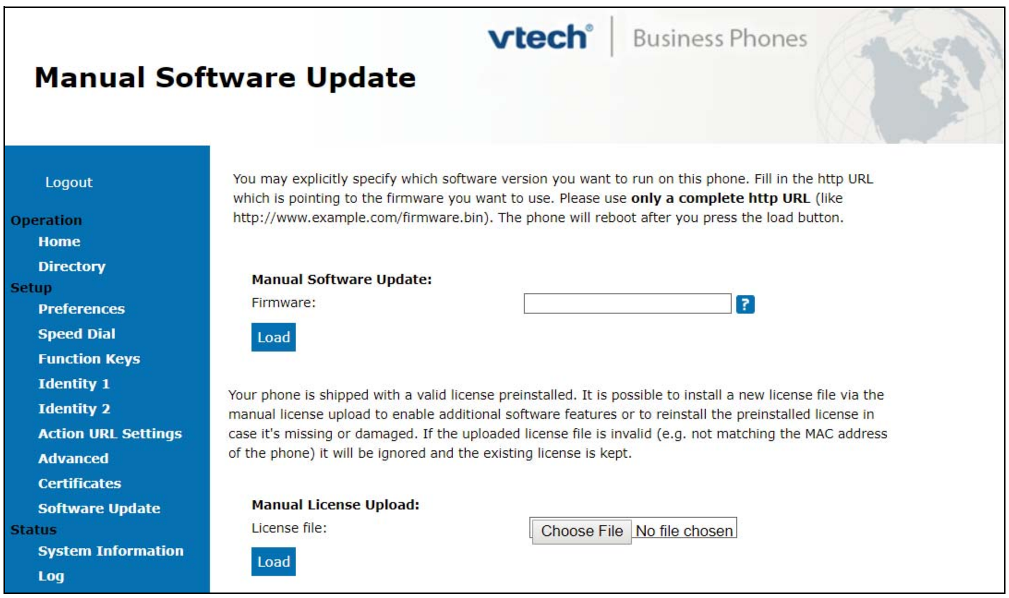
XML Syntax

Syntax	XML Format
Description	The syntax depends on the XML tag: ■ Container: <setting-files>, <settings> ■ Setting Files: <phone-settings>, <functionKeys>, <tbook>, <dialplan>, <ReplacementPlan> ■ Firmware File: <firmware-settings> ■ Language Files: <gui-languages>, <phrases>, <web-languages>, <w_phrases>
Coding	**UTF-8**
Hints	XML header is required. <?xml version=1.0 encoding=utf-8?>
Flags	Flags are defined as permission flags in the string perm within XML tags. Valid values are: ■ **perm=!**: The configuration parameter can be changed by the user and will not be overwritten by mass provisioning. **NOTE:** If administrators want to be able to overwrite user parameter definitions, they need to use perm=$. With perm=!, the settings can be changed by mass provisioning **only** if the end user has not made changes to the configuration on the phone itself or on its Web interface. ■ **perm=&** or **perm=R** or **perm=** : The configuration parameters are Read Only and cannot be changed by the end user. ■ **perm=$** or **perm=RW** or **perm="** The configuration parameters can be changed by the end user but will be overwritten by mass provisioning.

Manual Software Update

You can manually update the software of your phone by following these steps:

1. On a web browser, visit *businessphones.vtech.com* and open the ET605 downloads page.
2. Read any release notes that are available.
3. Copy the URL link to the firmware update file.
   This will be a .bin file. For example: VTechET605-SIP-8.10.1.22-0-SIP-r.bin
4. Open the ET605 WebUI interface, and open the **Software Update** page.
5. In the **Firmware** field, paste the link to the firmware update file.



6. Click **Load**.
   Your ET605 phone reboots and starts the software update.

   Note: Do not disconnect the power at any time during this process!
7. After your phone has finished the software update, check the firmware version.
   - From the WebUI: open the **System Information** page.

     The Firmware-Version is displayed on the page.
     For example, VTechET605-SIP-8.10.1.22-0
   - From the phone menu:
     - In Administrator mode: Select **6 Information** > **2 System info**

- In User mode: Select: **5 Information** > **2 System Info**

  The firmware version is displayed in the first line of the display.
  For example, VTechET605-SIP-8.10.1.22-0

CHAPTER 3

PHONE MENU REFERENCE

This chapter describes how to use the phone menu to configure the phone settings.

This chapter covers:

For more information about the other phone menus, see the ET605 User Guide.

Viewing the Phone Menu

To view the phone menu on the ET605 display:

- Press the navigation key

  –OR–

- Press the function key below , if the symbol is available.

To select menu items and settings on the phone menu:

- Press a number on the alphanumeric keypad

  –OR–

- Press and to scroll to the setting and press .

To cancel and return to the previous screen:

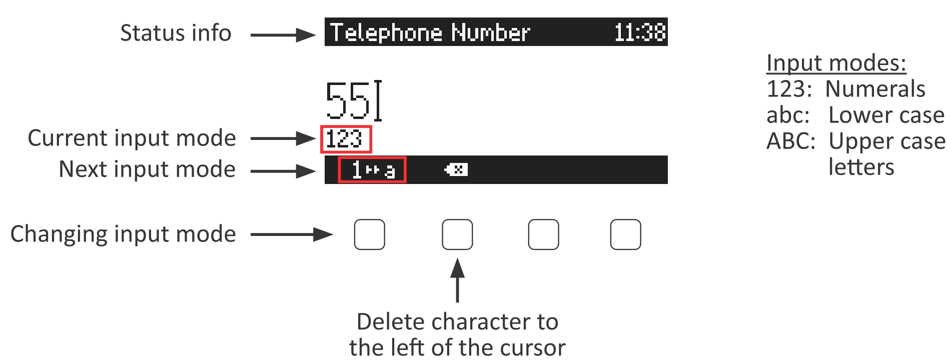
- Press .

To return to the idle screen:

- Press for two seconds.

Alphanumeric keypad

Input modes and navigation



On phone screens where you are keying in entries, the current input mode is indicated underneath the cursor. Press the left function key underneath the display to switch to the input mode indicated by the symbol directly above it in the function key line.

Text underneath cursor = current input mode	Press function key to switch to input mode indicated by symbol in function key line
123	1▸▸a
abc	a▸▸A
ABC	A▸▸1

Entering numerals, letters, special characters, and symbols

When entering letters and special characters, pause briefly after each character until the cursor has moved forward so that you won't overwrite the last character you entered. Pausing is not necessary when entering numerals.

Numerals: In numeral mode, press the respective number key to type the number printed on the key.

Letters: When in input modes lower and upper case letters, press the alphanumeric key with the respective letter one, two, three, or four times quickly to type the first, second, third, or fourth letter printed on the key. Pause briefly after each letter.

Example: In lower case letter mode, press the "2" key once to type an "a", twice to type a "b", and three times to type a "c".

Letters with accents and umlauts: When in input modes lower and upper case letters, press the alphanumeric key with the basic form of the respective letter as many times as necessary. Pause briefly after each letter. Available letters with accents and umlauts depend on the phone's language setting.

Example: If the phone language is German, press key "2" four times to type "ä".

Entering special characters and symbols: In input modes lower and upper case letters, press keys "0" and "1" one or more times quickly. Pause briefly after each character or symbol.

- Period. Press "1" once.
- Space (" "). Press "0" once.
- Underscore ("_"). Press "0" twice.
- Special characters listed in the following table. Press "1" as many times as indicated:

1x	.	2x	+	3x	@	4x	1	5x	:	6x	,		
7x	?	8x	!	9x	-	10x	_	11x	/	12x	\	13x	(
14x	)	15x	;	16x	&	17x	&	18x	*	19x	#		
20x	<	21x	=	22x	>	23x	$	24x	[	25x	]		

Using the Identity menu

The ET605 supports up to two accounts or "phone numbers" with one or more providers or within an office or organization network. On VTech phones, these accounts or phone numbers are called "identities".

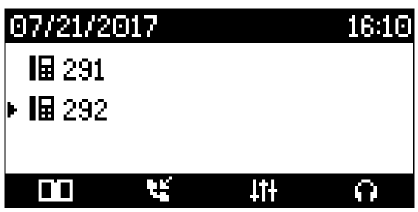
In Administrator mode, you can configure identies on the **3 Identity** phone menu.

Select Outgoing Identity menu

Use this menu item to select which identity the phone will use for outgoing calls.

1. Press > **3 Identity** > **1 Select Outgoing Identity**.
2. Select the identity you want for outgoing calls.
3. Press and hold for two seconds to return to the idle screen.

   The selected outgoing identity is indicated by an arrow.



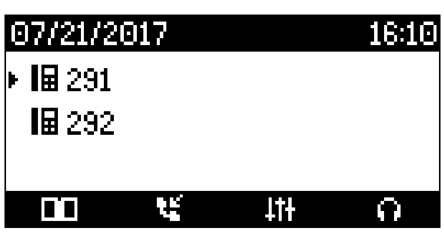
Reregister Identity menu

Use this menu item to reregister one or all identities.

1. Press > **3 Identity** > **2 Reregister identity**.
2. Select the identity you want to log off.

   –OR–

   Select **1 All Identities**.
3. The Identity menu appears.
4. Press and hold for two seconds to return to the idle screen.



After successful reregistration, the phone symbol is displayed beside each identity.

Edit Identity menu

The Edit Identity menu item enables you to configure or edit an identity.

Edit Identity (Hotdesking)

Use this menu item to configure or edit an identity for hotdesking (one phone shared by many users). If you need to enter more data, follow the steps described in *“Edit Identity” on page 59*.

1. Press > **3 Identity** > **3 Edit Identity** > **1 Hotdesking**.
2. Select a free identity with , or press its number in the menu.
3. Enter the account with which you register to a SIP registrar/proxy.

4. Enter the IP or DNS address of the registrar/proxy where you want to register this account.

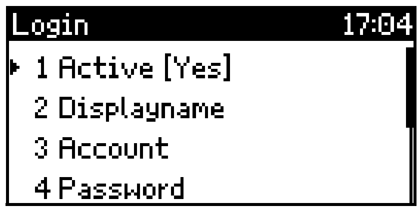
5. Enter the password for the account registered to a SIP registrar/proxy.

6. Press and hold for two seconds to return to the idle screen.

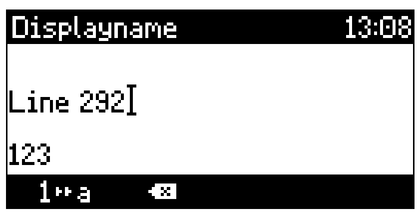
Edit Identity

Use this menu item to configure or edit an identity.

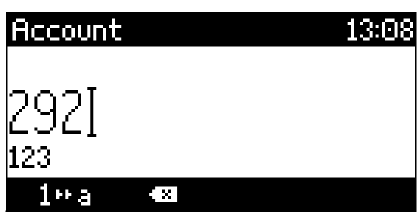
1. Press  > **3 Identity** > **3 Edit Identity** > **2 Edit Identity**.
2. Select a free identity with , or press its number in the menu.
3. Select each of the following menu items from the list, and enter the required information. **Note:** Some of these menu items might not be available.
   - **1 Active** - Select until **[Yes]** is displayed. This will make the identity active.



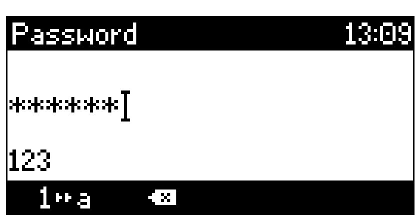
   - **2 Displayname** - Enter the name you would like to associate with the identity, e.g. “John Smith”.



   - **3 Account** - Enter the account with which you register to a SIP registrar/proxy.



   - **4 Password** - Enter the password for the account registered to a SIP registrar/proxy.



- **5 Registrar** - Enter the IP or DNS address of the registrar/proxy where you want to register this account.

- **6 Outbound Proxy** - Enter the outbound proxy in this field to ensure all SIP packets are sent via the specified communication point.

- **7 Authentication Username** - If your registrar environment needs a different user name for registration and authentication, then enter the user name for authentication. The user name in **3 Account** will be used for registration.

  If you leave this setting blank, then the user name in **3 Account** is used for both authentication and regstration.

- **8 Mailbox** - If you have set up a mailbox, specify the account name for that mailbox here to associate it with this particular SIP identity.

4. Press and hold for two seconds to return to the idle screen.
5. Follow the steps in *“Reregister Identity menu” on page 57*.

Logging off identity

Select this menu item to log off an identity or all identities.

1. Press > **3 Identity** > **4 Log off identity**.
2. Select the identity you want to log off.

   –OR–

   Select **1 Log Off All Identities**.

   If the "VTECH Welcome!" screen appears, it means there are no identities configured on the phone. You must press any button, and then enter the account, registrar, and SIP password to register an identity.

3. If the Identity phone menu appears, press and hold for two seconds to return to the idle screen.

   The idle screen shows the identity has been removed.

Using the Network menu

In Administrator mode, you can configure network settings on the **4 Network** phone menu.

IP Settings menu

Use this menu item to Internet Protocol (IP) settings for the phone.

NOTE: After changing these settings, you must reboot your phone.

IPv4 settings

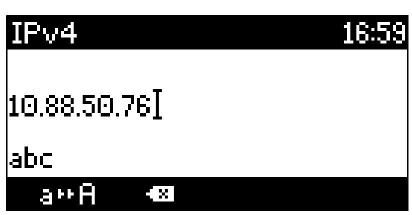
1. Press > **4 Network** > **1 IP Settings** > **1 IPv4**.
2. **To turn on DHCP:** Select **1 DHCP** until a check mark appears in the box ☑.

–OR–

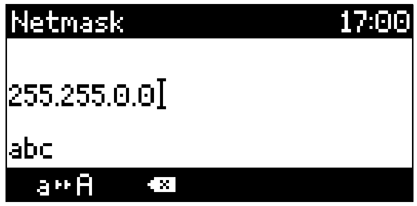
To turn off DHCP: Select each of the following menu items from the list, and enter the required information.

- **1 DHCP** - Select until there is no check mark in the box ☐.
  This will turn off DHCP.

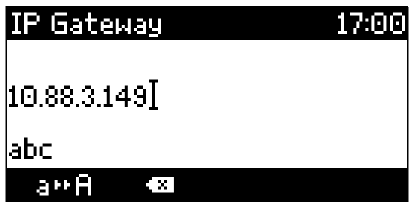
- **2 IPv4** - Enter the phone's IP address.



- **3 Netmask** - Enter the netmask for the phone.



- **4 IP Gateway** - Enter the IP address of the default IP gateway (NOT the VoIP gateway). It is the address to which the packets get routed when the desired packet address is not in the current subnet.



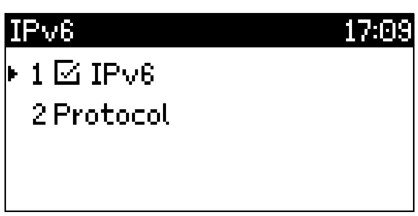
- **5 DNS Server1** - Enter the IP address of the DNS server for your network.

3. Press and hold ✕ for two seconds to return to the idle screen.

IPv6 Settings

NOTE: After changing these settings, you must reboot your phone.

1. Press > **4 Network** > **1 IP Settings** > **2 IPv6**.
2. Select **1 IPv6** until a check mark appears in the box ☑.



3. Select **2 Protocol**.
4. Select **1 DHCP & SLAAC** to assign the IP address with DHCPv6 and SLAAC (Stateless Address AutoConfiguration).

   –OR–

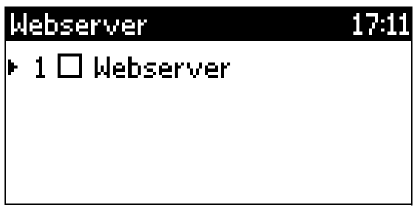
   Select **2 SLAAC** to assign the IP address with SLAAC only.

5. Press and hold for two seconds to return to the idle screen.

Webserver menu

Use this menu item to secure Web User Interface access to your phone.

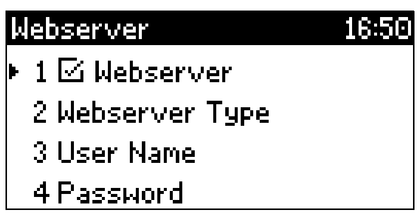
1. Press > **4 Network** > **2 Webserver**.
2. **To disable access to the web user interface (WebUI):** Select **1 Webserver** until there is no check mark in the box ☐.

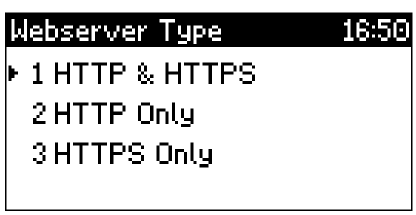
   –OR–

   To enable access to the Web user interface (WebUI): Select each of the following menu items from the list, and enter the required information.

   - **1 Webserver** - Select until a check mark appears in the box ☑.

   - **2 Webserver Type** - Select the type of connection the phone's web server is willing to answer to - HTTP & HTTPS, HTTP Only, or HTTPS only.

   - **3 User Name** - Enter a user name that will be required to access the web user interface.

- **4 Password** - Enter the password for the user name.

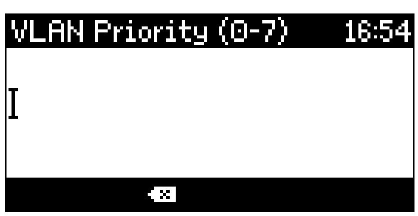
3. Press and hold for two seconds to return to the idle screen.

VLAN menu

Use this menu item to configure VLAN settings for your phone.

1. Press > **4 Network** > **3 VLAN**.
2. Select each of the following menu items from the list, and enter the required information.
   - **1 VLAN ID (1-4094)** - Enter the VLAN ID for the phone to connect to.

   - **2 VLAN Priority (0-7)** - Enter the VLAN priority.



3. Press and hold for two seconds to return to the idle screen.

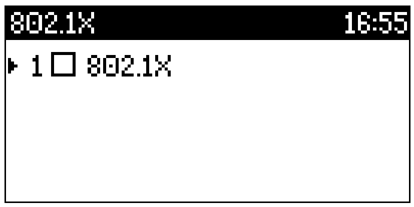
Advanced menu

Use this menu item to configure advanced settings for your phone.

802.1X menu

Use this menu item to configure 802.1X settings for your phone.

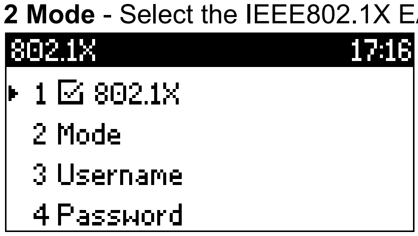
1. Press > **4 Network** > **4 Advanced** > **1 802.1X**.
2. **To disable 802.1X:** Select **1 802.1X** until there is no check mark in the box ☐.



–OR–

To enable 802.1X: Select each of the following menu items from the list, and enter the required information.

- **1 802.1X** - Select until a check mark appears in the box ☑.
- **2 Mode** - Select the IEEE802.1X EAP authentication method.



- **3 User Name** - Enter a user name that is used for IEEE802.1X EAP-MD5 authentication.

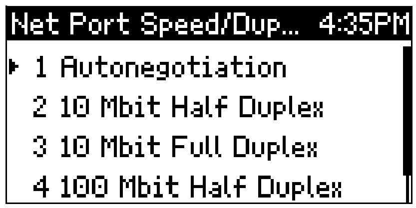
- **4 Password** - Enter the password that is used for IEEE802.1X EAP-MD5 authentication.

3. Press and hold ✕ for two seconds to return to the idle screen.

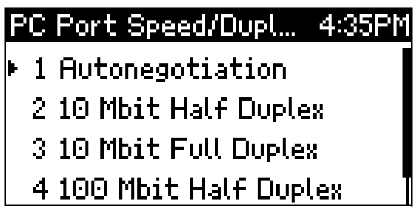
Hardware menu

Use this menu item to configure hardware settings for your phone.

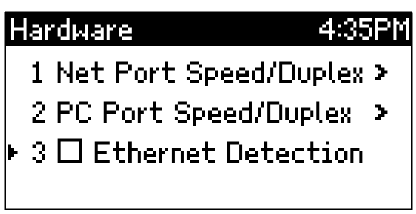
1. Press △ > **4 Network** > **4 Advanced** > **2 Hardware**.
2. Select each of the following menu items from the list, and enter the required information.
   - **1 Net Port Speed/Duplex** - Select the NET port speed/duplex.



   - **2 PC Port Speed/Duplex** - Select the PC port speed/duplex.



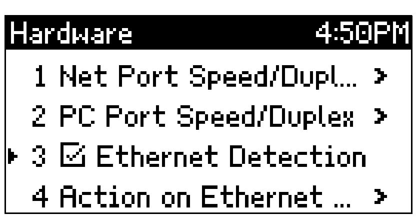
3. **To disable ethernet detection:** Select **3 Ethernet Detection** until there is no check mark in the box ☐.



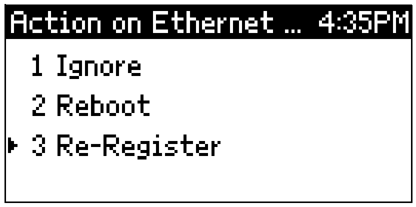
–OR–

To enable ethernet detection: Select the following menu items from the list, and enter the required information.

   - **3 Ethernet Detection** - Select until a check mark appears in the box ☑.



- **4 Action on Ethernet Cable Replug** - Select the action the phone should take when the ethernet cable is replugged.



4. Press and hold for two seconds to return to the idle screen.

NTP menu

NOTE: After changing these settings, you must reboot your phone.

Use this menu item to configure NTP settings for your phone.

1. Press > **4 Network** > **4 Advanced** > **3 NTP**.
2. Select each of the following menu items from the list, and enter the required information.

- **1 NTP Server** - Enter the domain name / IP address of the NTP server.

- **2 NTP Refresh Timer** - Enter the interval after the phone will re-synchronize the time from the NTP server, in seconds.

3. Press and hold for two seconds to return to the idle screen.

DNS menu

NOTE: After changing these settings, you must reboot your phone.

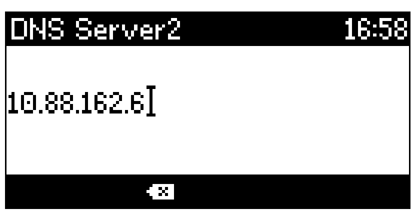
Use this menu item to configure NTP settings for your phone.

1. Press > **4 Network** > **4 Advanced** > **4 DNS**.
2. Select each of the following menu items from the list, and enter the required information.

   - **1 DNS Domain** - Enter the DNS domain for your phone.

   - **2 DNS Server1** - Enter the IP address of the DNS server for your network.

   - **3 DNS Server2** - Enter the IP address of a backup DNS server for your network.



3. Press and hold for two seconds to return to the idle screen.

Using the Maintenance menu

In Administrator mode, you can perform maintenance functions on the **5 Maintenance** phone menu.

Maintenance functions include switching between user/administrator mode, setting your keyboard lock PIN, rebooting your phone, or resetting your phone to factory default values.

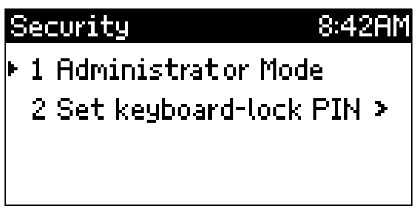
Security menu

Use this menu item to switch your phone between user mode and administrator mode, and to set your keyboard lock PIN.

Putting your phone in User Mode

1. Press > **5 Maintenance** > **1 Security** > **1 User Mode.**

   The phone is now in user mode. Menu item 1 changes to **Administrator Mode**.

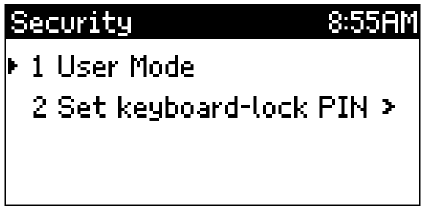


2. Press and hold for two seconds to return to the idle screen.

Putting your phone in Administrator Mode

1. Press > **4 Maintenance** > **1 Security** > **1 Administrator Mode**.
2. Enter the administrator password.

   If you entered the password correctly, the phone is now in administrator mode. Menu item 1 changes to **User Mode**.

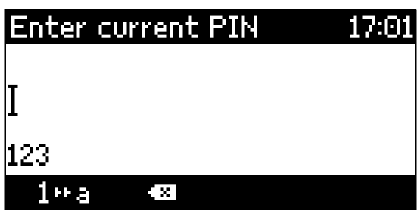


NOTE: If you forgot the administrator password, and the default administrator pasword 0000 (4 x zero) does not work, you can factory reset the phone – on the web interface: Go to the **Advanced** page > **Update** tab, and click the **Reset** button.

3. Press and hold for two seconds to return to the idle screen.

Changing the Keyboard Lock PIN

1. Press > **5 Maintenance** > **1 Security** > **2 Set keyboard-lock PIN**.
2. Enter the current PIN (if prompted).



3. Enter the new PIN or press to clear the PIN.

4. Re-enter the new PIN or press to clear the PIN.

5. Press and hold for two seconds to return to the idle screen.

Reboot

Use this menu item to reboot your phone.

1. Press > **5 Maintenance** > **2 Reboot**.
2. At the “Reboot?” prompt, press to reboot or to cancel.

   The phone reboots.

Reset Values

Use this menu item to reset your phone to factory default values.

1. Press > **5 Maintenance** > **3 Reset Values**.
2. Enter your administrator password.

The phone reboots. After rebooting, you will be prompted to select a language, time zone, dial tone scheme, and to register an identity.

Using the Information Menu

In Administrator mode, you can display information about your phone on the **6 Information** phone menu. The information you can display includes:

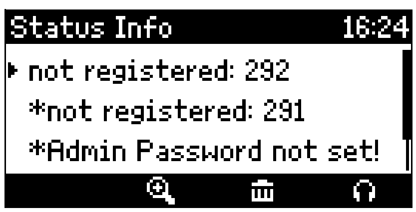
- Status messages
- Firmware version number
- IP address
- MAC address
- URL of the Web User Interface (WebUI)

Status Info

Use this menu item to display status messages - call forwarding status, passwords not set, missed calls, reboot required, etc.

1. Press > **6 Information** > **1 Status Info**.

   The status messages appear. If there are no status messages, the message “(no data available) is displayed.



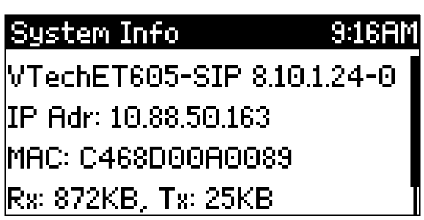
2. Press and hold for two seconds to return to the idle screen.

System Info

Use this menu item to display the firmware version number, IP address, and MAC address of the phone.

1. Press > **6 Information** > **2 System Info**.

   The phone displays the system info.



2. Press and to scroll through the information displayed on the screen.

3. Press and hold for two seconds to return to the idle screen.

Help

Use this menu item to display the URL for the phone’s web user interface (WebUI).

1. Press > **6 Information** > **3 Help**.

   The phone displays the WebUI information.

2. Press and to scroll through the information displayed on the screen.

3. Press and hold for two seconds to return to the idle screen.

CHAPTER 4

WEB USER INTERFACE (WEBUI) REFERENCE

The WebUI allows you to configure all aspects of ET605 Deskset operation, including account settings, programmable keys, network settings, contact lists, and provisioning settings. The WebUI is embedded in the ET605 operating system. When you access the WebUI, you are accessing it on the device, not on the Internet.

This chapter describes how to access the WebUI and configure ET605 settings.

This chapter covers:

- *"Using the Web User Interface (WebUI)" on page 76*.
- *"Operation pages" on page 78*.
- *"Setup pages" on page 83*.
- *"Status pages" on page 157*.

Using the Web User Interface (WebUI)

The Web User Interface (WebUI) resides on the ET605 Deskset. You can access it using a web browser. After you log in to the WebUI, you can configure the ET605 on the following pages.

Operation

- Home (see *page 78*)
- Directory (see *page 80*).

Setup

- Preferences (see *page 83*)
- Speed Dial (see *page 89*)
- Function Keys (see *page 91*)
- Identity n (see *page 99*)
- Action URL Settings (see *page 119*)
- Advanced (see *page 122*)
- Certificates (see *page 153*)
- Software Update (see *page 155*)

Status

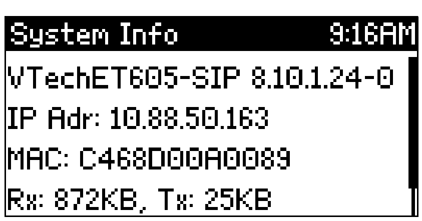
- System Information (see *page 157*)
- Log (see *page 157*)
- SIP Trace (see *page 158*)
- DNS Cache (see *page 159*)
- Subscriptions (see *page 159*)
- PCAP Trace (see *page 160*)
- Memory (see *page 160*)
- Settings (see *page 161*)

Many of these pages are available only if your phone is in Administrator mode.

Accessing the WebUI

1. Ensure that your computer is connected to the same network as the ET605. Your computer may already be connected to the network through the PC port on the back of the ET605.
2. Find the IP address of the ET605:
   - Press > **6 Information** > **2 System Info**.

   The phone displays the system info.



3. On your computer, open a web browser. (Depending on your browser, some of the pages presented here may look different and have different controls. Ensure that you are running the latest update of your preferred web browser).
4. Type the ET605 IP address, preceded by "http://" or "https://" in the web browser address bar (for example: http://192.168.10.115) and press **ENTER** on your computer keyboard.

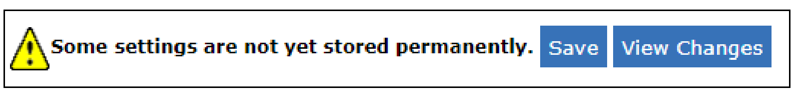
   The browser displays a window asking for your user name and password.
5. Enter your HTTP user name and password, if requested.
   You can set the user name and password later on the WebUI: **Advanced** > **Qos/Security** page > **HTTP Server**.
6. Click **OK**.

   The WebUI appears.
7. Click topics from the navigation bar on the left of the WebUI, and then click the links along the top to view individual pages.

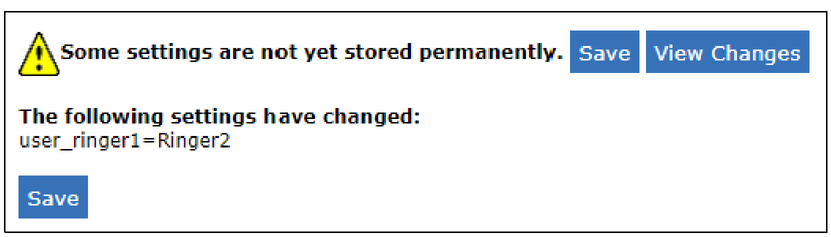
Changing settings in the WebUI

When you make changes to the phone's settings on the WebUI pages, click the **Apply** button to apply your changes.

If the WebUI displays the following message at the top of the page, it means you have not yet saved your changes to the phone.

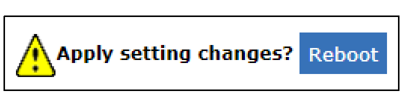


- Click the **View Changes** button to display what changes need to be saved.



- Click the **Save** button to save your changes to the phone.

Some changes to settings require the phone to be rebooted. The WebUI displays the following message.



- Click the **Reboot** button to reboot your phone.

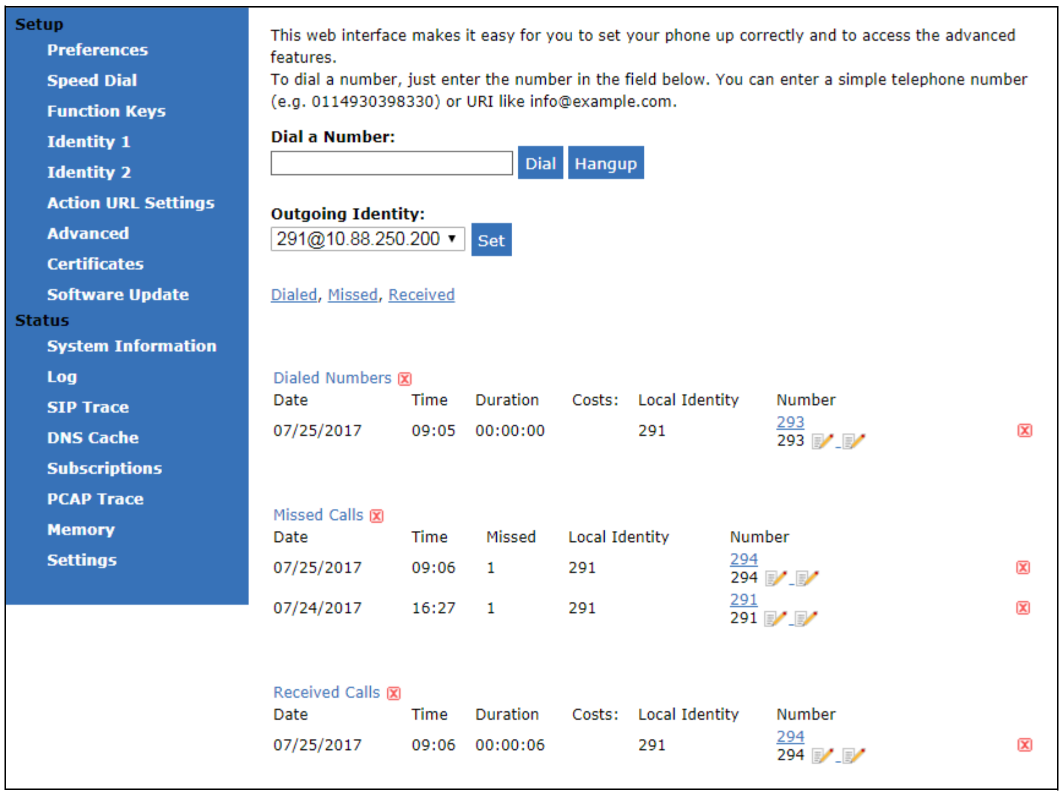
Operation pages

The Operation pages display information about the operation of your phone:

- Dialing a number
- Displaying Call History of dialed numbers, missed calls, and received calls.

Home page

The Home page enables you to dial a number or Uniform Resource identifier (URI) on your ET605 Deskset, and also displays call history of dialed numbers, missed calls, and received calls.



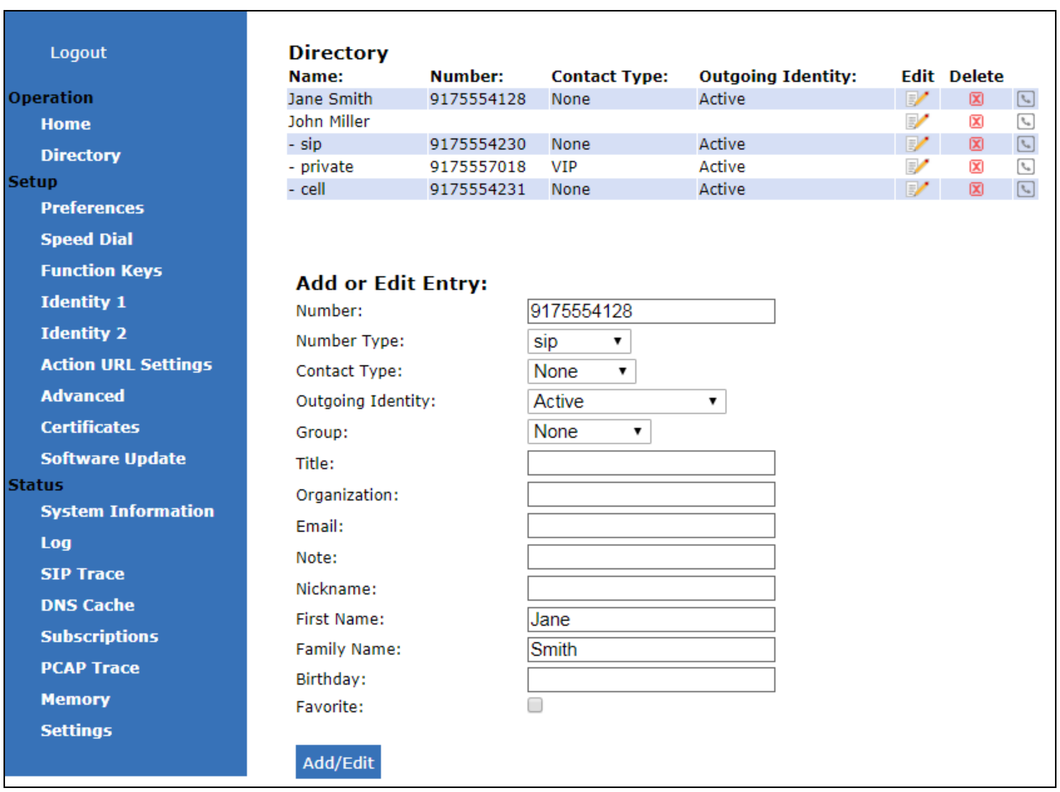
Setting	Description
Dial a number	Enter a phone number/SIP URI/IP address you want to call from your phone.
Outgoing Identity	Choose the outgoing identity of the number you want to call, and then click the **Set** button.

Setting	Description
	Dial button - Click to dial the number on your phone. The phone calls the number on the speakerphone. You can lift the handset or press the headset button on your phone. **Hangup** button - Click to disconnect the call. **Set** button - Click to set the Outgoing Identity.
	Dialed hyperlink - Go to the **Dialed Numbers** area of the page. **Missed** hyperlink - Go to the **Missed Calls** area of the page. **Received** hyperlink - Go to the **Received Calls** area of the page.
Dialed Numbers	This area of the page displays the call history of recent calls dialed from your phone. It shows date, time, and duration of the call as well as Local Identity and Number. Local Identity is the phone's outgoing identity chosen for the call, and Number is the phone number. ■ Click  next to **Dialed Numbers** to delete all entries. ■ Click  next to a line to delete the line. ■ Click the 1st  to add/edit the number in the Directory. ■ Click the 2nd  to add/edit the URI in the Directory.
Missed Calls	This area of the page displays the call history of recent calls missed by your phone. It shows date and time of the call as well as Missed, Local Identity, and Number. Local Identity is the phone's outgoing identity called by the phone number listed under Number, and Missed shows the number of times calls by this phone number were missed. ■ Click  next to **Missed Calls** to delete all entries. ■ Click  next to a line to delete the line. ■ Click the 1st  to add/edit the number in the Directory. ■ Click the 2nd  to add/edit the URI in the Directory.

Setting	Description
Received Calls	This area of the page shows the call history of calls received by your phone. It shows date, time, and duration of the call as well as Local Identity and Number. Local Identity is the phone's outgoing identity which received the call, and Number is the phone number it was received from. ■ Click next to **Received Calls** to delete all entries. ■ Click next to a line to delete the line. ■ Click the 1st to add/edit the number in the Directory. ■ Click the 2nd to add/edit the URI in the Directory.

Directory page

On the Local Directory page, you can manage your local directory entries. You can edit, delete, and add contact information for up to 1,000 entries. In order to back up your contacts or import another local directory file, the page also enables you to export and import your phone's local directory.



Setting	Description
Directory:	This area of the screen displays the entries in your phone's directory. ■ Click to edit the entry. ■ Click to delete the entry. ■ Click call the number on your phone.
Add or Edit Entry:	Displays information about the directory entry you are adding or editing.
Number	The person's phone number
Number Type	The number type - sip, cell, fixed, private, or business.
Contact Type	The contact type: ■ None ■ VIP - Enables calls from the number, even if Do Not Disturb (DND) is turned on. ■ Deny List - Blocks calls from the number, but the caller can still leave a voicemail message.
Outgoing Identity	The outgoing identity for this person's directory entry.
Group	A group in which the person belongs - None, Friends, Family, Work, Colleagues.
Title	The person's company title. For example, Head of Finances.
Organization	The organization/company for which the person works.
Email	The person's email address.
Note	A note about the person.
Nickname	The person's nickname
First Name	The person's first name
Family Name	The person's family name
Birthday	The person's birthday in either dd.mm.yyyy or mm/dd/yyyy format
Favorite	Marks the person as favorite

Setting	Description
	Add/Edit button - Click to either add a new entry, or save your changes to the currently selected entry. **Add Sub** button - Click to add a directory sub-entry. **Change** button - Click to save your changes to the currently selected entry.
Import directory (CSV):	This area of the screen enables you to import directory entries from a Comma-Separated Value (CSV) file.
Load from file:	
Filename	Select the file you want to upload.
Filetype	Select the format of the file - CSV format or Unicode TAB-separated.
Skip first Line	Select “on” to skip the first line of the import file, such as a heading line that describes field names.
	Load button - Click to import the file. The WebUI displays an import preview. • To delete your phone’s existing directory, select “on” for **Delete whole directory before**. • Make any required changes to the import field names, and click **Save**.
Delete whole directory	**Delete** button - Click to delete your phone’s directory. The WebUI displays a warning message asking if you really want to delete. Click the **Yes** or **No** button.
Click here to save the current directory.	Click the link to display the directory in CSV format. Right-click to save in your web browser.
Click here to save the current directory in XML format.	Click the link to display the directory in XML format. Right-click to save in your web browser.

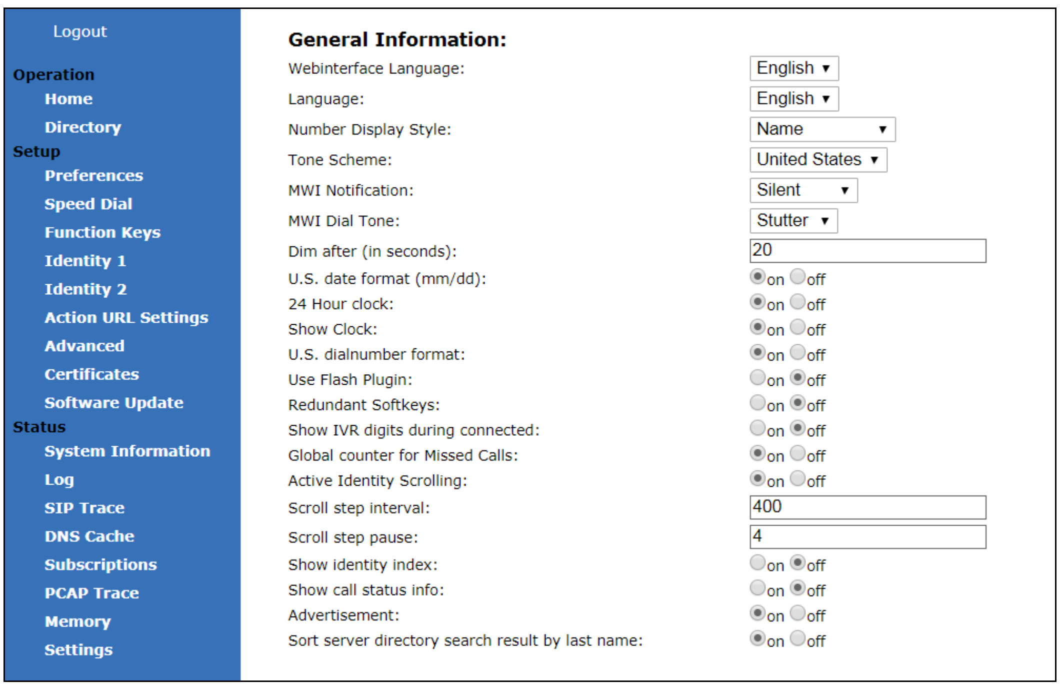
Setup pages

The Setup pages of the WebUI are for the setup and configuration of your phone:

- Setting phone preferences
- Assigning speed dial numbers
- Setting function keys
- Settings for Identities (accounts), Action URLs, and Advanced features
- Installing certificates
- Updating the phone's software

Preferences page

On the Preferences page, you can configure some basic settings for the phone and set hold ringtone, privacy, and keyboard settings. The Preferences page is also available to phone users when they log on to the WebUI in user mode.



Setting	Description
General Information:	
Webinterface Language	Select a language used on the Web User Interface (WebUI). This may be different from the language currently used on the phone.
Language	Select the language used on the Phone User Interface of your phone.

Setting	Description
Number Display Style	Specifies how incoming and outgoing calls are displayed: ▪ Full Contact: The complete URL is shown ▪ Name: Only the name is displayed ▪ Number: Only the number is displayed ▪ Name+Number: Name and number are displayed ▪ Number+Name: Number and name are displayed
Tone Scheme	Select the dialtone you prefer for your phone. Also, DTMF echo will differ (on/off) on different schemes.
MWI Notification	Specify the type of Message Waiting Indicator (MWI) notification that will inform you when a new message arrives. A short beep <beep> is reminding you once on having mailbox messages waiting in which <reminder> is doing that repeatedly. With <silent> this functionality can be switched off.
MWI Dial Tone	Your phone's dialtone can be changed into a stuttering tone <stutter> to remind you on having waiting mailbox messages. With <normal> this functionality can be switched off.
Dim after (in seconds)	Number of seconds after which to dim (phones with color display) or turn off the display backlight when nothing is happening.
U.S. date format (mm/dd)	With this setting, you can select either U.S. (month/day) or European (day.month) format for displaying the time and date.
24 Hour clock	When you select “on”, the timestamps will be formatted in 24-hour format, otherwise in 12-hour format.
Show Clock	Specifies whether or not clock and date should be displayed (at the idle screen usually). If <false>, phone name is displayed instead (if set).

Setting	Description
U.S. dialnumber format	When this setting is on AND the phone is set to a US time zone, any numbers you dial will be formatted on the display like the following examples: 1. National format: 9785550123 will be shown as (978) 555-0123; formatting will start when the 4th digit is entered. 2. Service numbers (depending on availability in your area): A service number beginning with 511, for example, will be shown as (511) -xxxx; formatting will start when the 4th digit is entered. 3. International access code (for dialing numbers outside NANP): Numbers beginning with the international access code 011 will be shown as 011-x-xxxxxx. Formatting will start when the 4th digit is entered; the country dialing code (the digit(s) enclosed by the two hyphens) can consist of one or more digits. Examples: ■ After you have entered the four digits 0114, the display will show them as 011-4. ■ Entering 9 as a fifth digit will result in 011-49- because 49 is an existing country dialing code (Germany). ■ Entering 2 as a fifth digit will result in 011-42 without the second hyphen because there is no 42 country dialing code; entering 0 as the sixth digit will result in 011-420- because 420 is an existing country dialing code (Czech Republic). Note: U.S. dialnumber format is the default setting, but will only be activated when the selected time zone on the phone is a US time zone.
Use Flash Plugin	If you want to have a live reaction on incoming or outgoing calls on the phone's "Home" page, switch this option to "on". Your web browser has to support the Macromedia flash movie format.
Redundant Softkeys	When showing a list in minibrowser while the minibrowser-xml does not define any context-keys on its own: this setting decides if to show navi-keys instead or no keys at all.

Setting	Description
Show IVR digits during connected	This setting controls whether digits pressed during a connected call are shown on the display or not. These digits are usually used to control IVR prompts and to enter user specific information e.g. calling card number, pin codes, credit card number, billing info etc. Turning this setting off ensures privacy by disabling the display of these digits. The actual keys are either not shown at all or replaced replaced by *.
Global counter for Missed Calls	When set to <on>, the phone will count missed calls on all registered lines and show them on the phone. If turned <off>, missed calls for the active identity will be shown on the display.
Active Identity Scrolling	Turn on/off active line scrolling using navigation key in idle state.
Scroll step interval	Time in ms to make the next step for text scrolling.
Scroll step pause	The setting describes for how many scroll-steps the scrolling is paused when its beginning of a scrolling text is shown. For phones that don't use circle-scroll-technique, but instead scroll to the end and then start up front again, this stop-time also describes the pause at the end.
Show identity index	Shows local sip line index during call states in adition to the remote user display name/number/url
Show call status info	if turned on the call progress is shown in the headline of the call progress window e.g. (100 Trying, 180 Ringing etc).
Advertisement	This setting distinguishes whether an Advertisement page is displayed on the VTech phone WebUI home page. This setting is related to the setting advertisement_url.
Sort server directory search result by last name	When set to 'on', the results returned from an on-line telephone directory search will be sorted by Last Name (Surname) then First Name (Given Name). When set to 'off', the results will be sorted by First Name (Given Name) then Last Name (Surname). If the record does not include a Last Name, the Display Name is used instead.
Ringtone defaults:	
Higher Ringer Volume	In loud environments, the ringer might not be loud enough. With this setting, you can digitally increase the ringer. A side-effect might be that a ringer sounds distorted on maximal volume. Please enable this feature only if it is really necessary.

Setting	Description
Ringer Device for Headset	If you want to hear the ring tone via the headset only, choose “headset”; otherwise, “speaker”. Both headset and speaker can be enabled. Then the configured ring tone will be played on the speaker of the phone and the headset plays it's own build in ring tone (e.g. three short beeps). Some headsets don't have a build in ring tone (most wired USB headsets). But some of them can give a visual indication.
Alert-Info Ringer:	
Alert Internal Text	Text which can be specified in Alert-Info to categorize an internal number.
Alert Internal Ringer	Melody to be played back on Alert Internal.
Alert External Text	Text which can be specified in Alert-Info to categorize the an external number.
Alert External Ringer	Melody to be played back on Alert External.
Alert Group Text	Text which can be specified in Alert-Info to categorize a group number.
Alert Group Ringer	Melody to be played back on Alert Group.
Directory Ringtones:	
Friends, Family, Colleagues, Work, VIP	Phone book contact type specific ringers. Specify the ringing melodies for different contact types of your personal directory entries (e.g., “friends”).
Custom Melody URL	If you have chosen Custom Melody URL in one of the pull-down menus, you can specify an URL to your own ringing melody. The type of file that should be supplied to the phone is: “PCM 8 kHz 16 bit/sample (linear) mono WAV”.
Customised Alert-Info using built-in melodies:	
Internal Ringer Text (0-10)	Text which can be specified in Alert-Info to categorize a specific ringtone melody.
Internal Ringer File (0-10)	Melody to be played back on the Internal Ringer Text.
Auto Answer:	
Auto Answer Indication	If you want to become informed with an audible indication when an incoming call (intercom call too) is automatically answered by your phone, select “on”.
Privacy Settings:	

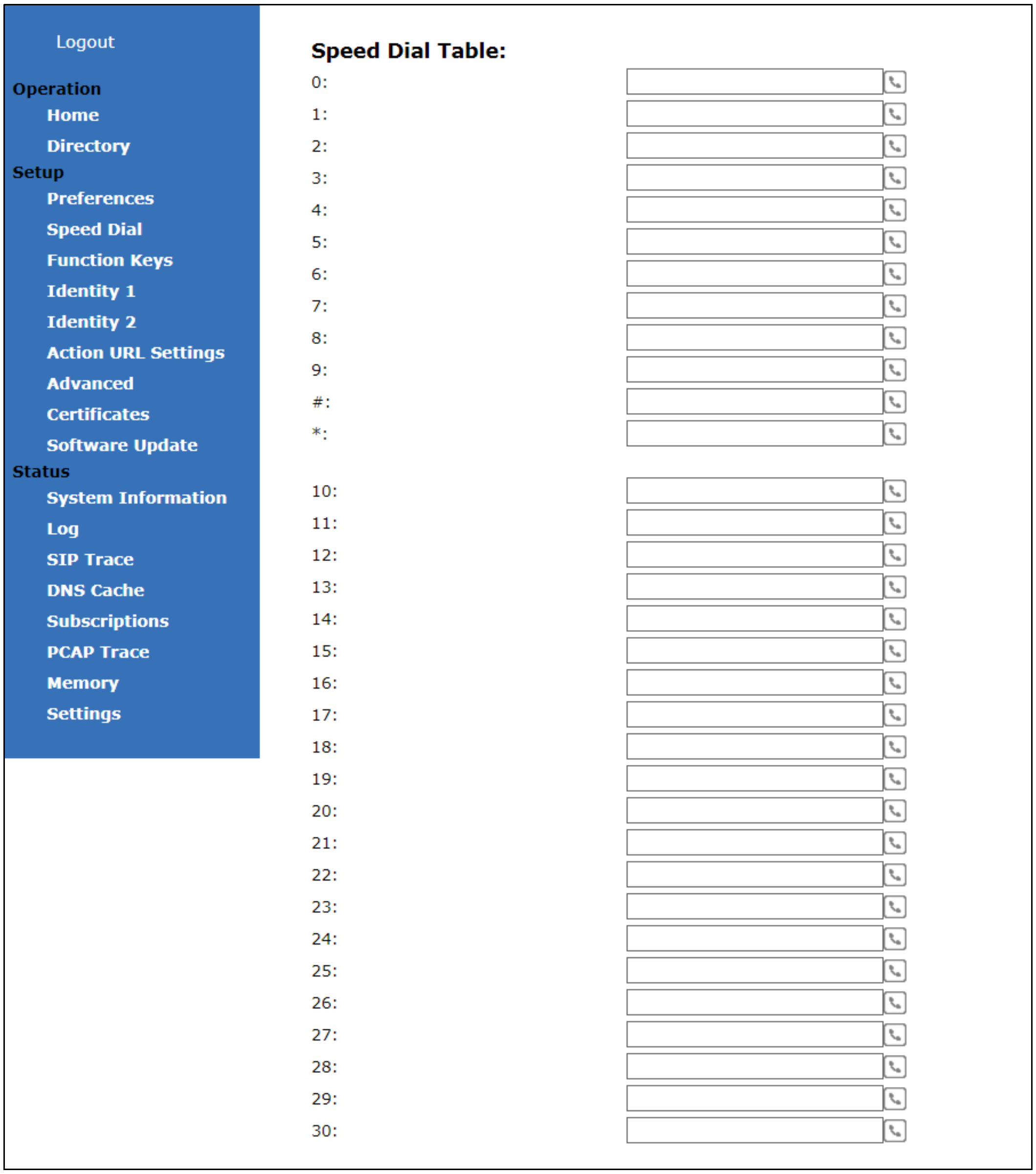
Setting	Description
Suppress own number (CLIP/CLIR)	Show or hide your own phone number on outgoing call.
Reject incoming anonymous calls	Reject or accept anonymous incoming calls.
Presence Inactivity Timeout (in minutes)	The time in min after which, if there is no activity, presence is set to “closed”. NOTE: If it is set to 0, the presence stays closed and nothing is published at all i.e. presence is disabled for all practical purposes.
Lock Keyboard:	
Allow keyboard locking	Enable keyboard locking via star-key or timeout. On OCS servers this setting is turned on if the inband provisioning parameter ucEnforcePinLock has a value of true. If its value is false this setting is left unchanged (i.e. it may be turned on or off at the user's discretion). Note that even when this setting is turned off, the user can still lock/unlock the phone via the web interface directly by changing the phone's lock state (see keyboard_lock).
Keyboard lock	By setting this option to 'on' the phone's keyboard will be locked. On the phone the keyboard can be locked/unlocked by pressing the star key for a few seconds (if enable_keyboard_lock is 'on'). This setting represents the current lock state of the phone. Therefore changing it can be used to lock or unlock the phone from the web interface regardless of whether the enable_keyboard_lock is on or off.
PIN to lock/unlock	The locked keyboard can be unlocked only by typing in the specified PIN. If this is empty, no PIN is needed to unlock the keyboard.
Lock Keyboard after sec. (0 = never)	This setting allows you to configure an inactivity timer (in seconds). If enable_keyboard_lock is set to on, the phone will automatically lock the keypad after the configured inactivity time. The user would then need to enter the configured PIN in order to unlock the keypad. On OCS servers this setting is provisioned via inband provisioning parameter ucPhoneTimeOut.
Emergency Numbers (space separated)	The specified space separated numbers can be dialled via keyboard even if the keyboard lock is enabled. Just dial them as usual without unlocking the keyboard before.
Outbound proxy for emergency numbers	Outbound proxy for emergency numbers.
Character Settings:	

Setting	Description
upper case char.sequence key (0-9)	Define which characters should appear one after another each time numpad key INDEX (INDEX ranges from 0 to 9) gets pressed (when input mode is set to upper case letters).
lower case char.sequence key (0-9)	Define which characters should appear one after another each time numpad key INDEX (INDEX ranges from 0 to 9) gets pressed (when input mode is set to lower case letters).

