Integrated Networking Solutions

IP PHONE

VP-30P

User manual

Firmware version 1.2.2

Username: admin

Password: password

Contents

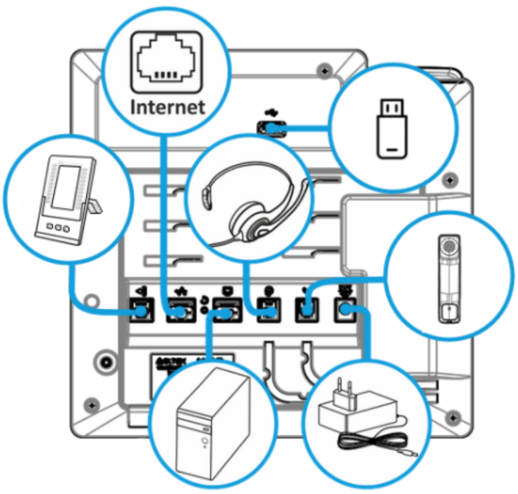
1 VP-30P description

1.1 Purpose

To provide VoIP services to network subscribers, VP series IP phones have been developed. The devices are aimed at offices, and are also suitable for organizations with high requirements to transmitted voice data, stability and usability.

VP-30P is an IP phone providing voice services and PC connection to IP network via single cable. The device supports PoE technology and has advanced functionality, high quality, and universal design.

VP-30P connection layout is shown below:



VP-30P connection layout

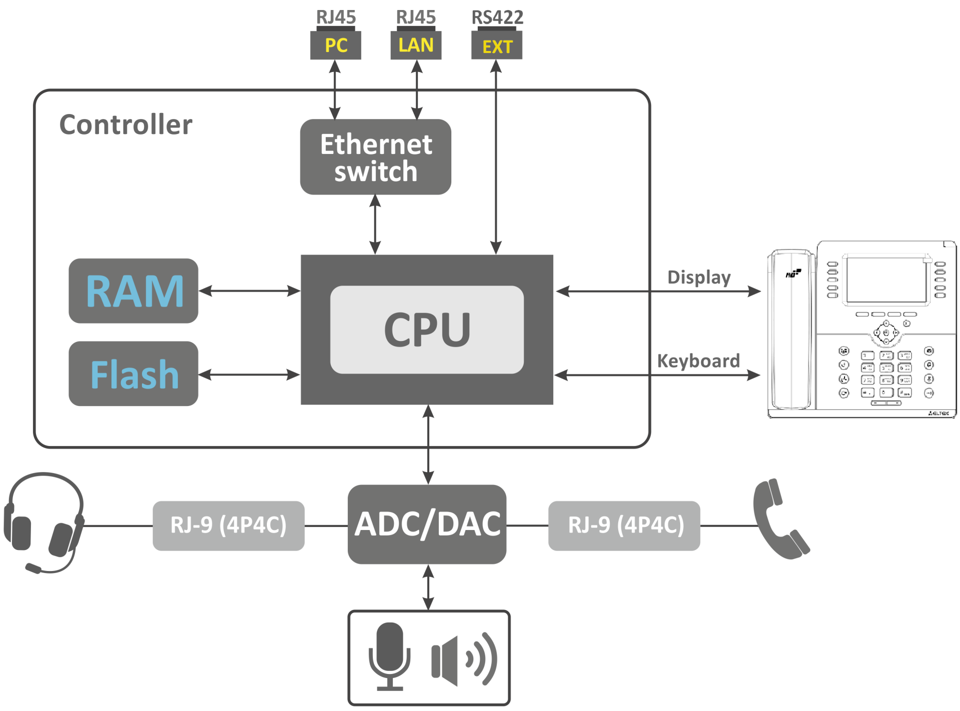
1.2 Operating principle

VP-30P IP phone includes the following subsystem:

- Controller featuring:
  - highly-integrated System-on-a-Chip (SoC), including:
    - CPU;
    - 1 Gbps switch with PHY embedded;
    - voice codec;
    - NAND Flash 512 MB;
    - DDR3 256 MB;

- Liquid crystal display (LCD) 800 × 480 px resolution;
- Fully-featured digital keyboard with additional function keys;
- 1 × EXT port: RJ-25 (6P6C) for connecting the VP-EXT22 extension console;
- 1 × LAN port: RJ-45 10/100/1000BASE-T;
- 1 × PC port: RJ-45 10/100/1000BASE-T;
- 1 × Handset port: RJ-9 (4P4C) for connecting a handset;
- 1 × Headset: RJ-9 (4P4C) for connecting a headset;
- 1 × USB port for connecting a USB device.

Block diagram for the device operating principle is shown below.



VP-30P operating principle

The device runs under Linux operating system. Basic control functions are performed by the processor.

1.3 Main specifications

General parameters	
Power supply	• power adapter (optional): input 100–240 V AC, output 12 V DC, 3 A • PoE+ support IEEE 802.3at (power class 4)
Maximum power consumption	• 15.6 W (without extension consoles) • 24 W (when three VP-EXT22 extension consoles connected)
Operating temperature range	from 0 to +40 °C
Relative humidity at 25 °C	no more than 80 %
Dimensions (W × H × D)	228 × 86 × 220
Weight	1.011 kg
Lifetime	no less than 5 years
Interfaces	• LAN: 1 port of Ethernet RJ-45 10/100/1000BASE-T • PC: 1 port of Ethernet RJ-45 10/100/1000BASE-T • Handset: 1 RJ-9 (4P4C) port for connecting a handset • Headset: 1 RJ-9 (4P4C) port for connecting a headset • EXT: 1 (6P6C) port for connecting the VP-EXT22 extension console • USB: 1 port for connecting a USB device
Ethernet LAN interface specification	
Number of ports	1
Electric port	RJ-45
Data transmission rate	10/100/1000 Mbps, autodetection
Standard support	BASE-T
Ethernet PC interface specification	
Number of ports	1
Electric port	RJ-45
Data transmission rate	10/100/1000 Mbps, autodetection
Standard support	BASE-T

Main features and capabilities

VoIP capabilities	
Supported protocols	SIP
Key features	• 6 SIP accounts configured independently • Support for up to 3 redundant SIP servers • Displaying of caller name and number (CallerID) • Mute • Redial • Call History • Local Phone book • Remote Phone book • LDAP Remote Phone book • Speakerphone mode • Busy Lamp Field (BLF) • Different ringtones for accounts[1] • Message Waiting Indicator (MWI) • SIP MESSAGE[1] • Call recording to USB drive[1] • Screenshots[1] • Remote Conference (RFC4579) • Support for up to three VP-EXT22 extension consoles
Voice features	• DTMF signals detection and generation
DTMF signals detection and generation	• Inband • RFC2833 • SIP INFO
Codecs	• G.711 (PCMA/PCMU) • G.722 • G.729
Supplementary services	• Call Hold • Call Transfer • Call Waiting • Call Forwarding Busy (CFB) • Call Forwarding No Reply (CFNR) • Call Forwarding Unconditional (CFU) • Do Not Disturb mode (DND) • Caller Line Identification Restriction (CLIR) • Hotline/Warmline • Conference • Stop dialing by pressing # • Call Pickup
Network features	
Protocols	• Static IP • DHCP

Support for DHCP option	• 3 — Router • 6 — Domain Name Server • 12 — Host Name • 15 — Domain Name • 33 — Static Route • 40 — Network Information Service Domain • 41 — Network Information Servers • 42 — Network Time Protocol Servers • 43 — Vendor-Specific Information • 66 — TFTP ServerName • 67 — Bootfile name • 120 — SIP Servers • 121 — Classless Static Route • 249 — Private/Classless Static Route (Microsoft)
Support for NTP	• Static NTP server address assignment • Obtaining NTP server address via DHCP dynamically
Management and monitoring	
Key features	Flexible settings for access to display menu
Interfaces	• Web interface • SSH • Display menu
Debug information output	• Syslog • File • Console
Loading/updating of firmware and configuration	• Autoupdate by schedule • Periodical autoupdate

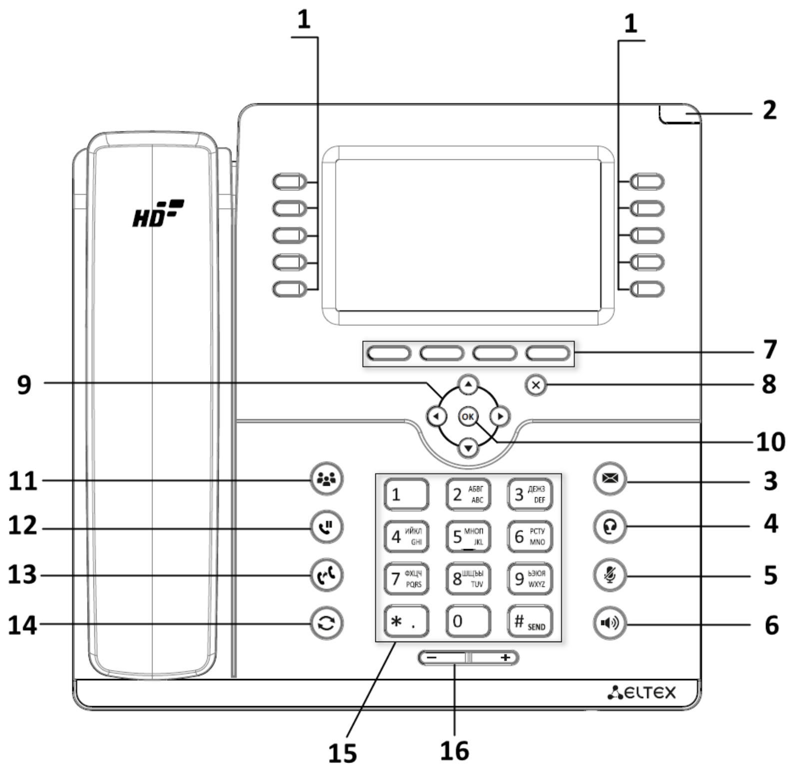
[1] The feature will be available in future firmware versions.

1.4 Design

VP-30P IP phone is enclosed into 228 × 86 × 220 mm plastic case.

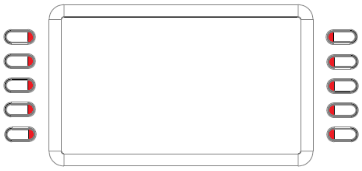
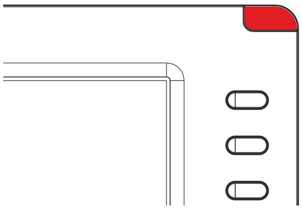
1.4.1 Top panel of the device. Light indication

The figure below shows VP-30P top panel layout.



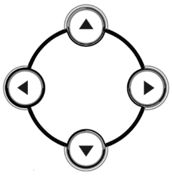
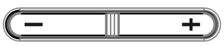
VP-30P top panel layout

VP-30P top panel is equipped with LED indicators:

Top panel element		Description	Indicator state	Device state
1		Programmable keys indicators	Depends on configuration	
2		System indicator	Depends on configuration	
3		New message indicator[1]	Flashes green	There are unread messages or new voice messages
			Off	There are no new messages
4		Headset mode indicator	Solid green	Headset mode is activated
			Off	Headset mode is not activated
5		Mute mode indicator	Solid green	Mute mode is activated for the current conversation
			Off	Mute mode is not activated
6		Speakerphone mode indicator	Solid green	Speakerphone mode is activated
			Off	Speakerphone mode is not activated

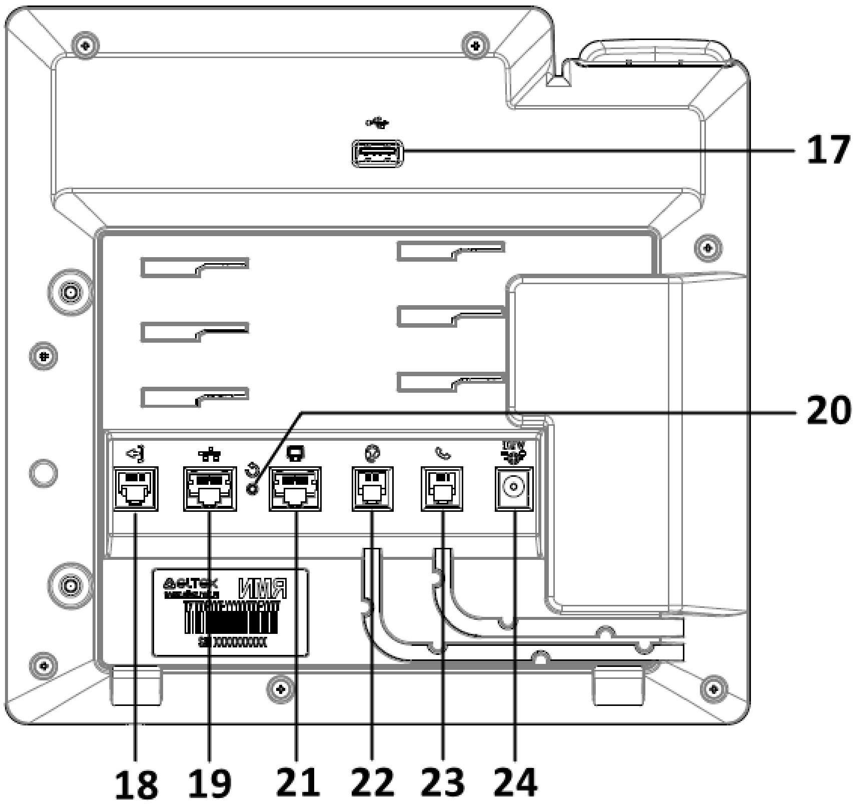
[1] The feature will be available in future firmware versions.

The following controls are also located on the device top panel:

Top panel element		Description
7		Context sensitive keys
8		Action cancelation
9		Navigation
10		Action confirmation
11		Conference call
12		Call Hold activation/disactivation
13		Call Transfer
14		Redial of the last dialled number
15		Keypad
16		Volume adjustment keys

1.4.2 Rear panel of the device

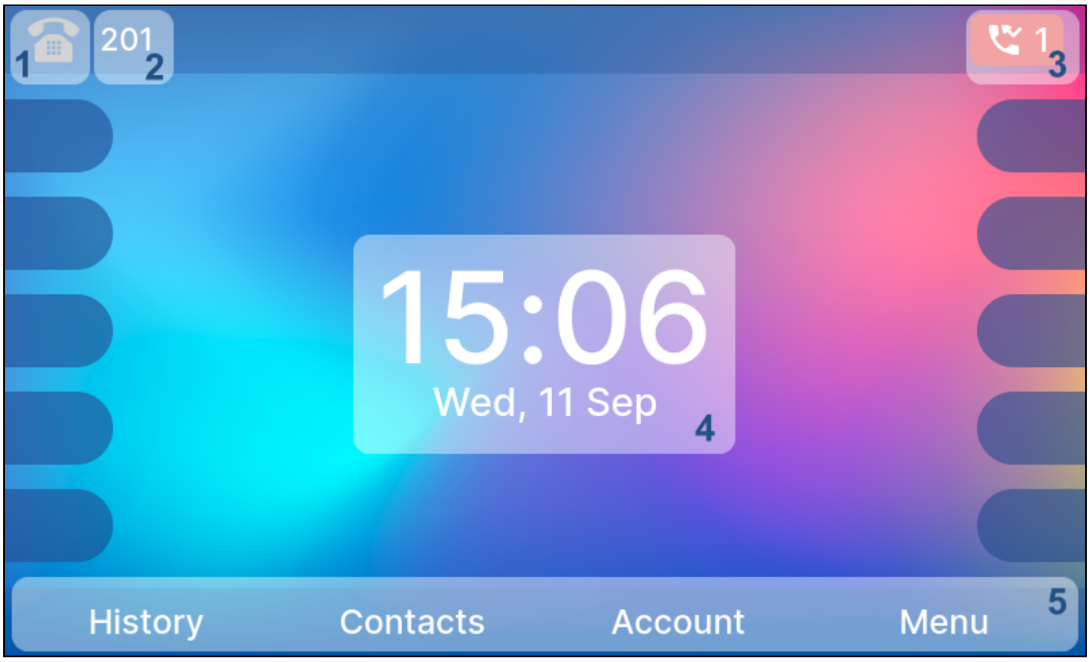
The figure below shows VP-30P rear panel layout.



VP-30P rear panel layout

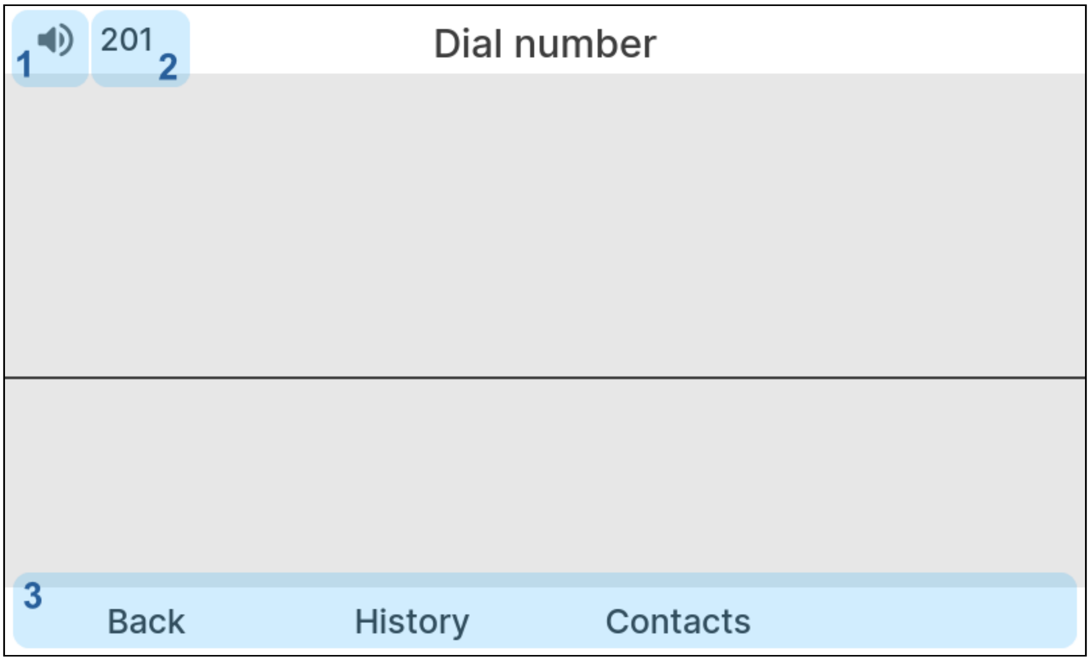
Rear panel element		Description
17	USB	Port for connecting an external memory
18	EXT	Port for connecting the VP-EXT22 extension console
19	LAN	10/100/1000BASE-T Ethernet (RJ-45) port for connection to LAN
20	Reset	Device reset button
21	PC	10/100/1000BASE-T Ethernet (RJ-45) port for connection to PC
22	Headset	RJ-9 port for connecting a headset
23	Handset	RJ-9 port for connecting a handset
24	DC	Port for connecting a power adapter 12 V, 3 A

1.5 Status indication on display



Main screen

No.	Description
1	Indicator of voice interface:  — Default interface 'handset'. Ringtone default mode (incoming call ringtone is played into speakerphone);  — Default interface 'headset'. Ringtone dual mode (incoming call ringtone is played both into headset and speakerphone).
2	Name of default account. If the account does not have a name, the phone number is displayed.
3	Missed call indicator or Mute mode indicator.
4	Current date and time.
5	Actions taken upon pressing soft keys.



Dialing screen

No.	Description
1	Indicator of voice interface:  — Active interface 'handset';  — Active interface 'speakerphone';  — Active interface 'headset'.
2	Name of default account. If the account does not have a name, the phone number is displayed.
3	Actions taken upon pressing soft keys.

1.6 Delivery package

VP-30P standard delivery package includes:

- VP-30P IP phone;
- Double-position stand;
- VP-EXT22 extension console (optional);
- Handset and cable for handset connection;
- 220/12 V, 3 A power adapter (optional);
- RJ-45 cable;
- Quick user manual and warranty certificate.

ⓘ Headphones might be added to delivery package upon a request.

1.7 Basic setup

> ⚠ It is recommended to update the firmware before the first use of the IP phone.

1.7.1 Obtaining IP address

Press the soft key <Menu>. You will see an obtained IP address in the 'Status' section. If the device has not obtained an IP address, the 'IP address' line in Menu → Status → Network will remain empty.

1.7.2 Basic setup using display menu

The IP phone can be managed with soft keys located under the IP phone screen, as well as to the right and left of it. The soft keys assignments are displayed at the bottom, on the left, and on the right of the screen.

1. Press the <Menu> soft key;
2. Go to 'Network settings' menu section: Menu → Settings → System → Network → IP/VLAN → VoIP IP;
3. Configure the necessary network settings;

> ⚠ A list of network parameters and their values can be obtained from your network administrator.

4. Go to 'Account settings': Menu → Settings → System → Accounts → Account 1..6;
5. Configure the necessary account settings.

> ⚠ A list of account parameters and their values can be obtained from your network administrator or the Internet telephony service provider.

1.7.3 Basic setup using web interface

1. Open web browser such as Firefox, Opera or Chrome on the PC;
2. Enter the device IP address in the browser address bar. When the device is successfully connected, login and password request page will be shown in the browser window;
3. Fill in the fields and click the '*Log in*' button. Default login is **admin**, password is **password**;
4. Change network parameters if needed in the Network → Internet tab;
5. Configure the VoIP parameters in the VoIP → SIP Accounts tab.

> ⚠ To enter the IP phone web interface, it is necessary that the computer from which you are logging in is located on the same local network as the IP phone or there is a route between the networks. For IP phone connection issues, consult your network administrator.

1.8 How to use the VP-30P IP phone

1.8.1 Outgoing calls

1.8.1.1 Ordinary call

Option 1: Lift the handset, dial a number on the IP phone keypad and press the <Call> soft key.

Option 2: Dial a number and then lift the handset. The call will be made when the S-timer expires.

1.8.1.2 Speakerphone mode

Dial a number on the IP phone keypad without lifting the handset and press <Call> or key.

To switch from speakerphone to handset:

Lift the handset. Speakerphone mode will turn off automatically.

1.8.1.3 Headset

Option 1: Dial a number on the IP phone keypad without lifting the handset and press key.

Option 2: Press key, dial a number and press <Call>.

To switch from headset to handset:

If the handset is on the phone, lift it. Headset will turn off automatically. If the handset is lifted, press key.

1.8.2 Incoming calls

How to answer a call:

- Lift the handset;
- Press <Call> soft key;
- Press or key.

To ignore a call, press <Silence> soft key. To decline a call, press <Hang up> soft key.

1.8.3 Mute mode

To turn off a microphone during a call so that the other person cannot hear you, press key. The MUTE indicator will light up green, and icon will appear on the display.

1.8.4 Call completion

How to complete a call:

- Press <Hang up> soft key;
- In speakerphone mode, press key;
- In headset mode, press key;
- In ordinary mode, replace the handset.

1.8.5 Call hold

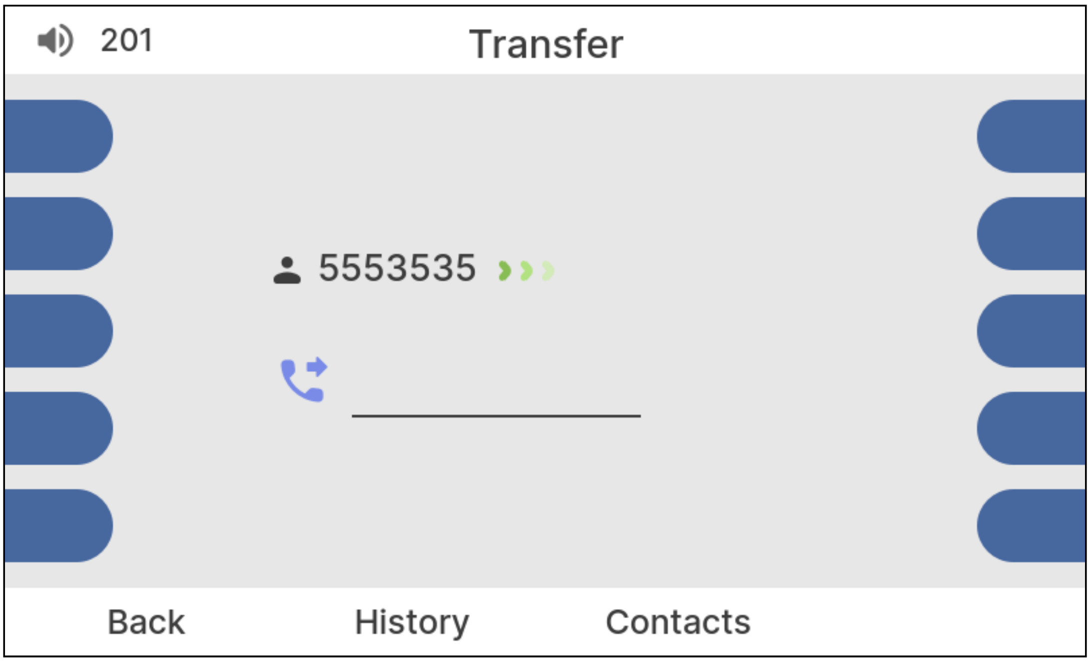
To put a call on hold, press <Hold> soft key or key. When a call is on hold, icon will appear on the display.

1.8.6 Call transfer

There are 2 options of call transfering when you are already on a call:

Option 1:

1. Press key or <Transfer> soft key;
2. Dial the number you want to transfer to and press <Call> soft key;
3. After subcriber answers, let them know that you intend to transfer the call. Press key or <Transfer> soft key.



