



H2/H2P SIP Phones Deployment Guide with Open SIP PBX Vendors



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1. Introduction

This document describes how to deploy the ALE Halo series DeskPhone sets on the open SIP PBX. ALE Halo series will include two models: H2 and H2P.

The following sets are covered: ALE H2/H2P DeskPhone



ALE H2/H2P SIP phones deployment guide provides general guidance on setting up phone network, provisioning and managing phones.

This guide is not intended for end users, but for administrator with experience in networking who understand the basis of open SIP networks and VoIP endpoint environments.

As an administrator, you can do the following with this guide:

- Set up a VoIP network and provisioning server.
- Provision the phones with features and settings.
- Troubleshoot, upgrade and maintain phones.

Glossary

DHCP	Dynamic Host Configuration Protocol
DM	Device Management = provisioning server
EDS	Easy Deployment Server
FQDN	Fully Qualified Domain Name
HTTP/HTTPS	Hypertext Transfer Protocol/Hypertext Transfer Protocol Secure
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
MMI	Man Machine Interface
PoE	Power over Ethernet
RAM	Random Access Memory
SIP	Session Initiation Protocol
SSH	Secure Shell
URL	Uniform Resource Locator
USB	Universal Serial Bus
VCI	Vendor Class Identifier
WBM	Web Based Management
WAN	Wide Area Network

2. Phone set provisioning overview

2.1 Provisioning method priority

This chapter gives general indication on the parameters that must be provisioned to start a set, the different ways to provision these parameters, and the priority rules between them.

Basically, parameters that must be provisioned are:

- IP parameters (IP address, netmask and router IP address)
- SIP account parameters (SIP server FQDN, domain, authentication, SIP User)
- DM URL, when SIP parameters are provisioned via SIP configuration files downloaded from a provisioning server

The different ways to provision these parameters are:

- IP parameters
 - Statically via MMI
 - Dynamically via DHCP
- SIP account parameters:
 - Manually
Via MMI: see Configuring IP parameters and SIP account parameters via MMI
Via WBM: see Configuring IP parameters and SIP account parameters via WBM
 - Automatically via SIP configuration files downloaded from a provisioning server
- DM URL (DM server-> DM URL (required only in case of initialization with SIP configuration files))
 - Manually:
Via WBM: see configuring the provisioning server URL via WBM
 - Automatically:
Via DHCP: see DHCP configuration for download path of SIP configuration files
Via EDS server: see Setting up auto-provisioning with EDS
Via SIP PnP: see Setting up auto-provisioning with SIP PnP

The method you use depends on how many phones need to be deployed and what features and settings to be configured. We recommend using manual provisioning as your primary provisioning method when just need several phones for testing.

2.2 Configuring IP parameters and SIP account parameters via MMI

The download modes (Provision DM URL) are sorted in descending order of priority as follows: DHCP, SIP PnP, and Static Provisioning Server.

The device supports SIP PnP, DHCP options, Static provision. If all 3 methods are enabled, the priority from high to low is as follows:

[PnP > DHCP > Static Provisioning](#)

2.2.1 Configuring IP parameters via MMI

The setting menu of the MMI can be accessed after the phone is initialized successfully:

- 1) Enter the admin password (the default admin password for the phone out of box is **123456**).
- 2) Press the softkeys **IP param > IP Config > IPv4 settings** to access the IP parameters setting page. This page allows to select the initialization mode (The default DHCP mode is dynamic) and to configure network parameters if static mode is selected.
- 3) Press the softkey “Left/Right” to switch to **Static**.
- 4) Complete the set IP parameters:
 - **IP**: enter the set IP address
 - **Mask**: enter the IP subnet mask
 - **Gateway**: enter the default router IP address
 - Press the **OK** key to save modifications.
 - Press return key to exit the settings menu.

2.2.2 Configuring SIP account parameters via MMI

The setting menu of the MMI can be accessed after the phone is initialized successfully:

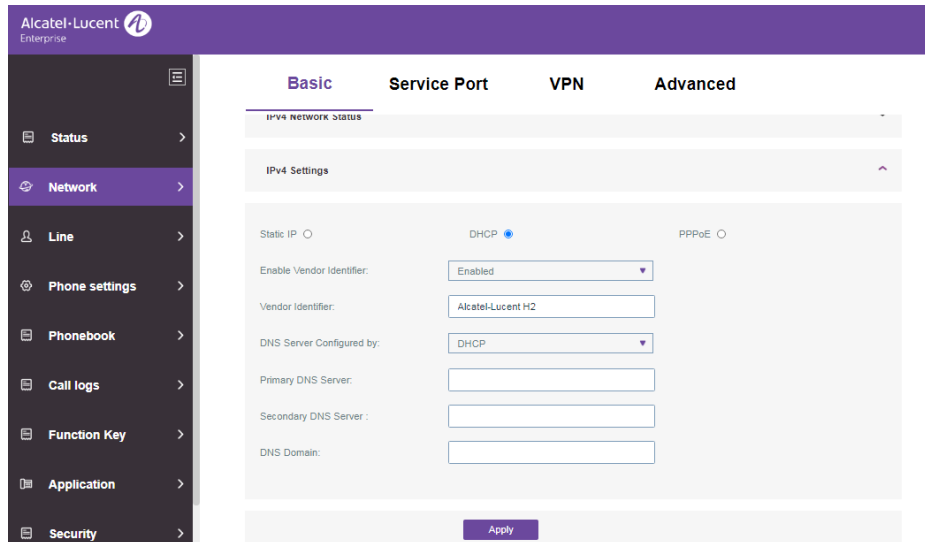
- 1) Enter the admin password (the default admin password for the phone out of box is **123456**).
- 2) Press down key on navigator keypad until the last page and press **SIP Account** softkey.
- 3) Select a SIP account and configure related SIP server connection parameters.
- 4) Press the OK key to save modifications.
- 5) Press return key to exit the settings menu.

2.3 Configuring IP parameters and SIP account parameters via WBM

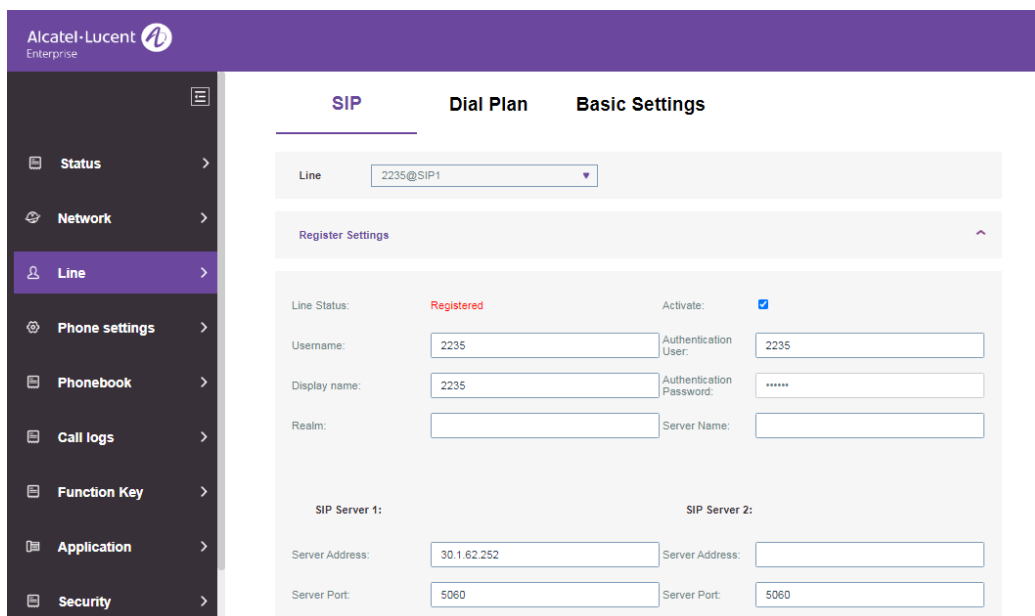
You can configure ALE H2/H2P DeskPhone sets via web user interface (WBM) when the phone has started up with proper IP parameters.

- 1) Press “OK” key to find the IP address of the set
- 2) In a web browser, enter the URL: **https://[IP address]**,
- 3) You are prompted to enter login/password:
- 4) login: enter **admin**
- 5) password: enter the password (default password is **123456**)
- 6) Click **Logon**

Go into Network->IPv4 Settings to configure IP parameters in web:



Go into Line->Register Settings to configure SIP account parameters in web

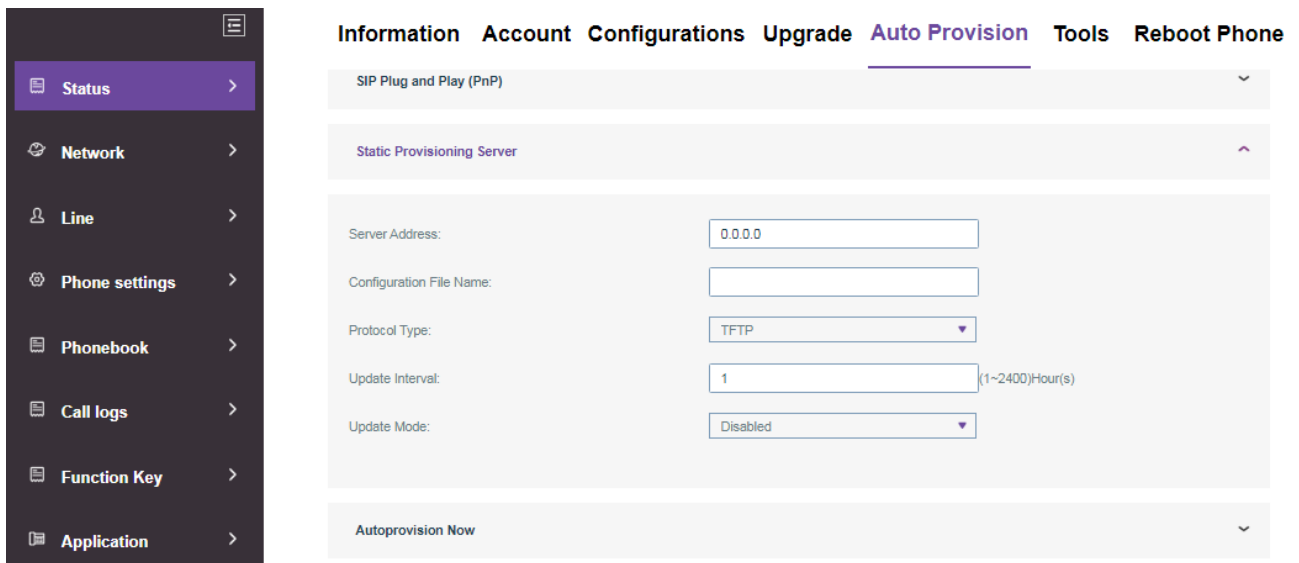


2.4 Configuring the provisioning server URL via WBM

You can configure ALE H2/H2P DeskPhone sets via web user interface (WBM) when the phone has started up with proper IP parameters.

- 1) Press "OK" key to find the IP address of the set
- 2) In a web browser, enter the URL: https://[IP address]
- 3) You are prompted to enter login/password:
- 4) login: enter **admin**
- 5) password: enter the password (default password is 123456)
- 6) Click **Logon**
- 7) Go to **Status > Auto Provision->Static Provisioning Server**
- 8) In the **Server Address** field, enter the complete URL (for example: ftp/tftp/http/https://<provisioning server IP address>/download).

9) In the Configuration File Name field, enter the file name (for example: \$config.mac.xml)



- 1) Press Apply to save modification.
- 2) Press Autoprovision Now.

The set will try to download the SIP configuration file from the provisioning server immediately.

Note: The configuration file name. If it is empty, the telephone will request the common file and device file which is named as its MAC address. The default file format is CFG format. The file name could be a common name such as \$mac.cfg, \$input.cfg. The file format supports CFG/TXT/XML. If you want to download a XML format device file, the file name can be \$config.mac.xml.

2.5 Central provisioning with SIP configuration files

You can set up ALE H2/H2P DeskPhone sets by a SIP configuration file, downloaded by the set from a provisioning server using FTP, TFTP, HTTP or HTTPS. The SIP configuration file contains necessary information to allow the phone to register to the SIP server. The file name is \$config.mac.xml.

For more information on the SIP configuration file content and structure, see:

- Description of the SIP Settings in configuration file
- SIP configuration file templates

To configure a provisioning server in your network environment, see: Provisioning server setup overview.

The download path of SIP configuration file can be provisioned:

- By DHCP: see Setting up a DHCP server
- By EDS: see Setting up auto-provisioning with EDS
- By SIP PnP: see Setting up auto-provisioning with SIP server
- Statically:

- Via WBM: see configuring the provisioning server URL via WBM

2.6 Classification of Configuration Files

1) By function

- General configuration file

A general configuration file takes effect for all terminals. The general configuration file is named differently on different terminal models.

The rules for naming the general configuration file are described as follows:

Model	Name of General Configuration File
H2P	A0V0H2P0000.cfg
H2	A0V0H20000.cfg

- Configuration file named by users

Users can define the name of a configuration file. For example, if a user names a device configuration file as name.xml, the telephone set initiates a request to the server to download the general configuration file name.xml. The user can enter the corresponding configuration file name and download the upgrade configuration from the server.

- Configuration file named after MAC addresses

A configuration file named after a terminal MAC address is valid only for the terminal with the MAC address contained in the configuration file name. For a configuration file named after a MAC address, the MAC address contained in the file name is one for which the connectors are removed. For example, the MAC address of an H2 terminal is 00:15:65:11:3a:f8 and the configuration file name is config.001565113af8.xml. A user can upgrade the specified telephone set with this file.

2) By format

- XML format
- CFG format
- TXT format
 - Supported file formats include cfg, txt, and xml.
 - Internal file format

The file header is 64 characters long and ends with a carriage return character (\r\n).

For example, <<VOIP CONFIG FILE>>Version: 2.0002

Pay attention to the part "Version: 2.0002". If a telephone set is successfully upgraded using the auto provision mode, the version number (for example 2.0002) is displayed in the version number position on the webpage. If no version is carried, the digest of the configuration file is displayed.

End of file

For example, <<END OF FILE>>

To update an option, the module header of this option must be carried.

For example, to modify "Host Name :<GLOBAL CONFIG MODULE> must be carried.

```
<<VOIP CONFIG FILE>>Version: 2.0002
```

```
<GLOBAL CONFIG MODULE>
```

```
Host Name : VOIP (not less than 20 characters)
```

```
<<END OF FILE>>
```

3) By encryption status

- Unencrypted configuration file
- Encrypted configuration file
 - Unencrypted configuration file

The content of an unencrypted configuration file is displayed in plaintext, as shown in Figure 1.

```
<<VOIP CONFIG FILE>>Version:2.0002
<GLOBAL CONFIG MODULE>
Time Zone :32
<AUTOUPDATE CONFIG MODULE>
Auto Pbook Url :tftp://123:123@172.16.6.70/500.csv
Auto Image Url :http://123:123@172.16.6.70:8000/x4.z
Auto Etc Url :tftp://172.16.6.70/sips.pem
<<END OF FILE>>
```

Figure 1

- Encrypted configuration file

The content of an encrypted configuration file can't be displayed in plaintext.

If a downloaded configuration file is encrypted using AES, an AES key is required to decrypt the configuration file. The key must contain 64 hexadecimal characters (0 to F). All configuration files can be encrypted. Log in to the webpage and choose Maintenance > Auto Provision. Enter the key in config Encryption Key if an encrypted general configuration file is to be downloaded and in Common Config Encryption Key if other encrypted configuration files are to be downloaded, as shown in Figure 3. If a configuration file to be downloaded is not encrypted but you enter a key in the corresponding position, the telephone set considers the configuration file as an encrypted one.

Auto Provision Settings

Current Config Version	2.0002
Common Config Version	2.0002
CPE Serial Number	00100400XH020010000000010e597052
User	<input type="text"/>
Password	<input type="text"/>
Config Encryption Key	<input type="text"/>
Common Config Encryption Key	<input type="text"/>
Save Auto Provision Information	<input type="checkbox"/>

DHCP Option Settings >>

Plug and Play (PnP) Settings >>

Phone Flash Settings >>

TR069 Settings >>

2.7 Connecting the phone set to the customer network

To connect the phone set to the customer network:

- If the phone set is powered by PoE:
Plug the RJ45 cable into the set LAN connector
Connect the RJ45 cable to the customer network via a PoE hub/switch (IEEE802.3af compliant)
- If the phone set is not powered by PoE:
Plug the AC/DC external adapter to the set power supply connector (DCSV) and connect the plug to the power supply

Once the phone set is connected and powered up, it automatically starts initializing.

The phone set begins the initialization process by following steps after connected the power and network:

- 1) The Line LEDs indicators glow blue.
- 2) The message “Alcatel-Lucent Enterprise” logo appears on the phone screen when the phone set starts up.
- 3) The main phone screen displays the following:
- 4) Press the softkey “Setting” to go into the settings menu.
- 5) Press “OK” key to check the phone IP address.

3. Accessing phone set information

3.1 Checking phone set hardware information

A label is present on the back of the phone set. It provides hardware information such as the phone model, S/N and MAC address.

3.2 Checking the software version of the phone set

You can check the software version of the phone set as below:

Go into the MMI menu : Setting->Version

Press the “OK” key to check the phone software version.

3.3 Checking the network settings of the phone set

You can check the network settings of the phone set via the MMI:

Go into the MMI menu : Setting->Network

Press the “OK” key to check the phone network settings.

4. Commissioning phone sets

This chapter describes basic initialization instructions of ALE H2/H2P DeskPhone sets. Four scenarios are described:

- Scenario 1: IP static initialization on LAN, no SIP configuration file
- Scenario 2: IP dynamic configuration on LAN, no SIP configuration file
- Scenario 3: IP dynamic configuration on LAN with SIP configuration file (zero touch)
- Scenario 4: IP dynamic configuration on WAN with SIP configuration file (zero touch)

4.1 Scenario 1: IP static initialization on LAN, no SIP configuration file

Scenario 1 describes the commissioning of a set on the LAN with IP static initialization (no DHCP server) and without SIP configuration files (no provisioning server). In this scenario, all the configuration is performed via the set MMI.

Before beginning: you must know the following:

- IP parameters of the set (IP address, netmask, router IP address)
- SIP parameters: SIP call server information (IP address, domain, authentication)

To commission the set:

- 1) Configure the phone set on the SIP call server as needed according to the SIP server documentation
- 2) Connect the set: see Connecting the phone set to the customer network
- 3) Access the phone set user interface (MMI) and configure the following:
 - Change the initialization mode from Dynamic (default mode) to Static: see: Configuring IP parameters via MMI
 - Configure the IP parameters of the phone set: see: Configuring IP parameters via MMI

- Configure SIP parameters (via a SIP account): see: Configuring SIP account parameters via MMI

4.2 Scenario 2: IP dynamic configuration on LAN, no SIP configuration file

Scenario 2 describes the commissioning of a set on the LAN with IP dynamic initialization (provision of standard IP parameters by DHCP server) and without SIP configuration files (no provisioning server). In this scenario, the set gets its IP parameters from the DHCP server and SIP parameters are configured manually via WBM.

Before beginning: you must know the following:

SIP parameters: SIP call server information (IP addressing, domain, authentication)

Prerequisites:

A DHCP server is operational on the LAN (no specific configuration required): see Setting up a DHCP server

To commission the set:

- 1) Configure the phone set on the SIP call server as needed according to the SIP server documentation
- 2) Connect the set: see Connecting the phone set to the customer network
- 3) Read the IP address on the phone set display
- 4) Access the phone set configuration via WBM: see Configuring IP parameters and SIP account parameters via WBM
- 5) Configure SIP parameters

4.3 Scenario 3: IP dynamic configuration on LAN with SIP configuration file (zero touch)

Scenario 3 describes the commissioning of a set on the LAN with IP dynamic initialization (provision of standard IP parameters by DHCP server) and with SIP configuration file which will be downloaded during set initialization from a provisioning server, whose URL is provided by the DHCP server: this requires a specific configuration on the DHCP server. In this scenario, the set starts without any manual operation via MMI or WBM (zero touch).