Speed Dial page

On the Speed Dial page, you can enter up to 32 speed dial numbers, which enable you to make a call without having to enter the complete phone number.

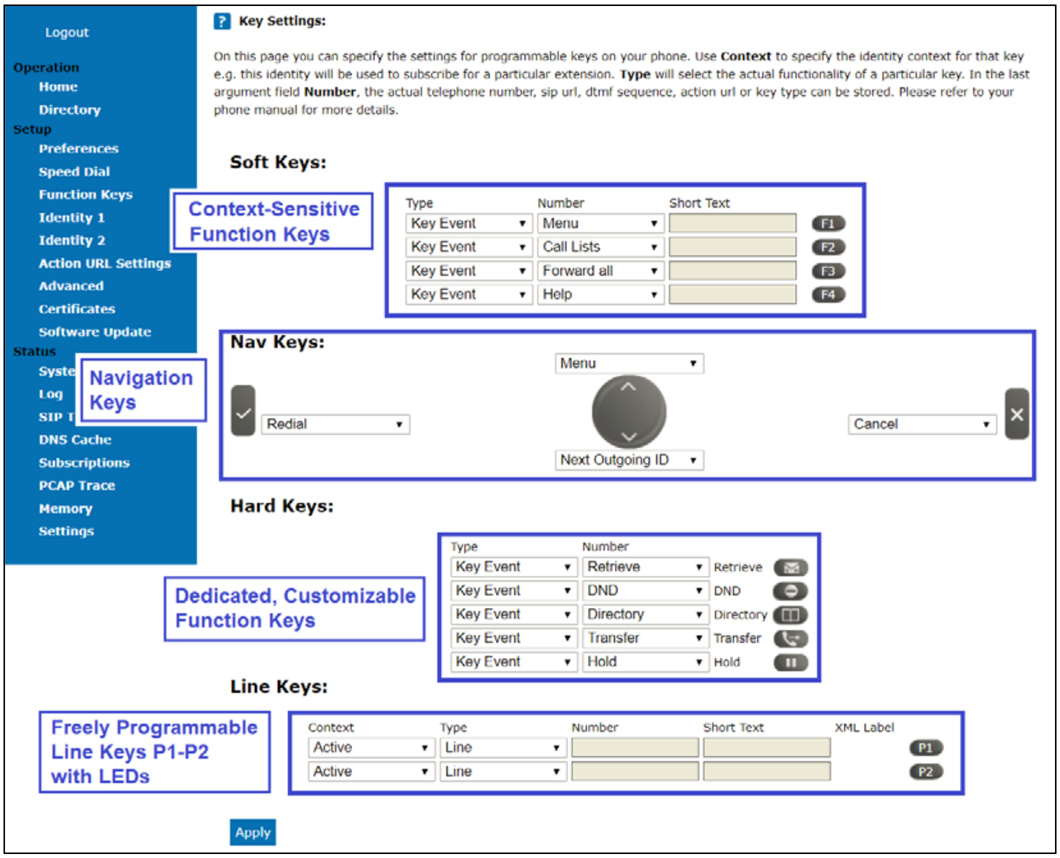
To dial a speed dial number, enter the speed dial number (0 to 30) or character (#, *) assigned to the phone number, and then press .



Setting	Description
0 to 9	Speed dial items 0-9 specifies the number which may be called via keys 0-9.
#	Speed dial item # specifies the number which may be called via key #.
*	Speed dial item * specifies the number which may be called via key *.
10 to 30	Speed dial items 10-30 specifies the number which may be called via numbers 10-30.

Function Keys page

On this page, you can specify the settings for programmable keys on your phone. Use **Context** to specify the identity context for that key; e.g. this identity will be used to subscribe for a particular extension. **Type** will select the actual functionality of a particular key. In the last argument field **Number**, the actual telephone number, SIP URL, DTMF sequence, action URL or key type can be stored. Please refer to your phone manual for more details.



For Freely Programmable LED line keys P1–P2:

- **Context:** The default setting is **<Active>**, i.e., the functionality chosen under **Type** will be applied to any currently active extension (SIP identity) for outgoing calls. If a specific extension (SIP identity) is chosen from the pull down menu, the functionality under **Type** will be applied only to the chosen extension (SIP identity).
- **Type:** The default setting is **<Line>**. When another setting is selected from the pull down menu **Types**, that functionality will be applied to the extension (SIP identity) chosen as **Context**.
- **Number:** The default setting is **<blank>**. You can enter a number / HTTP(S) URL / SIP URI as required by Type.

Type

The following table lists the available selections for **Type**.

Type	Description
Action URL	Action URLs are basically HTTP GET Requests. They can be used to send various data from the phone to a web server, like: ▪ usual settings stored on the phone. ▪ private settings e.g. passwords are replaced by empty strings ▪ $local for local URI (=own identity replaced at run-time) ▪ $remote for remote URI (=inbound/outbound caller ID replaced at run-time) ▪ $call-id for the current call ID (replaced at run-time) It is possible to configure two URLs per key, the first being triggered when the key is pressed, the second when the key is released. To configure two URLs, just separate them with a "|" character, for example "http://192.168.10.10/press.html|http://192.168.10.11/release.html"
Auto Answer	Press the key to enable or disable the auto answering of calls
BLF	The free function key types "Extension" and "BLF" allow users to monitor the dialog state of another phone/user extension. This is indicated by the LEDs adjacent to the particular key. This feature is called "Busy Lamp Field".
Button	This is a button that is connected to your PBX.
Call Agent	The phone can be used as a Call Agent that distinguishes five states: ▪ AgentLoggedOnEvent (Sign-In) ▪ AgentLoggedOffEvent (Sign-Out) ▪ AgentNotReadyEvent (Unavailable) ▪ AgentReadyEvent (Available) ▪ AgentWorkingAfterCallEvent (Wrap-Up) These states are governed by the function key ACD, which is configured in the Function Keys section of the webinterface.

Type	Description
Conference Server	This function key can be used for PBX-based conferences and for local conferences on the phone itself. ■ PBX-based conferences. When a conference room or conference account has been created on the server for an individual identity, you can dedicate a function key to calling and monitoring the conference room. Select the identity and the "Conference server" function from the respective drop-down menus and enter the SIP URI of the conference room in the "Number" text field. For information on how to use this key with your particular PBX, please check the PBX manual. ■ Phone-based conferences. If there is no SIP URI in the text field, pressing the function key will initiate a phone-based conference with all held calls and any active call.
Contact Presence (XMPP)	This feature allows you to publish a presence state to indicate your current communication status in order to inform your contacts of your availability and willingness to communicate.
DTMF	This option allows the specification of arbitrary key sequences (allowed digits: "0-9", "*", "#", "A-D" and flash: "!"), which will be sent via DTMF when this button is pressed. This can only be done during an active call.
Extension	This key can be used for: ■ Extension Monitoring (Busy Lamp Field (BLF)) & Call Pickup: This allows showing the status (idle, ringing, held call, busy) of a distinct phone extension on your phone ■ Speed Dial: Pressing this key during idle state will dial the programmed extension ("number"). ■ Call Deflection: Pressing this key during an incoming call will deflect the incoming call to the programmed extension ("number"). **Context:** can be assigned to any local SIP identity (account, registration, line) which had successfully registered at the same SIP domain. **Type:** extension (destination) **Number:** has to be assigned to the remote phone extension. Use the SIP URI format: extension@SIPdomain here.

Type	Description
Forward to	Press the key to enable or disable the forwarding of calls to the specified extension.
Intercom	Pressing the key bound to “Intercom” enables the intercom mode: the phone will be directly connected to the VTech phone if authentication is set up properly. This feature is useful in an office environment as a quick access key to connect to the operator or the secretary.
IVR	The argument is a number that is dialed on key press i.e. sending out an INVITE. Once the call has been established, pressing the same IVR key would send out dtmf digits comprising that number. This can be used to control IVR applications by one key only.
Key Event	Built-in key events may be mapped onto the predefined or the usual function keys. For a list of key events, see *“Key Events” on page 97*.

<table>
<tr><th>Type</th><th>Description</th></tr>
<tr><td>Line</td><td>“Line” key can behave as a private line or shared line key, according to the setting user_shared_line.

Private Line (user_shared_line = “off”):

This key can be used for:
SIP Identity Mapping:

This allows the customer to use different SIP identities (accounts, registrations, lines) similar as having several PSTN phone lines. Local SIP identities (lines) can be assigned to programmable keys from the list as Context via key Type "Line".
Free Key:

Line is also the default setting for the Freely Programmable LED Line Keys P1–P2. If no argument is set, the keys are treated as free. Outgoing and incoming calls not bound to any other key go to the first such key that is not already occupied.
Shared Line (user_shared_line = “on”):

The Bridged Line Appearance (BLA) feature allows subscribers to share SIP lines and also provides status monitoring of the shared line. The BLA feature is commonly offered in the IP Centrex services and IP-PBX offerings.

When a user places an outgoing call using such an appearance, all members belonging to that particular BLA group are notified of this usage, and are blocked from using this line appearance until the line goes back to idle state or if the call is placed on hold. Similarly, all members of the BLA group are notified of an incoming call and the call can be picked up on a line appearance associated with the BLA extension.

BLA members can monitor the status of the bridged line via the Function keys available on the VTech phones. For monitoring the status of a bridged line, the function key must be configured as a “Line” type. In addition, the “Number” must be set to the bridged line resource URI, and the “Context” must be set to a specific identity (not “active”). Once the phone has registered and subscribed successfully for the BLA resource, the LED corresponding to the programmed function key indicates the status of the bridged line. LED “on” indicates the line is in use, while LED “off” indicates an idle status.</td></tr>
</table>

Type	Description
Multicast Page	Supports paging via multicast IP. Set up the function key to generate a multicast stream.
Park+Orbit	This feature is useful for call center environments and all places where there is a great inflow of calls and some kind of queuing is required to manage them. Some PBX solutions provide its customers with the opportunity to set up parking orbits, where calls can be parked and picked up. The option “Park Orbit” enables the phone to provide this feature.
Presence	The phone will subscribe to the presence state of the destination URL with event type presence. The associated led will reflect the presence state of the destination e.g. ringing, available etc. Hitting the programmable key (usually when the destination is available and can receive a call) shall dial that number.
Push2Talk	Just like the Intercom option, the 'Push2Talk' feature enables users to make Intercom calls to a programmed destination via the function keys. This feature differs from the 'Intercom' option only in the sense that for this feature the intercom call will remain active as long as the programmed key is kept pressed. The call will be released as soon as the the 'Push2Talk' programmed key is released. This feature is particularly useful for group annoncements.
SendSipInfo	Send SIP INFO while call is connected. Message contains a "generic_value" header field with custom content.
Speed Dial	This key type behaves as a shortcut to a preset number the user may want to dial. In opposite to key type extension/destination, this key type does not subscribe to Dialog State changes. It is designed to speed up dialing numbers often used or hard to remember. A DTMF sequence can be appended that is dialed once the call has been established. A Comma represents a pause of one second. Normally, the number is dialed immediately after the function key is pressed. In some circumstances, this behaviour is not desired. e.g. if you place a prefix on the function key. In this case, pass number=incomplete as an argument.
Starcode	Making SIP calls without audiovisual indication on the phone user interface (PUI). ■ Select Starcode from the **Type** drop-down menu of the function key. ■ Enter the phone number, star code number, or SIP URI in the **Number** text field of the function key.

Type	Description
Transfer to	Press the key to transfer a call to the specified extension.
UserInputAndSendSipInfo	Send SIP INFO while call is connected. Message contains a "generic_value" header field with custom string. User will be prompted to input the custom string when key is pressed.
Voice+Recorder	This feature can be used to record a conversation during an active call or short messages or memos for personal use. Another possible usage is the recording of a debate or discussion, to keep audio minutes of a meeting, or to record a conference. This option can be set up with a valid voice recording account.
Xml Definition	XML Definition/Customizable via XML.
None	If you want to map a key to no functionality at all, use this type.

Key Events

The following table lists the available selections for Type **Key Event**.

Key Event	Description
Accepted Calls	(Accepted List) List of calls accepted on the phone.
Call Lists	Call history list (missed, received, dialed calls).
Conference	Enables the user to press the key to set up a conference call and select desired participants.
Contacts	Contact List, where the Presence State of selected users can be seen (online, busy, offline).
Delete Message	Deletes a text message.
Deny All	This key event will deny the incoming call and add the number to the deny list. All phones with call screen settings can alternatively do this by long-pressing the cancel key.
Directory	Internal phone directory.
DND	Turn "Do not disturb" function (DND) on an off.
Favorites	Favorites list.
Forward all	Forward all incoming calls to another extension or an external phone number.
Headset	Turn Headset mode on/off.
Help	Displays the URL of the phone's web interface and the URL to the web page.
Hold	Places an active call on "Hold".

Key Event	Description
Hold Private	Places an active call on "Private Hold".
Hoteling	Hoteling feature enables users (guests) within an office to use any cubicle phone (hosts) in the office by logging in to the host phone and having the host phone provisioned with guest's device profile settings.
Labels Backward	Opens the previous label page in a round-robin fashion on phones with self-labeling keys.
Labels Forward	Opens the next label page in a round-robin fashion on phones with self-labeling keys.
LDAP Directory	Enables the user to look up a remote directory while dialing.
Logoff Identities	Caution: This option will delete all account settings!! Usage: Mainly useful for call centers with frequently changing users.
Menu	Call up the settings menu of the phone.
Missed Calls	Missed call history list.
Monitor Calls	Show the list of monitored extensions active extensions that are active (i.e., busy or ringing). When there is no activity on any monitored extensions, the list is empty.
Multicast Zones	Multicast paging zones.
Multicast zones	Multicast paging zones.
Mute	Description: Mutes/Unmutes during an active call. Please note that on some phones the mute key can work as a DND when Idle. You can manage this feature through the **mute_is_dnd_in_idle** setting.
Next Outgoing ID	Select the next identity as the outgoing identity.
OCIP	Access the Broadsoft directory via the Open Client Interface-Provisioning (OCI-P) that allows third-party applications to peform all business functions peformed by BroadWorks.
Presence State	Provide access to a list where the Presence state of each registered SIP Identity can be defined (online, offline, busy, invisible).
Prev. Outgoing ID	Select the previous identity as the outgoing identity.
Reboot	Displays a screen on the phone asking if you want to reboot.
Record	Toggle recording on/off during an active call.
Redial	Dialed call history list (last call at the top).

Key Event	Description
Retrieve	Retrieves new mailbox messages. This key becomes active when the phone has received a message waiting indication (MWI) with a valid mailbox URI.
Ringer Silent	Turns the ringer off/on.
Server Directory	Provides access to an external phone directory.
Status messages	Display the currently available status messages.
Transfer	Transfers the current incoming/active call.
None	No function selected.

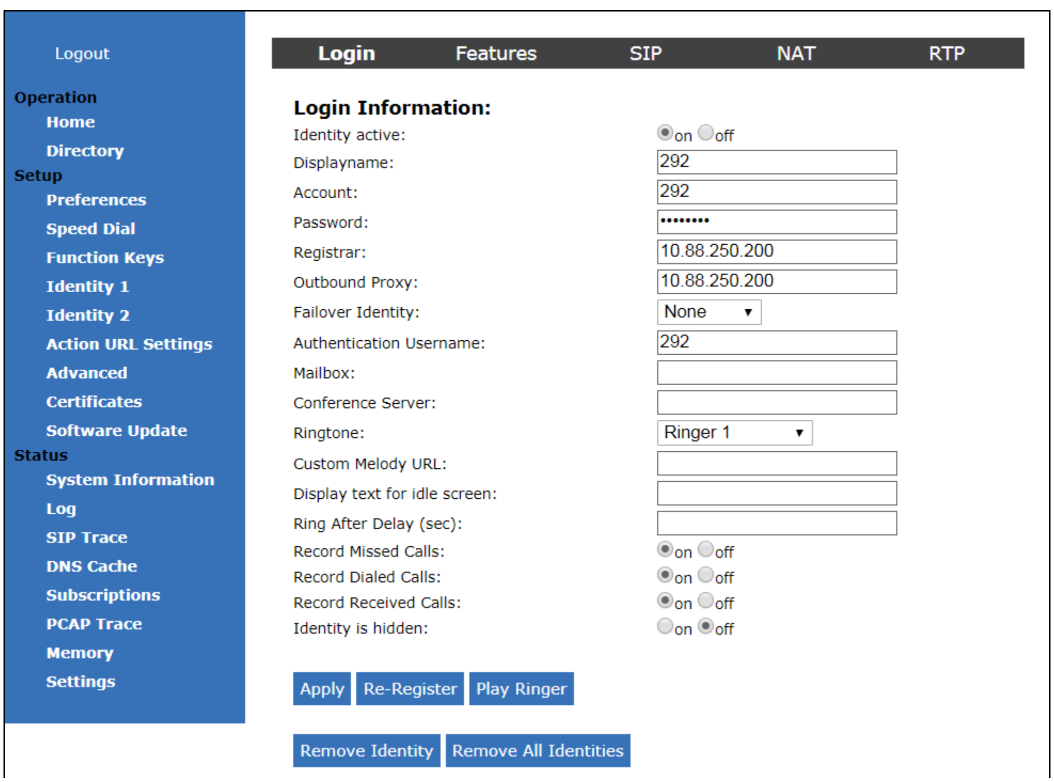
Identity n page

On the Identity n page, you can configure each identity (account) you have ordered from your service provider. You can configure up to 2 identities on the ET605 Deskset.

The WebUI pages are labeled Identity 1 and Identity 2, respectively. Each page has five tabs for configuring settings specific to the currently selected identity - Login, Features, SIP, NAT, and RTP. When you click the Identity n page, the Login tab is automatically selected.

Login tab

With the Login tab, you can add or remove an identity for the phone. You can enter information about your account, password, registrar, outbound proxy, and mailbox.

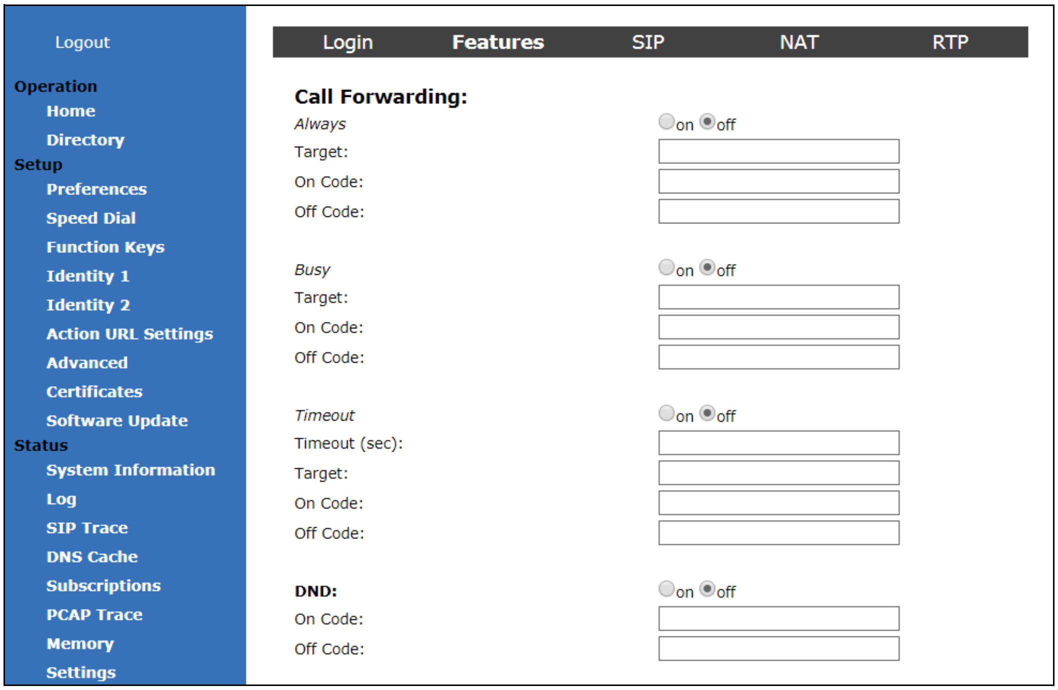


Setting	Description
Identity active	This identity can be disabled by disabling this option. This means this identity is not longer registered anymore.
Displayname	Set the name you would like to associate with each line. For example, “John Smith”. This information is also sent out to any party you are calling. Only the first 50 characters are used (when entering more than 50 characters).
Account	This is the account with which you register to a SIP registrar/proxy. It could be alphanumeric, for example: “js”, or based on digits like “445”. See also **Authentication Username**.
Password	This is the password to be used for challenge responses. In order to protect them from unauthorized use, passwords are not displayed in their true form, but as a series of asterisks.
Registrar	Specify the IP or DNS address of the registrar/proxy where you want to register this account. After a successful registration, the registrar knows how to reach this specific identity and can route requests (for example, incoming calls) from other registered parties to this phone.
Outbound Proxy	Specify the outbound proxy in this field (format: **addr:port**) to ensure all SIP packets are sent via the specified communication point.
Failover Identity	This identity will be used as a backup for failover. That is, if the current identity is not registered, this identity is used instead.
Authentication Username	Registrar environments may need different user names for registration and authentication. If user_pname is set, it is used for authentication and user_name is used for registration; otherwise **Account** is used for both.
Mailbox	If you have set up a mailbox, specify the account name for that mailbox here to associate it with this particular SIP identity. This is important for contacting your mailbox when the MWI message does not include the proper mailbox SIP URI.
Conference Server	Contains a sip-uri for a conference room. Used by pressing conference keys. This setting depends on an identity. If 'conference' key was pressed, the configured conference room of the active identity will be called. If no SIP-URI is configured, the default behaviour is a local conference on the phone (min. 2 participants connected).
Ringtone	Select a ring tone that will alert you when a call comes in for this particular identity.

Setting	Description
Custom Melody URL	Specify an URL to your own ringing melody. The type of file that should be supplied to the phone is: „PCM 8 kHz 16 bit/sample (linear) mono WAV”. This only has an effect when you have chosen “Custom Melody” from the “Ringtone” pull-down menu and when the incoming call matches this SIP identity.
Display text for idle screen	If you enter a name in this field, then this name will be shown on the idle screen associated with this particular line instead of the name you have entered in the **Displayname** field, if any. This information is not sent out to anyone, but is merely shown on the phone’s display for your information.
Ring After Delay (sec)	The phone delays playing the ringer for the given amount of seconds. But the message LED still rings from the beginning.
Record Missed Calls	Should be disabled, if incoming calls to this identity should not be taken into account for the number of missed calls. Also see record missed calls when cwi is off, sip cancel reasons to ignore missed call, ignore missed calls on busy
Record Dialed Calls	Should be disabled, if dialed calls from this identity should not be taken into account for the dialed calls list.
Record Received Calls	Should be disabled, if received calls to this identity should not be taken into account for the received calls list.
Identity is hidden	Setting this to 'true' will make the identity disappear from the idle-screen. This setting depends onto is_voice_identity, when that setting is disabled, the identity will automatically be hidden.
	Apply button - Click to apply your changes to the fields on the page. **Re-Register** button - Click to re-register the identity. **Play Ringer** button - Click to play the ringtone on the phone. To stop ringing, open another WebUI page or press the Cancel button on the phone. **Remove Identity** button - Click to remove the currently displayed identity from the phone. **Remove All Identities** button - Click to remove all identities from the phone. The “VTECH Welcome!” screen appears on your phone display. You must press any button, and then enter the account, registrar, and SIP password to register an identity. For more information, see step 3 to 5 in *“Edit Identity (Hotdesking)” on page 58*.

Features tab

With the Features tab, you can configure settings for call forwarding and SIP service prodivers.



Setting	Description
Call Forwarding:	
Always	If turned on, all calls to the associated identity are diverted to the number specified by Target. Diversion can either be handled by the phone or by a server - see parameter using_server_managed_fwd_all.
Target	The redirection target, when redirection is always active (Always is set to on).
On Code	When set, the given starcode, appended by the redirection target, will be dialed whenever redirection-always gets enabled or changes the target for the specific identity.
Off Code	If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.

Setting	Description
Busy	If turned on and a call is in progress while a 2nd one is incoming, the second caller is diverted to the number specified (Target). Note: This will only work if call waiting is disabled. Diversion can either be handled by the phone or by a server - see parameter using_server_managed_fwd_busy.
Target	Specifies the number to which calls will be diverted when the phone is busy. Note: This will only work if call waiting is disabled (WebUI: Identity n > SIP > Call Waiting Indication).
On Code	When set, the given starcode, appended by the redirection target, will be dialed whenever redirection when busy gets enabled or changes the target for the specific identity.
Off Code	If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.
Timeout	If turned any incoming call will be diverted to the specified number (Target) after the specified time (Timeout) has elapsed. Diversion can either be handled by the phone or by a server - see parameter using_server_managed_fwd_time.
Timeout (sec)	Specifies the timeout in seconds after which the call will be diverted.
Target	Specifies the number to which calls will be diverted after the specified time (Timeout) has elapsed.
On Code	When set, the given starcode, appended by the redirection target, will be dialed whenever redirection after timeout gets enabled or changes the target for the specific identity.
Off Code	If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.
DND:	<on> means that the phone is in do not disturb (DND) mode, <off> is normal behavior.
On Code	If the PBX is handling DND, it can be specified which star code enables this functionality at the PBX.
Off Code	If the PBX is handling DND, it can be specified which star code disables this functionality at the PBX.
Server Managed:	

Setting	Description
Call Forwarding Always	If this setting is on the server will be responsible for handling the global forwarding functionality. From the call perspective, the phone will act as if no forwarding was set (all is managed by the server). The PUI will show the current setup if the server synchronizes it's values with those on the phone (Call Forwarding: Always and Target). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes). This setting does not specify how the server changes the value of parameter server_managed_dnd_state, nor how the phone updates them (it may be done via TR69).
Call Forwarding Busy	If this setting is on the server will be responsible for handling the redirect on busy functionality. From the call perspective, the phone will act as if no redirect was set (all is managed by the server). The PUI will show the current setup if the server synchronizes it's values with those on the phone (Call Forwarding: Busy and Target). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes). This setting does not specify how the server changes the value of parameters server_managed_fwd_busy_state and server_managed_fwd_busy_nr, nor how the phone updates them (it may be done via TR69).
Call Forwarding Timeout	If this setting is on the server will be responsible for handling the redirect on timeout functionality. From the call perspective, the phone will act as if no redirect was set (all is managed by the server). The PUI will show the current setup if the server synchronizes it's values with those on the phone (Call Forwarding: Timeout, Target and Timeout [sec.]). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes). This setting does not specify how the server changes the value of parameters server_managed_fwd_time_state, server_managed_fwd_time_nr and server_managed_fwd_time_secs, nor how the phone updates them (it may be done via TR69).

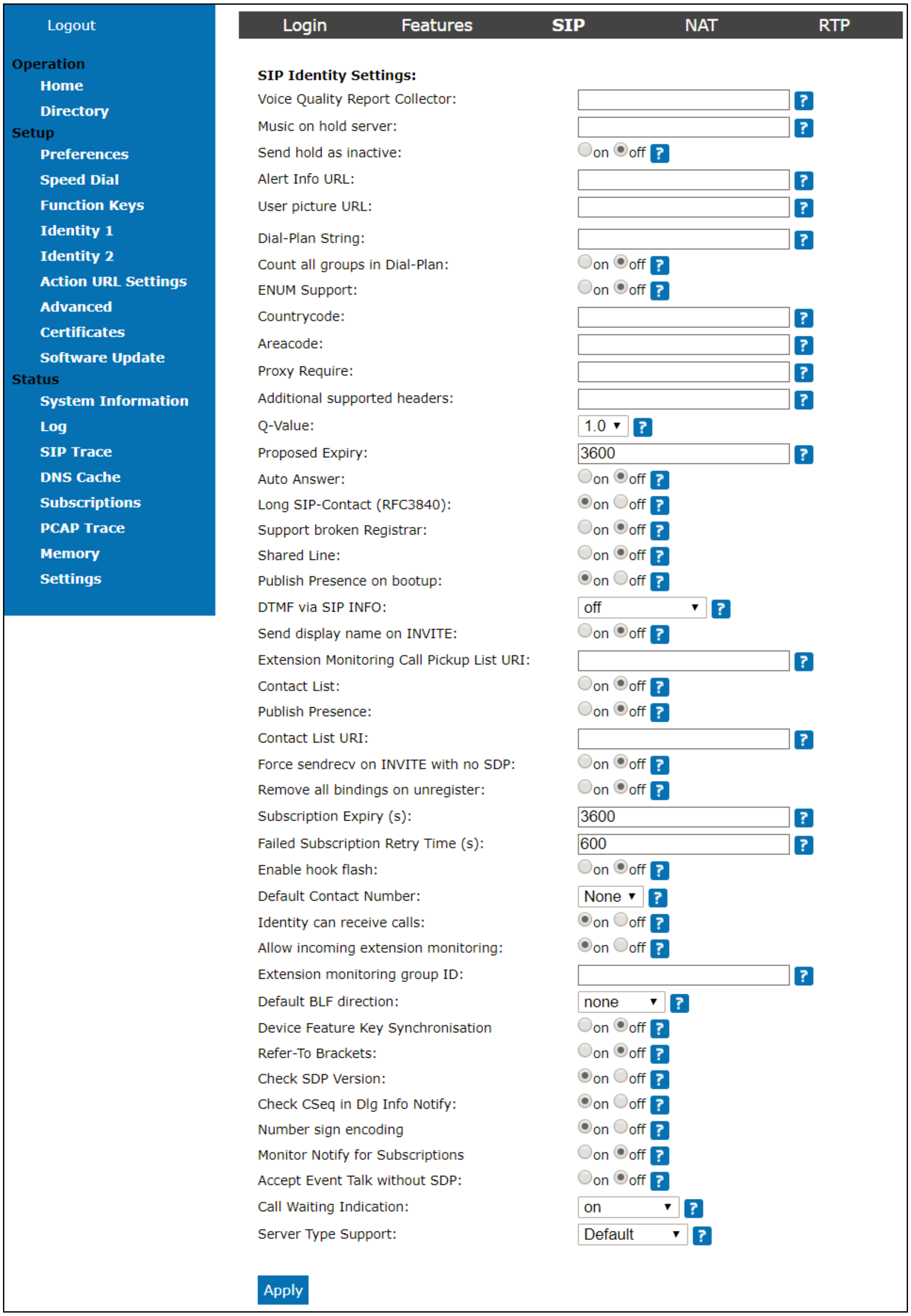
Setting	Description
DND	If this setting is on the server will be responsible for handling the DND(DO NOT DISTURB) functionality. From the call perspective, the phone will act as if no dnd was set (all is managed by the server). The phone user will see the value from DND:(on/off) as the current DND state, and this value can be changed at anytime by the server. This setting does not specify how the server changes the value of DND:(on/off), nor how the phone updates them (it may be done via TR69).
Call Logs	Specifies whether the call logs should be stored locally or on the server.
Directory Download:	
Phone Book Download Interval (Secs.)	Determines, in seconds, how much time should elapse before the phone initiates a Server Phonebook download. The interval is adjusted to a random value between 90 and 110 percent of the settings value. The interval time is capped at 1209600 seconds (= 14 days). If the setting is empty or contains an invalid value the download is never initiated. If the value is 0 the download is initiated exactly once after startup.
Server Directories (Download)	If the local telephone directory synchronization is to be limited to certain directories or groups these may be specified here, separated by a space. If left empty, all directories found will be downloaded.
Broadsoft Features:	
XSI Server	Specifies the Broadsoft XSI server
XSI User	The Broadsoft XSI account name.
XSI Password	The password of the Broadsoft XSI account.
XSI Retry Timer (Secs.)	If an error occurs during XSI session set up, this setting specifies after how many seconds the phone should retry setting up the XSI session (A value of zero means never).
XSI Events	Determines whether the phone should establish XSI event channels. Does not affect XSI Actions. For more information on XSI actions and events see Broadsoft's BroadWorks Interface Specification Xtended Services Interface.
XSI Action Polling Interval (Secs.)	Specifies the interval in seconds after which XSI action will be sent to retrieve related information from server.
XSI Conference Action Updating Interval (Secs.)	Controls how often the device polls the Broadsoft server for conference updates when idle.

Setting	Description
Server Directories (Search)	If the on-line telephone directory search is to be limited to certain directories or groups these may be specified here, separated by a space. If left empty, all directories found will be searched.
BLF Park Pick Up	Allows use different "Feature Access Codes" of service provider define to retrieve a parked call.
BLF Directed Call Picku	Allows use different "Feature Access Codes" of service provider define to directed call pickup.
Anywhere	Determines whether the phone should enable XSI Anywhere feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks Anywhere settings.
Visual Voicemail	This setting is used to enable / disable visual voicemail feature.
Call Center List	Determines whether the phone should enable XSI Call Center List feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks Call Center List settings.
Caller ID Blocking	If set to "on", outgoing caller ID blocking will be managed on Broadsoft server side through the use XSI. If set to "off", outgoing caller ID blocking will be managed locally.
Simultaneous Ring	Determines whether the phone should enable XSI Simultaneous Ringing feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's Simultaneous Ringing settings.
Remote Office	Determines whether the phone should enable XSI remote office feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks remote office settings.
Silent Alerting	Determines whether the phone should enable the Silent Alerting feature.
Hoteling	This setting enables and disables the Hoteling feature. The Hoteling feature enables a guest to login and use the host device.
Full Name Search	Determines whether the phone should perform a user's name search on both first and last name simultaneously. For more information on XSI search criteria see Broadsoft's BroadWorks Interface Specification Xtended Services Interface.

Setting	Description
Metaswitch Services:	
Web URL	The Metaswitch Web URL.
Directory Number	The Metaswitch Directory number.
Password	The Metaswitch password
Disconnect on Hook	Sometimes it is useful to disable the disconnection of a call when the handset is placed on hook, e.g., during conference calls or in handsfree-mode, etc. This is achieved by turning this setting off.

SIP tab

With the SIP tab, you can configure SIP identity settings for the phone.



Setting	Description
SIP Identity Settings:	
Voice Quality Report Collector	Specifies the collector to which a voice quality and registration reports are send to. The form of the report is specified by the setting rtcp_xr. For optional route headers on the notify request you might specify them with comma separated syntax and with a valid sip url.
Music on hold server	If you specify a SIP URI pointing to a media server account, the phone will, when a call is put on hold, invite this SIP URI to call the held phone to play music on hold.
Send hold as inactive	Specify if you want to indicate an hold request with sdp parameter sendonly or inactive. Some pbx's need the inactive setting for proper music on hold operation.
Alert Info URL	This URL should point to a web server where audio alert messages are accessible.
User picture URL	Specify an URL to a small JPEG picture. When the flash plugin feature (Preferences page) is enabled, this picture will be shown on the 'Home" web page during a call.
Dial-Plan String	You can set up the dial plan for this line here. With a dial plan, you can match user input (digits via keyboard) to specific actions like dialing, using a distinct outgoing identity, etc.

Count all groups in Dial-Plan	Defines how the backreferences (e.g. \3) inside our dialplan substitution patterns count. Historically, VTech only counted matched-groups that actually matched, ignoring the others.
	
	See this example
	
	`Input: hello`
	`RegEx: ((hell)(l?)(o))`
	`with this setting = false`
	` \0 : hello`
	` \1 : hell`
	` \2 : o`
	`with this setting = true`
	` \0 : hello`
	` \1 : hell`
	` \2 :`
	` \3  o`

Setting	Description
ENUM Support	ENUM means that a conventional E.164 number (normal phone number) is mapped to a SIP URI so that a pure IP call can be started instead of an IP/PSTN call. To use ENUM lookup not only this option has to be enabled, but also below options Countrycode and Areacode have to be setup properly before. Both options are used to build the above Dial Plan String which is mandatory to make the ENUM lookup work. NOTE: Part of the dialplan in order to set up ENUM support. 'ENUM 49 30' means the phone resides in the contry code 49 and area code 30 and is setup to use ENUM lookup.
Countrycode	The country code for ENUM lookup (e.g., 49 for Germany).
Areacode	The area code for ENUM lookup (e.g., 30 for Berlin).
Proxy Require	If your SIP proxy/registrar needs the 'SIP Proxy Require' header, it can be enabled here.
Additional supported headers	If your SIP proxy/registrar needs the additional header, it can be enabled here.
Q-Value	You can set up the probability of a registration for each line through this setting (the default is 1.0). This means that different registrations with different Q-values will ring in serial order (serial forking) in contrast to different registrations with the same Q-values, which will ring in parallel (parallel forking).
Proposed Expiry	The proposed expiry time of the registration in seconds for line x. Upon expiration of the registration, the phone will send a fresh re-registration request.
Auto Answer	If it is <on>, the phone will automatically answer incoming calls.
Long SIP-Contact (RFC3840)	When your SIP Registrar is not properly supporting long contacts specified in accordance with RFC 3840, you may want to switch this behavior off.

Setting	Description
Support broken Registrar	If your VoIP provider works only when you turn on 'Support broken registrar' on the phone's web interface, this means your provider does not call your phone the way the phone requested to be called. What happens is that incoming INVITEs from your VoIP provider do not contain the contact URI which was previously registered by your phone as its contact. Thus the phone cannot safely identify the target line of the incoming call. When you compare the URI in the first line of the incoming INVITE and the URI in the Contact of the REGISTER, which the phone sends to the registrar of your provider, they will presumably differ. This is what we mean by 'broken registrar'. It is as though your provider has sent a letter to an apartment building with the city, the street address, and the house number on it, but without the recipient’s name. When you turn on 'Support broken registrar', the phone tries to find the right apartment by guessing, but this guessing will fail when there are two parties with the same name in the building.
Shared Line	If you have to share your extension (identity) with somebody else, this has to be enabled.
Publish Presence on bootup	When this feature is set to “on”, the phone publishes the last presence state on bootup.
DTMF via SIP INFO	Some IVR systems may need DTMF events signalled via SIP INFO messages, this can be enabled here. Set it to <on> or <sip_info_only> to provide DTMF codes via SIP INFO messages. With <sip_info_only>, the in band and out of band DTMF codes stop going in RTP as they are sent only through SIP INFO messages. <sip_info_only> sends DTMF codes via SIP INFO messages only. <on> additionally sends DTMF via RTP!
Send display name on INVITE	When this option is enabled, the phone receiving a SIP INVITE message adds the 'display name' of the called identity to the reply message in order to allow the calling party to show this information on its display.

Setting	Description
Extension Monitoring Call Pickup List URI	The subscription URI for monitoring the dialog states of a number of extensions setup at the PBX. This setting turns on the mechanism) cause the phone to send a single subscription even for monitoring multiple extensions. The associated NOTIFY contains the extensions configured at the server for the user and their respective status if it active. When filling this setting with a simple sip-uri or number in the WUI, it will automatically be replaced by a complex XML-configuration that allows to auto-assign the received buddies onto keys of type Contact List Buddy.
Contact List	When this feature is set to 'on', the phone subscribes for the presence status of its contacts.
Publish Presence	When this feature is set to 'on", the phone sends out PUBLISH SIP messages showing the phone's status.
Contact List URI	The URI phone will subscribe for this identity's contact list.
Force sendrecv on INVITE with no SDP	INVITE Requests without SDP should not change the state of the SDP. However, some servers can not handle this and need a sendrecv in the response whatever the previous state was. If on, the phone sends sendrecv in the response for INVITE Requests with no SDP.
Remove all bindings on unregister	When enabled the phone sets the contact header to * in order to remove the old contact at the registrar on each DeREGISTER. A DeREGISTER will be done on each ReREGISTER as well.
Subscription Expiry (s)	This value specifies the desired expiration time in seconds for subscriptions to the following event packages: ■ dialog (individual and event list subscription) ■ call-info ■ message-sumary ■ presence The subscription will be refreshed after a time randomly chosen to be between 1/2 and 3/4 of the expiration time (which the server may have reduced in the 200 OK response). NOTE: Setting this value to zero will cause the subscription to become inactive. The line-seize event package subscription is not affected by this value. It is fixed to 15 seconds.

Setting	Description
Failed Subscription Retry Time (s)	When subscription fails this settings describes the value in seconds after which the phone will try again. Be aware: don't confuse this setting with the SUBSCRIBE expiration, which is defined by user_subscription_expiry
Enable hook flash	This setting enables support for the hookflash feature on Broadsoft's Broadworks servers. When enabled the phone will process incoming INFO messages with a content type of 'application/broadsoft' for call waiting indication. Additionally, when the line key is pressed in the connected state, a hookflash event is sent to the server inside an INFO message. This occurs in lieu of the hold action which is usually invoked when this feature is disabled.
Default Contact Number	OCS offers the user the possibility of publishing additional phone numbers under which he or she is reachable. This information will published along with the user's presence information. When traversing a contact list on a VTech phone, a contact may be called by selecting it (i.e. scrolling until it is highlighted) and then pressing 'enter' or going offhook. By default, the contact's SIP URI is used to place the call. This setting allows the default to be changed to one of the published phone numbers. This is paricularly useful in environments where OCS is used for presence only and voice is routed over a different server, as the OCS SIP URI cannot be used in this case to establish a voice call. This setting is used by server directories such as Metaswitch, LDAP, Broadsoft XSI and Broadsoft Xmpp Contacts, to control the behavior when user presses OK on a contact: ▪ If set to "none" (default), bring up the Contact Details screen of the contact ▪ If set to "main", directly dial the number that is considered the main one of the contact
Identity can receive calls	When this is disabled, invites for audio-calls will not be accepted by this identity. A non-voice-identity will automatically force setting hide_identity to be enabled.
Allow incoming extension monitoring	When this setting is 'off', all incoming dialog subscriptions for this identity are rejected with a '403 Forbidden' response. In other words, other users are blocked from monitoring your extension.

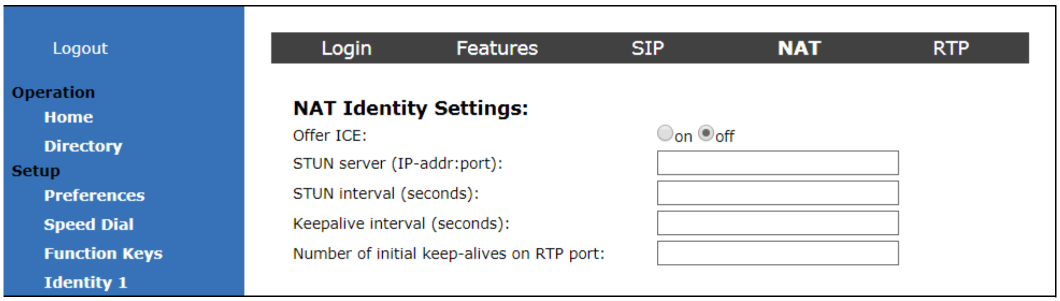
Setting	Description
Extension monitoring group ID	For this setting to have any effect, user_allow_inc_dialog_subscribe must be on. It allows the user to restrict extension monitoring to a group of users using one of two possible mechansism: shared secret or contact group. To use the shared secret mechanism simply enter a pass phrase into this field. All users using the same pass phrase can monitor each other's extension. Note that this mechanism does not work with OCS/Lync. Note also that the pass phrase must not start with '{'. The contact group mechanism is currently available only with OCS/Lync. Enter the name of a group on your contact list to allow all members of that group to monitor your extension. To distinguish a contact group from a pass phrase surround the group name with curly braces. For example: {My Pickup Group}. Entering empty braces {} allows everyone on your contact list to monitor your extension (this also works with non-OCS buddy lists).
Default BLF direction	RFC4235 the direction attribute of the XML dialog info is optional. With this setting you can define the default value.
Device Feature Key Synchronisation	Identity-Based. Many SIP phone users prefer to use the buttons on their phone to activate features, such as Do Not Disturb (DND), rather than any web portal. This feature permits these SIP phone users to use the buttons on their phones in just this way. With this feature installed, supported SIP phones can synchronize with the Application Server on the status of the following features: ■ Do Not Disturb ■ Call Forwarding Always (CFA) ■ Call Forwarding Busy (CFB) ■ Call Forwarding No Answer (CFNA). If a user changes the status of one of these features via the web portal or a feature access code (FAC), the Application Server notifies the phone about the status change. Conversely, if the user changes the feature status via a button on his/her phone, the phone notifies the Application Server of the status change. The synchronization protocol is based on the SIP events framework. To use this capability, the phone user must have a SIP phone that supports the 'as-feature-event" event package.
Refer-To Brackets	Switch additional brackets on or off in the Signaling for Refer-To. Some devices rely on this setting. Refer-To setting is per identity.

Setting	Description
Check SDP Version	Usually each received sdp-packet has a version number that identifies it. When receiving the same version again the phone can ignore it. However this versioning mechanism does not work reliably with all PBX'es so we introduced the option to keep the phone from checking the version. When version check is off, the phone will compare the entire sdp instead (except for the version). When setting user_server_type to nortel, ocs or broadsoft -> version-check will be disabled automatically.
Check CSeq in Dlg Info Notify	So as to prevent an incorrect LED status, a NOTIFY transporting a dialog-info event will be proceesed only if is not stale (i.e. the CSeq number in the NOTIFY request is greater than the last one received). Disable this setting if you want the CSeq counter to be ignored.
Number sign encoding	RFC 3261 states that the number sign (#) must be encoded inside a telephone subscriber. Therefore the default value of the setting is 'on'. Change it to 'off' if you need special cases for direct dialing and therefore not encoding the #.
Monitor Notify for Subscriptions	If we subscribe, we must get a NOTIFY indicating the current state of the dialog. But sometimes it might happen that the NOTIFY gets lost. For handling this error state, we introduced a new timer which monitors the receiving of the NOTIFY. If we don't get the NOTIFY, we un-subscribe the current subscription and set up a new fresh subscription to get the current state and resolve the error condition. Normally this setting should remain off. If you experience that the BLF gets frequently out of sync (staying on to long), or otherwise have the condition described above, you could give this setting a try.
Accept Event Talk without SDP	Accepts and processes the talk-NOTIFY also when the sdp isn't in the received INVITE, regardless of other settings.
Call Waiting Indication	Call Waiting Indication combines two functions: ■ 'Call Waiting (CW)' can be enabled ('on', 'visual only', 'ringer') or disabled ('off'). This function allows the phone to receive more than one call at one time. ■ 'Call Waiting Indication (CWI)' If Call Waiting is enabled ('on', 'visual only', 'ringer') the incoming caller extension is displayed in the lower left corner of the display. A short knocking signal can be heard simultaneously in the background of your current active call indicating another incoming call. Call Waiting Indication setting is per identity.

Setting	Description
Server Type Support	To enable PBX specific interoperability features you may specify the proper server type matching your PBX environment.

NAT tab

With the NAT tab, you can configure Network Address Translation (NAT) identity settings for the phone.

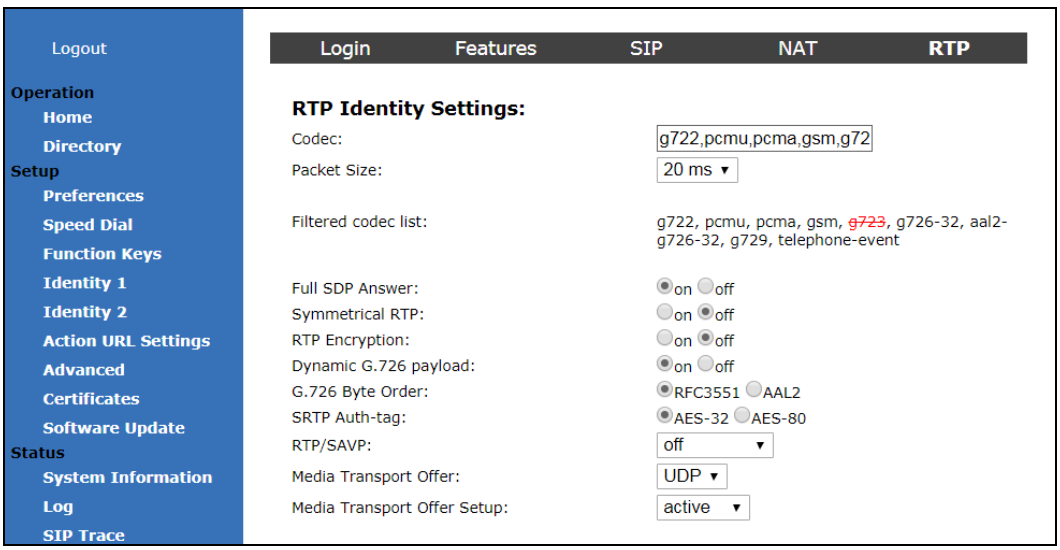


Setting	Description
NAT Identity Settings:	
Offer ICE	Choose whether or not you want to use ICE (Interactive Connectivity Establishment). ICE optimizes the media path. This would be the case, for example, when two phones in the same network are calling each other via a long media path through other, external networks. With ICE, the short media path in the same network would be chosen, which will presumably have better quality than the long one. Sometimes this feature will stop you from being able to make calls. When this occurs, switch it off. Note, that ICE currently will work reliable in OCS environment only.
STUN server (IP-addr:port)	We reintroduced a STUN keep-alive mechanism for SIP, which can be turned on manually by specifying the address of the STUN server followed by the port number. However, we strongly discourage you from using it, because it can not work properly in symmetrical NAT environments (i.e., linux-based router/firewall). The only general SIP NAT solution is a session border controller (SBC) on the service provider’s side.
STUN interval (seconds)	Sets the STUN interval time in seconds. After its expiration a new STUN requests will be send out. If it results in another IP/port the identity will be re-registered.