Transfer screen

Option 2:

1. Press key or <Transfer> soft key;
2. Dial the number you want to transfer to;
3. Press key or replace the handset.

1.8.7 Conference

To organize a remote conference via RFC4579 press <Conference> soft key on the call screen or press key. After pressing, the current call will be moved to the conference room. To get to the conference room press <Conference> soft key on the main screen. You can configure the remote conference room number via the web interface.

1.8.8 Group listening

Group listening mode allows duplicating the sound from the handset speaker to the speakerphone, while only the handset microphone is activated.

Option 1:

Lift the handset and press 🔊 key.

Option 2:

Lift the handset and press the configured 'Group listening' function key.

To complete group listening:

- Press 🔊 key, the phone will switch to handset mode;
- Replace the handset, the phone will switch to speakerphone mode.

2 VP-EXT22 extension console

2.1 Purpose

VP-EXT22 extends functionality of the VP-30P IP phone. The device includes 22 additional keys with LED indication, color LCD, and 3 virtual pages, which increase the number of programmable keys up to 66.

2.2 Main specifications

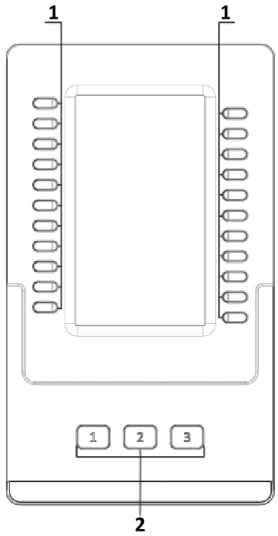
General parameters	
Power supply	• power adapter 12 V DC, 1.5 A (optional) • electrical connection RJ-25
Maximum power consumption	2.8 W
Operating temperature range	from 0 to +40 °C
Relative humidity at 25 °C	up to 80 %
Dimensions (W × H × D)	128 × 220 × 43 mm
Weight	0.52 kg
Lifetime	no less than 5 years
RS-422 interface parameters for IP phone or console connection	
Number of ports	2
Electrical connection	RJ-25 (6P6C)
Data rate	up to 2 Mbps

2.3 Design

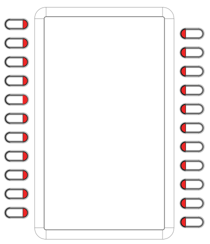
VP-EXT22 is enclosed into 128 × 220 × 43 mm plastic case.

2.3.1 Top panel of the device

The figure below shows VP-EXT22 top panel layout.

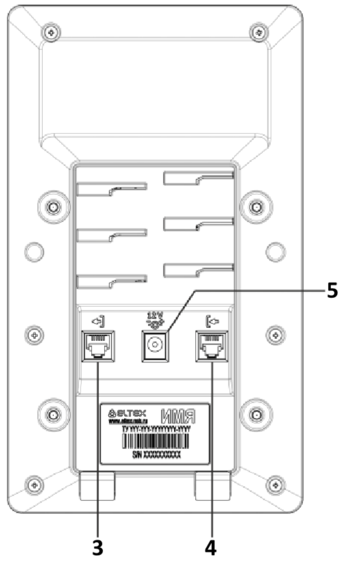


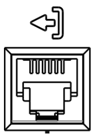
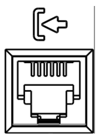
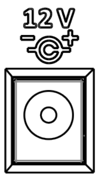
The following controls are located on the device top panel:

Top panel element		Description	Indicator state	Device state
1		Programmable keys indicators	Depends on configuration	
2		Virtual pages switch keys	If the key is lit green, the corresponding virtual page is activated	

2.3.2 Rear panel of the device

The figure below shows VP-EXT22 rear panel layout.



Rear panel element		Description
3		RJ-25 (6P6C) port for connecting the next extension console
4		RJ-25 (6P6C) port for connecting to the IP Phone or the previous extension console
5		port for connecting a power adapter, 12 V DC, 3 A

3 Managing via web interface

3.1 Getting started

3.1.1 Pre-starting procedures

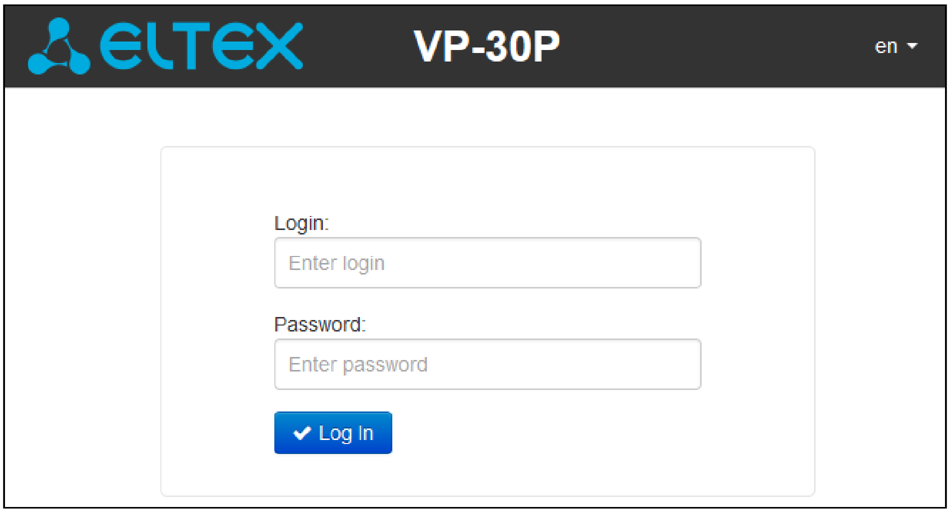
> It is recommended to reset the IP phone to factory settings when switching it on for the first time. Use display menu and buttons to reset the device and go to:
> **Menu** → **3. Settings** → **2. System** → **5. Reset settings** → **Yes**
> The device will automatically reset.

To start the operation, connect the device to PC via LAN interface. Use a web browser:

1. Open web browser, i.e. Firefox, Opera, Chrome.
2. Enter the device IP address in the browser address bar.

> By default, IP phone obtains an IP address and other network parameters automatically via DHCP.
> To get an obtained IP address, go to **Menu** → **1. Status** → **1. Network** using display menu.

When the device is successfully detected, login and password request page will be shown in the browser window:



> By default, login is **admin**, password is **password.**

3. Enter your login into *'Login'* field and password into *'Password'* field.
4. Click *'Log in'* button. Monitoring panel will be shown in the browser.

Before you start, please, upgrade the firmare. See 'Firmware upgrade' submenu.
You can download the up-to-date firmware version on the Downloads page of the Eltex website or contact ELTEX technical support. You can find contacts on TECHNICAL SUPPORT page at the end of the manual.

3.1.2 Web interface description

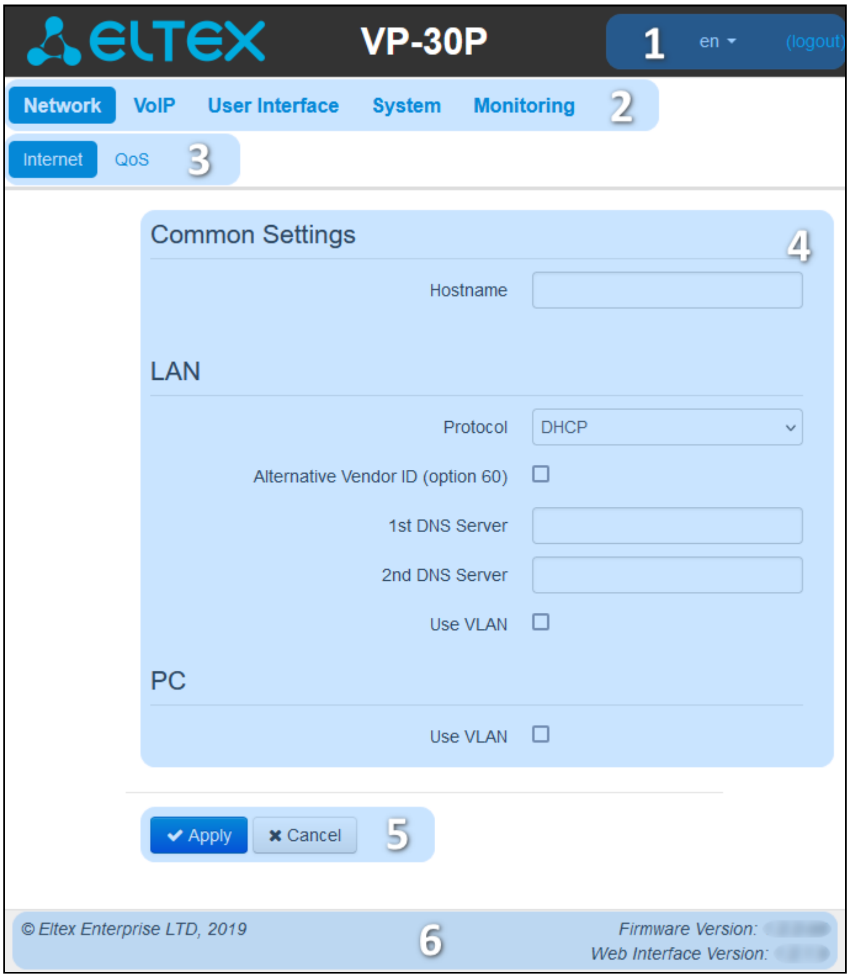
3.1.2.1 Web interface operation modes

Web interface of the VP devices can operate in two modes:

- **Configuration** is a system mode which enables full device configuration.
  The mode has four tabs:
  - Network;
  - VoIP;
  - User Interface;
  - System.
- **Monitoring** is a system monitoring mode which allows viewing various device operation information: Internet connection activity, phone port status, device information, etc.

3.1.2.2 Key elements of the web interface

User interface window is divided into 6 areas (see picture below).



Key elements of the web interface

1. User name for log in, session termination button in the web interface ('*logout*') for the current user and dropped down-menu for language changing.

2. Menu tabs allow you to select configuration and monitoring categories:

- **Network**;
- **VoIP**;
- **User Interface**;
- **System**;
- **Monitoring**.

3. Submenu tabs allow you to control settings field.

4. Device settings field based on the user selection; allows viewing device settings and entering configuration data.

5. Configuration management buttons. For detailed description see 'Applying configuration' submenu.

   - *Apply* — apply and save the current configuration into flash memory of the the device;
   - *Cancel* — discard changes (effective only until '*Apply*' button is clicked).

6. Informational field shows firmware version and web interface version.

3.1.2.3 Applying configuration

'Apply' button appears as follows: 

. Click it to save the configuration into the device flash memory and apply new settings. All settings will be accepted without device restart.

See the following table for detailed information on web interface visual indication of the current status of settings application process:

Appearance	Status description
	When you click the '*Apply*' button, settings will be applied and saved into the device memory. This is indicated by the icon in the tab name and on the '*Apply*' button.
	Successful settings saving and application are indicated by icon in the tab name.
	If the parameter value being specified contains an error, you will see a message with the reason description and icon will appear in the tab name, when you click '*Apply*' button.

3.1.2.4 Discarding changes

Discard changes button appears as follows: 

. Click it to restore values currently stored in the device memory.

Use '*Cancel*' button before clicking '*Apply*' button only. After you click '*Apply*', you will not be able to restore the previous settings.

3.2 Configuring

To move to configuration mode, select one of the following tabs: 'Network', 'VoIP', 'User Interface' or 'System', depending on the configuration goals:

- In the 'Network' menu, the network settings of the device are configured;
- In the 'VoIP' menu, the following is configured: SIP settings, accounts settings, codecs installation, VAS and dialplan settings;
- In the 'User Interface' menu, function keys and sound volume for different device operation modes are configured;
- In the 'System' menu, system, time, access to the device via different protocols are configured, passwords can be changed, firmware can be updated. Also, logging, autoupdate and LLDP protocol can be configured in this menu.

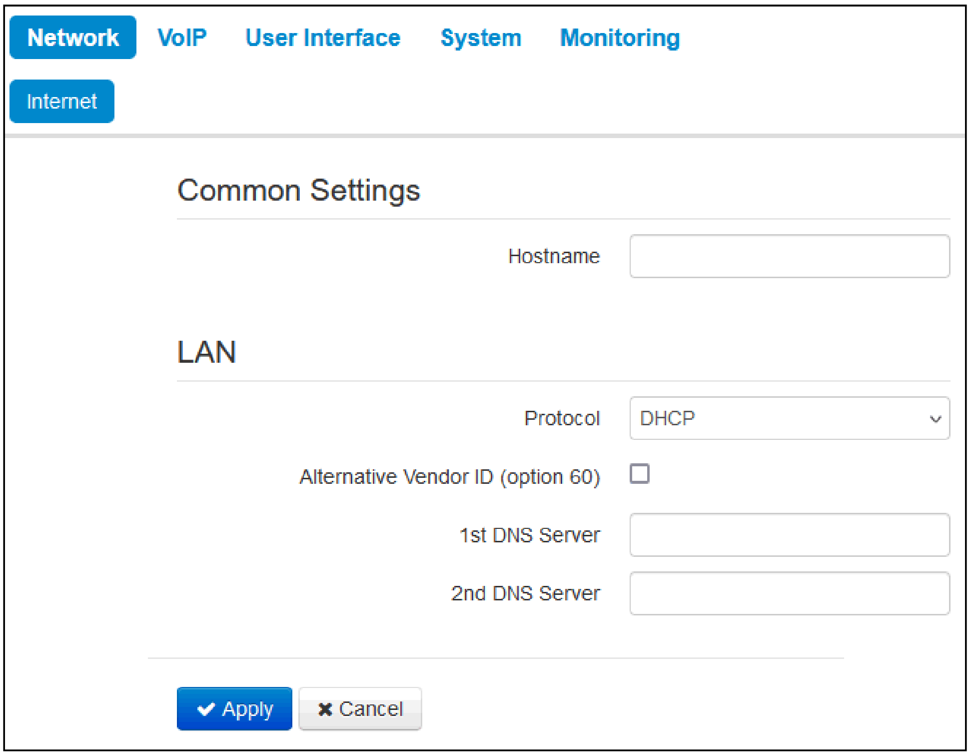
Configuration mode elements:

3.2.1 'Network' menu

In the 'Network' menu, the network settings of the device are configured.

3.2.1.1 'Internet' submenu

In the 'Internet' submenu, you can configure LAN via DHCP and Static.



3.2.1.1.1 Common settings

- *Hostname* — device network name.

3.2.1.1.2 LAN

- *Protocol* — select the protocol that will be used for device LAN interface connection to a data network:
  - *Static* — operation mode where IP address and all the necessary parameters for LAN interface are assigned statically;
  - *DHCP* — operation mode where IP address, subnet mask, DNS address, default gateway and other necessary settings for network operation are automatically obtained from DHCP server.

3.2.1.1.2.1 Static protocol

When 'Static' type is selected, the following parameters will be available for editing:

- *IP Address* — specify the device LAN interface IP address in the data network;
- *Netmask* — external subnet mask;
- *Default gateway* — address that the packet will be sent to, when route for it is not found in the routing table;
- *1st DNS Server, 2nd DNS Server* — domain name server addresses (allow identifying the IP address of the device by its domain name). You can leave these fields empty, if they are not required;
- *MTU*[1] — maximum size of the data unit transmitted on the network.

3.2.1.1.2.2 DHCP protocol

When 'DHCP' type is selected, the following parameters will be available for editing:

- *Alternative Vendor ID (Option 60)* — when selected, the device transmits Vendor ID (Option 60) field value in Option 60 DHCP messages (Vendor class ID). When not selected, a default value is transmitted in Option 60 in the following format:
  - **[VENDOR:** device vendor**][DEVICE:** device type**][HW:** hardware version**][SN:** serial number**][WAN:** WAN interface MAC address**][LAN:** LAN interface MAC address**][VERSION:** firmware version**]** Example: [VENDOR:Eltex][DEVICE:VP-17P][HW:2.0][SN:VI23000118] [WAN:A8:F9:4B:03:2A:D0] [LAN:02:20:80:a8:f9:4b][VERSION:#1.2.2].
  - *Vendor ID (Option 60)* — option 60 value (Vendor class ID) which is transmitted in DHCP messages. When the field is empty, option 60 is not transmitted in DHCP messages;
- *1st DNS Server, 2nd DNS Server* — domain name server addresses (allow identifying the IP address of the device by its domain name). Addresses, which are specified statically, have the higher priority than addresses obtained via DHCP;
- *MTU*[1] — maximum size of the data unit transmitted on the network.

You can manually assign the list of used DHCP options on each network interface.

> [1] The feature will be available in future firmware versions.

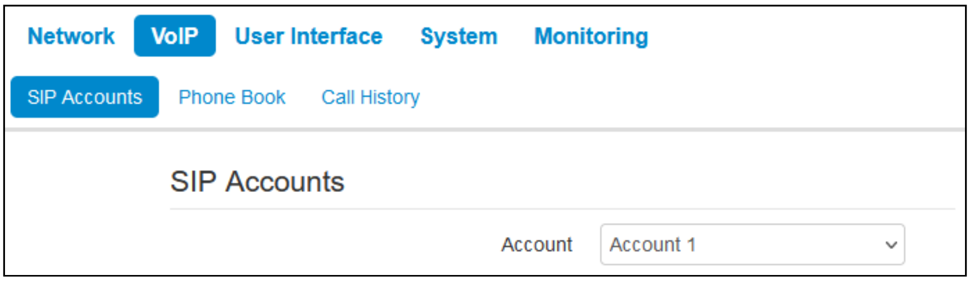
3.2.2 'VoIP' menu

In the 'VoIP' menu you can configure VoIP (Voice over IP):

- SIP protocol configuration;
- Account configuration;
- Codec installation;
- VAS configuration.

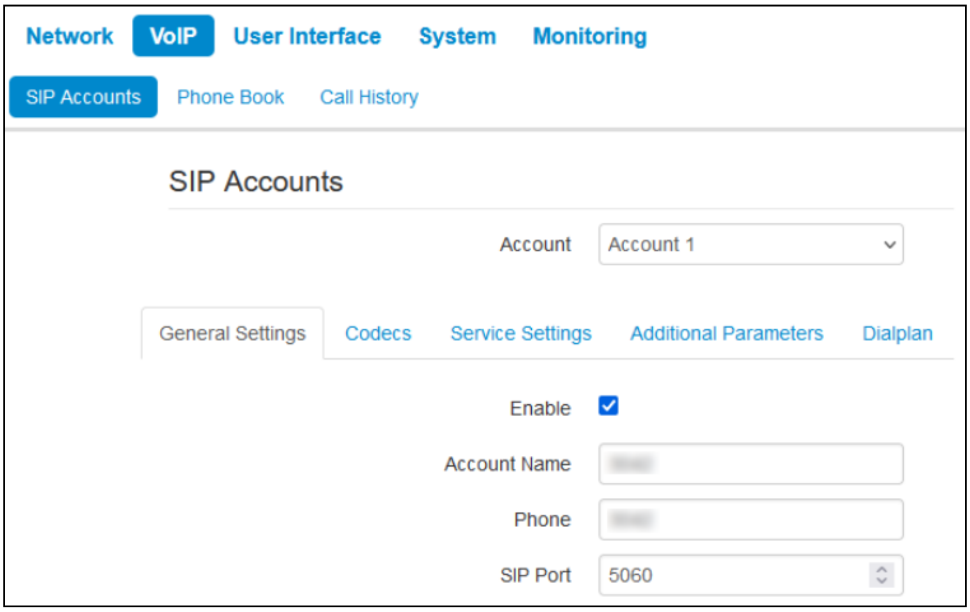
3.2.2.1 'SIP Accounts' submenu

Use drop-down 'Account' menu to select account for editing.



You can assign own SIP server addresses, registration servers, voice codecs, individualised dialing plan and other parameters for each account.

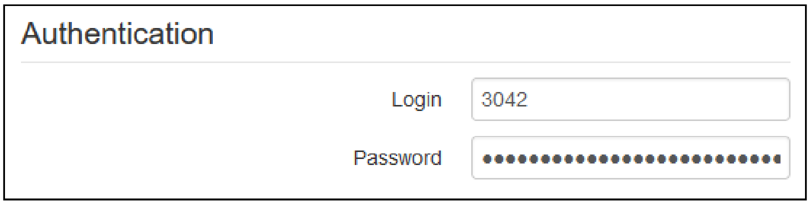
3.2.2.1.1 General settings



Enable — when selected, account is active;

- *Account Name* — an account tag, which will be used for identifying active account or account by default;
- *Phone* — subscriber number assigned to the account;
- *SIP Port* — UDP port for incoming SIP message reception for this account, and for outgoing SIP message transmission from this account. It can take values from 1 to 65535 (default value: 5060).

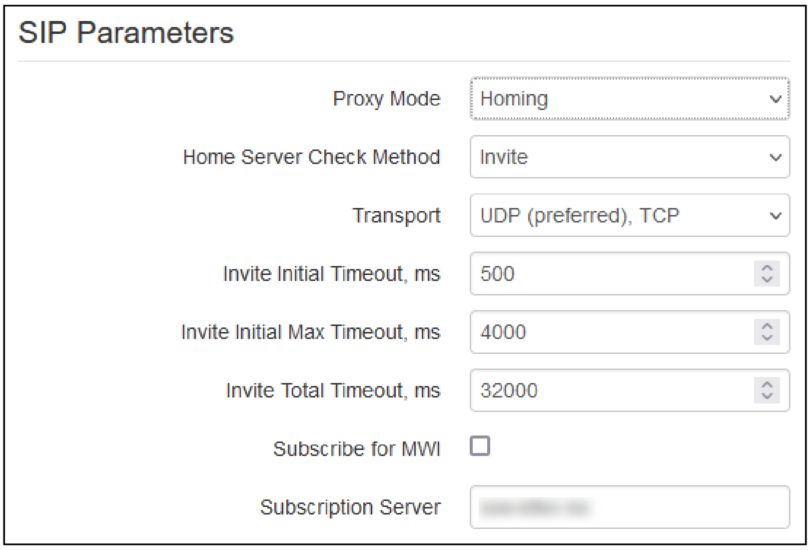
3.2.2.1.1.1 Authentication



- *Login* — user name used for subscriber authentication on SIP server and on registration server;
- *Password* — password used for subscriber authentication on SIP server and on registration server.

3.2.2.1.1.2 SIP parameters

Use '*SIP Parameters*' section to configure SIP parameters of the account.



Proxy Mode — you can select SIP server operation mode in the drop-down list:

- *Off;*
- *Parking* — SIP-proxy redundancy mode without main SIP-proxy management;
- *Homing* — SIP-proxy redundancy mode with main SIP-proxy management.

The phone can operate with a single main SIP-proxy and up to three redundant SIP-proxies. For exclusive operations with the main SIP-proxy, *'Parking'* and *'Homing'* modes are identical. In this case, if the main SIP-proxy fails, it will take time to restore its operational status.

For operations with redundant SIP-proxies, '*Parking*' and '*Homing*' modes will work as follows:

The gateway sends INVITE message to the main SIP-proxy address when performing outgoing call, and REGISTER message when performing registration attempt. If on expiration of '*Invite Total Timeout*' there is no response from the main SIP-proxy or response 408 or 503 is received, the phone sends INVITE (or REGISTER) message to the first redundant SIP-proxy address. If it is not available, the request is forwarded to the next redundant SIP-proxy and so forth. When available redundant SIP-proxy is found, registration will be renewed on that SIP-proxy.

Next, the following actions will be available depending on the selected redundancy mode:

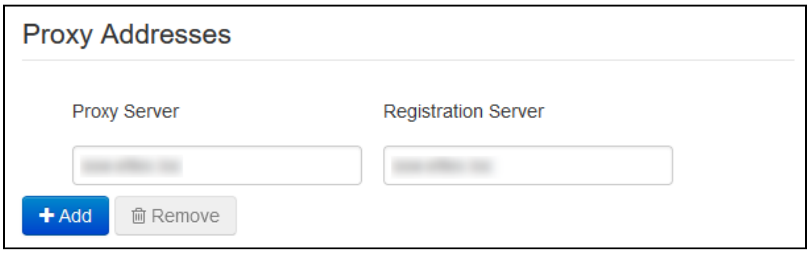
In the '*Parking*' mode, the main SIP-proxy management is absent, and the phone will continue operation with the redundant SIP-proxy even when the main proxy operation is restored. If the connection to the current SIP-proxy is lost, querying of the subsequent SIP-proxies will be continued using the algorithm described above. If the last redundant SIP-proxy is not available, the querying will continue in a cycle, beginning from the main SIP-proxy.

In the '*Homing*' mode, three types of the main SIP-proxy management are available: periodic transmission of OPTIONS messages to its address, periodic transmission of REGISTER messages to its address, or transmission of INVITE request when performing outgoing call. First of all, INVITE request is sent to the main SIP-proxy, and if it is unavailable, then the next redundant one, etc. Regardless of the management type, when the main SIP-proxy operation is restored, the gateway will use it to renew its registration. The gateway will begin operation with the main SIP-proxy.

- *Home Server Check Method* — select availability control method for the primary SIP server in *'Homing'* mode:
    - *Invite* — control via transmission of INVITE request to its address when performing an outgoing call;
    - *Register* — control via periodic transmission of REGISTER messages to its address;
    - *Options* — control via periodic transmission of OPTIONS messages to its address.
- *Home Server Keepalive Timeout* — periodic message transmission interval in seconds; used for home SIP server availability check.
- *Transport* — select protocol for SIP messages transport;
- *Invite Initial Timeout, ms* — a time interval between first INVITE transmission and the second one in case there is no answer on the first INVITE (ms). For the following INVITE requests (third, forth, etc.) the interval will be increased twice (i.e. if the value is 300 ms, the second INVITE will be sent in 300 ms, the third — in 600 ms, the forth — in 1200 ms, etc.);
- *Invite Initial Max Timeout, ms* — the maximum time interval for retransmitting non-INVITE requests and responses on INVITE requests;
- *Invite Total Timeout, ms* — common timeout of INVITE requests transmissition (ms). When the timeout is expired, it is defined that the route is not available. INVITE requests retranslation is limited for availability definition as well;
- *Subscribe for MWI* — when selected, the subscription request on 'message-summary' events is sent. After obtaining such request, subscription server will notify the device on new voice messages through sending NOTIFY requsts;
- *Subscriprion Server* — a network address, to which SUBSCRIBE requests are sent for subscription on 'message-summary' and 'dialog' events. It is possible to specify IP address as well as domain name (after colon, specify a UDP port of SIP server, default value is 5060).