Before beginning: you must know the following:

SIP parameters: SIP call server information (IP addressing, domain, authentication): this information is require to build the configuration file

Prerequisites:

The phone set must initialize in dynamic mode (default mode)

A DHCP is operational on the LAN and configured to provide the URL of the provisioning server (DM URL): see Setting up a DHCP server and DHCP configuration for download path of SIP configuration files

A provisioning server is operational on the LAN: see Setting up a provisioning server

To commission the set:

- 1) Configure the phone set on the SIP call server as needed according to the SIP server documentation

- 2) Create and configure the SIP configuration file: see: Building a SIP configuration file
- 3) Deploy the SIP configuration file in the provisioning server relative directory
- 4) Connect the set: see Connecting the phone set to the customer network

After the last initialization step, the set registers to the SIP server.

4.4 Scenario 4: IP dynamic configuration on WAN with SIP configuration file (zero touch)

Scenario 4 describes the commissioning of a set on the WAN with IP dynamic initialization (provision of standard IP parameters by DHCP server) and with SIP configuration files which will be downloaded during set initialization from a provisioning server, whose URL is provided by the EDS server: this requires a specific configuration on the EDS server. In this scenario, the set starts without any manual operation via MMI or WBM (zero touch).

Before beginning: you must know the following:

SIP parameters: SIP call server information (IP addressing, domain, authentication): this information is required to build the configuration file

Prerequisites:

The phone set can reach the WAN

The phone set must initialize in dynamic mode (default mode)

A DHCP is operational on the LAN (no specific configuration required): see [Setting up a DHCP server](#).

A provisioning server is operational on the WAN or Cloud: see [Setting up a provisioning server](#).

A profile associated to the phone MAC address has been created on the EDS server to provision DM URL and certificate relative URL: see [Setting up auto-provisioning with EDS](#)

To commission the set:

- 1) Configure the phone set on the SIP call server as needed according to the SIP server documentation
- 2) Create and configure the SIP configuration file: see: Building a SIP configuration file
- 3) Deploy the SIP configuration file in the provisioning server relative directory
- 4) Connect the set: see Connecting the phone set to the customer network

After the last initialization step, the set registers to the SIP server.

5 Setting up a DHCP server

This chapter details the configuration of the DHCP server to be performed when ALE H2/H2P DeskPhone sets initialize in dynamic mode.

The DHCP can be used to provide standard IP parameters only (see Commissioning phone sets: scenarios 2, 4) or standard IP parameters and the DM URL (see Commissioning phone sets: scenario 3). When the DM URL is not provisioned by the DHCP server, no specific configuration is required on the DHCP server: only standard IP parameters are required.

You can skip this section if ALE H2/H2P DeskPhone sets initialize in static mode (see [Commissioning phone sets: scenario 1](#)).

DHCP option configuration for IPv4 details the list of DHCP options supported by ALE H2/H2P DeskPhone sets.

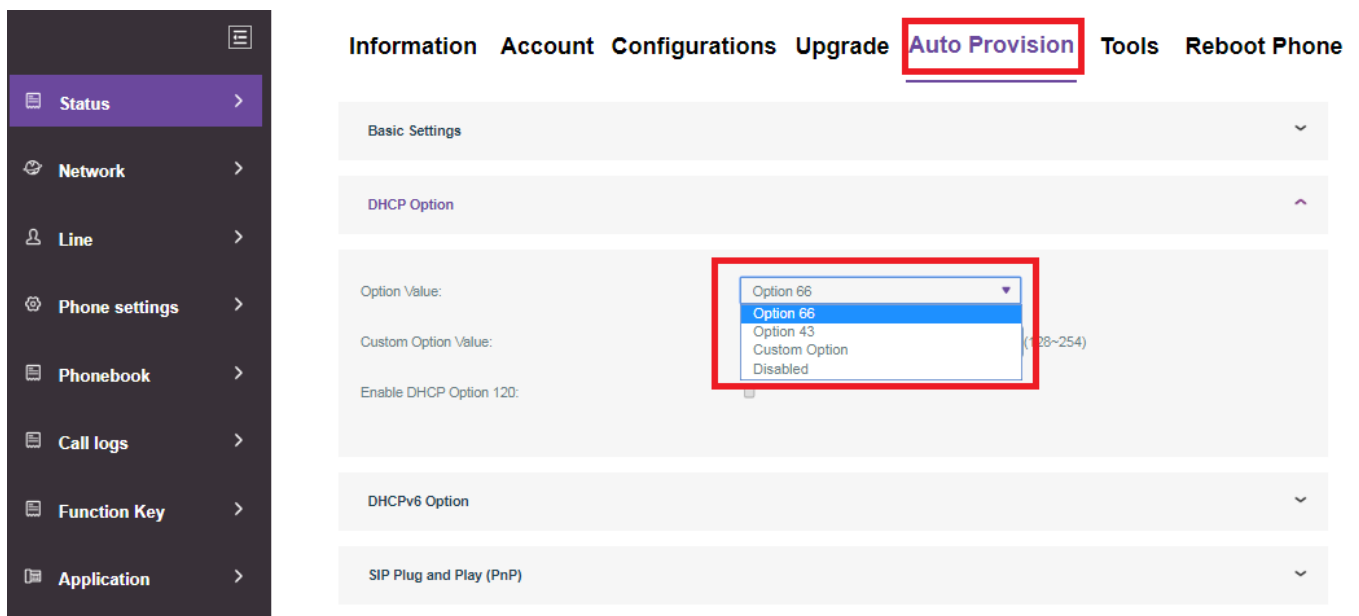
DHCP configuration for download path of SIP configuration files describes how to configure the download path of SIP configuration files on the DHCP server.

Note: Configuring the download path is not required when auto-provisioning with EDS is used.

5.1 DHCP option configuration for IPv4

DHCP Option

- 1) To use the DHCP Option mode, the network mode of the telephone set must be DHCP.
- 2) DHCP Option has four options: DHCP Option 66, DHCP Option 43, Custom DHCP Option, and DHCP Option Disable. The default is Option 66.



5.2 DHCP configuration for download path of SIP configuration files

If DHCP Option 43 is chosen when auto provision parameters are obtained through DHCP, the DHCP Discover and DHCP Request messages sent by the terminal to the server contain the following field values:

43 = Vendor-Specific Information

The DHCP Offer and DHCP ACK messages sent by the server to the terminal contain the following field values:

The value is for example: [http://172.16.6.45/\\$config.mac.xml](http://172.16.6.45/$config.mac.xml).

The auto provision parameters of DHCP Option 66 and Custom DHCP are the same as those of DHCP Option 43.

DHCP Option Disable indicates disabling DHCP Option.

6. Setting up a provisioning server

6.1 Provisioning server setup overview

A provisioning server is necessary when SIP configuration files are used (see Commissioning phone: scenarios 3, 4).

You can skip this section if ALE H2/H2P DeskPhone sets initialize without SIP configuration files (see Commissioning phone sets: scenarios 1, 2).

ALE H2/H2P DeskPhone sets support the following transport protocols for provisioning: FTP, TFTP, HTTP and HTTPS.

The HTTP/HTTPS provisioning server can be set up on the local LAN. Use the following procedure as a recommendation if this is your first provisioning server setup.

To set up the provisioning environment:

- 1) Install an FTP/TFTP/HTTP/HTTPS server application or locate a suitable existing server.
- 2) Create an account and home directory.
- 3) Set security permissions for the account.

Once the setup has been completed, create the SIP configuration files required for set commissioning (see: Building a SIP configuration file), and copy them in the FTP/TFTP/HTTP/HTTPS provisioning server relative directory.

If the phone has retrieved a DM URL from the DHCP server, it downloads the configuration file from the provisioning server.

Note: The DM URL which is configured in the DHCP server corresponds to the path of the SIP configuration files stored on the provisioning server.

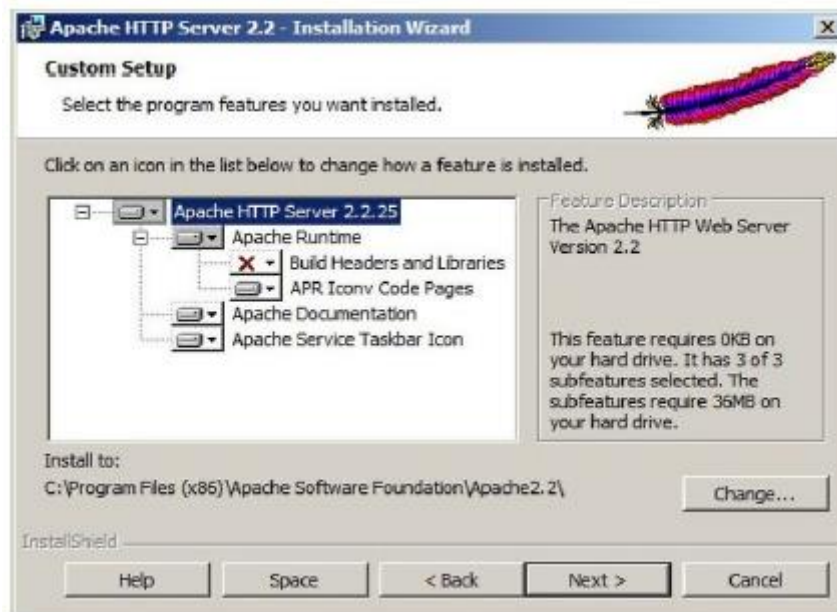
6.2 Example with the Apache HTTP server setup

Configure a Windows server system (or virtual system) to set up an Apache web server with following standard steps. When the Apache web server has been successfully installed, create a directory on this server to store the SIP configuration files or firmware binary files, and get the download URL for set commissioning.

- 1) To set up an Apache HTTP server
- 2) Install the Apache web server

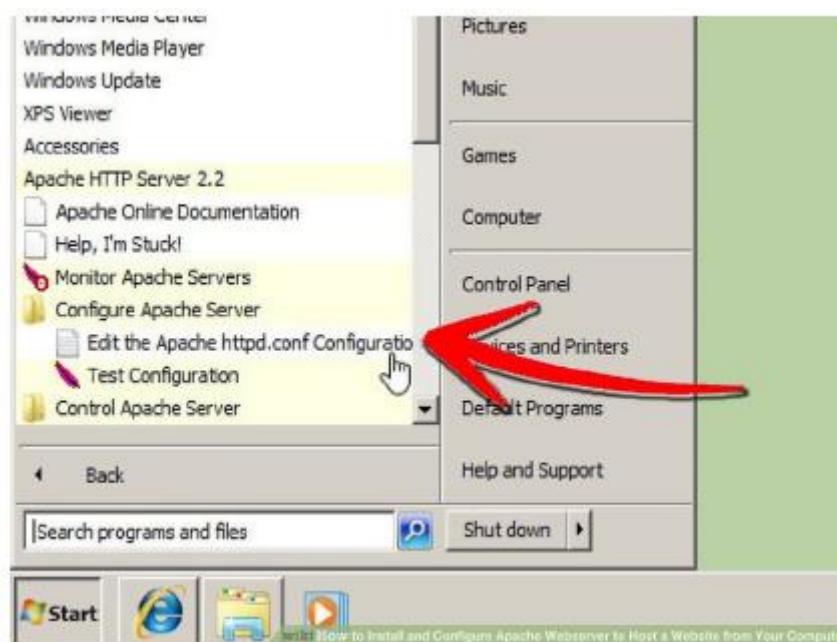
When installing Apache, you are asked to enter your domain name, network name, and e-mail address. You can add any value in these fields. The format must be:

- Domain name: example.com
 - Network name: www.example.com
 - E-mail address: user@example.com
- 3) Click Next
 - 4) Select the Apache HTTP server from the radio button list



An error message is displayed such as: "Apache could not be configured. Edit you Apache.conf file"

- 5) Go to: Start > Programs > Apache HTTP server <version number> > Configure Apache Server > Edit the Apache httpd.conf configuration file



- 6) Go to the DocumentRoot line
- 7) Change the document root to point to the location of your website folder, using the character "/", instead of "\"


```

httpd - Notepad
File Edit Format View Help
#
# ServerAdmin: Your address, where problems with the server should be
# e-mailed. This address appears on some server-generated pages, such
# as error documents. e.g. admin@your-domain.com
#
ServerAdmin anuj@wikiphow.sample.com
#
#
# ServerName gives the name and port that the server uses to identify itself.
# This can often be determined automatically, but we recommend you specify
# it explicitly to prevent problems during startup.
#
# If your host doesn't have a registered DNS name, enter its IP address here.
#ServerName www.wikiphow.sample.com:
#
# DocumentRoot: The directory out of which you will serve your
# documents. By default, all requests are taken from this directory, but
# symbolic links and aliases may be used to point to other locations.
#
DocumentRoot "121.30.225.250/root/users/shared"
#
# Each directory to which Apache has access can be configured with respect
# to which services and features are allowed and/or disabled in that
# directory (and its subdirectories).
#
# First, we configure the "default" to be a very restrictive set of
# features.
#
<Directory />

```

8) Repeat this operation for <Directory "drive:/location">

```

httpd - Notepad
File Edit Format View Help
</Directory>
#
# Note that from this point forward you must specifically allow
# particular features to be enabled, so if something's not working as
# you might expect, make sure that you have specifically enabled it
# below.
#
#
# This should be changed to whatever you set DocumentRoot to.
#
<Directory "121.30.225.250/root/users/shared">
#
# Possible values for the options directive are "None", "All",
# or any combination of:
#   Indexes Includes FollowsSymLinks SymLinksIfOwnerMatch ExecCGI MultiViews
#
# Note that "MultiViews" must be named *explicitly* --- "options All"
# doesn't give it to you.
#
# The options directive is both complicated and important. Please see
# http://httpd.apache.org/docs/2.2/mod/core.html#options
# for more information.
#
Options Indexes FollowsSymLinks
#
# AllowOverride controls what directives may be placed in .htaccess files.
# It can be "All", "None", or any combination of the keywords:
#   Options FileInfo AuthConfig Limit
#

```

9) To verify your configuration, go to Apache in your taskbar and stop the service

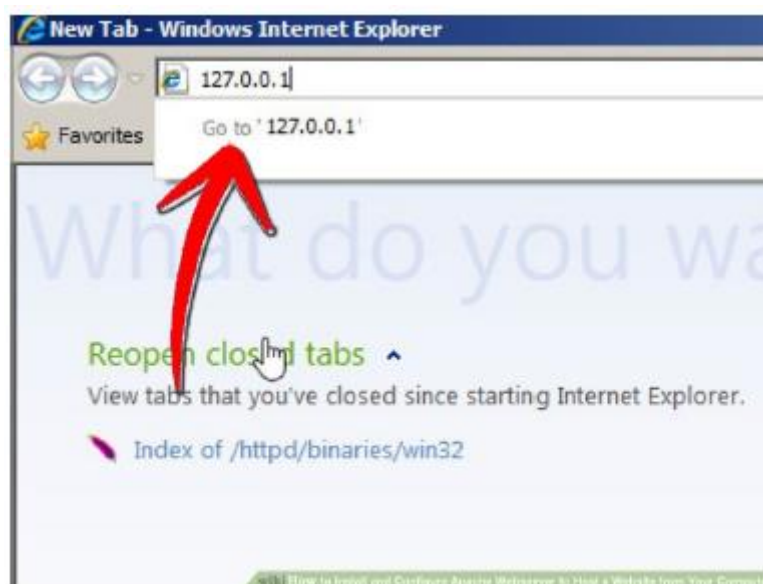


10) Restart the service



If the service does not start, modify the Apache httpd.conf configuration file properly

11) Once the service is restarted, open a web browser and enter localhost or 127.0.0.1 in address Bar



6.3 Building a SIP configuration file

Before beginning, you must have the following:

The MAC address of the phone set required for the name of the SIP configuration file ({mac address of the phone set}.xml, for example config.00809fe7021e.xml): see Checking phone set hardware information

A text editor, such as Notepad++, to create and edit configuration file Build the SIP configuration file required for set commissioning:

1) Install a FTP/TFTP/HTTP/HTTPS server application or locate a suitable existing server.

For details on the file structure, name and minimum settings, see: SIP configuration file templates

2) Complete the SIP configuration file according to your needs.

For details on the available settings, see: Description of the SIP Settings in configuration file.

Once creation is completed, copy the SIP configuration file in the FTP/TFTP/HTTP/HTTPS provisioning server relative directory.

7. Setting up auto-provisioning with EDS

ALE H2/H2P DeskPhones support Zero Touch Deployment using ALE Easy Deployment Server (EDS). You can contact the ALE EDS administrator account.eds@al-enterprise.com for more information. You can connect to <https://admin.eds.al-enterprise.com/register> to sign-up for a free EDS account.

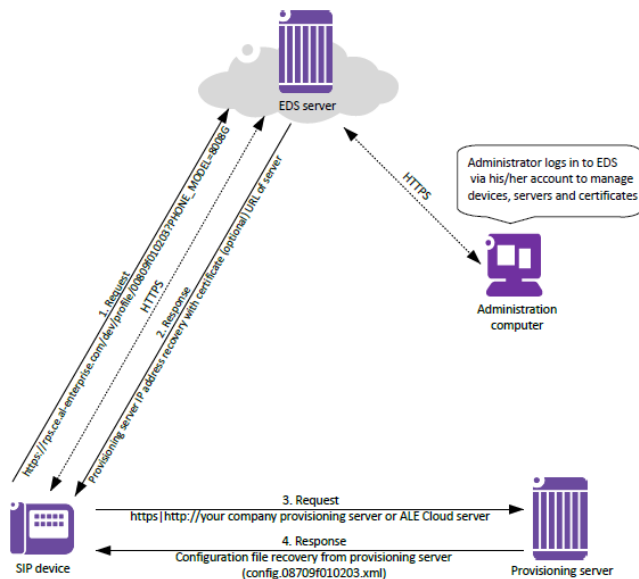
ALE EDS is a server side service that helps ALE H2/H2P DeskPhone sets to connect to the provisioning server on first startup. The service is deployed on the Internet Cloud.

The EDS server allows to provision sets with the DM URL and certificates, allowing them to initialize from the WAN (see Commissioning phone sets: scenario 4), without requiring a specific configuration of the DHCP server.

When the set starts in dynamic mode and no provisioning server URL (DM URL) is configured via MMI or received from DHCP, it tries to connect to the ALE EDS server, whose address is hard-coded in its software. The server verifies the set's MAC address, and searches a profile for the set in the database.

The provisioning server URL and certificate relative URL are provided in the profile. The set downloads certificate, connects to the provisioning server via this URL, and downloads its SIP configuration file.

Note: Auto-provisioning with EDS does not apply to phone sets initializing in static mode.