Setting	Description
Keepalive interval (seconds)	Specifies the number of seconds after which a new keepalive message will be sent out to the Registrar/Proxy port in order to have the port stay open and the phone remain reachable.
Number of initial keep-alives on RTP port	The number of keep-alives the phone should send out at the beginning of an RTP session. A keep-alive is an empty STUN Binding Request and serves to open a pin hole in the firewall. The phone sends one keep-alive by default, i.e. when the setting is empty. This is for backward compatibility. Set this to zero if you want no keep-alives. Note that if the phone receives such a Binding Request, it will answer it with a Binding Response.

RTP tab

With the RTP tab, you can configure Real-time Transport Protocol (RTP) identity settings for the phone.



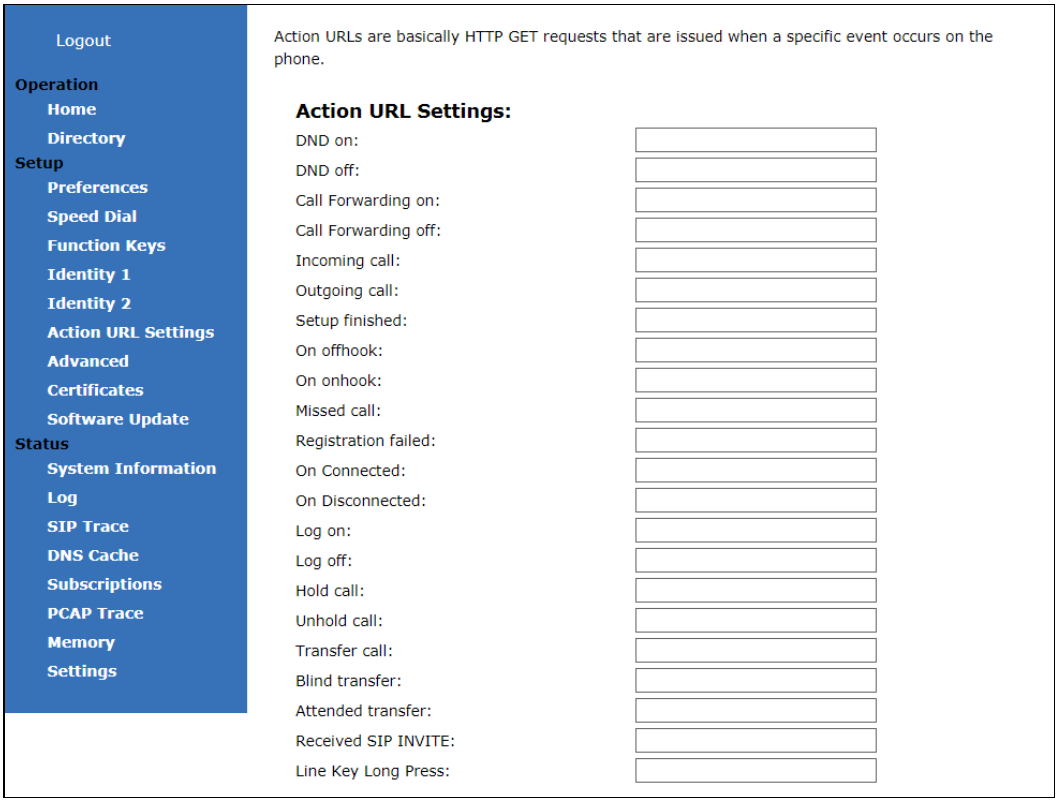
Setting	Description
RTP Identity Settings:	
Codec	Prioritize which codecs (audio-stream) the phone should use. Prioritizedma-separated list, most desired codec up front.

Setting	Description
Packet Size	Select the packet size in ms. Please note that the following codecs only work with certain packet time values: ■ g723: 30 or 60 ms ■ gsm: 20,40 or 60 ms
Filtered codec list	comma separated list of all configured codecs for this identity. All valid codecs are black and invalid codecs (e.g. configured with not supported packet size or wrong name) are red and crossed out.
Full SDP Answer	When the setting is turned 'on', the phone returns a list of all available codecs in the SDP in response to INVITE requests. Otherwise the first codec of the calling party that matches the configured codecs on the phone is returned.
Symmetrical RTP	This setting tells the phone to always send RTP packets to the same IP and port from where it receives them. It ignores the port which the remote party sent in the SDP details. If the two incoming and outgoing RTP (audio) streams of a single call should use the same port number, turn this setting on.
RTP Encryption	Your phone supports RTP encryption via SRTP. If you want to encrypt your outgoing audio (RTP) stream, this option must be “on”. Both parties have to enable the RTP Encryption option to establish an SRTP call. RTP encryption has nothing to do with SSL/TLS. The keys are sent in the SDP part of SIP messages. Certificates are not used for this. The default value is on. In order to obtain full security SIP call you have to use TLS as well. Then, a small lock sign is shown on the display which means that an secure SIP call is currently taking place (SIP secured + RTP encrypred).
Dynamic G.726 payload	**This setting is obsolete.** Previously turned on dynamic payload type for G726.
G.726 Byte Order	There are two types of byte order for G.726, namely RFC3551 and AAL2. With this setting you can choose the byte order in order to use the same order as the remote entity. Note: this setting has no effect on codec: AAL2-G726-32 !
SRTP Auth-tag	When the setting is set to AES-32 (default), the phone offers a 32-bit auth-tag for SRTP. Selecting AES-80 makes the phone offer an 80-bit auth-tag.

Setting	Description
RTP/SAVP	This setting is effective only when RTP encryption (SRTP) is also enabled and is used to specify whether the use of the RTP/SAVP profile by the phone should be off (for backward compatibility), optional or mandatory. When this setting is set to mandatory the phone will offer and accept only SDPs that contain m= lines with an audio profile of RTP/SAVP. When this setting is set to optional, the phone will offer SDPs containing two m= lines, one with an audio profile of RTP/SAVP the other with an audio profile of RTP/AVP and it will accept SDPs containing m= lines with either profile. The RTP/SAVP profile, being the preferred one, is listed first. Since some SIP proxies cannot handle RTP/SAVP profiles or multiple m= lines this setting may also be turned off. In this case the phone will send SDPs containing RTP/AVP audio profiles only. Whether or not the crypto attribute is included depends on whether RTP encryption is on or off. Note: When RTP encryption is turned off this setting has no effect.</p>
Media Transport Offer	Select the type of the rtp media transport. In mostly every case you should be fine with the default udp. However, RTP via TCP is also available according to RFC4145. If you choose tcp please pay also attention to user_media_setup_offer.
Media Transport Offer Setup	The chosen value has only affect if user_media_transport_offer has been set to TCP. It defines according to RFC4145 the local role on an SDP offer. ▪ active: local party is connecting to remote party (a=setup: active) ▪ passive: remote party is connecting to local party (a=setup: passive) ▪ any: remote party shall decide who is connecting (a=setup: actpass)

Action URL Settings page

On the Action URL Settings page, you can configure Action URLs, which are basically HTTP GET requests that are issued when a specific event occurs on the phone.



Setting	Description
Action URL Settings:	
DND on	In case the specific action has taken place (here DND has been switched on), a web GET to the specified URL is performed.
DND off	In case the specific action has taken place (here DND has been switched off), a web GET to the specified URL is performed.
Call Forwarding on	In case the specific action has taken place (here CFWD ON / redirection always has been activated), a web GET to the specified URL is performed.
Call Forwarding off	In case the specific action has taken place (here CFWD OFF / redirection always has been deactivated), a web GET to the specified URL is performed.
Incoming call	In case the specific action has taken place (here an incoming call is ringing), a web GET to the specified URL is performed.
Outgoing call	In case the specific action has taken place (here an outgoing call has been started to dial out), a web GET to the specified URL is performed.

Setting	Description
Setup finished	In case the specific action has taken place (here the end of the setup function has been reached after a reboot and the phone has finished starting up), a web GET to the specified URL is performed.
On offhook	In case the specific action has taken place (here the handset was lifted from the hook switch), a web GET to the specified URL is performed.
On onhook	In case the specific action has taken place (here the handset was put on the hook switch), a web GET to the specified URL is performed.
Missed call	In case the specific action has taken place (here an incoming call has been missed), a web GET to the specified URL is performed.
Registration failed	In case the specific action has taken place (here registration has failed), a web GET to the specified URL is performed.
On Connected	In case the specific action has taken place (here the call has been connected), a web GET to the specified URL is performed.
On Disconnected	In case the specific action has taken place (here the call has been disconnected), a web GET to the specified URL is performed.
Log on	In case the specific action has taken place (here one identity has been logged on), a web GET to the specified URL is performed.
Log off	In case the specific action has taken place (here all identities have been logged off), a web GET to the specified URL is performed.
Hold call	In case the specific action has taken place (here the active line is set to on hold), a web GET to the specified URL is performed.
Unhold call	In case the specific action has taken place (here an active line is set to connect to talk), a web GET to the specified URL is performed.
Transfer call	In case the specific action has taken place (here either a blind or an attended transfer of a call, not by the initiator), a web GET to the specified URL is performed.
Blind transfer	In case the specific action has taken place (here an initiation of a non attended transfer during call or ringing), a web GET to the specified URL is performed.

Setting	Description
Attended transfer	This event will be triggered on the phone (A) which received the REFER message during an attended transfer. Usually this is the calling party (A), while B is the called party, that performed the transfer and C is the party the call is transferred to.
Received SIP INVITE	This event is intended to be used on phone C in a typical attended transfer scenario where phone A calls phone B and phone B transfers to C. B sends a SIP REFER message to A which causes phone A to send a SIP INVITE message to phone C. Note: This event may also be triggered by another RE-INVITE during an existing Connection Dialog.
Line Key Long Press	This event is intended to be used for long press events of a function key (line key). If a line key is pressed longer than 2 seconds, a web GET to the specified URL is performed. By configuring the URL for example with a XML script, you can add an extra long press functionality for each line key. If you add the runtime variable $longpress_key to the query or the fragment part of the URL, you can use the line key name in the script to perform different actions for each line key. Example: http://<webserver-IP>/xml_test/test.xml#var:linekey=$longpress_key
Check for blacklisting	The action blacklist URL HTTP request is triggered when a call is received. If the HTTP server of the configured URL answers with 200 OK, then the caller is processed as remotely blacklisted, and the phone silently rejects the call. In case the server answers with an error, the call is accepted and the phone is ringing. In case it takes too long for the answer, the call should be accepted. This timeout can be configured with the setting remote_blacklist_action_timer. The blacklisting can be done via an Action URL, e.g.: `action_blacklist_url=http://myserver.com/blacklisted?caller=$remote`

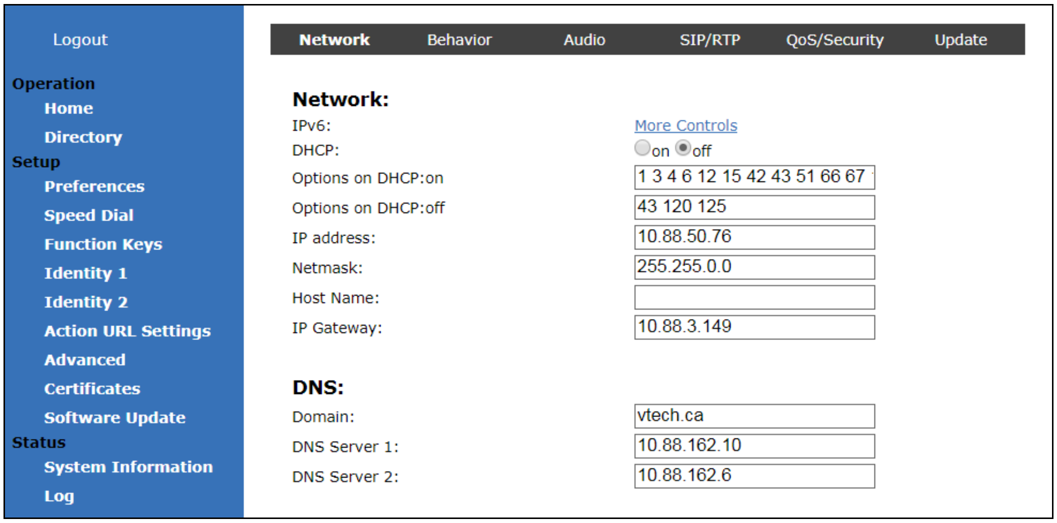
Advanced pages

On the Advanced page, you can configure various advanced settings for the phone.

The Advanced page has six tabs - **Network**, **Behavior**, **Audio**, **SIP/RTP**, **QoS/Security**, and **Update**. When you click the Advanced page, the **Network** tab is automatically selected.

Network tab

With the Network tab you can configure settings for the network IP addresses, DNS domains, NTP time server, HTTP proxy, LDAP, SIP trace, and SNMP port.



Setting	Description
Network:	
IPv6:	Click **More Controls** to see the IPv6 settings. See *"IPv6 settings" on page 128*.
DHCP:	Turn the use of DHCP for inquiring IP on or off with this option. The phone will still use DHCP to inquire other data when this setting is turned off. It does so by sending a DHCP-inform-message containing the list of the desired parameters. The list may be configured with this setting.
Options on DHCP:on	List of options to be inquired from dhcp-server when IP is fetched (dhcp = on). Should the server provide other options than stated in this list, they will be ignored (accept 53 and 54). See also Settings/dhcp_options_on_inform, which does something similar for when dhcp = off
Options on DHCP:off	List of options to be inquired from dhcp-server when no IP is to be fetched (dhcp = off). The phone will send an dhcp-inform during boot-up should this list not be empty. Should the server provide other options than stated in this list, they will be ignored (accept 53). See also Settings/dhcp_options_on_ip_aquire, which does something similar for when dhcp = on
IP address	You can change the IP address of the device through this setting. This parameter is mandatory in order to enable the Ethernet connection.

Setting	Description
Netmask	Change the netmask for the device.
Host Name	Change the hostname of the phone here. If set, the hostname is used to sign syslog packages and as the title of the webinterface webpages.
IP Gateway	This setting shows the IP address of the default IP gateway (NOT the VoIP gateway). It is the address to which the packets get routed when the desired packet address is not in the current subnet. Setting up this parameter is mandatory in order to reach an external network.
DNS:	
Domain	Specify the DNS domain for your phone here. This parameter is mandatory in order to enable DNS searching.
DNS Server 1	Specify the IP address of the DNS server for your network here. This parameter is extremely important for a proper functioning phone, so please make sure it is set up correctly.
DNS Server 2	Specify the IP address of a backup DNS server for your network here.
Time:	
NTP Time Server	Specify the domain name / IP address of the NTP server here.
NTP Refresh Time (sec)	The interval after the phone will re-synchronize the time from the NTP server, in seconds.
Timezone	Select the time zone of your geographical location through this option.
HTTP:	
HTTP Proxy	You can select the HTTP proxy address for your phone here. This is needed if you are also surfing the web via such a proxy. You can additionally define the Port Number e.g. 192.168.X.X:YYYY
HTTP port	Specify the HTTP port to be used by your phone through this setting. By default, it is port 80.
HTTPS port	Specify the HTTPS port to be used by your phone for HTTPS connections (default 443).

Setting	Description
Webserver connection type	Set up the type of connection the phone's web server is willing to answer to. Please be advised that you will no longer be able to use the web user interface of the phone when you select "off"! Press the menu key, use the navigation key to go to the submenu "Webinterface", and select "Server". Then change the type of connection to one of the other types. Note: activation of changes requires a reboot.
Auto Logout (min)	Specify the time in minutes after which the web interface shall ask you to login again.
LDAP:	
LDAP name filter	LDAP name filter is the search criteria for name look ups. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). The name prefix for search entered by the user is represented by the "%" symbol in the filter.
LDAP number filter	LDAP number filter is the search criteria for number look ups. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). The number prefix for search entered by the user is represented by the "%" symbol in the filter.
Server Address	This setting refers to the DNS name or IP address of the LDAP server.
Port	This setting specifies the LDAP server port. In case the setting is not configured, the default LDAP port (389) is taken.
Base	This setting specifies the LDAP search base (the distinguished name of the search base object) which corresponds to the location in the directory from which the LDAP search is requested to begin. The search base narrows the search scope and decreases directory lookup time. If you have multiple organizational units in your directory (for example, OU=Sales in O=COMPANY and OU=Development in O=COMPANY), but the OU=Sales organization never uses AOL AIM, you can restrict the lookup to the OU=Development subtree only by entering providing the following search base: OU=Development, O=COMPANY.

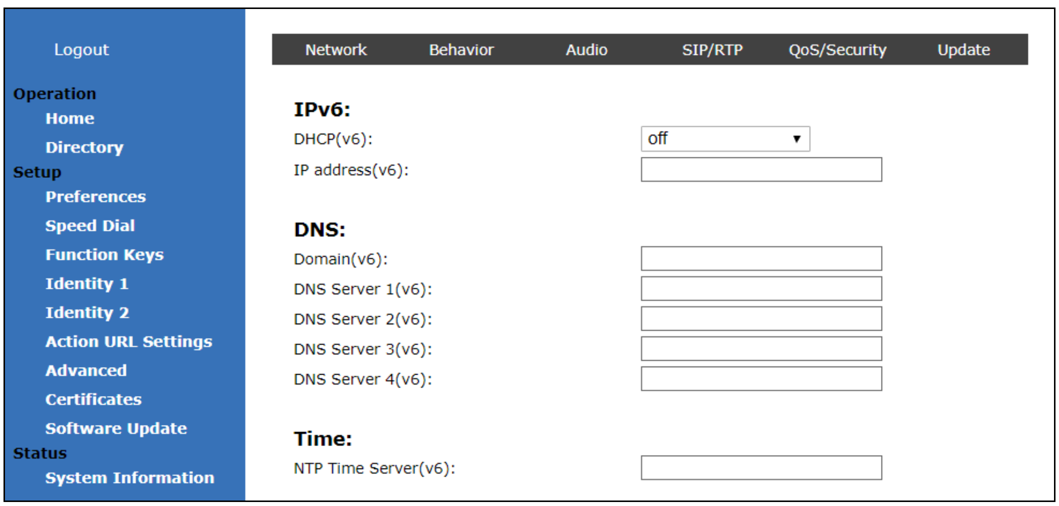
Setting	Description
Username	This setting specifies the bind “Username” for LDAP servers. Most LDAP servers allow anonymous binds in which case the setting can be left blank. However if the LDAP server does not allow anonymous binds, you will need to provide the Username and Password allowed to query the LDAP server.
Password	This setting specifies the bind “Password” for LDAP servers. VTech phones use “simple” authentication scheme for bind requests. This setting can be left blank in case the server allows anonymous binds. Otherwise you will need to provide the Password along with the Username in order to access the LDAP server.
Max. Hits	This setting specifies the maximum number of search results to be returned by the LDAP server. Please note that a very large value of the “Max. Hits” will slow down the LDAP lookup, therefore the setting should be configured according to the available bandwidth. The default value for this setting is 50.
LDAP name attributes	This setting can be used to specify the “name” attributes of each record which are to be returned in the LDAP search results. This setting compresses the search results, as the server only returns the attributes which are requested by the VTech phone. The setting allows the user to configure multiple space separated name attributes. Please consult your system administrator regarding which name attributes are to be configured.
LDAP number attributes	This setting can be used to specify the “number” attributes of each record which are to be returned in the LDAP search results by the LDAP server. This setting compresses the search results, as the server only returns the attributes which are requested. The user can configure multiple space separated number attributes by using this setting. Please consult your system administrator regarding which number attributes are to be configured.
LDAP display name	This setting specifies the format in which the “name” of each returned search result is to be displayed on the VTech phone. The setting allows combinations of various “name attributes” along with special characters.
Countrycode	This setting is used for specifying standard country codes which are to be substituted in LDAP search requests.
Areacode	This setting is used for specifying standard area codes which are to be substituted in LDAP search requests.

Setting	Description
LDAP over TLS	Specifies wether to use tcp (off) or tls (on) as LDAP transport.
Sort Results	This setting can be used to sort the LDAP result set.
Predict Text	Allows to quickly lookup names in the LDAP directory by using a technique similar to the one known as T9. In order to search John for example, you would press 5 6 4 6 consecutively. Note: With this option enabled you cannot toggle between letters by pressing the same key several times.
Do an initial Query	When entering the LDAP directory you can decide whether or not to query the server for an initial list of entries (query string = *).
Ethernet Ports:	
Net Port	This setting is used to configure the NET port of the phone's integrated Ethernet switch.
PC Port	This setting is used to configure the PC port of the phone's integrated Ethernet switch.
Detect Ethernet Cable Unplug	When this option is set to ‘on’, the phone will display a warning message and a status message when it loses ethernet connectivity. When WLAN is configured, only the status message is diplayed.
Action on Ethernet Cable Replug	Choose the action to be performed after the network connection is reestablished.
Debug:	
Syslog Server	Type in the host where a Syslog Server is running to store the log messages coming from the phone.
LCServer	Type in the IP address of the remote LCServer if you want your phone to connect to it. Usually, you do not need to make an entry here.
SIP Trace	Switches SIP tracing on or off.
SIP Trace for REGISTER/ SUBSCRIBE/NOTIFY	Set to 'off' when you do not want to log REGISTER-, SUBSCRIBE-, NOTIFY- nor SERVICE-SIP-messages in WUI-sip-trace.
SIP Trace Size (Number of Messages)	Determines the number of messages to keep in the trace. Once this number is reached, the oldest message is removed when a new on is added. If you want to trace only to a USB device (see usb_storage_siptrace), you may set this value to zero.

Setting	Description
Truncate SIP Body to this Size (in Bytes)	This setting determines how many bytes of the original body to keep in the trace. If you don't want the body to be truncated at all, set this setting to -1 (messages written to a USB storage device (see usb_storage_siptrace) are never truncated, irrepective of the value of this setting).
SNMP:	
Port	Type in the port to be used for SNMP communication.
Trusted Address	Specify the address range (in CIDR notation) solely from within which subnet SNMP requests will be accepted e.g. 192.168.0.0/16

IPv6 settings

To display these settings, go to the **Advanced** page > **Network** tab, and click **More Controls** under the Network area.

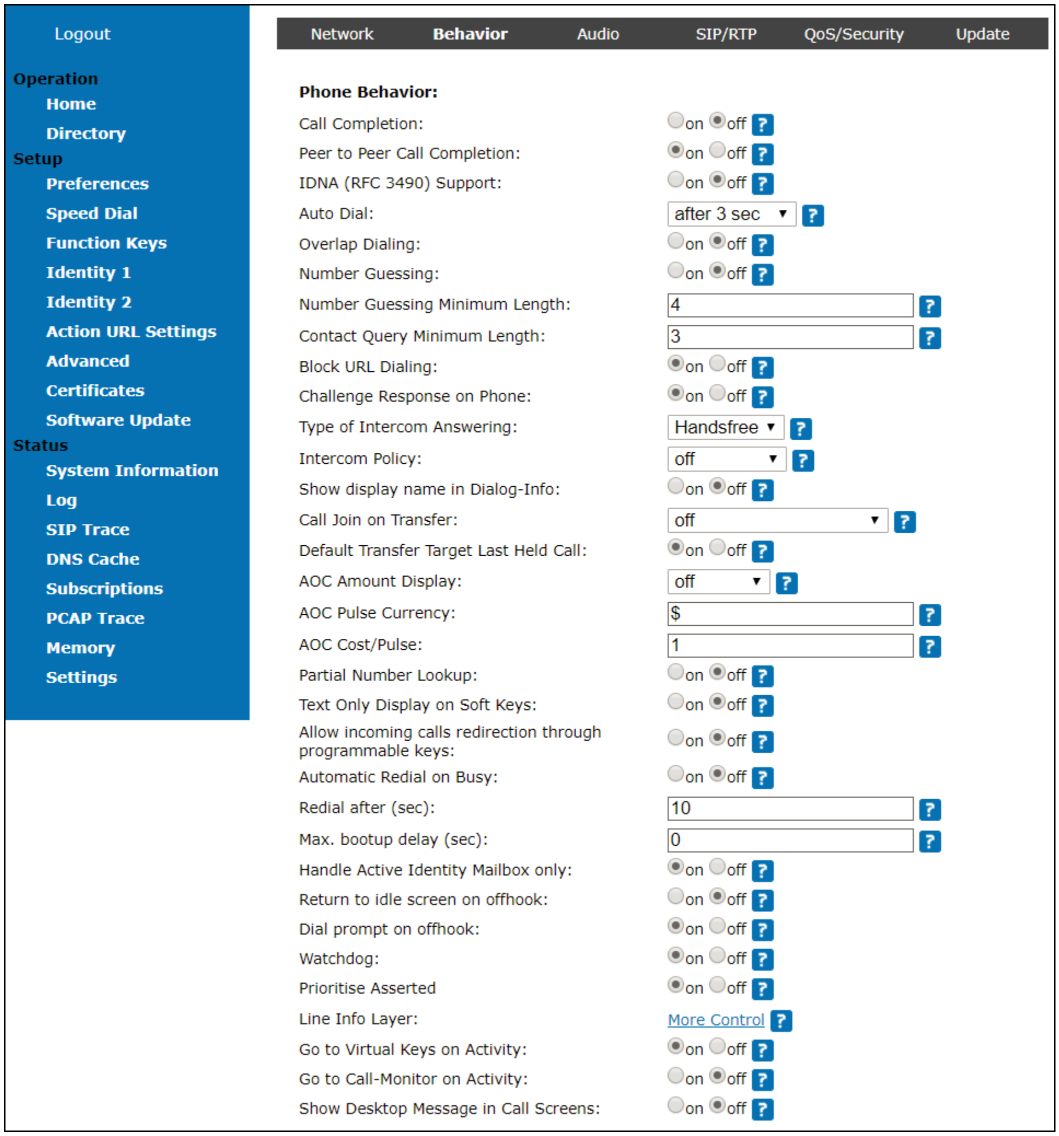


Setting	Description
IPv6:	
DHCP(v6):	This setting enables the use of ICMPv6 or DHCPv6 for inquiring IPv6 addresses. Currently this is the only way of assigning IPv6 addresses to your VTech phone. Setting up static IPv6 addresses is currently not supported. IPv6 address changes during operation cannot handled dynamically at the moment. Thus a restart of the phone is needed in order to use the new IPv6 address properly.

Setting	Description
IP address(v6):	This setting holds the current IPv6 address of the device. Note: Setting up static IPv6 addresses is currently not supported. See also dhcp_v6.
DNS:	
Domain(v6):	Additional domain name for IPv6 networks. See also dns_domain.
DNS Server 1(v6):	Additional DNS server for IPv6. See also DNS Server1.
DNS Server 2(v6):	Additional DNS server for IPv6. See also DNS Server1.
DNS Server 3(v6):	Additional DNS server for IPv6. See also DNS Server1.
DNS Server 4(v6):	Additional DNS server for IPv6. See also DNS Server1.
Time:	
NTP Time Server(v6):	Addtional NTP server for IPv6. Used only if ntp_server is empty.

Behavior tab

With the behavior tab, you can configure settings that control the phone's behavior.



Setting	Description
Phone Behavior:	
Call Completion	Turning this setting to “on” will prompt the user to activate call completion, if possible, while calling a number (see the CC soft key). When the called party becomes available again, your phone will be able to automatically redial the number. CCNR (on not reachable) as well as CCBS (on busy) is supported.
Peer to Peer Call Completion	Disable it if call completion is handled by the SIP proxy. Otherwise the phones are handling it directly between each other.

Setting	Description
IDNA (RFC 3490) Support	Switch on support for Internationalizing Domain Names in Applications (IDNA). IDNA support is the ability to handle domain names including international special characters.
Auto Dial	This setting is switched off by default. You can set a timeout after which a number is dialed automatically without pressing or taking the handset off the hook. Note: Apart from the predefined values in the drop-down form, you can set other values either through provisioning or single HTTP GET requests.
Overlap Dialing	If the connected SIP proxy supports this function, it can be enabled here. This will lead to the phone starting to dial each time a digit is entered and the SIP proxy replying with „Number incomplete“ until such time as the number has been entered and the call can be initiated successfully without the enter key having to be pressed.
Number Guessing	With this setting, the number guessing functionality can be enabled. This is the automatic number completion which will begin after you have entered the minimum number of digits.
Number Guessing Minimum Length	Specify the minimum number of digits that must be entered before 'Number Guessing' will begin. This setting also defines when ldap-lookup should begin when entering a number.
Contact Query Minimum Length	Minimum number of chars required before starting the query (LDAP, ABS, ...)
Block URL Dialing	You can block the dialing of SIP URLs by turning this setting on. In this case only numeric numbers will be allowed as input.
Challenge Response on Phone	VTech phones can handle challenge responses on the phone. Turning this setting off will disable this feature and you will only be able to handle authentication through the web interface of the phone.
Type of Intercom Answering	If the Alert-Info header is taken into account in order to allow auto answering behaviour like intercom, this option can be used to specify whether the phone answers in handset, headset, or handsfree Mode. Also see Auto Connect Type

Setting	Description
Intercom Policy	Incoming intercom-calls (i.e. those that use the Alert-Info SIP header, see intercom) do not ring but go directly to connected. That is if the situation and this setting allow it. ■ off - will disable auto-connect ■ always - will enable auto-connect without restrictions ■ idle - will allow auto-connect only when phone is in idle-screen ■ not_busy - will allow auto-connect except for when the user is in an active call, i.e. holding a call will allow for intercom-interuptions, while any call in one of the following states will disable auto-connect: ringing, calling, connected, being held by the other call-partner.
Show display name in Dialog-Info	When this setting is turned on, the call monitoring state shows display names for remote and local users found in the body of incoming dialog info notifies, as long as the display_method setting is set to name as well. If this setting is turned off, the user name in the uri's will be shown to maximize display space.
Call Join on Transfer	When this feature is turned "on" and you have exactly one person on hold, pressing transfer while in a call to another person will result in an immediate transfer of him/her to the held party. If it is set to "always" the immediate transfer is invoked also if there is more than one call on hold. In this case the transfer target is either the first or the last call to be put on hold, depending on the setting xfer_dest_order_lifo. In the same scenario with this feature turned off, you will be asked where to transfer to. You can then select between all the other calls you currently have and entering a number manually.
Default Transfer Target Last Held Call	Determines in which order held calls are presented to the user as destination during an attended transfer. When 'on' the most recent call on hold is presented first; when 'off' the oldest one is presented first.
AOC Amount Display	If your provider supports "Advice of Charge" (AOC) information (i.e., the costs for your call) during or at the end of the call, you can turn on this feature by selecting one of the following options: 1. Select "Charged" to show the accumulated amount of the current call on the display. 2. Select "Balance" to show the amount remaining on your account.

Setting	Description
AOC Pulse Currency	Sets the currency symbol that will be shown next to the amount (e.g., $).
AOC Cost/Pulse	Specify how much money one pulse costs (e.g., 0.12 means 12 cents per pulse).
Partial Number Lookup	When this option is set to 'on', the phone tries to match parts of the incoming call number to numbers stored in the address book displays the name belonging to the first number that matches partially. An integer value can be set too. If the value of the setting is n and n > 0, the phone sends a query to the LDAP server or to the internal address book. It matches with entries that end with that postfix of lenght n.
Text Only Display on Soft Keys	If enabled <on>, soft key icons are symbolized by text and not by icons anymore.
Allow incoming calls redirection through programmable keys	Allows to redirect an incoming call to a prespecified number using function keys e.g. Speed Dial, Extension etc. Can be turned off to disable such automatic transfers in a call centre environment.
Automatic Redial on Busy	In case of busy signal the phone will make an automatic attempt to redial the same number if this setting is turned on. The redial time is dependent on the setting auto_redial_value.
Redial after (sec)	If Automatic Redial on Busy is on, the value of this setting is used to redial the same number in case of busy signal.
Max. bootup delay (sec)	On reboot, the phone waits for a random number of seconds not exceeding the value set in this field, and then continues to boot up. This is to prevent DOS by provisioning servers etc. by preventing all the phones (that are rebooting) to send requests simultaneously in a given setup.
Handle Active Identity Mailbox only	If this setting is on, the Retrieve button will dial the mailbox of the active line. Otherwise the mailbox associated with the first MWI message in the queue is used. This setting also changes which type of status-msg is used for signaling messages on PBX. When set to on, the statuses CurrentIdentityHasTextMessages and CurrentIdentityHasVoiceMessages are used. When set to off the statuses PhoneHasTextMessages and PhoneHasVoiceMessages are used. I.e. changing this setting will automatically change the status-msg controlling settings: status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_blocked and status_msgs_that_are_important

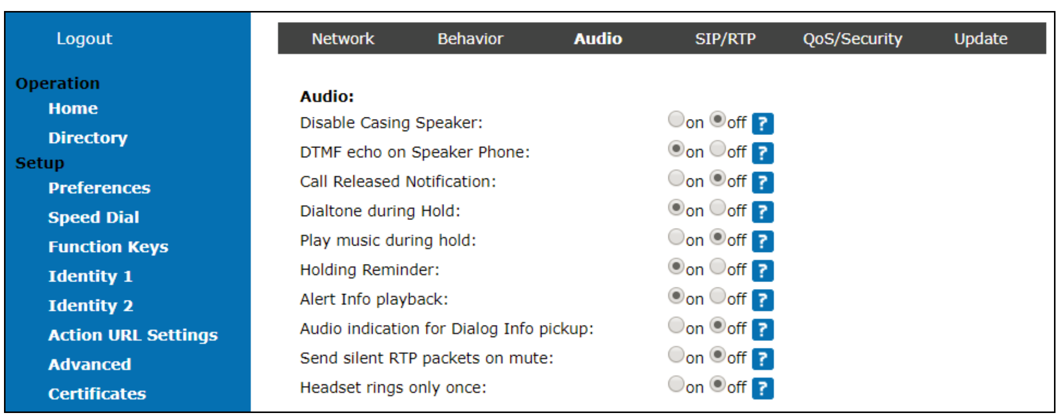
Setting	Description
Return to idle screen on offhook	If this setting is on, the phone will go to idle state even when the handset is offhook i.e. it will not prompt the user to dial a new number.
Dial prompt on offhook	If this setting is on, the phone will offer a dial prompt when the handset goes offhook. Otherwise the phone stays in idle state.
Watchdog	The watchdog will watch your phone, if the phone will freeze, the watchdog initiates a hard reboot of the phone. This watchdog is based on the linux software watchdog.
Prioritise Asserted	SIP messages like INVITE may include asserted information (p-asserted-identity). If this setting is enabled, the phone displays the name provided by the asserted information with the highest priority. Only if no asserted information is given the priority defined by the related setting contact_source_priority will be considered.
Go to Virtual Keys on Activity	When one of the virtual p-keys shows a monitored line that is not idle, the phone will automatically show the virtual key state. See also settings: states_ignored_in_goto_vkeys_on_activity, goto_monitor_state_on_line_activity, and pui_states_allowing_state_switch_on_activity
Go to Call-Monitor on Activity	When any of your monitored lines shows an activity (other than idle), the phone will automatically display the call-monitor state. See also pui_states_allowing_state_switch_on_activity and goto_virtual_keys_state_on_activity.
Show Desktop Message in Call Screens	Messages received via SIP MESSAGE outside an INVITE are displayed on the desktop of the idle screen. When this setting is enabled, the message will also appear in call screens. **NOTE:** Messages received inside an INVITE dialog are only displayed in the 'connected' screen.
Keys:	
Transfer on Onhook	If you want to transfer two calls by placing the handset onhook (one incoming call and one outgoing call), you can switch it on here.

Setting	Description
Independent transfer on Onhook	If you want to transfer two calls by placing the handset onhook (independent of call direction (incoming / outgoing): that will be not a Plain Old Telephone Service pots) , you can switch it on here. Condition: transfer_on_hangup must be set to on.
Transfer starcode picked up calls	If setting 'transfer on hangup' is set to on and the first call was picked up with a PBX starcode then the transfer will be done if this setting is set to on. Info: a picked up call with starcode is an outgoing call. But an incoming and an outgoing call is the condition for the 'transfer on hangup'.
Quick Transfer to Speed Dial/Extension	If set to **New Call**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will put the active call on hold and a new call will be initiated dialing out to the configured number associated with the key. If set to **Blind Transfer**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will blind transfer the active call to the configured number associated with the key. If set to **Attended Transfer**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will put the active call on hold and initiate a new call to the configured number for attended transfer. User can complete the transfer as early attended or attended transfer via the "Transfer" key.
Block DND	If you don't want the users of the phone to have the option to turn on the “Do not disturb” (DND) mode, set “Block DND” to “on”. This may be desirable in call center or switchboard environments.
Use Speaker Key to Dial	Usually the speaker key can be used to start a dial attempt, if this behaviour is unwanted, it can be disabled here.
Use Speaker/Headset Key to Receive Calls	Usually the speaker key can be used to receive an incoming call, if this behaviour is not desired, it can be disabled here. This setting is valid for headset key too.
Cancel Key on Held Call	When this option is set to ‘off’, a call on hold cannot be cancelled by pressing the CANCEL button, but has to be taken up again and then canceled. This prevents the accidental cancellation of calls on hold.
Clear Missed Calls on Cancel	When this option is set to ‘on’ the missed call list will be cleared when pressing the CANCEL button, otherwise it remains until phone reboots.

Setting	Description
Clear Desktop Message on Cancel	When this option is set to ‘on’ the desktop message will be cleared when pressing the CANCEL button, otherwise it remains until phone reboots.
Logon/Logoff:	
Logon Wizard	The Logon Wizard assists you during the SIP line registration process. Turn this setting on if you want to use the Logon wizard, switch it off if you don’t. <skip welcome>: enables the wizard but starts directly with editing the account
Automatically logoff all lines after inactivity (min)	After turning back to idle state and specified amount of time in minutes all identities are removed.
Preselection:	
Prefix	Specify the number to be prefixed to each dialled number. NOTE: If a number is entered in this option, the phone dials this pre-selected number automatically every time the phone is taken off the hook. This is particularly useful for using calling/prepaid cards etc.

Audio tab

With the Audio tab, you can configure audio settings for your phone.



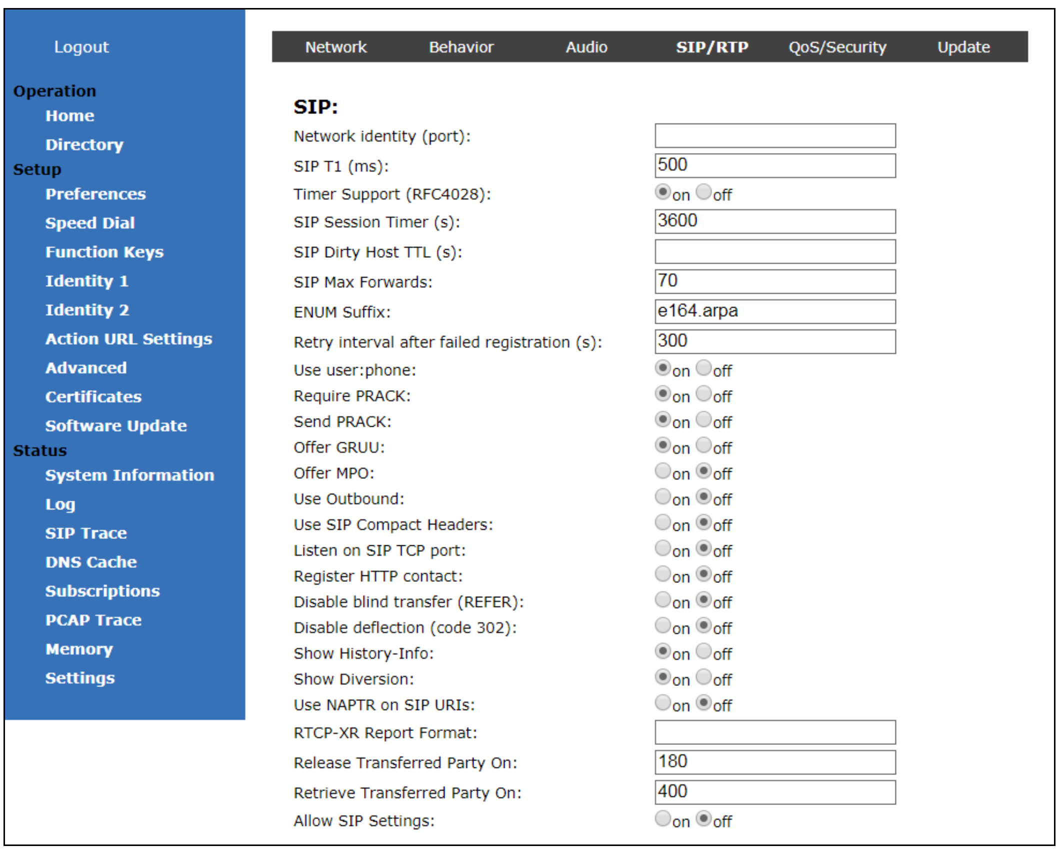
Setting	Description
Audio:	
Disable Casing Speaker	Turn this setting on to disable your speaker.

Setting	Description
DTMF echo on Speaker Phone	Hearing DTMF tones during call initiation (dialing) is enabled only for some tone schemes (due to each country's ISDN standardization). However, some customers asked to be able to disable these tones when in speaker mode, for privacy reasons. You can disable them by turning this setting to off. Please note that this setting has no effect on handset/headset mode. Here is the list of the tone schemes this feature will affect: ■ Australia ■ China ■ Denmark ■ Great Britain ■ India ■ Italy ■ Japan ■ Mexico ■ Netherlands ■ New Zealand ■ United States Note: During a call the DTMF echo is always audible.
Call Released Notification	Set this to “on” if the release sound should be played when the remote party terminates the call. Set this to “off” if no sound should be played when the remote party terminates the call. (A busy sound is played when the remote party is busy or denies an incoming call.)
Dialtone during Hold	Turning this setting “on” will play a dial tone when a call is being held, signalling the user that he/she is able to dial a second number. No dial tone is played when this setting is set to “off”.
Play music during hold	Enable this setting if you want to stream music from your local phone to the callers on hold. The music is stored on your phone and can be exchanged via provisioning.
Holding Reminder	When this option is set to ‘on’, the phone reminds you with a short beep that you still have somebody on hold.
Alert Info playback	If you want your phone to replay audio system messages when they are provided, set this option to “on”. Additionally, you will see a message on the display. When you set the option to “off”, you will only see the message on the display.

Setting	Description
Audio indication for Dialog Info pickup	Plays an acoustic indication when a call pickup is available. This only works when there are no active calls. If set to “on”, the value of setting status_msgs_with_audio_indication is changed to 'CallForPickupAvailable:10/2'.
Send silent RTP packets on mute	Setting this to on will allow RTP packets to be sent even on mute, although they will be silent because of the microphone mute. Turning it off will block the RTP packets altogether on microphone mute.
Headset rings only once	If on repeated ringing on headsets is disabled.

SIP/RTP tab

With the SIP/RTP tab, you can configure the phone’s SIP, RTP, and multicasting settings.



Setting	Description
SIP:	

Setting	Description
Network identity (port)	Set a static local port number, which is used to listen for SIP protocol communications. Please note that setting the value to 5060 also enables direct IP calls to the IP identity (see also sip_ip_dialin_content_types).
SIP T1 (ms)	Set the retry timer in milliseconds after which an unanswered request is resent. If it is set to 500, the phone will resend the unanswered request after 500, 1000, 2000, 4000, 6000 ... 31500 ms. If the request is still unanswered after this procedure, an error message will be shown on the display.
Timer Support (RFC4028)	Define whether sip-stack should support usage of timers. (includes adding headers Session-Expires and Min-SE)
SIP Session Timer (s)	If SIP Session Timer Support is enabled, this option specifies the SIP session timer in seconds. For instance, a Re-INVITE will be sent after 50% of its value has elapsed.
SIP Dirty Host TTL (s)	Specify the “Time to Live” (TTL) for dirty hosts in seconds. This means that, when a phone was unable to reach a host, the phone will not try to reach this host again until the time specified in this field has elapsed. If this setting is 0 or empty, it has no effect (the host is set as dirty but only for 0 seconds, which means it will have no effect on future requests). See also: sip_request_timeout, sip_retry_t1, sip_health_check.
SIP Max Forwards	If you set a maximum number of forwards in this field, each time a forward is sent the counter is reduced by one. When zero is reached, the forwarding will stop. This prevents the phone from running into a SIP message-forwarding loop.
ENUM Suffix	When using ENUM, you can specify a service suffix here, if desired. There is more than one service that supports ENUM lookups, and you can select here which one you want to use. You can enter a comma separated list of route domains for ENUM lookup. Leave the default value e164.arpa if you don’t know better.

Setting	Description
Retry interval after failed registration (s)	This value specifies after how many seconds the phone should attempt to reregister when the initial registration has failed. If this value is zero, the phone will make no such attempt. Value can be single integer value (range '1' to this value) or a range like '2,10'. Randomizing 10 percent if single value is configured (e.g. 300 +- 30sec) The value can also be, for example '3,6:300'. In this case when the phone loses the registration, a random value in seconds between 3 and 6 will be chosen and after this time the phone will try again. After that the value is doubled and the phone will try again until registration succeeds or the timer rechaed the second value. This is the maximum timer value. So basically the longer the phone is unregistered the longer it takes to reregister.
Use user:phone	Turn this setting on if you want to use user=phone in SIP URIs. This is to distinguish phones from different non-phone devices like gateways, etc. (RFC 2543 deprecated).
Require PRACK	Defines whether Required:100Rel will be send or not. This influences whether a early-dialog via PRACK will be established (if the opposite offers this by sending Supported:100Rel) or not. This could be useful for playing announcements or music/ring-back-tones during the time the call is in Ringing-state. Even if set to off, the phone will still offer 100Rel in the Supported-Header if it sends the INVITE (is the originator of the call). If B responses with Required: 100Rel it will send the ACK, independent of this setting. For preventing sending 100Rel as supported (and by that sending PRACK) you have to set additionally send_prack to off.
Send PRACK	Enables/Disables sending Supported:100Rel and by this whether early-dialogs by PRACK will be offered. Enabling this could be useful if the opposite wants to play music/ring-back-tone or announcements before the call is connected. On -> Supported:100Rel will be send (and opposite could initiate Early-Dialog by sending Required:100Rel) Off -> Supported:100Rel wont be send (and opposite gets no chance to initiate Early-Dialog) Note:This does not influences whether the phone itself will send Required:100Rel if from opposite Supported:100Rel is signaled and by this initiating a early-dialog. This behavior is influenced by require_prack -> see Settings/require_prack.

Setting	Description
Offer GRUU	This setting is used to toggle the support for GRUU (Globally Routable User agent URLs) in SIP. When several phones have the same account, each one of them can be identified by the proxy through this GRUU ID, which is unique for each phone and stays the same even after reboot.
Offer MPO	Using this setting, the user can turn the Media Path Optimization support on or off. Turning it on makes sense only when you have MPO-supporting session border controller devices in your environment (e.g., Jasomi).
Use Outbound	This setting is used to toggle the support for draft-ietf-sip-outbound-20. Enable this to force the reusage of connections, what VTech phones already do. However, in combination with Offer GRUU the phone will stick to the network flow created during line registration. Additionally you have to specify a value for Keep Alive.
Use SIP Compact Headers	In order to let the phone generate short compact SIP headers this option should be enabled. Otherwise the old usual style of SIP headers will be generated.
Listen on SIP TCP port	By default, the phone doesn't on the network_id_port for TCP connections. To change this behaviour, enable this option.
Register HTTP contact	This setting decides if the phone must add the http URL of the phone as additional contact information. WARNING: Turning this setting on may cause a complete loss of VoIP ability if the proxy/registrar does not support it. We urge you strongly to leave it on “off” if you are not absolutely sure that it is supported by your proxy/registrar.
Disable blind transfer (REFER)	A boolean to disable blind transfer. If it is on, instead of blind transfer, on hitting the transfer key, the only call is put on hold and a prompt offered to make second call and a normal consultative transfer would follow. This setting was introduced for PBXs that dont support REFER.
Disable deflection (code 302)	A boolean to stop 3xx codes (e.g. 302 Moved temporarily). If the setting is on, a Busy Here is returned. Turning this setting on will also disable Call Deflect.
Show History-Info	When this feature is set to “on”, the phone shows the information available through History-Info header in the incoming INVITE.

Setting	Description
Show Diversion	When this feature is set to “on”, the phone shows the information available through Diversion header in the incoming INVITE.
Use NAPTR on SIP URIs	When this feature is set to “on”, the phone converts SIP uri's according to the regular expression dialplan of the active outgoing line for numbers dialed through Received and Missed call lists. For normal phone operation it is best to leave it turned off, as a valid SIP uri need not be converted again. Only valid if the pbx used can not append the requisite leading digits to reach remote destination or if the number does not already contain the extra digits needed. e.g. adding 00 for an international call or 0 to access a number outside the local network.
RTCP-XR Report Format	Specifies which parts the voice quality report should be composed of. The report is encapsulated in a SIP PUBLISH message that is send if a call is terminated. See also parameter vq_report_collector.
Release Transferred Party On	When a call is transferred, the transferred party sends notifications to the tranferring party about the progess of the call towards the transfer target. The notifications include the current SIP response code received by the transferred party. This settings specifies at which point the transferring party will release the transferred call. This occurs when the reported SIP response code is equal to or greater than the value of this setting. This setting should be administered in tandem with the setting retrieve_xferred_call_on. Note that when marking a call with save transfer the phone will ignore the actuall setting value and instead act as if this was set to 200.
Retrieve Transferred Party On	When a call is transferred, the transferred party sends notifications to the tranferring party about the progess of the call towards the transfer target. The notifications include the current SIP response code received by the transferred party. This settings specifies at which point the transferring party will deem the transfer failed and retrieve the transferred call (which up to this point is still on hold). This occurs when the reported SIP response code is equal to or greater than the value of this setting. This setting should be administered in tandem with the setting release_xferred_call_on. Note that when marking a call with save transfer the phone will ignore the actuall setting value and instead act as if this was set to 200.

Setting	Description
Allow SIP Settings	For security reasons this setting disables the possibility to send XML settings via SIP MESSAGE. If it is on, the phone accepts settings via SIP MESSAGE. If it is off, the phone just sends a 200 OK but does not take over the settings. If enabled one must provide a secure environment. The SIP MESSAGE method is used to send settings. A SIP MESSAGE must have the following two headers, otherwise it is ignored for this purpose. Content-Type: application/xml Event: vtech-settings
Minibrowser:	
XML NOTIFY Support	Enables/Disables xml notifies (type: application/ciscoxml OR application/vtechxml).
RTP/RTCP:	
Dynamic RTP port start	If you want to set up the port range out of which the RTP ports will be dynamically taken, specify the start port number in this field.
Dynamic RTP port stop	If you want to set up the port range out of which the RTP ports will be dynamically taken, specify the end port number in this field.
DTMF Payload Type	This setting is obsolete.
RTCP Support	If enabled, the phone uses the Real Time Control Protocol (RTCP) to measure the quality of the audio (RTP) streams. This setting does not affect the RTCP XR functionality (for RTCP XR you must set rtcp_xr and vq_report_collector)
RTP Keepalive	On a hold call the phone sends out STUN packets to keep the RTP port open by default. Set this setting to off to switch this behaviour off.
Multicast:	
Multicast Support	If enabled, the phone receives RTP G.711 u-law (20 ms) packets sent to the given multicast addresses and plays them out. It can be used for listening, in handsfree mode, for streaming audio broadcasts or public announcements etc.
Zone (1-10) - Name	The name of the multicast zone is specified as an option: name=<zone name>

<table>
<tr><th>Setting</th><th>Description</th></tr>
<tr><td>(Zone (1-10) -
IP Address</td><td>The phone receives RTP packets destined for this multicast IP address and port and plays them out.