When using different timer values on accounts, the SIP ports of the accounts must also be different.

3.2.2.1.1.3 Proxy Addresses



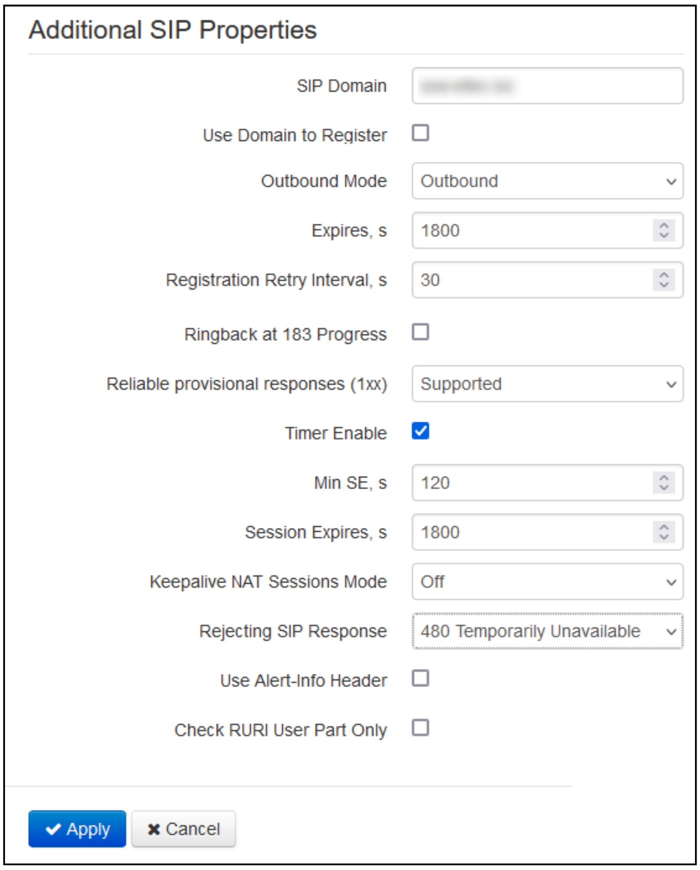
To add the main SIP proxy and registration server, enter the following settings:

- *Proxy Server* — network address of the SIP server: device that controls access of all subscribers to the provider's telephone network. You can specify an IP address as well as a domain name (specify SIP server UDP port after the colon, default value is 5060);
- *Registration Server* — network address of the device which registers all subscribers of the telephone network in order to grant them the right to use communication services (specify registration server UDP port after the colon, default value is 5060). You can specify an IP address as well as a domain name (specify UDP port after the colon, default value is 5060). Usually, the registration server is physically located with the SIP proxy server (they have the same addresses).

To add redundant SIP proxy, click '*Add*' button and enter the requested settings.

To delete the redundant SIP proxy, select the checkbox next to the specified address and click *'Remove'* button.

3.2.2.1.1.4 Additional SIP Properties



- *SIP Domain* — domain where the device is located (fill in, if needed);
- *Use Domain to Register* — when selected, apply SIP domain for registration (SIP domain will be inserted into the 'Request-Line' of REGISTER requests)
- *Outbound Mode*:
  - *Off* — calls will be routed according to the dialplan;
  - *Outbound* — dialplan is required for outgoing communications; however, all calls will be routed via SIP server; if there is no registration, PBX response will be sent to the subscriber in order to enable subscriber service management (VAS management).
- *Expires, s* — valid time of account registration on SIP server. At the average, account registration renewal will be performed after 2/3 of the specified period;
- *Registration Retry Interval, s* — time interval between unsucceful attempt of SIP server registration and the next try;
- *Ringback at 183 Progress* — when selected, 'ringback' tone will be sent upon receiving '183 Progress' message (w/o enclosed SDP);
- *Reliable provisional responses (1xx) (100rel)* — use reliable provisional responses (RFC3262):
  - *Supported* — reliable provisional responses are supported;
  - *Required* — reliable provisional responses are required;
  - *Off* — reliable provisional responses are disabled.

SIP protocol defines two types of responses for connection initiating requests (INVITE) — provisional and final. 2xx, 3xx, 4xx, 5xx and 6xx-class responses are final and their transfer is reliable, with ACK message confirmation. 1xx-class responses, except for *100 Trying* response, are provisional and transferred unreliable, without confirmation (RFC3261). These responses contain information on the current INVITE request processing step, therefore loss of these responses is unacceptable. Utilization of reliable provisional responses is also stated in SIP (RFC3262) protocol and defined by *100rel* tag presence in the initiating request. In this case, provisional responses are confirmed with PRACK message.

100rel setting operation for outgoing communications:

- *Supported* — send the following tag in INVITE request — *supported: 100rel*. In this case, communicating gateway can transfer provisional responses reliably or unreliably — as it deems fit;
- *Required* — send the following tags in INVITE request — *supported: 100rel* and *required: 100rel*. In this case, communicating gateway should perform reliable transfer of provisional replies. If communicating gateway does not support reliable provisional responses, it should reject the request with message 420 and provide the following tag — *unsupported: 100rel*. In this case, the second INVITE request will be sent without the following tag — *required: 100rel*;
- *Off* — do not send any of the following tags in INVITE request — *supported: 100rel* and *required: 100rel*. In this case, communicating gateway will perform unreliable transfer of provisional replies.

100rel setting operation for incoming communications:

- *Supported, Required* — when the following tag is received in INVITE request — *supported: 100rel*, or *required: 100rel* — perform reliable transfer of provisional replies. If there is no *supported: 100rel* tag in INVITE request, the gateway will perform unreliable transfer of provisional replies;
- *Off* — when the following tag is received in INVITE request — *required: 100rel*, reject the request with message 420 and provide the following tag — *unsupported: 100rel*. Otherwise, perform unreliable transfer of provisional replies.

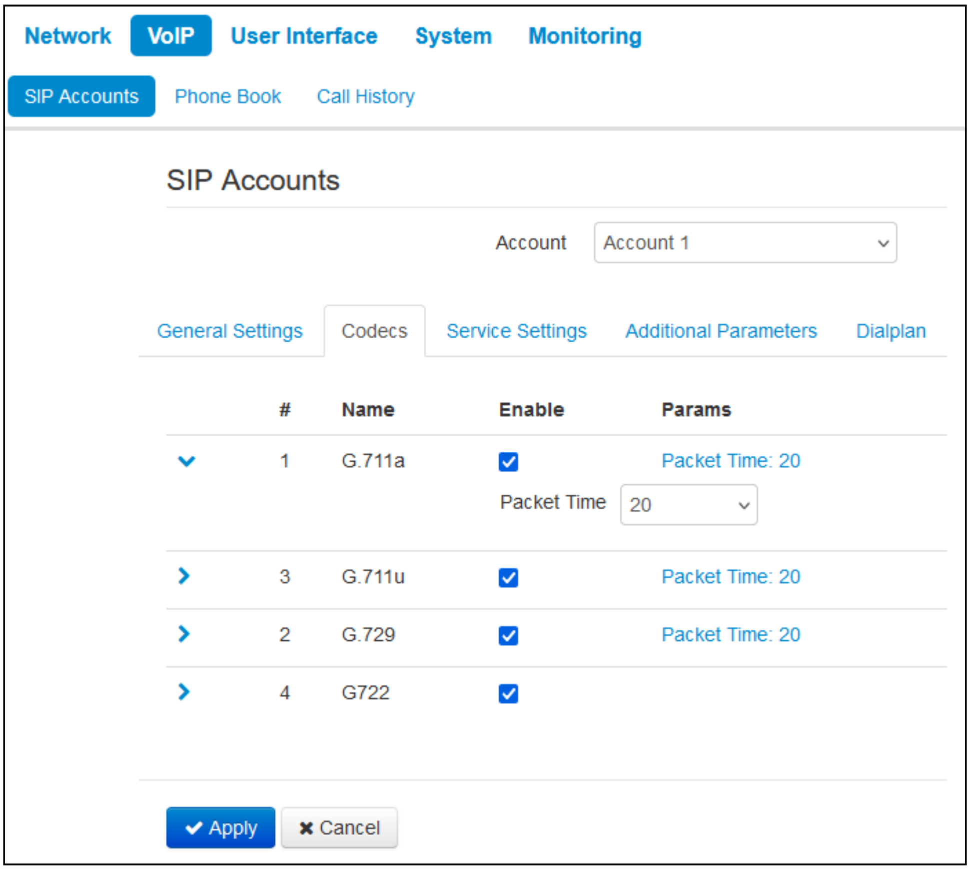
- *Timer Enable* — when selected, the 'timer' (RFC 4028) extension support is enabled. When connection is established, and both sides support 'timer' extension, one of them periodically sends re-INVITE requests for connection monitoring purposes (if both sides support UPDATE method, wherefore it should be specified in the 'Allow' header, the session update is performed by periodic transmission of UPDATE messages);
  - *Min SE, s* — minimal time interval for connection health checks in seconds (90 to 1800 s, 120 s by default);
  - *Session Expires, s* — period of time in seconds that should pass before the forced session termination if the session is not renewed in time (90 to 80000 s, recommended value — 1800 s, 0 — unlimited session);
- *Keepalive NAT Sessions Mode* — select SIP server polling method:
  - *Off* — SIP server will not be polled;
  - *Options* — SIP server polling with OPTIONS message;
  - *Notify* — SIP server polling with NOTIFY message;
  - *CLRF* — SIP server polling with an empty UDP packet.

- *Keepalive Timeout, s* — period of time in seconds that should pass before SIP server is polled. Available when '*Keepalive NAT Sessions Mode*' is on;
- *Rejecting SIP Response* — select SIP response on incoming call rejection;
- *Use Alert-Info Header*[1] — process INVITE request *'Alert-Info'* header to send a non-standard ringing to the subscriber port;
- *Check RURI User Part Only* — when selected, only subscriber number (user) will be analyzed, and if the number matches, the call will be assigned to the subscriber port. When cleared, all URI elements (user, host and port — subscriber number, IP address and UDP/TCP port) will be analyzed upon receiving an incoming call. If all URI elements match, the call will be assigned to the subscriber port.

> To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

> [1] The feature will be available in future firmware versions.

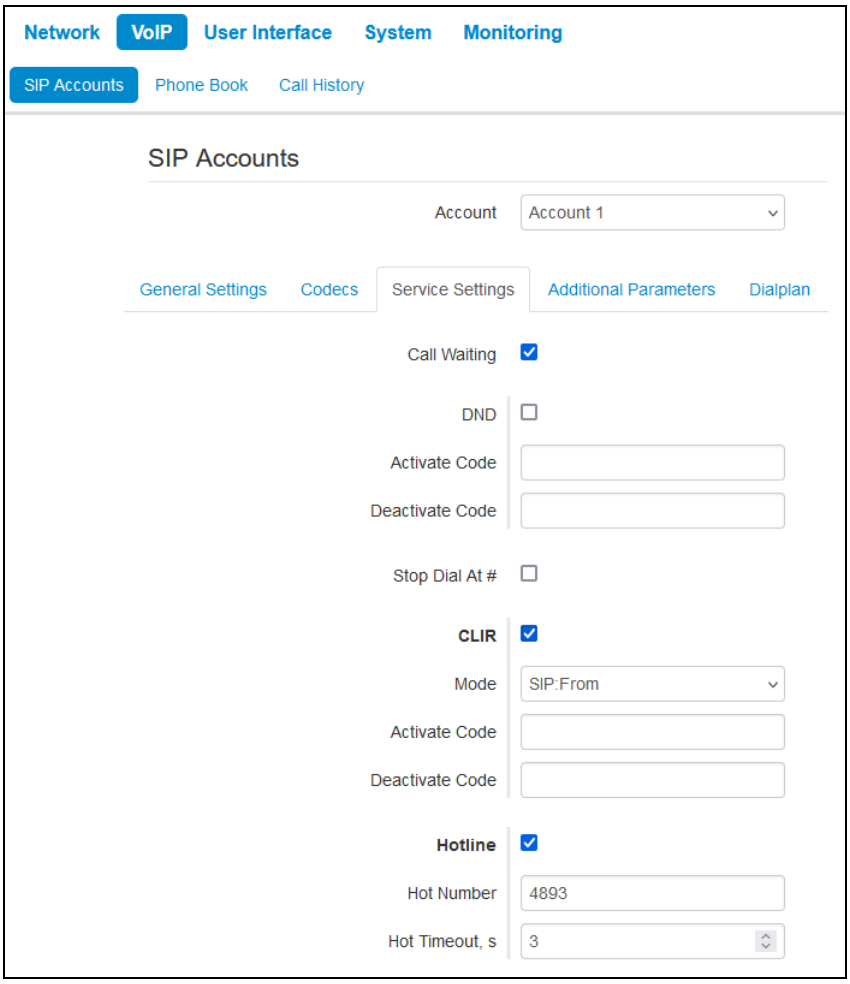
3.2.2.1.2 Codecs



- *Codec 1..5* — you can select codecs and an order of their usage. The highest priority codec should be dragged to the top of the list. For operation, you should select the checkbox '*Enable*' at least for one codec:
  - G.711a — use G.711A codec;
  - G.711u — use G.711U codec;
  - G.722 — use G.722 codec;
  - G.729 — use G.729 codec.
- *Params*:
  - *Packet Time* — amount of voice data in milliseconds (ms) transmitted in a single RTP packet for the corresponding codec G.711A, G.729, and G.726.

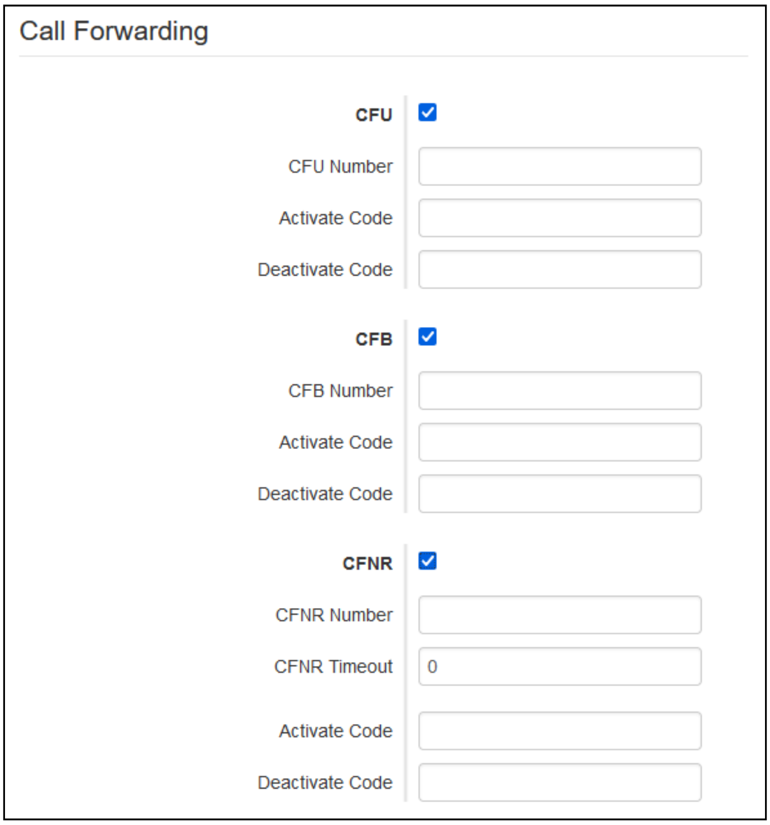
To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.2.1.3 Service settings



- *Call Waiting* — when selected, the subscriber will accept incoming calls while being in a call state, otherwise '484 Busy here' reply will be sent;
- *DND* — when selected, temporary restriction is placed for incoming calls (DND service — Do Not Disturb);
  - Activate Code — activation code for DND service on remote server;
  - Deactivate Code — deactivation code for DND service on remote server.
- *Stop Dial At #* — when selected, use '#' button on the phone unit to end the dialing, otherwise '#' will be recognized as a part of a number;
- *CLIR* — when selected, limitation of caller number identification is enabled:
  - *Mode:*
    - *SIP:From — Anonymous sip: anonymous@unknown.host* will be transmitted in the *'From'* header of SIP messages*;*
    - SIP:From and SIP:Contact — *Anonymous sip: anonymous@unknown.host* will be transmitted in the *'From'* and *'Contact'* headers of SIP messages.
  - Activate Code — activation code for CLIR service on remote server;
  - Deactivate Code — deactivation code for CLIR service on remote server.
- *Hotline* — when selected, 'Hotline' service is enabled. This service enables an outgoing connection automatically without dialing the number after the phone handset is picked up with the defined delay (in seconds). When selected, fill in the following fields:
  - *Hot Number* — phone number that will be used for connection establishment upon '*Hot timeout*' expiration after the phone handset is picked up (in SIP profile being used, a prefix for this direction should be defined in the dilaplan);
  - *Hot Timeout, s* — time interval that will be used for connection establishment with the opposite subscriber, in seconds.

3.2.2.1.3.1 Forwarding



- *CFU* — when selected, CFU (Call Forwarding Unconditional) service is enabled — all incoming calls will be forwarded to the specified *CFU Number*:
  - *CFU Number* — number that all incoming calls will be forwarded to when CFU service is enabled (in SIP profile being used, a prefix for this direction should be defined in the dialplan);
  - Activate Code — activation code for CFU service on remote server;
  - Deactivate Code — deactivation code for CFU service on remote server.

Call Forwarding Unconditional service can be configured via the phone display menu: Menu → Services → Account 1..6 → Call forwarding → Unconditional.

- *CFB* — when selected, CFB (Call Forwarding Busy) service is enabled — call forwarding to the specified *CFNR Number*, when the subscriber is busy:
  - *CFB Number* — number that incoming calls will be forwarded to when the subscriber is busy and CFB service is enabled (in SIP profile being used, a prefix for this direction should be defined in the dialplan);
  - Activate Code — activation code for CFB service on remote server;
  - Deactivate Code — deactivation code for CFB service on remote server.

Call Forwarding Busy service can be configured via the phone display menu: Menu → Services → Account 1..6 → Call forwarding → Busy.

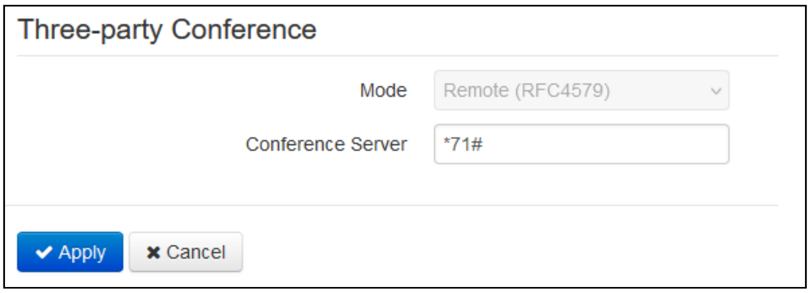
- *CFNR* — when selected, CFNR (Call Forwarding No Reply) service is enabled — call forwarding, when there is no answer from the subscriber:
  - *CFNR Number* — number that incoming calls will be forwarded to when there is no answer from the subscriber and CFNR service is enabled (in SIP profile being used, a prefix for this direction should be defined in the dialplan);
  - *CFNR Timeout* — time interval that will be used for call forwarding when there is no answer from the subscriber, in seconds;
  - Activate Code — activation code for CFNR service on remote server;
  - Deactivate Code — deactivation code for CFNR service on remote server.

Call Forwarding No Reply service can be configured via the phone display menu: Menu → Services → Account 1..6 → Call forwarding → No answer.

When multiple services are enabled simultaneously, the priority will be as follows (in the descending order):

1. CFU;
2. DND;
3. CFB, CFNR.

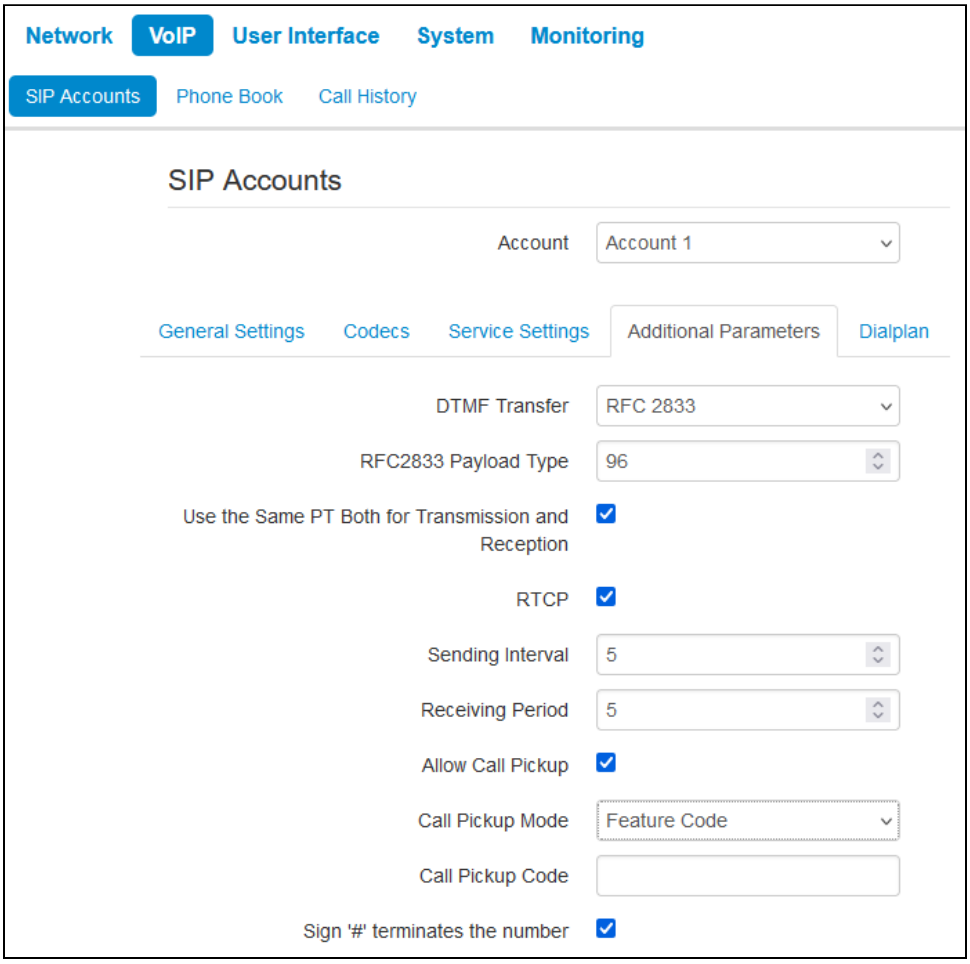
3.2.2.1.3.2 Three-party conference



- *Mode* — operation mode of three-party conference. Two modes are possible:
  - *Remote (RFC 4579)* — conference assembly is performed at the remote server; after pressing 'CONF' button, 'Invite' message will be sent to the server using number specified in the 'Conference Server' field. In this case, conference operation complies with the algorithm described in RFC 4579.
- *Conference Server* — in general, address of the server that establishes conference using algorithm described in RFC 4579. Address is specified in the following format SIP-URI: user@address:port. You can specify the 'user' URI part only — in this case, 'Invite' message will be sent to the SIP proxy address.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.2.1.4 Additional Parameters

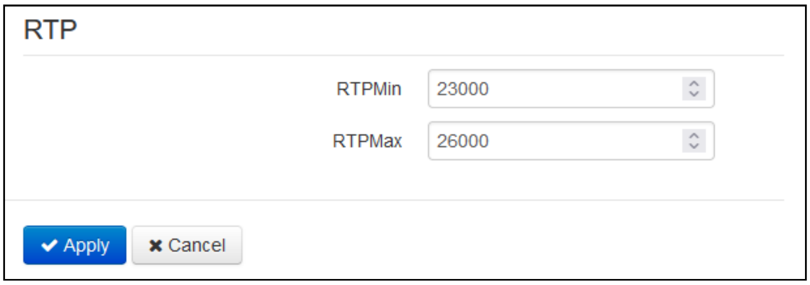


- *DTMF Transfer* — mode of DTMF signal transmission:
  - *Inband* — inband transmission;
  - *RFC2833* — according to RFC2833 recommendation as a dedicated payload in RTP voice packets:
    - *RFC2833 Payload Type* — payload type for packet transmission via RFC2833 (possible values: from 96 to 127);
    - *Use the Same PT Both for Transmission and Reception* — option is used in outgoing calls for payload type negotiation of events sent via RFC2833 (DTMF signals). When selected, event transmission and reception via RFC2833 is performed using the payload from 200Ok message sent by the opposite side. When cleared, event transmission is performed via RFC2833 using the payload from 200Ok being received, and reception — using the payload type from its own configuration (specified in the outgoing *Invite*).
  - *SIP info* — transfer messages via SIP in INFO requests.
- *RTCP* — when selected, use RTCP for voice channel monitoring:
  - *Sending Interval* — RTCP packet transmission period, in seconds;
  - *Receiving Period* — RTCP message reception period, measured in transmission period units; if there is no single RTCP packet received until the reception period expires, the IP Phone will terminate the connection.
- *Allow Call Pickup* — when selected, pressing the BLF key will initiate the interception of the incoming call to the subscriber on which the BLF key is configured;
  - *Call Pickup Mode* — the way the call is intercepted:
    - *Replaces* — call pickup using the Replaces header;

- *Feature Code* — call pickup using the prefix added to the number of the subscriber on which the BLF key is configured:
  - *Call Pickup Code* — prefix which will be added to the number of the subscriber to which the BLF key is configured;
  - *Sign '#' terminates the number* — adding the '#' symbol when intercepting a call after the number of the subscriber to which the BLF key is configured.