8. Setting up auto-provisioning with SIP server (SIP PnP)

PnP provides a SIP-based configuration upgrade/deployment method. Enter the server IP address and port and select Enable SIP PnP

The screenshot shows the configuration interface for SIP PnP. The 'Auto Provision' tab is active. The 'SIP Plug and Play (PnP)' section is expanded, and the following settings are visible:

- Enable SIP PnP:**
- Server Address:** 224.0.1.75
- Server Port:** 5060
- Transport Protocol:** UDP
- Update Interval:** 1 (1-99)Hour(s)

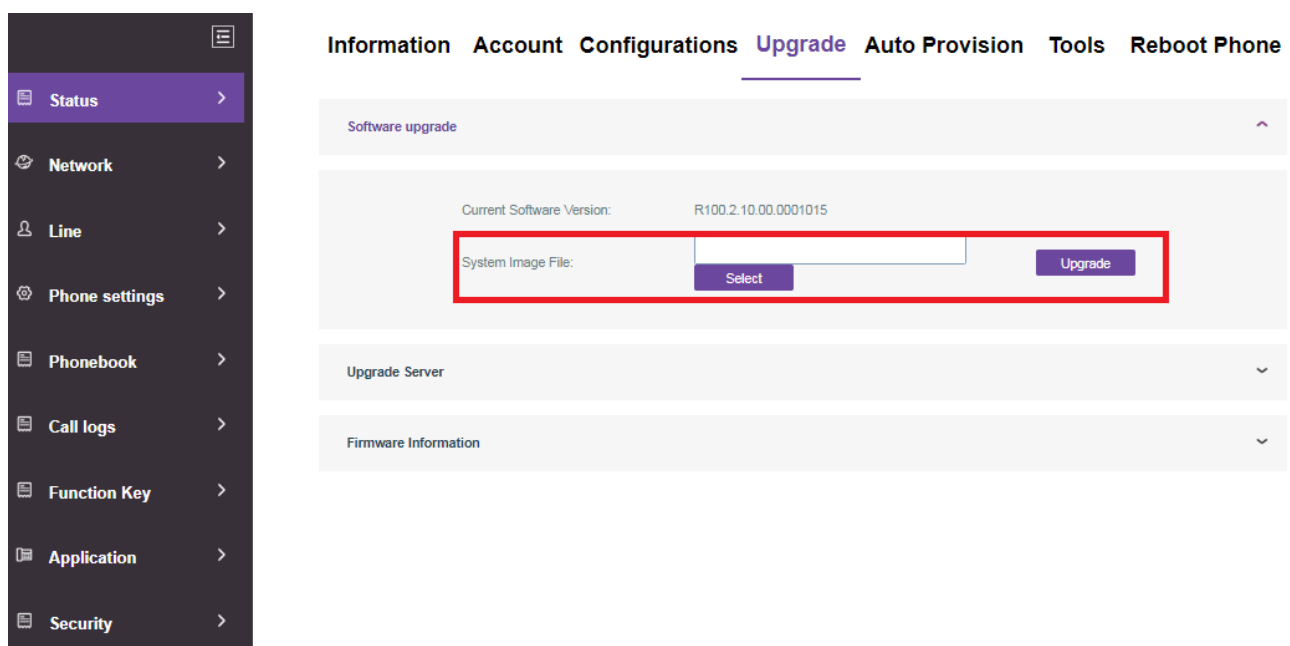
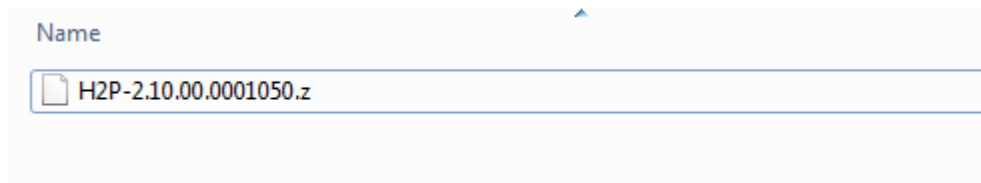
If PnP is enabled for a terminal, the terminal sends a SIP SUBSCRIBE message periodically in multicast mode. A SIP server supporting this message responds to the message and returns a SIP NOTIFY message carrying the path of the auto configuration/deployment server. The terminal can obtain the configuration file to be downloaded from this path. This auto configuration/deployment method applies to scenarios without a default auto configuration/deployment server or scenarios where a terminal uses a static IP address and cannot automatically obtain related parameters through DHCP Option.

9. Upgrading the firmware

This chapter details the firmware upgrade of ALE H2/H2P DeskPhone sets. You can have the binary by connected <http://www.aledevice.com/site/download>.

9.1 Upgrading by WBM

- 1) Download the binary file and save it on your local PC
- 2) Connect to the set WBM and go to: Status > Upgrade > Software Upgrade
- 3) Click Add binary files
- 4) Select the binary file which is one “.z” format file.



- 5) Click Upgrade

After download, the phone installs the new binary and reboots.

On the phone set, you can check the version from the settings menu (see: Checking the software version of the phone set).

9.2 Upgrading by configuration file

You can upgrade the H2/H2P phone using the SIP provisioning server and SIP configuration with relevant parameters.

H2/H2P phone can upgrade by downloading firmware binary files from a provisioning server whose URL must be defined in the SIP configuration file.

Settings:

<ota>

```
<FirmwareUrl> http://192.168.2.2/ale/firmware/ H2P-2.10.00.0001050.z </FirmwareUrl>
</ota>
```

Description:

Set up the upgrading URL. Put the firmware binary file in the directory of provisioning server, for example the URL could be http://192.168.2.2/ale/firmware/ H2P-2.10.00.0001050.z

Then this settings should be: FirmwareUrl: http://192.168.2.2/ale/firmware/ H2P-2.10.00.0001050.z

You can trigger an immediate upgrade by resetting manually the H2/H2P DeskPhone set.

Automatic update operates in the following way:

- The phone polls for config file at the time defined by “Flash Interval”
- If the directory contains a different version (older or newer), the set updates with this version
- If no binary file is available on the server, or the version is the same, nothing happens

The following settings for the SIP configuration file template used for the phone upgrade at phone startup.

```
<ota>
  <FirmwareUrl></FirmwareUrl>
</ota>
```

- 1) Prepare the SIP configuration file and put it on the SIP provisioning server
- 2) Put the firmware binary on the SIP provisioning server
- 3) Start the phone. The phone downloads the SIP configuration file and enter the upgrading process. The phone reboots automatically after finishing the whole upgrading process.
- 4) When the phone restarts, check that it had been upgraded to the desired version (refer to [Checking the software version of the phone set](#))

10. Troubleshooting

When the phone is not in normal use, the user can try the following methods to restore normal operation of the phone or collect relevant information and send a problem report to ALE.

10.1 Get Device System Information

Users can get information by pressing the [Setting] >> [Network/Version] option in the phone.

The following information will be provided:

The network information

Equipment information (model, software and hardware version), etc.

10.2 Reboot Device

Users can reboot the device from soft-menu, [Setting] >> [Reboot], and confirm the action by [OK].

Or, simply remove the power supply and restore it again.

10.3 Reset Device to Factory Default

Reset Device to Factory Default will erase all user's configuration, preference, database and profiles on the device and restore the device back to the state as factory default.

To perform a factory default reset, user should press [Setting] >> [Admin], and then input the password to enter the interface. Then choose [Restore Factory] and press [OK], and confirm the action by [OK]. The device will be rebooted into a clean factory default state.

10.4 Use Telnet

- Enable Telnet.

Telnet default is disable and can be enabled in web.

The screenshot shows the Alcatel-Lucent Enterprise web interface. The left sidebar contains navigation menus: Status, Network (selected), Line, Phone settings, Phonebook, Call logs, Function Key, Application, Security, and Device Log. The main content area is titled 'Service Port Settings' and includes tabs for Basic, Service Port (selected), VPN, and Advanced. The settings are as follows:

Setting	Value
Web Server Type:	HTTPS
Web Logon Timeout:	15 (10~30)Minute
web auto login:	<input type="checkbox"/>
HTTP Port:	80
HTTPS Port:	443
RTP Port Range Start:	10000
RTP Port Quantity :	1000
Enable Telnet:	<input type="checkbox"/>
Telnet Port:	23

An 'Apply' button is located at the bottom of the settings form.

Now, we just need to connect a PC to the phone same LAN. Type the command "telnet x.x.x.x" to access the phone system. "x.x.x.x" should be replaced by the phone IP address. Then, type root and will go to the phone system.

10.5 Command in Telnet

The command is similar like the Linux command.

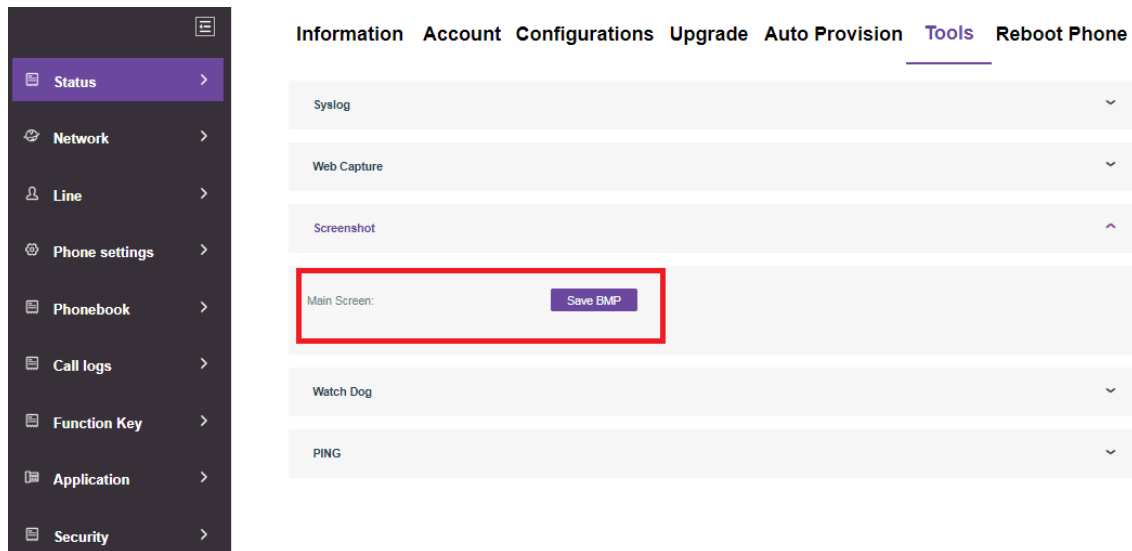
ps: view all the tasks

top: CPU , Memory information

ifconfig: network information

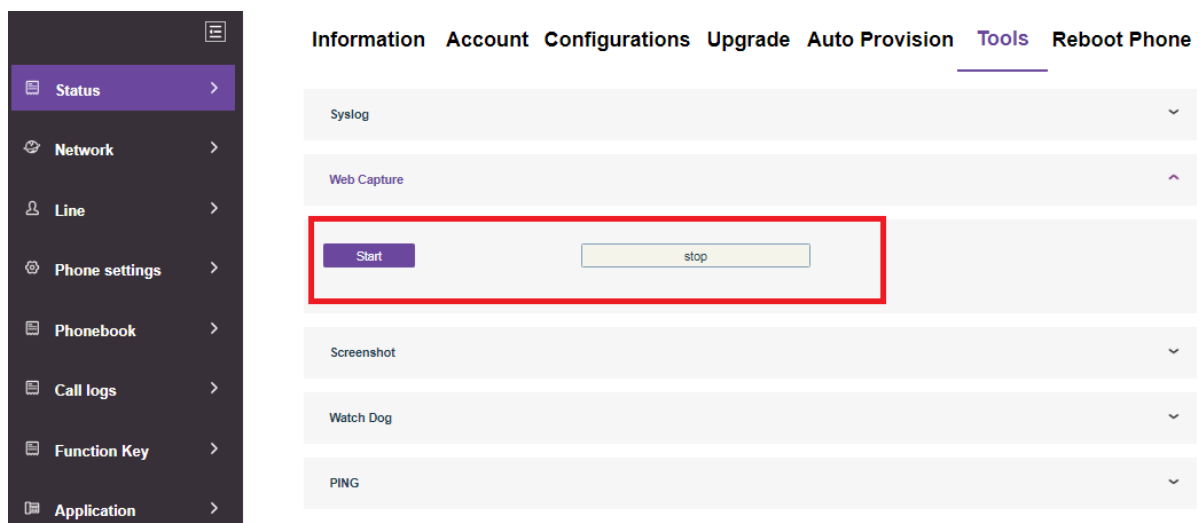
10.6 Screenshot

If there is a problem with the phone, the screenshot can help the technician locate the function and identify the problem. In order to obtain screen shots, log in the phone webpage [Status] >> [Tools], and you can capture the pictures of the main screen and the secondary screen (you can capture them in the interface with problems).



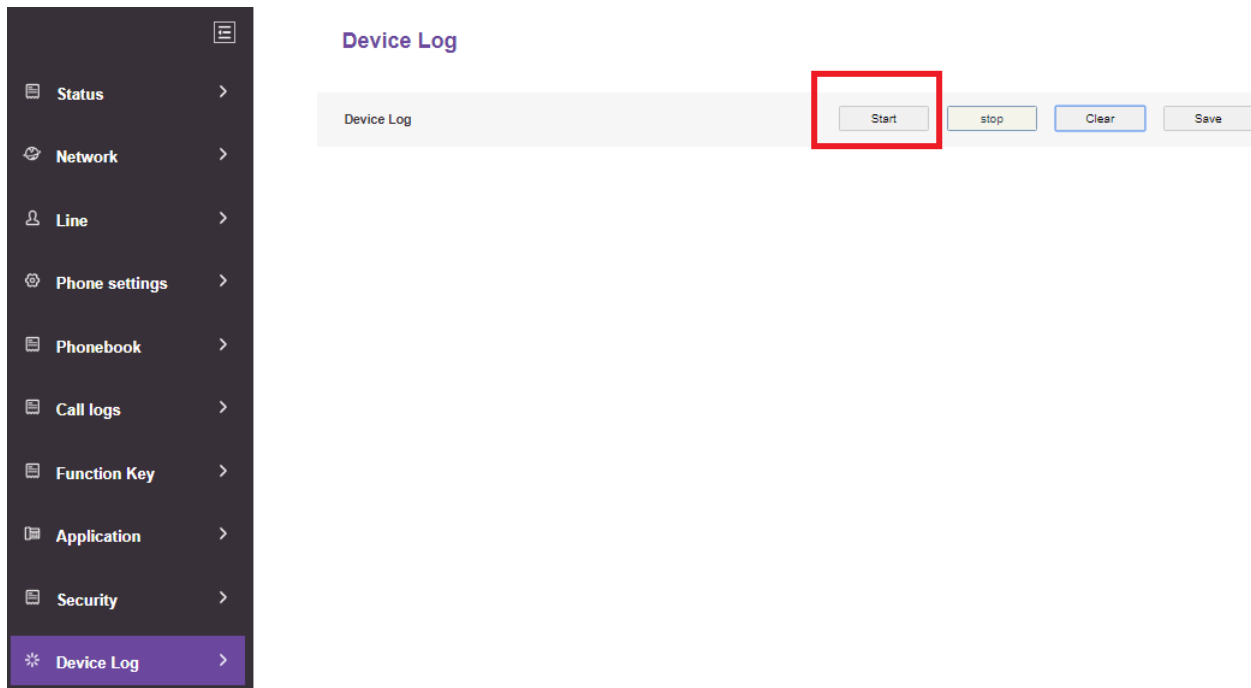
10.7 Network Packets Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, user needs to log in the device web portal, open page [Status] >> [Tools] and click [Start] in “Network Packets Capture” section. A pop-up message will prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [Stop] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file.



10.8 Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, the user can log in the phone web page, open the page [Device log], click the [Start] button, follow the steps of the problem until the problem appears, and then click the [End] button, [Save] to local analysis or send the log to the technician to locate the problem.



10.9 Syslog capture

Go to Status - tools - Syslog on phone web page:

The screenshot shows the 'Syslog' configuration page. It has a light blue background and contains the following fields and controls:

- Enable Syslog:** A checkbox that is checked.
- Server Address:** A text input field containing '172.16.12.19'.
- Server Port:** A text input field containing '514'.
- APP Log Level:** A dropdown menu set to 'Debug'.
- SIP Log Level:** A dropdown menu set to 'Debug'.
- Apply:** A button at the bottom of the form.

- 1) Enable syslog
- 2) Server Address is the syslog server address or the computer IP address that is running Wireshark.
- 3) Log level is debug
- 4) Click the "Apply" button
- 5) After applied, reproduce the issue. While the issue occurs, stop and save the log file.

If using a capture tool, start the capture tool before the syslog. While syslog is ending, make the capture tool stop and save the capture trace file.

11. Appendixes

11.1 SIP configuration file templates

```
<?xml version="1.0" encoding="UTF-8"?>
<sysConf>
  <Version>2.0000000000</Version>
  <net>
    <WANTYPE>0</WANTYPE>
    <WANIP>192.168.1.179</WANIP>
    <WANSubnetMask>255.255.255.0</WANSubnetMask>
    <WANGateway>192.168.1.1</WANGateway>
    <DomainName></DomainName>
    <PrimaryDNS>8.8.8.8</PrimaryDNS>
    <SecondaryDNS>202.96.134.133</SecondaryDNS>
    <EnableDHCP>1</EnableDHCP>
    <DHCPAutoDNS>1</DHCPAutoDNS>
    <DHCPAutoTime>0</DHCPAutoTime>
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    <UseVendorClassID>0</UseVendorClassID>
    <VendorClassID>ALE H2P</VendorClassID>
    <EnablePPPoE>0</EnablePPPoE>
    <PPPoEUser>user123</PPPoEUser>
    <PPPoEPassword>password</PPPoEPassword>
    <ARPCacheLife>2</ARPCacheLife>
    <MTU>1500</MTU>
    <WAN6IP></WAN6IP>
    <WAN6IPPREFIX></WAN6IPPREFIX>
    <WAN6Gateway></WAN6Gateway>
    <Domain6Name></Domain6Name>
    <PrimaryDNS6></PrimaryDNS6>
    <SecondaryDNS6></SecondaryDNS6>
    <EnableDHCP6>1</EnableDHCP6>
    <DHCP6AutoDNS>1</DHCP6AutoDNS>
    <DHCP6AutoTime>0</DHCP6AutoTime>
```

```
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<Vendor6ClassID></Vendor6ClassID>
</net>
<mm>
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  <ILBCPayloadLen></ILBCPayloadLen>
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  <G726-32PayloadType></G726-32PayloadType>
  <G726-40PayloadType></G726-40PayloadType>
  <DtmfPayloadType></DtmfPayloadType>
  <OpusPayloadType></OpusPayloadType>
  <OpusSampleRate></OpusSampleRate>
  <VAD></VAD>
  <ResvAudioBand></ResvAudioBand>
  <RTPInitialPort></RTPInitialPort>
  <RTPPortQuantity></RTPPortQuantity>
  <RTPKeepAlive></RTPKeepAlive>
  <RTPRelay></RTPRelay>
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  <RTCPCNAMEHost></RTCPCNAMEHost>
  <SelectYourTone></SelectYourTone>
  <SidetoneGAIN></SidetoneGAIN>
  <PlayEgressDTMF></PlayEgressDTMF>
  <DialTone></DialTone>
  <RingbackTone></RingbackTone>
  <BusyTone></BusyTone>
  <CongestionTone></CongestionTone>
  <CallwaitingTone></CallwaitingTone>
  <HoldingTone></HoldingTone>
  <ErrorTone></ErrorTone>
  <StutterTone></StutterTone>
  <InformationTone></InformationTone>
  <DialRecallTone></DialRecallTone>
```

```
<MessageTone></MessageTone>
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<NumberUnobtainable></NumberUnobtainable>
<WarningTone></WarningTone>
<RecordTone></RecordTone>
<AutoAnswerTone></AutoAnswerTone>
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  <AudioCodecSets>PCMU,PCMA,G726-16,G726-24,G726-32,G726-
40,G729,iLBC,opus,G722</AudioCodecSets>
</capability>
</mm>
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  <SIPWaitStunTime>800</SIPWaitStunTime>
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```

```
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<BusyFWDNum></BusyFWDNum>
<NoAnswerFWDNum></NoAnswerFWDNum>
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```

```
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<ServerType>0</ServerType>  
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```
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<EnableMACHeader>0</EnableMACHeader>
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  <Host></Host>
  <Port>0</Port>
```

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</addr>
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  <Port>0</Port>
  <Channel>0</Channel>
</addr>
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  <Host></Host>
  <Port>0</Port>
  <Channel>0</Channel>
</addr>
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  <Host></Host>
  <Port>0</Port>
  <Channel>0</Channel>
</addr>
<dynamic>
  <AutoExitExpires>60</AutoExitExpires>
</dynamic>
</mcast>
<dsskey>
  <SelectDsskeyAction>0</SelectDsskeyAction>
  <MemoryKeytoBXfer>3</MemoryKeytoBXfer>
  <FuncKeyPageNum>1</FuncKeyPageNum>
  <SideKeyPageNum>1</SideKeyPageNum>
  <DSSHomePage>0</DSSHomePage>
  <DisplayParkedInfo>0</DisplayParkedInfo>
  <DSSDIALSwitchMode>0</DSSDIALSwitchMode>
  <FirstCallWaitTime>16</FirstCallWaitTime>
  <FirstNumStartTime>360</FirstNumStartTime>
```