You can setup the multicast address with additional options: speaker=(0|1):

If this option is set and value is 1, then the multicast audio will be played always over speaker. If value is 0, then the current audio device will be used. If this option is not set, then value 0 is used as default value. interrupt=(0|1):

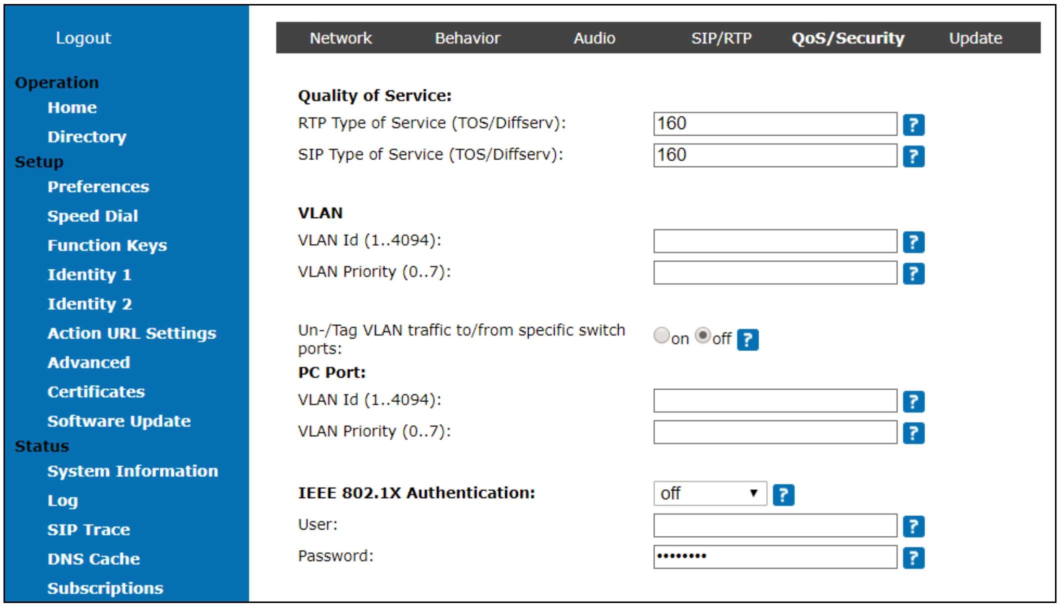
If this option is set and value is 1, then the multicast audio interrupts a running call. If multicast is finished, then the interrupted call continues. If value is 0, the multicast audio will only be played in idle state. If this option is not set, then value 0 is used as default value. volmax=(0|1):

If this option is set and value is 1, then the maximal volume will be used for multicast audio. If value is 0, then the current volume will be used. If this option is not set, then value 0 is used as default value.
priority=(0..10):

This option sets the priority of the multicast address. You can choose a priority between 0 and 10, where 0 is the lowest and 10 the highest priority. If the phone receives multicast from more than one configured port, then the multicast with the highest priority will be played. If they have the same priority then the multicast will be played, that was received first. If this option is not set, then a priority of 5 is used as default.
Please note: for hold scenarios an incoming multicast is blocked with cw_dialtone = on (default). In case it’s required to received the multicast also if calls on, please set this to off.</td></tr>
</table>

QoS/Security tab

With theQoS/Security tab, you can confiure the phone’s Quality of Service (QoS) and security settings. This tab’s page is where you configure the phone’s administrator userid/password, and the HTTP userid/password for accessing the WebUI.



Setting	Description
Quality of Service:	
RTP Type of Service (TOS/Diffserv)	This option enables the phone to support quality of service (QOS) for RTP traffic in a network. This makes sense only if all parts of the involved network also support QOS.
SIP Type of Service (TOS/Diffserv)	This option enables the phone to support quality of service (QOS) for SIP traffic in a network. This makes sense only if all parts of the involved network also support QOS.
VLAN:	
VLAN Id (1..4094)	This setting has to be set properly before the phone is able to connect to anything residing in a specific VLAN! Only the phone is residing in the specified VLAN, it has no effect to e.g. a PC connected with the second ethernet socket (PC). The VLAN tagging is done by the kernel (as opposed to vlan_net_id, which activates tagging by the phone's integrated switch).
VLAN Priority (0..7)	This is the priority of the VLAN.

Setting	Description
Un-/Tag VLAN traffic to/from specific switch ports	The ET605 Deskset has an internal ethernet-switch capable of handling vlan (set tags and unset them) This setting defines whether the switch will handle the vlan tagging or not. Handling means that pakets from the internal ports to the network are tagged (vlan id is added) and tagged pakets (vlan set) from the network are untagged (vlan id is removed) and assigned to the port they belong (selection by vlan id). **Example:** Pc-port is configured vlan 3 and the option is set to on, pakets arriving from the pc on the pc-port are tagged with vlan 3 and sent to the network. Pakets arriving from the network containing vlan id 3 will be assigned/send to pc-port, but before that the vlan id (3) is removed. So the pc will receive a paket without vlan id. Network --- VLAN ID 3 --- phone with int. switch ---- No Tag ---- PC **On:** Phone-internal switch handels the vlan-pakets. To Network direction -> vlan ids are set, From Network -> vlan id are unset **Off:** phone internal switch does not touch the pakets. Independent of vlan id set or not, pakets are not changed, connected device has to take care.
PC Port:	
VLAN Id (1..4094)	Any incoming packet on the PC port is tagged with this VLAN ID.
VLAN Priority (0..7)	This is the priority of the VLAN.
IEEE 802.1X Authentication:	This setting determines the IEEE802.1X EAP authentication method. When EAP-MD5 is selected, the settings ieee8021x_eap_md5_username and ieee8021x_eap_md5_password must be set appropriately. When EAP-TLS is selected, certificates and config file must be provided (Certificates -> 802.1X Certificates).
User	This setting specifies the username that is used for IEEE802.1X EAP-MD5 authentication.
Password	This setting specifies the password that is used for IEEE802.1X EAP-MD5 authentication.

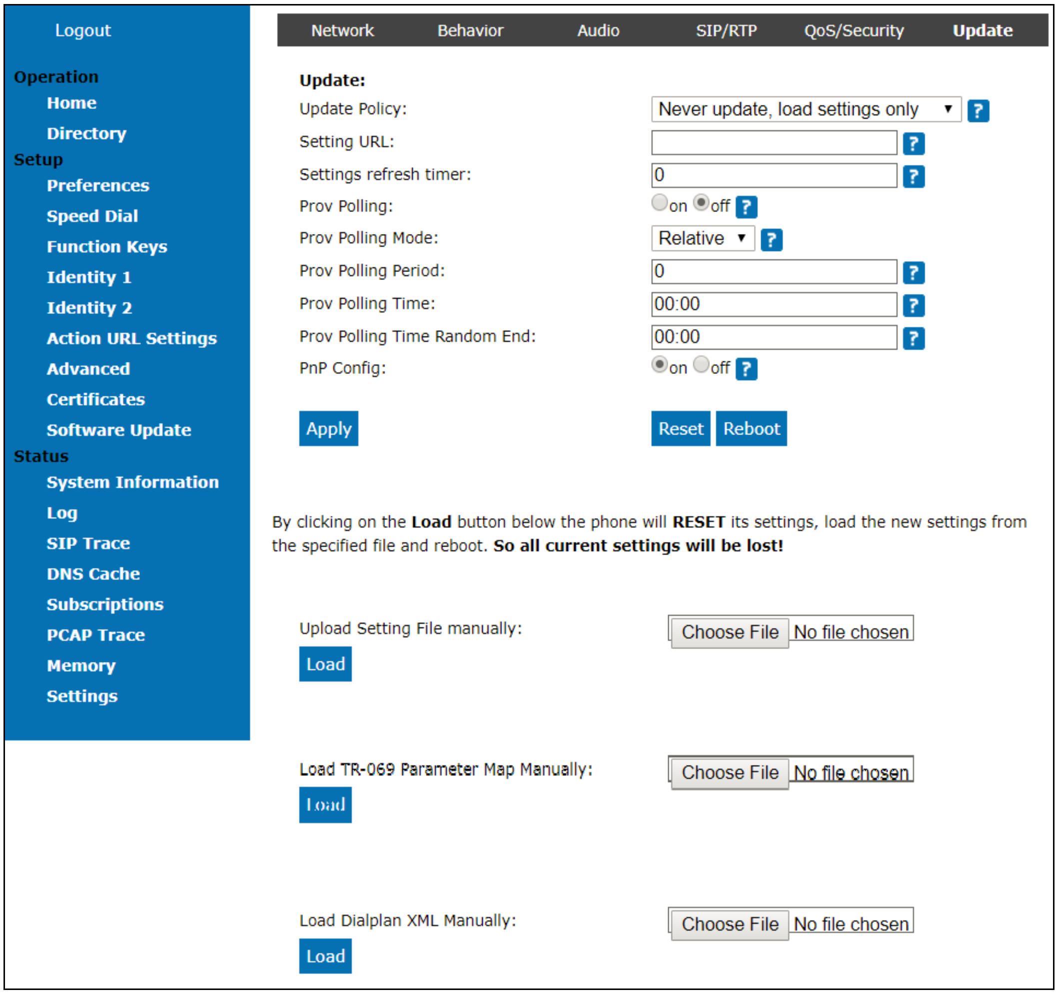
Setting	Description
Security:	
Ignore security advices	The security warning at the upper right hand corner of the web interface as well as the initial security advice web page can be switched off by setting this setting to on.
Use hidden tags	You can protect the phone's web interface with hidden security tags against remote attackers trying to change phone settings with faked HTTP POST requests (XSRF attack).
Restrict URI queries	By default, if admin_mode_password and http credentials (http_user and http_pass) are set and hidden tags are activated, query strings in URIs (the part after the ?) are restricted to a very limited number of cases. By setting restrict_uri_queries to false, query strings are not restricted anymore, so you can use hidden tags and passwords, even if you need stuff like dummy.htm?settings=save&....
Allow CSTA control	Allows to remotely control the phone via CSTA protocol. see also csta_challenge, sip_ip_dialin_content_types
Empty client cert	If this setting is on the phone will use empty client certificate in TLS connections.
Filter Packets from Registrar	If set to “on”, all SIP packets not coming from the registrar/proxy will be ignored. For security reasons, “on” is the default setting. This may cause big problems in an environment where SIP packets from other sources also have to be accepted for proper functionality! You have to disable it to make a call flow work which isn't going via the proxy only !
Authentication for SIP Reboot	This setting enables and disables challenge responses for remote reboot requests.
Authentication for SIP Check-Sync	Turning this setting on enables challenge responses for Check-Sync requests.
Administrator Mode	This setting allows to switch between user and administrator mode of the phone.

Setting	Description
Administrator Password	This setting is accessible when the phone is running in administrator (admin) mode. The default administrator password (admin PW) is “0000”. When the phone is running in user mode (i.e., many settings are not available), you need the admin PW to switch the phone to admin mode. This setting requires confirmation (see Settings/admin_mode_password_confirm). Note: We recommend that you replace the default admin PW by an individual one; if you do not, an unauthorized third party with access to the phone could set an admin PW unknown to you. In such a case, you would no longer be able to switch from user mode to administrator mode. If you set your own admin PW, be sure to write it down and store it in a secure place. If you lose your admin PW, you will not be able to return the phone to admin mode without a factory reset of all values.
Administrator Password (Confirmation)	This setting is required to confirm the admin password set at Settings/admin_mode_password to make sure that you have not made any typing errors when entering the password.
Mininum PIN length	Determines the minimum length that a PIN must have. A value of 0 indicates that a PIN is not required. If the length of the currently configured PIN is less than the value of this setting, the user will be prompted to create a new PIN which meets this requirement at the first attempt to manually lock or unlock the keyboard. On OCS servers this setting is provisioned via inband provisioning parameter ucMinPinLength, but only if its value is greater than the setting's current value.
Maximum PIN retries	Determines how many times the user may enter a wrong PIN before the keyboard is locked permanently. A value of zero indicates that there is no limit. Once the keyboard has been permanently locked, the user is prompted to reset the PIN when an attempt is made to unlock the keyboard. To reset the PIN the user must first enter the user password of the active identity. Then the user is prompted to create a new PIN. If the user cancels the PIN reset action, the keyboard remains locked.
HTTP Server:	
User	With this setting, you can select the HTTP username for your phone. Together with the HTTP Password option, it will protect your web interface.
Password	Set up the HTTP password for your phone here.

Setting	Description
Authentication Scheme	Define whether “Basic” or “Digest Authentication Scheme” should be used. Note: The latter is the more secure option.
HTTP Client:	
User	The build in web client can do authenticated HTTP(S) GET requests. Therefore, it uses this setting as user name and http_client_pass as password.
Password	HTTP Password for outgoing HTTP requests
HTTP Proxy:	
User	The build in web client can use an HTTP proxy which may ask for authentication credentials. Therefore it uses http_proxy_pass as password and this setting as user name.
Password	The build in web client can use an HTTP proxy which may ask for authentication credentials. Therefore it uses http_proxy_user as user name and this setting as password.
Upload Server Certificate	Enables you to upload your own signed web server certificate for TLS secured HTTP communication (->HTTPS). Web browsers using HTTPS to access the phone`s web interface will request this certificate from the phone´s HTTP server

Update tab

The Update tab enables you to set an update policy for auto provisioning, and manually upload a settings file (firmware update), TR-069 parameter map, or a dialplan XML file.



Setting	Description
Update:	

Setting	Description
Update Policy	Select the update policy you wish to adopt for your phone. (Only applicable when using mass deployment). ■ "**Update automatically**": load settings from settings server, but the user is not prompted to acknowledge the update, means full automatic provisioning. ■ "**Ask for update**": load settings from settings server and the user is prompted to acknowledge the update. ■ "**Never Update, load settings only**": load settings from settings server only, no update is initiated, means update disabled. ■ "**Never Update, do not load settings**": do not load any settings or updates from settings server at all, means provisioning disabled. Attention: update_policy affects all downloaded files: with "**Never Update, do not load settings**" value, the phone will not download any files (VPN config tarball, language files, etc.)
Setting URL	Enter the URL of the settings server from where you would like to obtain the configuration file to configure your phone.
Settings refresh timer	If a value greater than 0 is set (=number of seconds) the phone configuration will be requested from the setting server after the time has elapsed. After fetching the settings from the setting server URL they will be applied and the timer will be reset to the latest received value.
Prov Polling	If set to "on", automatic periodic provisioning server polling for upgrades is enabled.

<table>
<tr><th>Setting</th><th>Description</th></tr>
<tr><td>Prov Polling Mode</td><td>

Relative mode: enables phones to check for software or configuration upgrades after every X seconds. You can set the value of X in parameter prov_polling_period.
Absolute mode: enables phones to check for software or configuration upgrades at an exact time, based on the 24-hour clock. You can set the time in the parameter prov_polling_time.
Random mode: enables phones to check for software or configuration upgrades randomly. The randomness depends on the period set in prov_polling_period. If the period is less than one day, phones will check for upgrades at any time of the period randomly. If the period is greater than one day, for example 3 days, phones will check for upgrades within 3 days randomly and depend on the time period between the values in prov_polling_time and prov_polling_time_rand_end randomly also.

Random Case 1: prov_polling_period >= 1 day
<pre>prov_polling_enabled=on
prov_polling_mode=random
prov_polling_period=86400
prov_polling_time=18:00
prov_polling_time_rand_end=18:10</pre>
This case will have provisioning every day between 18:00-18:10, starting from the next day after setting being set. A general rule: If prov_polling_period >= 1 day, provisioning will occur randomly in specific time interval inside this prov_polling_period.
Random Case 1: prov_polling_period <= 1 day
<pre>prov_polling_enabled=on
prov_polling_mode=random
prov_polling_period=3600
prov_polling_time=18:00
prov_polling_time_rand_end=18:10</pre>
This case the period is 3600s and will have provisioning checked at intervals randomly selected between 0 and 3600 seconds, regardless of the time start and time end.
A general rule: if the period is less than one day, phones will check for upgrades at any time of the prov_polling_period randomly. Time start and end is not used in this case.
</td></tr>
</table>

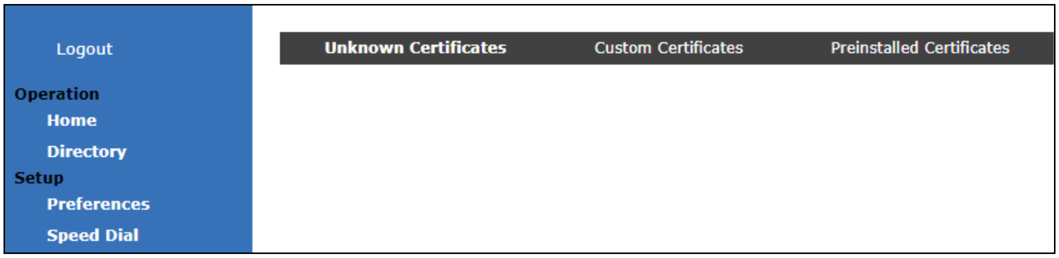
Setting	Description
Prov Polling Period	Check for software or configuration upgrades within this time interval(in seconds).
Prov Polling Time	Time to start polling of software or configuration upgrades.
Prov Polling Time Random End	Time to start polling of software or configuration upgrades.
PnP Config	If turned to on, the phone will try to retrieve its settings via a Plug-and-Play (PnP) Server. Modern SIP PBXs/Proxys can provide the PnP configuration data for the VTech phones. Please refer to the manual of your PBX/Proxy. If the PnP configuration fails, the phone will try to get the settings from a setting server.
	Apply buttton - Click to apply your changes to the **Update** area of the page. **Reset** button - Click to to reset your phone to factory default values. The WebUI displays a warning message asking if you really want to reset. Click the **Yes** or **No** button. **Reboot** button - Click to reboot your phone. The WebUI displays a warning message asking if you really want to reboot. Click the **Yes** or **No** button.
Upload Setting File manually	Select the filename of the setting file you want to upload manually.
Load TR-069 Parameter Map Manually	Select the filename of the TR-069 Parameter Map you want to load manually.
Load Dialplan XML Manually	Select the filename of the Dialplan XML you want to load manually.
	Load button - click to reset the phone’s settings, load the new settings from the specified file, and reboot. **All current settings on the phone will be lost**

Certificates page

The Certificates page enables you to manage certificates for your phone. It has the following tabs - Unknown Certificates, Custom Certificates, and Preinstalled Certificates.

Unknown Certificates tab

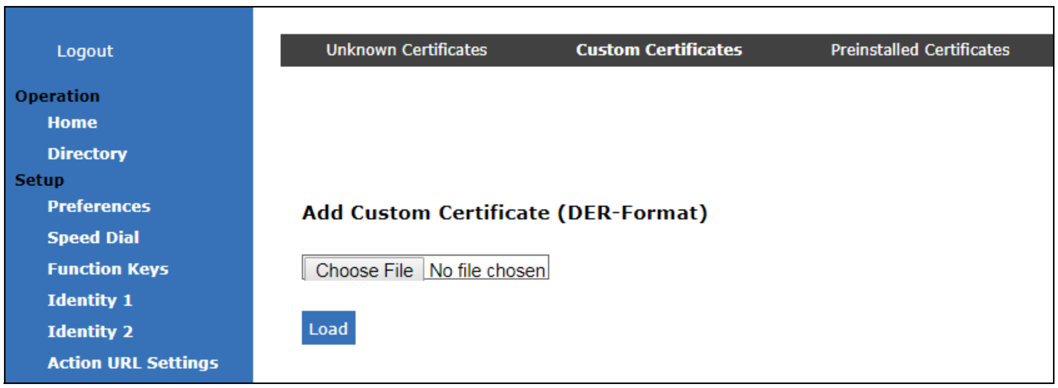
The Unknown Certificates tab displays a list of all rejected certificates.



If you want to permanently trust a certificate, you can click **Add Exception**. After adding it as an exception, a connection from a peer using this certificate will no longer be rejected. Currently, this is the only way to add unknown server certificates to the phone.

Custom Certificates

The Custom Certificates tab enables you to upload a certificate file.

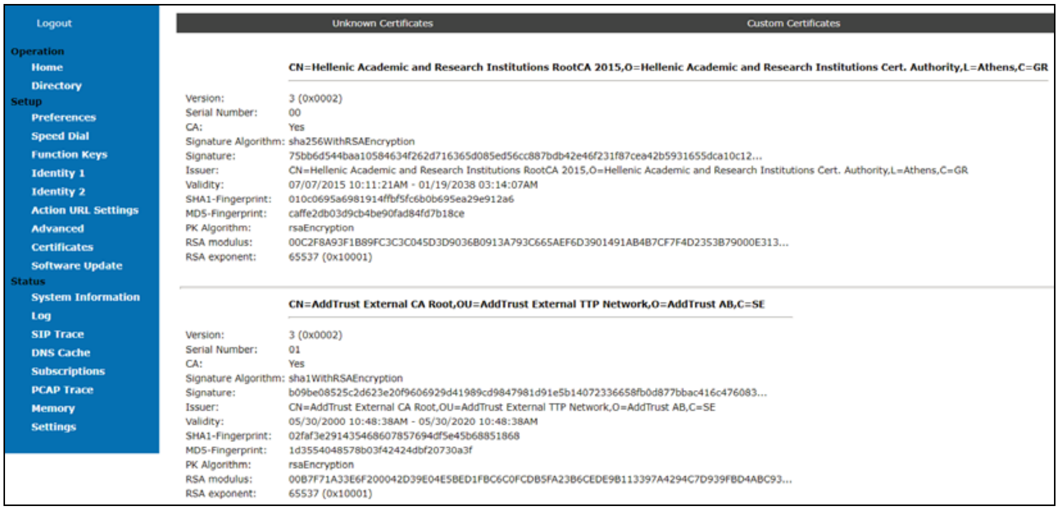


In administrator mode, you can manually upload certificates signed by one of the phone’s accepted authorities or server certificates. Every attempt to upload an unknown certificate will fail. In case of upload failures, please refer to the log and make sure your certificate is in DER format and is signed by one of phone's authorities or server certificates.

To upload a certificate, select the certificate file and click **Load**.

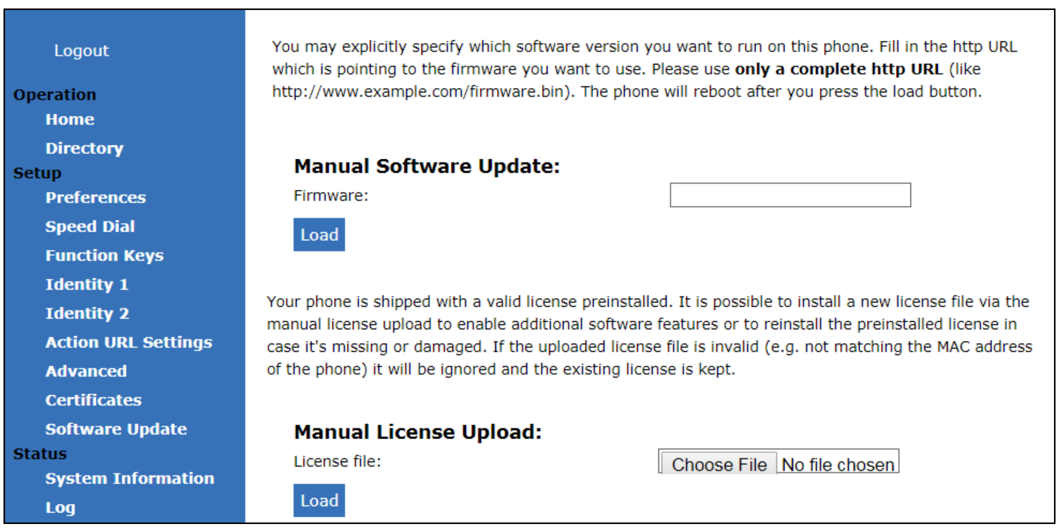
Preinstalled Certificates

The Preinstalled Certificates tab displays a list of certificates installed on your phone.



Software Update

The Software Update page enables you to manually update the ET605 firmware or manually upload a license.



Setting	Description
Manual Software Update:	
Firmware	Enter the URL for the firmware update file. This will be a .bin file. For example: VTechET605-SIP-8.10.1.24-0-SIP-r.bin You can copy and paste the URL from the ET605 downloads page on the VTech website: *businessphones.vtech.com*

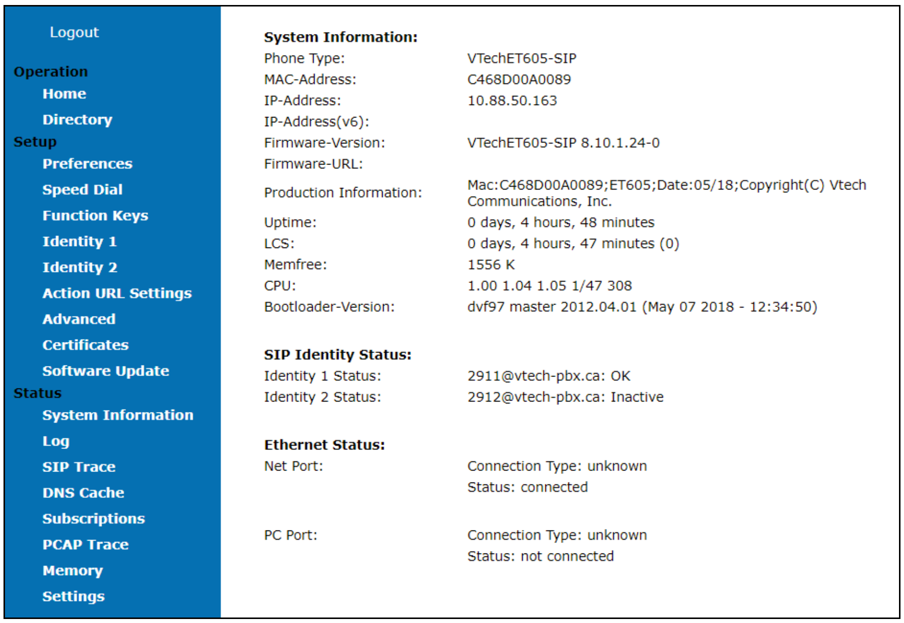
Setting	Description
	Load button - Click to update your phone’s firmware with the specified file. Your ET605 will reboot and start the software update. After it has rebooted, check the firmware version number in the WebUI: **System Information** page.
Manual License Upload:	
License file	Select the license file you want to upload.
	Load button - Click to load the license to your ET605.

Status pages

The Status pages of the WebUI are for displaying informaiton about your phone, downloading settings to a file, and performing diagnostics.

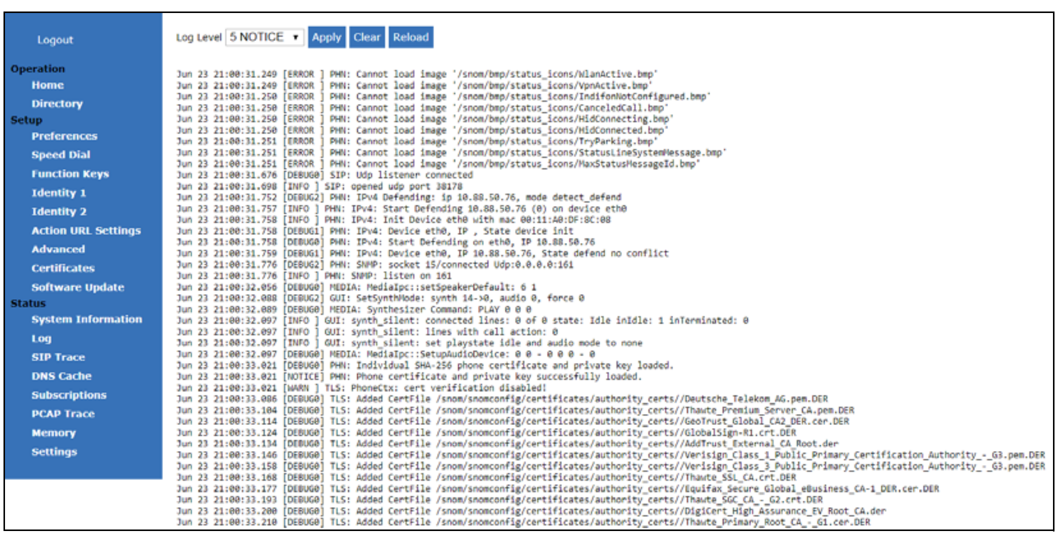
System Information page

The System Information page displays information about your ET605 Deskset, including the model, MAC address, IP address, and firmware verion number.



Log page

The Log page displays a system log.

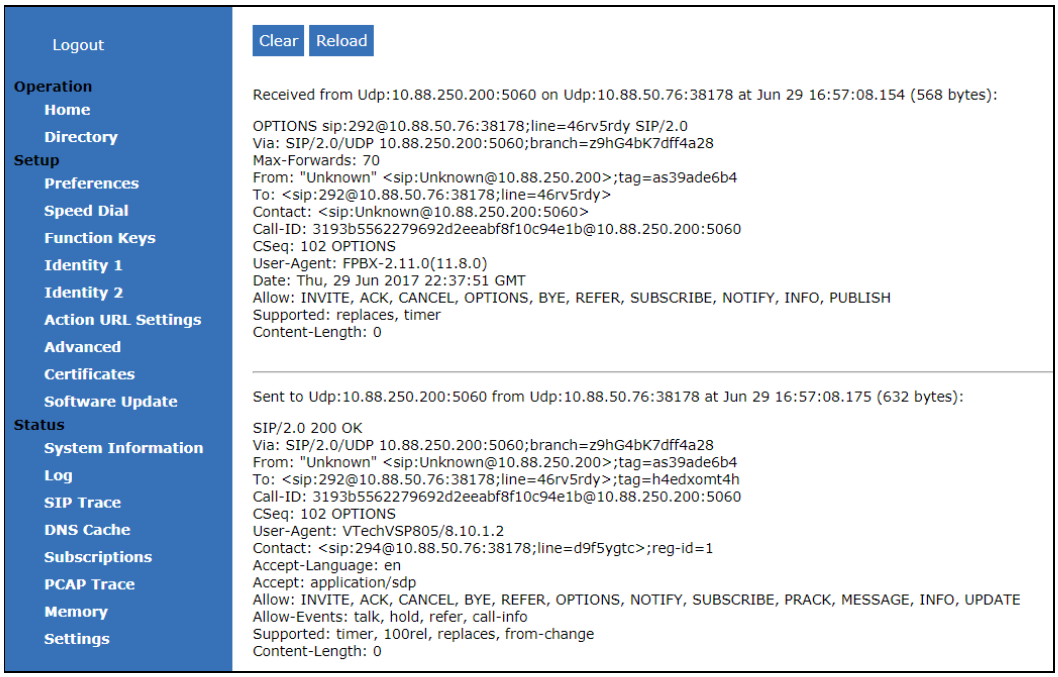


You can select the **Log Level** of the log messages you want to display, and then click **Apply**.

To reload the log, click **Reload**. To clear the log messages, click **Clear**.

SIP Trace page

The SIP Trace page is a log window which displays the SIP signaling. It becomes very important when analyzing the functionality of the phone, and is very helpful for troubleshooting support requests.



A SIP Trace is the most powerful tool to analyze all SIP related network traffic (application layer) that enters and leaves the phone’s built-in Ethernet switch.

To perform a SIP trace:

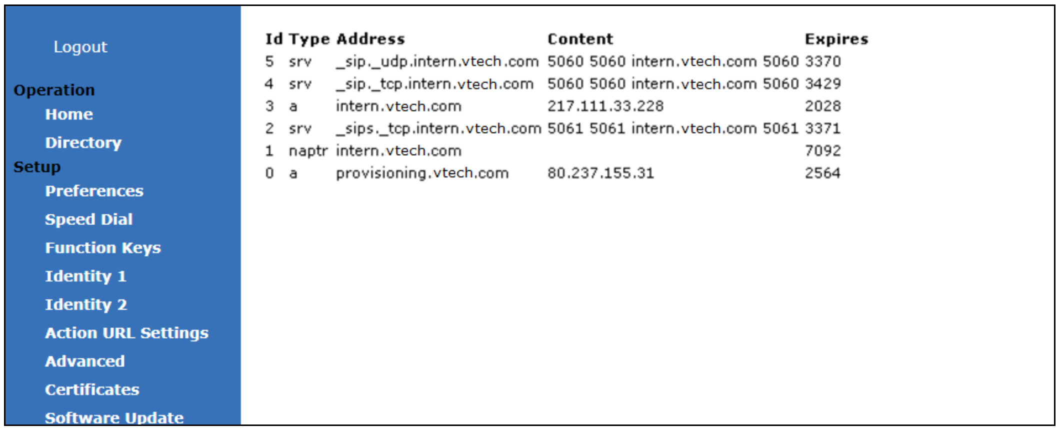
1. Open the SIP Trace page and click **Clear**.
2. Perform the scenario which caused the unexpected behavior in a basic environment.

   You can filter the displayed SIP messages via the Advanced > Network page setting **SIP Trace for REGISTER/SUBSCRIBE/NOTIFY**. You may enable the filter if the problem is not assumed to be related to Registration (REGISTER) or BLF Function (SUBSCRIBE and NOTIFY) but call issues.
3. In the SIP Trace page, click **Reload**.
4. Select and copy the content of the page and paste it into a plain text document (such as Notepad).

5. Save the textfile and name it in order to be identified easily. Attach the file to your support request.

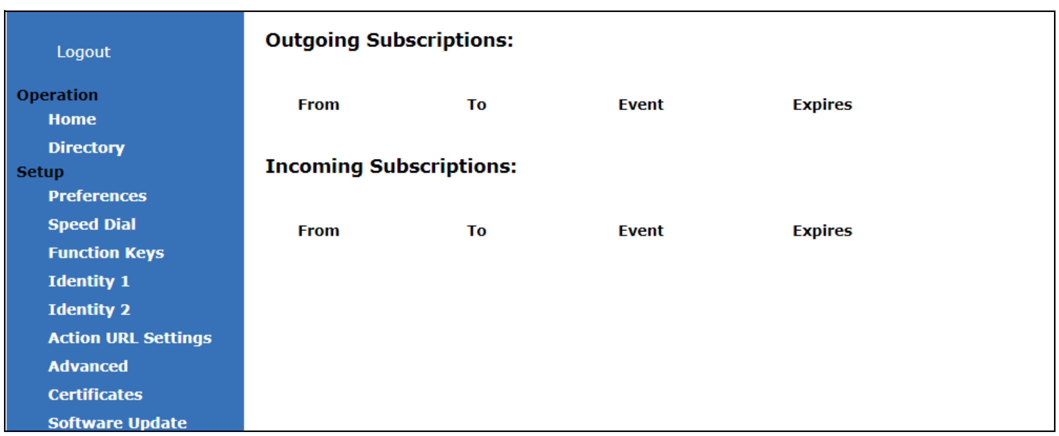
DNS Cache

This page displays the current Domain Name System (DNS) cache. It is highly recommended to copy and paste this page to a text file, and send it with your support request.



Subscriptions

This page shows subscriptions status information.



Outgoing/Incoming Subscriptions:

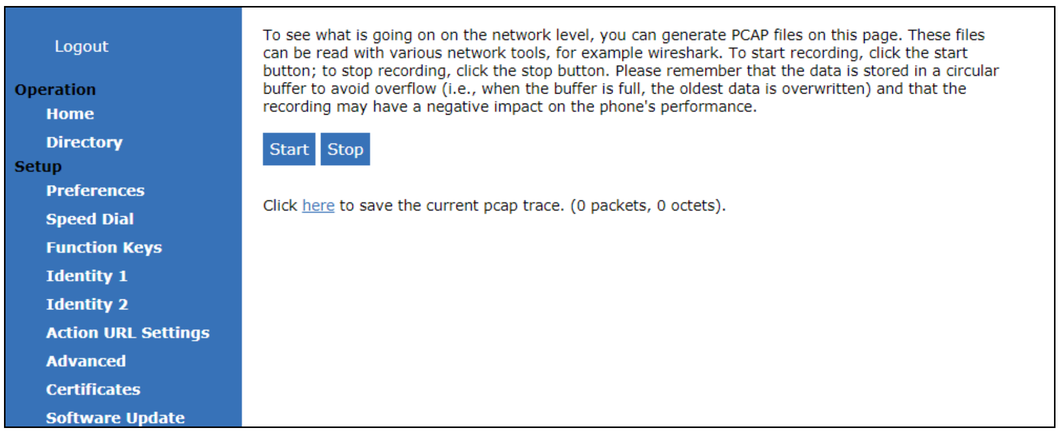
- **From**: column contains the **SIP identity** which initiated the subscription
- **To**: column contains the **SIP identity** which was subscribed
- **Event**: column contains the subscription **event**:
  - dialog (individual and event list subscription)
  - call-info

- message-summary
- presence

- **Expires**: column contains the **time in seconds** before the subscription ends

PCAP Trace page

On the PCAP Trace page, you can create IP packet traces from current network traffic directly on your phone. This tool is very powerful in order to analyze the network traffic on the phone´s ethernet interface.



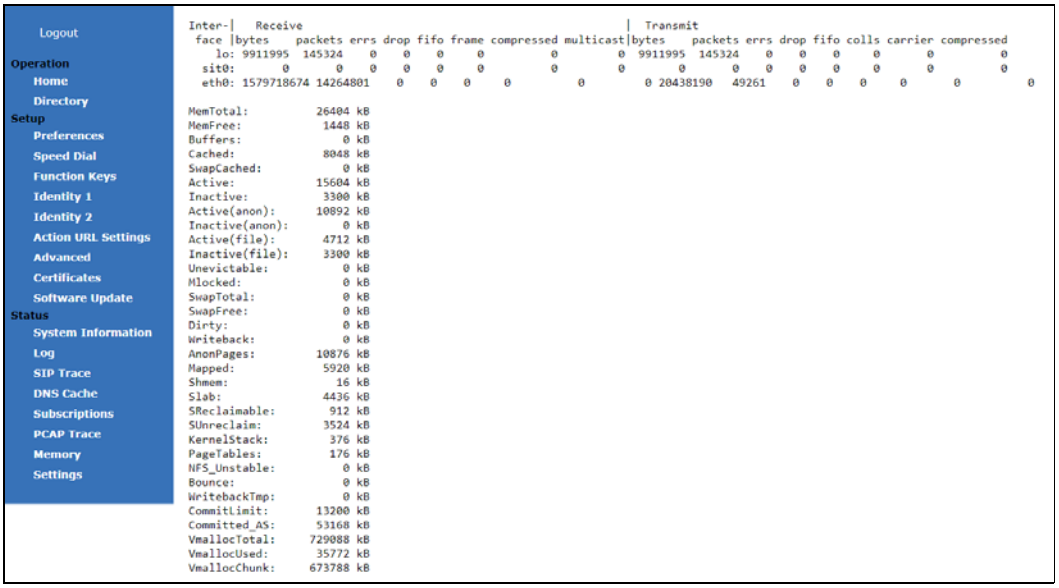
- Click the **Start** button to create IP packet traces from current network traffic directly on your phone.
- Click the **Stop** button to stop trace recording.
- Click the **here** link to save the trace to a file with the extension “pcap”. This file can be easily analyzed with tools like Ehtereal or Wireshark.

Note: Please be aware that the ring buffer size, where the information is stored during recording, is limited (515000-1 bytes). Especially when recording network traffic containing audio streams the buffer fills up quickly and as a result the first packets might be overwritten and disappear. Please try to record scenarios that are as short as possible!

Note: Performing this trace consumes memory and CPU power and may affect the phone behavior e.g. slowing down display refresh or ringtone distortion.

Memory page

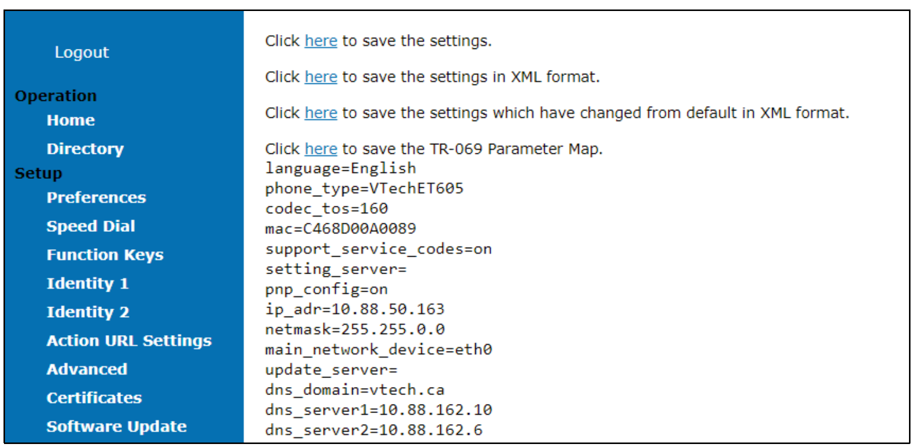
This page enables you to watch the current memory usage of your phone. You can copy and paste this information into a text file, which might be helpful for any support request.



Settings page

The settings page displays all available settings (configuration parameters) with their current values. System Internal settings are not displayed on this page.

It is a good starting point to create customized setting files for mass delployment.



- Click on “Click here to save the settings” to download the parameters in plain text format.
- Click on “Click here to save the settings in XML format” to download the parameters in XML format.

- Click on “Click here to save the settings which have changed from default in XML format” to download an XML file of those parameters which are different from the factory defaults. This file can be used to create your own setting files for Auto Provisioning.

CHAPTER 5

CONFIGURATION FILE PARAMETER GUIDE

This chapter lists the available options for all the settings (parameters) within the ET605 configuration file. Most settings in the configuration file have an equivalent in the WebUI . However, the options you must enter when editing the configuration file have a different syntax and format.

Configuration File Parameters

The following settings (parameters) are listed in alphabetical order:

Setting: accept_event_talk_without_sdp

Description: Accepts and processes the talk-NOTIFY also when the sdp isn't in the received INVITE, regardless of other settings.

Values: on, off

Default: off

Setting: acd_unavailable_req

Description: If set to "on", a call agent can select the reason code when going to the Unavailable state.

If set to "off", a call agent will not be presented with reason codes for selection when going to the Unavailable state.

Values: on, off

Default: on

Setting: ack_before_reinvite_when_holding

Description: When user wants to hold or retrieve a call, the phone will send a reinvite to change the state of the call. The user will not be able to issue another reinvite (i.e. to undo the hold/retrieval operation) until phone has received an 200-OK. Turning this setting "on" will extend that time to until the phone will have send the ACK for the received 200-OK.

Values: on, off

Default: off

Setting: ack_repetition_idle_time

Description: Time in miliseconds during which repeated ACKs on retransmitted 200-OKs will be blocked, i.e. not send.
0 disables this behaviour.
Time counts from the first ACK the phone sends.
These sort of retransmittions only occur in udp connections.
This setting only works for the reinvite-ping-pong caused by a hold-state-change originating from your phone. I.e.:
1) you press hold to place the person you are talking to on hold.
2) your phone sends reinvite to do so
3) pbx sends one or more (thru retransmittion) 200-OKs
4) your phone answers the first 200-OK with ACK and will refrain from sending any further ACKs (to any retransmitted 200-OKs) for the time set with this setting.

Values: positive integers

Default: 0

Setting: action_attended_transfer

Description: This event will be triggered on the phone (A) which received the REFER message during an attended transfer. Usually this is the calling party (A), while B is the called party, that performed the transfer and C is the party the call is transferred to. Compare this SIP call flow. In this case, a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_blacklist_url

Description: The action blacklist url HTTP request is triggered when a call is received. If the HTTP server of the configured url answers with 200 OK, then the caller is processed as remotely blacklisted and the phone silently rejects the call. In case the server answers with an error, the call is accepted and the phone is ringing. In case it takes too long for the answer, the call should be accepted. This timeout can be configured with the setting remote_blacklist_action_timer.

The blacklisting can be done via an Action URL, e.g.:

action_blacklist_url=http://myserver.com/blacklisted?caller=$remote

Values: HTTP URL

Default: blank

Setting:	action_blind_transfer
Description:	In case the specific action has taken place (here an initiation of a non attended transfer during call or ringing), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_connected_url
Description:	In case the specific action has taken place (here the call has been connected), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_disconnected_url
Description:	In case the specific action has taken place (here the call has been disconnected), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_dnd_off_url
Description:	In case the specific action has taken place (here DND has been switched off), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_dnd_on_url
Description:	In case the specific action has taken place (here one identity has been logged on), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting: action_firewall_test

Description: This setting is used to define an Action URL to be fired if the Computer Supported Telecommunications Applications (CSTA) message 'FireTest' is received. Useful to test whether a firewall blocks CSTA messages.

Values: HTTP URL

Default: blank

Setting: action_hold

Description: In case the specific action has taken place (here the active line is set to on hold), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_incoming_url

Description: In case the specific action has taken place (here an incoming call is ringing), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_log_off_url

Description: In case the specific action has taken place (here all identities have been logged off), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_log_on_url

Description: In case the specific action has taken place (here one identity has been logged on), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_longpress_url

Description: This event is intended to be used for long press events of a function key (line key). If a line key is pressed longer than two seconds, a web GET to the specified URL is performed. By configuring the URL for example with a XML script, you can add an extra long press functionality for each line key. If you add the runtime variable $longpress_key to the query or the fragment part of the URL, you can use the line key name in the script to perform different actions for each line key.
Example:

http://<webserver-IP>/xml_test/test.xml#var:linekey=$longpress_key

Values: HTTP URL

Default: blank

Setting: action_missed_url

Description: In case the specific action has taken place (here an incoming call has been missed), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_offhook_url

Description: In case the specific action has taken place (here the handset was lifted from the hook switch), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_onhook_url

Description: In case the specific action has taken place (here the handset was put on the hook switch), a web GET to the specified URL is performed.

Values: HTTP URL

Default: blank

Setting: action_outgoing_url

Description:	In case the specific action has taken place (here an outgoing call has been started to dial out), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_received_attended_transfer
Description:	This event is intended to be used on phone C in a typical attended transfer scenario where phone A calls phone B and phone B transfers to C. B sends a SIP REFER message to A which causes phone A to send a SIP INVITE message to phone C. In this case, a web GET to the specified URL is performed. Note: This event may also be triggered by another RE-INVITE during an existing Connection Dialog.
Values:	HTTP URL
Default:	blank

Setting:	action_received_subscr_notify_url
Description:	In case a notify for a subscription was received, http GET requests to the specified URL's are performed. When notifies with exact same content are received, only the first one will cause the action to be fired to minimize the workload for the phone.
Values:	HTTP URL or XML sub trees
Default:	blank

Setting:	action_redirection_off_url
Description:	In case the specific action has taken place (here CFWD OFF / redirection always has been deactivated), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_redirection_on_url
Description:	In case the specific action has taken place (here CFWD ON / redirection always has been activated), a web GET to the specified URL is performed.

Values:	HTTP URL
Default:	blank

Setting:	action_reg_failed
Description:	In case the specific action has taken place (here registration has failed), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_setup_url
Description:	In case the specific action has taken place (here the end of the setup function has been reached after a reboot and the phone has finished starting up), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_transfer
Description:	In case the specific action has taken place (here either a blind or an attended transfer of a call, not by the initiator), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	action_unhold
Description:	In case the specific action has taken place (here an active line is set to connect to talk), a web GET to the specified URL is performed.
Values:	HTTP URL
Default:	blank

Setting:	active_line
Description:	Number of the active SIP identity. This is the identity which is used as the originator of an outgoing call.

Values: 1, 2

Default: 1

Setting: admin_mode

Description: This setting allows to switch between user and administrator mode of the phone.

Values: on, off

Default: on

Setting: admin_mode_login

Description: Stores the admin login password typed in by the user to become admin.

System Internal.

Values: String

Default: blank

Setting: admin_mode_password

Description: This setting is accessible when the phone is running in administrator (admin) mode. The default administrator password (admin PW) is “0000”. When the phone is running in user mode (that is, many settings are not available), you need the admin PW to switch the phone to admin mode. This setting requires confirmation. See parameter **admin_mode_password_confirm**.

Note: VTech recommends that you replace the default admin PW by an individual one. If you do not, an unauthorized third party with access to the phone could set an admin PW unknown to you. In such a case, you would no longer be able to switch from user mode to administrator mode. If you set your own admin PW, be sure to write it down and store it in a secure place. If you lose your admin PW, you will not be able to return the phone to admin mode without a factory reset of all values.

Valid values:

1. Numbers of unspecified length. For example: 1234
2. Character strings of unspecified length. For example: nhcndeve
3. Special characters of unspecified length:
   . + @ : , ? ! - _ / ( ) ; & $ * # < > [ ] =
4. A mixture of 1), 2), 3) of unspecified length

Values: String

Default: blank

Setting: admin_mode_password_confirm

Description: This setting is required to confirm the admin password set at paremater **admin_mode_password** to make sure that you have not made any typing errors when entering the password.

Valid values:

1. Numbers of unspecified length. For example: 1234
2. Character strings of unspecified length. For example: nhcndeve
3. Special characters of unspecified length:
   . + @ : , ? ! - _ / ( ) ; & $ * # < > [ ] =
4. A mixture of 1), 2), 3) of unspecified length

Values: String

Default: blank

Setting:	admin_mode_upon_http_login
Description:	This setting determines whether the admin mode should be enabled, when the administrator credentials are used for HTTP login to the web user interface (WUI). Logging out from the WUI will disable the admin mode again.
Values:	on, off
Default:	off

Setting:	advertisement
Description:	This setting distinguishes whether an Advertisement page is displayed on the VTech phone WebUI home page. This setting is related to the parameter **advertisement_url**.
Values:	on, off
Default:	off

Setting:	advertisement_url
Description:	Advertisement page to be displayed on the VTech phone WebUI home page. This setting is related to the parameter **advertisement**. {web_lng_iso_code} will be replaced by the ISO code of the current selected WebUI language So you may provide language specific advertisements. **System Internal**
Values:	HTTP URL
Default:	blank

Setting:	alert_external_ring_sound
Description:	Melody to be played back on Alert External.
Values:	Ringer1, Ringer2, Ringer3, Ringer4, Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, silent
Default:	Ringer1

Setting:	alert_external_ring_text

Description: Text which can be specified in Alert-Info to categorize the an external number.

Values: String

Default: alert-external

Setting: alert_group_ring_sound

Description: Melody to be played back on Alert Group.

Values: Ringer1, Ringer2, Ringer3, Ringer4, Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, silent

Default: Ringer1

Setting: alert_group_ring_text

Description: Text which can be specified in Alert-Info to categorize a group number.

Values: String

Default: alert-group

Setting: alert_info_playback

Description: If you want your phone to replay audio system messages when they are provided, set this option to “on”. Additionally, you will see a message on the display. When you set the option to “off”, you will only see the message on the display.

Values: on, off

Default: on

Setting: alert_internal_ring_sound

Description: Melody to be played back on Alert Internal.

Values: Ringer1, Ringer2, Ringer3, Ringer4, Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, silent

Default: Ringer1

Setting: alert_internal_ring_text

Description: Text which can be specified in Alert-Info to categorize an internal number.

Values: String

Default: alert-internal

Setting: allow_mismatched_sdp_answers

Description: RFC 3264 stipulates that an SDP "answer MUST contain exactly the same number of "m=" lines as the offer", and that "existing media streams are removed by creating a new SDP with the port number for that stream set to zero" (that is, m= lines may be added, but not removed from the SDP). Some UAs don't adhere to this and drop disabled streams in SDP answers or new SDP offers within an existing session (for example, when putting the peer on hold). SDP offers or answers missing an m= line will normally cause the VTech phone to end the session, unless this setting is enabled.

Values: on, off

Default: off

Setting: allow_rtp_on_mute

Description: Setting this to "on" will allow RTP packets to be sent even on mute, although they will be silent because of the microphone mute. Turning it "off" will block the RTP packets altogether on microphone mute.

Values: on, off

Default: off

Setting: allow_sip_settings

Description: For security reasons this setting disables the possibility to send XML settings via SIP MESSAGE. If it is "on", the phone accepts settings via SIP MESSAGE. If it is "off", the phone just sends a 200 OK but does not take over the settings. If enabled one must provide a secure environment.
The SIP MESSAGE method is used to send settings. A SIP MESSAGE must have the following two headers, otherwise it is ignored for this purpose.
Content-Type: application/xml
Event: vtech-settings

The body of the SIP message contains XML like:

```
<settings>
  <phone-settings>
    <setting_name>setting_value</setting_name>
    ...
  </phone-settings>
</settings>
```

Values: on, off

Default: off

Setting: allow_sip_xml_action

Description: For security reasons this setting disables the possibility to parse vtech-XMLs received via SIP MESSAGE. When activated the phone accepts an entire xml-configuration within special SIP MESSAGEs. If it is "off", the phone just sends a 200 OK but does not parse the xml-configuration. If enabled one should provide a secure environment.
A SIP MESSAGE must have the following two headers, otherwise it is ignored for this purpose.
Content-Type: application/xml
Event: vtech-action

The body of the SIP message contains an xml as described here. Most likely one would make it contain only an action-section that holds one or more actions that fire "on notify"

Values: on, off

Default: off

Setting: allow_wizard_abort

Description: Turn this setting on if you want to abort the logon or initial setup wizard. Switch it off if you want only a system information. To abort a wizard make a long press on the 'cancel' key.