BLF is assigend only for buttons with LED indicator. LED indicates line status of the subscriber selected in the *Additional parameters* field in User Interface → Buttons submenu. If the subscriber is in a call (indicator is solid/blinking red), pressing the key initiates call pickup; if the subscriber is not busy (indicator is solid green), the call is initiated.

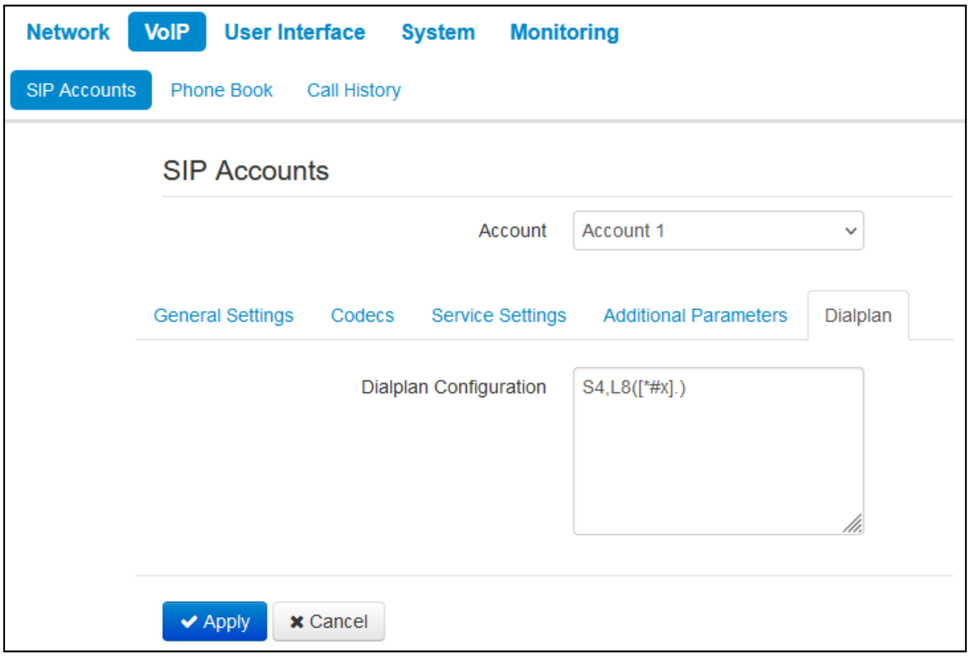
3.2.2.1.4.1 RTP



- *RTPMin* — lower limit of the RTP ports range used for voice traffic transmission;
- *RTPMax* — upper limit of the RTP ports range used for voice traffic transmission.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.2.1.5 Dialplan



To define a dialplan, use regular expressions in the '*Dialplan Configuration*' field. The structure and format of regular expressions that enable different dialing features are listed below.

Structure of regular expressions:

S xx , L xx (Rule1 | Rule2 | ... | RuleN)

where:

- **xx** — arbitrary values of S and L timers;
- **()** — dialplan margins;
- **|** — delimiter for dialplan rules;
- **Rule1, Rule 2, Rule N** — numbers templates which are allowed or forbidden to be called.

Routing rules structure:

Sxx Lxx prefix@optional(parameters)

where:

- **xx** — arbitrary value of S and L timer. Timers inside rules could be dropped; in this case, global timer values, defined before the parentheses, will be used.
- **prefix** — prefix part of the rule;
- **@optional** — optional part of the rule (might be skipped);
- **(parameters)** — additional options (might be skipped).

3.2.2.1.5.1 Timers

- *Interdigit Long Timer ('L' character in a dialplan record)* — entry timeout for the next digit, if there are no templates that correspond to the dialed combination.
- *Interdigit Short Timer ('S' character in a dialplan record)* — entry timeout for the next digit, if the dialed combination fully matches at least one template and if there is at least one template that requires an extension dialing for the full match.

The timers values might be assigned either for the whole dialplan or for a certain rule. The timers values specified before round brackets are applied for the whole dialplan.

<u>Example:</u> S4 (8XXX.) or S4, L8 (XXX)

If the values of timers are specified in a rule, they are applied to this rule. The value might be located at any position in a template.

<u>Example:</u> (S4 8XXX. | XXX) or ([1-5] XX S0) — an entry requests instant call transmission when 3-digit number dialing; a number should begin with 1,2, … ,5.

3.2.2.1.5.2 Prefix part of the rule

Prefix part might consist of the following elements:

Prefix part elements	Description
X or x	Any digit from 0 to 9, equivalent to [0-9] range.
0 - 9	Digits from 0 to 9.
*	Symbol *.
#	Symbol #. The use of '#' symbol in a dialplan can cause blocking of dial completion with the help of '#' key!

Prefix part elements	**Description**
[ ]	Specify a range (using dash), enumeration (without spaces, comas and other symbols between digits) or combination of range and enumeration. Example of a range: ([1-5]) — any digit from 1 to 5. Example of enumeration : ([1239]) — any digit out of 1, 2, 3 or 9. Example of a range and enumeration combination: ([1-39]) — the same as in the previous example but in another form. The entry corresponds to any digit from 1 to 3 and 9.
{a,b}	Specify the number of reiteration of the symbol placed before round brackets, range or *# symbols. The following entries are possible: • {,max} — equal to {0,max}, • {min,} — equal to {min,∞}. Where: • min — minimum number of reiteration, • max — maximum. Example 1: 6{2,5} — 6 might be dialed from 2 to 5 times. The entry equals to the followings: 66 | 666 | 6666 | 66666 Example 2: 8{2,} — 8 might be dialed 2 and more times. The entry equals to the followings: 88 | 888 | 8888 | 88888 | 888888 | ... Example 3: 2{,4} — 2 might be dialed up to 4 times. The entry equals to the followings: 2 | 22 | 222 | 2222.
.	Special symbol 'dot' defines the possibility of reiteration of the previous digit, range or *# symbols from 0 ad infinitum times. It is equal to {0,} entry. Example: 5x.* — x in this rule can either be absent or present as many times as needed. It is equal to 5* | 5x* | 5xx* | 5xxx* | ...
+	Special symbol 'plus' — repeat the previous digit, range or *# symbols from 1 ad infinitum times. It is equal to {1,} entry. Example: 7x+ — x is supposed to present in the rule at least 1 time. It is equal to 7x | 7xx | 7xxx | 7xxxx | ...
<arg1:arg2>	Replace dialed sequence. The dialed sequence (arg1) in SIP request to SIP server is changed to another one (arg2). The modification allows deleting — <xx:>, adding — <:xx>, or replacing — <xx:xx> of digits and symbols. Example 1: (<9:8383>XXXXXXX) — the entry corresponds the following dialed digits 9XXXXXXX, but in the transmitted request to SIP server, 9 digit will be replaced to 8383 sequence. Example 2: (<83812:>XXXXXX) — the entry corresponds the following dialed digits 83812XXXXXX, but the sequence 83812 will be omitted and will not be transmitted to a SIP server.

Prefix part elements	Description
,	Paste tone to dialing. When ringing to intercity numbers (or to city number using an office phone) usually, you can hear a dial tone. The dial tone can be realized by putting coma at the needed position in a sequence. Example: (8, 770) — while dialing 8770 sequence you will hear a continuous dial tone ('station responce') after dialing 8 digit.
!	Forbid number dialing. If you put '!' symbol at the end of the number template, dialing of numbers corresponding to the template will be blocked. Example: (8 10X xxxxxxx ! | 8 xxx xxxxxxx) — expression allows long-distance dialing only and denies outgoing international calls. Prohibition rules must be written first.

3.2.2.1.5.3 Optional part of rules

The optional part of a rule might be omitted. This part might consist the following elements:

Optional part of rules element	Description
@host:[port]	Direct address dialing (IP Dialing). '@' symbol placed after the number defines that the dialed call which will be sent to the subsequent server address. Also, IP Dialing address format can be used for numbers intended for the call forwarding. If *@host:port* is not specified, calls are routed via SIP-proxy. Example: (**1xxxx@192.168.16.13:5062**) — all five-digit dials, beginning with 1, will be routed to 192.168.16.13 IP address to 5062 port.

3.2.2.1.5.4 Additional parameters

Format: **(param1: value1, .., valueN; .. ;paramN: value1, .., valueN)**

- *param* — parameter name; several parameters are semicolon-separated and all parameters are enclosed in parentheses;
- *value* — parameter value; several values of one parameter are comma-separated.

3.2.2.1.5.5 Examples

Example 1: (8 xxx xxxxxxx) — 11-digit number beginning with 8.

Example 2: (8 xxx xxxxxxx | <:8495> xxxxxxx) — 11-digit number beginning with 8; if 7-digit number is dialed, add 8495 to the number being sent.

Example 3: (0[123] | 8 [2-9]xx [2-9]xxxxxx) — dialing of emergency call numbers and unusual sets of long-distance numbers.

Example 4: (S0 <:82125551234>) — quickly dial the specified number, similar to 'Hotline' mode.

Example 5: (S5 <:1000> | xxxx) — this dialplan allows you to dial any number that contains digits, and if there was no entry in 5 seconds, dial number '1000' (for example, it belongs to a secretary).

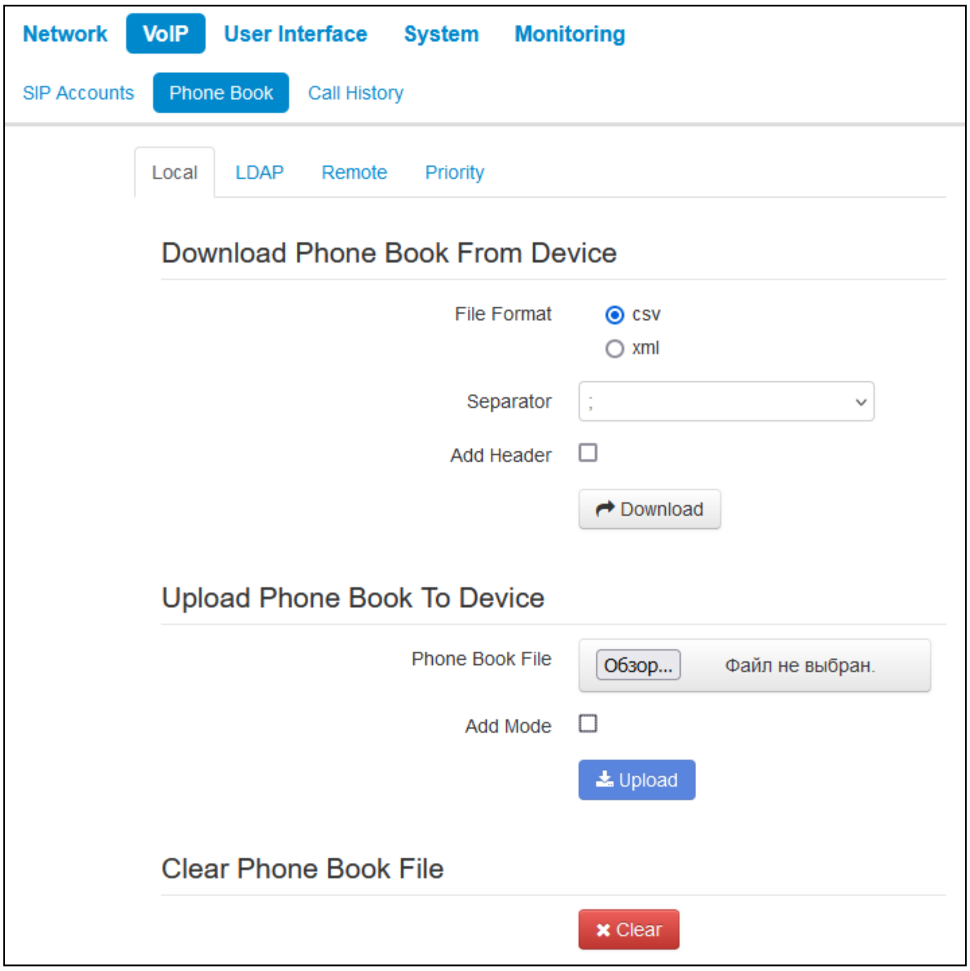
Example 6: (8, 10x.|1xx@10.110.60.51:5060) — this dialplan allows you to dial any number beginning with 810 and containing at least one digit after '810' (after entering '8', 'station reply' tone will be generated) as well as 3-digit numbers beginning with 1. Subscriber calls with 3-digit numbers beginning with 1 will be sent to IP address 10.110.60.51 and port 5060.

Example 7: (S3 *xx#|#xx#|#xx#|*xx*x+#) — managment and usage of VAS.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.2.2 'Phone Book' submenu

3.2.2.2.1 Local phone book management



3.2.2.2.1.1 Download Phone Book From Device

Use the section to download a phone book stored on the device.

- *File Format* — select a format of the file you want to download. The following formats are available:
  - *csv* — text file format where all the contacts are written in the table. The values in the table are separated by the selected separator;
    - *Separator* — the symbol for separating data in the table in csv format;
    - *Add Header* — when selected, downloaded csv file will have a header — the first line.
- *xml* — an eXtensible Markup Language.

3.2.2.2.1.2 Upload Phone Book To Device

This section is used to configure parameters of restoring a phone book from the backup copy.

- *Phone Book File* — choose file;
- *Add Mode* — when selected, the conacts from the uploaded file will be added to existing ones.

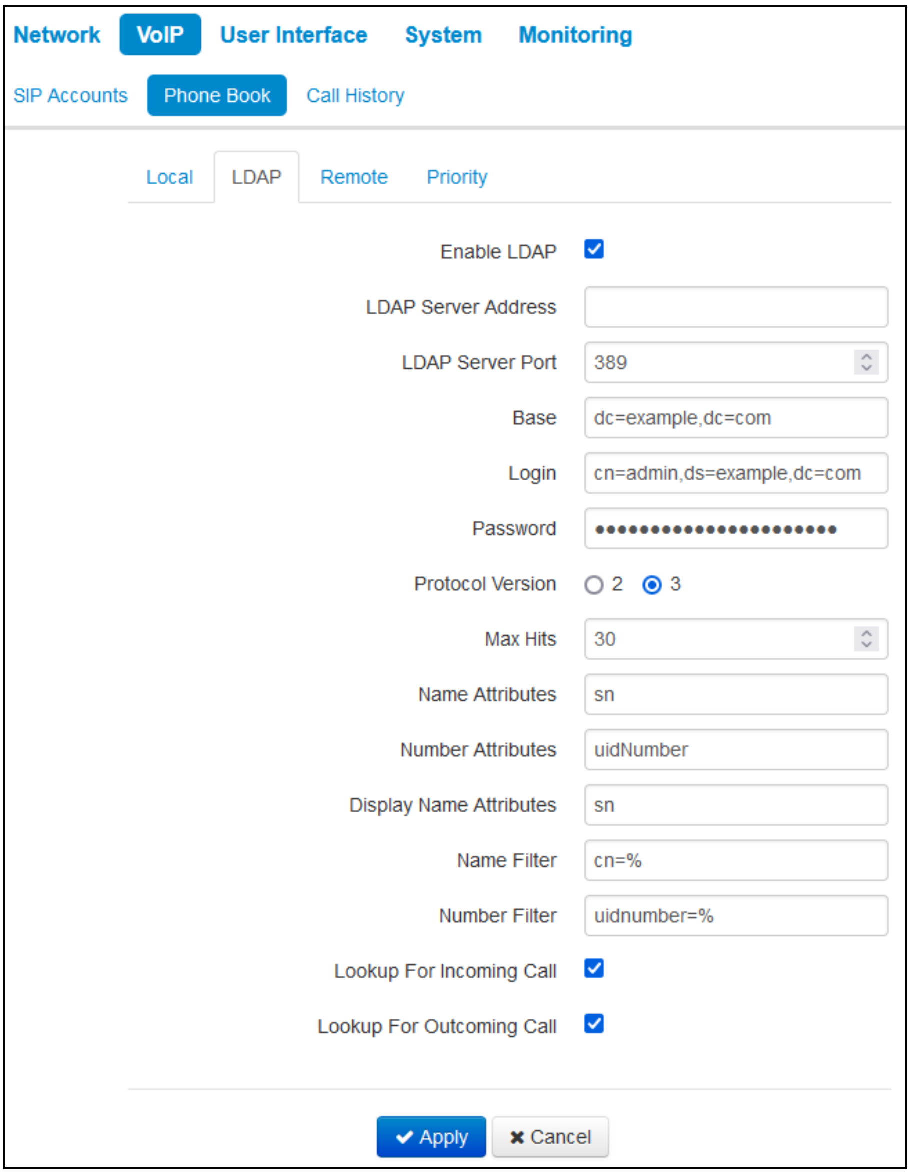
> If 'Add Mode' box is not selected, contacts from the loaded file will replace the existing one.

3.2.2.2.1.3 Clear Phone Book File

To clean the phone book, click '*Clear*' button.

3.2.2.2.2 LDAP Phone Book management

In the 'Phone book' submenu, you can set up the connection to LDAP server and search parameters.



- *Enable LDAP* — when selected, the phone book is accessible via display menu;
  - *LDAP Server Address* — domain name or IP address of LDAP server;
  - *LDAP Server Port* — port of LDAP server transport protocol;

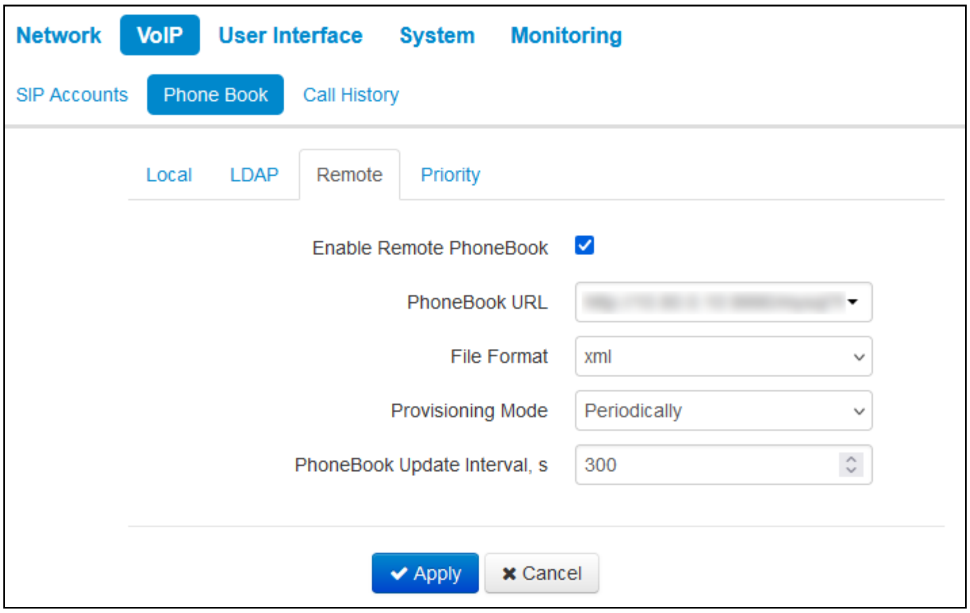
- *Base* — indicates the location of base directory, that contains the phone book, and from which the search begins, in the LDAP directory. Specifying this parameter narrows the search and thereby reduces the time it takes to search for a contact;
- *Login* — username that will be used when authorizing on LDAP server;
- *Password* — password that will be used when authorizing on LDAP server;
- *Protocol Version* — LDAP protocol version of formed requests;
- *Max Hits* — the parameter indicating the maximum amount of search results that will be returned by LDAP server;

Too big ‘Max Hits’ value reduces the LDAP search rate, that is why the parameter is to be configured according to the available bandwidth.

- *Name Attributes* — name attribute of each record returned by the LDAP server;
- *Number Attributes* — number attribute of each record returned by the LDAP server;
- *Display Name Attributes* — display name attribute of each record returned by the LDAP server;
- *Name Filter* — the filter used to lookup for the names. The '*' character in the filter indicates any character. The '%' character in the filter indicates the input string used as the filter condition prefix;
- *Number Filter* — the filter used to lookup for the number. The '*' character in the filter indicates any character. The '%' character in the filter indicates the input string used as the filter condition prefix;
- *Lookup For Incoming Call* — when selected, lookup for a name using a number during incoming calls;
- *Lookup For Outcoming Call* — when selected, lookup for a name using a number during outcoming calls.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

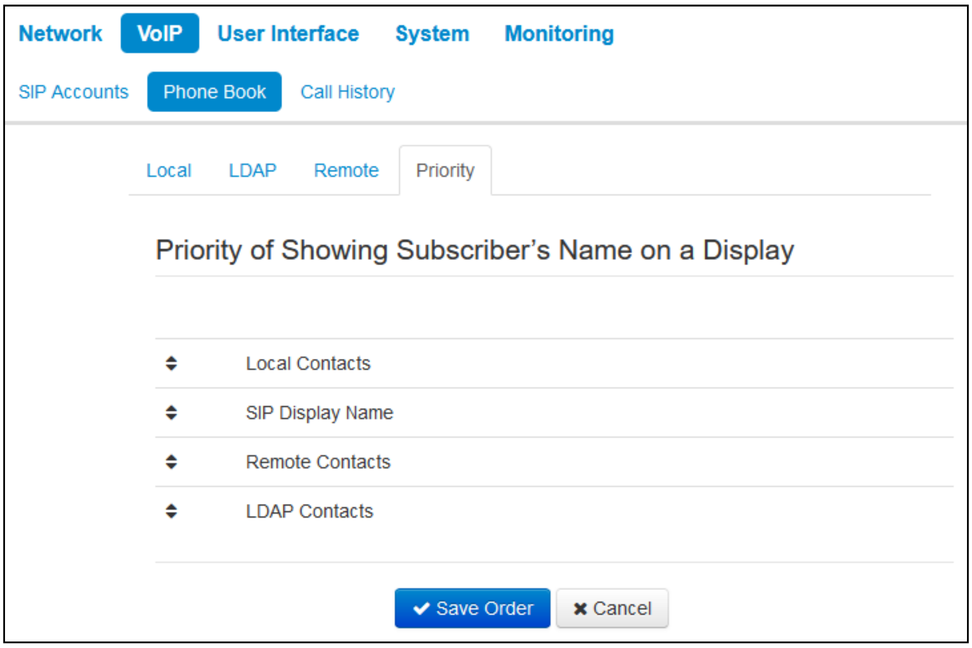
3.2.2.2.3 Remote Phone Book management



- *Enable Remote PhoneBook* — when selected, remote phone book is loaded automatically;
  - *PhoneBook URL* — a full path to the remote phone book — is set in URL format (the following protocols are available to be used for phone book loading through: *TFTP, FTP, HTTP* and *HTTPS*);
  - *File Format* — select a format of the file you want to download. The following formats are available:
    - *csv* — text file format where all the conacts are written in the table. The values in the table are separated by the selected separator;
      - *Separator* — the symbol for separating data in the table in csv format;
      - *Add Header* — when selected, downloaded csv file will have a header — the first line.
    - *xml* — an eXtensible Markup Language.
  - *Provisioning Mode* — select a mode for phone book loading:
    - *Periodically* — the device phone book will be automatically updated after defined period of time;
      - *PhoneBook Update Interval, s* — time interval between phone book updates. If the parameter is set to 0, the phone book is updated once — right after device loading.
    - *Scheduled* — the device phone book will be automatically updated at specific time and on specific days:
      - *Days Of PhoneBook Update* — weekdays when the phone book will be automatically updated;
      - *Time Of PhoneBook Update* — time in 24-hours format, when the phone book will be automatically updated.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.2.2.4 Phone Book Priority management



In the 'Priority' submenu, you can configure the priority for displaying the subscriber's name on the display.

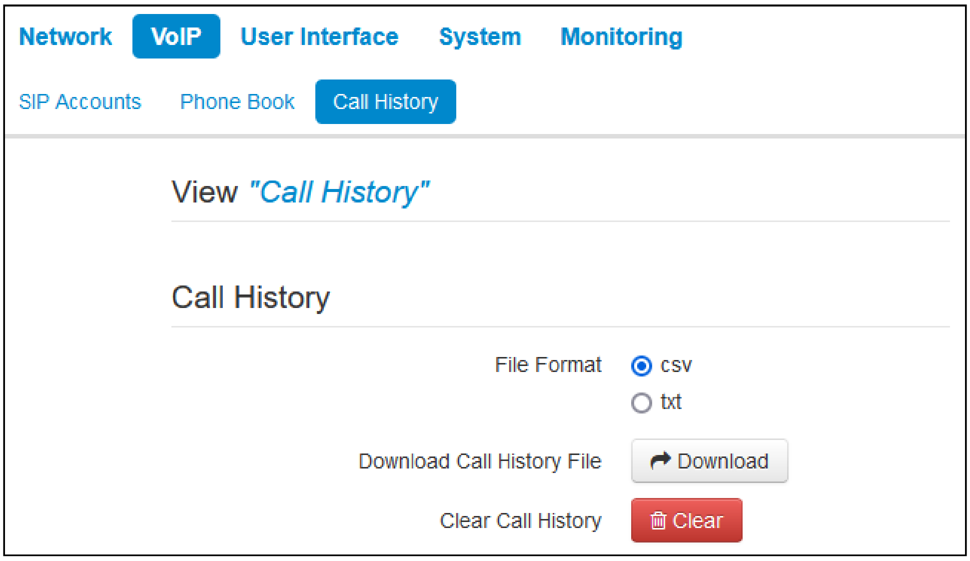
- *Local Contacts* — displaying of names from local pnonebook;
- *SIP Display Name* — displaying of names received via SIP protocol;
- *Remote Contacts* — displaying of names from remote pnonebook;
- *LDAP Contracts* — displaying of names from LDAP pnonebook.