```
<FirstNumEndTime>1080</FirstNumEndTime>
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<Extern2PageBelong>0</Extern2PageBelong>
<Extern3PageBelong>0</Extern3PageBelong>
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<internal index="1">
  <Fkey index="1">
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    <Value>SIP1</Value>
    <Title></Title>
    <ICON>Green</ICON>
  </Fkey>
  <Fkey index="2">
    <Type>2</Type>
    <Value>SIP2</Value>
    <Title></Title>
    <ICON>Green</ICON>
  </Fkey>
</internal>
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  <Type>3</Type>
  <Value>F_PBOOK</Value>
  <Title></Title>
```



```
<ICON>Green</ICON>
</dssSoft>
<dssSoft index="2">
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  <Value></Value>
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  <Value></Value>
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  <Value></Value>
  <Title></Title>
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</dssSoft>
<dssSoft index="7">
  <Type>0</Type>
```

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</dssSoft>
<dssSoft index="8">
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  <Value></Value>
  <Title></Title>
  <ICON>Green</ICON>
</dssSoft>
</dsskey>
<web>
  <WebServerType>1</WebServerType>
  <WebPort>80</WebPort>
  <HttpsWebPort>443</HttpsWebPort>
  <RemoteControl>1</RemoteControl>
  <EnableMMIFilter>0</EnableMMIFilter>
  <WebAuthentication>0</WebAuthentication>
  <EnableTelnet>0</EnableTelnet>
  <TelnetPort>23</TelnetPort>
  <TelnetPrompt></TelnetPrompt>
  <LogonTimeout>15</LogonTimeout>
```

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<account index="1">
  <Name>admin</Name>
  <Password>123456</Password>
  <Level>10</Level>
</account>
<account index="2">
  <Name>guest</Name>
  <Password>guest</Password>
  <Level>5</Level>
</account>
</web>
<log>
  <Level>ERROR</Level>
  <Style>level,tag</Style>
  <OutputDevice></OutputDevice>
  <FileName>platform.log</FileName>
  <FileSize>512KB</FileSize>
  <SyslogTag>platform</SyslogTag>
  <SyslogServer>0.0.0.0</SyslogServer>
  <SyslogServerPort>514</SyslogServerPort>
</log>
<tr069>
  <TR069Tone>1</TR069Tone>
  <CPESerialNumber>3c28a600012c</CPESerialNumber>
  <ACSServerType>1</ACSServerType>
  <EnableTR069>0</EnableTR069>
  <ACSURL>0.0.0.0</ACSURL>
  <ACSUserName>admin</ACSUserName>
  <ACSPassword></ACSPassword>
  <ACSBackupURL>0.0.0.0</ACSBackupURL>
  <ACSBackupUserName></ACSBackupUserName>
  <ACSBackupPassword></ACSBackupPassword>
  <CPEUserName>dps</CPEUserName>
  <CPEPassword>dps</CPEPassword>
```

```
<PeriodixInterval>3600</PeriodixInterval>
<TLSVersion>2</TLSVersion>
<AreaCode></AreaCode>
<STUNEnable>0</STUNEnable>
<STUNServerAddr></STUNServerAddr>
<STUNServerPort>3478</STUNServerPort>
<STUNLocalPort>30000</STUNLocalPort>
</tr069>
<hotspot>
  <EnableHotspot>0</EnableHotspot>
  <Mode>1</Mode>
  <ListenType>0</ListenType>
  <ListenIP>224.0.2.0</ListenIP>
  <ListenPort>16360</ListenPort>
  <OwnName>SIP Hotspot</OwnName>
  <RingMode>0</RingMode>
  <EnableManageMode>1</EnableManageMode>
  <EnableConfigMode>0</EnableConfigMode>
  <hs index="1">
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    <ExtPrefix></ExtPrefix>
  </hs>
  <hs index="2">
    <Enable>1</Enable>
    <ExtPrefix></ExtPrefix>
  </hs>
</hotspot>
<vpn>
  <VPNmode>-1</VPNmode>
  <EnableVPN>0</EnableVPN>
  <EnableNat>0</EnableNat>
  <Openvpnmode>0</Openvpnmode>
  <L2TPServerAddress>0.0.0.0</L2TPServerAddress>
  <L2TPUserName></L2TPUserName>
```

```
<L2TPPassword></L2TPPassword>
<L2TPNegotiateDNS>1</L2TPNegotiateDNS>
<PPTPServerAddress>0.0.0.0</PPTPServerAddress>
<PPTPUserName></PPTPUserName>
<PPTPPassword></PPTPPassword>
</vpn>
<mt>
  <ContactUpdateMode>0</ContactUpdateMode>
  <AutoServerDigest>0</AutoServerDigest>
</mt>
<ap>
  <AutoPbookUrl></AutoPbookUrl>
  <AutoEtcUrl></AutoEtcUrl>
  <DefaultUsername></DefaultUsername>
  <DefaultPassword></DefaultPassword>
  <InputCfgFileName></InputCfgFileName>
  <DeviceCfgFileKey></DeviceCfgFileKey>
  <CommonCfgFileKey></CommonCfgFileKey>
  <DownloadCommonConf>1</DownloadCommonConf>
  <SaveProvisionInfo>0</SaveProvisionInfo>
  <CheckFailTimes>5</CheckFailTimes>
  <FlashServerIP></FlashServerIP>
  <FlashFileName></FlashFileName>
  <FlashProtocol>2</FlashProtocol>
  <FlashMode>0</FlashMode>
  <FlashInterval>1</FlashInterval>
  <updatePBInterval>720</updatePBInterval>
  <APPswdEncryption>0</APPswdEncryption>
  <pnp>
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    <PNPIP>224.0.1.75</PNPIP>
    <PNPPort>5060</PNPPort>
    <PNPTransport>0</PNPTransport>
    <PNPInterval>1</PNPInterval>
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</pnp>
<opt>
  <DHCPOption>66</DHCPOption>
  <DHCPv6Option>0</DHCPv6Option>
  <DhcpOption120>0</DhcpOption120>
</opt>
</ap>
<ota>
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</ota>
<fwCheck>
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  <UpgradeServer1></UpgradeServer1>
  <UpgradeServer2></UpgradeServer2>
  <AutoUpgradeInterval>24</AutoUpgradeInterval>
</fwCheck>
<qos>
  <EnableVLAN>0</EnableVLAN>
  <VLANID>256</VLANID>
  <EnablePVID>0</EnablePVID>
  <PVIDValue>254</PVIDValue>
  <SignallingPriority>0</SignallingPriority>
  <VoicePriority>0</VoicePriority>
  <VideoPriority>0</VideoPriority>
  <LANPortPriority>0</LANPortPriority>
  <EnablediffServ>0</EnablediffServ>
  <SingallingDSCP>46</SingallingDSCP>
  <VoiceDSCP>46</VoiceDSCP>
  <VideoDSCP>46</VideoDSCP>
  <LLDPTransmit>1</LLDPTransmit>
  <LLDPRefreshTime>60</LLDPRefreshTime>
  <LLDPLearnPolicy>1</LLDPLearnPolicy>
  <LLDPSaveLearnData>0</LLDPSaveLearnData>
  <CDPEnable>0</CDPEnable>
```

```
<CDPRefreshTime>60</CDPRefreshTime>
<DHCPOptionVlan>132</DHCPOptionVlan>
</qos>
<dot1x>
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  <XsupUser></XsupUser>
  <XsupPassword></XsupPassword>
  <sslMode>
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    <CommonName>0</CommonName>
    <CTFmode>0</CTFmode>
    <DeviceCertMode>0</DeviceCertMode>
  </sslMode>
</dot1x>
<ale>
  <AleEnable>1</AleEnable>
  <AleUrl>https://device.eds.al-enterprise.com</AleUrl>
</ale>
<pubApp>
  <WatchDogEnabled>1</WatchDogEnabled>
  <EnableInAccess>0</EnableInAccess>
  <EnableOutAccess>0</EnableOutAccess>
</pubApp>
<record>
  <Enabled>1</Enabled>
  <VoiceCodec>G729</VoiceCodec>
  <RecordType>0</RecordType>
  <FileSizeLimit>8</FileSizeLimit>
  <ServerAddr>0.0.0.0</ServerAddr>
  <ServerPort>10000</ServerPort>
</record>
<background>
  <TimeoutToScreensaver>0</TimeoutToScreensaver>
  <UserChangeBackground>0</UserChangeBackground>
```