Values: on, off

Default: on

Setting: always_delegate_forward

Description: This setting is only available for LYNC. It can make a delegate always reachable on behalf of the boss. Even if the Boss turns of call forwarding/simultaneous ringing, we reset to call forwarding on if always_deleg_forw is active. If always_deleg_sim is active, we reset to simultaneous ringing.

Values: on, off

Default: off

Setting: always_show_active_call

Description: This setting is used to configure the default behaviour in call waiting scenarios. Default value on will keep the active call on the display, regardless of any incoming calls. All user actions such as hold or transfer will effect the active call. Disabling this setting will display the latest incoming call (all actions will be applied to the call displayed)

Values: on, off

Default: on

Setting: answer_after_policy

Description: Incoming intercom-calls (i.e. those that use the Alert-Info SIP header) do not ring but go directly to connected. That is if the situation and this setting allow it.

- **off** - will disable auto-connect
- **always** - will enable auto-connect without restrictions
- **idle** - will allow auto-connect only when phone is in idle-screen
- **not_busy** - will allow auto-connect except for when the user is in an active call, i.e. holding a call will allow for intercom-interuptions, while any call in one of the following states will disable auto-connect: ringing, calling, connected, being held by the other call-partner.

Values: off, idle, not busy, always

Default: off

Setting: aoc_amount_display

Description: If your provider supports "Advice of Charge" (AOC) information (that is, the costs for your call) during or at the end of the call, you can turn on this feature by selecting one of the following options:

1. Select "Charged" to show the accumulated amount of the current call on the display
2. Select "Balance" to show the amount remaining on your account.

Values: off, charged, balance

Default: off

Setting: aoc_cost_pulse

Description: Specify how much money one pulse costs (for example, 0.12 means 12 cents per pulse).

Values: float

Default: 1

Setting: aoc_pulse_currency

Description: Sets the currency symbol that will be shown next to the amount (for example, $).

Values: character

Default: $

Setting: area_code

Description: This setting is used for specifying standard area codes which are to be substituted in LDAP search requests.

Values: valid area code

Default: blank

Setting: attended_transfer_on_ringing

Description: Setting has been introduced to select between two different call transfer behaviours.

Consider the following flow:
A calls B
B picks up
A and B converse (A and B have an confirmed dialog)
...
B puts A on hold
...
B calls C
C is ringing, but does not yet pick up (B and C have an early dialog)
B transfers A to C:

B sends C a CANCEL (only if attended_transfer_on_ringing = off [old behaviour])

B sends A a REFER without replaces.

A sends an INVITE to C

...
A and C converse

So, setting this value to "on" will avoid the CANCEL request and thus avoiding a possible "missed call entry" in some environments on party C.

Values: on, off

Default: off

Setting: auth_tmp_pass

Description: **Internal**

This setting holds temporarily used data which should not be set or changed by any means. This setting cannot be provisioned.

Values: Do not change the vaue of this setting.

Default: empty

Setting: auth_tmp_realm

Description: **Internal**

This setting holds temporarily used data which should not be set or changed by any means. This setting cannot be provisioned.

Values:	Do not change the vaue of this setting.
Default:	empty

Setting:	auto_connect_indication
Description:	If you want to become informed with an audible indication when an incoming call (intercom call too) is automatically answered by your phone, select “on”.
Values:	on, off
Default:	on

Setting:	auto_connect_indication_tone
Description:	Optional specify the autoconnect indication tone Builtin value is "528 500 100 1", where the first value is the frequency in Hz, second value is the duration the tone will be played (milliseconds), third value is the duration the tone won't be played (milliseconds), fourth value is the loop count, starting by 1 (played one time).
Values:	{integer, integer, integer, integer}
Default:	blank

Setting:	auto_dial
Description:	This setting is switched off by default. You can set a timeout (in seconds) after which a number is dialed automatically without pressing or taking the handset off the hook. **Note:** Apart from the predefined values in the drop-down form, you can set other values either through provisioning or single HTTP GET requests.
Values:	off, integer
Default:	3

Setting:	auto_logoff_time
Description:	After turning back to idle state and specified amount of time in minutes, all identities are removed.
Values:	integer

Default: blank

Setting: auto_reboot_on_setting_change

Description: This setting may be used to enable the auto reboot feature during provisioning but preserve old behaviour if needed. Some settings need a reboot to get applied (i.e. vlan, dhcp, ip_address, etc.).
When using this setting in the provisioning file, please remember:

- **A change of this setting takes effect on the settings following it in the provisioned settings file only, so if you like to have it effect all settings in the provisioned settings file, put it at the top of the file.**
- **This is a setting just like any other setting. If this setting is turned on, it stays on. So after a reboot, the setting is still on, even if it isn't mentioned at all in the new settings file. If you experience a constantly rebooting phone, set log level to 7 and see (via syslog server) which setting causes the loop.**

Values: on, off

Default: off

Setting: auto_redial

Description: In case of busy signal, the phone will make an automatic attempt to redial the same number if this setting is turned on. The redial time is dependent on the parameter **auto_redial_value**.

Values: on, off

Default: off

Setting: auto_redial_value

Description: If the parameter **auto_redial** is on, the value of this setting is used to redial the same number in case of busy signal.

Values: integer

Default: 10

Setting: automatic_key_configuration_targets

Description: Helper for parameter user_keys_to_be_configured_on_first_registration that defines where first to look for free keys that can be re-configured.

Valid Values:
Space-separated list of key-locations/-blocks:

- **side:** these are the keys on the right side of the display
- **expansion:** these are the keys on attached expansion modules, i.e. the VSP08
- **line_block:** these are the array of line keys on most of our models that are not related to the main display

Values: Space-separated list of key-locations/-blocks

Default: side expansion line_block

Setting: away_timeout

Description: Determines the number of minutes of inactivity after which the phone will report its state as "away". Activity is defined as going off-hook. A value of zero means "away" will never be reported. If the value of this setting is smaller than that of inactive_timeout, the setting has no effect.

Values: integer

Default: 40

Setting: backlight

Description: Sets the display-brightness/backlight intensity for when the phone is active.

Values: integer between 3 and 15

Default: 7

Setting: backlight_idle

Description: Sets the display-brightness/backlight intensity for when the phone is doing nothing. See also parameter dim_timer.

Values: integer between 3 and 15

Default: 1

Setting: blf_directed_call_pickup

Description:	Allows use of different "Feature Access Codes" of service provider defined to Directed Call Pickup.
Values:	Feature Access Codes
Default:	*97

Setting:	blf_park_pickup
Description:	Allows use of different "Feature Access Codes" of service provider defined to Call Park Retrieve.
Values:	Feature Access Codes
Default:	*88

Setting:	block_url_dialing
Description:	You can block the dialing of SIP URLs by turning this setting on. In this case, only numeric numbers will be allowed as input.
Values:	on, off
Default:	on

Setting:	cache_contact_details
Description:	This parameter is used to deactivate the caching of specific contact details beyond call boundaries. When set to "off", subsequent calls from the same contact (determined by the SIP URI) do not use cached contact details. **Note:** Currently, only the display name is affected by this setting. For server type **Broadsoft**, the default is "off".
Values:	on, off
Default:	on

Setting:	cache_sip_authorization

Description: When this setting is set to 'on', the phone will cache the 'nonce', 'qop', 'opaque' and 'realm' parameters from the initial challenge, as well as the user credentials, and present unbidden an Authorization header (or Proxy-Authorization, depending on the challenge it received) calculated from these cached credentials in the next request it sends on the same identity. The nonce count is incremented on each subsequent message. The server may send a 'nextnonce' in the (Proxy-)Authentication-Info header of the response. The phone will replace the cached nonce with the value of the 'nextnonce' parameter and reset the nonce count. When this setting is set to 'off' the phone will not include any credentials in the next request and must be re-challenged by the server if continued authentication is desired.

Values: on, off

Default: on

Setting: call_completion

Description: Turning this setting to “on” will prompt the user to activate call completion, if possible, while calling a number. When the called party becomes available again, your phone will be able to automatically redial the number. CCNR (on not reachable) as well as CCBS (on busy) is supported.

Values: on, off

Default: off

Setting: call_join_xfer

Description: When this feature is turned to “on” and you have exactly one person on hold, pressing transfer while in a call to another person will result in an immediate transfer of him/her to the held party. In the same scenario with this feature turned off or in scenarios with multiple parties on hold, you will be asked where to transfer to. You can then select between all the other calls you currently have and entering a number manually setting blind transfer).

Values: on, off

Default: off

Setting: call_logs

Description: Specifies whether the call logs should be stored locally or on the server.

Values: local, server

Default: local

Setting: call_screen_fkeys_on_connected

Description: This setting describes which function keys are shown on-screen when the phone displays a connected call (includes conferences).

- The function keys are listed in order from left to right. Example: With the setting "F_REC F_HOLD", the Record function key is shown on the first position from the left, the Hold key on the second one. The Left and Right keys (F_LEFT/F_RIGHT) are an exception to this rule; Left is always put on the first and Right on the last position.
- Some function keys are automatically hidden when they are of no use under current circumstances. Example: F_CONF_ON will not be shown when there are not enough parties available for a conference or when the maximum number of parties within the conference has been reached.
- It is possible to restrict each function key to certain states (Calling, Conference, Connected, Holding, Ringing and Transfer) or to special conditions (have_incoming_call: there is an incoming ringing call), have_only_connected_calls: all the calls on the device are in connected state (i.e. in a single 1o1 call or a conference) or have_multiple_established_calls: more than 1 call on the device is in connected or holding state). For example, if you want a function key to be available only during the transfer and holding states, add them to the function key settings in parentheses, e.g. "F_CONTACTPOOL(transfer,holding)".
- It is also possible to negate a state by placing the operator "not" in front of it. For example, "F_CONF_ON(not:Transfer)" hides the function key Start conference when there is more than one party on hold and you have pressed the Transfer key. The *not* must be in front of each keyword/state that is to be negated.

Attention: Use spaces only to separate the function keys. Do not use them before or inside the brackets or parentheses.

Related settings:
call_screen_fkeys_on_outgoing
call_screen_fkeys_on_incoming
call_screen_fkeys_on_holding
fkeys_on_dialing

Values: A space separated list of F_-keys

Default: F_CONF_ON F_HOLD F_TRANSFER(not:Transfer) F_PARKORBIT F_DUAL_AUDIO(not:Conference) F_NEXT_CALL_SCREEN F_DELETE_MSG HEADSET

Setting: call_screen_fkeys_on_holding

Description: This setting describes which function keys are shown on-screen when the phone displays a locally held call.

- The function keys are listed in order from left to right. Example: With the setting "F_REC F_HOLD", the Record function key is shown on the first position from the left, the Hold key on the second one. The Left and Right keys (F_LEFT/F_RIGHT) are an exception to this rule; Left is always put on the first and Right on the last position.
- Some function keys are automatically hidden when they are of no use under current circumstances. Example: F_CONF_ON will not be shown when there are not enough parties available for a conference or when the maximum number of parties within the conference has been reached.
- It is possible to restrict each function key to certain states (Calling, Conference, Connected, Holding, Ringing and Transfer) or to special conditions (have_incoming_call: there is an incoming ringing call), have_only_connected_calls: all the calls on the device are in connected state (i.e. in a single 1o1 call or a conference) or have_multiple_established_calls: more than 1 call on the device is in connected or holding state). For example, if you want a function key to be available only during the transfer and holding states, add them to the function key settings in parentheses, e.g. "F_CONTACTPOOL(transfer,holding)".
- It is also possible to negate a state by placing the operator "not" in front of it. For example, "F_CONF_ON(not:Transfer)" hides the function key Start conference when there is more than one party on hold and you have pressed the Transfer key. The *not* must be in front of each keyword/state that is to be negated.

Attention: Use spaces only to separate the function keys. Do not use them before or inside the brackets or parentheses.

Related settings:
call_screen_fkeys_on_outgoing
call_screen_fkeys_on_incoming
call_screen_fkeys_on_connected
fkeys_on_dialing

Values: A space separated list of F_-keys

Default: F_CONF_ON(not:Transfer) F_DIAL(Transfer) F_HOLD F_TRANSFER(not:Transfer) F_CONTACTPOOL(Holding,Transfer) F_NEXT_CALL_SCREEN F_ABS F_DELETE_MSG HEADSET

Setting: call_screen_fkeys_on_incoming

Description: This setting describes which soft keys are shown when phone displays an incoming ringing call.

- The function keys are listed in order from left to right. Example: With the setting "F_REC F_HOLD", the Record function key is shown on the first position from the left, the Hold key on the second one. The Left and Right keys (F_LEFT/F_RIGHT) are an exception to this rule; Left is always put on the first and Right on the last position.
- Some function keys are automatically hidden when they are of no use under current circumstances. Example: F_CONF_ON will not be shown when there are not enough parties available for a conference or when the maximum number of parties within the conference has been reached.
- It is possible to restrict each function key to certain states (Calling, Conference, Connected, Holding, Ringing and Transfer) or to special conditions (have_incoming_call: there is an incoming ringing call), have_only_connected_calls: all the calls on the device are in connected state (i.e. in a single 1o1 call or a conference) or have_multiple_established_calls: more than 1 call on the device is in connected or holding state). For example, if you want a function key to be available only during the transfer and holding states, add them to the function key settings in parentheses, e.g. "F_CONTACTPOOL(transfer,holding)".
- It is also possible to negate a state by placing the operator "not" in front of it. For example, "F_CONF_ON(not:Transfer)" hides the function key Start conference when there is more than one party on hold and you have pressed the Transfer key. The *not* must be in front of each keyword/state that is to be negated.

Attention: Use spaces only to separate the function keys. Do not use them before or inside the brackets or parentheses.

Related settings:
call_screen_fkeys_on_outgoing
call_screen_fkeys_on_connected
call_screen_fkeys_on_holding
fkeys_on_dialing

Values: A space separated list of F_-keys

Default: F_CONTACTPOOL(Transfer) F_NEXT_CALL_SCREEN F_DELETE_MSG HEADSET

Setting: call_screen_fkeys_on_outgoing

Description: This setting describes which soft keys are shown when phone displays a outgoing ringing call.

- The function keys are listed in order from left to right. Example: With the setting "F_REC F_HOLD", the Record function key is shown on the first position from the left, the Hold key on the second one. The Left and Right keys (F_LEFT/F_RIGHT) are an exception to this rule; Left is always put on the first and Right on the last position.
- Some function keys are automatically hidden when they are of no use under current circumstances. Example: F_CONF_ON will not be shown when there are not enough parties available for a conference or when the maximum number of parties within the conference has been reached.
- It is possible to restrict each function key to certain states (Calling, Conference, Connected, Holding, Ringing and Transfer) or to special conditions (have_incoming_call: there is an incoming ringing call), have_only_connected_calls: all the calls on the device are in connected state (i.e. in a single 1o1 call or a conference) or have_multiple_established_calls: more than 1 call on the device is in connected or holding state). For example, if you want a function key to be available only during the transfer and holding states, add them to the function key settings in parentheses, e.g. "F_CONTACTPOOL(transfer,holding)".
- It is also possible to negate a state by placing the operator "not" in front of it. For example, "F_CONF_ON(not:Transfer)" hides the function key Start conference when there is more than one party on hold and you have pressed the Transfer key. The *not* must be in front of each keyword/state that is to be negated.

Attention: Use spaces only to separate the function keys. Do not use them before or inside the brackets or parentheses.

Related settings:
call_screen_fkeys_on_incoming
call_screen_fkeys_on_connected
call_screen_fkeys_on_holding
fkeys_on_dialing

Values: A space separated list of F_-keys

Default: F_CALL_COMPLETION F_NEXT_CALL_SCREEN F_DELETE_MSG HEADSET

Setting: call_states_when_knocking

Description: List of call states in which knocking is played. When there is at least one connection which state is in the list, knocking is played otherwise it is not played.

Values: space-separated list of the following call states: connected holding on_hold calling ringback offhook

Default: connected calling holding on_hold ringback

Setting: call_states_with_local_party

Description: Names the call-states that will display the local identity involved in a call. Not Displaying the local party will result in more space and a cleaner/simpler look. If you are using your phone with only one identity, you'll probably want to set this setting to empty.

Values: space-separated list of the following call states:

- connected (you are connected to a remote party and can talk)
- holding (you have placed remote party on hold)
- on_hold (the remote party has placed you on hold)
- ringing (incoming call, ringing at your device)
- calling (outgoing call, not ringing yet)
- ringback (outgoing ringing call)

Default: ringing calling ringback

Setting: call_waiting

Description: Call Waiting Indication combines two functions:

"Call Waiting (CW)" can be enabled ("on", "visual only", "ringer") or disabled ("off"). This function allows the phone to receive more than one call at one time.

"Call Waiting Indication (CWI)" If Call Waiting is enabled ("on", "visual only", "ringer") the incoming caller extension is displayed in the lower left corner of the display. A short knocking signal can be heard simultaneously in the background of your current active call indicating another incoming call.

This setting is per identity.

Values:

- on -> Call Waiting enabled -> Visual and audio indication
- visual -> Visual but NO audio indication
- ringer -> same as "on" -> reserved for future ringtone audio indication
- off -> Call Waiting disabled -> only ONE call can be received

Default: on

Setting: calling_title

Description: SYSTEM INTERNAL

The title that appears in the calling state.

Values: string

Default: lang_calling

Setting: callrecord_dialed_costs

Description: Cost of the most recent dialed call records. The element with the lowest index marks the most recent call record.

Internal

Values: string

Default: blank

Setting: callrecord_dialed_local

Description: Caller local identity for the most recent dialed call records. The element with the lowest index marks the most recent call record.

Values: SIP URI string

Default: blank

Setting: callrecord_dialed_remote

Description: Destination string of the most recent dialed call records. The element with the lowest index marks the most recent call record.

Values: SIP URI string

Default: blank

Setting: callrecord_missed_costs

Description: Cost for the most recent missed call records. The element with the lowest index marks the most recent call record.

Values: string

Default: blank

Setting: callrecord_missed_local

Description: Destination local identity for the most recent missed call records. The element with the lowest index marks the most recent call record.

Values: SIP URI string

Default: blank

Setting: callrecord_missed_remote

Description: **Internal**

String representing the caller for the most recent missed call records. The element with the lowest index marks the most recent call record.

Values: SIP URI string

Default: blank

Setting: callrecord_received_costs

Description:	**Internal**
	Cost of the most recent received call records. The element with the lowest index marks the most recent call record.
Values:	String
Default:	blank

Setting:	callrecord_received_local
Description:	**Internal**
	Destination local identity for the most recent received call records. The element with the lowest index marks the most recent call record.
Values:	SIP URI string
Default:	blank

Setting:	callrecord_received_remote
Description:	**Internal**
	String representing the caller of the most recent dialed call records. The element with the lowest index marks the most recent call record.
Values:	SIP URI string
Default:	blank

Setting:	cancel_conference
Description:	When this setting is turned on, pressing the CANCEL-key will cause call-termination with all parties in conference.
	When this setting is turned off all parties will be held instead. HOLD-key always holds all conference members.
	For onhook/offhook it can be combined with setting "conf_hangup".
Values:	on, off
Default:	on

Setting:	cancel_desktop

Description: When this option is set to 'on' the desktop message will be cleared when pressing the CANCEL button, otherwise it remains until phone reboots.

Values: on, off

Default: off

Setting: cancel_missed

Description: When this option is set to 'on' the missed call list will be cleared when pressing the CANCEL button, otherwise it remains until phone reboots.

Values: on, off

Default: on

Setting: cancel_on_hold

Description: When this option is set to 'off', a call on hold cannot be cancelled by pressing the CANCEL button , but has to be taken up again and then canceled. This prevents the accidental cancellation of calls on hold.

Values: on, off

Default: off

Setting: cc_token

Description: **SYSTEM INTERNAL**

Temporary setting to store the value returned by registrar in X-VTECH-CCTOKEN header. It is used while dialing and later for call completion.

Values: Do not change the vaue of this setting.

Default: empty

Setting: cert_provisioning_service

Description:	This setting applies only to the UC edition. It is used to store the HTTP address of the certificate provisioning service provided in option 43 of the DHCP response. The phone will query for this information on start-up by broadcastig a DHCP INFORM message with the vendor class identifier (option 60) set to "MS-UC-Client" (UC edition only). This setting may be provisioned manually if the phone is in an environment where the DHCP server does not provide this information, however if the server response does contain the requested information, the setting will be overwritten. Without this setting sign-in with extension number and PIN is not possible.
Values:	HTTP URI
Default:	blank

Setting:	challenge_checksync
Description:	Turning this setting on enables challenge responses for Check-Sync requests.
Values:	on, off
Default:	off

Setting:	challenge_reboot
Description:	This setting enables and disables challenge responses for remote reboot requests.
Values:	on, off
Default:	off

Setting:	challenge_response
Description:	VTech phones can handle challenge responses on the phone. Turning this setting off will disable this feature and you will only be able to handle authentication through the web interface of the phone.
Values:	on, off
Default:	on

Setting:	chars_in_lower_case

Description: Define which characters should appear one after another each time numpad key INDEX (INDEX ranges from 0 to 9) gets pressed (when input mode is set to lower case letters).

Values: character strings

Default: blank

Setting: chars_in_upper_case

Description: Define which characters should appear one after another each time numpad key INDEX (INDEX ranges from 0 to 9) gets pressed (when input mode is set to upper case letters).

Values: character strings

Default: blank

Setting: check_fqdn_against_server_cert

Description: When on, the phone checks whether the FQDN of the server it is trying to connect to via TLS appears either as CN in the subject field or is listed in the DNS names of the Subject Altnernative Names extension of the certificate presented by the server. If the name is not found the certificate is rejected. Note: This is setting has no effect if TLS Server Authentication is turned off. The host name validation can be controlled with the setting host_name_validation_flags.

Values: on (UC Edition), off (Non-UC Edition)

Default: on

Setting: codec_priority_list

Description: Prioritize which codecs (audio-stream) the phone should use. Prioritized coma-separated list, most desired codec up front.

Values: Comma separated list of codec tokens

Default: g722,pcmu,pcma,gsm,g723,g726-32,aal2-g726-32,g729,telephone-event

---

Setting: codec_size

Description: Select the packet size in ms.

Please note that the following codecs only work with certain packet time values:

g723: 30 or 60 ms

gsm: 20,40 or 60 ms

Values: 10, 20, 30, 40, 60

Default: 20

Setting: codec_tos

Description: This option enables the phone to support quality of service (QOS) for RTP traffic in a network. This makes sense only if all parts of the involved network also support QOS.

Values: integer [0 - 255]

Default: 160

Setting: colleagues_ring_sound

Description: Phone book contact type specific ringers. Selection of the ring tone style that signals incoming calls dependent on the contact type of the caller in the local phone book.

Values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>, <Silent>, <Custom>

Default: Ringer1

Setting: conf_hangup

Description: Sometimes it is useful to disable the disconnection of a call when the handset is placed on hook, e.g., during conference calls or in handsfree-mode, etc. This is achieved by turning this setting off.

If set to "on" the behaviour is like the setting "cancel_conference". Otherwise only the audio device will switch with onhook/offhook.

This setting is per identity.

Values: on, off

Default: off

Setting: conferencing

Description: Contains a sip-uri for a conference room. Used by pressing conference keys. This setting depends on an identity. If 'conference' key was pressed the configured conference room of the active identity will be called. If no SIP-URI is configured the default behaviour is a local conference on the phone (min. 2 participants connected).

Values: SIP URI string

Default: blank

Setting: connected_title

Description: SYSTEM INTERNAL

The title that appears in the connected state.

Values: character strings

Default: lang_connected

Setting: contact_source_priority

Description: Prioritise which source for looking up details (names) to show in PUI takes priority. First one in list has highest priority.

See also related setting Prioritise PBX number lookup. When it is set to true, the SIP-source is put to the front of the list.

Values: Space seperated list containing: Memory, Abs, OcsContactList, Ldap, Ocip, InternalTbook, Sip, Vcard

Default: Ldap Tbook Sip Vcard Memory

Setting: contactquery_start_length

Description: Minimum number of chars required before starting the query (LDAP, ABS, ...)

Values: Integer >0

Default: 3

Setting: contrast

Description: Determines the display contrast, but should not be used, because each phone reacts differently to it dependend by example from the temperature etc. Its better to set it manually.

Values: Integer [1-15]

Default: 8

Setting: country_code

Description: This setting is used for specifying standard country codes which are to be substituted in LDAP search requests.

Values: standard country codes

Default: blank

Setting: csta_challenge

Description: This setting enables/disables the challenge of incoming sip requests on csta sessions like INVITE and INFO. If enabled and no user_pass or user_hash has been provided the request will be rejected.

0 - disabled, no challenge at all

1 - only the initial incoming csta INVITE will be challenged

2 - all incoming sip requests for csta sessions will be challenged

see also csta_control, sip_ip_dialin_content_types

Values: 0, 1, 2

Default: 0

Setting: csta_control

Description: Allows to remotely control the phone via CSTA protocol.

see also csta_challenge, sip_ip_dialin_content_types

Values: on, off

Default: on

Setting: custom_melody_url

Description: If you have chosen Custom Melody URL in one of the pull-down menus, you can specify an URL to your own ringing melody. The type of file that should be supplied to the phone is: PCM 8 kHz 16 bit/sample (linear) mono WAV

Values: HTTP URL

Default: blank

Setting: cw_dialtone

Description: Turning this setting on will play a dial tone when a call is being held, signalling the user that he/she is able to dial a second number. No dial tone is played when this setting is set to off.

Values: on, off

Default: on

Setting: date_us_format

Description: With this setting, you can select either U.S. (month/day) or European (day.month) format for displaying the time and date.

Values: on, off

Default: on

Setting: dfks

Description: Identity-Based setting.

Many SIP phone users prefer to use the buttons on their phone to activate features, such as Do Not Disturb (DND), rather than any web portal. This feature permits these SIP phone users to use the buttons on their phones in just this way. With this feature installed, supported SIP phones can synchronize with the Application Server on the status of the following features:

Do Not Disturb

Call Forwarding Always (CFA)

Call Forwarding Busy (CFB)

Call Forwarding No Answer (CFNA).

If a user changes the status of one of these features via the web portal or a feature access code (FAC), the Application Server notifies the phone about the status change. Conversely, if the user changes the feature status via a button on his/her phone, the phone notifies the Application Server of the status change. The synchronization protocol is based on the SIP events framework. To use this capability, the phone user must have a SIP phone that supports the as-feature-event event package.

Values: on, off

Default: off

Setting: dhcp

Description: Turn the use of DHCP for inquiring IP on or off with this option.

The phone will still use DHCP to inquire other data when this setting is turned off. It does so by sending a DHCP-inform-message containing the list of the desired parameters. The list may be configured with the setting dhcp_options_on_ inform.

Values: on, off

Default: on

Setting: dhcp_options_on_inform

Description: List of options to be inquired from dhcp-server when no IP is to be fetched (dhcp = off). The phone will send an dhcp-inform during boot-up should this list not be empty. Should the server provide other options than stated in this list, they will be ignored (accept 53).

See also Settings/dhcp_options_on_ip_aquire, which does something similar for when dhcp = on

Values: List of space separated integers 0 - 255

Default: 43 120 125

Setting: dhcp_options_on_ip_aquire

Description: List of options to be inquired from dhcp-server when IP is fetched (dhcp = on). Should the server provide other options than stated in this list, they will be ignored (accept 53 and 54).

See also Settings/dhcp_options_on_inform, which does something similar for when dhcp = off

Values: List of space separated integers 0 - 255

Default: 1 3 4 6 12 15 42 43 51 66 67 120 125 132 133

Setting: dhcp_v6

Description: This setting enables the use of ICMPv6 or DHCPv6 for inquiring IPv6 addresses.

Note:

Currently this is the only way of assigning IPv6 addresses to your VTech phone. Setting up static IPv6 addresses is currently not supported.

IPv6 address changes during operation cannot handled dynamically at the moment. Thus a restart of the phone is needed in order to use the new IPv6 address properly.

VALIDVALUE

autoconf --> SLAAC (ICMPv6) only

on --> autoconf & DHCPv6

off --> IPv6 switched off completely

Values: autoconf, on, off

Default:	off

Setting:	dial_from_wui
Description:	This setting controls whether dialing from the web UI is allowed, allowed only in admin mode (admin_only) or completely disabled.
Values:	admin_only, on, off
Default:	on

Setting:	dialnumber_us_format
Description:	When this setting is "on" AND the phone is set to a US time zone, any numbers you dial will be formatted on the display like the following examples: 1. National format: 9785550123 will be shown as (978) 555-0123; formatting will start when the 4th digit is entered. 2. Service numbers (depending on availability in your area): A service number beginning with 511, for example, will be shown as (511) -xxxx; formatting will start when the 4th digit is entered. 3. International access code (for dialing numbers outside NANP): Numbers beginning with the international access code 011 will be shown as 011-x-xxxxxx. Formatting will start when the 4th digit is entered; the country dialing code (the digit(s) enclosed by the two hyphens) can consist of one or more digits. Examples: After you have entered the four digits 0114, the display will show them as "011-4". Entering 9 as a fifth digit will result in "011-49-" because 49 is an existing country dialing code (Germany). Entering 2 as a fifth digit will result in "011-42" without the second hyphen because there is no "42" country dialing code; entering 0 as the sixth digit will result in "011-420-" because 420 is an existing country dialing code (Czech Republic). Note: U.S. dialnumber format is the default setting, but will only be activated when the selected time zone on the phone is a US time zone.
Values:	on, off
Default:	on

Setting: dialplan_count_failed_match_groups

Description: Defines how the backreferences (e.g. \3) inside our dialplan substitution patterns count. Historically, they only counted matched-groups that actually matched, ignoring the others.

See this example

Input: hello

RegEx: ((hell)(l?)(o))

with this setting = false

\0 : hello

\1 : hell

\2 : o

with this setting = true

\0 : hello

\1 : hell

\2 :

\3 : o

Values: on, off

Default: off

Setting: dialplan_for_keypaddial_only

Description: If set this setting to "on", dial plan will be applied to keypad dialing only, outgoing calls from call history or phonebook should ignore the dial plan.

If set this setting to "off", dial plan will be applied to all the dialing.

Values: on, off

Default: off

Setting: dim_timer

Description: Number of seconds after which to dim (phones with color display) or turn off the display backlight when nothing is happening.

Values: Integer

Default:	20

Setting:	directory_search_config
Description:	**Internal** Internal setting used to set up on-line telephone directory searches. The parameters are determined by the server type of the identity.
Values:	string
Default:	blank

Setting:	dirty_host_ttl
Description:	Specify the Time to Live (TTL) for dirty hosts in seconds. This means that, when a phone was unable to reach a host, the phone will not try to reach this host again until the time specifed in this field has elapsed. If this setting is 0 or empty, it has no effect (the host is set as "dirty" but only for 0 seconds, which means it will have no effect on future requests) See also: sip_request_timeout, sip_retry_t1, sip_health_check
Values:	integer
Default:	blank

Setting:	disable_blind_transfer
Description:	A boolean to disable blind transfer. If it is on, instead of blind transfer, on hitting the transfer key, the only call is put on hold and a prompt offered to make second call and a normal consultative transfer would follow. This setting was introduced for PBXs that dont support REFER.
Values:	on, off
Default:	off

Setting:	disable_deflection
Description:	A boolean to stop 3xx codes (e.g. 302 Moved temporarily). If the setting is on, a Busy Here is returned. Turning this setting on will also disable Call Deflect.
Values:	on, off

Default:	off

Setting:	disable_speaker
Description:	Turn this setting on to disable your speaker.
Values:	on, off
Default:	off

Setting:	disable_storing_changes
Description:	When turning this on, neither setting changes nor changes to the internal address book are ever saved to the permanent memory of the phone. Everything will be lost after reboot.
Values:	on, off
Default:	off

Setting:	disconnect_on_onhook
Description:	Sometimes it is useful to disable the disconnection of a call when the handset is placed on hook to switch to speaker audio. This is achieved by turning this setting off.
Values:	on, off
Default:	on

Setting:	disconnected_title
Description:	**Internal** Title that appears when a call is disconnected.
Values:	string
Default:	lang_terminated_finished

Setting:	disconnected_url_on_reject
Description:	If value is set to 'on', an action url for disconnect will be fired in case of rejecting a call.

Values:	on, off
Default:	off

Setting:	display_method
Description:	Specifies how incoming and outgoing calls are displayed: Full Contact: The complete URL is shown Name: Only the name is displayed Number: Only the number is displayed Name+Number: Name and number are displayed Number+Name: Number and name are displayed Please also note user_pui_treats_uri_username_as_fallback_for
Values:	full_contact, display_name, display_number, display_name_number, display_number_name
Default:	display_name

Setting:	dkey_directory
Description:	This is the value preprogrammed for the function key labeled "Directory".
Values:	valid keyevent ID
Default:	keyevent F_ADR_BOOK

Setting:	dkey_dnd
Description:	This is the value preprogrammed for the function key labeled "DND".
Values:	valid keyevent ID
Default:	keyevent F_DND

Setting:	dkey_fkey1

Description: Configures the corresponding softkey of idle screen like other "dkeys" (SETTINGS, HELP, PGM, ...). If it is set, it overrides the configured keys in the gui_fkey* settings.

CAUTION: The gui_fkey* settings are still and always used for choosing the label text/icon of the softkeys!

Values: valid keyevent ID

Default: blank

Setting: dkey_fkey2

Description: Configures the corresponding softkey of idle screen like other "dkeys" (SETTINGS, HELP, PGM, ...). If it is set, it overrides the configured keys in the gui_fkey* settings.

CAUTION: The gui_fkey* settings are still and always used for choosing the label text/icon of the softkeys!

Values: valid keyevent ID

Default: blank

Setting: dkey_fkey3

Description: Configures the corresponding softkey of idle screen like other "dkeys" (SETTINGS, HELP, PGM, ...). If it is set, it overrides the configured keys in the gui_fkey* settings.

CAUTION: The gui_fkey* settings are still and always used for choosing the label text/icon of the softkeys!

Values: valid keyevent ID

Default: blank

Setting: dkey_fkey4

Description: Configures the corresponding softkey of idle screen like other "dkeys" (SETTINGS, HELP, PGM, ...). If it is set, it overrides the configured keys in the gui_fkey* settings.

CAUTION: The gui_fkey* settings are still and always used for choosing the label text/icon of the softkeys!

Values: valid keyevent ID

Default:	blank

Setting:	dkey_hold
Description:	This is the value preprogrammed for the function key labeled "HOLD".
Values:	valid keyevent ID
Default:	keyevent F_HOLD

Setting:	dkey_retrieve
Description:	This is the value preprogrammed for the function key labeled “Retrieve”.
Values:	valid keyevent ID
Default:	keyevent F_RETRIEVE

Setting:	dkey_transfer
Description:	This is the value preprogrammed for the function key labeled "TRANSFER".
Values:	valid keyevent ID
Default:	keyevent F_TRANSFER

Setting:	dnd_mode
Description:	<on> means that the phone is in do not disturb (DND) mode, <off> is normal behavior. This setting is per identity.
Values:	on, off
Default:	off

Setting:	dnd_off_code
Description:	If the PBX is handling DND, it can be specified which star code disables this functionality at the PBX. VALIDVALUE e.g. <*74>, <*74>.
Values:	dialing string

Default: blank

Setting: dnd_on_code

Description: If the PBX is handling DND, it can be specified which star code enables this functionality at the PBX.

VALIDVALUE

e.g. <*74>, <*74>.

Values: dialing string

Default: blank

Setting: dns_a_queries_only

Description: Setting the value to on will force the phones dns stack to skip all DNS SRV and DNS NAPTR queries and only perform DNS A queries. Not recommended.

Values: on, off

Default: off

Setting: dns_cache_clear_timeout

Description: Specifies the optional amount of time before the phones internal dns cache gets completly cleared. On default the dns cache entries times out after their individual TTL given from the dns server has passed. You can shorten their TTL with this setting value.

Values: 0 (off) - 1209600

Default: blank

Setting: dns_domain

Description: Specify the DNS domain for your phone here. This parameter is mandatory in order to enable DNS searching.

Values: URL

Default: vtech.ca

Setting:	dns_domain_v6
Description:	Additional domain name for IPv6 networks. See also dns_domain.
Values:	URL
Default:	blank

Setting:	dns_fallback_time
Description:	Specifies an optional fallback time from secondary dns server to primary dns server. Default dns request failover behavior is to always query the server with the fastest response time. The value for this setting is set to "65535". If you want to always query the primary dns server first, specify a value of "0" here. Please note that if the primary server is really down, this also will delay your phone reactions up to 3 seconds for each new dns request during the server downtime. Or you can switch back after a defined time from secondary to primary server again (this will reduce the load and give your dns administrator some time to fix it). To do so please specify the given time value in seconds here.
Values:	0 - 65535
Default:	65535

Setting:	dns_server1
Description:	Specify the IP address of the DNS server for your network here. This parameter is extremely important for a proper functioning phone, so please make sure it is set up correctly.
Values:	IP address
Default:	10.88.162.10

Setting:	dns_server1_v6
Description:	Additional DNS server for IPv6. See also dns_server1.
Values:	IPv6 address
Default:	blank

Setting:	dns_server2
Description:	Specify the IP address of a backup DNS server for your network here.
Values:	IP address
Default:	10.88.162.6

Setting:	dns_server2_v6
Description:	Additional DNS server for IPv6. See also dns_server1.
Values:	IPv6 address
Default:	blank

Setting:	dns_server3_v6
Description:	Additional DNS server for IPv6. See also dns_server1.
Values:	IPv6 address
Default:	blank

Setting:	dns_server4_v6
Description:	Additional DNS server for IPv6. See also dns_server1.
Values:	IPv6 address
Default:	blank

Setting:	documentation_link
Description:	**SYSTEM INTERNAL** This setting holds the base link the questionmark icon shown at the web interface behind each setting is pointing to.
Values:	Any valid HTTP(S) URL; leaving this value blank switches off the questionmark icons at the web interface.
Default:	blank

Setting:	dst

Description:	**Internal** - Format 1 (usually used): offset -> time difference in sec mm.ww.dd -> start date of daylight saving (mm: month [01..12]; ww:week [01..05] e.g. 05 = last week in month; dd:day of the week [01..07]) hh:mm:ss -> start time of daylight saving (hh: hours [00..23]; mm:minutes [00..59]; ss:seconds [00..59]] mm.ww.dd -> end date of daylight saving (mm: month [01..12]; ww:week [01..05] e.g. 05 = last week in month; dd:day of the week [01..07]) hh:mm:ss -> end time of daylight saving (hh: hours [00..23]; mm:minutes [00..59]; ss:seconds [00..59]] - Example: e.g. for Germany -> Daylight saving starts on a Sunday (07) of the last week (05) in March (03) at 2 o clock in the morning (2 am (02:00:00)) and ends on a Sunday (07) of the last week (05) of October (10) at 3 o´clock in the morning (3 am (03:00:00)): <3600 03.05.07 02:00:00 10.05.07 03:00:00> - Format 2 (seldomly used): offset -> time difference in sec dd.mm -> start date of daylight saving (dd: day [01..31]; mm: month [01..12]) hh:mm:ss -> start time of daylight saving (hh: hours [00..23]; mm:minutes [00..59]; ss:seconds [00..59]] dd.mm -> end date of daylight saving (dd: day [01..31]; mm: month [01..12]) hh:mm:ss -> end time of daylight saving (hh: hours [00..23]; mm:minutes [00..59]; ss:seconds [00..59]] - Example: In the below example string Daylight saving starts on 22. March at 3 o clock in the morning (3 am (03:00:00)) and ends on 22. September at 4 o´clock in the morning (4 am (04:00:00)): <3600 22.03 03:00:00 22.09 04:00:00>
Values:	time format string
Default:	blank

Setting: dtmf_handset_phone

Description: Hearing DTMF tones during call initiation (dialing) is enabled only for some tone schemes (due to each country's ISDN standardization). However, some customers asked to be able to disable these tones when in handset mode, for privacy reasons. You can disable them by turning this setting to off. Please note that this setting has no effect on speaker/headset mode.

Here is the list of the tone schemes this feature will affect:

Australia, China, Denmark, Great Britain, India, Italy, Japan, Mexico, Netherlands, New Zealand, United States

Note: During a call the DTMF echo is always audible.

Values: on, off

Default: on

Setting: dtmf_micro_delay

Description: Specifies the delay in milliseconds after a DTMF tone has been played and the microphone becomes active again.

If a greater value than 1000 milliseconds is needed, just delete the local DTMF output entirely with the setting: dtmf_volume.

Values: 0 (off) - 1000 (max)

Default: 0

Setting: dtmf_speaker_phone

Description: Hearing DTMF tones during call initiation (dialing) is enabled only for some tone schemes (due to each country's ISDN standardization). However, some customers asked to be able to disable these tones when in speaker mode, for privacy reasons. You can disable them by turning this setting to off. Please note that this setting has no effect on handset/headset mode.

Here is the list of the tone schemes this feature will affect:

Australia, China, Denmark, Great Britain, India, Italy, Japan, Mexico, Netherlands, New Zealand, United States

Note: During a call the DTMF echo is always audible.

Values: on, off

Default: on

Setting: dtmf_volume

Description: Specifies the volume of local played DTMF key tones .

Values: 0 (off) -15 (max)

Default: 8

Setting: edit_mode_for_passwords

Description: Specifies the default edit-mode used for inputting passwords in PUI.

Values: 123, abc, ABC

Default: 123

Setting: emergency_accepted_callkeys

Description: Comma separated list of keys who will be accepted in an emergency call.

Values: comma separated keynames

Default: STATE_AUTO_LEAVE,OFFHOOK,ONHOOK,CANCEL,F_CANCEL,F_HOLD,VOLUME_UP,VOLUME_DOWN,SPEAKER,HEADSET,*,#,0,1,2,3,4,5,6,7,8,9

Setting: emergency_proxy

Description: Outbound proxy for emergency numbers.

Values: URI

Default: blank

Setting: empty_tls_client_cert

Description: If this setting is on the phone will use empty client certificate in TLS connections.

Values: on, off

Default: off

Setting: enable_e164_substitution

Description: Setting used for LDAP directory search. Substitutes + for 00 etc.

Values: on, off

Default: on

Setting: enable_keyboard_lock

Description: Enable keyboard locking via star-key or timeout. On OCS servers this setting is turned on if the inband provisioning parameter ucEnforcePinLock has a value of "true". If its value is "false" this setting is left unchanged (i.e. it may be turned on or off at the user's discretion). Note that even when this setting is turned off, the user can still lock/unlock the phone via the web interface directly by changing the phone's lock state (see keyboard_lock).

Values: on, off

Default: on

Setting: enable_predial_mode

Description: This setting is used to enable the pre-dialing mode. In pre-dialing mode, if users operate the keypad, it will not activate any Line key or Speakerphone key. And it will not dial out any number even the Auto-Dial was enabled.

Values: on, off

Default: off

Setting: enable_rport_rfc3581

Description: Enables or disables rport parameter for the Via header field. The default setting allows a client to request that the server send the response back to the source IP address and port from which the request originated. However in some environments it might be desired to switch this parameter off. In order to do so, please turn this setting <OFF> via mass deployment.

Values: on, off

Default: on

Setting: enter_number_title

Description: **SYSTEM INTERNAL**

Title that appears in the edit state for dialing a number.

Values: string

Default: lang_enter_number

Setting: enum_suffix

Description: When using ENUM, you can specify a service suffix here, if desired. There is more than one service that supports ENUM lookups, and you can select here which one you want to use. You can enter a comma separated list of route domains for ENUM lookup. Leave the default value e164.arpa if you don't know better.

Values: comma separated list of route domains

Default: e164.arpa

Setting: eth_net

Description: This setting is used to configure the NET port of the phone's integrated Ethernet switch. The setting value is a comma-separated list of three items: <speed>,<pause>,<advertisement>

Whereas each item has the following meaning:

<speed> - setting forced Ethernet speed or enabling auto-negotiation

<pause> - enable Ethernet flow control via PAUSE frame (empty value leaves the feature disabled)

<advertisement> - space-separated list of properties to advertise (empty advertises all supported properties)

For example, the following setting value would auto-negotiate the Ethernet speed, while leaving the pause feature untouched (empty value between the two commas) and advertising that only 1000MBit and 100MBit full duplex can be auto-negotiated:

auto,,auto 1000full 100full

Note: The values 1000full and 1000half are only supported by phones with an integrated Gigabit Ethernet switch.

Values: A comma-sparated list with these three items (<pause> and <advertisement> may be left blank):

- <speed> - one of the following values:
  - auto
  - 10half
  - 10full
  - 100half
  - 100full
  - 1000full
- <pause> - one of the following values:
  - tx_rx_off
  - tx_on
  - rx_on
  - tx_rx_on
- <advertising> - a combination of the following values (space-separated):
  - auto
  - 10half
  - 10full
  - 100half
  - 100full
  - 1000full

Default: auto

Setting: eth_pc

Description: This setting is used to configure the PC port of the phone's integrated Ethernet switch. The setting value is a comma-separated list of three items: <speed>,<pause>,<advertisement>

Whereas each item has the following meaning:

<speed> - setting forced Ethernet speed or enabling auto-negotiation

<pause> - enable Ethernet flow control via PAUSE frame (empty value leaves the feature disabled)

<advertisement> - space-separated list of properties to advertise (empty advertises all supported properties)

For example, the following setting value would auto-negotiate the Ethernet speed, while leaving the pause feature untouched (empty value between the two commas) and advertising that only 1000MBit and 100MBit full duplex can be auto-negotiated:

auto,,auto 1000full 100full

Note: The values 1000full and 1000half are only supported by phones with an integrated Gigabit Ethernet switch.

Values: A comma-sparated list with these three items (<pause> and <advertisement> may be left blank):

- <speed> - one of the following values:
    - auto
    - 10half
    - 10full
    - 100half
    - 100full
    - 1000full
- <pause> - one of the following values:
    - tx_rx_off
    - tx_on
    - rx_on
    - tx_rx_on
- <advertising> - a combination of the following values (space-separated):
- auto
- 10half
- 10full
- 100half
- 100full
- 1000full

Default: auto

Setting: ethernet_detect

Description: When this option is set to 'on', the phone will display a warning message and a status message when it loses ethernet connectivity. When WLAN is configured, only the status message is diplayed.

Values: on, off

Default: on

Setting: ethernet_replug

Description:	Choose the action to be performed after the network connection is reestablished: Ignore Reboot Reregister all active Identities.
Values:	nothing, reboot, reregister
Default:	reregister

Setting:	exchange_refresh_in_secs
Description:	Currently the phone is polling the exchange server for latest 'appointments for today' related data each exchange_refresh_in_secs seconds. To disable the 'click to join' and 'appointments for today' functionality, set setting 'exchange_refresh_in_secs' to '0'. Then no calendar items are retrieved anymore. Thus the menu item is made invisible as well.
Values:	unsigned integer
Default:	60

Setting:	extension_monitoring_group
Description:	For this setting to have any effect user_allow_inc_dialog_subscribe must be on. It allows the user to restrict extension monitoring to a group of users using one of two possible mechansism: shared secret or contact group. To use the shared secret mechanism simply enter a pass phrase into this field. All users using the same pass phrase can monitor each other's extension. Note that this mechanism does not work with OCS/Lync. Note also that the pass phrase must not start with '{'. The contact group mechanism is currently available only with OCS/Lync. Enter the name of a group on your contact list to allow all members of that group to monitor your extension. To distinguish a contact group from a pass phrase surround the group name with curly braces. For example: {My Pickup Group}. Entering empty braces {} allows everyone on your contact list to monitor your extension (this also works with non-OCS buddy lists).
Values:	string
Default:	blank

Setting: family_ring_sound

Description: Phone book contact type specific ringers. Selection of the ring tone style that signals incoming calls dependent on the contact type of the caller in the local phone book.

Values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>, <Silent>, <Custom>

Default: Ringer1

Setting: filter_registrar

Description: If set to on, all SIP packets not coming from the registrar/proxy will be ignored. For security reasons, on is the default setting. This may cause big problems in an environment where SIP packets from other sources also have to be accepted for proper functionality! You have to disable it to make a call flow work which isn't going via the proxy only !

Values: on, off

Default: on

Setting: firmware

Description: **SYSTEM INTERNAL**

URL of the firmware image file

Values: URL

Default: blank

Setting: firmware_interval

Description: This setting specifies the time interval (in minutes) for polling the firmware configuration file. The start time counter is reset on each reboot.

Values: integer

Default: blank

Setting: firmware_status

Description: URL of the firmware configuration file

Values: URL

Default:	blank

Setting:	firmware_version
Description:	**SYSTEM INTERNAL** Contains the version string of the currently installed application firmware.
Values:	String
Default:	VTechET605-SIP x.x.x

Setting:	fkey
Description:	Defines the type of the free programmable function key x.
Values:	auto_answer, blf, button, BW_Anywhere, BW-ACD, BW-RemoteOffice, BW-ServerBLF, call_agent, conference, Contact List Buddy , dest, dtmf, icom, ivr, keyevent F_ACCEPTED_LIST, keyevent F_ADR_BOOK, keyevent F_CALL_LIST, keyevent F_CONFERENCE, keyevent F_CONTACTS, keyevent F_DELETE_MSG, keyevent F_DENYALL, keyevent F_DIALOG, keyevent F_DIRECTORY_SEARCH, keyevent F_DND, keyevent F_FAVORITES, keyevent F_HOLD, keyevent F_HOLD_PRIVATE, keyevent F_HOTELING, keyevent F_LABEL_PAGE_NEXT, keyevent F_LABEL_PAGE_PREV, keyevent F_LOGOFF_ALL, keyevent F_MISSED_LIST, keyevent F_MUTE, keyevent F_NEXT_ID, keyevent F_NONE, keyevent F_OCIP, keyevent F_PRESENCE, keyevent F_PREV_ID, keyevent F_REBOOT, keyevent F_REC, keyevent F_REDIAL, keyevent F_REDIRECT, keyevent F_RETRIEVE, keyevent F_RINGER_SILENT, keyevent F_SERVER_AB, keyevent F_SETTINGS, keyevent F_STATUS, keyevent F_SUPPORT, keyevent F_TRANSFER, keyevent F_ZONES, keyevent HEADSET, line, multicast, none, orbit, p2t, presence, recorder, redirect, SendSipInfo, speed, Starcode, transfer, url, UserInputAndSendSipInfo, xml, XMPP-ContactPresence
Default:	line

Setting:	fkey_delay_timeout
Description:	This setting is measured in seconds and applies for keys set to type "Park+Orbit". It will prohibit repeated pressing of this key-type for the time set.
Values:	integer
Default:	5

Setting:	fkey_label_overrides_xml_label
Description:	When both the **fkey_label** setting and the **XML description** setting provide a label for a self labeling key, this setting determines which takes precedence. When true, the contents of the fkey_label setting is used, else the contents generated in the XML description. This setting has no effect if only one of the two are set.
Values:	on, off
Default:	off

Setting:	fkeys_on_dialing

Description: This setting describes which soft keys are shown when phone displays the dial screen.

- This setting is available on all models with a screen.

- The function keys are listed in order from left to right. Example: With the setting "F_DIALMODE F_BACK", the edit mode function key is shown on the first position from the left, the Backspace key on the second one.

- Some function keys are automatically hidden when they are of no use under current circumstances. Example: F_REDIAL will not be shown when there are no numbers in the redial-list.

- It is possible to restrict each function key to certain conditions (edit_for_transfer: entering target for a blind transfer, have_incoming_call: there is an incoming ringing call, have_only_connected_calls: all the calls on the device are in connected state (i.e. in a single 1o1 call or a conference) or have_multiple_established_calls: more than 1 call on the device is in connected or holding state). For example, if you want a function key to be available only when there is an incoming ringing call, add the keyword to the function key settings in parentheses, e.g. "F_WHATEVER(have_incoming_call)".

--It is also possible to negate this by placing the operator "not" up front. For example, "F_WHATEVER(not:have_incoming_call)" only shows the function key when there isn't an incoming ringing call.

--You may also combine the keywords like this: "F_WHATEVER(edit_for_transfer,not:have_incoming_call)". In this case the key only shows when you are either entering the target for a blind transfer or there isn't an incoming ringing call.