The caller's name will be displayed according to the selected priority. For example, in this case, if the local phone book has the name of the caller, the display will show the name from the local phone book, if not — the name designated in the SIP protocol. If the name is not designated in the SIP protocol, it will be displayed from the remote phone book, etc.

To apply a new configuration and store settings into the non-volatile memory, click *'Save Order'* button. To discard changes, click *'Cancel'* button.

3.2.2.3 'Call History' submenu

In the 'Call History' submenu you can configure call history logging.



- *File Format* — select a format of the file you want to download. The following formats are available:
  - *csv* — text file format where call history is written in a table. The values in the table are separated by the selected separator;
  - *txt* — text file format that contains call history organized by lines.
- *Download Call History File* — to save 'voip_history' file on a local PC, click 'Download' button;
- *Clear Call History* — to clear call history, click *'Clear'* button.

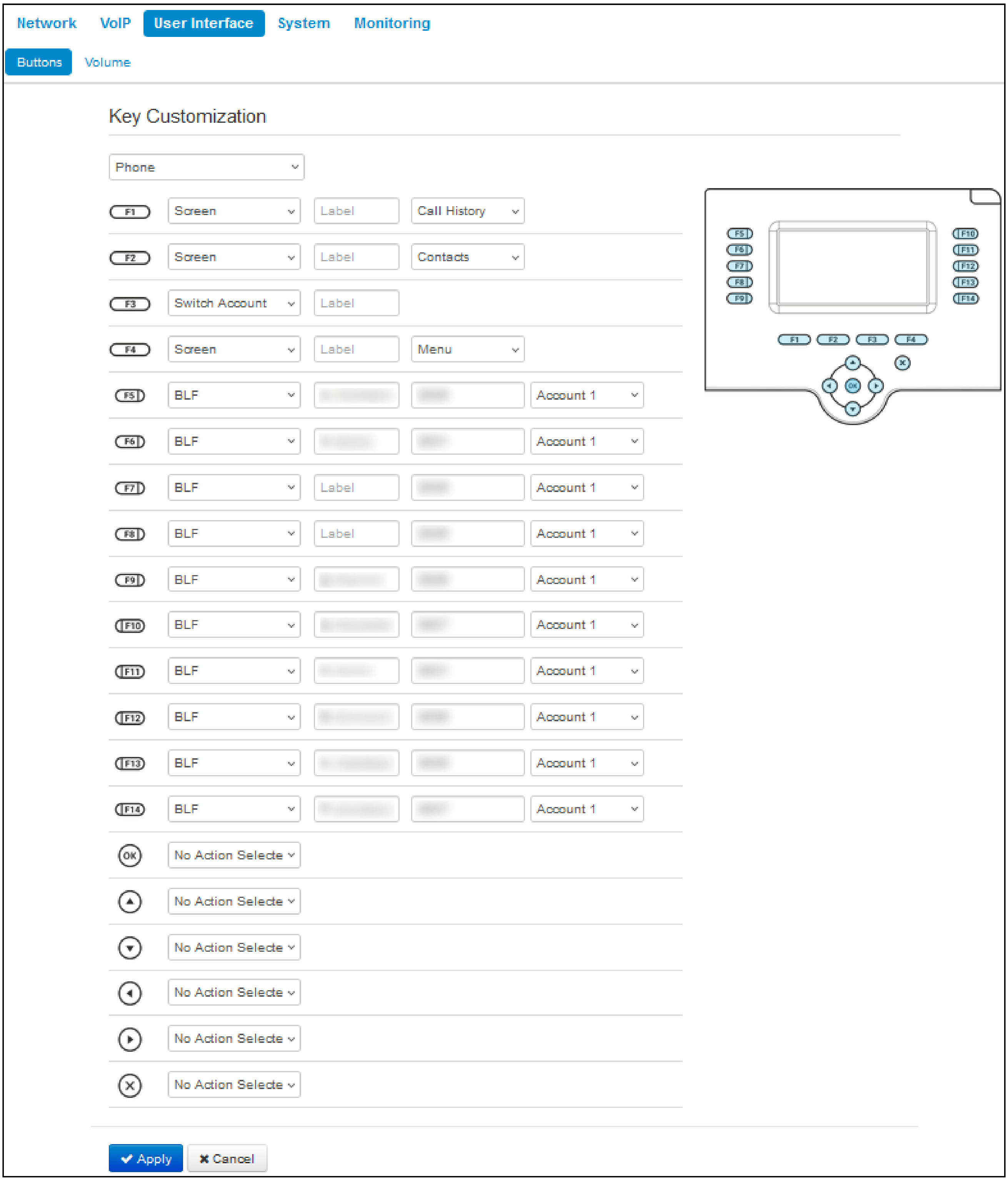
To view the call history, follow the *View 'Call History'* link. For parameter monitoring description, see section 'Viewing call history'.

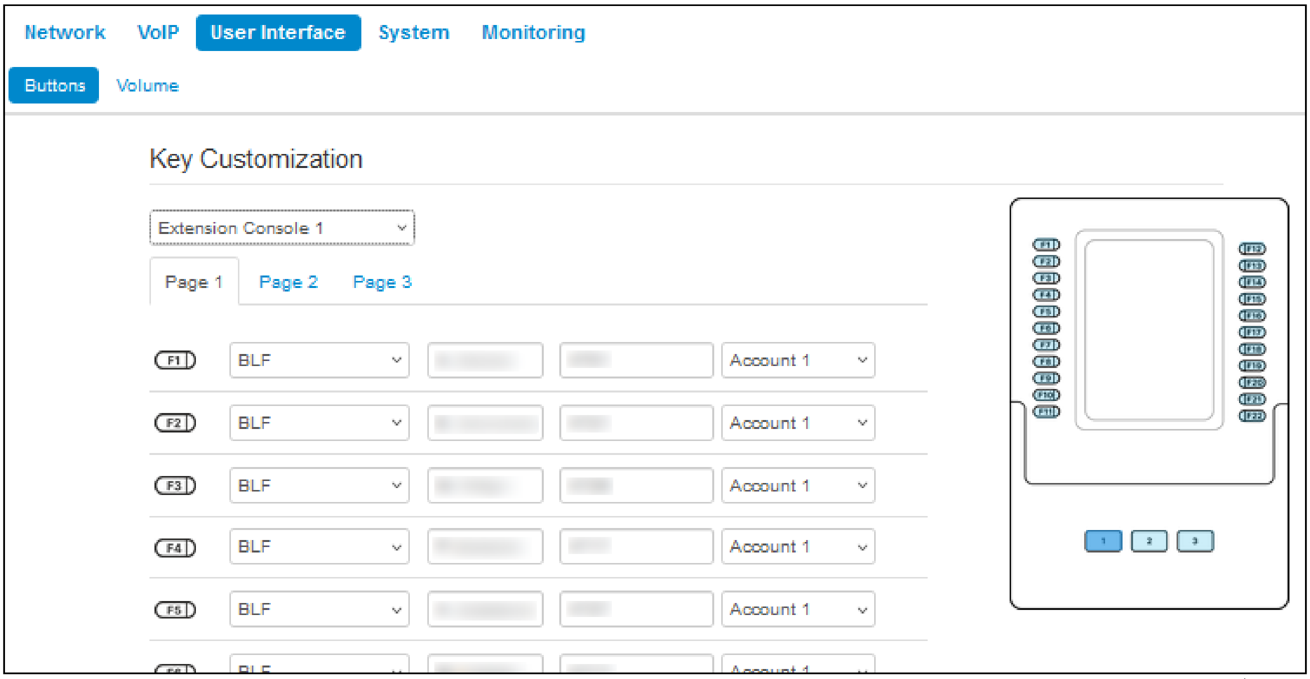
To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.3 'User Interface' menu

3.2.3.1 'Buttons' submenu

This submenu allows you to select the actions that occur when you press keys on the phone and VP-EXT22 extension consoles.





To switch between consoles and phone, select the desired option in the selector above the columns (*Phone, Extension Console 1–3*). Three virtual pages can be configured for consoles.

The settings are presented as a table with the following columns:

1. *Button*;
2. *Action* — select action to be performed on the button pressing. The followings are available:
   a. *No Action Selected* — pressing this button will not be processed;
   b. *Screen* — open a screen selected in the additional parameters;
   c. *Call* — call the number selected in the additional parameters;
   d. *Switch Account* — change the account by default;
   e. *BLF* — pressing the button in stand-by mode initiates a call. In conversation mode, pressing the button redirects the call to the selected subscriber.

   BLF is assigend only for buttons with LED indicator. LED indicates line status of the subscriber selected in the *Additional parameters* field in User Interface → Buttons submenu.

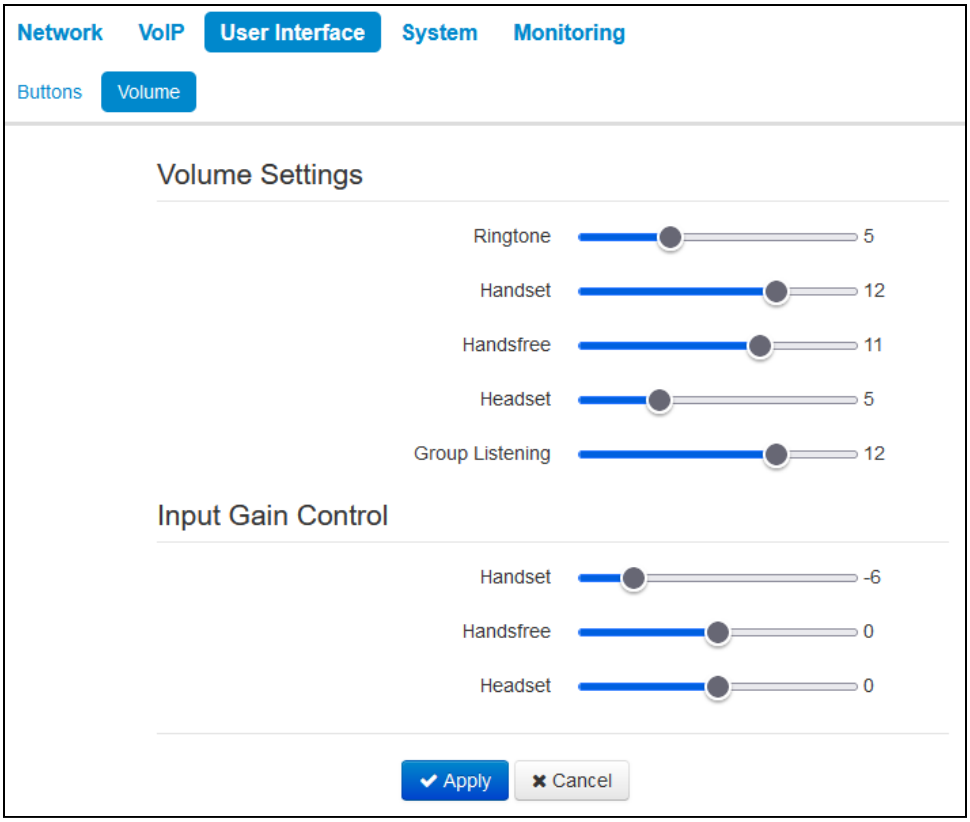
   To activate BLF function, you should specify subscription server in SIP account settings.

   f. *Account* — open the dialer for the specified account;
   g. *Forward* — activate forwarding to a specified number;
   h. *Group Listening* — activate Group listening service.
3. *Label* — button label, which is displayed on the screen next to the button;
4. *Additional parameters* — select additional parameters for the button (options depend on the action selected).

To apply a new configuration and store settings into the non-volatile memory, click '*Apply*' button. To discard changes, click '*Cancel*' button.

3.2.3.2 'Volume' submenu

In the 'Volume' submenu you can configure the volume in various device operation modes.



Volume Settings

- *Ringtone* — ringtone volume:
- *Handset* — handset volume during conversation;
- *Handsfree* — speakerphone volume during conversation;
- *Headset* — headset volume during conversation;
- *Group Listening* — volume during group listening.

Input Gain Control

- *Handset* — specifies the value by which a signal from the handset will be amplified (valid values -9, … 9 dB, at a pitch of 1.5 dB);
- *Handsfree* — specifies the value by which a signal from the speakerphone will be amplified (valid values -9, … 9 dB, at a pitch of 1.5 dB);
- *Headset* — specifies the value by which a signal from the headset will be amplified (valid values -9, … 9 dB, at a pitch of 1.5 dB).

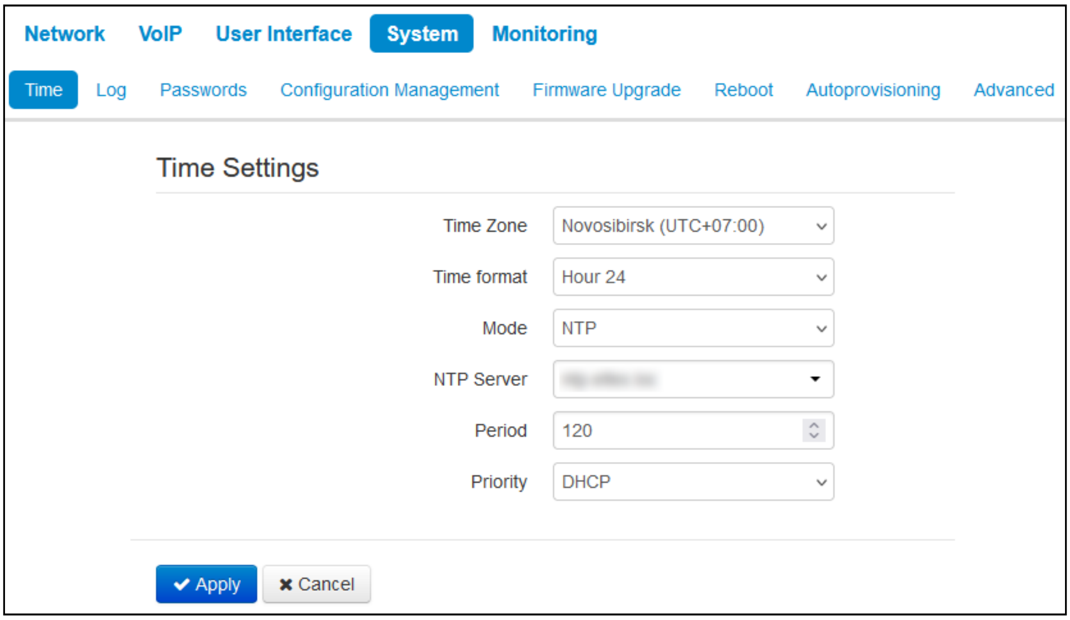
To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.4 'System' menu

In the 'System' menu you can configure settings for system, time and access to the device via various protocols, change the device password and update the device firmware.

3.2.4.1 'Time' submenu

In the 'Time' submenu you can configure time synchronization protocol (NTP) and manual date and time setting.



Time Zone — select a timezone from the list according to the nearest city in your region;

- *Time format* — set time format: *Hour 24* or *Hour 12*;
- *Mode* — time setting mode: NTP synchronization or manual.
- *NTP Server* — time synchronization server IP address/domain name. Manual entering of server address or selection from a list are available;
- *Period* — the device time will be automatically updated after the specified period of time;
- *Priority* — allows selection of priority of obtaining the NTP server address:
  - *DHCP* — when selected, the device uses the NTP server address from DHCP messages in option 42 (Network Time Protocol Servers). DHCP protocol must be set for the main interface;
  - *Config* — when selected, the device uses the NTP server address from '*NTP Server*' parameter.

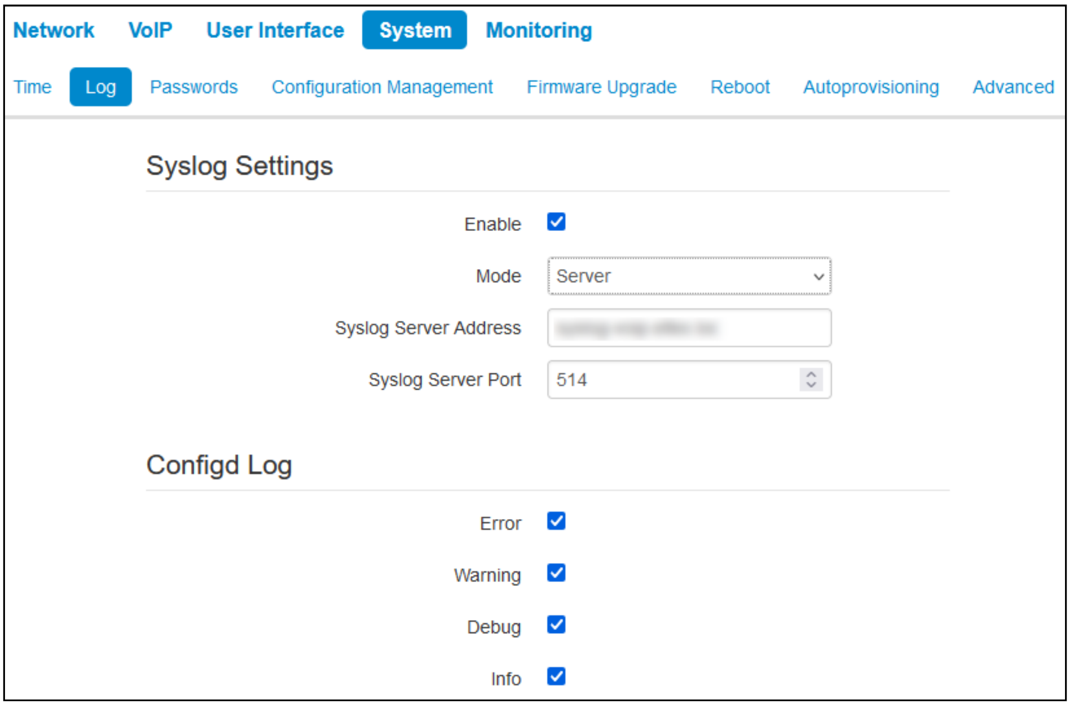
To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.4.2 'Log' submenu

In the 'Log' submenu you can configure output for various debug messages intended for device troubleshooting. Debug information is provided by the following device firmware modules:

- *Configd Log* — deals with the configuration file operations (config file reads and writes from various sources) and the device monitoring data collection;
- *Networkd Log* — deals with network configuration;
- *VoIP Log* — deals with VoIP functions operation;
- *Phone Manager Log* — deals with the device user interface operation (such as extension console, keyboard, display, speaker phone, handset etc.);
- *Media Manager Log* — deals with media operation;
- *Auto-update Log* — deals with auto-updating.

3.2.4.2.1 Syslog Settings

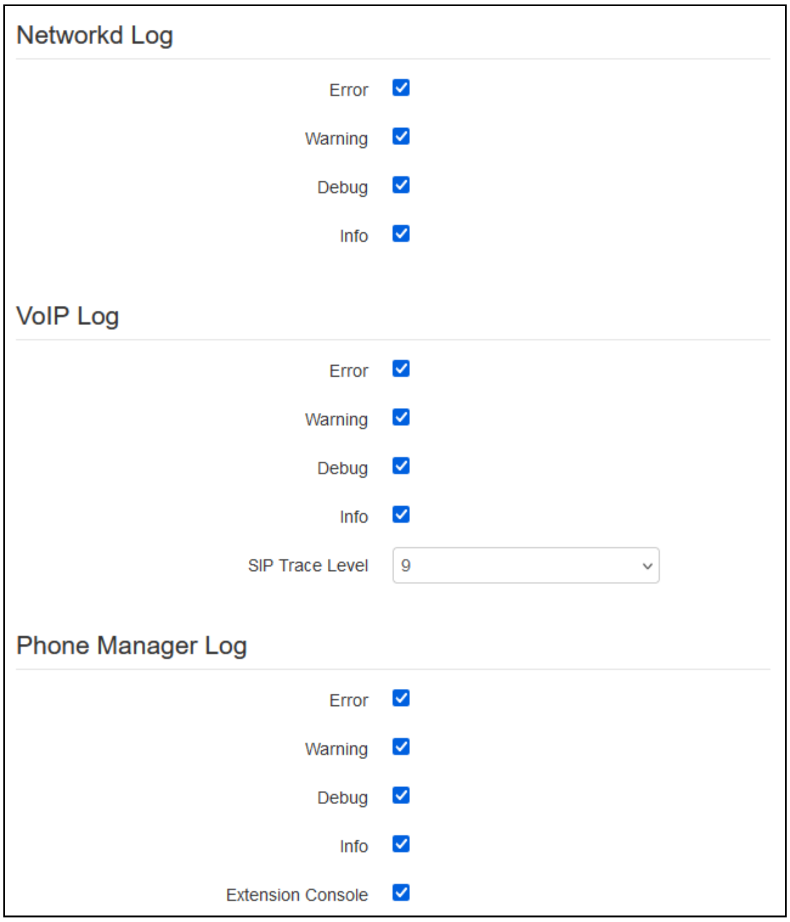


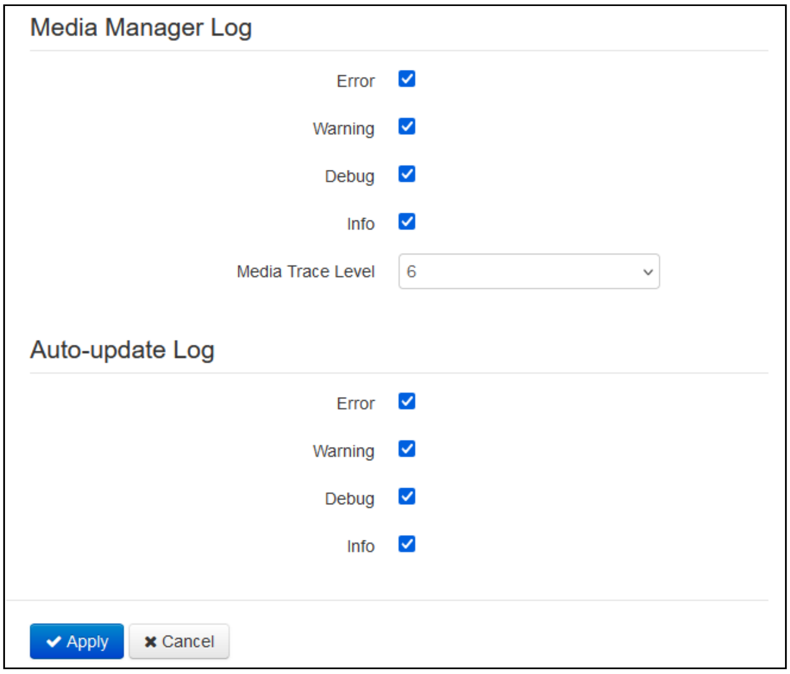
If there is at least a single log configured for Syslog output, it is necessary to enable Syslog agent that will intercept debug messages from the respective manager and send them to remote server or save them to a local file in Syslog format.

- *Enable* — when selected, user Syslog agent is launched;
- *Mode* — Syslog agent operation mode:
  - *Server* — log information will be sent to the remote Syslog server (this is the 'remote log' mode):
    - *Syslog Server Address* — Syslog server IP address or domain name (required for 'Server' and 'Server and File' modes);
    - *Syslog Server Port* — port for Syslog server incoming messages (default value is 514; required for 'Server' and 'Server and File' modes).
  - *Local File* — log information will be saved to the local file:
    - *File Name* — name of the file to store log in Syslog format (required for 'Local File' and 'Server and File' modes);
    - *File Size, kB* — maximum log file size (required for 'Local File' and 'Server and File' modes).

- *Server and File* — log information will be sent to the remote Syslog server and saved to the local file:
  - *Syslog Server Address* — Syslog server IP address or domain name (required for 'Server' and 'Server and File' modes);
  - *Syslog Server Port* — port for Syslog server incoming messages (default value is 514; required for 'Server' and 'Server and File' modes);
  - *File Name* — name of the file to store log in Syslog format (required for 'Local File' and 'Server and File' modes);
  - *File Size, kB* — maximum log file size (required for 'Local File' and 'Server and File' modes).
- *Console* — log information will be sent to the device console (connection via a COM port adapter is required).

3.2.4.2.2 Configd Log, Network Log, VoIP Log, Phone Manager Log, Media Manager Log, Auto-update Log





- *Error* — select this checkbox, if you want to collect 'Error' type messages;
- *Warning* — select this checkbox, if you want to collect 'Warning' type messages;
- *Debug* — select this checkbox, if you want to collect debug messages;
- *Info* — select this checkbox, if you want to collect information messages;
- *SIP Trace Level* — defines output level of VoIP SIP manager stack messages;
- *Media Trace Level* — defines output level of media manager stack messages.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

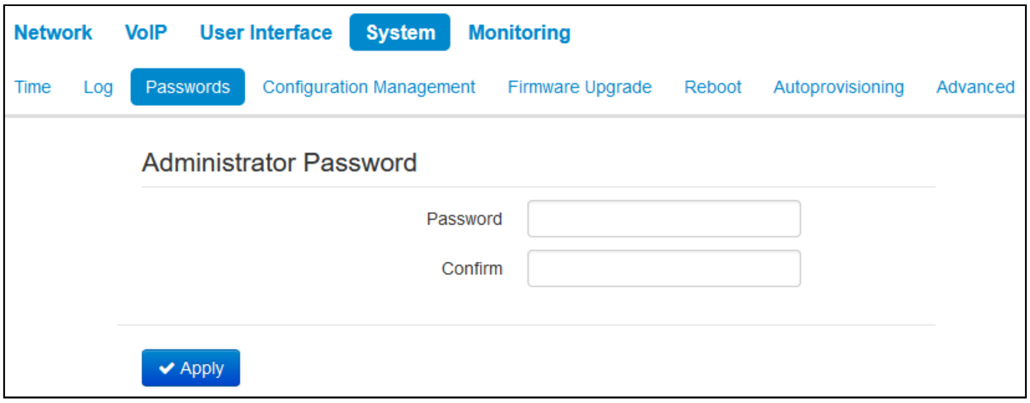
3.2.4.3 'Passwords' submenu

In the 'Passwords' submenu you can define passwords for administrator.