```
<UserChangeSubBackground></UserChangeSubBackground>  
<EHSHeadsettype>0</EHSHeadsettype>  
<MissedCallPopup>1</MissedCallPopup>  
<MWIPopup>1</MWIPopup>  
<DeviceConnectPopup>1</DeviceConnectPopup>  
<SMSPopup>0</SMSPopup>  
<OtherPopup>1</OtherPopup>  
<EnablePortMirror>0</EnablePortMirror>  
</background>  
</sysConf>
```


11.2 Description of the SIP Settings in configuration file

Below is the detail description for the parameters in configuration file:

Parameter	Options	Description	Configuration Item Path
<<VOIP CONFIG FILE>>Version:2.0000000000			
<NET CONFIG MODULE>			
WAN TYPE :0	0: ipv4 1: ipv6 2: ipv4&ipv6		Network>>Basic>>Network Mode>>IPv4 Only
WAN IP :192.168.1.179	IPv4 address	Static IPv4 address intended for this device on the network.	Network>>Basic>>IPv4 Settings>>Static IP>>IP
WAN Subnet Mask :255.255.255.0	IPv4 address	Subnet for the network that the phone is to communicate on.	Network>>Basic>>IPv4 Settings>>Static IP>>Subnet mask
WAN Gateway :192.168.1.1	IPv4 address	IPv4 address of the default gateway on the network.	Network>>Basic>>IPv4 Settings>>Static IP>>Default gateway
Domain Name :			Network>>Basic>>IPv4 Settings>>Static IP>>Primary DNS Server
Primary DNS :8.8.8.8	IPv4 address	IPv4 address of the primary DNS server	Network>>Basic>>IPv4 Settings>>Static IP>>Secondary DNS Server
Secondary DNS :202.96.134.133	IPv4 address	IPv4 address of the secondary DNS server	Network>>Basic>>IPv4 Settings>>Static IP>>DNS Domain
Enable DHCP :0	0: Static 1: DHCP	Set DHCP or Static	Network>>Basic>>IPv4 Settings>>Static IP/DHCP

DHCP Auto DNS :1	0: Disable 1: Enable	Get DNS address by DHCP server automatically,	Network>>Basic>>IPv4 Settings>>DNS Server Configured by
DHCP Auto Time :0	0: Disable 1: Enable	Getting time data from DHCP server.	Phone settings>>Time/Date>>Time Synchronized via DHCP
DHCP Option 100-101:1	0: Disable 1: Enable	DHCP Option 100 (POSIX Timezone) Time	
Use Vendor Class ID:0	0: Disable 1: Enable	Whether to enable manufacturers information.	Network>>Basic>>IPv4 Settings>>DHCP>>Enable Vendor Identifier
Vendor Class ID :Alcatel-Lucent H2P	Max 20 characters	Set the manufacturer information.	Network>>Basic>>IPv4 Settings>>DHCP>>Vendor Identifier
Enable PPPoE :0	0: Disable 1: Enable	Set phone to be PPPOE mode. *NOTE* Must match WAN_TYPE	Network>>Basic>>IPv4 Settings>>PPPoE
PPPoE User :user123	Max 60 characters	PPPoE username from ISP	Network>>Basic>>IPv4 Settings>>PPPoE Username
PPPoE Password :password	Max 60 characters	PPPoE password from ISP	Network>>Basic>>IPv4 Settings>>PPPoE Password
ARP Cache Life :2	Valid Value:0~99	Set ARP aging time. DO NOT change this value unless you know it exactly.	Network>>Advanced>>ARP Cache Life>>ARP Cache Life
WAN6 IP :	IPv6 address		Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>IP
WAN6 IP PREFIX :	Valid Value:0~128		Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>Prefix Length

WAN6 Gateway :	IPv6 address	IPv6 address of the default gateway on the network.	Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>Default gateway
Domain6 Name :	IPv6 address		Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>DNS Domain
Primary DNS6 :	IPv6 address		Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>Primary DNS Server
Secondary DNS6 :	IPv6 address		Network>>Basic>>IPv6 Only>>IPv6 Settings>>Static IP>>Secondary DNS Server
Enable DHCP6 :1	0: Disable 1: Enable		Network>>Basic>>IPv6 Only>>IPv6 Settings>>DHCP/Static IP
DHCP6 Auto DNS :1	0: Disable 1: Enable	Get DNS address by DHCP server automatically,	Network>>Basic>>IPv6 Only>>IPv6 Settings>>DHCP>>DNS Server Configured by
DHCP6 Auto Time :0	0: Disable 1: Enable	Getting time data from DHCP server.	Phone settings>>Time/Date>>Time Synchronized via DHCPv6
Use Vendor6 Class ID:0	0: Disable 1: Enable		Network>>Basic>>IPv6&IPv4>>DHCP>>Enable Vendor Identifier
Vendor6 Class ID :			Network>>Basic>>IPv6&IPv4>>DHCP>>Vendor Identifier
<MM CONFIG MODULE>			
ILBC Payload Type :97	valid value: 96~127	It configures the ILBC payload type.	Phone settings>>Media Settings>>Media Settings>>ILBC Payload Type

ILBC Payload Length: 20	20:20ms 30:30ms	It configures the ILBC Payload Length.	Phone settings>>Media Settings>>Media Settings>>ILBC Payload Length
G726-16 Payload Type:103		settings Media codes.	Phone settings>>Media Settings>>Codecs Settings
G726-24 Payload Type:104		settings Media codes.	Phone settings>>Media Settings>>Codecs Settings
G726-32 Payload Type:102		settings Media codes.	Phone settings>>Media Settings>>Codecs Settings
G726-40 Payload Type:105		settings Media codes.	Phone settings>>Media Settings>>Codecs Settings
Dtmf Payload Type :101	valid value: 96~127	It configures the DTMF payload type	Phone settings>>Media Settings>>Media Settings>>DTMF Payload Type
Opus Payload Type :107	valid value: 96~127	It configures the OPUS payload type	Phone settings>>Media Settings>>Media Settings>>OPUS Payload Type
Opus Sample Rate :0	0:OPUS-NB 1:OPUS-WB		Phone settings>>Media Settings>>Media Settings>>OPUS Sample Rate
VAD :0	0: Disable 1: Enable	It enables or disables the VAD (Voice Activity Detection) feature on the IP phone.	Phone settings>>Media Settings>>Media Settings>>Enable VAD
Resv Audio Band :0			
RTP Initial Port :10000	valid value: 1025~65530	the rtp port	Network>>Service Port>>RTP Port Range Start

RTP Port Quantity :1000	valid value: 1~1000	the RTP Port Quantity	Network>>Service Port>>RTP Port Range Quantity
RTP Keep Alive :0	0: Disable 1: Enable	When call hold, send rtp to stay connected	Phone settings>>Media Settings>>RTP Settings>>RTP Keep Alive
RTCP CNAME User :	Max 39 characters	It configures the CNAME username.	Phone settings>>Media Settings>>RTP Control Protocol Settings>>CNAME user
RTCP CNAME Host :	Max 39 characters	It configures the CNAME host.	Phone settings>>Media Settings>>RTP Control Protocol Settings>>CNAME host
Select Your Tone :11	300: custom 15: Australia 22: Austria 0: Belgium 16: Brazil 18: Canada 20: Chile 1: China 10: China Taiwan 17: Croatia 12: Czech 23: Denmark 24: Finland 25: France 2: Germany 26: Greece 27: Hungary 28: Lithuania 29: India 3: Israel 21: Italy 4: Japan 30: Mexico 31: New Zealand 5: Netherlands 6: Norway 32: Portugal 19: Russia 14: South Africa 7: South Korea 33: Spain 8: Sweden 9: Switzerland	Select Your Tone	Phone settings>>Tone>>Tone Settings>>Select Your Tone

	13: United Kingdom 11: United States		
Sidetone GAIN :1	valid value: 1~9		
Play Egress DTMF :0			
Dial Tone :350+440/0		Dial tone	Phone settings>>Tone>>Tone Settings>>Dial Tone
Ringback Tone :440+480/2000,0/4000		Ring back tone	Phone settings>>Tone>>Tone Settings>>Ring Back Tone
Busy Tone :480+620/500,0/500		Busy Tone	Phone settings>>Tone>>Tone Settings>>Busy Tone
Congestion Tone :		Congestion Tone	Phone settings>>Tone>>Tone Settings>>Congestion Tone

Call waiting Tone :440/300,0/10000,440/300,0/10000,0/0		It enables or disables the phone to play the call waiting tone when the phone receives an incoming call during a call.	Phone settings>>Tone>>Tone Settings>>Call waiting Tone
Holding Tone :		It enables or disables the phone to play the holding tone when the hold key of the phone is pressed during a call.	Phone settings>>Tone>>Tone Settings>>Holding Tone
Error Tone :		Error tone.	Phone settings>>Tone>>Tone Settings>>Error Tone
Stutter Tone :		Repeat tone.	Phone settings>>Tone>>Tone Settings>>Stutter Tone
Information Tone :		Information tone.	Phone settings>>Tone>>Tone Settings>>Information Tone
Dial Recall Tone :350+440/100,0/100,350+440/100,0/100,350+440/100,0/100,350+440/0		Callback tone.	Phone settings>>Tone>>Tone Settings>>Dial Recall Tone
Message Tone :		Message tone.	Phone settings>>Tone>>Tone Settings>>Message Tone
Howler Tone :		Howler tone.	Phone settings>>Tone>>Tone Settings>>Howler Tone
Number Unobtainable:400/500,0/6000		Number Unobtainable tone.	Phone settings>>Tone>>Tone Settings>>Number Unobtainable Tone
Warning Tone :1400/500,0/0		Warning tone.	Phone settings>>Tone>>Tone Settings>>Warning Tone
Record Tone :440/500,0/5000		Record tone.	Phone settings>>Tone>>Tone Settings>>Record Tone

Auto Answer Tone :		Auto answer tone.	Phone settings>>Tone>>Tone Settings>>Auto Answer Tone
--PHONE CONFIG-- :			
Audio Codec Sets :PCMU,PCMA,G726-16,G726-24,G726-32,G726-40,G729,iLBC,opus,G722		Select the corresponding code	
<SIP CONFIG MODULE>			
SIP Port :5060	valid value: 1~65535	The SIP port used by the phone.	Line>>SIP>>SIP Global Settings>>Local SIP Port
STUN Server :	stun server	stun server address	Line>>Basic Settings>>Server Address
STUN Port :3478	valid value: 1~65535		Line>>Advanced Settings>>Server Port
STUN Refresh Time :50	valid value: 31~65535		Line>>Basic Settings>>Binding Period
SIP Wait Stun Time :800	valid value: 300~9999999 (ms)		Line>>Basic Settings>>SIP Waiting Time
Extern NAT Addr :	external nat address		
Reg Fail Interval :32	valid value: 1~3600	The registration failure retries time, if the SIP account fails to register, the chance to register half of the retransmission time is registered until the registration is successful.	Line>>SIP>>SIP Global Settings>>Registration Failure Retry Time
SIP Pswd Encryption:0	0: Disable 1: Enable		
Strict BranchPrefix:0	0: Disable 1: Enable	Strictly match the Branch field.	Line>>SIP>>SIP Global Settings>>Strict Branch

Video Mute Attr :0	0:Sendrecv 1:Inactive		
Enable Group Backup:0	0: Disable 1: Enable	Enable SIP group server function as server backup	Line>>SIP>>SIP Global Settiings>>Enable Group
Enable RFC4475 :1	0: Disable 1: Enable	After enabling, strictly observe RFC4475	Line>>SIP>>SIP Global Settiings>>Enable RFC4475
Strict UA Match :1	0: Disable 1: Enable	Open a strict UA match and only accept requests from the server.	Line>>SIP>>SIP Global Settiings>>Enable Strict UA Match
CSTA Enable :0	0: Disable 1: Enable		Line>>SIP>>SIP Global Settiings>>Enable uaCSTA
Notify Reboot :0	0: Disable 1: Enable	Receive notify check - sync, don't judge directly restart reboot = true or false	
--SIP Line List-- :			
SIP1 Phone Number :	Max 39 characters	It configures the display name for account X.	Line>>SIP>>Register Settings>>Username
SIP1 Display Name :	Max 20characters	It configures the display name for account X.	Line>>SIP>>Register Settings>>Display name
SIP1 Sip Name :	Max 39characters	It configures the address of Domain Name	Line>>SIP>>Register Settings>>Realm
SIP1 Register Addr :	Max 39characters	It configures the SIP server.	Line>>SIP>>Register Settings>>Server Address
SIP1 Register Port :5060	valid value: 0~65535	It configures the port of the SIP server.	Line>>SIP>>Register Settings>>Server Port
SIP1 Register User :	String within 80 characters.	It configures the register user name for account X. Valid Value:	Line>>SIP>>Register Settings>>Authenticati on User
SIP1 Register Pswd :	Max 22characters	It configures the password for register authentication for account X.	Line>>SIP>>Register Settings>>Authenticati on Password

SIP1 Register TTL :3600	valid value: 1~65535	It configures the interval (in seconds) for the IP phone to retry to re-register account X when registration fails.	Line>>SIP>>Register Settings>>Registration Expiration
SIP1 Backup Addr :			Line>>SIP>>Register Settings
SIP1 Backup Port :5060			Line>>SIP>>Register Settings
SIP1 Backup Transport :0			
SIP1 Backup TTL :3600			
SIP1 Enable Reg :0	0: Disable 1: Enable	It enables or disables the account X.	
SIP1 Proxy Addr :	Max 39 characters	It configures the proxy server.	Line>>SIP>>Register Settings>>Proxy Server Address
SIP1 Proxy Port :5060	valid value: 0~65535	It configures the port of the proxy server.	Line>>SIP>>Register Settings>>Proxy Server Port
SIP1 Proxy User :	Max 80characters	It configures the Proxy user name for account X.	Line>>SIP>>Register Settings>>Proxy User
SIP1 Proxy Pswd :	Max 39characters	It configures the password for Proxy authentication for account X.	Line>>SIP>>Register Settings>>Proxy Password
SIP1 BakProxy Addr :	Max 39characters	It configures the IP address (or domain name) of the outbound proxy server 1 for account X.	Line>>SIP>>Register Settings>>Backup Proxy Server Address
SIP1 BakProxy Port :5060	valid value: 0~65535	It configures the port of the outbound proxy server 1 for account X.	Line>>SIP>>Register Settings>>Backup Proxy Server Port
SIP1 Enable Failback :0	0: Disable 1: Enable	When the main server is available, whether switch to the master server	Line>>SIP>>Basic Settings>>Enable Failback

SIP1 Failback Interval :1800		Probe the primary backup server availability interval	Line>>SIP>>Basic Settings>>Signal Failback
SIP1 Signal Failback :0	0: Disable 1: Enable	Allow multiple backup servers to switch	Line>>SIP>>Basic Settings>>Failback Interval
SIP1 Signal Retry Counts:3	The integers 1 to 10	Probe the number of times the primary backup server is available	Line>>SIP>>Basic Settings>>Signal Retry Counts
SIP1 Enable RFC5939 :0			
SIP1 Signal Crypto :0	0: Disable 1: Enable	When this configuration item is opened, sip communication encryption will be encrypted, and the sip packet re will not be captured in the call.	Line>>SIP>>Advance Settings>>SIP Encryption
SIP1 SigCrypto Key :	Max 40 characters		
SIP1 Media Crypto :0	0: Disable 1: Enable	RTP encryption. When this configuration is open, the voice of the phone is encrypted, the conversation is normal, and the phone is not heard by the grab bag.	Line>>SIP>>Advance Settings>>RTP Encryption
SIP1 MedCrypto Key :	Max 80 characters		
SIP1 SRTP Auth-Tag :0	0:auth-tag:AES-80, 1:auth-tag:AES-32		
SIP1 Local Domain :			
SIP1 Always FWD :0	0: Disable 1: Enable	It enables or disables the always forward feature for account X.	Line>>SIP>>Basic Settings>>Call Forward Unconditional
SIP1 Busy FWD :0	0: Disable 1: Enable	It enables or disables the busy forward feature for account X.	Line>>SIP>>Basic Settings>>Call Forward on Busy
SIP1 No Answer FWD :0	0: Disable 1: Enable	It enables or disables the no answer forward	Line>>SIP>>Basic Settings>>Call Forward on No Answer

		feature for account X.	
SIP1 Always FWD Num :	Max 30 characters	It configures the AlwaysFWD number for account X.	Line>>SIP>>Basic Settings>>Call Forward Number for Unconditional
SIP1 Busy FWD Num :	Max 30 characters	It configures the BusyFWD number for account X.	Line>>SIP>>Basic Settings>>Call Forward Number for Busy
SIP1 NoAnswer FWD Num :	Max 30 characters	It configures the NoAnswerFWD number for account X.	Line>>SIP>>Basic Settings>>Call Forward Number for No Answer
SIP1 FWD Timer :5	valid value: 0~120	It configures ring times (N) to wait before forwarding incoming calls.	Line>>SIP>>Basic Settings>>Call Forward Delay for No Answer
SIP1 Hotline Num :	0: Disable 1: Enable	It configures the hotline number that the IP phone automatically dials out when lifting the handset, pressing the speakerphone key or the line key. Leaving it blank disables hotline feature.	Line>>SIP>>Basic Settings>>Hotline Number
SIP1 Enable Hotline :0	Max 20 characters	It enables or disables the hotline feature.	Line>>SIP>>Basic Settings>>Enable Hotline
SIP1 WarmLine Time :0	valid value: 0~9		Line>>SIP>>Basic Settings>>Hotline Delay
SIP1 Pickup Num :	Max 39 characters	set Pickup Num	Line>>SIP>>Advance Settings>>PickUp Number
SIP1 Join Num :	Max 39 characters	set Join Num	Line>>SIP>>Advance Settings>>JoinCall Number
SIP1 Intercom Num :	Max 39 characters	set Intercom Num	Line>>SIP>>Advance Settings>>Intercom Number

SIP1 Ring Type :default	default,type 1~9	It sets the ringtone type of the current line.	Line>>SIP>>Advance Settings>> Ring Type
SIP1 NAT UDPUpdate :2	0: Disable 1: sip option 2: UDP		Line>>SIP>>Advance Settings>>Keep Alive Type
SIP1 UDPUpdate TTL :60	Max 6 characters	It sets the online cycle, in seconds.	Line>>SIP>>Advance Settings>>Keep Alive Interval
SIP1 Server Type :0	0: COMMON 1: NET2PHONE 6: BOTE 14: NORTEL 17: MITEL 22: MS_RP 23: CONFIG 24: FUGITSU 26: SOFTX3000 28: BroadSoft 29: Karel UCAP 30: Cellcom 31: 3CX 32: Fortinet	Select server type.	Line>>SIP>>Advance Settings>>Specific Server Type
SIP1 User Agent :	\$vendor \$model \$mac \$version \$date \$systemName	The user agent, which is set up, will take this value in the SIP package to distinguish which phone.	Line>>SIP>>Advance Settings>>User Agent
SIP1 PRACK :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>> Enable PRACK
SIP1 Keep AUTH :0	0: Disable 1: Enable	To be certified, When this configuration is open, it will register with the certificate information	Line>>SIP>>Advance Settings>>Keep Authentication
SIP1 Session Timer :0	0: Disable 1: Enable	When the call timer is enabled, when the configuration item is switched on, the phone periodically sends the message and terminates the call without a reply.	Line>>SIP>>Advance Settings>>Enable Session Timer

SIP1 S Timer Expires :0	valid value: 0~999999		Line>>SIP>>Advance Settings>>Session Timeout
SIP1 Enable GRUU :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Enable GRUU
SIP1 DTMF Mode :3	0:inband 1:RFC2833 2:SIP INFO 3:auto	It configures the DTMF type for account X.	Line>>SIP>>Basic Settings>>DTMF Type
SIP1 DTMF Info Mode :0	0:10/11 1:*#	SIP-Info * and # model.	Line>>SIP>>Basic Settings>>DTMF SIP INFO Mode
SIP1 NAT Type :0	0: Disable 1: Enable		Line>>SIP>>Basic Settings>>Use STUN
SIP1 Enable Rport :1	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Enable Rport
SIP1 Subscribe :0	0: Disable 1: Enable		Line>>SIP>>Basic Settings>>Subscribe For Voice Message
SIP1 Sub Expire :3600	valid value: 60~65535	It configures the voice mail number for account X.	Line>>SIP>>Basic Settings>>Voice Message Subscribe Period
SIP1 Single Codec :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Response Single Codec
SIP1 CLIR :0	0: Disable 1: RFC3323 2: RFC3325	Select anonymous call criteria.	Line>>SIP>>Advance Settings>>Anonymous Call Standard
SIP1 Strict Proxy :1	0: Disable 1: Enable	It is Compatible with special servers.	Line>>SIP>>Advance Settings>>Enable Strict Proxy
SIP1 Direct Contact :0	0: Communicate through the server 1: Direct communication via telephone		
SIP1 History Info :0	0: Disable 1: Enable		
SIP1 DNS SRV :0	abandoned		

SIP1 DNS Mode :0	0: DNS R, 1: DNS SRV, 2: DNS NAPTR	Select the DNS mode.	Line>>SIP>>Advance Settings>>DNS Mode
SIP1 XFER Expire :0	valid value: 0~60		
SIP1 Ban Anonymous :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Blocking Anonymous Call
SIP1 Dial Off Line :0	0: Disable 1: Enable	Allow not registered call	Line>>SIP>>Basic Settings>>Dial Without Registered
SIP1 Quota Name :0	0: Disable 1: Enable	Configure display name in quotes	Line>>SIP>>Advance Settings>>Use Quote in Display Name
SIP1 Presence Mode :0	0: Disable 1: Enable		
SIP1 RFC Ver :1	0:RFC2543 1:RFC3261	To select the SIP version you use, the phone needs to be configured RFC2543 to communicate properly with the SIP1.0 gateway.	Line>>SIP>>Advance Settings>>SIP Version
SIP1 Phone Port :0			
SIP1 Signal Port :5060			
SIP1 Transport :0	0: UDP 1: TCP 3: TLS	Select transfer protocol.	Line>>SIP>>Register Settings>>Transport Protocol
SIP1 Use SRV Mixer :0	0: local conference 1: server conference	It configures the network conference type for account X.	Line>>SIP>>Basic Settings>>Conference Type
SIP1 SRV Mixer Uri :	Max 20 characters	It configures the network conference URI for account X.	Line>>SIP>>Basic Settings>>Server Conference Number
SIP1 Long Contact :0	0: Disable 1: Enable	Long Contact field. After enable, configure the Contact field to carry more parameters.	Line>>SIP>>Advance Settings>>Enable Long Contact
SIP1 Auto TCP :0	0: Disable 1: Enable	Automatically using TCP transports, the TCP	Line>>SIP>>Advance Settings>>Auto TCP

		protocol transports are automatically used when the message body exceeds 1,300 bytes.	
SIP1 Uri Escaped :1	0: Disable 1: #->%23	Enable URI conversion Whether or not.	Line>>SIP>>Advance Settings>> Convert URI
SIP1 Click to Talk :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Enable Click To Talk
SIP1 MWI Num :	Max 20 characters		Line>>SIP>>Basic Settings>>Voice Message Number
SIP1 CallPark Num :	Max 20 characters	CallPark number. Keep the call to the configured number, record the number of voice broadcasts, and then use the number of other terminal call records to retrieve the call.	Line>>SIP>>Advance Settings>>CallPark Number
SIP1 Retrieve Num :	Max 32 characters		
SIP1 MSRPHelp Num :	Max 32 characters		
SIP1 User Is Phone:1			
SIP1 Auto Answer :0	0: Disable 1: Enable	It enables or disables the auto answer feature for account X	Line>>SIP>>Basic Settings>>Enable Auto Answering
SIP1 NoAnswerTime :5	valid value: 0~120	It configures ring times (N) to wait before forwarding incoming calls.	Line>>SIP>>Basic Settings>>Auto Answering Delay
SIP1 MissedCallLog :1	0: Disable 1: Enable	It enables or disables the IP phone to save the call log.	Line>>SIP>>Basic Settings>>Enable Missed Call Log
SIP1 SvcCode Mode :0	0: Disable 1: Enable	It enables or disables the IP phone to display the feature name instead of the feature access code when dialing and in talk	Line>>SIP>>Advance Settings>>Use Feature Code