--Attention: Use spaces only to separate the function keys. Do not use them before or inside the brackets or parentheses.

Related settings:

call_screen_fkeys_on_incoming

call_screen_fkeys_on_outgoing

call_screen_fkeys_on_connected

call_screen_fkeys_on_holding

Values: space separated list of F keys

Default: F_DIALMODE F_BACK F_DEFLECT(not:edit_for_transfer) F_ACCEPT_CALL(not:edit_for_transfer) F_SAFETRANSFER(edit_for_transfer) F_CONTACTPOOL F_REDIAL

Setting: flood_tracing

Description: Set to 'off' when you do not want to log REGISTER-, SUBSCRIBE-, NOTIFY- nor SERVICE-SIP-messages in WUI-sip-trace.

Values: on, off

Default: on

Setting: friends_ring_sound

Description: Phone book contact type specific ringers. Selection of the ring tone style that signals incoming calls dependent on the contact type of the caller in the local phone book.

Values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>, <Silent>, <Custom>

Default: Ringer1

Setting: fwd_all_enabled

Description: If turned on all calls to the associated identity are diverted to the number specified.

Diversion can either be handled by the phone or by a server, see setting using_server_managed_fwd_all.

Values: on, off

Default: off

Setting: fwd_all_off_code

Description: If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.

Values: starcode

Default: blank

Setting: fwd_all_on_code

Description: If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.

Values: starcode

Default:	blank

Setting:	fwd_all_target
Description:	The redirection target, when redirection is always active (setting fwd_all_enabled).
Values:	SIP URI or number
Default:	blank

Setting:	fwd_busy_enabled
Description:	If turned on and a call is in progress while a 2nd one is incoming, the second caller is diverted to the number specified. Note: This will only work if call waiting is disabled. Diversion can either be handled by the phone or by a server, see setting using_server_managed_fwd_busy.
Values:	on, off
Default:	off

Setting:	fwd_busy_off_code
Description:	If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.
Values:	starcode
Default:	blank

Setting:	fwd_busy_on_code
Description:	When set, the given starcode, appended by the redirection target, will be dialed whenever redirection when busy gets enabled or changes the target for the specific identity.
Values:	starcode
Default:	blank

Setting:	fwd_busy_target

Description: Specifies the number to which calls will be diverted when the phone is busy (setting fwd_busy_enabled). Note: This will only work if call waiting (setting call_waiting) is disabled .

Values: SIP URI or number

Default: blank

Setting: fwd_time_enabled

Description: If turned any incoming call will be diverted to the specified number (setting fwd_time_target) after the specified time (setting fwd_time_enabled) has elapsed.

Diversion can either be handled by the phone or by a server, see setting using_server_managed_fwd_time.

Values: on, off

Default: off

Setting: fwd_time_off_code

Description: If the PBX is handling the redirection, it can be specified which star code disables this functionality at the PBX for the specific identity.

Values: starcode

Default: blank

Setting: fwd_time_on_code

Description: When set, the given starcode, appended by the redirection target, will be dialed whenever redirection after timeout gets enabled or changes the target for the specific identity.

Values: starcode

Default: blank

Setting: fwd_time_secs

Description: Specifies the timeout in seconds after which the call will be diverted.

Values: integer

Default: blank

Setting: fwd_time_target

Description: Specifies the number to which calls will be diverted after the specified time (setting fwd_time_secs) has elapsed.

Values: SIP URI or number

Default: blank

Setting: garbage_timeout

Description: Time to call the internal garbage collection for the contact pool or presence informations cyclic. Have a look on the memory webside of the phone. The contacts and presence memory usage are listed on this page.

Values: integer

Default: 300

Setting: gateway

Description: This setting shows the IP address of the default IP gateway (NOT the VoIP gateway). It is the address to which the packets get routed when the desired packet address is not in the current subnet. Setting up this parameter is mandatory in order to reach an external network.

Values: IP address

Default: 10.88.3.149

Setting: gateway_vlan

Description: SYSTEM INTERNAL (Reboot required)

This setting shows the IP address of the default VLAN IP gateway (NOT the VoIP gateway). It is the address to which the packets get routed when the desired packet address is not in the current subnet.

Values: IP address

Default: blank

Setting: general_purpose_xml_descriptions

Description:	There are several (varies by fw-version) of these general purpose xml descriptions (gp-xml) available. They offer a way of creating xml-entites without tying it to a specific key. You can also decide to use a gp-xml as context-key on screen by inserting "GP_XML[n]" (with n being the index of the gp-xml, first one is 0) into one of these settings: call_screen_fkeys_on_incoming call_screen_fkeys_on_outgoing call_screen_fkeys_on_connected call_screen_fkeys_on_holding fkeys_on_dialing
Values:	XML definition
Default:	blank

Setting:	global_missed_counter
Description:	When set to <on>, the phone will count missed calls on all registered lines and show them on the phone. If turned <off>, missed calls for the active identity will be shown on the display.
Values:	on, off
Default:	on

Setting:	goto_monitor_state_on_line_activity
Description:	When any of your monitored lines shows an activity (other than idle), the phone will automatically display the call-monitor state. See also settings: pui_states_allowing_state_switch_on_activity and goto_virtual_keys_state_on_activity.
Values:	on, off
Default:	off

Setting:	goto_virtual_keys_state_on_activity

Description: When one of the virtual p-keys shows a monitored line that is not idle, the phone will automatically show the virtual key state.

Please also see: states-ignored-in-goto-vkeys-on-activity, goto_monitor_state_on_line_activity and pui_states_allowing_state_switch_on_activity

Values: on, off

Default: on

Setting: guess_number

Description: With this parameter, the number guessing functionality can be enabled. This is the automatic number completion which will begin after you have entered the minimum number of digits.

Values: on, off

Default: off

Setting: guess_start_length

Description: Specify the minimum number of digits that must be entered before 'Number Guessing' will begin. This setting also defines when ldap-lookup should begin when entering a number.

Values: integer

Default: 4

Setting: gui_fkey_label

Description: Defines the short label to be used to describe the dkey. The index ranged from 0 to 3, where 0 is the first dkey on the left.

Values: string

Default: blank

Setting: gui_fkey1

Description: Context-Sensitive (S) keys can be predefined for the Idle Screen.

Values:	F_ADR_BOOK (Directory), F_ACCEPTED_LIST (Accepted Calls), F_CALL_LIST (Call Lists), F_CONTACTS (Contacts), F_DIALOG (Monitor Calls), F_DIRECTORY_SEARCH (LDAP Directory), F_DND (DND), F_MISSED_LIST (Missed Calls), F_NEXT_ID (Next Outgoing ID), F_PREV_ID (Prev. Outgoing ID), F_REDIAL (Redial), F_REDIRECT (Forward All), F_RETRIEVE (Retrieve), F_SETTINGS (Menu), F_SUPPORT (Help), F_TRANSFER (Transfer)
Default:	keyevent F_SETTINGS

Setting:	gui_fkey2
Description:	Context-Sensitive (S) keys can be predefined for the Idle Screen.
Values:	F_ADR_BOOK (Directory), F_ACCEPTED_LIST (Accepted Calls), F_CALL_LIST (Call Lists), F_CONTACTS (Contacts), F_DIALOG (Monitor Calls), F_DIRECTORY_SEARCH (LDAP Directory), F_DND (DND), F_MISSED_LIST (Missed Calls), F_NEXT_ID (Next Outgoing ID), F_PREV_ID (Prev. Outgoing ID) |F_REDIAL (Redial), F_REDIRECT (Forward All), F_RETRIEVE (Retrieve), F_SETTINGS (Menu), F_SUPPORT (Help), F_TRANSFER (Transfer)
Default:	keyevent F_CALL_LIST

Setting:	gui_fkey3
Description:	Context-Sensitive (S) keys can be predefined for the Idle Screen.
Values:	F_ADR_BOOK (Directory), F_ACCEPTED_LIST (Accepted Calls), F_CALL_LIST (Call Lists), F_CONTACTS (Contacts), F_DIALOG (Monitor Calls), F_DIRECTORY_SEARCH (LDAP Directory), F_DND (DND), F_MISSED_LIST (Missed Calls), F_NEXT_ID (Next Outgoing ID), F_PREV_ID (Prev. Outgoing ID) , F_REDIAL (Redial), F_REDIRECT (Forward All), F_RETRIEVE (Retrieve), F_SETTINGS (Menu), F_SUPPORT (Help), F_TRANSFER (Transfer)
Default:	keyevent F_REDIRECT

Setting:	gui_fkey4
Description:	Context-Sensitive (S) keys can be predefined for the Idle Screen.

Values:	F_ADR_BOOK (Directory), F_ACCEPTED_LIST (Accepted Calls), F_CALL_LIST (Call Lists), F_CONTACTS (Contacts), F_DIALOG (Monitor Calls), F_DIRECTORY_SEARCH (LDAP Directory), F_DND (DND), F_MISSED_LIST (Missed Calls), F_NEXT_ID (Next Outgoing ID), F_PREV_ID (Prev. Outgoing ID) |F_REDIAL (Redial), F_REDIRECT (Forward All), F_RETRIEVE (Retrieve), F_SETTINGS (Menu), F_SUPPORT (Help), F_TRANSFER (Transfer)
Default:	keyevent F_SUPPORT

Setting:	handset_agc
Description:	Turn this setting off to disable the Automatic Gain Control (AGC) of the handset.
Values:	on, off
Default:	on

Setting:	headset_active
Description:	This setting activates the headset.
Values:	on, off
Default:	off

Setting:	headset_agc
Description:	Turn this setting off to disable the Automatic Gain Control (AGC) of the headset.
Values:	on, off
Default:	on

Setting:	headset_cmd_pause
Description:	Defines the time in milliseconds that the phone waits between sending commands to the headset. Different Headset types have different timing. If you experience problems like your Headset is sometimes not 'online' like it should be, increase this pause.
Values:	positive integer
Default:	700

Setting: headset_rings_once

Description: If "on" repeated ringing on headsets is disabled.

Values: on, off

Default: off

Setting: held_by_title

Description: SYSTEM INTERNAL

Title that appears when a call is held by the remote party.

Values: String

Default: lang_held_by

Setting: hide_identity

Description: Setting this to 'true' will make the identity disappear from the idle-screen.

This setting depends on is_voice_identity, when that setting is disabled, the identity will automatically be hidden.

Values: on, off

Default: off

Setting: high_mic_gain

Description: With this setting you can increase the microphone volume. The default microphone volume is inside the TIA norm. If you need a higher microphone sensibility you can set this setting to on. But this is at your own risk and then you are above the TIA norm.

Values: on, off

Default: off

Setting: holding_reminder

Description: When this option is set to 'on', the phone reminds you with a short beep that you still have somebody on hold.

Values: on, off

Default:	on

Setting:	host_name_validation_flags
Description:	governs to which degree the use of wild cards is permitted when doing host name validation as a part of validating a server certificate. This is done by setting one or more flags. For a description of what the flags mean, see the OpenSSL documentation. The value of the flags is as follows: 0 (no flags set) --> Wildcards are supported and they match only in the left-most label; but they may match part of that label with an explicit prefix or suffix. For example the host name "www.example.com" would match a certificate with a SAN or CN value of "*.example.com", "w*.example.com" or "*w.example.com". X509_CHECK_FLAG_ALWAYS_CHECK_SUBJECT = 1 --> Always check subject name for host match even if subject alt names present X509_CHECK_FLAG_NO_WILDCARDS = 2 --> Disable wildcard matching for dnsName fields and common name. X509_CHECK_FLAG_NO_PARTIAL_WILDCARDS = 4 --> Wildcards must not match a partial label. X509_CHECK_FLAG_MULTI_LABEL_WILDCARDS = 8 --> Allow (non-partial) wildcards to match multiple labels. X509_CHECK_FLAG_SINGLE_LABEL_SUBDOMAINS = 16 --> Constrain verifier subdomain patterns to match a single label. To set multiple flags add up their values. This setting is only effective if setting check_fqdn_against_server_cert is enabled.
Values:	0, 1, 2, 4, 8, 16 or the sum of a subset of these values
Default:	0

Setting:	hoteling
Description:	This setting enables and disables the Hoteling feature. The Hoteling feature allows a guest to login and use the host device.
Values:	on, off
Default:	off

Setting:	http_client_hash
Description:	Hash value used in reponses for a challenge if no password is given.
Values:	String
Default:	blank

Setting:	http_client_pass
Description:	HTTP Password for outgoing HTTP requests
Values:	String
Default:	blank

Setting:	http_client_save_credentials
Description:	if set to "on" http client credentials will be saved after challenge.
Values:	on, off
Default:	on

Setting:	http_client_user
Description:	The build in web client can do authenticated HTTP(S) GET requests. Therefore it uses this setting as user name and http_client_pass as password.
Values:	String
Default:	blank

Setting:	http_pass
Description:	Set up the HTTP password for your phone here.
Values:	String
Default:	blank

Setting:	http_port
Description:	Specify the HTTP port to be used by your phone through this setting. By default, it is port 80.

Values: Valid Port Number

Default: 80

Setting: http_proxy

Description: You can select the HTTP proxy address for your phone here. This is needed if you are also surfing the web via such a proxy. You can additionally define the Port Number e.g. 192.168.X.X:YYYY

Values: IP Address

Default: blank

Setting: http_proxy_hash

Description: Hash value used in reponses for a challenge if no password is given.

Values: String

Default: blank

Setting: http_proxy_pass

Description: The build in web client can use an HTTP proxy which may ask for authentication credentials. Therefore it uses http_proxy_user as user name and this setting as password.

Values: String

Default: blank

Setting: http_proxy_save_credentials

Description: if set to "on" http proxy credentials will be saved after challenge.

Values: on, off

Default: on

Setting: http_proxy_user

Description: The build in web client can use an HTTP proxy (setting http_proxy) which may ask for authentication credentials. Therefore, it uses setting http_proxy_pass as password and this setting as user name.

Values: String

Default: blank

Setting: http_scheme

Description: Define whether Basic or Digest Authentication Scheme should be used. Note: The latter is the more secure option.

Values: on, off

Default: on

Setting: http_user

Description: With this setting, you can select the HTTP username for your phone. Together with the HTTP Password option, it will protect your web interface.

Values: String

Default: blank

Setting: http_user_agent_string

Description: The contents of this setting is used for the User-Agent header in HTTP requests sent by the phone. By using substitution, the content of other (system) settings can give a hint about the hardware in provisioning requests (see DEFAULTVALUE for syntax).

Values: User-Agent Header String

Default: !!$(::)!!User-Agent: Vtech Vesa ET605 X.X.X.X $(mac_lower_case)

Setting: https_port

Description: Specify the HTTPS port to be used by your phone for HTTPS connections (default 443).

Values: HTTPS Port

Default: 443

Setting: ice_diagnostics

Description:	Here you can set the filter for ICE(Interactive Connectivity Establishment). ICE optimizes the media path. This would be the case, for example, when two phones in the same network are calling each other via a long media path through other, external networks. With ICE, the short media path in the same network would be chosen, which will presumably have better quality than the long one. Sometimes this feature will stop you from being able to make calls. When this occurs, switch it off.
Values:	0 - 9
Default:	0

Setting:	idle_cancel_key_action

Description: The navigation key labeled "Cancel" can be programmed for additional functions in idle state, please see the list below.

- none
- F_ACCEPTED_LIST (Accepted Calls)
- F_ADR_BOOK (Directory)
- F_CALL_LIST (Call Lists)
- F_CANCEL (Clear Pickup Info)
- F_CONFERENCE (Conference)
- F_CONTACTS (Contacts)
- F_DELETE_MSG (Delete Message)
- F_DENYALL (Deny All)
- F_DIALOG (Monitor Calls)
- F_DIRECTORY_SEARCH (LDAP Directory)
- F_DND (DND)
- F_FAVORITES (Favorites)
- F_HOLD (Hold)
- F_LABEL_PAGE_NEXT (Next Label Page)
- F_LABEL_PAGE_PREV (Previous Label Page)
- F_LOGOFF_ALL (Logoff Identities)
- F_MISSED_LIST (Missed Calls)
- F_MUTE(Mute)
- F_NEXT_ID (Next Outgoing ID)
- F_PREV_ID (Prev. Outgoing ID)
- F_REBOOT (Reboot)
- F_RECORD (Record)
- F_REDIAL (Redial)
- F_REDIRECT (Forward All)
- F_REGS (Change Active ID)
- F_RETRIEVE (Retrieve)
- F_RINGER_SILENT (Turn ringer off)
- F_SETTINGS (Menu)
- F_SUPPORT (Help)
- F_TRANSFER (Transfer)
- F_VKEY (Virtual Keys)
- Xml description

Values: Valid KeyEvent ID

Default:	keyevent none

Setting:	idle_down_key_action

Description: The navigation key labeled "Down" can be programmed for additional functions in idle state, please see the list below.

- none
- F_ACCEPTED_LIST (Accepted Calls)
- F_ADR_BOOK (Directory)
- F_CALL_LIST (Call Lists)
- F_CANCEL (Clear Pickup Info)
- F_CONFERENCE (Conference)
- F_CONTACTS (Contacts)
- F_DELETE_MSG (Delete Message)
- F_DENYALL (Deny All)
- F_DIALOG (Monitor Calls)
- F_DIRECTORY_SEARCH (LDAP Directory)
- F_DND (DND)
- F_FAVORITES (Favorites)
- F_HOLD (Hold)
- F_LABEL_PAGE_NEXT (Next Label Page)
- F_LABEL_PAGE_PREV (Previous Label Page)
- F_LOGOFF_ALL (Logoff Identities)
- F_MISSED_LIST (Missed Calls)
- F_MUTE(Mute)
- F_NEXT_ID (Next Outgoing ID)
- F_PREV_ID (Prev. Outgoing ID)
- F_REBOOT (Reboot)
- F_RECORD (Record)
- F_REDIAL (Redial)
- F_REDIRECT (Forward All)
- F_REGS (Change Active ID)
- F_RETRIEVE (Retrieve)
- F_RINGER_SILENT (Turn ringer off)
- F_SETTINGS (Menu)
- F_SUPPORT (Help)
- F_TRANSFER (Transfer)
- F_VKEY (Virtual Keys)
- Xml description

Values: Valid KeyEvent ID

Default: keyevent F_CALL_LIST

Setting: idle_offhook

Description: If this setting is on, the phone will go to idle state even when the handset is offhook i.e. it will not prompt the user to dial a new number.

Values: on, off

Default: off

Setting: idle_ok_key_action

Description: The navigation key labeled "Ok" can be programmed for additional functions in idle state, please see the list below.

- none
- F_ACCEPTED_LIST (Accepted Calls)
- F_ADR_BOOK (Directory)
- F_CALL_LIST (Call Lists)
- F_CANCEL (Clear Pickup Info)
- F_CONFERENCE (Conference)
- F_CONTACTS (Contacts)
- F_DELETE_MSG (Delete Message)
- F_DENYALL (Deny All)
- F_DIALOG (Monitor Calls)
- F_DIRECTORY_SEARCH (LDAP Directory)
- F_DND (DND)
- F_FAVORITES (Favorites)
- F_HOLD (Hold)
- F_LABEL_PAGE_NEXT (Next Label Page)
- F_LABEL_PAGE_PREV (Previous Label Page)
- F_LOGOFF_ALL (Logoff Identities)
- F_MISSED_LIST (Missed Calls)
- F_MUTE(Mute)
- F_NEXT_ID (Next Outgoing ID)
- F_PREV_ID (Prev. Outgoing ID)
- F_REBOOT (Reboot)
- F_RECORD (Record)
- F_REDIAL (Redial)
- F_REDIRECT (Forward All)
- F_REGS (Change Active ID)
- F_RETRIEVE (Retrieve)
- F_RINGER_SILENT (Turn ringer off)
- F_SETTINGS (Menu)
- F_SUPPORT (Help)
- F_TRANSFER (Transfer)
- F_VKEY (Virtual Keys)
- Xml description

Values: Valid KeyEvent ID

Default:	keyevent F_REDIAL

Setting:	idle_status_btn_index
Description:	Define on which context key to put the status-button. This Button overwrites the normal context-key at that position whenever there are statuses available. To not see this button, set it to -1. See also settings: status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_blocked, status_msgs_that_are_important and status_msgs_with_audio_indication
Values:	-1,1,2,3,4
Default:	4

Setting:	idle_up_key_action

Description: The navigation key labeled "Up" can be programmed for additional functions in idle state, please see the list below.

- none
- F_ACCEPTED_LIST (Accepted Calls)
- F_ADR_BOOK (Directory)
- F_CALL_LIST (Call Lists)
- F_CANCEL (Clear Pickup Info)
- F_CONFERENCE (Conference)
- F_CONTACTS (Contacts)
- F_DELETE_MSG (Delete Message)
- F_DENYALL (Deny All)
- F_DIALOG (Monitor Calls)
- F_DIRECTORY_SEARCH (LDAP Directory)
- F_DND (DND)
- F_FAVORITES (Favorites)
- F_HOLD (Hold)
- F_LABEL_PAGE_NEXT (Next Label Page)
- F_LABEL_PAGE_PREV (Previous Label Page)
- F_LOGOFF_ALL (Logoff Identities)
- F_MISSED_LIST (Missed Calls)
- F_MUTE(Mute)
- F_NEXT_ID (Next Outgoing ID)
- F_PREV_ID (Prev. Outgoing ID)
- F_REBOOT (Reboot)
- F_RECORD (Record)
- F_REDIAL (Redial)
- F_REDIRECT (Forward All)
- F_REGS (Change Active ID)
- F_RETRIEVE (Retrieve)
- F_RINGER_SILENT (Turn ringer off)
- F_SETTINGS (Menu)
- F_SUPPORT (Help)
- F_TRANSFER (Transfer)
- F_VKEY (Virtual Keys)
- Xml description

Values: Valid KeyEvent ID

Default: keyevent F_SETTINGS

Setting: ieee8021x_eap_auth_method

Description: This setting determines the IEEE802.1X EAP authentication method.

When "EAP-MD5" is selected, the settings ieee8021x_eap_md5_username and ieee8021x_eap_md5_password must be set appropriately.

When "EAP-TLS" is selected, certificates and config file must be provided (Certificates -> 802.1X Certificates).

Values: off, EAP-MD5, EAP-TLS

Default: off

Setting: ieee8021x_eap_logoff

Description: This setting enables the EAP Logoff mechanism. When enabled, the phone sends an EAPOL Logoff on behalf of an attached client, when the client got disconnected and had no chance to send an EAPOL Logoff by itself.

The phone extracts the client's MAC address from the last received EAPOL Start and EAP Response Identity packet.

Values: on, off

Default: on

Setting: ieee8021x_eap_md5_password

Description: This setting specifies the password that is used for IEEE802.1X EAP-MD5 authentication.

Values: String

Default: blank

Setting: ieee8021x_eap_md5_username

Description: This setting specifies the username that is used for IEEE802.1X EAP-MD5 authentication.

Values: String

Default:	blank

Setting:	ignore_asserted_in_gui
Description:	In certain environments the sip-servers might fill the asserted-headers in sip-dialogs with information that should not be displayed on the phone. In these cases set this setting to on. This setting is not available for all server-types. Current single exception is Microsoft-OCS, which dictates to always use the asserted headers.
Values:	on, off
Default:	off

Setting:	ignore_dhcp_findings
Description:	**This setting is obsolete.** Please use setting dhcp_options_on_ip_aquire instead. A space separated list of all those settings that are not to be overwritten by what DHCP discovers that they should be.
Values:	dns_domain, dns_server1, dns_server2, gateway, http_proxy, ip_adr, netmask, ntp_server, phone_name, sip_proxy, update_filename, update_server, vlan_id, vlan_value
Default:	blank

Setting:	ignore_missed_calls_on_busy
Description:	Inhibits the phone to add an incoming call to the missed calls if the user is in dialing state and denies an incoming call See also settings: record_missed_calls, record_missed_calls_cwi_off, sip_cancel_reasons_to_ignore_missed_call
Values:	on, off
Default:	off

Setting:	ignore_security_warning
Description:	The security warning at the upper right hand corner of the web interface as well as the initial security advice web page can be switched off by setting this setting to "on".

Values: on, off

Default: off

Setting: inactive_stream_alert_info_text

Description: When the info parameter of the Alert-Info header contains the text specified in this setting, the audio stream will be set to inactive on accepting the call. This is useful for reducing the connect time when transferring calls from a queue to an agent. For example:

Alert-Info: <http://www.notused.invalid>;info=queue

Setting this setting to "queue" would suppress the audio stream in the initial INVITE containing the above header.

Values: String

Default: blank

Setting: inactive_timeout

Description: Determines the number of minutes of inactivity after which the phone will report its state as "inactive". Activity is defined as going off-hook. A value of zero means "inactive" will never be reported.

Values: Integer

Default: 15

Setting: increased_ringer_volume

Description: In loud environments, the ringer might not be loud enough. With this setting, you can digitally increase the ringer. A side-effect might be that a ringer sounds distorted on maximal volume. Please enable this feature only if it is really necessary.

Values: on, off

Default: off

Setting: initial_rtp_keep_alives

Description:	The number of keep-alives the phone should send out at the beginning of an RTP session. A keep-alive is an empty STUN Binding Request and serves to open a pin hole in the firewall. The phone sends one keep-alive by default, i.e. when the setting is empty. This is for backward compatibility. Set this to zero if you want no keep-alives. Note that if the phone receives such a Binding Request, it will answer it with a Binding Response.
Values:	0 - 256, blank
Default:	blank

Setting:	intercom_connect_type
Description:	If the Alert-Info header is taken into account in order to allow auto answering behaviour like intercom, this option can be used to specify whether the phone answers in handset, headset, or handsfree Mode. See also setting auto_connect_type.
Values:	intercom_connect_type_ handsfree, intercom_connect_type_headset, intercom_connect_type_handset
Default:	intercom_connect_type_handsfree

Setting:	internal_ringer_file
Description:	Melody to be played back on the Internal Ringer Text.
Values:	Ringer1, Ringer2, Ringer3, Ringer4, Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, silent
Default:	Ringer1

Setting:	internal_ringer_text
Description:	Text which can be specified in Alert-Info to categorize a specific ringtone melody.
Values:	String
Default:	blank

Setting:	ip_adr
Description:	You can change the IP address of the device through this setting. This parameter is mandatory in order to enable the Ethernet connection.

Values: IP address

Default: blank

Setting: ip_adr_v6

Description: This settings holds the current IPv6 address of the device.

Note: Setting up static IPv6 addresses is currently not supported. See also setting dhcp_v6.

Values: IPv6 Address

Default: blank

Setting: ip_adr_vlan

Description: SYSTEM INTERNAL (Reboot required).

This setting defines the VLAN IP address of the phone.

Values: IP address

Default: blank

Setting: ip_call_identity

Description: Number of the identity who supports ip calls.

Values: 1,2, blank

Default: blank

Setting: ip_frag_enable

Description: If this setting is on, the IP fragmentation bit in IP packets will be set, allowing network devices to fragment the IP packet.

Values: on, off

Default: on

Setting: ipv4_conflict_detection

Description: Configures the IPv4 conflict detection module according to RFC 5227. Normally there is no need the change the default behaviour.

- detect_defend: the phone detect possible conflicts before using the selected IPv4

- address and after using it defends the address via arp announcements.

- detect_only: the phone detect possible conflicts before using the selected IPv4 address only

- defend_only: the phone defends the address via arp announcements only

off: the IPv4 conflict detection module is disabled.

Changes to this setting will only affect after a reboot of the phone.

Values: off, detect_only, defend_only, detect_defend

Default: detect_defend

Setting: is_voice_identity

Description: When this is disabled, invites for audio-calls will not be accepted by this identity. A non-voice-identity will automatically force setting hide_identity to be enabled.

Values: on, off

Default: on

Setting: keepalive_interval

Description: Specifies the number of seconds after which a new keepalive message will be sent out to the Registrar/Proxy port in order to have the port stay open and the phone remain reachable.

Values: Integer

Default: blank

Setting: key_0_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 0

Setting: key_1_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 1

Setting: key_2_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 2

Setting: key_3_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 3

Setting: key_4_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 4

Setting: key_5_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 5

Setting: key_6_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 6

Setting: key_7_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 7

Setting: key_8_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 8

Setting: key_9_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: 9

Setting: key_cancel_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: CANCEL

Setting: key_directory_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: PHONE_BOOK

Setting: key_dnd_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: DND

Setting: key_down_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: DOWN

Setting: key_enter_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: ENTER

Setting: key_f1_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: F1

Setting: key_f2_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: F2

Setting: key_f3_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: F3

Setting: key_f4_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: F4

Setting: key_f5_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: F5

Setting: key_hash_remapped

Description:

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: #

Setting: key_headset_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: HEADSET

Setting: key_left_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: LEFT

Setting: key_mute_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: MUTE

Setting: key_retrieve_remapped

Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	RETRIEVE

Setting:	key_right_remapped
Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	RIGHT

Setting:	key_settings_remapped
Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	SETTINGS

Setting:	key_speaker_remapped

Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	SPEAKER

Setting:	key_star_remapped
Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	*

Setting:	key_transfer_remapped
Description:	The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key. Remapping is done at the lowest level, that is also the reason why these settings require reboot.
Values:	f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #
Default:	TRANSFER

Setting:	key_up_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: UP

Setting: key_vol_down_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: VOLUME_DOWN

Setting: key_vol_up_remapped

Description: The key_.._remapped settings are used for OEM customers that decide to have a different key-layout then the VTech standard, e.g.: switching position of OK and Cancel key.

Remapping is done at the lowest level, that is also the reason why these settings require reboot.

Values: f1, f2, f3, f4, f5, cancel, enter, left, right, up, down, record, retrieve, mute, speaker, headset, redial, dnd, directory, help, menu, pgm, transfer, settings, conference, hold, volume up, volume down, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, *, #

Default: VOLUME_UP

Setting: keyboard_event_time_limit

Description:	Key press events within this time limit in milliseconds will be ignored.
Values:	Integer
Default:	80

Setting:	keyboard_lock
Description:	By setting this option to 'on' the phone's keyboard will be locked. On the phone the keyboard can be locked/unlocked by pressing the star key for a few seconds (if enable_keyboard_lock is 'on'). This setting represents the current lock state of the phone. Therefore changing it can be used to lock or unlock the phone from the web interface regardless of whether the enable_keyboard_lock is on or off.
Values:	on, off
Default:	off

Setting:	keyboard_lock_accepted_keys
Description:	Comma-separated list of keys which will be accepted if phone keyboard is locked.
Values:	Comma separated list of key names
Default:	STATE_AUTO_LEAVE,F_HOLD,MUTE,VOLUME_UP,VOLUME_DOWN

Setting:	keyboard_lock_emergency
Description:	The specified space separated numbers can be dialled via keyboard even if the keyboard lock is enabled. Just dial them as usual without unlocking the keyboard before.
Values:	Strings separated by spaces
Default:	911 112 110 999

Setting:	keyboard_lock_pw
Description:	The locked keyboard can be unlocked only by typing in the specified PIN. If this is empty, no PIN is needed to unlock the keyboard.
Values:	Numerical String
Default:	blank

Setting: keyboard_lock_timeout

Description: This setting allows you to configure an inactivity timer (in seconds). If enable_keyboard_lock is set to on, the phone will automatically lock the keypad after the configured inactivity time. The user would then need to enter the configured PIN in order to unlock the keypad. On OCS servers this setting is provisioned via inband provisioning parameter ucPhoneTimeOut.

Values: integer, blank

Default: blank

Setting: language

Description: This is the language used on the Phone User Interface of your phone. Choose a language from the drop-down menu.

Values: Language, blank

Default: blank

Setting: lastexit

Description: **SYSTEM INTERNAL**

This is a variable set by the phone and it displays the last exit code of lcs. Shown on support.htm

Values: String

Default: 0

Setting: lastkey

Description: **SYSTEM INTERNAL**

This is a variable set by the phone and it displays the last pressed key. Shown on support.htm

Values: String

Default: 0

Setting: lastmethod

Description:	**SYSTEM INTERNAL**
	This is a variable set by the phone and it displays the last state method. Shown on support.htm'
Values:	String
Default:	0

Setting:	lastsignal
Description:	**SYSTEM INTERNAL**
	This is a variable set by the phone and it displays the the last signal that kills the lid. Shown on support.htm
Values:	String
Default:	0

Setting:	laststate
Description:	**SYSTEM INTERNAL**
	This is a variable set by the phone and it displays the last lcs state. Shown on support.htm
Values:	String
Default:	0

Setting:	lcs_core_dump
Description:	When this setting is on a core dump is written on flash in case the phone LCS crashes.
Values:	on, off
Default:	off

Setting:	lcserver1
Description:	Type in the IP address of the remote LCServer if you want your phone to connect to it. Usually, you do not need to make an entry here.
Values:	String
Default:	blank

Setting:	ldap_answer_timeout
Description:	Define how many milliseconds the phone should wait on answers from the ldap server before cancelling the request.
Values:	10-3600000
Default:	7000

Setting:	ldap_base
Description:	This setting specifies the LDAP search base (the distinguished name of the search base object) which corresponds to the location in the directory from which the LDAP search is requested to begin. The search base narrows the search scope and decreases directory lookup time. If you have multiple organizational units in your directory (for example, OU=Sales in O=COMPANY and OU=Development in O=COMPANY), but the "OU=Sales" organization never uses AOL AIM, you can restrict the lookup to the OU=Development subtree only by entering providing the following search base: OU=Development, O=COMPANY. Other examples see below.
Values:	String
Default:	blank

Setting:	ldap_display_name
Description:	This setting specifies the format in which the name of each returned search result is to be displayed on the VTech phone. The setting allows combinations of various name attributes along with special characters.
Values:	LDAP name attributes
Default:	blank

Setting:	ldap_max_hits
Description:	This setting specifies the maximum number of search results to be returned by the LDAP server. Please note that a very large value of the Max. Hits will slow down the LDAP lookup, therefore the setting should be configured according to the available bandwidth. The default value for this setting is 50.
Values:	1 - 200

Default:	50

Setting:	ldap_name_attributes
Description:	This setting can be used to specify the name attributes of each record which are to be returned in the LDAP search results. This setting compresses the search results, as the server only returns the attributes which are requested by the VTech phone. The setting allows the user to configure multiple space separated name attributes. Please consult your system administrator regarding which name attributes are to be configured.
Values:	space separated LDAP name attributes
Default:	blank

Setting:	ldap_number_attributes
Description:	This setting can be used to specify the number attributes of each record which are to be returned in the LDAP search results by the LDAP server. This setting compresses the search results, as the server only returns the attributes which are requested. The user can configure multiple space separated number attributes by using this setting. Please consult your system administrator regarding which number attributes are to be configured.
Values:	space separated number attributes
Default:	blank

Setting:	ldap_number_filter
Description:	LDAP number filter is the search criteria for number look ups. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). The number prefix for search entered by the user is represented by the % symbol in the filter.
Values:	LDAP Filters
Default:	blank

Setting:	ldap_over_tls
Description:	Specifies wHether to use tcp (off) or tls (on) as LDAP transport.
Values:	on, off

Default: off

Setting: ldap_password

Description: This setting specifies the bind Password for LDAP servers. VTech phones use simple authentication scheme for bind requests. This setting can be left blank in case the server allows anonymous binds. Otherwise you will need to provide the Password along with the Username in order to access the LDAP server.

Values: String

Default: blank

Setting: ldap_port

Description: This setting specifies the LDAP server port. In case the setting is not configured, the default LDAP port (389) is taken.

Values: 0 - 65535

Default: blank

Setting: ldap_predict_text

Description: Allows to quickly lookup names in the LDAP directory by using a technique similar to the one known as T9.

In order to search John for example, you would press 5 6 4 6 consecutively.

Note: With this option enabled you cannot toggle between letters by pressing the same key several times.

Values: on, off

Default: off

Setting: ldap_queue_requests

Description: As of introduction of this setting the phone is capable of sending multiple ldap-queries in parallel over the network. Setting this setting to false enables this behaviour which might result in a speedier experience.

Values: true, false

Default: true

Setting:	ldap_search_filter
Description:	LDAP name filter is the search criteria for name look ups. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). The name prefix for search entered by the user is represented by the % symbol in the filter.
Values:	LDAP filters
Default:	blank

Setting:	ldap_server
Description:	This setting refers to the DNS name or IP address of the LDAP server.
Values:	IP Address or domain
Default:	blank

Setting:	ldap_sort_results
Description:	This setting can be used to sort the LDAP result set.
Values:	on, off
Default:	off

Setting:	ldap_telephonenumber_mapping
Description:	Set the number type used for ldap telephoneNumber entries. When the value of the setting is not one of the valid values the number type of ldap telephoneNumber entries will be set to unqualified.
Values:	office, home, mobile, unqualified
Default:	office

Setting:	ldap_username
Description:	This setting specifies the bind Username for LDAP servers. Most LDAP servers allow anonymous binds in which case the setting can be left blank. However if the LDAP server does not allow anonymous binds, you will need to provide the Username and Password allowed to query the LDAP server.

Values: String

Default: blank

Setting: led_blink_fast

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to start blinking fast.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will blink very fast when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: RINGING PICKUP PhoneHasCallInStateRinging alerting_local alerting_remote

---

Setting: led_blink_medium

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to start blinking at a medium speed.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will blink when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: RECORDING MESSAGE DateOngoing DateReminding

Setting: led_blink_slow

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to start blinking slowly.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will blink slowly when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: PARKED HOLDING I-Am-Almost-Ready I-Am-Busy PhoneHasCallInStateHolding held_local held_remote

---

Setting: led_blue

Description: The only blue LED in VTech phones is the call-indication-LED of the MeetingPoint. The setting is used in conjuction with the led_call_indicator_usage setting to determine its color.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: Blank

Setting: led_call_indicator_usage

Description: This setting defines what events/states the call-indicator-LED should signal.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: PhoneHasCallInStateRinging PhoneHasCallInStateCalling PhoneHasCallInStateRingback PhoneHasCallInStateConnected PhoneHasCallInStateOffhook PhoneHasCallInStateHolding PhoneHasCall PhoneHasMissedCalls CurrentIdentityHasVoiceMessages PhoneHasVoiceMessages DateOngoing DateReminding

---

Setting: led_colors_used_for_green_only_leds

Description: The ET605 Deskset has green line-LEDs, however you can also attach a ET6 expansion module which has multi colored LEDs. The default LED settings are configured in such a manner that they setup the standard multicolor behavior. This setting controls which of the multi color behaviors are represented on the green-only line-LEDs of the phone itself.

For example, if some sort of key-setup would signal its state by blinking orange on a multi colored LED you can hereby determine whether or not the same key-setup would cause a green-only line-LED to blink or just stay off. I.e. if you include orange into this setting's value the green LED would blink, if orange is not listed it will stay off.

Values: a space separated list of colors "green" "red" and "orange"

Default: green

Setting: led_green

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to become green.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will become green when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: AVAILABLE I-Am-Ready I-Am-Almost-Ready seized_local alerting_local active_local held_local

---

Setting: led_message_usage

Description: This setting defines what events/states the message-LED should signal.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: CurrentIdentityHasVoiceMessages PhoneHasVoiceMessages

---

Setting: led_on

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to turn on.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will turn on when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: ON IN_A_CALL CALLING IN_A_MEETING URGENT_INTERRUPTIONS_ONLY BUSY I-Am-Busy DND_ALL DND_SELF ACTIVE INACTIVE BE_RIGHT_BACK SEIZED CONNECTED ON_HOLD OFFHOOK RINGBACK I-Am-Ready PhoneHasCall PhoneHasMissedCalls CurrentIdentityHasVoiceMessages PhoneHasVoiceMessages seized_local seized_remote active_local active_remote

---

Setting: led_orange

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to change its color into orange.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will change its color into orange when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default: AWAY INACTIVE BE_RIGHT_BACK

Setting: led_red

Description: This setting is used in conjuction with the presence feature. Whenever the state of the monitored extension changes, the assigned Led will change its state. This setting defines the state in which the monitored extension must be in order for the Led to change its color into red.

For example, if the content of this setting is 'AWAY OFFLINE', the Led will change its color into red when the monitored extension's state becomes away or offline.

Values: AVAILABLE, BUSY, IN_A_CALL, IN_A_MEETING, HOLDING, DND, INACTIVE, AWAY, OFFLINE, UNKNOWN, UNAVAILABLE, ACTIVE, RINGING, PhoneHasCall, PhoneHasCallInState..., alerting_local, alerting_remote, DateOngoing, DateReminding, I-Am-Almost-Ready, I-Am-Busy, held_local, held_remote, PhoneHasVoiceMessages, CurrentIdentityHasVoiceMessages, seized_local, seized_remote, active_local, active_remote

Default:	BUSY IN_A_CALL IN_A_MEETING URGENT_INTERRUPTIONS_ONLY DND I-Am-Busy UNAVAILABLE seized_remote alerting_remote active_remote held_remote

Setting:	leftnav_hidden_admin
Description:	Any menu entry in the navigation sidebar of the web interface can be hidden with this setting. This setting is for the admin mode, the according setting for user mode is leftnav_hidden_user.
Values:	operation,home,addressbook,setup,preferences,speeddial,functionkeys,ocs_account,lineone,linetwo,linethree,linefour,linefive,linesix,lineseven,lineeight,linenine,lineten,lineeleven,linetwelve,action,advanced,trusted_cert,softupdate,status,sysinfo,log,siptrace,dnscache,subscriptions,pcaptrace,memory,settings
Default:	blank

Setting:	lid_core_dump
Description:	When this setting is on a core dump is written on flash in case the phone LID crashes.
Values:	on, off
Default:	off

Setting:	line_info_at_auto_redial
Description:	Shows the line key info when the phone is displaying the auto-redial screen, where it states it's going to attempt to redial in XX seconds. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_buddies

Description:	Shows the line key info in when browsing through your buddy-contacts. Thats the state the phone is after you have pressed the "Contacts" softkey. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_call_completion
Description:	Shows the line key info while you are waiting for call completion. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_calling
Description:	Shows the line key info when you are calling someone and the phone shows the calling screen. This setting cannot apply, when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_conference

Description:	Shows the line key info when you are in a conference, that is after you've established that conference with your phone. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_connected
Description:	Shows the line key info when you are in a call and talking (or listening, or doing whatever). This setting cannot apply, when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_directory
Description:	Shows the line key info in when looking at your telephone nmbr. directory. Thats the state the phone is after you have pressed the "Directory" key. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_edit_number

Description: Shows the line key info in when entering a phone number you want to dial.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_holding

Description: Shows the line key info when the phone shows the on-hold screen. That usually happens when you put someone on hold.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_idle

Description: Shows the line key info in idle. Idle is the state the phone is usually in, when nothing is happening.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_mailbox_info

Description:	Shows the line key info when the phone is displaying the mailbox information. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_minibrowser
Description:	Shows the line key info in when a minibrowser document is shown. You can decide the behavior within each minibrowser document by setting the attribute show_line_info_layer of the main-tag to on, off or smart. If a document doesn't have this attribute, the value defined by this setting is used. This setting cannot apply when all 4 line-keys are set to “line” or "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting:	line_info_at_presence
Description:	Shows the line key info in when selecting your current presence state. That is the state the phone is after you have pressed the "Presence State" softkey. This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown. For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.
Values:	on, off, smart
Default:	off

Setting: line_info_at_registration

Description: Shows the line key info in when changing the active identity. Thats the state the phone is in when you have pressed the "Change active Id" softkey and are choosing the active identity.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_ringing

Description: Shows the line key info when a call is incoming and the phone shows the ringing screen.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_settings

Description: Shows the line key info in when in settings menu. That is the state the phone is in when you are scrolling thru the settings menu and look at or edit its settings.

This setting cannot apply when all 4 line-keys are set to "none". When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_terminated

Description: Shows the line key info when a call gets terminated.

This setting cannot apply when all 4 line-keys are set to "none".
When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_at_transfer

Description: Shows the line key info when you are transferring a call.

This setting cannot apply when all 4 line-keys are set to "none".
When all are "none", the line info layer will never be shown.

For information about how to temporarily overwrite this setting, please see line_info_overwrite_time.

Values: on, off, smart

Default: off

Setting: line_info_overwrite_time

Description: When the line info layer is shown, one can hide it temporarily through long-press of line-key P4. When it is hidden, long-press P4 will make it appear temporarily. How long this change will stay on screen is determined by this setting. Its value is measured in milliseconds.

Values: A positive integer

Default: 5000

Setting: lldp_asset_id

Description: LLDP asset ID

Values:

Default: VTechET605

Setting: lldp_reboot_timeout

Description:	This setting defines the amount of time in seconds that a reboot should be deferred after a new network policy has been published via LLDP. This helps to avoid continuous reboot loops in network environments where new network devices are first put into a retention VLAN and after successful authentication gain access to their designated production VLAN (e.g. voice VLAN). Note: The default value of 60 seconds seems to be a reasonable value to grant enough time for the authentication process to complete, or a fallback mechanism (e.g. MAC Authentication Bypass (MAB)) to take place.
Values:	Integer
Default:	60

Setting:	location_template

Description: This setting defines the template needed for displaying the location information automatically retrieved on phones registered with a Lync server. To display the location information press the menu button on the phone and select Information > Location.

This information is returned from the Location Information Server as a PIDF document with the location information included in the 'civic address' extension of the PIDF document. For details about this extension see RFC 5139.

The location information is essentially an address. Because the 'civic address' format contains a very high level of detail, particularly the elements describing a street address, the usage of the various elements will vary widely form country to country as well as the order in which these elements are typically presented to the user. This template is therefore used to select the required elements from the 'civic address' element inside the PIDF document and embed them in some explanatory text.

To create a template simply combine regular text, 'civic address' elements and line breaks. 'Civic address' elements are identified by surrounding the element name from the civicAddress structure with curly braces ('{' and '}'); a line break is represented by '\n'.

For example, the template

City: {A3}\nPostal Code: {PC}

might result in the following output:

City: Berlin

Postal Code: 10117

For a list of all available civic address elements see RFC 5139 (and RFC 4119 which it extends). Note that not all civic address elements are necessarily populated by the Location Information Server.

To include a curly brace or a backslash (\) in the regular text it must be preceded by the escape character '\'.

This template extracts 'civicAddress' elements only. Any elements from higher level PIDF structures within this template are ignored.

Values: Strings separated by spaces

Default: {NAM}\n{LOC}\n{HNO}{HNS} {PRD} {RD} {STS} {POD}\n{A3}, {A1} {PC}\n{country}

Setting: log_level

Description: SYSTEM INTERNAL

Log level of the maintenance web page, 9 is the most verbose mode.

Values: -1 (off) to 9

Default: 5

Setting: logoff_all_no_confirm

Description: Disable/Enable the display confirmation query after Logoff_All event

Values: on, off

Default: off

Setting: logon_wizard

Description: The Logon Wizard assists you during the SIP line registration process. Turn this setting on if you want to use the Logon wizard, switch it off if you don't. <skip welcome> enables the wizard, but starts directly with editing the account.

Values: on, off, skip welcome

Default: on

Setting: long_cancel_is_blocking_caller

Description: With long press cancel, you can cancel an incoming call. If this setting is set to 'on,' the caller will be additionally added to the deny list of the internal phone book. In this case, the caller is blocked forever. If he/she tries to call you again, he/she will always get an busy. To remove the caller from the deny list, the phone book entry can be removed completely or the contact type of the entry has to be changed from 'Deny List' to an other value. If the setting is set to 'off,' the caller will not be added to the deny list.

Values: on, off

Default: on

Setting: mac_info_in_sip_register

Description: If set to on, a new sip header Mac is added to the register, and also added to the user-agent.

Values: on, off

Default: off

Setting: mailbox_active

Description: If this setting is on, the Retrieve button will dial the mailbox of the active line. Otherwise the mailbox associated with the first MWI message in the queue is used.

This setting also changes which type of status-msg is used for signaling messages on PBX.

When set to on, the statuses CurrentIdentityHasTextMessages and CurrentIdentityHasVoiceMessages are used.

When set to off, the statuses PhoneHasTextMessages and PhoneHasVoiceMessages are used. i.e. changing this setting will automatically change the status-msg controlling settings: status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_blocked and status_msgs_that_are_important

Values: on, off

Default: on

Setting: max_boot_delay

Description: On reboot, the phone waits for a random number of seconds not exceeding the value set in this field, and then continues to boot up. This is to prevent DOS by provisioning servers etc. by preventing all the phones (that are rebooting) to send requests simultaneously in a given setup.

Values: Integer

Default: 0

Setting: max_dialed_calls

Description: Defines how many dialed calls the phone keeps track of (size of redial-list).

There are also settings for received, missed and parked calls - see settings: max_received_calls, max_missed_calls, and max_parked_calls.

Values: Integer >=0

Default: 30

Setting:	max_forwards
Description:	If you set a maximum number of forwards in this field, each time a forward is sent the counter is reduced by one. When zero is reached, the forwarding will stop. This prevents the phone from running into a SIP message-forwarding loop.
Values:	Integer
Default:	70

Setting:	max_missed_calls
Description:	Defines how many missed calls the phone keeps track of. There are also settings for received, dialed and parked calls - see settings: max_received_calls, max_dialed_calls, and max_parked_calls.
Values:	Integer >=0
Default:	100

Setting:	max_parked_calls
Description:	Defines how many parked calls the phone keeps track of. There are also settings for received, dialed and missed calls - see settings: max_received_calls, max_dialed_calls, and max_missed_calls.
Values:	Integer >=0
Default:	30

Setting:	max_pin_retry
Description:	Determines how many times the user may enter a wrong PIN before the keyboard is locked permanently. A value of zero indicates that there is no limit. Once the keyboard has been permanently locked, the user is prompted to reset the PIN when an attempt is made to unlock the keyboard. To reset the PIN the user must first enter the user password of the active identity. Then the user is prompted to create a new PIN. If the user cancels the PIN reset action, the keyboard remains locked.
Values:	Integer, or blank
Default:	blank

Setting: max_received_calls

Description: Defines how many received calls the phone keeps track of.

There are also settings for missed, dialed and parked calls - see settings: max_missed_calls, max_dialed_calls, and max_parked_calls.

Values: Integer >=0

Default: 30

Setting: mb_trusted_hosts

Description: Some features of the Minibrowser - like changing settings, for instance - are security relevant, and can not be used in XMLs from arbitrary sources. The XML must come from a trusted source to be allowed to use these features. By default only XMLs stored on the phone are trusted. With this setting you can extend that list of trusted sources with a list of hostnames or IP addresses. Caution: the hostname or IP address must appear exactly like the host in the URLs of the trusted XMLs, that is no resolution from hostname to IP address or vice versa is done.

Values: Space separated list of hostnames and/or IP addresses.

Default: blank

Setting: mc_address

Description: The phone receives RTP packets destined for this multicast IP address and port and plays them out.

You can setup the multicast address with additional options:

speaker=(0|1):

If this option is set and value is 1, then the multicast audio will be played always over speaker. If value is 0, then the current audio device will be used. If this option is not set, then value 0 is used as default value.

interrupt=(0|1):

If this option is set and value is 1, then the multicast audio interrupts a running call. If multicast is finished, then the interrupted call continues. If value is 0, the multicast audio will only be played in idle state. If this option is not set, then value 0 is used as default value.

volmax=(0|1):

If this option is set and value is 1, then the maximal volume will be used for multicast audio. If value is 0, then the current volume will be used. If this option is not set, then value 0 is used as default value.

priority=(0..10):

This option sets the priority of the multicast address. You can choose a priority between 0 and 10, where 0 is the lowest and 10 the highest priority. If the phone receives multicast from more than one configured port, then the multicast with the highest priority will be played. If they have the same priority then the multicast will be played, that was received first. If this option is not set, then a priority of 5 is used as default.

Values: Valid multicast IP and port or a comma separated key-value string with IP and port and optional parameters

Default: blank

Setting: min_pin_length

Description: Determines the minimum length that a PIN must have. A value of 0 indicates that a PIN is not required. If the length of the currently configured PIN is less than the value of this setting, the user will be prompted to create a new PIN which meets this requirement at the first attempt to manually lock or unlock the keyboard. On OCS servers this setting is provisioned via inband provisioning parameter ucMinPinLength, but only if its value is greater than the setting's current value.

Values: Integer, or blank

Default: blank

Setting: monitor_notify_for_subscription_refresh

Description: If we subscribe, we must get a NOTIFY indicating the current state of the dialog. But sometimes it might happen that the NOTIFY gets lost.