The set password is used for the device access via web interface and when connection via SSH.

When signing into web interface, administrator (default password: 'password') has the full access to the device: read/write any settings, full device status monitoring.

> Administrator login: **admin**;
> Administrator password: **password**.

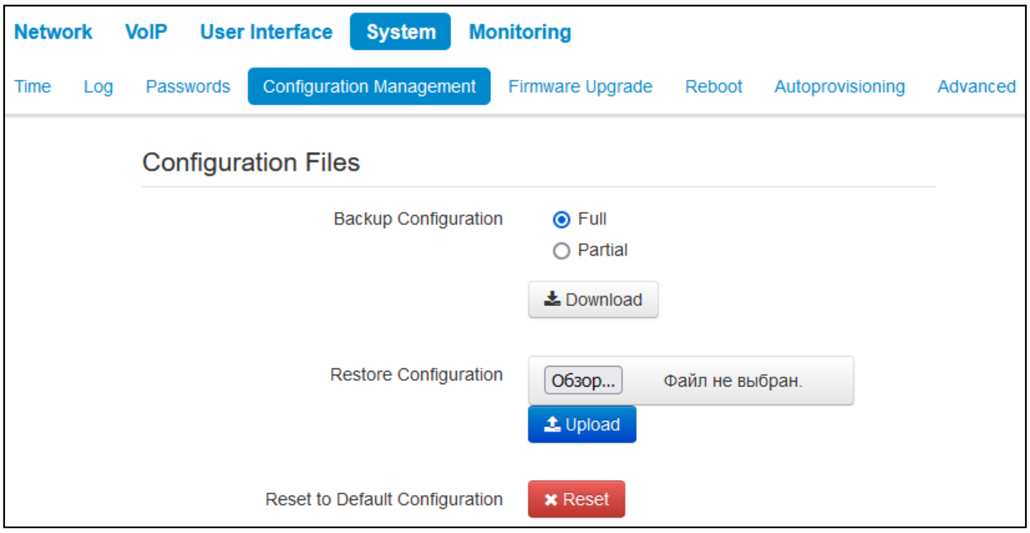


- *Password, Confirm* — enter administrator password in the respective fields and confirm it.

> To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.4.4 'Configuration Management' submenu

In the 'Configuration management' submenu you can save and update the current configuration.



3.2.4.4.1 Backup Configuration

- *Full* — download full device configuration archive;
- *Partial* — download partial device configuration archive, which contains only user configuration.

To save the current device configuration to a local PC, click *'Download'* button.

3.2.4.4.2 Restore Configuration

Select configuration file stored on a local PC. To update the device configuration, click '*Choose File*' button, specify a file (in .tar.gz format) and click '*Upload*' button. Uploaded configuration will be applied automatically without device reboot.

3.2.4.4.3 Reset to Default Configuration

To reset the device to default settings, click '*Reset*' button.

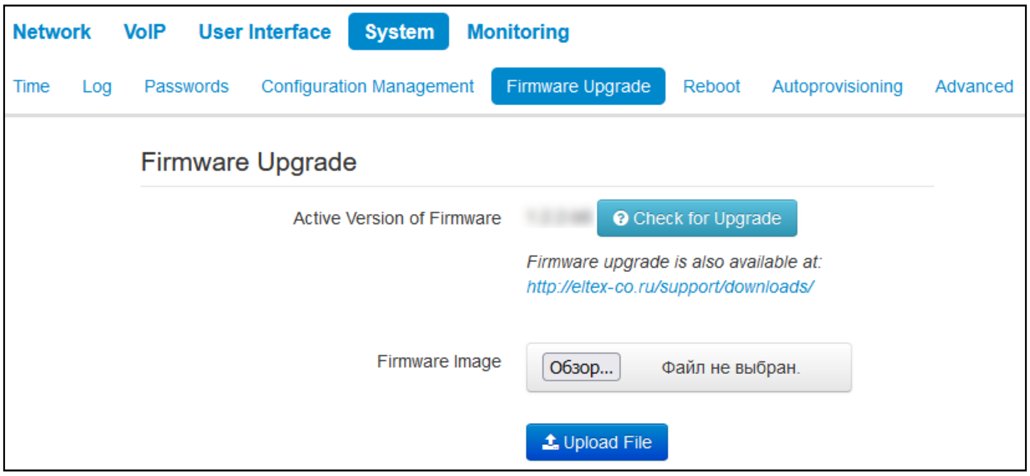
When you reset the device configuration, the followings will be also reset:

- contacts;
- call history;
- text messages[1].

[1] The feature will be available in future firmware versions.

3.2.4.5 'Firmware Upgrade' submenu

In 'Firmware upgrade' submenu you can update the firmware of the device.



- *Active Version of Firmware* — installed firmware version;
    - *Check for Upgrade* — click this button to check the availability of the latest firmware version. With this function, you can quickly check the latest firmware version and update the firmware, if necessary.

Firmware upgrade check function requires Internet access.

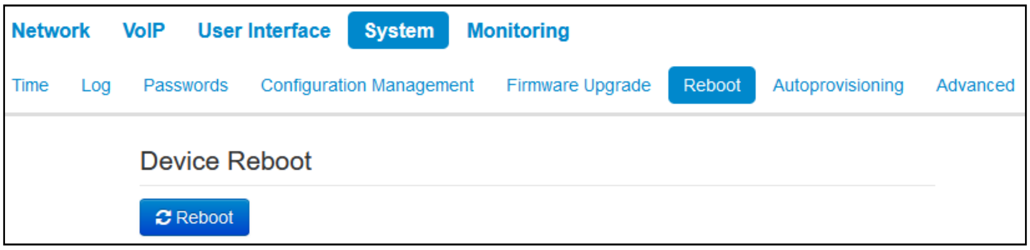
- Firmware Image — You can upgrade the device firmware manually by downloading the firmware file from the web site https://eltex-co.com/support/downloads/ and saving it on the computer. To do this, click the '*Choose File*' button in *Firmware Image* field and specify path to firmware in .tar.gz format file.

To launch the update process, click '*Upload File*' button. The process can take several minutes (its current status will be shown on the page). The device will be automatically rebooted when the update is completed.

Do not switch off or reboot the device during the firmware upgrade.

3.2.4.6 'Reboot' submenu

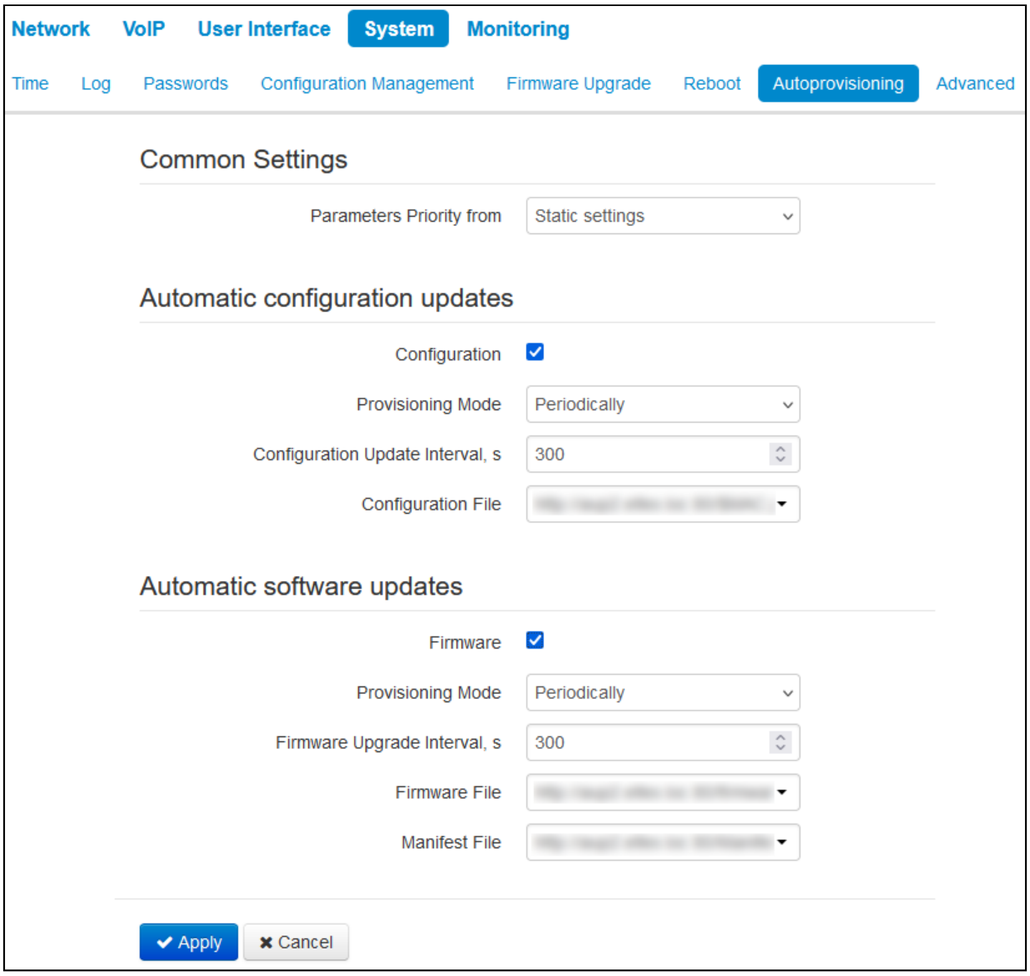
In the 'Reboot' submenu you can reboot the device.



Click the *'Reboot'* button to reboot the device. Device reboot process takes approximately 1 minute to complete.

3.2.4.7 'Autoprovisioning' submenu

In the 'Autoprovisioning' submenu you can configure DHCP-based autoprovisioning algorithm.



Common Settings

- *Parameters Priority from* — this parameter manages names and locations of configuration and firmware files:
  - *Static settings* — paths to configuration, firmware, and manifest files are defined by the *'Configuration File'* and *'Firmware File', and 'Manifest File'* settings;
  - *DHCP options* — paths to configuration and firmware files are defined by the DHCP Option 43, 66, and 67 (it is necessary to select DHCP for the Internet service).

3.2.4.7.1 Automatic configuration updates

3.2.4.7.1.1 Configuration

- *Provisioning Mode* — to update configuration, you can specify one of the several update modes:
  - *Periodically* — the device configuration will be automatically updated after defined period of time;
    - *Configuration Update Interval, s* — time period in seconds that will be used for periodic device configuration update; if 0 is selected, device will be updated only once — immediately after startup;
  - *Scheduled* — the device configuration will be automatically updated at specific times and on specific days:
    - *Time of Configuration Update* — time on 24-hour format that will be used for configuration autoupdate;
    - *Days of Configuration Update* — week days with defined time that will be used for configuration autoupdate.
- *Configuration File* — full path to configuration file; defined in URL format:
  - http://<server address>/<full path to cfg file>
  - https://<server address>/<full path to cfg file>
  - ftp://<server address>/<full path to cfg file>
  - tftp://<server address>/<full path to cfg file>

where <server address> — HTTP, HTTPS, FTP or TFTP server address (domain name or IPv4),

< full path to cfg file > — full path to configuration file on server.

3.2.4.7.2 Automatic software updates

3.2.4.7.2.1 Firmware

- *Provisioning Mode* — to update firmware, you can separately specify one of the several update modes:
  - *Periodically* — the device firmware will be automatically updated after defined period of time;
    - *Firmware Update Interval, s* — time period in seconds that will be used for periodic device firmware update; if 0 is selected, device will be updated only once — immediately after startup;
  - *Scheduled* — the device firmware will be automatically updated at specific times and on specific days:
    - *Time of Firmware Update* — time on 24-hour format that will be used for firmware autoupdate;
    - *Days of Firmware Update* — week days with defined time that will be used for firmware autoupdate.
- *Firmware File* — full path to firmware file; defined in URL format:
  - http://<server address>/<full path to firmware file>
  - https://<server address>/<full path to firmware file>
  - ftp://<server address>/<full path to firmware file>
  - tftp://<server address>/<full path to firmware file>

where <server address> — HTTP, HTTPS, TFTP or FTP server address (domain name or IPv4),

<full path to firmware file> — full path to firmware file on server.

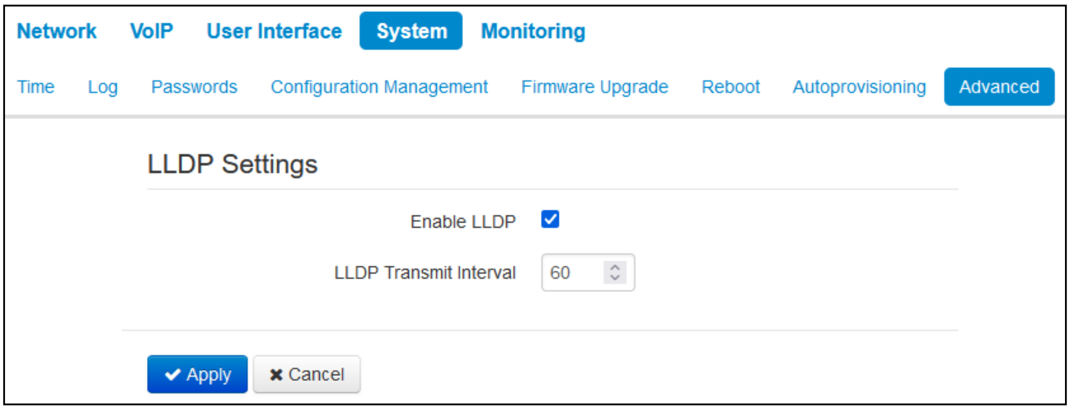
- *Manifest File* — full path to manifest file; defined in URL format. The use of the manifest file is due to the large size of the firmware file, which is downloaded periodically using the firmware auto-update algorithm. To reduce load on the network in such cases, it is recommended to use the Manifest file. The file structure is a line that specifies the firmware version identifier that is available for downloading and updating.
  For example, the contents of the Manifest file could be as follows: '1.2.2-b8'.

For detailed DHCP-based automatic update algorithm, see Appendix 'Device automatic update algorithm based on DHCP'.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

3.2.4.8 'Advanced' submenu

Use the menu to configure additional device settings.



3.2.4.8.1 Settings LLDP

- *Enable LLDP* — use LLDP when selected;
- *LLDP Transmit Interval* — time interval for messages transmission through LLDP. Default value is 60 seconds.

To apply a new configuration and store settings into the non-volatile memory, click *'Apply'* button. To discard changes, click *'Cancel'* button.

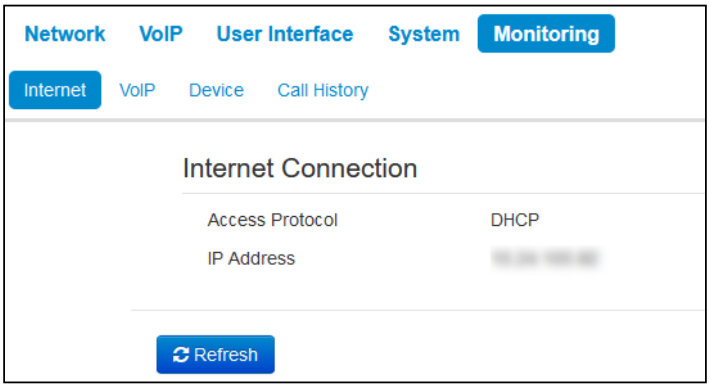
3.3 Monitoring

To enter the system monitoring mode, select 'Monitoring' tab from the panel.

> Some pages do not feature automatic update of the device monitoring data. To obtain the current information from the device, click Refresh button.

3.3.1 Network parameters monitoring

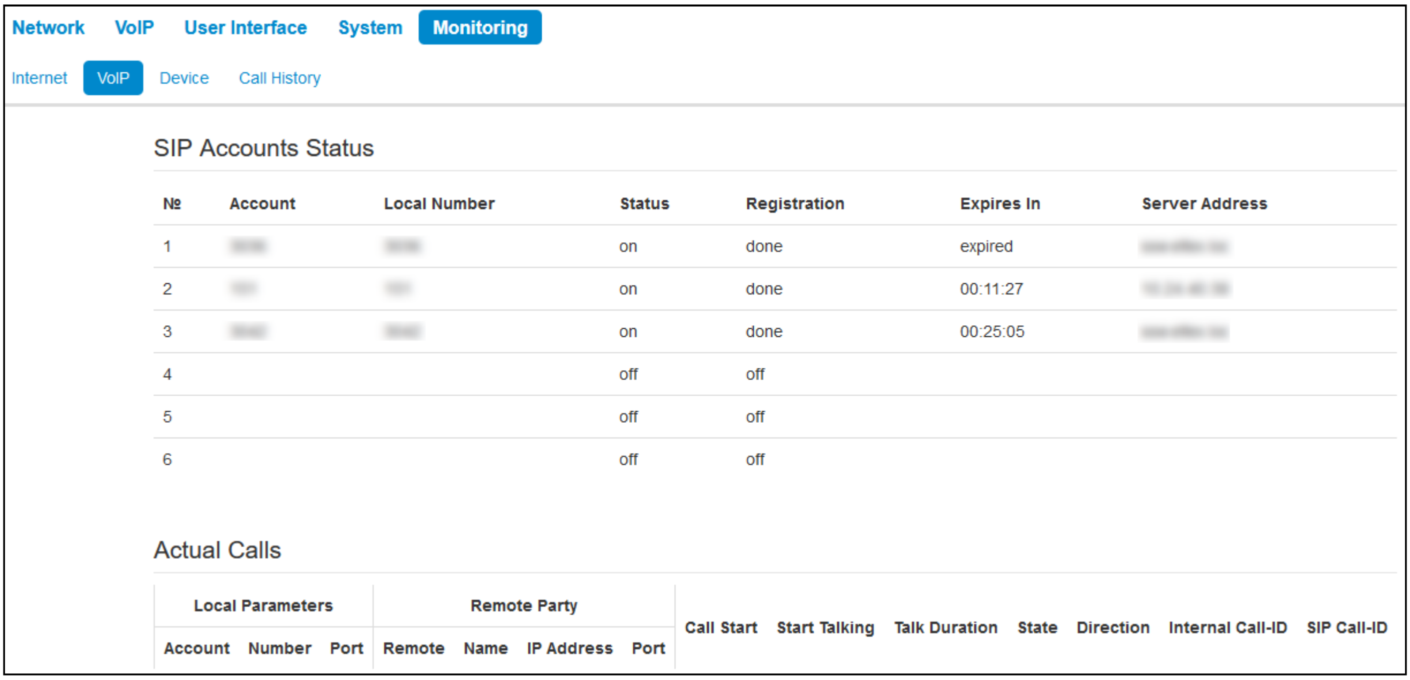
In the 'Internet' submenu you can view basic network settings of the device.



- *Access Protocol* — protocol used for LAN access.
- *IP Address* — device IP address in the local network.

3.3.2 VoIP connection monitoring

In 'VoIP' submenu you can view VoIP network interface status and monitor accounts.



3.3.2.1 SIP Accounts Status

- *№* — number of account;
- *Account* — name of account;
- *Local Number* — subscriber phone number assigned to the current account;
- *Status* — account status:
  - *on*;
  - *off*.
- *Registration* — state of registration on proxy server for the group phone number:
  - *off* — SIP server registration function is disabled in SIP profile settings;
  - *error* — registration was unsuccessful;
  - *done*— registration on SIP server successfully completed.
- *Expires In* — expiration time of account registration on SIP server;
- *Server Address* — address of the server on which the subscriber line has been registered at the last time.

3.3.2.2 Actual Calls

Actual Calls

Local Parameters			Remote Party				Call Start	Start Talking	Talk Duration	State	Direction	Internal Call-ID	SIP Call-ID
Account	Number	Port	Remote	Name	IP Address	Port							
[illegible]	[illegible]	23008	[illegible]	[illegible]	[illegible]	13564	17:12:48 13.08.2024	17:12:52 13.08.2024	00:00:26	holded	outgoing	5	1b270616-d43a-123d-5baf-0050c2010203
[illegible]	[illegible]	23012	[illegible]	[illegible]	[illegible]	13378	17:12:59 13.08.2024	17:13:05 13.08.2024	00:00:13	talking	incoming	6	06bb84ceb6c45024

Local Parameters

- *Account* — name of account through which a call is implemented or on which the call was received;
- *Number* — phone number assigned on the account;
- *Port* — RTP stream local port.

Remote Party

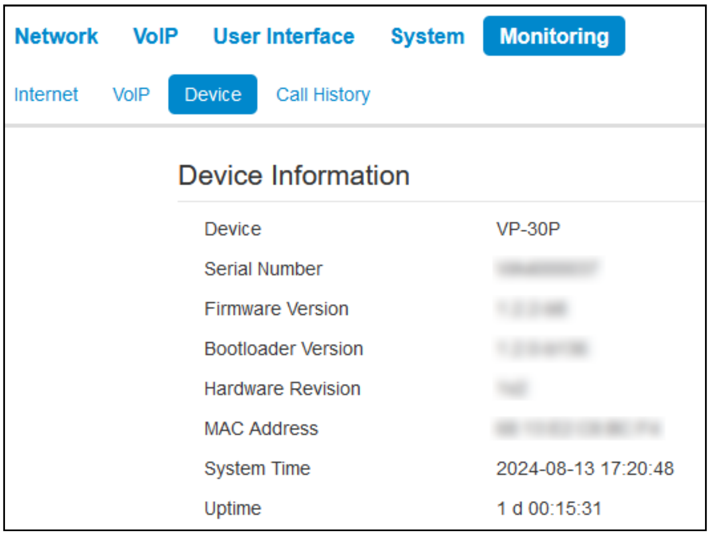
- *Remote* — phone number of opposite party;
- *Name* — opposite party name;
- *IP Address* — IP address of opposite party used for RTP;
- *Port* — UDP port of opposite party used for RTP stream.

Common parameters

- *Call Start* — call start time;
- *Start Talking* — conversation start time;
- *Talk Duration* — conversation duration;
- *State* — call state. Call might be in the following states:
  - *call* — ringback tone is issued (during outcoming call);
  - *incoming call* — ring tone is issued (during incoming call);
  - *talking;*
  - *holded;*
  - *conference*.
- *Direction* — call type:
  - *incoming;*
  - *outgoing*.
- *Internal Call-ID;*
- *SIP Call-ID*.

3.3.3 Viewing information on the device

In the 'Device' submenu you can find general device information.



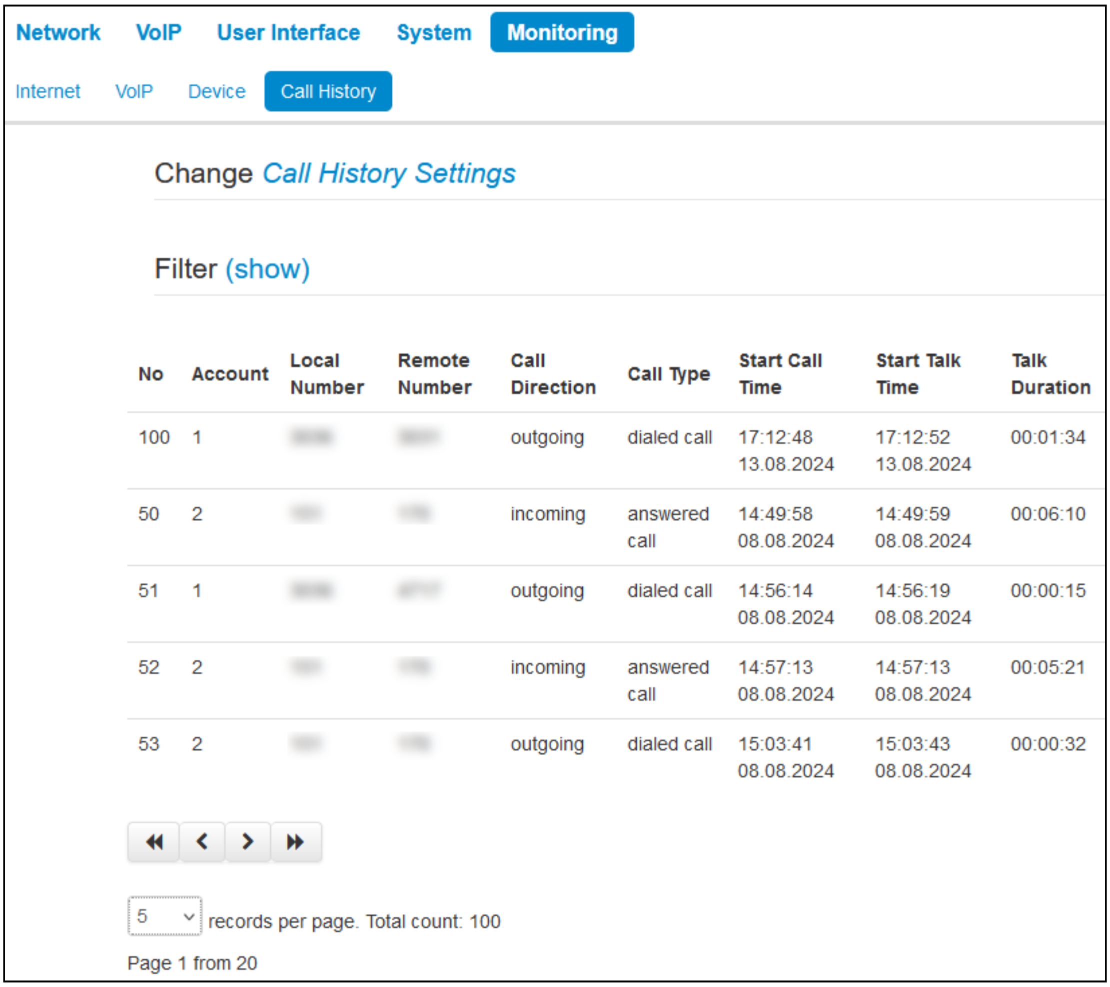
- *Device* — device model name;
- *Serial Number* — device serial number defined by the manufacturer;
- *Firmware Version* — device firmware version;
- *Bootloader Version* — software version of the device bootstrap;
- *Hardware Revision* — device revision;
- *MAC Address* — device MAC address defined by the manufacturer;
- *System Time* — current date and time defined in the system;
- *Uptime* — time of operation since the last startup or reboot of the device.