SIP1 DNDOOn SvcCode :	Max 19 characters	It configures the DND on code to activate the server-side DND feature	Line>>SIP>>Advance Settings>>Enable DND
SIP1 DNDOff SvcCode :	Max 19 characters	It configures the DND off code to deactivate the server-side DND feature	Line>>SIP>>Advance Settings>> DND Disabled
SIP1 CFUOn SvcCode :	Max 19 characters	It configures the always forward on code to activate the server-side always forward feature.	Line>>SIP>>Advance Settings>>Enable Call Forward Unconditional
SIP1 CFUOff SvcCode :	Max 19 characters	It configures the always forward off code to deactivate the server-side always forward feature	Line>>SIP>>Advance Settings>>Disable Call Forward Unconditional
SIP1 CFBOn SvcCode :	Max 19 characters	It configures the busy forward on code to activate the server-side busy forward feature	Line>>SIP>>Advance Settings>>Enable Call Forward on Busy
SIP1 CFBOff SvcCode :	Max 19 characters	It configures the busy forward off code to deactivate the server-side busy forward feature.	Line>>SIP>>Advance Settings>>Disable Call Forward on Busy
SIP1 CFNOn SvcCode :	Max 19 characters	It configures the no answer forward on code to activate the server-side no answer forward feature	Line>>SIP>>Advance Settings>>Enable Call Forward on No Answer
SIP1 CFNOff SvcCode :	Max 19 characters	It configures the no answer forward off code to deactivate the server-side no answer forward feature	Line>>SIP>>Advance Settings>> Disable Call Forward on No Answer
SIP1 ANCOOn SvcCode :	Max 19 characters	It configures the Blocking Anonymous Call On code to activate the server-side Blocking Anonymous Call feature	Line>>SIP>>Advance Settings>>Enable Blocking Anonymous Call

SIP1 ANCOff SvcCode :	Max 19 characters	It configures the Blocking Anonymous Call Off code deactivate the server-side Blocking Anonymous Call feature	Line>>SIP>>Advance Settings>>Disable Blocking Anonymous Call
SIP1 Send ANOn Code :	Max 19 characters	It configures the Anonymous Call On code activate the server-side Anonymous Call feature	Line>>SIP>>Advance Settings>>Send Anonymous On Code
SIP1 Send ANOffCode :	Max 19 characters	It configures the Anonymous Call Off code deactivate the server-side Anonymous Call feature	Line>>SIP>>Advance Settings>>Send Anonymous Off Code
SIP1 CW On Code :	Max 19 characters	It configures the call waiting on code to activate the server-side call waiting feature	Line>>SIP>>Advance Settings>>Call Waiting On Code
SIP1 CW Off Code :	Max 19 characters	It configures the Call Waiting Off code to deactivate the server-side call waiting feature	Line>>SIP>>Advance Settings>> Call Waiting Off Code
SIP1 VoiceCodecMap :PCMU,PCMA,G726-32,G729,G723,iLBC,AMR,G722,AMR-WB		It enables or disables the specified codec for account X.	Line>>SIP>>Codecs Settings
SIP1 VideoCodecMap :		set Video Codec	Line>>SIP>>Video Settings
SIP1 BLFList Uri :	Max 50 characters	Set the BLF List number and bind the BLF List of this number to the DSSKEY.	Line>>SIP>>Advance Settings>>BLF List Number
SIP1 BLF Server :	Max 39 characters	The BLF server is used in conjunction with the BLF list.	Line>>SIP>>Advance Settings>>BLF Server
SIP1 Respond 182 :0	0: Disable 1: Enable	Set the call waiting response code (182), enable, and then return to the 182 bell.	Line>>SIP>>Advance Settings>>Use 182 Response for Call waiting

SIP1 Enable BLFList :0	0: Disable 1: Enable	Open the BLF List function.	Line>>SIP>>Advance Settings>>Enable BLF List
SIP1 Caller Id Type :4	1:FROM 2:PAI-FROM 3:RPID-FROM 4:PAI-RPID-FROM 5:RPID-PAI-FROM	Set the caller display header field.	Line>>SIP>>Advance Settings>>Caller ID Header
SIP1 Syn Clock Time :0	0: Disable 1: Enable	Synchronous machine time, enable, synchronous server time	Line>>SIP>>Advance Settings>>Enable Feature Sync
SIP1 Use VPN :1	0: Disable 1: Enable	It configures the OpenVPN feature	Line>>SIP>>Basic Settings>>Use VPN
SIP1 Enable DND :0	0: Disable 1: Enable	It allows IP phones to ignore incoming calls	Line>>SIP>>Basic Settings>>Enable DND
SIP1 Inactive Hold :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Enable Use Inactive Hold
SIP1 Req With Port :1	0: Disable 1: Enable	Whether in the Request uri add port information	
SIP1 Update Reg Expire :1	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Server Expire
SIP1 Enable SCA :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>> Enable SCA
SIP1 Sub CallPark :0	0: Disable 1: Enable		
SIP1 Sub CC Status :0	0: Disable 1: Enable		
SIP1 Feature Sync :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Enable Feature Sync
SIP1 Enable XferBack :0	0: Disable 1: Enable	Received transfer calls, whether the timeout time transferred calls to transferors	
SIP1 XferBack Time :35	valid value: 0~60		

SIP1 Use Tel Call :0	0: Disable 1: Enable		Line>>SIP>>Advance Settings>>Use Tel Call
SIP1 Enable Preview :0	0: Disable 1: Enable		
SIP1 Preview Mode :1	0:18x 1:2xx		
SIP1 TLS Version :0	0: TLS 1.0 1: TLS 1.1 2: TLS 1.2	TLS Version	Line>>SIP>>Advance Settings>>TLS Version
SIP1 CSTA Number :	Max 20 characters		Line>>SIP>>Advance Settings>> uaCSTA Number
SIP1 Enable ChgPort :0	0: Disable 1: Enable	The phone will change the port registration when the registration fails	Line>>SIP>>Advance Settings>> Enable Chgport
SIP1 VQ Name :	Max 48 characters	It configures PUBLITH report name	Line>>SIP>>Advance Settings>>VQ Name
SIP1 VQ Server :	Max 48 characters	It configures PUBLITH report Server	Line>>SIP>>Advance Settings>>VQ Server
SIP1 VQ Server Port :5060	valid value: 0~65535	It configures PUBLITH report Server Port	Line>>SIP>>Advance Settings>>VQ Server Port
SIP1 VQ HTTP Server :	Max 96 characters	It configures PUBLITH report HTTP Server	Line>>SIP>>Advance Settings>>VQ Http/Https server
SIP1 Flash Mode :0	0: normal 1: sip info	Phone whether to send custom SIP Info messages or not when beating Plug spring.	Line>>SIP>>Advance Settings>>Flash Mode
SIP1 Content Type :	Max 63 characters	The Content Type of SIP Info by phone sending When beating Plug spring.	Line>>SIP>>Advance Settings>>Flash Info Content-Type
SIP1 Content Body :	Max 127 characters	The Content Body of SIP Info by phone sending When beating Plug spring.	Line>>SIP>>Advance Settings>>Flash Info Content-Body
SIP1 Unregister On Boot :0	0: Disable 1: Enable	Whether to cancel the registration when the telephone is restarted.	Line>>SIP>>Advance Settings>>Unregister On Boot

SIP1 Enable MAC Header :0	0: Disable 1: Enable	Whether or not to bring a Mac field in the sip message	Line>>SIP>>Advance Settings>>Enable MAC Header
SIP1 Record Start :Record:on	Recording server	Select SIP INFO type to record. At the beginning of recording, the field in the SIP INFO message sent by the telephone is recorded.	
SIP1 Record Stop :Record:off	Recording server	Select SIP INFO type to record. At the end of recording, the field in the SIP INFO message sent by the telephone.	
Transaction TimerT1:500	valid value: 500~10000: ms		
Transaction TimerT2:4000	valid value: 2000~40000: ms		
Transaction TimerT4:5000	valid value: 2500~60000: ms		
SIP1 Unavailable Mode :0	0 500,503 1 4xx,5xx,6xx		
SIP1 BLF Dialog Match :1	1 4xx,5xx,6xx	When subscribing to BLF, notify issued by the server carries multiple Dialog Ids, disable BLF Dialog Strict Match, and enable it when carrying one.	
SIP1 Ptime :0		Whether or not the ptime field is carried in the Invite message.	
SIP1 Session Timer T1 :500	valid value: 500~10000 : ms		
SIP1 Session Timer T2 :4000	valid value: 2000~40000: ms		
SIP1 Session Timer T4 :5000	valid value: 2500~60000: ms		

SIP1 Unavailable Mode :0	0 500,503 1 4xx,5xx,6xx		
<CALL FEATURE MODULE>			
--Port Config-- :			
P1 Enable XferDPlan :1	0: Disable 1: Enable	It configures whether to use the Configured Digital Plan rule when transferring.	
P1 Enable FwdDPlan :0	0: Disable 1: Enable	It configures whether to use the Configured Digital Plan rule when forwarding.	
P1 Enable Pre DPlan :0	0: Disable 1: Enable	It configures whether to use the Configured Digital Plan rule when predictive dialer.	
P1 IP Dial Prefix :		Configure the prefix for IP dialing. For example, when configured as "172.16.7.", the phone dials # 165 # and calls out 172.16.7.165.	Phone settings>>Feature>>Basic settings>>P2P IP Prefix
P1 Enable DND :1	0: Disable 1: Enable	Allow DND functionality	Line>>SIP>>Basic settings>>Enable DND
P1 DND Mode :0	0: Disable 1: Phone 2: Line	when enable Enable DND ,and set DND Mode to 1 or 2,the phone will open DND.	
P1 Enable Space DND :0	0: Disable 1: Phone	Enable DND Timing Function.	
P1 DND Start Time :1500		Set the time to open DND.	Phone settings>>Feature>>DND Settings>>DND Start Time
P1 DND End Time :1730		Set the time to close DND.	Phone settings>>Feature>>DND Settings>>DND End Time

P1 Enable White List :1	0: Disable 1: Enable	Enable White List Function.	Phone settings>>Feature>>Basic settings>>Enable Restricted Incoming List
P1 Enable Black List :1	0: Disable 1: Enable	Enable Black ListFunction.	Phone settings>>Feature>>Basic settings>>Enable Restricted Outgoing List
P1 Enable CallBar :1	0: Disable 1: Enable	Allow the function of setting a list that prohibits incoming calls.	Phone settings>>Feature>>Basic settings>>Enable Allowed Incoming List
P1 Mute Ringing :0	0: Disable 1: Enable	It has the function of silencing the bell when calling.	Phone settings>>Feature>>Basic settings>>Enable Silent Mode
P1 Ban Dial Out :0	0: Disable 1: Enable	No exhaling numbers.	Phone settings>>Feature>>Basic settings>>Ban Outgoing
P1 Ban Empty CID :0	0: Disable 1: Enable	It will reject calls that CID is empty.	
P1 Accept Any Call :1	0: Disable 1: Enable	IP calls cannot be answered after shutdown.	
P1 Enable CLIP :1	0: Disable 1: Enable		
P1 CallWaiting :1	0: Disable 1: Enable	It configures whether the call waiting function is enabled, and when enabled, the phone can receive multiple calls.	Phone settings>>Feature>>Basic settings>>Enable Call Waiting
P1 CallTransfer :1	0: Disable 1: Enable	It configures whether the transfer function is enabled.	Phone settings>>Feature>>Basic settings>>Enable Call Transfer
P1 CallSemiXfer :1	0: Disable 1: Enable	It configures whether the semi-attendance transfer function is enabled.	Phone settings>>Feature>>Basic settings>>Semi-Attended Transfer
P1 CallConference :1	0: Disable 1: Enable	It configures whether conference functions are enabled.	Phone settings>>Feature>>Ba

			Phone settings>>Enable 3-way Conference
P1 Auto PickupNext :0			
P1 Busy No Line :1			
P1 Auto Onhook :1	0: Disable 1: Enable	It configures whether the automatic hang-up function is enabled.	Phone settings>>Feature>>Basic settings>>Enable Auto on Hook
P1 Auto Onhook Time :3	valid value: 0~30	It configures the automatic hang-up time.	Phone settings>>Feature>>Basic settings>> Auto HangUp Delay
P1 Enable Intercom :1	0: Disable 1: Enable	It configures whether to receive Intercom call.	Phone settings>>Feature>>Intercom Settings>>Enable Intercom
P1 Intercom Mute :0	0: Disable 1: Enable	It configures whether Intercom calls are silent.	Phone settings>>Feature>>Intercom Settings>>Enable Intercom Mute
P1 Intercom Tone :1	0: Disable 1: Enable	It configures whether there is a prompt tone when an Intercom call is made.	Phone settings>>Feature>>Intercom Settings>>Enable Intercom Tone
P1 Intercom Barge :1	0: Disable 1: Enable	It configures whether to answer an Intercom call when there is already a call.	Phone settings>>Feature>>Intercom Settings>>Enable Intercom Barge
P1 Use Auto Redial :0	0: Disable 1: Enable	It configures whether to turn on the auto redial function. After enable, when the phone calls and replies 486 to the other end, the phone auto redial according to the set number and time interval.	
P1 Redial EnterCallLog :0	0: Disable 1: Enable	Redial enter CallLogs	Phone settings>>Feature>>Redial Settings>>Redial Enter CallLog

P1 AutoRedial Delay :30		It configures the time interval for auto redial.	
P1 AutoRedial Times :5		It configures the times for auto redial.	
P1 Call Complete :0	0: Disable 1: Enable	It configures whether to turn on the call complete function. After enable, when the phone calls and the other end replies 486, the phone will redial after the other end not busy.	
P1 CHolding Tone :1	0: Disable 1: Enable	It configures whether to play the prompt tone when hold calls.	Phone settings>>Feature>>Tone Settings>>Enable Holding Tone
P1 CWaiting Tone :1	0: Disable 1: Enable	It configures whether the prompt sound is played while the call is waiting.	Phone settings>>Feature>>Tone Settings>>Enable Call Waiting Tone
P1 Hide DTMF Type :0	0:disable 1:all 2:delay 3:last show	It configures the call by pressing the DTMF key to hide or delay the display.	Phone settings>>Feature>>Basic Settings>>Hide DTMF
P1 Talk DTMF Tone :1	0: Disable 1: Enable	It configures whether to play the prompt sound by pressing the DTMF key when talking.	Phone settings>>Feature>>Tone Settings>>Enable Holding Tone
P1 Dial DTMF Tone :1	0: Disable 1: Enable	It configures whether to play prompt tone by pressing DTMF key when dialing.	Phone settings>>Feature>>Tone Settings>>Play Dialing DTMF Tone
P1 Psw Dial Mode :0	0: Disable 1: Enable	It configures whether to turn on password dialing. When turned on, it encrypts the corresponding portion of the display number according to the password prefix and length of the configuration.	Phone settings>>Feature>>Password Dial Settings>>Enable Password Dial
P1 Psw Dial Length :0	valid value: 0~31	It configures the length of the password dial.	Phone settings>>Feature>>Password Dial Settings>>Encryption Number Length

P1 Psw Dial Prefix : :	Max 31 characters	It configures the prefix for password dialing.	Phone settings>>Feature>>Password Dial Settings>>Password Dial Prefix
P1 Enable MultiLine :1	0: Disable 1: Enable	Up to 10 calls can be set up after enable it, and up to two calls can be set up after disable.	Phone settings>>Feature>>Basic settings>>Enable Multi Line
P1 Allow IP Call :1	0: Disable 1: Enable	It configures whether to receive IP calls.	Phone settings>>Feature>>Basic settings>>Allow IP Call
P1 Caller Name Type :0		The phone displays the name according to the priority set by Caller Name Priority.	Phone settings>>Feature>>Basic settings>>Caller Name Priority
P1 Mute For Ring :0	0: Disable 1: Enable	It configures whether the ringtone is silent when the call is made.	Phone settings>>Feature>>Basic settings>>Disable Mute for Ring
P1 Auto Handle Video :1			
P1 Default Ans Mode :2	1: audio 2: video	Set answer mode,Default video answer the call	Phone settings>>Feature>>Basic settings>>Default Ans Mode
P1 Default Dial Mode :2	1: audio 2: video	Set dial mode,Default video dial call	Phone settings>>Feature>>Basic settings>>Default Dial Mode
P1 Hold To Transfer :0	0: Disable 1: Enable	Whether it can transfer calls when it configures Hold.	
P1 Enable PreDial :0	0: Disable 1: Enable	It configures whether pre-dialing is allowed. After closing, the standby input number enters the dial directly, and after opening, the standby input number enters the pre-dialing interface.	Phone settings>>Feature>>Basic settings>>Enable Pre-Dial
P1 Default Ext Line :1	0: Disable 1: Enable	It configures default SIP lines.	Phone settings>>Feature>>Basic settings>>Default Ext Line

P1 Enable Def Line :1	0: Disable 1: Enable	It configures whether to turn on the default line.	Phone settings>>Feature>>Basic settings>>Enable Default Line
P1 Enable SelLine :1		It configures whether to turn on the default line.	Phone settings>>Feature>>Basic settings>>Enable Auto Switch Line
P1 Ring in Headset :0	0: disable 1: enable 2: group ring	Enable ring headset,it will be a hint“headset is not plugged in”, when Headset is not inserted. Enable group ring, it will be Speaker and headset or handle and headset ring at the same time	Phone settings>>Feature>>Basic settings>>Ring From Headset
P1 Auto Headset :0	0: Disable 1: Enable	It is equipped to press the answering button when the phone connects to the headset, and whether to use the headset to answer automatically when there is an incoming call.	Phone settings>>Feature>>Basic settings>>Enable Auto Headset
P1 DND Return Code :480	0:404 1:480 2:486 3:603	dnd return code	Phone settings>>Feature>>Response Code Settings>>DND Response Code
P1 Busy Return Code :486	0:404 1:480 2:486 3:603	busy return code	Phone settings>>Feature>>Response Code Settings>>Busy Response Code
P1 Reject Return Code :603	0:404 1:480 2:486 3:603	reject return code	Phone settings>>Feature>>Basic Settings>>Reject Response Code
P1 Contact Type :0	0:None 1:BOTH 2:DND White List 3:FWD White List	It configures contacts as the type of whitelist. DND White List takes effect only when DND is turned on, and FWD White List takes effect only when forwarding. Both works in both cases.	

P1 Enable Country Code:0		It configures whether to turn on the function of identifying country code.After enable it, phone can identify the number with the country code and Area Code.	Phone settings>>Feature>>Basic Settings>>Enable Country Code
P1 Country Code :		It configures the Country Code.	Phone settings>>Feature>>Basic Settings>>Country Code
P1 Call Area Code :		It configures the Area Code.	Phone settings>>Feature>>Basic Settings>>Area Code
P1 Number Privacy :0	0: Disable 1: Enable	It is configured to enable encrypted display of caller numbers.	Phone settings>>Feature>>Basic Settings>>Enable Number Privacy
P1 Privacy Rule :		It configures the rules for encrypting caller numbers.	Phone settings>>Feature>>Basic Settings>>Match Direction
P1 Transf DTMF Code :		Press the transfer key ,the phone will send dtmf you set.	
P1 Hold DTMF Code :		Press the hold key ,the phone will send dtmf you set.	
P1 Conf DTMF Code :		Press the conference key ,the phone will send dtmf you set.	
P1 Disable Dial Search :1	0: Disable 1: Enable	It configures whether enable the search function when dialing.	
--Basic DialPlan-- :			

Dial by Pound :1	0: Disable 1: Enable	Press the # key to dial out the number, for example, 123 # phone will call 123	Line>>Dial Plan>>Basic Settings>>Press # to invoke dialing
BTransfer by Pound :0	0: Disable 1: Enable	Enter the number in the transfer dial, then enter # ,the phone will blind transfer.	Line>>Dial Plan>>Basic Settings>>Press # to Do Blind Transfer
Onhook to BXfer :0	0: Disable 1: Enable	After the transfer of the dial number, Onhook is blind transfer.	Line>>Dial Plan>>Basic Settings>>Blind Transfer on Onhook
Onhook to AXfer :0	0: Disable 1: Enable	After the transfer of the dial number, Onhook is Attend transfer.	Line>>Dial Plan>>Basic Settings>>Attended Transfer on Onhook
Conf Onhook to Xfer:0	0: Disable 1: Enable	After the conference call, Onhook is attend transfer	Line>>Dial Plan>>Basic Settings>>Attended Transfer on Conference Onhook
Dial Fixed Length :0	0: Disable 1: Enable	When the length of the input number reaches the value of the fixed length, it automatically call the number.	Line>>Dial Plan>>Basic Settings>>Dial Fixed Length
Fixed Length Nums :11	valid value: 1~32	the length of numbers	

Dial by Timeout :1	0: Disable 1: Enable	Input number,when timeout and the phone will automatically call the number.	
Dial Timeout value :10	valid value: 3~30	the timeout value	Line>>Dial Plan>>Basic Settings>>Send after
Enable E OneSixFour:0	0: Disable 1: Enable	It configures to enable the E.164 rule.	Line>>Dial Plan>>Basic Settings>>Enable E.164
--Alert Info Ring--:			
Alert1 Text :	Max 63 characters	It configures Alert Text.	
Alert1 Ring Type :Type 1	valid value: Type1~9	It configures Ring Type.When the text of incoming call is same with Alert Text, it will play the corresponding bells.	
<PHONE FEATURE MODULE>			
Menu Password :123646	Max 6 characters	It configures the password for unlocking the menu, not support""	
KeyLock Password :123646	Max 6 characters	It configures the password for unlocking the phone	

Fast Keylock Code :		It is configured to quickly lock the keyboard code, in the pre-dial interface, enter this number will quickly lock the keyboard.	
Enable KeyLock :0	0: Disable 1: Enable	It configures whether to enable key lock	
KeyLock Timeout :0		It configures the lock standby time.	
KeyLock Status :0	0: Disable 1: Enable	It is equipped with keyboard lock master switch. When it closes, neither the keyboard lock nor the long press # can be unlocked.	
Emergency Call :110		It configures emergency call number. After the keyboard is locked, only the emergency call number can be called.	Phone settings>>Feature>>Basic settings>>Emergency Call Number
Push XML IP :		Set the server address for pushing XML files to the terminal	Phone settings>>Feature>>Basic settings>>Push XML Server
SIP Number Plan :0			
LDAP Search :0	0: Disable 1: Enable	when enable it, you call choose ldap that you want.	Phone settings>>Feature>>Basic settings>>LDAP Search
Search Path :0		select contacts Search range.	Phone settings>>Feature>>Basic settings>>Search path