For handling this error state, we introduced a new timer which monitors the receiving of the NOTIFY. If we don't get the NOTIFY, we un-subscribe the current subscription and set up a new fresh subscription to get the current state and resolve the error condition. Normally this setting should remain off. If you experience that the BLF gets frequently out of sync (staying on to long), or otherwise have the condition described above, you could give this setting a try.

Values: on, off

Default: off

Setting: ms_before_returning_to_idle_xml

Description: Only needed if an xml-idle-screen is configured to access the springboard.

Allows to show standard-idle screen for the defined number of milliseconds whenever user presses cancel or touches the screen.

Values: Integer >=0

Default: 10000

Setting: msw_cp_pat

Description: This is the encrypted persistent authentication token. It must be generated by the CommPortal server and must only be set through provisioning.
If available, it will be used (instead of directory number and password) by the first identity to login to Metaswitch CommPortal.

Values: Encrypted token string provisioned by Metaswitch CommPortal server.

Default: blank

Setting: msw_directory_number

Description: The Metaswitch Directory number.

Values: Integer, or blank

Default: blank

Setting:	msw_password
Description:	The Metaswitch password.
Values:	String
Default:	blank

Setting:	msw_web_url
Description:	Specifies the Metaswitch Server.
Values:	URL
Default:	blank

Setting:	multicast_listen
Description:	If enabled, the phone receives RTP G.711 u-law (20 ms) packets sent to the given multicast addresses and plays them out. It can be used for listening, in handsfree mode, for streaming audio broadcasts or public announcements etc.
Values:	on, off
Default:	off

Setting:	mute_is_dnd_in_idle
Description:	In idle state the mute button acts as DND button.
Values:	on, off
Default:	off

Setting:	mwi_dialtone
Description:	Your phone's dialtone can be changed into a stuttering tone <stutter> to remind you on having waiting mailbox messages. With <normal> this functionality can be switched off.
Values:	normal, stutter
Default:	stutter

Setting:	naptr_sip_uri
Description:	When this feature is set to on, the phone converts SIP uri's according to the regular expression dialplan of the active outgoing line for numbers dialed through Received and Missed call lists. For normal phone operation it is best to leave it turned off, as a valid SIP uri need not be converted again. Only valid if the pbx used can not append the requisite leading digits to reach remote destination or if the number does not already contain the extra digits needed. e.g. adding 00 for an international call or 0 to access a number outside the local network.
Values:	on, off
Default:	off

Setting:	navikey_event_time_limit
Description:	Navikey press events in different directions within this time limit in milliseconds will be ignored. Subsequent press events in the same direction (e.g. when scrolling down a list in the PUI) are not affected by this setting.
Values:	Integer
Default:	300

Setting:	netmask
Description:	Change the netmask for the device.
Values:	IP Address, or blank
Default:	255.255.0.0

Setting:	netmask_vlan
Description:	SYSTEM INTERNAL (Reboot required). This setting defines the netmask for the device.
Values:	IP Address, or blank
Default:	blank

Setting:	network_id_port

Description:	Set a static local port number, which is used to listen for SIP protocol communications. Please note that setting the value to 5060 also enables direct IP calls to the IP identity (see also setting sip_ip_dialin_content_types).
Values:	Valid port number
Default:	blank

Setting:	no_dnd
Description:	If you don't want the users of the phone to have the option to turn on the Do not disturb (DND) mode, set Block DND to on. This may be desirable in call center or switchboard environments.
Values:	on, off
Default:	off

Setting:	ntp_refresh_timer
Description:	Specify the time in seconds after which the phone again contacts the NTP server to refresh the time.
Values:	60-32400
Default:	3600

Setting:	ntp_server
Description:	Specify the domain name / IP address of the NTP server here.
Values:	IP Address, or blank
Default:	192.53.103.104

Setting:	ntp_server_v6
Description:	Addtional NTP server for IPv6. Used only if setting ntp_server is empty.
Values:	IPv6 Address or FQDN or blank
Default:	blank

Setting: number_sign_encoding

Description: RFC 3261 states that the number sign (#) must be encoded inside a telephone subscriber. Therefore the default value of the setting is 'on'. Change it to 'off' if you need special cases for direct dialing and therefore not encoding the #.

Values: on, off

Default: on

Setting: number_simultaneous_calls

Description: Overrides the default maximum of simultaneous calls.

Values: Integer or off

Default: off

Setting: ocip_max_hits

Description: This setting specifies the maximum number of search results to be returned by the OCI-P server. Please note that a very large value of the Max. Hits will slow down the OCI-P lookup, therefore the setting should be configured according to the available bandwidth. The default value for this setting is 50.

Values: Integer

Default: 50

Setting: ocip_password

Description: This setting specifies the OCI-P server password.

Values: String

Default: blank

Setting: ocip_port

Description: This setting specifies the OCI-P server port.

Values: integer or blank

Default: 2208

Setting: ocip_server

Description: This setting refers to the DNS name or IP address of the OCI-P server.

Values: IP Address, hostname, blank

Default: blank

Setting: ocip_username

Description: This setting specifies the OCI-P username.

Values: String

Default: blank

Setting: offer_gruu

Description: This setting is used to toggle the support for GRUU (Globally Routable User agent URLs) in SIP. When several phones have the same account, each one of them can be identified by the proxy through this GRUU ID, which is unique for each phone and stays the same even after reboot.

Values: on, off

Default: on

Setting: offer_mpo

Description: Using this setting, the user can turn the Media Path Optimization support on or off. Turning it on makes sense only when you have MPO-supporting session border controller devices in your environment (e.g., Jasomi).

Values: on, off

Default: off

Setting: offer_outbound

Description: This setting is used to toggle the support for draft-ietf-sip-outbound-20. Enable this to force the reusage of connections, what VTech phones already do. However, in combination with setting offer_gruu, the phone will stick to the network flow created during line registration. Additionally you have to specify a value for setting keepalive_interval.

Values: on, off

Default: off

Setting: offhook_accept_calls

Description: If set to 'on' going offhook accepts an incoming call.

Values: on, off

Default: on

Setting: offhook_dial_prompt

Description: If this setting is on, the phone will offer a dial prompt when the handset goes offhook. Otherwise, the phone stays in idle state.

Values: on, off

Default: on

Setting: onhook_debounce_timeout

Description: Delay in milliseconds for debouncing of the mechanical hook switch. On phones with electronic hook switch, this setting should be zero.

Values: Integer >=0

Default: 150

Setting: outgoing_identity

Description: Contains the number of the outgoing identity. This value is retrieved automatically from the active_line configuration.

Values: 1-

Default: 1

Setting: overlap_dialing

Description:	If the connected SIP proxy supports this function, it can be enabled here. This will lead to the phone starting to dial each time a digit is entered and the SIP proxy replying with Number incomplete until such time as the number has been entered and the call can be initiated successfully without the enter key having to be pressed.
Values:	on, off
Default:	off

Setting:	pair_tcp_relay_only
Description:	When enabled, this setting causes only local TCP relay ICE candidates to be paired with remote TCP relay candidates, and thus prevents local TCP host candidates from being paired with remote TCP relay candidates.
Values:	on, off, true, false
Default:	off

Setting:	partial_lookup
Description:	When this option is set to 'on', the phone tries to match parts of the incoming call number to numbers stored in the address book displays the name belonging to the first number that matches partially. An integer value can be set too. If the value of the setting is n and n > 0, the phone sends a query to the LDAP server or to the internal address book. It matches with entries that end with that postfix of lenght n.
Values:	on, off, <unsigned integer>
Default:	off

Setting:	pbx_buttons
Description:	This setting allows for sending a message containing a button name to your PBX whenever the handset is placed on hook. For this to work, you'll need to set up one of your line keys (for example P1) as type button, with the number-field set to "message". The PBX will have to set up the number where the message should be sent to.
Values:	on, off
Default:	off

Setting:	peer_to_peer_cc
Description:	Disable it if call completion is handled by the SIP proxy. Otherwise the phones are handling it directly between each other.
Values:	on, off
Default:	on

Setting:	perform_initial_query_in_ldap_state
Description:	When entering the LDAP directory you can decide whether or not to query the server for an initial list of entries (query string = *).
Values:	on, off
Default:	on

Setting:	phone_name
Description:	Change the hostname of the phone here. If set, the hostname is used to sign syslog packages and as the title of the webinterface webpages.
Values:	String
Default:	blank

Setting:	phone_type
Description:	**SYSTEM INTERNAL** This setting shows the type of phone.
Values:	String
Default:	VTechET605

Setting:	play_music_during_hold
Description:	Enable this setting if you want to stream music from your local phone to the callers on hold. The music is stored on your phone and can be exchanged via provisioning.
Values:	on, off
Default:	off

Setting: pnp_config

Description: If turned to on, the phone will try to retrieve its settings via a Plug-and-Play (PnP) Server. Modern SIP PBXs/Proxys can provide the PnP configuration data for the VTech phones. Please refer to the manual of your PBX/Proxy. If the PnP configuration fails, the phone will try to get the settings from a setting server.

Values: on, off

Default: on

Setting: pnp_server

Description: SYSTEM INTERNAL

If a potential setting server URL has been delivered via SIP PnP, it will be stored in this setting.

Values: URL

Default: blank

Setting: prefer_saved_over_received_photo

Description: This setting is used to decide which photo to show, when you have a photo in your local address book and the server sends another one over in the SIP INVITE package.

Values: on, off

Default: on

Setting: preselection_nr

Description: Specify the number to be prefixed to each dialled number.

Values: Dialing String

Default: blank

Setting: presence_lookup_number

Description: When this setting is set to 'on' the phone will use presence information to look up contacts from the server.

Values: on, off

Default: off

Setting: presence_timeout

Description: The time in min after which, if there is no activity, presence is set to closed.

Values: Integer

Default: 15

Setting: prioritise_asserted

Description: SIP messages like INVITE may include asserted information (p-asserted-identity). If this setting is enabled, the phone displays the name provided by the asserted information with the highest priority. Only if no asserted information is given the priority defined by the related setting contact_source_priority will be considered.

Values: on, off

Default: on

Setting: privacy_in

Description: Reject or accept anonymous incoming calls.

Values: on, off

Default: off

Setting: privacy_out

Description: Show or hide your own phone number on outgoing call.

Values: on, off

Default: off

Setting: prov_back_off_timer

Description: With this setting a repetition mechanism ('back off timer') of HTTP/HTTPS based provisioning requests can be realized, which is using a list of random based growing timeouts. A time value list can be initialized by different formats. Time values are expressed in seconds.

Values:

- ‘120' the number will be stored to the list as only entry.
- '3,6:300' a random number between 3 and 6 will be build which is the first entry. This is followed by doubled values respectively. Last entry is the maximum limit (300).
- '5,10;10,20;20,40;40,80' out of each of the pairs separated by ';' a random number of this range gets calculated respectively.

Default: blank (old behavior is enabled)

Setting: prov_polling_enabled

Description: If set to ‘on’, automatic periodic provisioning server polling for upgrades is enabled.

Values: on, off

Default: off

Setting: prov_polling_mode

Description	■ **rel:** Relative mode, enables phones to check for software or configuration upgrades after every X seconds. You can set the value of X in parameter prov_polling_period.
	■ **abs:** Absolute mode, enables phones to check for software or configuration upgrades at an exact time, based on the 24-hour clock. You can set the time in the parameter prov_polling_time.
	■ **random:** Random mode, enables phones to check for software or configuration upgrades randomly. The randomness depends on the period set in prov_polling_period. If the period is less than one day, phones will check for upgrades at any time of the period randomly. If the period is greater than one day, for example 3 days, phones will check for upgrades within 3 days randomly and depend on the time period between the values in prov_polling_time and prov_polling_time_rand_end randomly also.
	Random Case 1: prov_polling_period >= 1 day
	`prov_polling_enabled=on`
	`prov_polling_mode=random`
	`prov_polling_period=86400`
	`prov_polling_time=18:00`
	`prov_polling_time_rand_end=18:10`
	This case will have provisioning every day between 18:00-18:10, starting from the next day after setting being set. A general rule: If prov_polling_period >= 1 day, provisioning will occur randomly in specific time interval inside this prov_polling_period.
	Random Case 1: prov_polling_period <= 1 day
	`prov_polling_enabled=on`
	`prov_polling_mode=random`
	`prov_polling_period=3600`
	`prov_polling_time=18:00`
	`prov_polling_time_rand_end=18:10`
	In this case, the period is 3600s and will have provisioning checked at intervals randomly selected between 0 and 3600 seconds, regardless of the time start and time end. A general rule: if the period is less than one day, phones will check for upgrades at any time of the prov_polling_period randomly. Time start and end is not used in this case.

Values:	rel, abs, random
Default:	rel

Setting:	prov_polling_period
Description:	Check for software or configuration upgrades within this time interval (in seconds).

Values:	Time in seconds. e.g. 3600 (1 hour).
Default:	0

Setting:	prov_polling_time
Description:	Time to start polling of software or configuration upgrades.
Values:	hh:mm (24-hour clock format) e.g. 00:00, 23:00
Default:	00:00

Setting:	prov_polling_time_rand_end
Description:	Time to stop polling of software or configuration upgrades.
Values:	hh:mm (24-hour clock format) e.g. 00:00, 23:00
Default:	00:00

Setting:	provisioning_order

Description: One can determine what provisioning types in which order the phone is attempting from these given provisioning types: **redirection pnp dhcp tr69**. With the key words **stop** or **proceed** after the specific provisioning type, one is specifying what to do after the respective step:

- key word: **stop** - after the respective provisioning type was finished successfully, the provisioning process is stopped. If the provisioning type fails, the provisioning process continues to the next type.
- key word: **proceed** - the provisioning process always continues after the respective provisioning type, even if the provisioning type was successful.

The provisioning type redirection is taken as successfully finished if a different setting server has been accessed successfully. The other types are taken as successfully finished if arbitrary URLs have been accessed with success regardless whether it lead to a different setting server or not.

When the value of this setting is changed, the phone immediately restarts the provisioning process using the new order.

```
Example:

    Value: redirection:stop pnp:stop tr69:stop

    Description: Always the redirection service will be
    accessed first regardless of what PNP has delivered
    before.
        If redirection fails PNP and/or TR69 will be used for
        provisioning in this order.
        In this case the DHCP request is still made, but
        provided redirection server information is ignored.
```

Values: redirection:stop/proceed pnp:stop/proceed dhcp:stop/proceed tr69:stop/proceed

Default: redirection:stop pnp:stop dhcp:stop tr69:stop

Setting: publish_presence

Description: When this feature is set to on, the phone sends out PUBLISH SIP messages showing the phone's status.

Values: on, off

Default: off

Setting: pui_states_allowing_state_switch_on_activity

Description:	Lists all PUI states that may allow auto-switching to activity-state. Values (below) shows the list of all the possible PUI states. See also settings goto_monitor_state_on_line_activity and goto_virtual_keys_state_on_activity
Values:	Space separated list of keywords: Menu Addressbook TBook_entry List_pkeys Select_active_line Status_messages Status_msg_details clock Confirm Wizard Edit_number Calling Call_completion Ringing Connected Transfer Holding Terminated Edit Change_volume Ringtone Settings Mwi Info Auto_redial Conference Details Change_presence Traverse_buddy Dialog Multicast Minibrowser_Message Idle Minibrowser
Default:	idle

Setting:	quick_transfer
Description:	If quick_transfer=**new_call**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will put the active call on hold and a new call will be initiated dialing out to the configured number associated with the key. If quick_transfer=**blind**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will blind transfer the active call to the configured number associated with the key. If quick_transfer=**attended**, pressing a Speed Dial / BLF / Extension / BLF (Server) key during call active will put the active call on hold and initiate a new call to the configured number for attended transfer. User can complete the transfer as early attended or attended transfer via the "Transfer" key.
Values:	new_call, blind, attended
Default:	new_call

Setting:	reactivate_wireless_offhook_pause
Description:	In most cases the headset is already offhook before the hook button is pressed on the headset to make a call. This is necessary to e.g. play dtmf tones or the dial tone. But by pressing the hook button in this state the headset goes onhook. That's why the phone sends an offhook command automatically to the headset after a defined time. This time is defined by this setting in milliseconds. Different Headset types needs different timing. If the time is to short, then an endless toggling between onhook and offhook could be the result.

Values:	Positive Integer
Default:	1100

Setting:	reboot_after_nr
Description:	SYSTEM INTERNAL If the phone becomes unregistered and this setting is set to a value bigger 0, the phone will reboot after the amount of time has elapsed this setting is set to. This may be useful because a restart of the phone may fix the issue why the phone fell unregistered before.
Values:	Integer
Default:	0

Setting:	reciprocal_hold
Description:	This is for PBX that holds against client hold. Must be set to 'on' to invite "held by" lines for conference. Normally you don't want this because otherwise music on hold (MOH) could be possible in conference. But we can't differentiate between the hold request of the server or a participant. Typical PBX is Metaswitch.
Values:	on, off
Default:	off

Setting:	record_dialed_calls
Description:	Should be disabled, if dialed calls from this identity should not be taken into account for the dialed calls list.
Values:	on, off
Default:	on

Setting:	record_missed_calls
Description:	Should be disabled, if incoming calls to this identity should not be taken into account for the number of missed calls. See also settings: record_missed_calls_cwi_off, sip_cancel_reasons_to_ignore_missed_call, gnore_missed_calls_on_busy

Values: on, off

Default: on

Setting: record_missed_calls_cwi_off

Description: When this setting is 'on', the missed calls are recorded even if call waiting indication is off.

See also settings: record_missed_calls_cwi_off, sip_cancel_reasons_to_ignore_missed_call, gnore_missed_calls_on_busy

Values: on,off

Default: on

Setting: record_received_calls

Description: Should be disabled, if received calls to this identity should not be taken into account for the received calls list.

Values: on, off

Default: on

Setting: recording_mechanism

Description: Controls how to record calls, these keywords are allowed:

SIP -> sends sip INFO with "Record: on" or "Record: off"

DTMF4242 -> sends DTMF-sequence, the same one for starting and stopping - please substitute 4242 with the sequence you need in your environment.

NONE -> no recording at all

SIP_CALL:42@pbx.com -> make a conference by calling the configured SIP-URI. Behind this URI should be a recorder that auto-answers all calls and that records them.

Values: SIP, DTMF____, NONE, SIP_CALL:______

Default: SIP

Setting: redirect_ringing

Description:	Allows to redirect an incoming call to a prespecified number using function keys e.g. Speed Dial, Extension etc. Can be turned off to disable such automatic transfers in a call centre environment.
Values:	on, off
Default:	off

Setting:	refer_brackets
Description:	Switch additional brackets on or off in the Signaling for Refer-To. Some devices rely on this setting. This setting is per identity.
Values:	on, off
Default:	off

Setting:	referred_by_brackets
Description:	If value is set to "on", for the REFER SIP message, the Referred-By URI is enclosed with angled brackets. Some servers (e.g. Jive) rely on these brackets. See also refer_brackets.
Values:	on, off
Default:	on

Setting:	refuse_call_pickup_of_connected_calls
Description:	If enabled, the phone prohibits to send out an INVITE of a pickup call that has already been established.
Values:	on, off
Default:	off

Setting:	register_http_contact
Description:	This settings decides if the phone must add the http URL of the phone as additional contact information WARNING: Turning this setting on may cause a complete loss of VoIP ability if the proxy/registrar does not support it. We urge you strongly to leave it on off if you are not absolutely sure that it is supported by your proxy/registrar.
Values:	on,off

Default:	off

Setting:	reject_calls_with_603
Description:	When call is rejected (i.e. using the X button), the phone usually sends failure SIP reply "486 Busy Here". If this setting is on, the phone will send "603 Declined" instead of "486 Busy Here" when the call is rejected. Please note that this not affect the case when the call is rejected because the phone is busy. This setting is usefull if you want to have two different failure replies: "486 Busy Here" in case the phone is busy; "603 Declined" when the call is rejected.
Values:	on, off
Default:	off

Setting:	release_sound
Description:	Set this to on if the release sound should be played when the remote party terminates the call. Set this to off if no sound should be played when the remote party terminates the call. (A busy sound is played when the remote party is busy or denies an incoming call.)
Values:	on, off
Default:	off

Setting:	release_xferred_call_on
Description:	When a call is transferred, the transferred party sends notifications to the tranferring party about the progess of the call towards the transfer target. The notifications include the current SIP response code received by the transferred party. This settings specifies at which point the transferring party will release the transferred call. This occurs when the reported SIP response code is equal to or greater than the value of this setting. This setting should be administered in tandem with the setting retrieve_xferred_call_on. Note that when marking a call with save transfer the phone will ignore the actuall setting value and instead act as if this was set to 200.

Values:	SIP response code
Default:	180

Setting:	remote_3264_hold
Description:	Assume the remote side supports RFC 3264 style hold when a=sendrecv is missing in SDP. When set to off, the phone will signal hold as specified in RFC 2543 when the remote side's SDP is missing the a=sendrecv attribute, i.e. by setting the IP in the SDP to 0.0.0.0.
Values:	on, off
Default:	on

Setting:	remote_blacklist_action_timer
Description:	Time in seconds, the phone will take to make sure whether the caller is blacklisted or not in remote / server side black list. Regarding the action URL related to this timer, see action_blacklist_url
Values:	Numeric value. Time in seconds.
Default:	1

Setting:	remote_contact_header_field
Description:	By default, the phone uses the SIP URI provided in the "From" header field of an incoming SIP INVITE message to store the entry in the missed or received call list. When this setting is set to "contact", the SIP URI in the "Contact" header field is used instead. When the "Contact" header field is not present, the default is used.
Values:	from, contact
Default:	from

Setting:	replace_header_fire_action_url
Description:	If on, action URLs for "Incoming call" and "On disconnected" will be fired after transfer with replace headers
Values:	on, off
Default:	blank

Setting: require_prack

Description: Defines whether Required:100Rel will be send or not.

This influences whether a early-dialog via PRACK will be established (if the opposite offers this by sending Supported:100Rel) or not.

This could be useful for playing announcements or music/ring-back-tones during the time the call is in Ringing-state.

Even if set to off, the phone will still offer 100Rel in the Supported-Header if it sends the INVITE (is the originator of the call). If B responses with Required: 100Rel it will send the ACK, independent of this setting.

For preventing sending 100Rel as supported (and by that sending PRACK) you have to set additionally setting send_prack to off.

Values: on, off

Default: on

Setting: reset_settings

Description: You can provide one or several of the below values space separated in order to reset only network, SIP stack, user, function key, speeddial related or other settings.

Values: main, net, stack, user, fkey, speeddial, phonebook

Default: blank

Setting: restrict_uri_queries

Description: By default, if setting admin_mode_password and http credentials (settings http_user and http_pass) are set and hidden tags are activated (setting use_hidden_tags), then query strings in URIs (the part after the "?") are restricted to a very limited number of cases.

By setting restrict_uri_queries to false, query strings are not restricted anymore, so you can use hidden tags and passwords, even if you need stuff like "dummy.htm?settings=save&...".

Values: on, off

Default: on

Setting: retrieve_xferred_call_on

Description: When a call is transferred, the transferred party sends notifications to the tranferring party about the progess of the call towards the transfer target. The notifications include the current SIP response code received by the transferred party. This settings specifies at which point the transferring party will deem the transfer failed and retrieve the transferred call (which up to this point is still on hold). This occurs when the reported SIP response code is equal to or greater than the value of this setting. This setting should be administered in tandem with the setting release_xferred_call_on.

Note that when marking a call with save transfer the phone will ignore the actual setting value and instead act as if this was set to 200.

Values: SIP response code

Default: 400

Setting: retry_after_failed_register

Description: This value specifies after how many seconds the phone should attempt to reregister when the initial registration has failed. If this value is zero, the phone will make no such attempt.

Value can be single integer value (range '1' to this value) or a range like '2,10'. Randomizing 10 percent if single value is configured (e.g. 300 +- 30sec)

The value can also be, for example '3,6:300'. In this case when the phone loses the registration, a random value in seconds between 3 and 6 will be chosen and after this time the phone will try again. After that the value is doubled and the phone will try again until registration succeeds or the timer rechaed the second value. This is the maximum timer value. So basically the longer the phone is unregistered the longer it takes to reregister.

Values: 1 - 1209600

Default: 300

Setting: retry_after_failed_subscribe

Description: When subscription fails this settings describes the value in seconds after which the phone will try again.

Be aware: don't confuse this setting with the SUBSCRIBE expiration, which is defined by setting user_subscription_expiry

Values: Positive Integer

Default: 600

Setting: ring_after_delay

Description: The phone delays playing the ringer for the given amount of seconds. But the message LED still rings from the beginning.

Values: Integer, blank

Default: blank

Setting: ring_count

Description: This setting is used for synchronisation of Call Forwarding Timeout/NoAnswer for Broadsoft.

Values: Integer

Default: 5

Setting: ringer_animation

Description: The ringer animation can be switched off by <off> to save space for displaying longer numbers by applying a line break. There is also a different title displayed, which allows to determine the SIP identity called: To: <SIP Identity Number>

Values: on, off

Default: on

Setting: ringer_headset_device

Description: If you want to hear the ring tone via the headset only, choose headset; otherwise, speaker. Both headset and speaker can be enabled. Then the configured ring tone will be played on the speaker of the phone and the headset plays its own build in ring tone (e.g. 3 short beeps). Some headsets don't have a build in ring tone (most wired USB headsets). But some of them can give a visual indication.

Values: speaker, headset, headsetloud

Default: speaker

Setting: ringing_time

Description: **SYSTEM INTERNAL**

Time in seconds how long an incoming call should ring before the phone denies it.

Values: 0 - 86400

Default: 120

Setting: ringing_title

Description: SYSTEM INTERNAL

The title that appears in the ringing state

Values: String

Default: lang_ringing

Setting: rtcp_xr

Description: Specifies of which parts the voice quality report should be composed of. The report is encapsulated in a SIP PUBLISH message that is send if a call is terminated.

See also setting vq_report_collector

Values: loss, dup, jitt

Default: blank

Setting: rtp_codec_size

Description: This is the codes-packet-size measured in milliseconds used when initiating rtp-streams that are independant of any sip-identity. Only current use-case: multicasts.

Values: 1 - 60

Default: 20

Setting: rtp_codec_type

Description: This codec is used when initiating rtp-streams that are independant of any sip-identity. Only current use-case: multicasts.

Values: pcmu,pcma,gsm,g723,g726-32,aal2-g726-32,g729-annexb=no,g729,g722

Default: pcmu

Setting: rtp_early_media_ring_fallback

Description: Time in milliseconds until the phone plays the internal ringer after early media announcement is finished.

Example:

```
<--- INVITE (outgoing phone call)
---> 180 Ringing (phone plays internal ringtone)
...
---> 183 Progress + SDP (phone plays the incoming early media
                         instead of internal ringtone)
...
---> 180 Ringing (if early media is disrupted for x seconds
                the phone will play the internal ringtone again)
and so on
```

Values: Positive Integer

Default: 4100

Setting: rtp_keepalive

Description: On a hold call the phone sends out STUN packets to keep the RTP port open by default. Set this setting to off to switch this behaviour off.

Values: on, off

Default: on

Setting: rtp_port_end

Description: If you want to set up the port range out of which the RTP ports will be dynamically taken, specify the start port (setting rtp_port_start) and end port number, respectively, in these fields.

Values: valid port number

Default: 65534

Setting: rtp_port_start

Description: If you want to set up the port range out of which the RTP ports will be dynamically taken, specify the start port and end port number (setting rtp_port_end), respectively, in these fields.

Values: valid port number

Default: 49152

Setting: save_latest_callrecords_to_flash

Description: If "on" the call records (missed/received/redial) will be saved in the settings callrecord_... so that they'll be available after reboot.

Values: on, off

Default: on

Setting: scroll_outgoing

Description: Turn on/off active line scrolling using navigation key in idle state.

Values: on, off

Default: on

Setting: scroll_text_interval

Description: Time in ms to make the next step for text scrolling.

Values: Integer

Default: 250

Setting: scroll_text_step_count

Description: Defines the number of steps a text is scrolled, e.g. when =1 a scrolling text would first show it's beginning and next its end. For smoother scrolling you will need a high number. Text always scrolls at least 1 pixel per step.

Possible scroll pause when showing beginning or end do not count as extra scroll steps.

Values: Integer > 1

Default:	12

Setting:	scroll_text_wait_multiplier
Description:	The setting describes for how many scroll-steps the scrolling is paused when its beginning of a scrolling text is shown. For phones that don't use circle-scroll-technique, but instead scroll to the end and then start up front again, this stop-time also describes the pause at the end.
Values:	Integer > 1
Default:	4

Setting:	secondary_dialtone_when
Description:	When user enters a number into the dial-screen and it matches one of the groups defined herein, a dial-tone will be played.
Values:	space separated list of dial strings
Default:	blank

Setting:	seconds_to_show_transfer_success_for
Description:	This setting makes it possible to have the phone display a success message when a transfer has been completed successfully. The setting defines for how many seconds the message will be shown. The default setting is 0 (zero seconds), i.e., no success message will be shown.
Values:	integer >= 0
Default:	0

Setting:	send_prack

Description: Enables/Disables sending Supported:100Rel and by this whether early-dialogs by PRACK will be offered.

Enabling this could be useful if the opposite wants to play music/ring-back-tone or announcements before the call is connected.

- On -> Supported:100Rel will be send (and opposite could initiate Early-Dialog by sending Required:100Rel)
- Off -> Supported:100Rel wont be send (and opposite gets no chance to initiate Early-Dialog)

Note:This does not influence whether the phone itself will send Required:100Rel if from opposite Supported:100Rel is signaled and by this initiating a early-dialog. This behavior is influenced by require_prack -- see setting require_prack.

Values: on, off

Default: on

Setting: send_starcodes_with_audio

Description: When enabled the phone will make an actual call with audio instead of just sending an sip invite whenever it has to dial starcodes (see these starcode settings for redirect_always_on, redirect_always_off, redirect_busy_on, redirect_busy_off, redirect_time_on, redirect_time_off, dnd_on, dnd_off). If the PBX plays a confirmation message for certain starcodes (for example 'Do-not-disturb activated') and this setting is on, the user will be able to hear this confirmation message.

Be aware that there can only be one outgoing audio-call at a time, so this setting doesn't work well when issuing starcodes for multiple identities at once.

Values: on, off

Default: off

Setting: server_directories

Description: If the on-line telephone directory search is to be limited to certain directories or groups these may be specified here, separated by a space. If left empty, all directories found will be searched.

Values: space separated list of strings

Default: blank

Setting:	server_enforced_kb_lock
Description:	This setting determines whether the provisioning parameters received via OCS inband provisioning are mirrored on the phone. These are: ucEnforcePinLock -> setting enable_keyboard_lock ucMinPinLength -> setting min_pin_length ucPhoneTimeOut-> setting keyboard_lock_timeout
Values:	on, off
Default:	on

Setting:	service_mode_login
Description:	With this setting, you can specify the username for the service mode login. Together with the setting service_mode_password, it provides an additional maintenance account apart from the administrator login. Note: This setting should be provisioned with read-only permission.
Values:	String
Default:	blank

Setting:	service_mode_password
Description:	With this setting, you can specify the password for the service mode login. It is used together with setting service_mode_login, to provide an additional maintenance account apart from the administrator login. Note: This setting should be provisioned with read-only permission.
Values:	String
Default:	blank

Setting:	session_timer
Description:	If SIP Session Timer Support is enabled, this option specifies the SIP session timer in seconds. For instance, a Re-INVITE will be sent after 50% of its value has elapsed.
Values:	Integer
Default:	3600

Setting: setting_server

Description: Enter the URL of the settings server from where you would like to obtain the configuration file to configure your phone.

Values: URL

Default: blank

Setting: settings_cyclic_store_timer

Description: Automatically store all settings to flash at the specified interval (measured in hours). Disable the setting with 0 (zero).

ET605 phones save settings to the Flash memory only upon certain events. This setting prevents the loss of call records (missed, received, dialed) when power is lost.

Values: 0 - 595

Default: 0

Setting: settings_refresh_timer

Description: If a value greater than 0 is set (=number of seconds) the phone configuration will be requested from the setting server after the time has elapsed. After fetching the settings from the "setting server URL" they will be applied and the timer will be reset to the latest received value.

Values: Integer

Default: 0

Setting: short_cancel_denies_call

Description: If value is true a short key press of cancel key will deny an incoming call. A long press (2sec.) cancels the connected call. If value set to false it works vice versa.

Note: This setting will only take effect on phone models **without** call screens settings. For all other phones, you can select which call to cancel by navigating through the list of available calls.

Values: on, off

Default: on

Setting: short_form

Description: In order to let the phone generate short compact SIP headers this option should be enabled. Otherwise the old usual style of SIP headers will be generated.

Values: on, off

Default: off

Setting: show_call_status

Description: If turned on, the call progress is shown in the headline of the call progress window e.g. (100 Trying, 180 Ringing etc).

Values: on, off

Default: off

Setting: show_clock

Description: Specifies whether or not clock and date should be displayed (at the idle screen usually).

If <false>, the value of setting phone_name is displayed instead (if set).

Values: on, off

Default: on

Setting: show_connected_call_in_monitor_view

Description: Show or hides the connected calls within the call monitor view.

Values: on, off

Default: on

Setting: show_desktop_msg_in_call_screens

Description: Messages received via SIP MESSAGE outside an INVITE are displayed on the desktop of the idle screen. When this setting is enabled, the message will also appear in call screens.

Note: Messages received inside an INVITE dialog are only displayed in the 'connected' screen.

Values: on, off

Default: off

Setting: show_diversion

Description: When this feature is set to on, the phone shows the information available through Diversion header in the incoming INVITE.

Values: on, off

Default: on

Setting: show_history_info

Description: When this feature is set to on, the phone shows the information available through History-Info header in the incoming INVITE.

Values: on, off

Default: on

Setting: show_ivr_digits

Description: This setting controls whether digits pressed during a connected call are shown on the display or not. These digits are usually used to control IVR prompts and to enter user specific information e.g. calling card number, pin codes, credit card number, billing info etc.

Turning this setting off ensures privacy by disabling the display of these digits. The actual keys are either not shown at all or replaced replaced by *.

Values: on, off

Default: off

Setting: show_local_line

Description: Shows local sip line index during call states in adition to the remote user display name/number/url

Values: on, off

Default: off

Setting: show_name_dialog

Description: When this setting is turned on, the call monitoring state shows display names for remote and local users found in the body of incoming dialog info notifies, as long as the display_method setting is set to name as well. If this setting is turned off, the user name in the uri's will be shown to maximize display space.

Values: on, off

Default: off

Setting: show_redundant_context_keys

Description: When showing a list in minibrowser while the minibrowser-xml does not define any context-keys on its own: this setting decides if to show navi-keys instead or no keys at all.

Values: on, off

Default: off

Setting: signaling_tos

Description: This option enables the phone to support quality of service (QOS) for SIP traffic in a network. This makes sense only if all parts of the involved network also support QOS.

Values: 0-255

Default: 160

Setting: sip_body_trace_size

Description: This setting determines how many bytes of the original body to keep in the trace. If you don't want the body to be truncated at all, set this setting to -1 (messages written to a USB storage device (see setting usb_storage_siptrace) are never truncated, irrepective of the value of this setting).

Values: Integer >= -1

Default: -1

Setting: sip_cancel_reasons_to_ignore_missed_call

Description: When phone misses an incoming call, it usually records it in it's missed calls list so the user can call the caller back when he/she sees the missed call. There are certain scenarios where this is not desired. E.g. imagine you are logged in with your account on two places (e.g. office phone and at home). If you get a call, both phones will ring. If you pick up the call on one phone, you don't need the wrong missed-entry in the other. If the PBX usually includes the reason in it's cancel-message to thew other phone which might look like this:

```
CANCEL <your account> SIP/2.0
Via: ...
From: ..
To: ...
Reason: SIP ;cause=200 ;text="Call completed elsewhere"
...
```

With the help of this setting you can determine which reasons will inhibit creating a missed record. Each reason is evaluate separately, if any one matches the one inside the SIP-Cancel the call will not be treated as missed.

See also settings record_missed_calls, record_missed_calls_cwi_off, ignore_missed_calls_on_busy

Values: space separated list of reasons

Default: text='Call completed elsewhere' text='Call was replaced' cause<300

Setting: sip_failover_response

Description: Defines a certain SIP Response code and reason phrase for Register and Invite requests.

It allows you to smoothly move the phone between service hosts.

Never use this option unless you exactly know what you are doing!

If the phone receives that response for an Register request, it

- clears the Dirty Host Cache

- add the response transport:host:port to the dirty host cache for

  a) Retry-After: time

  b) configured dirty host ttl

  c) 5 minutes

- restart the registration process for all other hosts indicated by DNS SRV responses

5 minutes is choosed to avoid an sip registration loop.

If the phone receives that response for an Invite request, it

- clears the Dirty Host Cache

- add the response transport:host:port to the dirty host cache for

  a) Retry-After: time

  b) configured dirty host ttl

  c) 5 minutes

- restart the registration process for all other hosts indicated by DNS SRV responses

- on successfull registration restart the Invite request

Values: <response code><space><response phrase>

Default: blank

Setting: sip_failover_response_reg

Description: Defines a certain SIP Response code and reason phrase for Invite requests. It allows you to force a registration with an invite response.

Never use this option unless you exactly know what you are doing! Do not interfere with existing response codes and their handling!

If the phone receives that response for an Invite request, it

- acknowledges the response

- initiates an registration against the response sender

- on successfull registration restart the Invite request to the response header

Values: <response code><space><response phrase>[<pipe><response code><space><response phrase>...]

Default: blank

Setting: sip_force_sendrecv_on_invite_wo_sdp

Description: INVITE Requests without SDP should not change the state of the SDP. However, some servers can not handle this and need a sendrecv in the response whatever the previous state was. If on, the phone sends sendrecv in the response for INVITE Requests with no SDP.

Values: on, off

Default: off

Setting: sip_health_check

Description: Enables/Disables the status polling of primary SBC's if the phone has been failed over to the backup SBC's.

Not recommended due to additional traffic.

If enabled the phone will send Option Requests within the account_health_check to the primary SBC. Any SIP Response will be taken as host is available again and the entry will then change to quarantine state. If the quarantine period timer finally fires, it will trigger a reregistration of all accounts to the primary SBC again.

The following settings configure the timing and show their default values (all in seconds):

sip_health_check: off // en/disables the health check

sip_health_check_base_time: 30

sip_health_check_max_time: 300

sip_health_check_static_time: 300

The value of dirty_host_ttl needs to be chosen "large enough", lets say a couple of hours or something similiar. The SIP Options resend time is then calculated as

health_check_ubw = min(health_check_max_time, base_time * 2^num_retries)

health_check_ubw *= rand(50..100%)

health_check_ubw += health_check_static_time

The same algorithm is used for the quarantine_period of the primary SBC, except that the static and max times are adjustable:

sip_quarantine_max_time: 600

sip_quarantine_static_time: 1800

Values: on, off

Default: off

Setting: sip_health_check_base_time

Description: See setting sip_health_check.

Values: positive integer

Default: 30

Setting:	sip_health_check_max_time
Description:	See setting sip_health_check.
Values:	positive integer
Default:	600

Setting:	sip_health_check_static_time
Description:	See setting sip_health_check.
Values:	positive integer
Default:	300

Setting:	sip_ip_dialin_content_types
Description:	Phones can be called without account and by ip directly if network_id_port has been configured to port 5060. By default and due to security concerns only application/sdp sessions are allowed to this builtin ip identity. To allow other session types like application/csta+xml (remote control) add the desired type to this filter (e.g. "application/sdp, application/csta+xml"). See also settings: network_id_port, csta_control, csta_challenge.
Values:	<empty>, application/sdp, application/csta+xml
Default:	application/sdp

Setting:	sip_max_challenges

Description: Value controls how many times the phones tries to answer an sip response indicating that the phones sip request did not pass authorization (challenged).

Example with default value equal 1

```
<-- REGISTER Request (no authorization header)
--> 407 Response
<-- REGISTER Request (with authorization header)
--> 200 Response
```

Example with value equal 2

```
<-- REGISTER Request (no authorization header)
--> 407 Response
<-- REGISTER Request (with authorization header)
--> 407 Response again
<-- REGISTER Request (with authorization header)
--> 200 Response
```

Values: integer >=1

Default: 1

Setting: sip_proxy

Description: If DHCP option 120 has been provided, the content will be stored in this setting.

Values: URL

Default: blank

Setting: sip_quarantine_max_time

Description: See setting sip_health_check.

Values: positive integer

Default: 600

Setting: sip_quarantine_static_time

Description: See setting sip_health_check.

Values: positive integer

Default: 1800

Setting: sip_reconnect_on_rejected_refer

Description: Defines if the phone does automatic reconnect to A party if a REFER (blind/attended transfer) has been rejected.

Suppose the following call flow:

- A calls B, A and B talking

- B puts A on hold

- B calls C, B and C talking

- B presses transfer key twice to initiate transfer A <-> C

- the call transfer (REFER request) will be rejected, e.g. with SIP Response Code 603

now the value of this settings decides if:

- B will be automatically connected to A again, while C is on hold

(value "on": old behaviour, not default anymore)

or

- B holds A and C to select the party to talkagain after the transfer failure

(value: off: new and default behaviour introduced with this setting).

Values: on, off

Default: off

Setting: sip_request_timeout

Description: Specifies the amount of time before a sip client transaction will be timed out.

Builtin value is "64", which means the max transaction time is calculated as '64 * sip_retry_t1' before the transaction is considered to be failed. After that the routing tries to send the request to the next possible server or the request will be canceled at all.

Values: 1-64

Default: 64

Setting: sip_retry_t1

Description: Set the retry timer in milliseconds after which an unanswered request is resent. If it is set to 500, the phone will resend the unanswered request after 500, 1000, 2000, 4000, 6000 ... 31500 ms. If the request is still unanswered after this procedure, an error message will be shown on the display.

Values: Integer >= 100

Default: 500

Setting: sip_shutdown_timeout

Description: Time in seconds how long the phone waits to handle unregister/unsubscribe during reboot process.

Values: integer

Default: 10000

Setting: sip_stop_subscriptions_on_register_failure

Description: Starting with the above versions, all outgoing subscriptions will be silently stopped on a registration failure. If the registration succeded again the subscription will be restarted from scratch. This behaviour is helpfull for all pbx's who link registration and subscriptions together.

However, from a pure sip perspective view registration and outgoing subscriptions are not related to each other so you might turn off this behaviour by configuring this option to off.

Values: on, off

Default: on

Setting: sip_trace_size

Description: Determines the number of messages to keep in the trace. Once this number is reached, the oldest message is removed when a new on is added. If you want to trace only to a USB device (see setting usb_storage_siptrace), you may set this value to zero.

Values: 0-500

Default: 100

Setting:	sip_tracing
Description:	Switches SIP tracing on or off.
Values:	on, off
Default:	on

Setting:	skip_provisioning_urls_on_tls_error
Description:	If this setting is enabled, skip any URL which fails due to a TLS error and continue with the next one (if any) instead of retrying. This setting was introduced for testing purposes, it is not advised to enable it in a production environment.
Values:	on, off
Default:	off

Setting:	snmp_port
Description:	Type in the port to be used for SNMP communication.
Values:	valid port number
Default:	161

Setting:	snmp_trusted_addresses
Description:	Specify the address range (in CIDR notation) solely from within which subnet SNMP requests will be accepted e.g. 192.168.0.0/16
Values:	Subnet in CIDR notation
Default:	blank

Setting:	sort_server_dir_result_by_last_name
Description:	When set to 'on', the results returned from an on-line telephone directory search will be sorted by Last Name (Surname) then First Name (Given Name). When set to 'off', the results will be sorted by First Name (Given Name) then Last Name (Surname). If the record does not include a Last Name, the Display Name is used instead.

Values: on, off

Default: on

Setting: soundcard_event_map

Description: This setting contains necessary parameters for soundcards (in this special case USB headsets):

Headset Value

Plantronics Blackwire C620 VID=047f:PID=aa00:MUTE=101:VOL+=104:VOL-=105:HOOK=100

Plantronics Savi W430 (Dect D100) VID=047f:PID=ab01:HOOK=10f

Plantronics CS540a (plus APU-70) VID=047f:PID=0410:HOOK=100

Plantronics Voyager PRO UC BlueTooth VID=0a12:PID=100d:HOOK=38/1

Values: VID=<vendorid>:PID=<productid>:VOL+=<vol-up-code>:VOL-=<vol-down-code>:HOOK=<hookcode>:MUTE=<mutecode>

Default: blank

Setting: speaker_dialer

Description: Usually the speaker key can be used to start a dial attempt, if this behaviour is unwanted, it can be disabled here.

Values: on, off

Default: on

Setting: speaker_receive_call

Description: Usually the speaker key can be used to receive an incoming call, if this behaviour is not desired, it can be disabled here. This setting is valid for headset key too.

Values: on, off

Default: on

Setting: speed

Description:	Speed dial items 0-9, 10, 11, 12-32 are specifying the number which may be called via keys 0-9, *, * and numbers 12-32 respectively.
Values:	phone number
Default:	blank

Setting:	startup_presence
Description:	When enabled, the phone's XMPP client will report the user's presence status when the phone starts up.
Values:	on, off
Default:	off

Setting:	states_ignored_in_goto_vkeys_on_activity
Description:	List the state-strings that don't signal activity. When goto-virtual-keys-state-on-activity is set to true, any state-change of a virtual-key to a state not listed here will make the virtual-key-screen pop up.
Values:	Comma-separated list of states that should be ignored.
Default:	on,off,,undefined,free,no buddy,initial

Setting:	status_msgs_that_are_blocked
Description:	Lists all statuses that should never appear in PUI. See also settings: s_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_important, status_msgs_with_audio_indication, status_msgs_to_pop_up and idle_status_btn_index

Values: space separated list of keywords

PhoneHasFirmwareUpdate PhoneWantsReboot PhoneHasDisabledSipStack PhoneHasVpnError PhoneHasLowMemory PhoneRefusedHugeXcapSync CurrentIdentityIsNotRegistered Identity01IsNotRegistered Identity02IsNotRegistered Identity03IsNotRegistered Identity04IsNotRegistered Identity05IsNotRegistered Identity06IsNotRegistered Identity07IsNotRegistered Identity08IsNotRegistered Identity09IsNotRegistered Identity10IsNotRegistered Identity11IsNotRegistered Identity12IsNotRegistered Identity PhoneIsWaitingForCallCompletion CurrentIdentityForewardsWhenBusy CurrentIdentityForewardsAfterTimeout CurrentIdentityForewardsAlways CurrentIdentityIsDnd PhoneWaitsOnNtpServer PhoneCannotReachNtpServer PhoneHasNoHttpPassword PhoneHasNoAdminPassword PhoneIsLocked PhoneHasIncomingPublicAnnouncement CurrentIdentityHasTextMessages PhoneHasTextMessages CurrentIdentityHasVoiceMessages PhoneHasVoiceMessages ThoneHasMissedCalls ServerMessageToBeShownDirectly EthernetUnplugged FirmwareUpdateFailed VisionConnectionLost PhoneWantsToUpdate DfksFailed IPv4Conflict AudioDeviceIsSpeaker AudioDeviceIsHeadset AudioIsMuted During call On incoming calls PhoneProvisioningStarting PhoneProvisioningInProgress PhoneProvisioningFailed Identity01 Identity02 Identity03 Identity04 Identity05 Identity06 Identity07 Identity08 Identity09 Identity10 Identity11 Identity12 ActiveLocations RemoteOfficeEnabled CallForPickupAvailable DateReminding DateOngoing ExpDeviceCabelingBroken ExpDeviceLimitExceeded ActiveBluetoothConnection UsbDiskConnected CallBackOnBusyInProgress Lync CallBackOnBusyAvailable Lync BtoeStateUnpaired Lync BtoeStatePairing Lync UxmConnected WlanActive CanceledCall HidConnecting HidConnected TryParking StatusLineSystemMessage

Default: PhoneHasVoiceMessages PhoneHasTextMessages PhoneProvisioningFailed

---

Setting: status_msgs_that_are_essential

Description: Lists all statuses that are essential. These messages cannot be deleted from message-list-view.

See also settings: status_msgs_that_show_directly, status_msgs_that_are_blocked, status_msgs_that_are_important, status_msgs_with_audio_indication, status_msgs_to_pop_up and idle_status_btn_index

Values: space separated list of keywords

See setting status_msgs_that_are_blocked

Default: AudioDeviceIsSpeaker AudioDeviceIsHeadset AudioIsMuted PhoneHasNoHttpPassword PhoneHasNoAdminPassword PhoneHasIncomingPublicAnnouncement PhoneIsLocked PhoneHasDisabledSipStack CurrentIdentityIsNotRegistered PhoneIsWaitingForCallCompletion CurrentIdentityIsDnd RingerIsSilent CurrentIdentityForewardsAlways ServerMessageToBeShownDirectly IPv4Conflict

---

Setting: status_msgs_that_are_important

Description: Lists all important status messages. Important messages will make the status-button blink and get listed before the other messages in status message view.

See also status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_blocked, status_msgs_with_audio_indication, status_msgs_to_pop_up and idle_status_btn_index

Values: space separated list of keywords

See setting status_msgs_that_are_blocked

Default: ActiveLocations RemoteOfficeEnabled EthernetUnplugged PhoneHasFirmwareUpdate PhoneWantsToUpdate PhoneWantsReboot PhoneHasDisabledSipStack PhoneHasLowMemory PhoneRefusedHugeXcapSync FirmwareUpdateFailed VisionConnectionLost UsbDiskConnected CurrentIdentityIsNotRegistered Identity01IsNotRegistered Identity02IsNotRegistered Identity03IsNotRegistered Identity04IsNotRegistered Identity05IsNotRegistered Identity06IsNotRegistered Identity07IsNotRegistered Identity08IsNotRegistered Identity09IsNotRegistered Identity10IsNotRegistered Identity11IsNotRegistered Identity12IsNotRegistered PhoneCannotReachNtpServer PhoneHasNoHttpPassword PhoneHasNoAdminPassword Identity01ExtendedRegInfo Identity02ExtendedRegInfo Identity03ExtendedRegInfo Identity04ExtendedRegInfo Identity05ExtendedRegInfo Identity06ExtendedRegInfo Identity07ExtendedRegInfo Identity08ExtendedRegInfo Identity09ExtendedRegInfo Identity10ExtendedRegInfo Identity11ExtendedRegInfo Identity12ExtendedRegInfo CallBackOnBusyInProgress CallBackOnBusyAvailable TbookDownloadFailed TbookFull

Setting: status_msgs_that_show_directly

Description: Lists all statuses that should make it into the statusbar (space separated list). The statusbar only holds one status, so the first one in the list that applies is shown.

You can add the duration to a status (statusmessage[:duration in seconds]). No duration means forever.

An active status message with short duration can't be interrupted, but interrupts a status message with long duration.

Valid duration range:

- Short duration messages: 1 - 30 seconds

- Long duration messages: 31 second - forever

TryParking, CanceledCall and StatusLineSystemMessage can only be used as short duration messages. Wrong values will be set automatically to the minimal or maximal value.