3.3.4 Viewing Call History

In the 'Call History' submenu you can view the list of phone calls and the summary for each call.

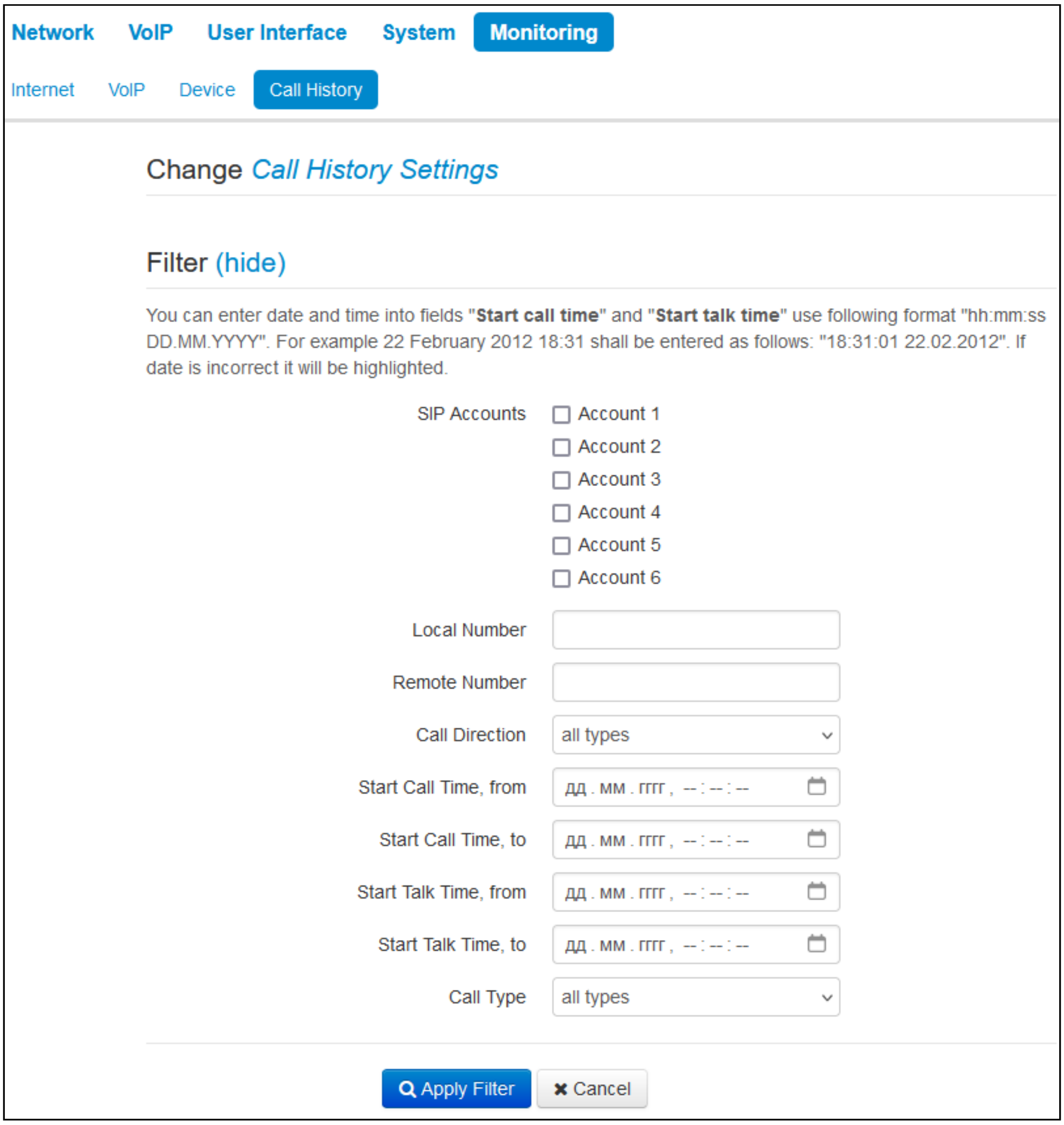
The device RAM can store up to 100 records for performed calls. If the record number exceeds 100 the oldest records (at the top of the table) will be removed, and new ones will be added at the end of the file.

Call log statistics will not be collected, when the history size is zero.



'Call history' table field description:

- *No* — sequence number of the record in the table;
- *Account* — device subscriber port number;
- *Local Number* — subscriber number assigned to the current subscriber port;
- *Remote Number* — remote subscriber number that the phone connection has been established with;
- *Call Direction* — *outgoing* or *incoming*;
- *Call Type* — *dialed, missed* or *answered*;
- *Start Call Time* — call received/performed time and date;
- *Start Talk Time* — call start time and date;
- *Talk Duration* — call duration.



In the call history table you can search records by different parameters; to do this, click the '*Filter (show)*' link. Filtering can be performed by account, local or remote number, call direction, start call time, call talk time, and call type. For filtering parameter description, see call history table field description above.

To hide the table record filtration parameter settings, click the '*Filter (hide)*' link.

To configure call history parameters, click '*Change Call History Settings*' link. For detailed parameter configuration description, see 'Phone book' submenu.

Click button to proceed to the table showing the first record.

Click button to proceed to the previous page with the call history table.

Click button to proceed to the next page with the call history table.

Click button to proceed to the table showing the last record.

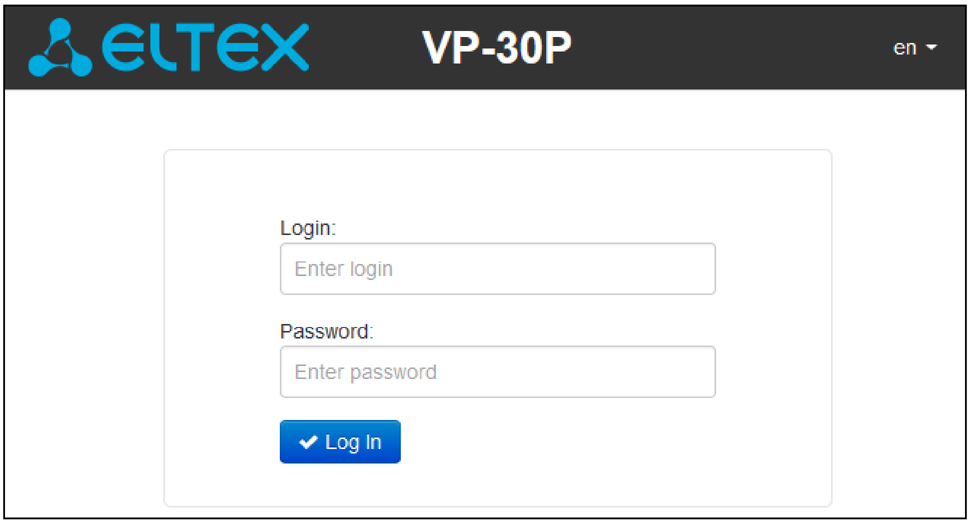
You can select the number of displayed records at the bottom of the page.

4 Example of device configuration

1. Open web browser such as Firefox, Opera or Chrome on the PC.
2. Enter the device IP address in an address line of a browser.

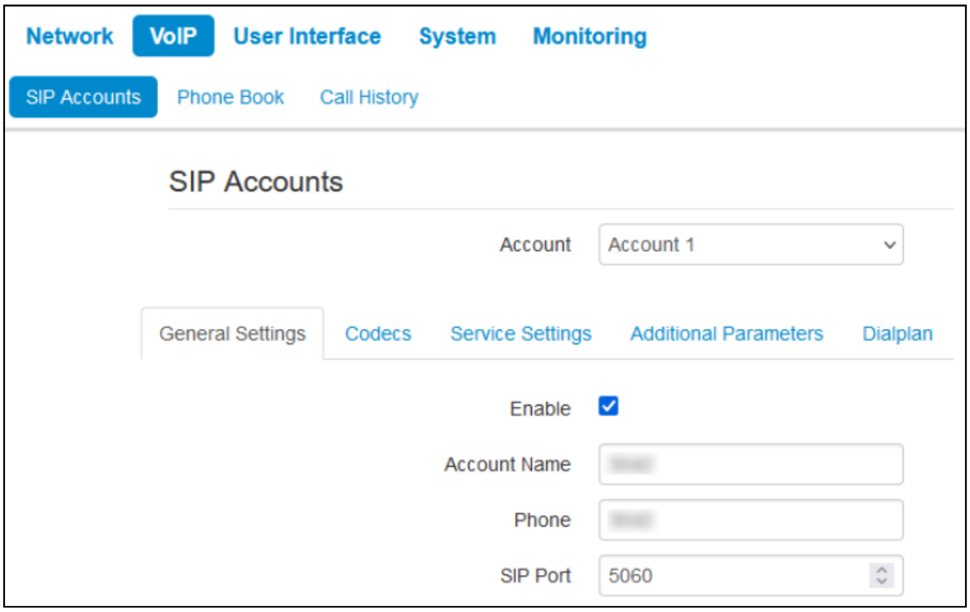
> By default, the device receives IP address and other network parameters via DHCP. For further work, you should know IP address received by IP phone from DHCP server. To do it, use display menu:
> 1. Press 'Menu' soft key.
> 2. Check the IP address assigned to the phone in 'State' → 'Network' section.
>
> If IP address is 0.0.0.0, it means IP phone has not received IP address from DHCP server. In this case, you should manually configure network parameters by using display menu.

If the device is successfully connected, you will see a pop-up window with login and password request. Fill in the fields and click '*Log in*' button.

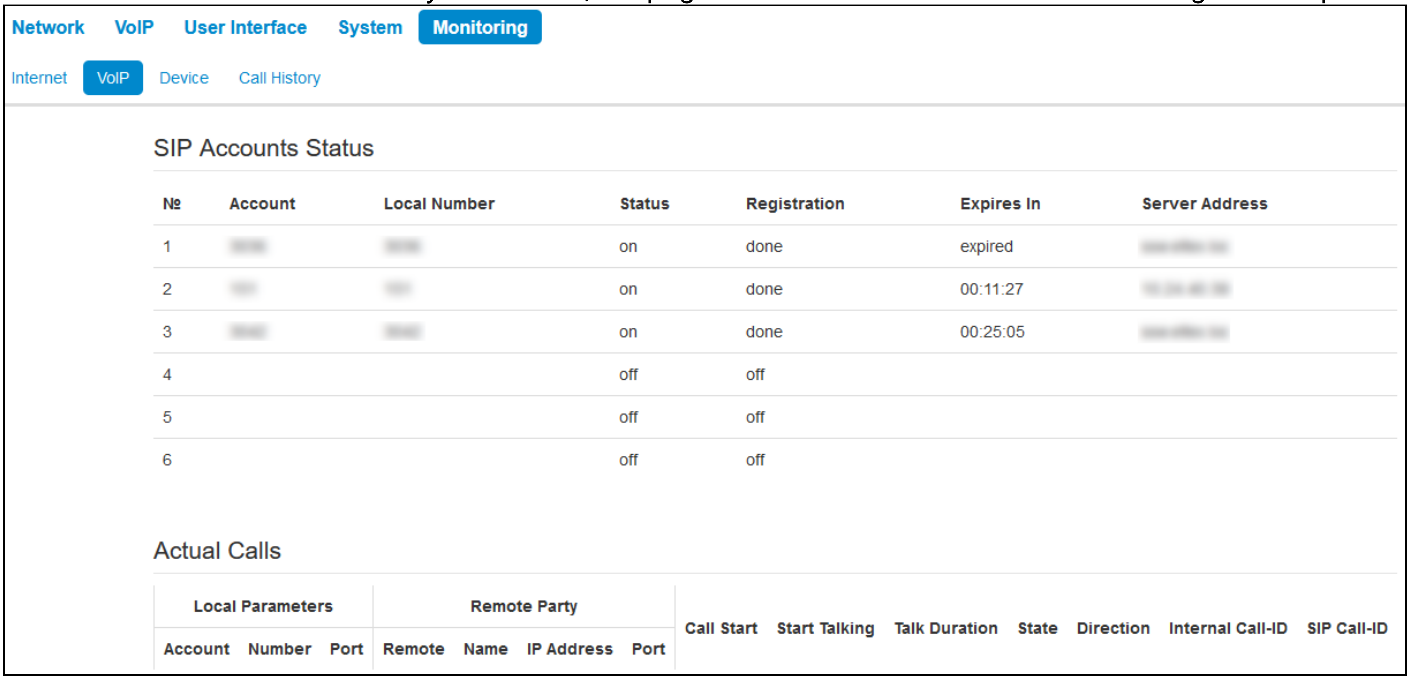


> By default, login: **admin**, password: **password**.

You can change web interface language at the top right corner, see below:



If the device has been successfully authorized, the page of the device current state monitoring will be opened:

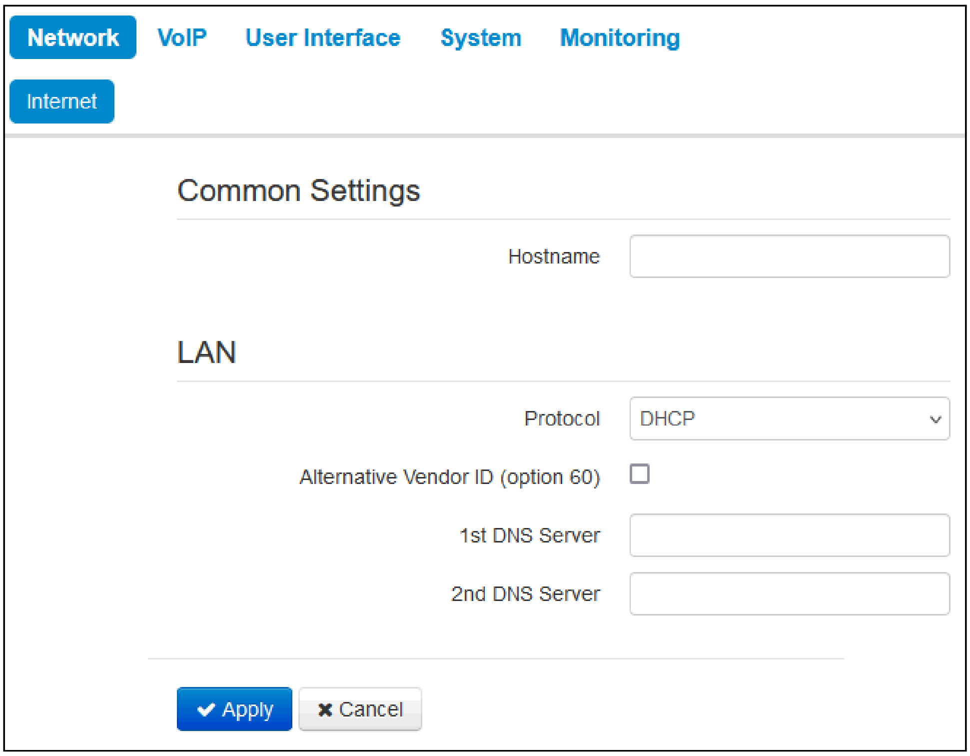


3. To change the device network settings go to 'Network' → 'Internet' section.

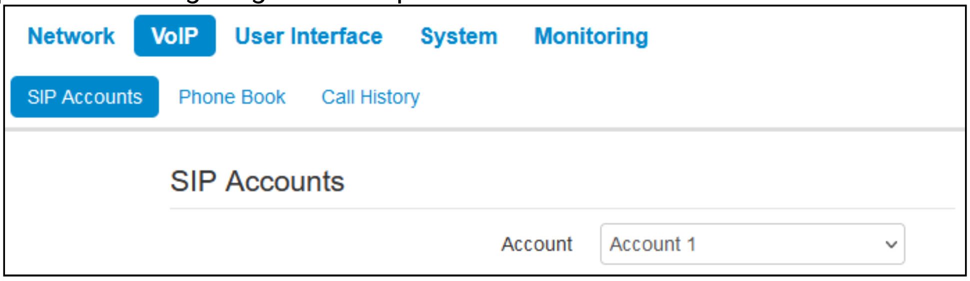
Select protocol used by your Internet provider in '*Protocol*' field and enter necessary data according to provider guidelines. If static settings are used for connection to a provider network, select '*Static*' value in the '*Protocol*' field and fill in '*IP address*', '*Netmask*', '*Default gateway*', '*1st DNS Server*' and '*2nd DNS Server*' fields (parameter values are given by service provider).

To save and apply settings, click 

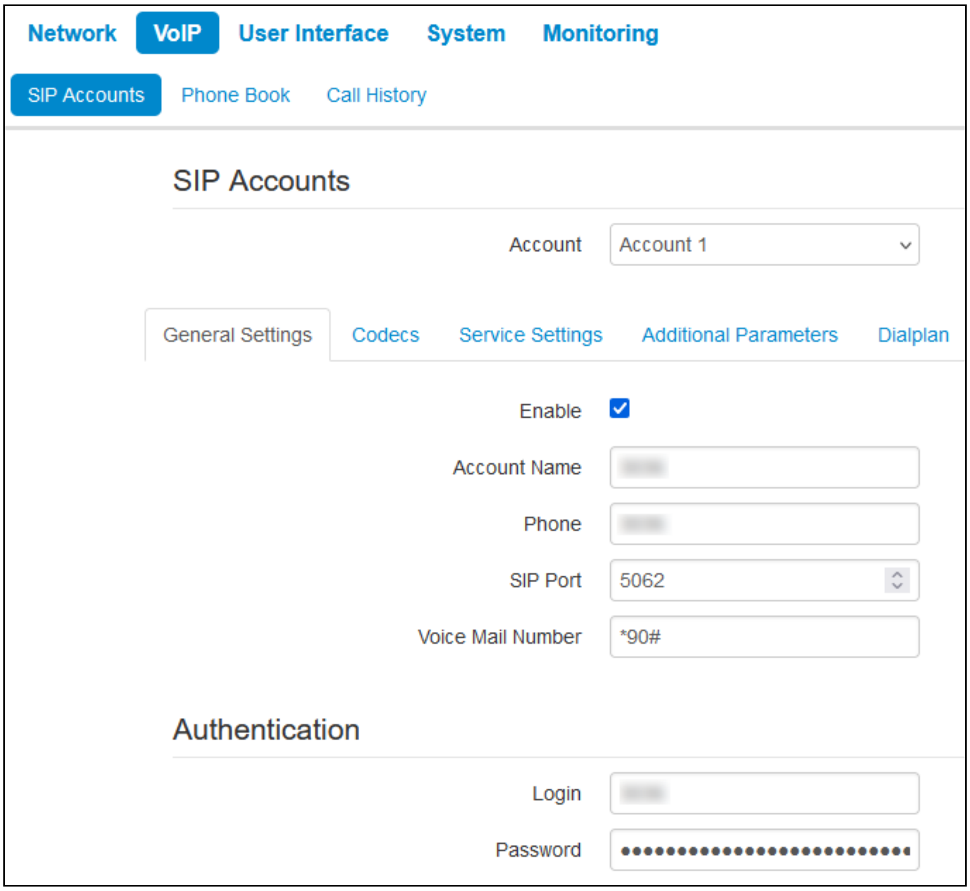
.



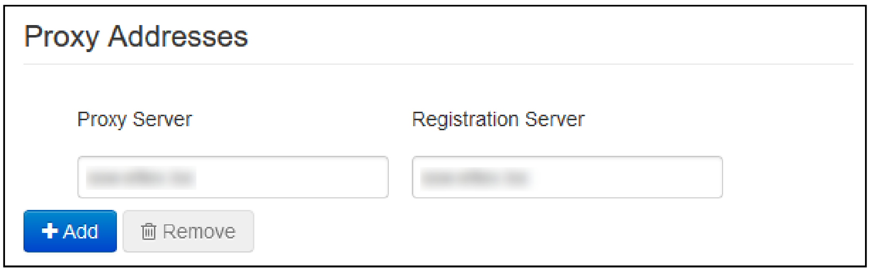
Go to 'VoIP' → 'SIP Accounts' tab to configure accounts for operation via SIP. To do it, select account number in '*Account*' field, required for configuring in the drop-down list.



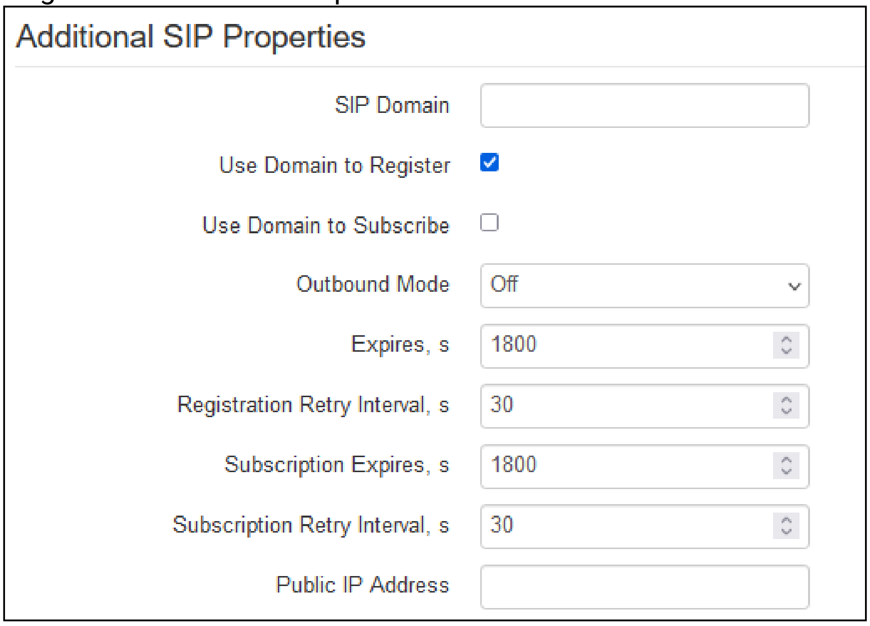
In '*General Settings'* tab select '*Enable*' checkbox, enter phone number assigned to the current account and specify login and password for SIP server authorization.



Specify IP address or SIP server domain name and registration servers (if required) in relevant fields in section 'Proxy Addresses'. If port numbers used on servers are different than 5060, specify alternative port colon separated.



Specify SIP domain (if required) in relevant field in 'Additional SIP Properties' section. To use domain to register, set the relevant flag in 'Additional SIP Properties' section.



To save and apply settings click 

button.

5 Appendixes

5.1 Description of phone books supported structures

5.1.1 EltexIPPhoneDirectory phone book of .xml format

The EltexIPPhoneDirectory phone book in .xml format consists of two parts: a prologue and a root element.

The prologue contains an XML declaration indicating that this is an XML document, and also includes the XML version number and encoding:

```
<?xml version="1.0" encoding="UTF-8" ?>
```

The root element is a description of the EltexIPPhoneDirectory phone book, which includes a list of all groups, contacts and numbers belonging to them. The opening tag of the root element looks like this:

```
<EltexIPPhoneDirectory>
```

The value of the root element <EltexIPPhoneDirectory> contains the following tags.

The <Title> tag describes the name of the phone book, which indicates its affiliation with a specific vendor:

```
<Title>EltexPhones</Title>
```

The next tag is <Prompt>, which is used for hints, the value of the parameter can be any text message:

```
<Prompt>Prompt</Prompt>
```

Next are the paired tags <Grouplist> and </Grouplist>, which include the self-closing tags <Group/>. They contain the group name in attribute-value pairs (name="Development"):

```
<Grouplist>
    <Group name="Development"/>
    <Group name="Testing"/>
</Grouplist>
```

Then there is a list of contacts. Each of them is separated by paired tags – <DirectoryEntry> and </DirectoryEntry>. The value of the contact name is specified inside <Name> tag. Below, inside <Telephone> tag, the contact number is specified. If the contact has 2 or 3 numbers, then each of them is written in a new separate line inside a similar <Telephone> tag.

The maximum quantity of numbers for one contact is 3. Next, inside <Group> tag, the group to which the contact belongs is specified. The entry for the contact, its numbers, and the group to which the contact belongs is shown below:

```
<DirectoryEntry>
    <Name>Ivan Ivanov</Name>
    <Telephone>2000</Telephone>
    <Telephone>2001</Telephone>
    <Telephone>2002</Telephone>
    <Group>Testing</Group>
</DirectoryEntry>
```

The similar syntax is used for the rest of contacts.