Caller Display T :0	0: Display name priority 1: Display number only 4:Display Blank 5:Normal	It configures the type of name and number to be displayed during a call.	Phone settings>>Feature>>Basic settings>>Contact As White List Type
CallLog DisplayType :0		It configures the type of call logs.	
Enable Recv SMS :1	0: Disable 1: Enable	It configures whether to accept SMS messages.	

Enable Call History:1	0: Disable 1: Enable	If the value is 1, the call history is recorded in the terminal's call record; if the value is 0, the call history is not displayed in the terminal's call record. Default 1, enabled	Phone settings>>Feature>>Basic settings>> Enable CallLog
Line Display Format :\$name@\$protocol\$instance		It configures the display format of registered Line on the phone.	Phone settings>>Feature>>Basic settings>> Line Display Format
Enable MWI Tone :0	0: Disable 1: Enable		
All Pswd Encryption:0	0: Disable 1: Enable	It configures all password encryption in the configuration file.	
SIP Notify XML :1	0: Disable 1: Enable	when enabled, when the phone receives relevant notify content, the corresponding information will be displayed.	
Block XML When Call:1	0: Disable 1: Enable	Blocked Push XML When Call	
XML Update Interval:0		Configure the time interval for XML updates.	
--Display Input-- :			
LCD Title :VOIP PHONE		Configuration shows welcome words in the upper left corner of the phone.	Phone settings>>Advance>>Greeting Words>>Greeting Words
LCD Constrast :5		Set the LCD Contrast level of the phone.	
Enable Energysaving:0	0~16	It configures the Backlight Inactive Level.	Phone settings>>Advance>>Greeting Words>>Backlight Inactive Level
LCD Luminance Level:16	1~16	It configures the Backlight active Level.	Phone settings>>Advance>>Greeting

			Words>>Backlight Active Level
Backlight Off Time :30	0~120second	It configures the delay time (in seconds) to change the intensity of the LCD screen when the IP phone is inactive.	Phone settings>>Advance>>Greeting Words>>Backlight Time
Disable CHN IME :0			
Phone Model :		It shows the model of the telephone.	
Host Name :dvf97		Hostname is a kernel parameter under Linux system. The hostname of X6 is the user name of Linux system by default.	
Default Language :en		It configures the language displayed on the phone.	
Enable Greetings :0			
--Power LED-- :			
Power :0	0: Disable 1: Enable	the default state of power led.	
MWI Or SMS :2	0: Disable 1: Enable 2: slow blink 3: fast blink	Set the state of the power light when there is a MWI	Phone settings>>Feature>>Power LED>> SMS/MWI
In Using :0	0: Disable 1: Enable	settings the power led state when talking	Phone settings>>Feature>>Power LED>>Talk/Dial
Ring :0	0: Disable 1: Enable 2: slow blink 3: fast blink	settings power led state when the phone ring	Phone settings>>Feature>>Power LED>>Ringing
Hold :0	0: Disable 1: Enable 2: slow blink 3: fast blink	settings the power led state when the call hold	Phone settings>>Feature>>Power LED>>Hold/Held

Mute :0	0: Disable 1: Enable 2: slow blink 3: fast blink	settings the power led state when talking was mute	Phone settings>>Feature>>Power LED>>Mute
Missed Call :0	0: Disable 1: Enable 2: slow blink 3: fast blink	Set the state of the power light when there is a Missed call	Phone settings>>Feature>>Power LED>>Missed call
--Voice Volume-- :			
Handset Vol :3	valid value: 1~9	Configure the volume of the call when using the handset.	Phone settings>>Media Settings>>Handset Volume
Handset Mic Vol :3		It is equipped with a handset microphone gain.	
Headset Vol :3	valid value: 1~9	It is configured to use the volume of a headset call.	Phone settings>>Media Settings>>Headset Volume
Headset Mic Vol :3	valid value: 1~9	It configures headphone MIC gain.	Phone settings>>Media Settings>>Headset Mic Gain
Headset Ring Vol :5	valid value: 0~9	The volume of the bell that rings through the earphone.	Phone settings>>Media Settings>>Headset Ring Volume
HandFree Vol :7	valid value: 1~9	Configure the volume of hands-free calls.	Phone settings>>Media Settings>>Speakerphone Volume
HandFree Mic Vol :3		It configures handfree MIC gain.	
HandFree Ring Vol :5	valid value: 0~9	The volume of the bell that rings through the handfree.	
Ring Type :ring00.wav	valid value: Type 1~9	Default ring tone type.	Phone settings>>Media Settings>>Media

			Settings>>Default Ring Type
--DateTime Config--:			
Enable SNTP :1	0: Disable 1: Enable	sync time by sntp	Phone settings>>Time/Date>>Network Time Server Settings>>Time Synchronized via SNTP
SNTP Server :0.pool.ntp.org	sntp server address	sntp server address	Phone settings>>Time/Date>>Network Time Server Settings>>Primary Time Server
Second SNTP Server :time.nist.gov	sntp server address	Backup sntp server address	Phone settings>>Time/Date>>Network Time Server Settings>>Secondary Time Server
Time Zone :32	valid value: (-44) ~ (56)	the number represents different time zone, follow the time zone name changes	Phone settings>>Time/Date>>Network Time Server Settings>>Time zone
Time Zone Name :UTC+8	valid value: (UTC-11) ~ (UTC+14)	the time zone	Phone settings>>Time/Date>>Network Time Server Settings>>Time zone
SNTP Timeout :60	valid value: 60~9999	Set the time for SNTP time synchronization.	Phone settings>>Time/Date>>Network Time Server Settings>>Resync Period
DST Type :0	0: Disable 1: automatic 2: manual	the type of using DST	Phone settings>>Time/Date>>Daylight Saving Time Settings>>DST Set Type
DST Location :0		the country of Daylight Saving Time	Phone settings>>Time/Date>>Time/Date Format
DST Rule Mode :0	0: Disable 1: by date 2: by week	the fixed type of manual.	Phone settings>>Time/Date>>Time/Date Format

DST Min Offset :60	valid value: 1~1440	the time offset of DST	Phone settings>>Time/Date>>Time/Date Format
DST Start Mon :3	valid value: 1~12	the start month of DST	Phone settings>>Time/Date>>Time/Date Format
DST Start Week :5	valid value: 1~7	the start week of DST	Phone settings>>Time/Date>>Time/Date Format
DST Start Wday :0	valid value: 0~6	the start Wday of DST	Phone settings>>Time/Date>>Time/Date Format
DST Start Hour :2	valid value: 0~23	the start hour of DST	Phone settings>>Time/Date>>Time/Date Format
DST End Mon :10	valid value: 1~12	the end month of DST	Phone settings>>Time/Date>>Time/Date Format
DST End Week :5	valid value: 1~7	the end week of DST	Phone settings>>Time/Date>>Time/Date Format
DST End Wday :0	valid value: 0~6	the end wday of DST	Phone settings>>Time/Date>>Time/Date Format
DST End Hour :2	valid value: 0~23	the end hour of DST	Phone settings>>Time/Date>>Time/Date Format
--DateTime Display--:			
Enable TimeDisplay :0			
Time Display Style :0	0:12 1:24	Configuration time shows 12-hour clock or 24-hour clock.	
Date Display Style :0	0:DD MMM WW 1: MMM DD WW 2: WW DD MMM 3: WW MMM DD 4: DD MM YY 5: DD MM YYYY 6: MM DD YY 7: MM DD YYYY 8: YY MM DD 9: YYYY MM DD	Configure the type of date display.	

Date Separator :0	0:/ 1:- 2:. 3:space	Configure the interval symbol for the time display.	
--ScreenSaver Config-- :			
Screen Saver Type :0	0:Disable 1:Lcd Power off 2:Start screen saver	Configure the type of screen saver.	
Screen Timeout :0		Timeout screen saver into the interface	
Enable ActivePeriod:0			
Period One Start :0			
Period One End Time:0			
Period Two Start :0			
Period Two End Time:0			
Screen Saver App :		App icon displayed on screen saver interface	
Sleep After Active :0			
Sleep Timeout :0		The time to sleep after Screen Saver Type set to LCD Power off	
--Softkey Config-- :			
Softkey Mode :1		It shows setting the display mode of softkey when the number of choices is greater than 4. More, the last softkey is more, press the more key to switch to display softkey	
SoftKey Exit Style :2		Exit key location Settings.	
Desktop Softkey :menu;dnd;history;dss1;headset;		It configures the desktop SoftKey.	Function key>>Softkey>>Softkey settings>>Screen
Talking Softkey :hold;xfer;conf;end;headset;		It configures the call interface SoftKey.	Function key>>Softkey>>Softkey settings>>Screen

Ringing Softkey :accept;none;forward;reject;headset;		It configures the softkey for incoming call.	Function key>>Softkey>>Softkey settings>>Screen
Alerting Softkey :end;none;none;none;headset;		It configures the softkey for outgoing call.	
XAlerting Softkey :end;none;none;xfer;		It configures the softkey when transferring calls.	
Conference Softkey :hold;none;split;end;headset;		It configures the softkey for conference.	Function key>>Softkey>>Softkey settings>>Screen
Waiting Softkey :xfer;accept;reject;end;headset ;		It configures the softkey of the call waiting interface.	Function key>>Softkey>>Softkey settings>>Screen
Ending Softkey :repeat;none;none;end;		It configures the softkey for ending a call.	Function key>>Softkey>>Softkey settings>>Screen
DialerPre Softkey :send;2aB;delete;exit;headset;		It configures the softkey for pre-dial call.	Function key>>Softkey>>Softkey settings>>Screen
DialerCall Softkey :send;2aB;delete;exit;headset;		It configures the softkey for dialer interface.	Function key>>Softkey>>Softkey settings>>Screen
DialerXfer Softkey :delete;xfer;send;exit;		It configures the softkey for transfer Dial-up Interface.	Function key>>Softkey>>Softkey settings>>Screen
DialerCfwd Softkey :send;2aB;delete;exit;		It configures the forward dial-up interface's softkey.	Function key>>Softkey>>Softkey settings>>Screen
Desktop Click :history;none;none;none;status;		Configure the function of pressing navigation keys on standby desktop. They correspond to the Up, Down, Left, Right and OK keys, respectively.	Function key>>Softkey>>Softkey settings>>Screen
Dailer Click :pline;nline;none;none;none;		Configure the function of pressing navigation keys on dialer Interface. They correspond to the Up, Down, Left, Right and OK keys, respectively.	Function key>>Softkey>>Softkey settings>>Screen
Ringing Click :none;none;none;none;none;		Configure the function of pressing navigation keys on call Interface. They correspond to the Up,	

		Down, Left, Right and OK keys, respectively.	
Call Click :none;none;none;none;none;		Configure the function of long press navigation key on standby desktop. They correspond to the Up, Down, Left, Right and OK keys, respectively.	
Desktop Long Press :status;none;none;none;reset;			
DialerConf Softkey :contact;clogs;redial;video;cancel ;		It configures the softkey of the conference dial-up interface.	
-- Agent Config-- :			
Agent Username :		It configures the Agent Username.	
Agent Password :		It configures the Agent Password.	
Agent Number :		It configures the Agent Number.	
Agent Sipline :0	Valid Value:0~5	It configures the Agent corresponding lines.	
Agent Status :0	0:logon 1: logoff 2: UNAVAILABLE 3: AVAILABLE 4: WRAPUP	It configures the Agent Status.	
Agent Status Reason:			
Agent Clear CallLog:0		Whether to clear all call records of agent account when log off it.	
--BW Directory-- :			
BWDir1 Title :		It configures the Broadsoft phonebook name.	Phonebook>>Cloud phonebook>>Broadsoft Directory Settings>>Display Title

BWDir1 URL :		It configures the Broadsoft phonebook URL.	Phonebook>>Cloud phonebook>>Broadsoft Directory Settings>>Server Address
BWDir1 Username :		It configures the Broadsoft phonebook Username.	Phonebook>>Cloud phonebook>>Broadsoft Directory Settings>>Username
BWDir1 Password :		It configures the Broadsoft phonebook Password.	Phonebook>>Cloud phonebook>>Broadsoft Directory Settings>>Password
BWDir1 SipLine :0		It configures the Broadsoft phonebook Corresponding lines.	Phonebook>>Cloud phonebook>>Broadsoft Directory Settings>>Password
--BW Calllogs-- :			
BWCLog1 Title :		It configures the Broadsoft calllog name.	Phonebook>>Cloud phonebook>>Broadsoft Call logs Settings>>Display Title
BWCLog1 URL :		It configures the Broadsoft calllog URL.	Phonebook>>Cloud phonebook>>Broadsoft Call logs Settings>>Server Address
BWCLog1 Username :		It configures the Broadsoft calllog Username.	Phonebook>>Cloud phonebook>>Broadsoft Call logs Settings>>Username
BWCLog1 Password :		It configures the Broadsoft calllog Password.	Phonebook>>Cloud phonebook>>Broadsoft Call logs Settings>>Password
BWCLog1 SipLine :0		It configures the Broadsoft calllog Corresponding lines.	Phonebook>>Cloud phonebook>>Broadsoft Call logs Settings>>SIP Line
--LDAP Config-- :			
LDAP1 Title :	Max 31 characters	It configures the search criteria for LDAP contact names look up.	Phonebook>>Cloud phonebook>>LDAP Settings>>Display Title

LDAP1 Server :	Max 31 characters	It configures the IP address or domain name of the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Server Address
LDAP1 port :389	valid value: 1~65535	It configures the port of the LDAP server	Phonebook>>Cloud phonebook>>LDAP Settings>>Server Port
LDAP1 Base :	within 39 characters.	It configures the LDAP search base which corresponds to the location of the LDAP phone book from which the LDAP search request begins.	Phonebook>>Cloud phonebook>>LDAP Settings>>Search Base
LDAP1 Use SSL :0	0: LDAP 1: LDAPS 2:LDAP TLS Start	It configures the connection mode between the LDAP server and the IP phone.	Phonebook>>Cloud phonebook>>LDAP Settings>>LDAP TLS Mode
LDAP1 Version :3	2: version 2 3: version 3	It configures the LDAP protocol version supported by the IP phone.	Phonebook>>Cloud phonebook>>LDAP Settings>>Version
LDAP1 Calling Line :-1	valid value: -1~6 1-6:line1~6	It selects the lines that LDAP uses.	Phonebook>>Cloud phonebook>>LDAP Settings>>Calling Line
LDAP1 Bind Line :-1	valid value: -1~6 1-6:line1~6	use LDAP lines for queries. The line that is configured to AUTO is searched without the check	Phonebook>>Cloud phonebook>>LDAP Settings>>Search Line
LDAP1 In Call Search :0	0: Disable 1: Enable	The number is queried in LDAP when called in	Phonebook>>Cloud phonebook>>LDAP Settings>>Enable In Call Search
LDAP1 Out Call Search :0	0: Disable 1: Enable	The number is queried in LDAP when called out	Phonebook>>Cloud phonebook>>LDAP Settings>>Enable Out Call Search
LDAP1 Authenticate :3	0: None 1: Digest-MD5 2: CRAM-MD5 3: Simple	It selects the authentication type.	Phonebook>>Cloud phonebook>>LDAP Settings>>Authentication
LDAP1 Username :	Max 39 characters	It configures the user name used to login the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Username

LDAP1 Password :	Max 63 characters	It configures the password to login the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Password
LDAP1 Tel Attr :telephoneNumber	Max 39 characters	It configures the telephoneNumber attributes of each record to be returned by the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Telephone
LDAP1 Mobile Attr :mobile	Max 39 characters	It configures the mobileNumber attributes of each record to be returned by the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Mobile
LDAP1 Other Attr :other	Max 39 characters	It configures the otherNumber attributes of each record to be returned by the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>>Other
LDAP1 Name Attr :cn sn ou	Max 39 characters	It configures the display name of the contact record displayed on the LCD screen.	Phonebook>>Cloud phonebook>>LDAP Settings>>Name Attr
LDAP1 Sort Attr :cn	Max 39 characters	It configures the display name of the contact record displayed on the LCD screen.	Phonebook>>Cloud phonebook>>LDAP Settings>>Sort Attr
LDAP1 Displayname :cn	Max 39 characters	It configures the display name of the contact record displayed on the LCD screen.	Phonebook>>Cloud phonebook>>LDAP Settings>>Display name
LDAP1 Number Filter :((telephoneNumber=%)(mobile=%)(other=%))	((telephoneNumber=%)(mobile=%)(other=%)) (&(telephoneNumber=%)(mobile=%)(other=%))	LDAP contact number filtering rules	Phonebook>>Cloud phonebook>>LDAP Settings>>Number Filter
LDAP1 Name Filter :((cn=%)(sn=%))	((cn=%)(sn=%)) (&(cn=%)(sn=%))	LDAP contact name filtering rules	Phonebook>>Cloud phonebook>>LDAP Settings>>Name Filter
LDAP1 Max Hits :50	valid value: 1~65535	It configures the maximum number of search results to be returned by the LDAP server.	Phonebook>>Cloud phonebook>>LDAP Settings>> Max Hits
--Xml PhoneBook-- :			

XML-PBook1 Name :	Max 29 characters	Sets the XML phone book name	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Cloud phonebook name
XML-PBook1 Addr :	Max 89 characters	Sets the XML phone book server address	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Cloud phonebook URL
XML-PBook1 UserName :	Max 30 characters	Sets the XML phone book server name	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Authenti cation Name
XML-PBook1 PassWd :	Max 16 characters	Sets the XML phone book server password	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Authenti cation Password
XML-PBook1 Sipline :-1	valid value: -1~6 1-6:line1-6	Set up the XML phone book to call the line	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Calling Line
XML-PBook1 BindLine :-1	valid value: -1~6 1-6:line1-6	Set up an XML phone book search line	Phonebook>>Cloud phonebook>>Manage Cloud Phonebooks>>Search Line
<DEVICE MANAGER MODULE>			
Onhook Time :120	valid value: 100~1000(ms)		Hidden configuration
<CTI CONFIG MODULE>			
Enabled Active Uri :1	0: Disable 1: Enable	Allows you to manipulate the phone through the Active url	Hidden configuration
Enabled Action Url :1	0: Disable 1: Enable	Allows you to manipulate the phone through the Actiion url	Hidden configuration

Active Uri IP :	IPv4 address		
Start Reboot Url :	Max 128 characters	It configures the action URL, the IP phone sends after startup	Phone settings>>Action>>Start Reboot
Boot Completed Url :	Max 128 characters	It configures the action URL, the IP phone sends after Boot Completed	Phone settings>>Action>>Set up Completed
IP Change Url :	Max 128 characters	It configures the action URL, the IP phone sends when changing the IP address of the IP phone	Phone settings>>Action>>IP Changed
Reg On Url :	Max 128 characters	It configures the action URL, the IP phone sends after an account is registered	Phone settings>>Action>>Registration Succeeded
Reg Off Url :	Max 128 characters	It configures the action URL, the IP phone sends after an account is unregistered.	Phone settings>>Action>>Registration Disabled
Reg Failed Url :	Max 128 characters	It configures the action URL, the IP phone sends after an account failed to register	Phone settings>>Action>>Registration Failed
PhoneState Idle Url:	Max 128 characters	It configures the action URL, the IP phone sends after idle.	Phone settings>>Action>>Phone State Idle
PhoneState Talking :	Max 128 characters	It configures the action URL, the IP phone sends after talking	Phone settings>>Action>>Phone State Talking
PhoneState Ringing :	Max 128 characters	It configures the action URL, the IP phone sends after ring	Phone settings>>Action>>Phone State Ringing
DND On Url :	Max 128 characters	It configures the action URL, the IP phone sends when DND feature is enabled.	Phone settings>>Action>>DND Enabled
DND Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when DND feature is disabled.	Phone settings>>Action>>DND Disabled

Always FWD On Url :	Max 128 characters	It configures the action URL, the IP phone sends when always forward feature is enabled.	Phone settings>>Action>>Unc onditional Call Forward Enabled
Always FWD Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when always forward feature is disabled.	Phone settings>>Action>>Unc onditional Call Forward Disabled
Busy FWD On Url :	Max 128 characters	It configures the action URL, the IP phone sends when busy forward feature is enabled.	Phone settings>>Action>>Call Forward on Busy Enabled
Busy FWD Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when busy forward feature is disabled.	Phone settings>>Action>>Call Forward on Busy Disabled
No Ans FWD On Url :	Max 128 characters	It configures the action URL, the IP phone sends when no answer forward feature is enabled.	Phone settings>>Action>>Call Forward on No Answer Enabled
No Ans FWD Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when no answer forward feature is disabled.	Phone settings>>Action>>Call Forward on No Answer Disabled
Mute On Url :	Max 128 characters	It configures the action URL, the IP phone sends when Mute on.	Phone settings>>Action>>Call Mute
Mute Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when Mute off.	Phone settings>>Action>>Call Unmute
Incoming Call Url :	Max 128 characters	It configures the action URL, the IP phone sends when receiving an incoming call.	Phone settings>>Action>>Inco ming Calls
Outgoing Call Url :	Max 128 characters	It configures the action URL, the IP phone sends when outgoing call.	Phone settings>>Action>>Out going Calls

Call Active Url :	Max 128 characters	It configures the action URL, the IP phone sends when call active.	Phone settings>>Action>>Call Established
Call Stop Url :	Max 128 characters	It configures the action URL, the IP phone sends when call stop.	Phone settings>>Action>>Call Terminated
Transfer Url :	Max 128 characters	It configures the action URL, the IP phone sends when performing a transfer.	Phone settings>>Action>>Call transfer