See also status_msgs_that_are_essential, status_msgs_that_are_blocked, status_msgs_that_are_important, status_msgs_with_audio_indication, status_msgs_to_pop_up and idle_status_btn_index

Values: space separated list of keywords

See setting status_msgs_that_are_blocked

Default: StatusLineSystemMessage:3 AudioDeviceIsSpeaker AudioDeviceIsHeadset AudioIsMuted CallBackOnBusyInProgress CallBackOnBusyAvailable PhoneProvisioningStarting PhoneProvisioningInProgress PhoneHasIncomingPublicAnnouncement PhoneIsLocked EthernetUnplugged PhoneHasFirmwareUpdate FirmwareUpdateFailed PhoneWantsToUpdate VisionConnectionLost PhoneWantsReboot PhoneHasDisabledSipStack PhoneHasLowMemory PhoneRefusedHugeXcapSync CurrentIdentityIsNotRegistered PhoneIsWaitingForCallCompletion CurrentIdentityForewardsAlways CurrentIdentityForewardsWhenBusy CurrentIdentityForewardsAfterTimeout CurrentIdentityIsDnd PhoneWaitsOnNtpServer PhoneCannotReachNtpServer ActiveLocations PhoneHasNoHttpPassword PhoneHasNoAdminPassword ServerMessageToBeShownDirectly CurrentIdentityHasVoiceMessages PhoneHasMissedCalls CurrentIdentityHasTextMessages TryParking:5 UxmConnected:5 SxmConnected:5 WlanActive:5 ContactSaved:2 Hoteling:2 HidConnecting:10 HidConnected:5 ExpDeviceCabelingBroken ExpDeviceLimitExceeded

Setting:	status_msgs_to_pop_up
Description:	Lists all statuses that should pop up (full screen) they are active. The list is prioritized, the first active status will pop-up depending on there parameters. How to define the pop-up parameters: statusmessage[:full screen time in ms] parameters values are: 0 < - full screen as long as the status is enabled 0 - can be confirmed by any key > 0 - will be shown full screen for the given time in ms and closed automatically See also settings: status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_important, status_msgs_that_are_blocked and idle_status_btn_index
Values:	space separated list of keywords See setting status_msgs_that_are_blocked
Default:	blank

Setting:	status_msgs_with_audio_indication

Description:	Lists all statuses that should make the phone beep in idle (i.e. no calls) whenever they are active. The list is prioritized, the first active status found determines the beep-mechanism. The beep set of every active status will be played one after the other. How to define the beep-mechanism: statusmessage[:reminder time in s][/index of beep set] beep sets are: 1 - beep one time 2 - beep three times 3 - beep five times e.g.: EthernetUnplugged PhoneWantsReboot/2 CurrentIdentityIsDnd:10/3 PhoneHasMissedCalls:300 1 beep for ethernet cable is unplugged, no repetition 3 beeps for phone wants to reboot, no repetition 5 beeps for do not disturb current identity, repeating them every 10 seconds 1 beep for missed calls, repeating it every 5 minutes See also status_msgs_that_show_directly, status_msgs_that_are_essential, status_msgs_that_are_important, status_msgs_that_are_blocked, status_msgs_to_pop_up and idle_status_btn_index
Values:	space separated list of keywords See setting status_msgs_that_are_blocked
Default:	PhoneHasIncomingPublicAnnouncement

Setting:	stun_binding_interval
Description:	Sets the STUN interval time in seconds. After its expiration a new STUN requests will be send out. If it results in another IP/port the identity will be re-registered.
Values:	integer
Default:	blank

Setting:	stun_server

Description: We reintroduced a STUN keep-alive mechanism for SIP, which can be turned on manually by specifying the address of the STUN server followed by the port number. However, we strongly discourage you from using it, because it can not work properly in symmetrical NAT environments (i.e., linux-based router/firewall). The only general SIP NAT solution is a session border controller (SBC) on the service provider's side.

Values: IP Address:Port

Default: blank

Setting: stutter_timeout

Description: In alphanumeric edit mode the cursor changes after this is the time. Pressing a phone key twice or more in less then this timeout the key value changes to the next character.

E.g.: Timeout set to 300: Press '2' - wait 200ms - press '2' - wait 500ms - press '2'. Result will be 'ba'.

Values: integer

Default: 1000

Setting: subscription_delay

Description: Selects a random number around the given value in seconds to send delayed batch subscriptions. Useful at bootup for certain servers. Its not set by default.

Values: integer

Default: 0

Setting: support_idna

Description: Switch on support for Internationalizing Domain Names in Applications (IDNA). IDNA support is the ability to handle domain names including international special characters.

Values: on, off

Default: off

Setting: support_rtcp

Description:	If enabled, the phone uses the Real Time Control Protocol (RTCP) to measure the quality of the audio (RTP) streams. This setting does not affect the RTCP XR functionality (for RTCP XR you must set the settings rtcp_xr and vq_report_collector).
Values:	on, off
Default:	on

Setting:	support_service_codes
Description:	Disable this setting if you want to prevent the phone to react to the following service code inputs (e.g. in IVR key input scenarios): '*','volume up','*','volume down','#' - reset and reboot phone All other phones: '*','*','#','#' - reboot '*','*','#','*' - restart phone application
Values:	on, off
Default:	on

Setting:	suppress_ringing_during_hold
Description:	Enable this setting if you want to suppress the ringtone when you have one or more callers on hold. Note: When this setting is turned "off" and the ring tone should be played during hold, please also check that the setting call_states_when_knocking does not contain the holding state, otherwise knocking is played instead of the ring tone. This setting is per identity.
Values:	on, off
Default:	on

Setting:	suppress_sip_messages
Description:	If this setting is on, the information received inside SIP MESSAGE requests is discarded. If such a request is received, the phone replies with 200 OK but nothing is displayed on the phone screen.

Values: on, off

Default: blank

Setting: swupd_curl_timeouts

Description: The normal firmware update process downloads firmware images via the unix tool curl. This setting allows to modify some curl options which control the timeout and retry behavior in case of slow downloads and/or errors.

The following curl options get their values from this setting:

--retry

--connect-timeout

--max-time

--retry-max-time

Example: The value "12;30;60;120" would result in the following curl options:

--retry 12 --connect-timeout 30 --max-time 60 --retry-max-time 120

Values: 4 positive integers separated by semicolons

Default: 4;600;600;3600

Setting: swupd_failed

Description: SYSTEM INTERNAL

This setting gets set to failed if a software update has failed.

Values: blank, failed

Default: blank

Setting: sxm_count

Description: **SYSTEM INTERNAL**

Indicates how many Serial eXpansion Modules are currently attached to the phone. This setting cannot be provisioned.

There should be no need to change this setting. As an end-user, please contact your reseller for further details in this regard. As a VAR, please ask VTech support.

Values: 0-3

Default: 0

Setting: syslog_server

Description: Type in the host where a Syslog Server is running to store the log messages coming from the phone.

Values: IP address

Default: blank

Setting: tbook_download_interval

Description: Determines, in seconds, how much time should elapse before the phone initiates a Server Phonebook download. The interval is adjusted to a random value between 90 and 110 percent of the settings value. The interval time is capped at 1209600 seconds (= 14 days). If the setting is empty or contains an invalid value, the download is never initiated. If the value is 0, the download is initiated exactly once after startup.

Values: blank, 0-1209600

Default: blank

Setting: tbook_sort

Description: This settings defines the field used to sort the internal directory (eg. by name, birthday, title, ..). Sorting is done alphabetically. Vaules are numbers representing one of the possible sort-options.

Values: 0 - 13

Integer numbers from 1 to 9 have the following meaning:

1: sort by firstname

2: sort by last name

3: sort by: member, number

4: sort by nickname

5: sort by outgoingId

6: sort by birthday

7: sort by title

8: sort by group

9: sort by organization

Default: 0

Setting: tcp_failover

Description: Toggles the usage of the following settings: tcp_keepidle, tcp_keepcnt, tcp_keepintvl. If set to 'on', the settings are used. If set to 'off', the settings are ignored.

Values: on, off

Default: off

Setting: tcp_keepcnt

Description: The maximum number of keepalive probes TCP should send before dropping the connection.

Values: integer

Default: 5

Setting: tcp_keepidle

Description: The time (in seconds) the connection needs to remain idle before TCP starts sending keepalive probes.

Values: integer

Default:	30

Setting:	tcp_keepintvl
Description:	The time (in seconds) between individual keepalive probes.
Values:	integer
Default:	20

Setting:	tcp_listen
Description:	By default the phone doesn't listen on the network ID port for TCP connections (setting: network_id_port). To change this behaviour, enable this option.
Values:	on, off
Default:	off

Setting:	terminate_ongoing_calls_on_user_deactivation
Description:	When set to true, will cancel all ongoing calls when the associated identity is deactivated via setting user_active. First the deregistration is done and afterwards the calls are canceled.
Values:	on, off
Default:	blank

Setting:	terminate_subscribers_on_reboot
Description:	**SYSTEM INTERNAL** The default setting causes the phone to un- subscribe (SUBSCRIBE & Expire:0) from all open dialog state subscriptions established on function keys (key type "extension" or "destination") before rebooting the phone. However in some environments it might be desired to keep all existing dialog state subscriptions untouched in case of rebooting. In order to do so, please turn this setting <OFF> via mass deployment.
Values:	on, off
Default:	on

Setting:	text_softkey
Description:	If enabled <on>, soft key icons are symbolized by text and not by icons anymore.
Values:	on, off
Default:	off

Setting:	tftp_secret
Description:	Please ask VTech support for details.
Values:	Key which is used to decrypt provisioned encrypted setting files.
Default:	blank

Setting:	time_24_format
Description:	When you select on, the timestamps will be formatted in 24-hour format, otherwise in 12-hour format.
Values:	on, off
Default:	on

Setting:	timer_support
Description:	Define whether sip-stack should support usage of timers. (includes adding headers "Session-Expires" and "Min-SE")
Values:	on, off
Default:	on

Setting:	timezone
Description:	Select the time zone of your geographical location through this option.
Values:	Time zone code
Default:	blank

Setting:	tone_scheme

Description: Select the dialtone you prefer for your phone. Also, DTMF echo will differ (on/off) on different schemes.

Values: country code

Default: blank

Setting: tr69_acs_passwd

Description: Password to be used for the ACS connection.

Values: String

Default: blank

Setting: tr69_acs_url

Description: URL of the TR-069 ACS.

Values: URL

Default: blank

Setting: tr69_acs_user

Description: Username to use for the ACS connection.

Values: String

Default: blank

Setting: tr69_bootstrap

Description: Send a BOOTSTRAP to the ACS. This must be set to on when a new ACS is contacted.

Values: on, off

Default: on

Setting: tr69_cnr_pass

Description: Password for incoming connection requests according to TR-111.

Values: String

Default:	blank

Setting:	tr69_cnr_user
Description:	Username for incoming connection requests according to TR-111.
Values:	String
Default:	blank

Setting:	tr69_download_status
Description:	Auxillary setting for the TR-069 provisioning. Do not manually change it as it is automatically set by the phone.
Values:	String
Default:	blank

Setting:	tr69_events
Description:	Auxillary setting for the TR-069 provisioning. Do not manually change it as it is automatically set by the phone.
Values:	String
Default:	blank

Setting:	tr69_log
Description:	Turn on the logging of TR-069 SOAP envelopes for debugging purposes.
Values:	on, off
Default:	off

Setting:	tr69_params
Description:	Auxillary setting for the TR-069 provisioning. Do not manually change it as it is automatically set by the phone.
Values:	String
Default:	blank

Setting: tr69_use_acs

Description: Toggle use of TR-069 for configuration.

Values: on, off

Default: off

Setting: transfer_dialing_on_other

Description: There is a transfer dial-screen. You get to it by pressing transfer during a connected call. Dialing a number here and then pressing OK or Transfer would normally blind transfer the previously active call to that entered number. This setting allows to instead do a normal outgoing call in that scenario when pressing the OK-key (set this setting to attended if you desire this alternative behaviour).

See also setting transfer_dialing_on_transfer which defines the path to be taken when pressing the transfer-key to confirm the dialing.

Values: blind, attended

Default: attended

Setting: transfer_dialing_on_transfer

Description: There is a transfer dial-screen. You get to it by pressing transfer during a connected call. Dialing a number here and then pressing OK or Transfer would normally blind transfer the previously active call to that entered number. This setting allows to instead do a normal outgoing call in that scenario when pressing the Transfer-key (set this setting to attended if you desire this alternative behaviour).

See also setting transfer_dialing_on_other which defines the path to be taken when pressing non-transfer keys to confirm the dialing.

Values: blind, attended

Default: blind

Setting: transfer_on_hangup

Description: If you want to transfer two calls by placing the handset onhook (one incoming call and one outgoing call), you can switch it on here.

Values: on, off

Default: off

Setting: transfer_on_hangup_non_pots

Description: If you want to transfer two calls by placing the handset onhook (independent of call direction (incoming / outgoing): that will be not a Plain Old Telephone Service "pots") , you can switch it on here. Condition: "transfer_on_hangup" must be set to "on".

Values: on, off

Default: off

Setting: transfer_on_hangup_with_starcode

Description: If setting 'transfer on hangup' is set to on and the first call was picked up with a PBX starcode then the transfer will be done if this setting is set to on. Info: a picked up call with starcode is an outgoing call. But an incoming and an outgoing call is the condition for the 'transfer on hangup'.

Values: on, off

Default: off

Setting: uboot_lock

Description: **Internal**

The uboot lock feature allows to protect the phone from using the uboot/rescue mode update/reset mechanism by unknown users.

Values: Integer

Default: blank

Setting: uboot_version

Description: **SYSTEM INTERNAL**

Contains the version string of the uboot used on the phone. Is a read-only setting

Values: String

Default: dvf99 master 2012.04.01

Setting: update_after_idle_timeout

Description: Timespan in minutes which the phone needs to be idle before an potential software update gets applied.

Values: Positive integer

Default: 0

Setting: update_filename

Description: **SYSTEM INTERNAL**

If the DHCP parameter is enabled and the supported DHCP options have been received in the DHCP offer :

- The value found in **Option 66** will be stored in parameter update_server, e.g. http://server
- The value found in **Option 67** will be stored in parameter update_filename, e.g. vtech/vtech.xml

Values: Path to file

Default: blank

Setting: update_host_f

Description: **SYSTEM INTERNAL**

Internally used only. Must not be changed externally!

Values: N/A

Default: blank

Setting: update_policy

Description: auto_update (Update automatically: load settings from settings server, but the user is not prompted to acknowledge the update, means full automatic provisioning)

ask_for_update (Ask for update: load settings from settings server and the user is prompted to acknowledge the update)

settings_only (Never Update, load settings only: load settings from settings server only, no update is initiated, means update disabled)

never_update (Never Update, do not load settings: do not load any settings or updates from settings server at all, means provisioning disabled)

Attention: update_policy affects all downloaded files: with never_update value the phone will not download any files (VPN config tarball, language files, etc..)

Values: auto_update, ask_for_update, settings_only, never_update

Default: settings_only

Setting: update_server

Description: **SYSTEM INTERNAL**

If the DHCP parameter is enabled and the supported DHCP options have been received in the DHCP offer :

- The value found in **Option 66** will be stored in parameter update_server, e.g. http://server
- The value found in **Option 67** will be stored in parameter update_filename, e.g. vtech/vtech.xml

Values: URL

Default: blank

Setting: upload_font

Description: **SYSTEM INTERNAL**

Specifies a URL pointing to an uncompressed TAR archive allowing PUI font customization. The TAR archive has to contain the fonts, named according to the language scheme which should be replaced:

de.ttf (German)

en.ttf (English)

The tarfile MUST be named "fonts.tar".

Values: URL

Default: blank

Setting: upload_gui

Description: **SYSTEM INTERNAL**

Specifies a URL pointing to an uncompressed TAR archive allowing full PUI customization. The TAR archive shall only contain the images which have to be changed, unchanged files must be omitted!

Values: URL

Default: blank

Setting: upload_license

Description: **SYSTEM INTERNAL**

Used to store the url provisioned by the file upload type license. Prevents refetching the license unless the url changes.

Values: N/A

Default: blank

Setting: upload_moh

Description: **SYSTEM INTERNAL**

Specifies a URL pointing to an wav file allowing MOH file customization.

Values: URL

Default: blank

Setting:	upload_web
Description:	**SYSTEM INTERNAL** Specifies a URL pointing to an uncompressed TAR archive allowing full WUI customization. The TAR archive shall only contain the images which have to be changed (icons, background, etc.), unchanged files must be omitted!
Values:	URL
Default:	blank

Setting:	use_backlight
Description:	On: Backlight is turned off or dimmed after the phone has been inactive for approximately 20 seconds (default setting) or after time in seconds set in text field of Preferences > Dim after. On some phone models, it is additionally possible to adjust the intensity of the backlight in active and idle mode. Off: Backlight is turned off completely Always: Backlight is turned on permanently.
Values:	on, off
Default:	on

Setting:	use_contact_in_refer_to_hdr
Description:	This setting determines which header to use as the source for the Refer-To header in a REFER. When this setting is on the Contact header is used otherwise the remote AoR (taken from To or From header, depending on the direction of the call).
Values:	on, off
Default:	blank

Setting:	use_hidden_tags
Description:	You can protect the phone's web interface with hidden security tags against remote attackers trying to change phone settings with faked HTTP POST requests (XSRF attack).
Values:	on, off

Default:	off

Setting:	use_NTLMv2
Description:	This OCS-only setting determines whether NTLMv2 should be used for authentication. Normally this should be on. Note that NTLMv2 requires the system time and may not work in an environment where no NTP server is available. In this case NTLMv1 should be attempted (the server must allow it) by turning this setting off.
Values:	on, off
Default:	on

Setting:	user_active
Description:	This identity can be disabled by disabling this option. This means this identity is not longer registered anymore.
Values:	on, off
Default:	on

Setting:	user_additional_supported_header
Description:	If your SIP proxy/registrar needs the additional header, it can be enabled here.
Values:	comma separated headers
Default:	blank

Setting:	user_admin_mode
Description:	If set to 0, the admin is allowed to see and edit the users call lists and directory. Besides the user cannot change his/her password. If set to 1, the admin has no access to the useres dictionary and call lists. The user can change his/her own password in the advanced settings of the web interface.
Values:	0,1
Default:	blank

Setting:	user_alert_info
Description:	This URL should point to a web server where audio alert messages are accessible.
Values:	URL
Default:	blank

Setting:	user_allow_inc_dialog_subscribe
Description:	When this setting is 'off', all incoming dialog subscriptions for this identity are rejected with a '403 Forbidden' response. In other words, other users are blocked from monitoring your extension.
Values:	on, off
Default:	on

Setting:	user_auth_tag
Description:	When the setting is set to AES-32 (default), the phone offers a 32-bit auth-tag for SRTP. Selecting AES-80 makes the phone offer an 80-bit auth-tag.
Values:	on, off
Default:	on

Setting:	user_auto_connect
Description:	If it is <on>, the phone will automatically answer incoming calls.
Values:	on, off
Default:	off

Setting:	user_check_cseq_dlginfo_notify
Description:	So as to prevent an incorrect LED status, a NOTIFY transporting a dialog-info event will be proceesed only if is not stale (i.e. the CSeq number in the NOTIFY request is greater than the last one received). Disable this setting if you want the CSeq counter to be ignored.
Values:	on, off
Default:	on

Setting: user_custom

Description: Specify an URL to your own ringing melody. The type of file that should be supplied to the phone is: "PCM 8 kHz 16 bit/sample (linear) mono WAV. This only has an effect when you have chosen Custom Melody from the Ringtone pull-down menu and when the incoming call matches this SIP identity.

Values: URL

Default: blank

Setting: user_default_blf_direction

Description: RFC4235 the direction attribute of the XML dialog info is optional. With this setting you can define the default value.

Values: initiator, recipient, blank

Default: blank

Setting: user_default_contact_uri

Description: This setting is used by server directories such as Metaswitch, LDAP, Broadsoft XSI and Broadsoft Xmpp Contacts to control the behavior when user presses OK on a contact:

- If set to "none" (default), bring up the Contact Details screen of the contact.
- If set to "main", directly dial the number that is considered the main one of the contact.

Values: none, main

Default: none

Setting: user_descr_contact

Description: When your SIP Registrar is not properly supporting long contacts specified in accordance with RFC 3840, you may want to switch this behavior off.

Values: on, off

Default: on

Setting: user_dp

Description: Here, one can specify the regular expression dialplan for a particular SIP identity. **This setting has been replaced by 'user_dp_str' !**

Values: dial plan string

Default: blank

Setting: user_dp_exp

Description: ENUM means that a conventional E.164 number (normal phone number) is mapped to a SIP URI so that a pure IP call can be started instead of an IP/PSTN call. To use ENUM lookup not only this option has to be enabled, but also below options Countrycode and Areacode have to be setup properly before. Both options are used to build the above Dial Plan String which is mandatory to make the ENUM lookup work.

Values: ENUM lookup string

Default: blank

Setting: user_dp_str

Description: You can set up the dial plan for this line here. With a dial plan, you can match user input (digits via keyboard) to specific actions like dialing, using a distinct outgoing identity, etc.

Values: reg ex string

Default: blank

Setting: user_dtmf_info

Description: Some IVR systems may need DTMF events signalled via SIP INFO messages, this can be enabled here. Set it to <on> or <sip_info_only> to provide DTMF codes via SIP INFO messages.

With <sip_info_only>, the in band and out of band DTMF codes stop going in RTP as they are sent only through SIP INFO messages.

<sip_info_only> sends DTMF codes via SIP INFO messages only.

<on> additionally sends DTMF via RTP!

Values: sip_info_only, on, off

Default: off

Setting: user_dynamic_payload

Description: **This setting is obsolete.**

Previously turned on dynamic payload type for G726.

Values: on, off

Default: on

Setting: user_enable_hookflash

Description: This setting enables support for the hookflash feature on Broadsoft's Broadworks servers. When enabled the phone will process incoming INFO messages with a content type of 'application/broadsoft' for call waiting indication. Additionally, when the line key is pressed in the connected state, a hookflash event is sent to the server inside an INFO message. This occurs in lieu of the hold action which is usually invoked when this feature is disabled.

Values: on, off

Default: off

Setting: user_event_list_uri

Description: The subscription URI for monitoring the dialog states of a number of extensions setup at the PBX. This setting turns on the mechanism) cause the phone to send a single subscription even for monitoring multiple extensions. The associated NOTIFY contains the extensions configured at the server for the user and their respective status if it active.

When filling this setting with a simple sip-uri or number in the WUI, it will automatically be replaced by a complex XML-configuration that allows to auto-assign the received buddies onto keys of type Contact List Buddy.

Values: URI or XML sub trees

Default: blank

Setting: user_expiry

Description: The proposed expiry time of the registration in seconds for line x. Upon expiration of the registration, the phone will send a fresh re-registration request.

Values: Integer

Default: 3600

Setting: user_failover_identity

Description: This identity will be used as a backup for failover i.e. if the current identity is not registered, this identity is used instead.

Values: none, 1, 2, 3, 4

Default: none

Setting: user_full_sdp_answer

Description: When the setting is turned 'on', the phone returns a list of all available codecs in the SDP in response to INVITE requests. Otherwise the first codec of the calling party that matches the configured codecs on the phone is returned.

Values: on, off

Default: on

Setting: user_g726_packing_order

Description: There are two types of byte order for G.726, namely RFC3551 and AAL2. With this setting you can choose the byte order in order to use the same order as the remote entity. Note: this setting has no effect on codec: AAL2-G726-32 !

Values: on, off

Default: on

Setting: user_hash

Description: This is the password to be used for challenge responses. In order to protect them from unauthorized use, passwords are not displayed in their true form, but as a series of asterisks.

Values: String

Default: blank

Setting:	user_hold_inactive
Description:	Specify if you want to indicate an hold request with sdp parameter sendonly or inactive. Some PBX's need the inactive setting for proper music on hold operation.
Values:	on, off
Default:	off

Setting:	user_host
Description:	Specify the IP or DNS address of the registrar/proxy where you want to register this account. After a successful registration, the registrar knows how to reach this specific identity and can route requests (e.g., incoming calls) from other registered parties to this phone.
Values:	host string
Default:	blank

Setting:	user_ice
Description:	Choose whether or not you want to use Interactive Connectivity Establishment (ICE). ICE optimizes the media path. This would be the case, for example, when two phones in the same network are calling each other via a long media path through other, external networks. With ICE, the short media path in the same network would be chosen, which will presumably have better quality than the long one. Sometimes this feature will stop you from being able to make calls. When this occurs, switch it off. Note, that ICE currently will work reliable in OCS environment only.
Values:	on, off
Default:	off

Setting:	user_idle_number
Description:	This setting only works with the new color UI. If you enter a name or number in this field, the entered value replaces the account number / identity shown in the subtext of the idle screen for this particular identity. This information is not sent out to anyone, but is merely shown on the phone’s display for your information.
Values:	String e.g. 123, provider-abc, my extension: 123, Company A, +49 30 398 33 123

Default:	blank

Setting:	user_idle_text
Description:	If you enter a name in this field, then this name will be shown on the idle screen associated with this particular line instead of the name you have entered in the Displayname field, if any. This information is not sent out to anyone, but is merely shown on the phone's display for your information.
Values:	String
Default:	blank

Setting:	user_keys_to_be_configured_on_first_registration
Description:	The keys listed here get automatically distributed over all free keys whenever the associated identity registers for the first time. Free keys in this context are keys of type none or line without an specific identity context (i.e. == active). See also setting automatic_key_configuration_targets
Values:	space separated list of key types
Default:	blank

Setting:	user_mailbox
Description:	If you have set up a mailbox, specify the account name for that mailbox here to associate it with this particular SIP identity. This is important for contacting your mailbox when the MWI message does not include the proper mailbox SIP URI.
Values:	String
Default:	blank

Setting:	user_media_setup_offer

Description: The chosen value has only affect if setting user_media_transport_offer has been set to TCP. It defines according to RFC4145 the local role on an SDP offer.

active: local party is connecting to remote party (a=setup: active)

passive: remote party is connecting to local party (a=setup: passive)

any: remote party shall decide who is connecting (a=setup: actpass)

Values: active, passive, any

Default: active

Setting: user_media_transport_offer

Description: Select the type of the rtp media transport. In mostly every case you should be fine with the default "udp". However, RTP via TCP is also available according to RFC4145.

If you choose "tcp", please pay also attention to setting user_media_setup_offer.

Values: udp, tcp

Default: udp

Setting: user_moh

Description: If you specify a SIP URI pointing to a media server account, the phone will, when a call is put on hold, invite this SIP URI to call the held phone to play music on hold.

Values: SIP address

Default: blank

Setting: user_name

Description: This is the account with which you register to a SIP registrar/proxy. It could be alphanumeric, e.g. js, or based on digits like 445. See also setting user_pname.

Values: String

Default: blank

Setting:	user_no_auto_logoff
Description:	Identity survives the auto logoff timer. This can be used e.g. for emergency lines.
Values:	on, off
Default:	off

Setting:	user_outbound
Description:	Specify the outbound proxy in this field (format: addr:port) to ensure all SIP packets are sent via the specified communication point.
Values:	Address:Port
Default:	blank

Setting:	user_pass
Description:	This is the password to be used for challenge responses. In order to protect them from unauthorized use, passwords are not displayed in their true form, but as a series of asterisks.
Values:	String
Default:	blank

Setting:	user_phone
Description:	This is the password to be used for challenge responses. In order to protect them from unauthorized use, passwords are not displayed in their true form, but as a series of asterisks.
Values:	on, off
Default:	on

Setting:	user_pic
Description:	Specify an URL to a small JPEG picture. When the flash plugin feature (Preferences page) is enabled, this picture will be shown on the Home web page during a call.
Values:	URL
Default:	blank

Setting:	user_pic_tie_to_tbook
Description:	When this setting is on, the setting 'user_pic' is handled automatically so it always points to the photo from the directory that describes the identity
Values:	on, off
Default:	off

Setting:	user_pname
Description:	Registrar environments may need different user names for registration and authentication. If user_pname is set, it is used for authentication and setting user_name is used for registration; otherwise setting user_name is used for both.
Values:	String
Default:	blank

Setting:	user_presence_buddy_list_uri
Description:	The URI phone will subscribe for this identity's contact list.
Values:	SIP URI
Default:	blank

Setting:	user_presence_host
Description:	The address to which the phone sends its Presence updates (using web service requests). This setting is only used if setting user_server_type is Telepo
Values:	URL
Default:	blank

Setting:	user_presence_identity
Description:	Indicates from which idenity the OCS presence status which is displayed in the idle screen is to be taken. This is useful for installations where voice and OCS presence are signalled over a separate identities using different servers.

Values:	none, 1 - 12
Default:	none

Setting:	user_presence_subscription
Description:	When this feature is set to on, the phone subscribes for the presence status of its contacts.
Values:	on, off
Default:	off

Setting:	user_presence_uri
Description:	The address to which the SUBSCRIBE for Buddylist is sent
Values:	URI
Default:	blank

Setting:	user_proxy_require
Description:	If your SIP proxy/registrar needs the 'SIP Proxy Require' header, it can be enabled here.
Values:	Proxy-Require header
Default:	blank

Setting:	user_pui_treats_uri_username_as_fallback_for

Description: The Number display style setting (display_method) specifies how incoming and outgoing calls are displayed, for example with the name and/or phone number of the calling party. But sometimes this information is not available. For these cases, this setting makes it possible to display the username of the SIP URI instead.

Using the username as fallback for a name: Set this setting to name. When, for example, there is no name information available for an incoming call with URI "John.Doe@pbx.com", the display would show "John.Doe" instead.

Using the username as fallback for a phone number: Please note that SIP URIs like "4711@pbx.com" will automatically detect "4711" as the number. Setting this setting to number is only needed for cases where you'd want to display "a101" of "a101@pbx.com" as the number string.

Leave this setting empty if you do not want to use the username as fallback.

Values: name, number, empty

Default: number

Setting: user_q

Description: You can set up the probability of a registration for each line through this setting (the default is 1.0). This means that different registrations with different Q-values will ring in serial order (serial forking) in contrast to different registrations with the same Q-values, which will ring in parallel (parallel forking).

Values: Values between <0.0> and <1.0>

Default: 1.0

Setting: user_realname

Description: Set the name you would like to associate with each line, e.g. John Smith. This information is also sent out to any party you are calling. Only the first 50 characters are used (when entering more than 50 characters).

Values: String

Default: blank

Setting: user_remove_all_bindings

Description: When enabled the phone sets the contact header to * in order to remove the old contact at the registrar on each DeREGISTER. A DeREGISTER will be done on each ReREGISTER as well.

Values: on, off

Default: off

Setting: user_replaces_when_referring_to_conference_server

Description: Switches whether or not to add the replaces-query to the refer-to-uri when refering calls to the conference server.

Related Setting (also controls content of refer-to): refer brackets

Values: on, off

Default: on

Setting: user_report_machine_state

Description: This is an OCS specific setting. When on, the phone will publish its machine state to the OCS server as well as its device capabilities. The machine state is initially 'available'. If the settings inactive_timeout and away_timeout are set, it will eventually move to 'inactive' and then to 'away'. Note that if you set your phone to not report the machine state it cannot not be part of a response group (since the phone will never become available and therefore no calls will be routed to it).

Values: on, off

Default: on

Setting: user_report_phone_state

Description: This is an OCS specific setting. When on, the phone will publish its phone state to the OCS server. This is published in addition to the machine state (if this is enabled, see setting user_report_machine_state) when the user goes off-hook. The phone state alwas has an availability of 'busy' and an activity of 'in-a-call'. When the user goes back on-hook the phone state is deleted. The phone state will be visible to others only if at least one device on which the user is logged on also reports the machine state. If you want the phone state to be visible only while you are also logged on to Communicator, then set user_report_machine_state to off. When you then log out of communicator and make a call on the phone, others will see your state as 'offline'.

Values:	on, off
Default:	on

Setting:	user_ringer
Description:	Select a ring tone from this pull-down menu that will alert you when a call comes in for this particular identity.
Values:	Ringer1, Ringer2, Ringer3 , Ringer4 , Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, Silent ,Custom
Default:	Ringer1

Setting:	user_ringer_forwarded_calls
Description:	This setting applies only to the UC edition. Select from this pull-down menu which ring tone to use to alert you that the incoming call was originally intended for another target. Retargeting may occur as a result of call fowarding, delegation, team call, and Automatic Call Distribution (Response Groups).
Values:	Ringer1, Ringer2, Ringer3 , Ringer4 , Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, Silent ,Custom
Default:	Ringer3

Setting:	user_ringer_private_line
Description:	This setting applies only to the UC edition. Select from this pull-down menu which ring tone to use to alert you to a call coming in on your private line.
Values:	Ringer1, Ringer2, Ringer3 , Ringer4 , Ringer5, Ringer6, Ringer7, Ringer8, Ringer9, Ringer10, Silent ,Custom
Default:	Ringer2

Setting:	user_savp

Description:	This setting is effective only when RTP encryption (SRTP) is also enabled and is used to specify whether the use of the RTP/SAVP profile by the phone should be off (for backward compatibility), optional or mandatory. When this setting is set to "mandatory" the phone will offer and accept only SDPs that contain m= lines with an audio profile of RTP/SAVP. When this setting is set to "optional", the phone will offer SDPs containing two m= lines, one with an audio profile of RTP/SAVP the other with an audio profile of RTP/AVP and it will accept SDPs containing m= lines with either profile. The RTP/SAVP profile, being the preferred one, is listed first. Since some SIP proxies cannot handle RTP/SAVP profiles or multiple m= lines this setting may also be turned off. In this case the phone will send SDPs containing RTP/AVP audio profiles only. Whether or not the crypto attribute is included depends on whether RTP encryption is on or off. Note: When RTP encryption is turned off this setting has no effect.</p>
Values:	off, optional, mandatory
Default:	off

Setting:	user_sdp_version_check
Description:	Usually each received sdp-packet has a version number that identifies it. When receiving the same version again the phone can ignore it. However this versioning mechanism does not work reliably with all PBX'es so we introduced the option to keep the phone from checking the version. When version check is off, the phone will compare the entire sdp instead (except for the version). When setting user_server_type to nortel, ocs or broadsoft -> version-check will be disabled automatically.
Values:	on, off
Default:	on

Setting:	user_send_local_name
Description:	When this option is enabled, the phone receiving a SIP INVITE message adds the display name of the called identity to the reply message in order to allow the calling party to show this information on its display.
Values:	on, off
Default:	off

Setting:	user_server_type
Description:	To enable PBX specific interoperability features you may specify the proper server type matching your PBX environment.
Values:	Default , Asterisk, Bria (custom solution for Telekom Austria), Broadsoft, CCM, MetaSwitch, Nortel, PBXnSIP, snomONE, Sutus BC, Sylantro, Telepo, Teles
Default:	Default

Setting:	user_shared_line
Description:	If you have to share your extension (identity) with somebody else, this has to be enabled.
Values:	on, off
Default:	off

Setting:	user_sipusername_as_line
Description:	If your VoIP provider works only when you turn on Support broken registrar on the phone's web interface, this means your provider does not call your phone the way the phone requested to be called. What happens is that incoming INVITEs from your VoIP provider do not contain the contact URI which was previously registered by your phone as its contact. Thus the phone cannot safely identify the target line of the incoming call. When you compare the URI in the first line of the incoming INVITE and the URI in the Contact of the REGISTER, which the phone sends to the registrar of your provider, they will presumably differ. This is what we mean by broken registrar. It is as though your provider has sent a letter to an apartment building with the city, the street address, and the house number on it, but without the recipient's name. When you turn on Support broken registrar, the phone tries to find the right apartment by guessing, but this guessing will fail when there are two parties with the same name in the building.
Values:	on, off
Default:	off

Setting:	user_srtp

Description:	Your phone supports RTP encryption via SRTP. If you want to encrypt your outgoing audio (RTP) stream, this option must be on. Both parties have to enable the RTP Encryption option to establish an SRTP call. RTP encryption has nothing to do with SSL/TLS. The keys are sent in the SDP part of SIP messages. Certificates are not used for this. The default value is "on". In order to obtain full security SIP call you have to use TLS as well. Then, a small lock sign is shown on the display which means that an secure SIP call is currently taking place (SIP secured + RTP encrypted).
Values:	on, off
Default:	on

Setting:	user_stream
Description:	This setting is obsolete. Please use setting user_moh instead.
Values:	
Default:	

Setting:	user_subscription_expiry
Description:	This value specifies the desired expiration time in seconds for subscriptions to the following event packages: dialog (individual and event list subscription) call-info message-sumary presence The subscription will be refreshed after a time randomly chosen to be between 1/2 and 3/4 of the expiration time (which the server may have reduced in the 200 OK response). NOTE Setting this value to zero will cause the subscription to become inactive. The line-seize event package subscription is not affected by this value. It is fixed to15 seconds.
Values:	0 - 1209600
Default:	3600

Setting:	user_symmetrical_rtp
Description:	This setting tells the phone to always send RTP packets to the same IP and port from where it receives them. It ignores the port which the remote party sent in the SDP details. If the two incoming and outgoing RTP (audio) streams of a single call should use the same port number, turn this setting "on".
Values:	on, off
Default:	off

Setting:	user_tel_nr
Description:	This setting assigns a telephone-number to an identity. This feature is currently used for one CSTA-service only: The sip-urise in our answer to GetSwitchingFunctionDevices will be enhanced by the tel-parameter, when a phone-number is configured. E.g.: sip:foo@gar.com;tel=4711
Values:	phone number
Default:	blank

Setting:	user_tlsdsk_store
Description:	This setting applies only to the UC edition and is for the phone's internal use only to persistently store data required for TLS-DSK authentication. The setting is cleared when the "Logoff User" function is invoked.
Values:	String
Default:	blank

Setting:	user_uid
Description:	The user_uid value is generated and stored in the setting on a fresh phone when an account is setup. If you reboot the phone afterwards it will use the same uuid value as the one generated/stored in the settings. Naturally if you reset the phone this setting will also be erased and the next account setup will generate a new uuid. If you provision the user_uid setting the phone will use that value instead of generating a new one on its own. The uuid is used in the contact header of SIP REGISTER messages.
Values:	a sequence of randomly generated bytes according RFC 4122
Default:	blank

Setting: user_wait_for_ntp_before_register

Description: In some environments it is essential for the registration process, that the phone has the correct time. When this setting is turned on, the phone will wait for the reception of the time from the ntp server before trying to register the associated identity.

Values: on, off

Default: off

Setting: user_was_registered

Description: **SYSTEM INTERNAL**

Flag showing whether identity was ever registered since last identity reset.

This is the identity-based version of setting was_never_registered.

Values: true, false

Default: false

Setting: user_xml_screen_url

Description: The HTTP URL pointing to a XML idle screen description is used to design your own idle screen. Per identity a different XML idle screen can be specified and will be shown if this identity is the current active outgoing one.

Values: Any HTTP URL pointing to a valid XML idle screen description.

Default: blank

Setting: using_server_managed_dnd

Description: If this setting is "on" the server will be responsible for handling the DND(DO NOT DISTURB) functionality. From the call perspective the phone will act as if no dnd was set (all is managed by the server).

The phone user will see the value from dnd_mode as the current DND state, and this value can be changed at anytime by the server.

This setting does not specify how the server changes the value of setting dnd_mode nor how the phone updates them (it may be done via TR69).

Values: on, off

Default: off

Setting: using_server_managed_fwd_all

Description: If this setting is "on" the server will be responsible for handling the global forwarding functionality. From the call perspective the phone will act as if no forwarding was set (all is managed by the server).

The PUI will show the current setup if the server synchronizes it's values with those on the phone (settings fwd_all_enabled and fwd_all_target). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes).

Values: on, off

Default: off

Setting: using_server_managed_fwd_busy

Description: If this setting is "on" the server will be responsible for handling the redirect on busy functionality. From the call perspective the phone will act as if no redirect was set (all is managed by the server).

The PUI will show the current setup if the server synchronizes it's values with those on the phone (settings fwd_all_enabled and fwd_all_target). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes).

Values: on, off

Default: off

Setting: using_server_managed_fwd_time

Description: If this setting is "on" the server will be responsible for handling the redirect on timeout functionality. From the call perspective the phone will act as if no redirect was set (all is managed by the server).

The PUI will show the current setup if the server synchronizes it's values with those on the phone (settings fwd_all_enabled, fwd_all_target and fwd_time_secs). This setting does not specify how the server changes the redirection values nor how the phone updates them. (it may be done via TR69, DFKS or starcodes).

Values: on, off

Default: off

Setting: utc_offset

Description: **SYSTEM INTERNAL**

Signed UTC offset in seconds. This value is retrieved automatically from the timezone configuration. Usually there will be no need to change this setting.

Values: Integer

Default: blank

Setting: vip_ring_sound

Description: Phone book contact type specific ringers. Selection of the ring tone style that signals incoming calls dependent on the contact type of the caller in the local phone book.

Values: Ringer1, Ringer2, Ringer3, Ringer4, Ringer5, Ringer6, Ringer7, Silent, Custom

Default: Ringer1

Setting: vlan_id

Description: This setting has to be set properly before the phone is able to connect to anything residing in a specific VLAN ! Only the phone is residing in the specified VLAN, it has no effect to e.g. a PC connected with the second ethernet socket (PC).

The VLAN tagging is done by the kernel (as opposed to setting vlan_net_id, which activates tagging by the phone's integrated switch).

Values: 1-4095

Default: blank

Setting: vlan_pc_id

Description: Any incoming packet on the PC port is tagged with this VLAN ID.

Values: 1-4095

Default: blank

Setting: vlan_pc_priority

Description: This is the priority of the VLAN.

Values: 0-7

Default: blank

Setting: vlan_port_tagging

Description: VTech ET6xx phones have an internal ethernet-switch capable of handling vlan (set tags and unset them)

This setting defines whether the switch will handle the vlan tagging or not.

Handling means that pakets from the internal ports to the network are tagged (vlan id is added) and tagged pakets (vlan set) from the network are untagged (vlan id is removed) and assigned to the port they belong (selection by vlan id).

Example: Pc-port is configured vlan 3 and the option is set to on, pakets arriving from the pc on the pc-port are tagged with vlan 3 and sent to the network.

Pakets arriving from the network containing vlan id 3 will be assigned/send to pc-port, but before that the vlan id (3) is removed. So the pc will receive a paket without vlan id.

Network --- VLAN ID 3 --- phone with int. switch ---- No Tag ---- PC

On: Phone-internal switch handels the vlan-pakets.

To Network direction -> vlan ids are set, From Network -> vlan id are unset

Off: phone internal switch does not touch the pakets.

Independend of vlan id set or not, pakets are not changed, connected device has to take care.

Values: on, off

Default: off

Setting: vlan_qos

Description:	Priority (802.1p) has to be set properly before the phone is able to connect to anything residing in a specific VLAN ! Only the phone is residing in the specified VLAN, it has no effect to e.g. a PC connected with the second ethernet socket (PC). The VLAN tagging is made by the kernel (as opposed to setting vlan_net_priority, which sets tagging made by the phone's incorporated swich)
Values:	0-7
Default:	blank

Setting:	vol_handset
Description:	Selection of the handset speaker volume
Values:	0-15
Default:	13

Setting:	vol_headset
Description:	Selection of the headset speaker volume.
Values:	0-15
Default:	10

Setting:	vol_ringer
Description:	Determines the volume of the ringer.
Values:	1-15
Default:	10

Setting:	vol_speaker
Description:	Selection of the casing speaker volume.
Values:	1-15
Default:	8

Setting:	vq_local_group
Description:	The value of this setting will be used as value of "Local Group" in any voice quality report to the voice quality report collector.'
Values:	String
Default:	blank

Setting:	vq_report_collector
Description:	Specifies the collector to which a voice quality and registration reports are send to. The form of the report is specified by the setting rtcp_xr. For optional route headers on the notify request you might specify them with comma separated syntax and with a valid sip url.
Values:	sip:vqr.voip.intern:5099
Default:	blank

Setting:	was_never_registered
Description:	**SYSTEM INTERNAL** Traces whether somebody ever was registered at the phone since last factory reset.
Values:	true, false
Default:	true

Setting:	watchdog
Description:	The watchdog will watch your phone, if the phone will freeze, the watchdog initiates a hard reboot of the phone. This watchdog is based on the linux software watchdog.
Values:	on, off
Default:	on

Setting:	web_language
Description:	Your phone is able to show all web GUI texts in a number of different languages. Select the language of your choice which may be different from the one currently used on the phone.

Values: Language Code

Default: English

Setting: web_logout_timer

Description: Specify the time in minutes after which the web interface shall ask you to login again.

Values: Integer

Default: blank

Setting: webserver_cert

Description: With this setting, one can upload its own signed web server certificate for TLS secured HTTP communication (->HTTPS).

Web browsers using HTTPS to access the phone

s web interface will request this certificate from the phone´s HTTP server

Values: base 64 encoded certificate along with the private key

Default: blank

Setting: webserver_max_data_size

Description: The maximum size of HTTP POST requests accepted by the internal webserver. For requests which exceed the limit an error code 413 will be returned by the server.

The maximum value can be changed but will use the current memory of the phone. If e.g. an upload of an address book is done, please make sure you split it into smaller uploads instead of increasing the maximum value.

Values: Integer

Default: 524288

Setting: webserver_type

Description: Set up the type of connection the phone's web server is willing to answer to. Please be advised that you will no longer be able to use the web user interface of the phone when you select off! Press the menu key, use the navigation key to go to the submenu Webinterface, and select Server. Then change the type of connection to one of the other types.
Note: activation of changes requires a reboot.

Values: http, https, http_https, off

Default: http_https

Setting: with_flash

Description: If you want to have a live reaction on incoming or outgoing calls on the phone's Home page, switch this option to on. Your web browser has to support the Macromedia flash movie format.

Values: on, off

Default: off

Setting: work_ring_sound

Description: Phone book contact type specific ringers. Selection of the ring tone style that signals incoming calls for contacts of type 'Work' in the local phone book.

Values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>, <Silent>, <Custom>

Default: Ringer1

Setting: wui_admin_only

Description: List the WUI-pages that are not accessible in user-mode.

Values: List of WUI-pages (like e.g. log.htm) separated by space. Pages may include a query like line_login.htm?l=1.

Default:	screen.bmp settings.cfg settings.xml settings_wo_default.xml tbook.xml tbook.csv param_map param_map_structs state_of_gui.htm state_of_identity.htm dirty_hosts.htm dialplan.xml trace.pcap dummy.htm strings.csv log.htm certificates_unknown_certs.htm subscriptions.htm trace.htm http_trace.htm memstat.htm support.htm line_login.htm action.htm pcap.htm dnscache.htm update.htm settings.htm line_sip.htm line_nat.htm line_rtp.htm line_features.htm changed_settings.htm contacts.htm debug.htm modules.htm media.htm xml_entities.htm state_of_edit.htm

Setting:	xcap_dir_doc_name
Description:	Document name used to construct the xcap contact-list-url. Only used when setting 'user_server_type' is set to bria.
Values:	Document name
Default:	contacts-resource-list.xml

Setting:	xcap_directory_auid
Description:	Directory used to construct the xcap contact-list-url. Only used when setting 'user_server_type' is set to bria.
Values:	String
Default:	services/resource-lists

Setting:	xcap_server_name
Description:	Server name used to construct the xcap contact-list-url. Only used when setting 'user_server_type' is set to bria.
Values:	String
Default:	blank

Setting:	xcap_server_port
Description:	Port number used to construct the xcap contact-list-url. Only used when setting 'user_server_type' is set to bria.
Values:	valid port
Default:	8080

Setting: xcap_tbook_sync_interval

Description: This setting defines the number of seconds after which a syncronization between the XCAP server and internal directory must be done, even when there is no indication for change (usually a SIP message informs us of changes on server side).

Values: integer

Default: 7200

Setting: xcap_via_tls

Description: Define whether to connect to the XCAP server using http or https.

Values: on, off

Default: on

Setting: xfer_dest_order_lifo

Description: Determines in which order held calls are presented to the user as destination during an attended transfer. When 'on' the most recent call on hold is presented first; when 'off' the oldest one is presented first.

Values: on, off

Default: off

Setting: xml_notify

Description: Enables/Disables xml notifies (type: application/ciscoxml OR application/vtechxml)

Values: on, off

Default: on

Setting: xsi_anywhere

Description: Determines whether the phone should enable XSI Anywhere feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks Anywhere settings.

Values: on, off

Default:	on

Setting:	xsi_auth_pass
Description:	The password of the Broadsoft XSI account.
Values:	String
Default:	blank

Setting:	xsi_auth_user
Description:	The Broadsoft XSI account name.
Values:	String
Default:	blank

Setting:	xsi_callcenter_list
Description:	Determines whether the phone should enable XSI Call Center List feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks Call Center List settings.
Values:	on, off
Default:	on

Setting:	xsi_caller_id_blocking
Description:	If set to "on", outgoing caller ID blocking will be managed on Broadsoft server side through the use XSI. If set to "off", outgoing caller ID blocking will be managed locally.
Values:	on, off
Default:	off

Setting:	xsi_conf_timer
Description:	Controls how often the device polls the Broadsoft server for conference updates when idle.
Values:	time in seconds

Default: 30

Setting: xsi_directory_fullsearch

Description: Determines whether the phone should perform a user's name search on both first and last name simultaneously. For more information on XSI search criteria see Broadsoft's BroadWorks Interface Specification Xtended Services Interface.

Values: on, off

Default: off

Setting: xsi_events

Description: Determines whether the phone should establish XSI event channels. Does not affect XSI Actions. For more information on XSI actions and events see Broadsoft's BroadWorks Interface Specification Xtended Services Interface.

Values: on, off

Default: off

Setting: xsi_polling_interval

Description: Specifies the interval in seconds after which XSI action will be sent to retrieve related information from server.

Values: Integer value >= 0; while there is no explicit maximum value, intervals are limited to two weeks.

Default: 60

Setting: xsi_protocol_version

Description: Determines the XSI Interface version.

Values: Valid XSI Interface version number, like 22.0, 19.0

n/a means the latest XSI Interface.

Default: blank

Setting: xsi_remote_office

Description: Determines whether the phone should enable XSI remote office feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's BroadWorks remote office settings.

Values: on, off

Default: on

Setting: xsi_retry_timer

Description: If an error occurs during XSI session set up, this setting specifies after how many seconds the phone should retry setting up the XSI session (A value of zero means never).

Values: positive integer

Default: 300

Setting: xsi_server

Description: Specifies the Broadsoft XSI server.

Values: String

Default: blank

Setting: xsi_silent_alert

Description: Determines whether the phone should enable the Silent Alerting feature.

Values: on, off

Default: on

Setting: xsi_simultaneous_ring

Description: Determines whether the phone should enable XSI Simultaneous Ringing feature. When set to 'on', the phone UI will allow user to get/set the Broadsoft's Simultaneous Ringing settings.

Values: on, off

Default: on

Setting: xsi_unknown_call_list_name_text

Description: If the remote name in the call list entry is matching the value of this setting, then this name will be replaced by the remote number of the call list entry.

Values: Character string

Default: Unavailable

Setting: xsi_visual_voicemail

Description: This setting is used to enable / disable visual voicemail feature.

Values: on, off

Default: on

Setting: xsi_visual_voicemail_dial_offhook

Description: This setting is used to influence behaviour on offhook.

If user goes offhook while presenting visual voicemail:

- on = dial number of caller
- off = listen to voicemail

Values: on, off

Default: on

CHAPTER 6

TROUBLESHOOTING

If you have difficulty with your ET605 Deskset, please try the suggestions below.

NOTE: For customer service or product information, contact the person who installed your system. If your installer is unavailable, visit our website at *businessphones.vtech.com* or call **1 (888) 370-2006**.

Common Troubleshooting Procedures

Follow these procedures to resolve common issues. For more troubleshooting information, see the user‘s manual for your product.

Screen is blank.

- Ensure power is connected. If powered by an AC adapter, check that the adapter is plugged into a wall socket and the ET605 power jack. If powered by PoE, ensure that the network switch is providing power through the correct ports.

My computer can't connect to the network after plugging the Ethernet cable through the PC port.

- Make sure the ET605 is connected to power. The PC port does not work when the ET605 does not have power source or during a power outage.
- Make sure you plug the Ethernet cable connected to the router into the ET605 Ethernet port and the Ethernet cable connected to the computer into the ET605 PC port.

The firmware upgrade or configuration update isn't working.

- Before using the WebUI, ensure you have the latest version of your web browser installed. Some menus and controls in older browsers may operate differently than described in this manual.
- Ensure you have specified the correct path to the firmware and configuration files on the WebUI: **Software Update** page and the **Advanced** > **Update** page.

Provisioning: Use DHCP Option is enabled, but the ET605 is not getting a provisioning URL from the DHCP Server.

- Ensure that **DHCP** is set to “on” in the WebUI: **Advanced** > **Network** .

Pages are not received.

- The **Intercom Policy** setting is set to “off”. Check this setting in the WebUI: **Advanced** > **Behavior**.

APPENDIXES

Appendix A: Maintenance

Taking care of your telephone

- Your ET605 Deskset contains sophisticated electronic parts, so you must treat it with care.
- Avoid rough treatment.
- Place the corded handset down gently.
- Save the original packing materials to protect your ET605 Deskset if you ever need to ship it.

Avoid water

- You can damage your ET605 Deskset if it gets wet. Do not use the corded handset in the rain, or handle it with wet hands. Do not install the ET605 Deskset near a sink, bathtub or shower.

Electrical storms

- Electrical storms can sometimes cause power surges harmful to electronic equipment. For your own safety, take caution when using electric appliances during storms.

Cleaning your telephone

- Your ET605 Deskset has a durable plastic casing that should retain its luster for many years. Clean it only with a soft cloth slightly dampened with water or a mild soap.
- Do not use excess water or cleaning solvents of any kind.

Remember that electrical appliances can cause serious injury if used when you are wet or standing in water. If the ET605 Deskset should fall into water, DO NOT RETRIEVE IT UNTIL YOU UNPLUG THE POWER CORD AND NETWORK CABLE FROM THE WALL, then pull the unit out by the unplugged cords.