After listing all groups and contacts, the closing tag of the root element is specified, which looks like this:

```
</EltexIPPhoneDirectory>
```

Example of EltexIPPhoneDirectory phone book of .xml format:

```
<?xml version="1.0" encoding="UTF-8" ?>
<EltexIPPhoneDirectory>
    <Title>EltexPhones</Title>
    <Prompt>Prompt</Prompt>
    <Grouplist>
        <Group name="Development"/>
        <Group name="Testing"/>
    </Grouplist>
    <DirectoryEntry>
        <Name>Fedor Fedorov</Name>
        <Telephone>1001</Telephone>
        <Telephone>1002</Telephone>
        <Telephone>1003</Telephone>
        <Group>Development</Group>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Ivan Ivanov</Name>
        <Telephone>2000</Telephone>
        <Telephone>2001</Telephone>
        <Telephone>2002</Telephone>
        <Group>Testing</Group>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Peter Petrov</Name>
        <Telephone>8002</Telephone>
        <Telephone>6008</Telephone>
        <Group>Development</Group>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Boris Borisov</Name>
        <Telephone>4020</Telephone>
        <Group>Testing</Group>
    </DirectoryEntry>
</EltexIPPhoneDirectory>
```

5.1.2 EltexIPPhoneDirectory phone book of .xml format of another vendor

The EltexIPPhoneDirectory phone book of another vendor in .xml format starts with prologue, also known as a opening tag.

The element's opening tag describes the book's vendor and also contains the xmlns attribute, which is used to define a namespace. Defining a namespace for an element means that all child elements with the same prefix are associated with the same namespace.

```
<VendorIPPhoneBook xmlns:b="urn:crystal-reports:schemas:report-detail">
```

<Title> tag describes the title of the phone book, which indicates its affiliation with a particular vendor:

```
<Title>Vendor</Title>
```

Next is a description of the first group in the phone book.

The title tag for designating a group of contacts is <Menu> tag, which contains the name of the group in attribute-value pairs (Name="Job"):

```
<Menu Name="Job">
```

This is followed by a list of contacts. Each contact is separated by a self-closing <Unit/> tag, which has attribute-value pairs:

- For contact numbers: Phone1...3="8782" (the maximum quantity of numbers for one contact is 3)
- For default photo: default_photo="Resource:" (this tag is not processed on VP-30P)
- For contact name: Name="Oleg Alekseev"

Below is a complete entry of information for one contact:

```
<Unit Phone3="8782" default_photo="Resource:" Name="Oleg Alekseev" Phone1="4467" Phone2="7621"/
>
```

The similar syntax is used for the rest of contacts.

After listing all contacts in a given group, a closing tag is specified, which looks like this:

```
</Menu>
```

The similar syntax is used for the rest of contact groups.

The phone book structure is completed by a closing tag that looks like this:

```
</VendorIPPhoneBook>
```

Example of VendorIPPhoneBook phone book of .xml format:

```
<VendorIPPhoneBook xmlns:b="urn:crystal-reports:schemas:report-detail">
<Title>Vendor</Title>
<Menu Name="Job">
<Unit Phone3="" default_photo="Resource:" Name="Boss-group" Phone1="2517" Phone2=""/>
<Unit Phone3="8782" default_photo="Resource:" Name="Oleg Alekseev" Phone1="4467" Phone2="7621"/
>
<Unit Phone3="2081" default_photo="Resource:" Name="Mariya Ivanova" Phone1="86338531113"
 Phone2="2080"/>
<Unit Phone3="" default_photo="Resource:" Name="Ivan Maksimov" Phone1="2214" Phone2="2215"/>
<Unit Phone3="" default_photo="Resource:" Name="Stanislav Petrov" Phone1="8003" Phone2="8004"/>
<Unit Phone3="" default_photo="Resource:" Name="Natalya Haritonova" Phone1="9010" Phone2=""/>
<Unit Phone3="" default_photo="Resource:" Name="Anna Shishkina" Phone1="9120" Phone2="9809"/>
<Unit Phone3="4752" default_photo="Resource:" Name="Vladimir Yurov" Phone1="2931" Phone2="7820"
/>
<Unit Phone3="8432" default_photo="Resource:" Name="Irina Yavolova" Phone1="1010" Phone2="8600"
/>
</Menu>
</VendorIPPhoneBook>
```

5.1.3 EltexIPPhoneDirectory phone book of .csv format

The EltexIPPhoneDirectory phone book in .csv format consists of two parts: a header (optional) and lines, which consist of separate columns. Each line is a separate table row, and columns are separated from each other by special separator characters.

If there is a header, the first line of a file specifies a list of the following fields:

```
Name,Group,Phone1,Phone2,Phone3
```

Header values:

- Name — contact name
- Group — contact group
- Phone 1..3 — contact numbers (maximum quantity of numbers for one contact is 3)

Below is a complete entry of information for one contact using a header:

```
Name,Group,Phone1,Phone2,Phone3
Mikhail,Management,4002,4004,4006
```

The similar syntax is used for the rest of contacts.

The comma (,) character is not the only character that can be used as a column value separator. Phone books in .csv format also support the following characters as separators: (;), (.), (:), (|), and the tab character.

An example of using the colon (:) character as a separator:

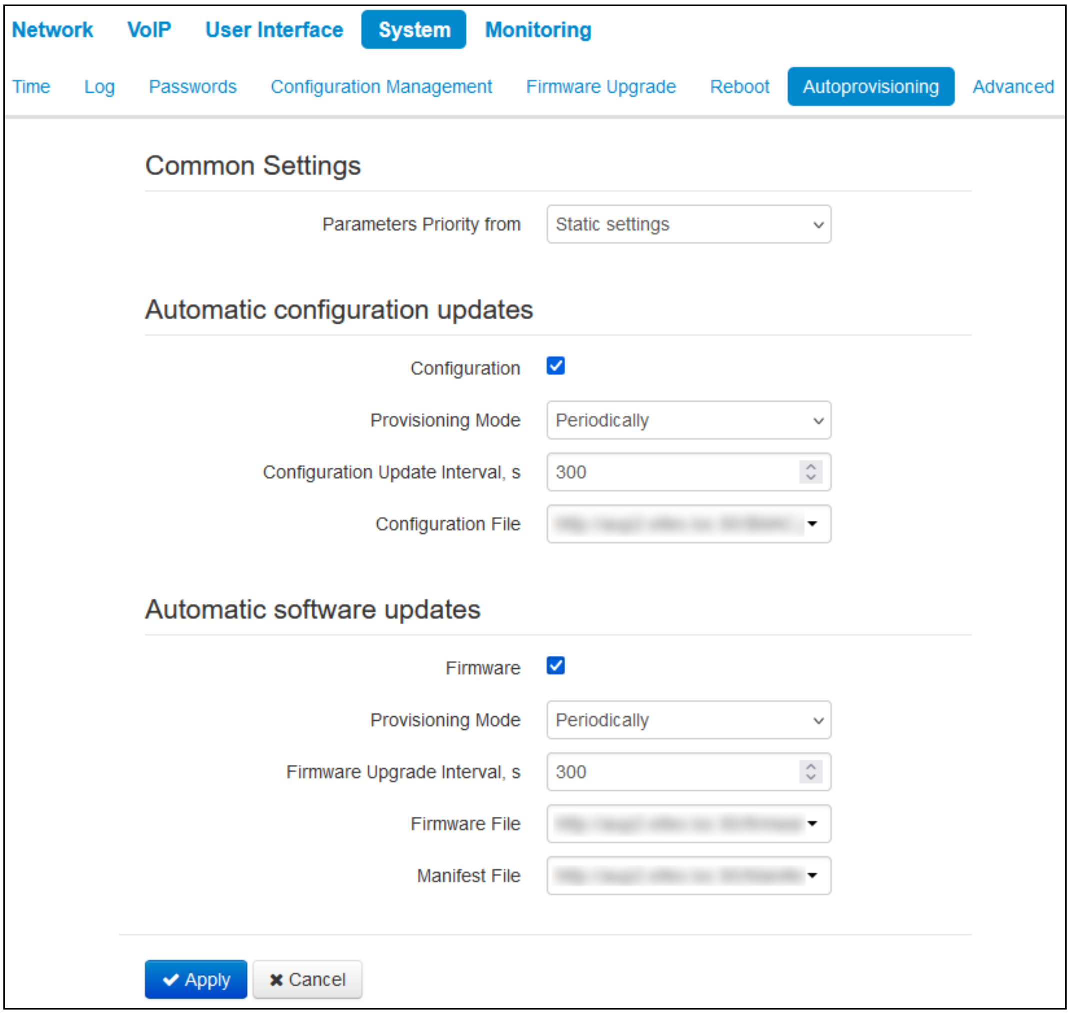
```
Stepan:Security:7021:7022:7023
```

The similar syntax is used for the rest of contacts.

Example of EltexIPPhoneDirectory phone book of .csv format:

```
Name,Group,Phone1,Phone2,Phone3
Ivan,Management,7020,9020,
Mikhail,Management,4002,4004,4006
Stepan,Security,7021,7022,7023
Irina,Logistics,7008,7009,7010
```

5.2 Device automatic update algorithm based on DHCP



Device automatic update algorithm is defined by the '*Parameters Priority from*' value.

If the '*Static settings*' value is selected, then the full path (including access protocol and server address) to configuration file and firmware file will be defined by '*Configuration File*' and '*Firmware File*' parameters. The full path should be specified in URL format (HTTP, HTTPS, FTP and TFTP protocols are supported):

<protocol>://<server address>/<path to file>, where

- <protocol> — protocol used for downloading corresponding files from the server;
- <server address> — address of the server with a file to be downloaded (domain name or IPv4);
- <path to file> — path to file on the server, the file must be of tar.gz extension.

URL examples:

tftp://download.server.loc/firmware.tar.gz

http://192.168.25.34/configs/vp-30/mycfg.tar.gz, and so on

You may use the following macro in URL (reserved words substituted with the specific values):

- *$MA* – MAC address — this macro in file URL is substituted by the native device MAC address;
- *$SN* – Serial number — this macro in file URL is substituted by the native device serial number;
- *$PN* – Product name — this macro in file URL is substituted by the model name (e.g., VP-30P);
- *$SWVER* — Software version — this macro in file URL is substituted by the firmware version number;
- *$HWVER* — Hardware version — this macro in file URL is substituted by the device hardware version number.

For MAC address, serial number and model name, see the 'Device' section on the 'Monitoring' tab.

URL examples:

tftp://download.server.loc/firmware.tar.gz,

http://192.168.25.34/configs/VP-30/mycfg.tar.gz,

tftp://server.tftp/$PN/config/$SN.tar.gz,

http://server.http/$PN/firmware/$MA.tar.gz etc.

If the system is unable to extract all the necessary file downloading parameters (protocol, server address or path to file on server) from configuration URL file or firmware file, it will attempt to extract an unknown parameter from DHCP Option 43 (Vendor specific info) or 66 (TFTP server) and 67 (Boot file name), when address obtaining via DHCP is enabled for the Internet service (DHCP option format and analysis are provided below):

tftp://update.local/VP-30.fw.

If '*DHCP options*' value is selected, configuration file and firmware file URLs will be extracted from DHCP Option 43 (Vendor specific info) or 66 (TFTP server) and 67 (Boot file name), wherefore address obtaining via DHCP should be enabled for the Internet service (DHCP option format and analysis is provided below).

It is possible to upload a text file of configuration, the format of the text file must be *.json*.

5.2.1 Option 43 format (Vendor specific info)

1|<acs_url>|2|<pcode>|3|<username>|4|<password>|5|<server_url>|6|<config.file>|7|<firmware.file>|9|<manifest.file>

1 — TR-069 autoconfiguration server address code;

2 — 'Provisioning code' parameter specification code;

3 — code of the username for TR-069 server authorization;

4 — code of the password for TR-069 server authorization;

5 — server address code; server address URL should be specified in URL format: tftp://address or http://address. The first version represents TFTP server address, the second version represents HTTP server address;

6 — configuration file name code;

7 — firmware file name code;

9 — manifest file name code;

'|' — mandatory separator used between codes and suboption values.

> For TR-069 autoconfiguration, suboptions 1, 3 and 4 will be applied when a priority from DHCP options is selected in the DHCP-based autoconfiguration section.

5.2.2 Algorithm of identification for configuration file and firmware file URL parameters from DHCP Options 43 and 66

1. DHCP exchange initialization
   Device initializes DHCP exchange after the startup.

2. Option 43 analysis
   When Option 43 is received, codes 5, 6, 7 and 9 suboptions are analyzed in order to identify the server address and the configuration, firmware file, and Minifest file names.

3. Option 66 analysis
   If Option 43 is not received from DHCP server or it is received but the system fails to extract the server address, Option 66 will be discovered. If the system fails to obtain the firmware file name, Option 67 will be discovered. They are used for TFTP server address and the firmware file path extraction respectively. Next, configuration and firmware files will be downloaded from Option 66 address via TFTP.

5.2.3 Special aspects of configuration updates

Configuration file should be in **.tar.gz** format (this format is used when configuration is saved from the web interface in the 'System' → 'Configuration Management' tab). Configuration downloaded from the server will be applied automatically and does not require device reboot.

5.2.4 Special aspects of firmware updates

Firmware file should be in **.tar.gz** format. When the firmware file is loaded, the device unpacks it and checks its version (using 'version' file in **tar.gz** archive).

If the current firmware version matches the version of the file obtained via DHCP, firmware will not be updated. Update is performed only when firmware versions are mismatched. The process of writing a firmware image to the device's flash memory is indicated by the 'Firmware update in progress...' screen on the phone display.

> Do not power off or reboot the device, when the firmware image is being written into the flash memory. These actions will interrupt the firmware update that will lead to the device boot partition corruption. If this happens, restore power to the device, and it will boot from the backup firmware image.
> There are two ways to restore a damaged image:
> 1. Update the firmware again;
> 2. Wait 10 minutes after successful booting from the backup image (in this case, it is not needed to update the firmware again, the images are synchronized automatically).
>
> If the backup firmware area was also damaged at the time of the update, the device can only be restored in a specialized service center.

5.3 Description of VP-30P configuration file cfg.json (+WEB)

Description of VP-17P configuration file is available at this link.

5.4 Options of network configuring using VP-30P IP phone

> ⚠ Before turning on the IP phone, it is neccessary to pre-configure the switch.

IP phone traffic is divided into 2 types:

1. Phone — traffic intended for the IP phone itself and all its services. Depending on settings, the traffic can be either tagged or untagged;
2. Transit — traffic intended for devices behind the phone's PC port. It comes from the external network to the LAN port, then passes through IP phone in transit and exits from the PC port, and so in the opposite direction. By default, the traffic is untagged along the entire path through the phone's bridge. If VLAN is enabled for transit traffic, then this traffic is tagged on the LAN port side, and untagged on the PC port side, that is, the phone removes or adds a tag for this type of traffic.

Network settings are made in the web interface menu 'Network' → 'Internet'.

It is possible to activate VLAN for Phone traffic in the section 'LAN' → 'Use VLAN' section or via the LLDP protocol.

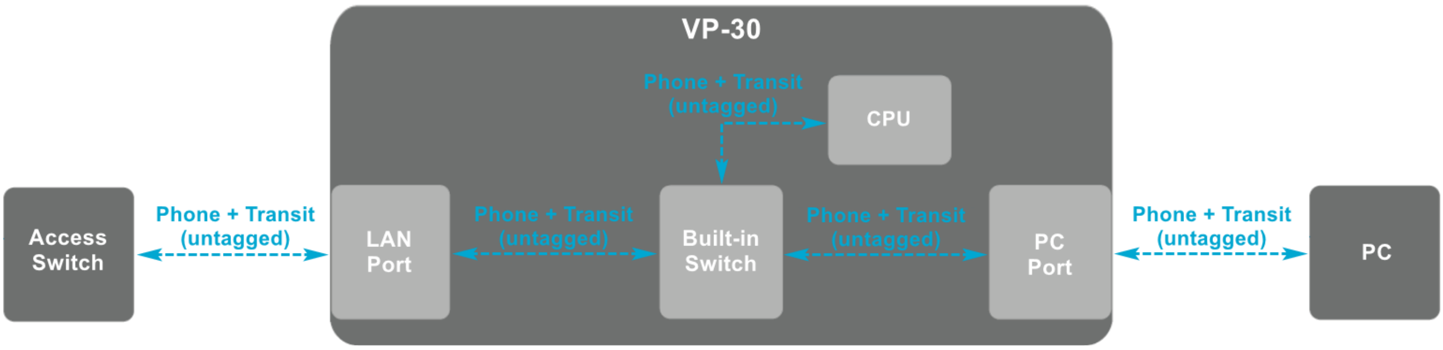
> ⚠ The LLDP protocol setting has priority over the configuration settings: that is, if a VLAN is configured for the phone traffic in the configuration, but the phone has received a network policy via LLDP, the network will be configured via LLDP.

It is possible to activate VLAN for Transit traffic in the section 'PC' → 'Use VLAN'.

> ⚠ A PC connected to the PC port has untagged traffic regardless of the configured scheme. When VLAN for Transit traffic is activated, the untagged traffic from the PC will be tagged when passing through the phone. And the tag will be removed when passing through the phone in the opposite direction.

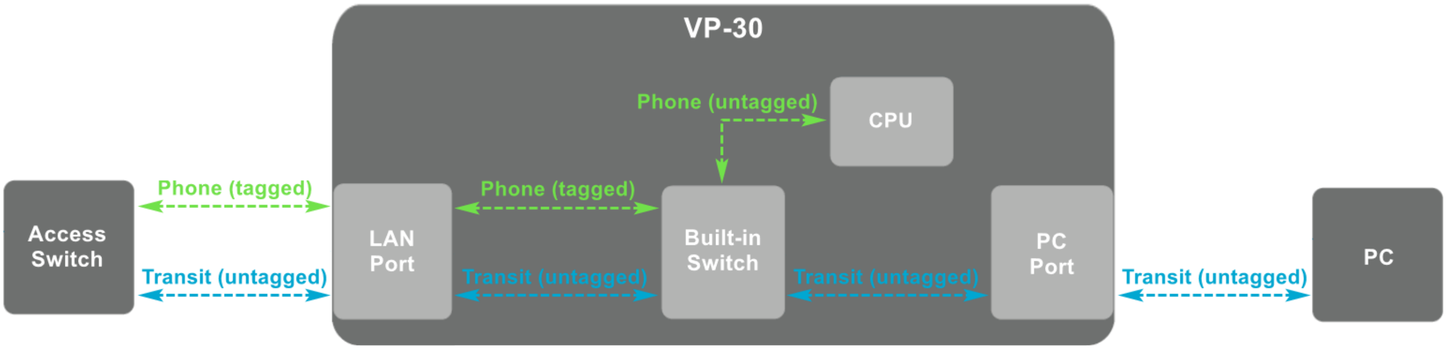
Diagrams of network configuration options are shown below. The color shows the distribution of traffic by ports if different traffic comes to the ingress. If the traffic is not divided by purpose, then it is shown in one color in the diagram.

1. Diagram without using VLAN for Phone and Transit traffic:



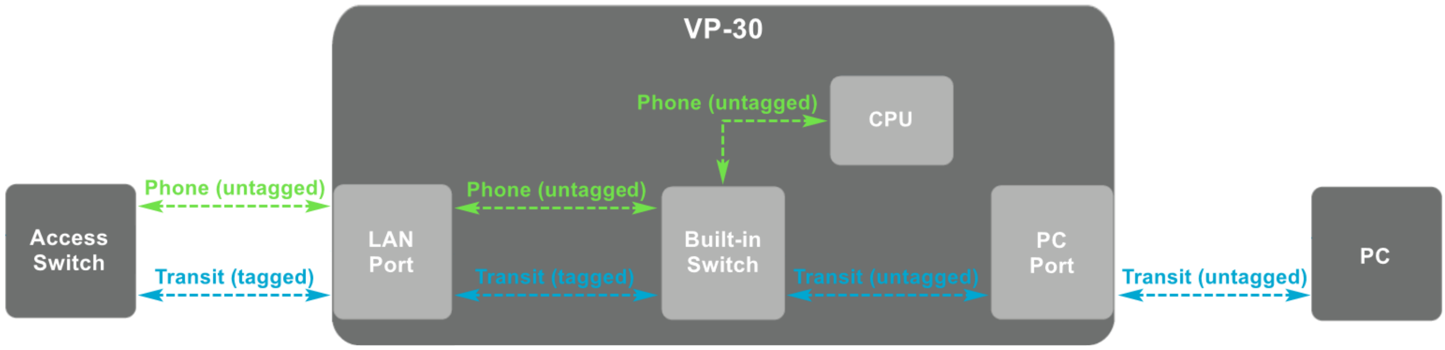
In this case, the network is configured using DHCP or Static with VLAN disabled for Phone and Transit traffic. All traffic (shown as 'Phone + Transit' in the diagram) passing through the device will be untagged.

2. Diagram of using VLAN only for Phone traffic



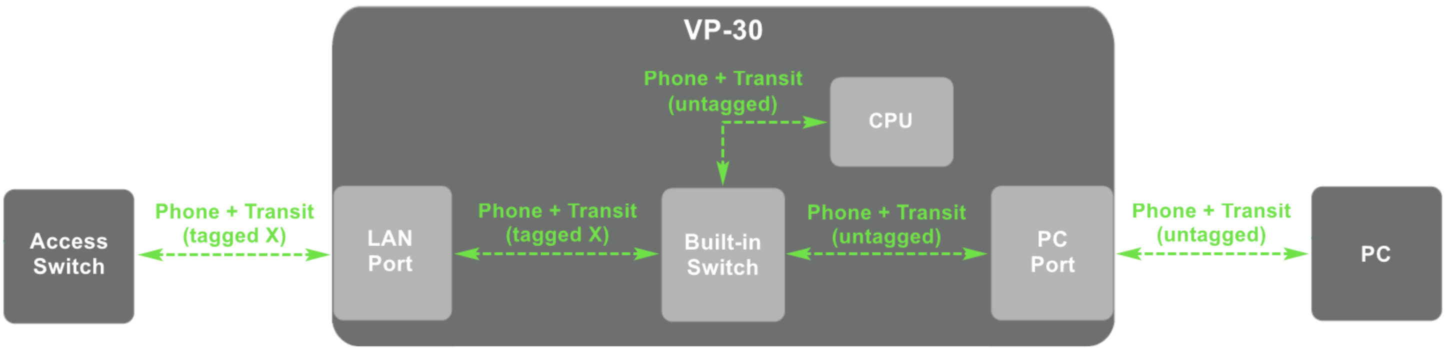
In this case, the network is configured using DHCP or Static protocol with VLAN configured for Phone traffic. VLAN is disabled for Transit traffic. Traffic intended for the phone will be tagged. Transit traffic will be untagged.

3. Diagram of using VLAN only for Transit traffic



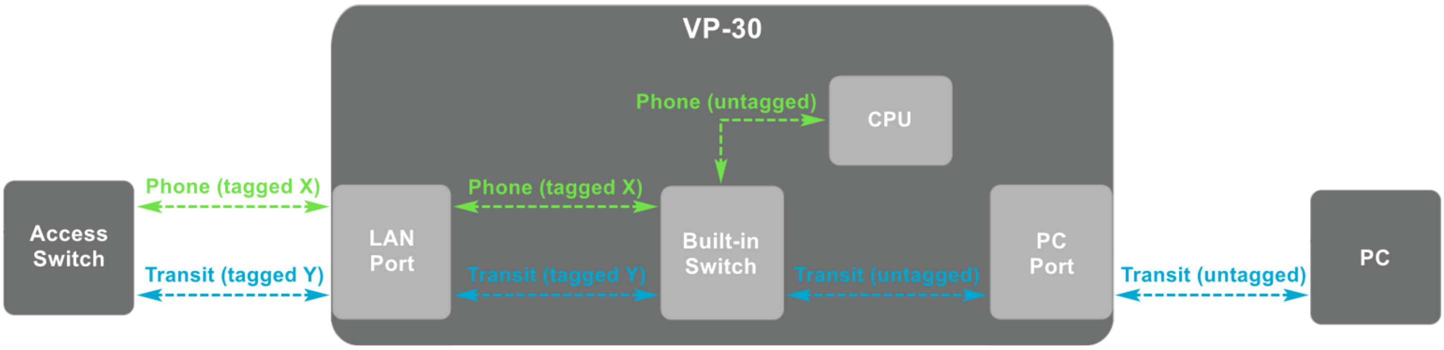
In this case, the network is configured using DHCP or Static protocol with VLAN disabled for Phone traffic. VLAN is enabled for Transit traffic. Traffic intended for the phone will be untagged. Transit traffic at the ingress and egress of the LAN port will be tagged, and at the ingress and egress of the PC port it will be untagged.

4. Diagram of using the same VLAN ID for Phone and Transit traffic



In this case, the network is configured using DHCP or Static protocol with VLAN enabled for Phone and Transit traffic with the same VLAN ID. Traffic intended for the phone will be tagged. Transit traffic at the ingress and egress of the LAN port will be tagged with the same VLAN ID as Phone traffic, and at the ingress and egress of the PC port it will be untagged.

5. Diagram of using different VLAN ID for Phone and Transit traffic



In this case, the network is configured using DHCP or Static protocol with VLAN enabled for Phone and Transit traffic with different VLAN ID. Traffic intended for the phone will be tagged with one VLAN ID. Transit traffic at the ingress and egress of the LAN port will be tagged with another VLAN ID, and at the ingress and egress of the PC port it will be untagged.

TECHNICAL SUPPORT

For technical assistance in issues related to handling Eltex Ltd. equipment, please, address to Service Center of the company:

https://eltex-co.com/support/

You are welcome to visit Eltex official website to get the relevant technical documentation and software, to use our knowledge base or consult a Service Center Specialist.

https://eltex-co.com/

https://eltex-co.com/support/downloads/