Hold On Url :	Max 128 characters	It configures the action URL, the IP phone sends when call hold.	Phone settings>>Action>>Call hold
Hold Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when call resume.	Phone settings>>Action>>Call resume
Held On Url :	Max 128 characters	It configures the action URL, the IP phone sends when call held.	
Held Off Url :	Max 128 characters	It configures the action URL, the IP phone sends when call resume.	
Mute On Call Url :	Max 128 characters	t configures the action URL, the IP phone sends when call Mute on.	Phone settings>>Action>>Call Mute
Mute Off Call Url :	Max 128 characters	t configures the action URL, the IP phone sends when call Mute off.	Phone settings>>Action>>Call Unmute
New Missed call Url:	Max 128 characters	t configures the action URL, the IP phone sends when receive new missed call.	Phone settings>>Action>>Missed Calls
New MWI Url :	Max 128 characters	t configures the action URL, the IP phone sends when receive a MWI.	Phone settings>>Action>>MWI
New SMS Url :	Max 128 characters	t configures the action URL, the IP phone sends when receive a message.	Phone settings>>Action>>SMS
--CTI AT Config-- :			

At Enabled :0		Special server configuration	Hidden configuration
At Server :		Special server configuration	Hidden configuration
<MCAST CFG MODULE>			
Priority :0	Valid Value:-1~9 -1: Disable	Set the priority of normal call and multicast	Phone settings>>MCAST>>MCAST Settings>>Priority
Enable Priority :0	0: Disable 1: Enable	Whether to enable priority configuration	Phone settings>>MCAST>>MCAST Settings>>Enable Page Priority
--Mcast Addr-- :			
MCAST1 Name :	Max 40 characters	Sets the multicast name	Phone settings>>MCAST>>MCAST Settings
MCAST1 Host :	Max 40 characters	Sets the multicast HOST	Phone settings>>MCAST>>MCAST Settings
MCAST1 Port :0	Valid Value:1~65535	Sets the multicast Port	Phone settings>>MCAST>>MCAST Settings
<MMI CONFIG MODULE>			
Web Server Type :1	0: http 1: https	Choose Web login type	Network>>Service Port>>Service Port Settings>>Web Server Type
Web Port :80	Valid Value:1024~65535	http: login web port	Network>>Service Port>>Service Port Settings>>HTTP Port
Https Web Port :443	Valid Value:1024~65535	https: login web port	Network>>Service Port>>Service Port Settings>>HTTPS Port
Remote Control :1			
Enable MMI Filter :0	0: Disable 1: Enable	Whether to enable web filter	Security>>Web Filter>>Web Filter Setting

Web Authentication :0	0:JS Authentication 1:Basic Authentication	Whether to enable automatic web page login	Network>>Service Port>>Service Port Settings>>web auto login
Enable Telnet :0	0: Disable 1: Enable	Whether to enable telnet	Hidden configuration
Telnet Port :23		Connect telnet port	Hidden configuration
Telnet Prompt :		Whether to open remote connection prompt	Hidden configuration
Logon Timeout :15	Valid Value:10~30	Web access timeout setting.	Network>>Service Port>>Service Port Settings>>Web Logon Timeout
--MMI Account-- :			
Account1 Name :123456	Max 31 characters	Web login user name	Login
Account1 Password :123646	Max 31 characters	Web login user password	Login
Account1 Level :10	5:users 10:admin	Account Type	Login
<TR069 CONFIG MODULE>			
TR069 Tone :1	0: Disable 1: Enable	TR069 Warning Tone,connection success or lost	System>>Auto Provision>>TR069>>Enable TR069 Warning Tone
CPE SerialNumber :3c28a6000081		Device's TR069 serial number	
ACS Server Type :1	0:CTC 1:common	TR069 Type.	System>>Auto Provision>>TR069>>ACS Server Type
Enable TR069 :0	0: Disable 1: Enable	Enable TR069.	System>>Auto Provision>>TR069>>Enable TR069
ACS URL :0.0.0.0	Max 128 characters	TR069 Server Address	System>>Auto Provision>>TR069>>ACS Server URL
ACS UserName :admin	Max 59 characters	ACS authentication User name.	System>>Auto Provision>>TR069>>ACS User

ACS Password :admin	Max 59 characters	ACS authentication User password	System>>Auto Provision>>TR069>>ACS Password
ACS Backup URL :0.0.0.0	Max 128 characters	ACS Backup server address	Hidden configuration
ACS BackupUserName :	Max 59 characters	ACS Backup server username	Hidden configuration
ACS BackupPassword :	Max 59 characters	ACS Backup server password	Hidden configuration
CPE UserName :dps			Hidden configuration
CPE Password :dps			Hidden configuration
Periodix Interval :3600	Valid Value:1~9999	TR069 message cycle	System>>Auto Provision>>TR068>>INFORM Sending Period
TLS Version :0	0: TLS 1.0 1: TLS 1.1 2: TLS 1.2	Select the TLS version	System>>Auto Provision>>TR069>>TLS Version
Area Code :	Empty		
STUN Enable :0	0: Disable 1: Enable		System>>Auto Provision>>TR069>>STUN Enable
STUN Server Addr :		STUN Server Address	System>>Auto Provision>>TR070>>STUN Server Address
STUN Server Port :3478	Valid Value:1~65535	STUN Server Port	Hidden configuration
STUN Local Port :30000	Valid Value:1~65535	STUN Local Port	Hidden configuration
<SIP Hotspot MODULE>			
Enable Hotspot :0	0: Disable 1: Enable	Whether to turn on hotspot function	Line>>SIP Hotspot>>SIP Hotspot Settings>>Enable Hotspot
Mode :1	0:Hotspot 1:Client	Sip hotspot type	Line>>SIP Hotspot>>SIP Hotspot Settings>>Mode
Listen Type :0	0: Broadcast 1: Multicast	Set Listen Type	Line>>SIP Hotspot>>SIP Hotspot

			Settings>>Monitor Type
Listen IP :224.0.2.0		Listen Ip address	Line>>SIP Hotspot>>SIP Hotspot Settings>>Monitor Address
Listen Port :16360	Valid Value:1~65535	Listen port	Line>>SIP Hotspot>>SIP Hotspot Settings>>Local Port
Own Name :SIP Hotspot	Max 40 characters	SIP hotspot name,The same hot the name of the client can connect	Line>>SIP Hotspot>>SIP Hotspot Settings>>Name
--Line Conf List-- :			
HS1 Enable :1	0: Disable 1: Enable	SIP enable hotspot	Line>>SIP Hotspot>>Line Settings
<VPN CONFIG MODULE>			
VPN mode :-1	1:L2TP 2:openVPN	Set VPN mode type	Network>>VPN>>VPN Mode
Enable VPN :0	0: Disable 1: Enable	Set whether to enable VPN	Network>>VPN>>VPN Mode>>Enable VPN
Enable Nat :0	0: Disable 1: Enable	Set whether to enable Nat	Network>>VPN>>VPN Mode>>Enable NAT
Openvpn mode :0	0: tun 1: tap	Set Openvpn mode	Network>>VPN>>VPN Mode>>Open VPN mode
L2TP Server Address:0.0.0.0	IP address and domain name	L2TP Server Address	Network>>VPN>>Layer 2 Tunneling Protocol >>L2TP Server Address
L2TP User Name :	Max 40 characters	L2TP User Name	Network>>VPN>>Layer 2 Tunneling Protocol >>Authentication Name
L2TP Password :	Max 40 characters	L2TP Password	Network>>VPN>>Layer 2 Tunneling Protocol >>Authentication Password
L2TP Negotiate DNS :1		L2TP Negotiate DNS	Hidden configuration
PPTP Server Address:0.0.0.0		PPTP Server Address	Hidden configuration

PPTP User Name :		PPTP User Name	Hidden configuration
PPTP Password :		PPTP Password	Hidden configuration
<MAINTENANCE CONFIG MODULE>			
Contact Update Mode:0		Set contact update mode	
Auto Server Digest :0	0: Disable 1: Enable	computer digest by server before downloading	System>>Auto Provision>>Basic Settings>>Enable Server Digest
<AUTOUPDATE CONFIG MODULE>			
Default Username :	Max 32 characters	Authentication username for Auto Provisioning.	System>>Auto Provision>>Basic Settings>>Authentication Name
Default Password :	Max 22 characters	Automatically upgraded passwords.	System>>Auto Provision>>Basic Settings>>Authentication Password
Input Cfg File Name:			
Auto Image Url :	ftp://username:password@server IP/x3s.z	Need to open FTP server, URL format for the User name, Password, phone IP, version full name, where the User and Password is the server request when the User name and Password, this is not a mandatory item	System>>Auto Provision>>Static Provisioning Server
Auto Pbook Url :	ftp://username:password@server IP/phonebook.vcf	Need to open FTP server, URL format for the User name, Password, phone IP, phone book, where the User and Password is the server request when the User name and Password, this is not a mandatory item	System>>Auto Provision>>Static Provisioning Server

Auto etc Url :	ftp://username:password@server IP/ca.crt	Need to open the FTP server, the URL format for the User name, Password, phone IP, certificate, where the User and Password is the request to the server when the User name and Password, this is not a mandatory item	System>>Auto Provision>>Static Provisioning Server
Device Cfg File Key :	Max 64 characters	Decryption key for Configuration file.	System>>Auto Provision>>Basic Settings>>Configuration File Encryption Key
Common Cfg File Key:	Max 64 characters	Decryption key for common Configuration file.	System>>Auto Provision>>Basic Settings>>General Configuration File Encryption Key
Download CommonConf:0	0: Disable 1: Enable	Download Common configuration file	System>>Auto Provision>>Basic Settings>>Download CommonConfig enabled
Save Provision Info:0	0: Disable 1: Enable	Save manually input parameters for auto provisioning. Including Authentication user name/password, and InputID	System>>Auto Provision>>Basic Settings>>Save Auto Provision Information
Check FailTimes :5	Max 4 characters	Download failed check times	System>>Auto Provision>>Basic Settings>>Download Fail Check Times
Flash Server IP :	Max 512 characters	Downloaded server address	System>>Auto Provision>>Static Provisioning Server>>Server Address
Flash File Name :	Max 63 characters,\$mac, \$input,or some othe assigned file name. leave blank is optional. File suffix might be CFG/TXT/XML.	Downloaded configuration file's name	System>>Auto Provision>>Static Provisioning Server>>Configuration File Name

Flash Protocol :2	1:FTP 2:TFTP 4:HTTP 5:HTTPS	Downloaded protocol	System>>Auto Provision>>Static Provisioning Server>>Protocol Type
Flash Mode :0	0: Disable 1: Update After reboot 2: Update at time interval	Update mode.	System>>Auto Provision>>Static Provisioning Server>>Update Mode
Flash Interval :1	valid value:1~9999	Downloading interval period	System>>Auto Provision>>Static Provisioning Server>>Download Fail Check Times
update PB Interval :720	valid value:1~9999	Local phonebook file's downloaded periodic interval	System>>Auto Provision>>Static Provisioning Server>>Update Contact Interval
AP Pswd Encryption :0	0: Disable 1: Enable	Sets whether the automatic upgrade turns on encryption	Hidden configuration
--Sip Pnp List-- :			
PNP Enable :1	0: Disable 1: Enable	Whether to turn on sip PNP	System>>Auto Provision>>SIP Plug and Play (PnP)>>Enable SIP PnP
PNP IP :224.0.1.75	IPv4 Address	PNP IP address	System>>Auto Provision>>SIP Plug and Play (PnP)>>Server Address
PNP Port :5060	valid value:1~65535	PNP server port	System>>Auto Provision>>SIP Plug and Play (PnP)>>Server Port
PNP Transport :0	0: UDP 1: TCP	PNP transportation protocol	System>>Auto Provision>>SIP Plug and Play (PnP)>>Transport Protocol
PNP Interval :1	valid value:1~99	PNP update period.	System>>Auto Provision>>SIP Plug and Play (PnP)>>Update Interval

--Net Option-- :			
DHCP Option :66	0: Disable 66:Option 66 43:Option 43 128~254:Option 128~254	DHCP OPTION type for Auto Provisioning	System>>Auto Provision>>DHCP Option>>Option Value
DHCPv6 Option :0	0: Disable 66:Option 66 43:Option 43 128~254:Option 128~254	DHCP ipv6 OPTION type for Auto Provisioning	System>>Auto Provision>>DHCP Option>>Custom Option Value
Dhcp Option 120 :0	0: Disable 1: Enable	Set sip register server address through dhcp option 120	System>>Auto Provision>>DHCP Option>>Enable DHCP Option 120
--MDNS Config-- :			
<OTA CONFIG MODULE>			
<RPS CONFIG MODULE>			
Rps Name :fdps			
<FIRMWARE CHECK MODULE>			
Enable Auto Upgrade:0	0: Disable 1: Enable	Firmware auto upgrade	System>>Upgrade>>U pgrade Server>>Enable Auto Upgrade
Upgrade Server 1 :		Firmware auto upgrade server address 1	System>>Upgrade>>U pgrade Server>>Upgrade Server Address1
Upgrade Server 2 :		Firmware auto upgrade server address 2	System>>Upgrade>>U pgrade Server>>Upgrade Server Address2
Auto Upgrade Interval:24		Set the automatic upgrade time	System>>Upgrade>>U pgrade Server>>Update Interval

<QOS CONFIG MODULE>			
Enable VLAN :0	0: Disable 1: Enable	Enable VLAN to let system access to VLAN network with vlan tagged	Network>>Advanced>>WAN VLAN Settings>>Enable VLAN
VLAN ID :256	valid value:0~4095	LAN ID for system WAN port	Network>>Advanced>>WAN VLAN Settings>> WAN VLAN ID
Enable PVID :1	0: follow WAN 1:Disable 2: Enable	It configures LAN port mode	Network>>Advanced>>WAN VLAN Settings>>LAN VLAN Mode
PVID Value :254	valid value:0~4095	It configures VLAN for the Internet (LAN) port.	Network>>Advanced>>WAN VLAN Settings>>LAN VLAN ID
Signalling Priority:0	valid value:0~7	802.1P priority for SIP messages	Network>>Advanced>>WAN VLAN Settings>>802.1p Signal Priority
Voice Priority :0	valid value:0~7	Set the use of voice priority	Network>>Advanced>>WAN VLAN Settings>>802.1p Media Priority
Video Priority :0	valid value:0~7	Set the use of video priority	Network>>Advanced>>WAN VLAN Settings>>802.1p Media Priority
LAN Port Priority :0	valid value:0~7	Set the local area network VLAN priority	Network>>Advanced>>WAN VLAN Settings>>LAN VLAN PRIORITY
Enable diffServ :0	0: Disable 1: Enable	Enable DSCP to get best effort QoS for voice quality	Network>>Advanced>>Quality of Service (QoS) Settings>>Enable DSCP
Signalling DSCP :46	valid value:0~63	DSCP value for SIP messages	Network>>Advanced>>Quality of Service (QoS) Settings>>Signal DSCP

Voice DSCP :46	valid value:0~63	DSCP value for RTP streams.	Network>>Advanced>>Quality of Service (QoS) Settings>>Audio DSCP
Video DSCP :46	valid value:0~63		Network>>Advanced>>Quality of Service (QoS) Settings>>Audio DSCP
LLDP Transmit :0	0: Disable 1: Enable	Enables LLDP (Linker Layer Discovery Protocol) function.	Network>>Advanced>>Link Layer Discovery Protocol (LLDP) Settings>>Enable LLDP
LLDP Refresh Time :60	Valid Value:1~3600	LLDP Refresh Time	Network>>Advanced>>Link Layer Discovery Protocol (LLDP) Settings>>Packet Interval
LLDP Learn Policy :0	0: Disable 1: Enable	Enable VLAN settings learned via LLDP-MED	Network>>Advanced>>Link Layer Discovery Protocol (LLDP) Settings>>Enable Learning Function
LLDP Save Learn Data :0	0: Disable 1: Enable		Network>>Advanced>>Link Layer Discovery Protocol (LLDP) Settings>>Audio DSCP
CDP Enable :0	0: Disable 1: Enable	Enables CDP (Cisco Discovery Protocol) function to learn VLAN settings automatically	Network>>Advanced>>Cisco Discovery Protocol (CDP) Settings>>Enable CDP
CDP Refresh Time :60	Valid Value:1~3600	CDP Refresh Time	Network>>Advanced>>Cisco Discovery Protocol (CDP) Settings>> Packet Interval
DHCP Option Vlan :0	0:Disable 128~254:option vlan 128~254	The DHCP option for VLAN discovery	Network>>Advanced>>Cisco Discovery Protocol (CDP) Settings
<LOG CONFIG MODULE>			
Level :INFO	INFO:information DEBUG:debug WARN: warning ERROR:error	Output logs' level.	System>>Tools>>APP Log Level

Style :level,tag		Output logs' level.	System>>Tools>>APP Log Level
Output Device :stdout		Output logs' level.	System>>Tools>>APP Log Level
File Name :platform.log		Syslog file name	System>>Tools>>APP Log Level
File Size :512KB		Syslog file size	System>>Tools>>APP Log Level
Syslog Tag :platform		Syslog type	System>>Tools>>APP Log Level
Syslog Server :0.0.0.0		Syslog server address	System>>Tools>>Server Address
Syslog Server Port :514		Syslog server port	System>>Tools>>Server Port
<APP CONFIG MODULE>			
Watch Dog Enabled :1	0: Disable 1: Enable	The phone will restart automatically when it encounters an abnormal problem	System>>Tools>>Enable Watch Dog
Enable In Access :0	0: Disable 1: Enable		Hidden configuration
Enable Out Access :0	0: Disable 1: Enable		Hidden configuration
<VQM CONFIG MODULE>			
Session Report :1	0: Disable 1: Enable	Enable/Disable RTCP-XR session report	Line>>RTCP-XR>>VQ RTCP-XR Session Report
Interval Report :1	0: Disable 1: Enable	Enable/Disable RTCP-XR interval report	Line>>RTCP-XR>>VQ RTCP-XR Interval Report
Interval Period :60	valid value:5~99	Set the interval for RTCP-XR interval report	Line>>RTCP-XR>>Period for Interval Report(5~99)
MOS-LQ Warning :40	valid value:15~40	Set the warning threshold for Moslq	Line>>RTCP-XR>>Warning threshold for Moslq(15~40)

MOS-LQ Critical :25	valid value:15~40	Set the critical threshold for Moslq	Line>>RTCP-XR>>Critical threshold for Moslq(15~40)
Delay Warning :150	valid value:10~2000	Set the warning threshold for Delay	Line>>RTCP-XR>>Warning threshold for Delay(10~2000)
Delay Critical :200	valid value:10~2000	Set the critical threshold for Delay	Line>>RTCP-XR>>Critical threshold for Delay(10~2000)
Phone Report :1	0: Disable 1: Enable	Enable/Disable display report on phone	Hidden configuration
WEB Report :1	0: Disable 1: Enable	Enable/Disable display report on web	Line>>RTCP-XR>>Display Report options on Web
<DOT1X CONFIG MODULE>			
Xsup Mode :0	0:Disable 1:EAP-MD5 2:EAP-TLS 3:EAP-MSCHAPV2	802.1X Authentication type	Network>>Advanced>>802.1X Settings>>802.1x Mode
Xsup User :admin	Max 50 characters	802.1X Authentication username	Network>>Advanced>>802.1X Settings>>Identity
Xsup Password :admin	Max 50 characters	802.1X Authentication password	Network>>Advanced>>802.1X Settings>>Password
--SSL Mode-- :			
Permission CTF :0	0: Disable 1: Enable	It enables or disables the phone to only trust the server certificates in the Trusted Certificates list.	Security>>Trust Certificates>>Permissi on Certificate
Common Name :0	0: Disable 1: Enable	It shows whether to enable common name validation.	Security>>Trust Certificates>>Common Name Validation
CTF mode :0	0: Disable 1: Enable	It configures the type of certificates in the Trusted Certificates list for the phone to authenticate for TLS connection.	Security>>Trust Certificates>>Certificat e mode

Device Cert Mode :0	0: Default Certificates 1: Custom Certificates	Default certificates are built into the device	Security>>Device Certificates>>Device Certificates
<IPCAMERA CONFIG MODULE>			
<RECORD CONFIG MODULE>			
Enabled :1	0: Disable 1: Enable	Whether to turn on the recording	Application>>Manage Recording
Voice Codec :G729	PCMA PCMU G726-32 G729 G723 Ilbc AMR AMR-WB opus G722	Select recording code	Application>>Manage Recording
Record Type :0	0: local 1: network 2: sip info	Select recording type	Application>>Manage Recording
File Size Limit :8	local record file size, no limit	Record file size limit	Application>>Manage Recording
Server Addr :	server ip	Recording server address	Application>>Manage Recording
Server Port :0	valid value:1-65535	Recording server port	Application>>Manage Recording
<ATE CONFIG MODULE>			
ATE Id :0000000000000000			
<UI DEFINED CONFIG MODULE>			
Bluetooth Adapter Name :Fortinet FortiFone FON-475		Name of external bluetooth adapter	
Bluetooth Enabled:0		Turn on bluetooth	
BT Last Connected			
Addr:00:00:00:00:00:00		Last connected bluetooth MAC address	

BT Audio Codec:1		Bluetooth USES voice coding	
<UI MAINTAIN CONFIG MODULE>			
Timeout To Screensaver :0			
User Change Background :0			
EHS Headset type :0	0: Disable 1: EHS Plantronics	When enabled, you can use ehs headset to answer the call	
<DSSKEY CONFIG MODULE>			
Select DsskeyAction:0	0: none 1:call hold 2:hangup	Select the function of the memory key function in the call	Function key>>Function key >>Function key settings
Memory Key to BXfer:3	1: blind transfer 2: attended transfer 3: make a new call 4: conference call 5: play DTMF	One key setting function key	Function key>>Function key >>Function key settings
FuncKey Page Num :1	valid value:0~5	Default shortcut number of pages	Function key>>Function key >>Function key settings
DSS Home Page :0	valid value:0~5 0: none 1~5: functionkey1~5	One key setting function key	Function key>>Function key >>Function key settings
Expand Board Enable:0	0:disable 1:enable	Whether to open the connection extension board	Function key>>EXT key
Extern1 Page Belong :0	valid value:0~5	Extension board configuration page	Function key>>EXT key
Extern2 Page Belong :0	valid value:0~5	Extension board configuration page	Function key>>EXT key
Extern3 Page Belong :0	valid value:0~5	Extension board configuration page	Function key>>EXT key
Extern4 Page Belong :0	valid value:0~5	Extension board configuration page	Function key>>EXT key
Extern5 Page Belong :0	valid value:0~5	Extension board configuration page	Function key>>EXT key

DSS Extend1 MAC :		Extension board MAC	
DSS Extend1 IP :		Extension board IP	
--Dsskey Config1--:			
Fkey1 Type :2	0: none 1: Memory key 2: Line 3: Key Event 4: DTMF 7: URL 13: BLF List Key 14: Multicast 20:action url	First page shortcut key configuration	Function key>>Function key >>Function key settings
Fkey1 Value :SIP1	Max 32 characters	Add the soft dsskey name	
Fkey1 Title :	Max 33 characters	Add the soft dsskey title	
Fkey1 ICON :Green			
--SoftDss Config-- :			
Fkey1 Type :0	0:None 1:memory key 2:line 3:key event 4:DTMF 7:URL 13:BLF LIST 14:MULTICAST 20:Action URL 21:XML browser	Select the soft dsskey type	Function key >>Softkey >>Soft DSS Key Settings
Fkey1 Value :		Add the soft dsskey name	Function key >>Softkey >>Soft DSS Key Settings
Fkey1 Title :		Add the soft dsskey title	Function key >>Softkey >>Soft DSS Key Settings