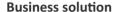


Integrated networking solutions

- Power over Ethernet (PoE) for VP-12P
- 2 SIP accounts
- Autoprovision (TR-069, DHCP)
- Headset connection
- Big LCD display

VP-12, VP-12P – up-to-date IP phones with integrated switch, which provides PC and IP phone connection through one physical line. The VP-12P phone supports Power over Ethernet technology.



VP-12, VP-12P are designed for companies with high quality voice data transmission requirements. The IP phones also provide reliability of connections and usability.

High quality of sound

The high performance of VP-12, VP-12P phones is provided by the hardware based on advanced Realtek chip. All the main audio codecs used in VoIP networks (G.711, G.722, G.723, G.726, G.729) are supported by the VP-12, VP-12P phones. Echo cancellation, Silence Detector, Comfort Noise Generator, DTMF signals reception and generation as well as traffic prioritization (QoS) ensure high quality of voice data.

Redundancy

In case of main Softswitch connection failure, VP-12 and VP-12P are switched automatically to a redundant SIP server with main server state control.

Eltex.ACS management system (TR-069)

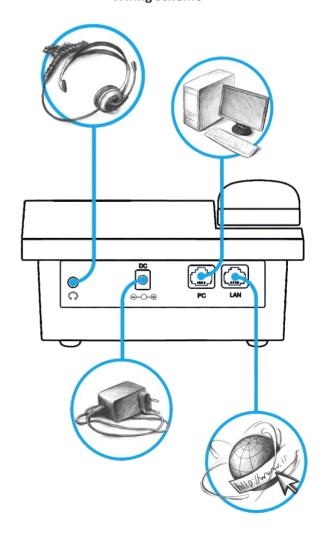
Auto Configuration system Eltex.ACS based on TR-069 specification provides easy-to-use functionality for firmware versions management and phone configuration. Eltex.ACS loads automatically individual configuration to the device according to the subscriber credentials. Intellectual firmware updating function works consistently with devices on the network, updating device groups according to the settled queue. Eltex.ACS has NorthBound Interface for OSS integration.

Usability

Convenient key location, intuitive menu and user-friendly management web interface with multilingual support provides usability in corporate telephone networks.



Wiring scheme



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Features and capabilities

Phone functions

- 2 SIP accounts configuring independently
- Geographical redundancy of a SIP server (support for up to 4 SIP redundant servers)
- Flexible dialplan
- Operation without a SIP server
- Caller name and number displaying (Caller ID)
- Mute
- Redial
- Different ring-tones for different accounts
- Call history
- Local phonebook for 200 phone numbers
- LDAP Remote Phonebook
- Speakerphone mode
- Short text messages transmitting and receiving (SIP MESSAGE)

Value Added services

- Call Hold
- Call Transfer
- Call Waiting
- Call Forward on Busy
- Call forward on no response
- Call forward unconditional
- Do not Disturb mode (DND)
- Caller Line Identification Restriction (CLIR)
- 3 Way-conference
- Hotline/warmline

Display and indicators

- 3.2 inch backlit monochrome display, 128x64 resolution
- Interface language selection (English and Russian)
- Account registration status indicator
- Functional keys: message (with the light indicator), do not disturb, transfer, conference, call hold, headset, mute, redial, speakerphone, volume + and -
- Programmable keys¹
- 2 SIP lines keys with status indicators

VoIP protocol

- SIP

Audio codec support

- G.711 a-law, μ-law
- -G.723.1
- -G.726
- G.729

Voice standards

- Voice activity detection (VAD)
- Comfort Noise Generation (CNG)
- Echo Cancellation, G.165, G.168 (AEC)

Monitoring and management

- Management web interface (Russian and English versions)
- SSH
- Telnet
- Autoprovision (TR-069, DHCP)

Diagnostics

- Device status monitoring via the web interface
- Displaying of debugging information in Syslog, Telnet

Network protocols and security

- Data network connection: Static, DHCP, PPPoE
- Time and date synchronization via NTP
- 802.1p, DSCP traffic marking
- Support for NAT traversal: STUN mode, Public IP
- Support for SIP over TLS
- Support for SRTP

Specifications

- RFC 3261 SIP 2.0
- RFC 3262 SIP PRACK
- RFC 4566 Session Description Protocol (SDP)
- RFC 3263 Locating SIP servers for DNS lookup SRV and A records
- RFC 3264 SDP Offer/Answer Model
- RFC 3311 SIP Update
- RFC 3515 SIP REFER
- RFC 3891 SIP Replaces Header
- RFC 3892 SIP Referred-By Mechanism
- RFC 4028 SIP Session Timer
- RFC 2976 SIP INFO Method
- RFC 2833 RTP Payload for DTMF Digits, Flash event
- RFC 3108 Attributes ecan and silenceSupp in SDP
- RFC 4579 SIP Call Control Conferencing for User Agents
- RFC 3361 DHCP Option 120
- RFC 3550 RTP A Transport Protocol for Real-Time Applications
- RFC 3611 RTP Control Protocol Extended Reports (RTCP XR)

Technical features

- Realtek 8972C processor
- SDRAM 128 MB
- SPI Flash 16 MB
- Linux software
- Software redundancy

Physical features

- 2xRJ45 10/100Mbps Ethernet ports
- 1xRJ9 (4P4C) for handset connection
- 1xJack 3.5 mm or 1xRJ49 (4P4C) for hands-free connection²
- Power over Ethernet technology (for VP-12P)
- Power consumption: no more than 3.5 W (maximum current consumption 0.7 A)
- Power adapter: 5 VDC, 2A
- Operating temperature: from +5°C to +40°C
- Operating relative humidity at +25°C: up to 80%
- Desktop case
- Dimensions: 223x178x89,5 mm
- Weight: no more than 0,52 kg

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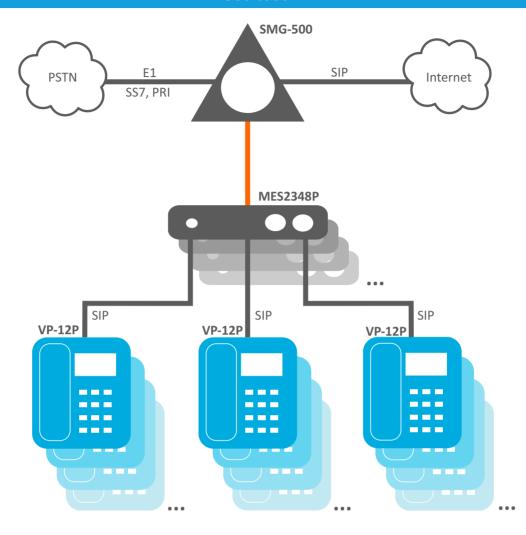
¹ Not available in the firmware version 1.5.0

²Depends on the hardware version



Integrated networking solutions

Use case



Name	Description	Image
VP-12	IP phone VP-12: 2 SIP accounts, 2x100M, LCD display	
VP-12P	IP phone VP-12P: 2 SIP accounts, 2x100M, LCD display, PoE	
Jointing products		
ACS-CPE-256	Option ACS-CPE-256 of Eltex.ACS system for Eltex CPE auto configuration: 256 devices	ACS
ACS-CPE-512	Option ACS-CPE-512 of Eltex.ACS system for Eltex CPE auto configuration: 512 devices	ACS ACS
ACS-CPE-1024	Option ACS-CPE-1024 of Eltex.ACS system Eltex CPE auto configuration: 1024 devices	ACS

Contact Us About EltexAlatau







EltexAlatau company is one of the first communication equipment manufacturers in Kazakhstan established in 2012. The main focus of the enterprise is a set of solutions and the opportunity of their seamless connection to the customer's infrastructure.