- Power over Ethernet (PoE)
- 2 SIP accounts
- Gigabit Ethernet ports
- High quality of sound
- Programmable keys
- Autoprovision (DHCP)
- Headset connection
- Large LCD
- Modern design

VP-17P — up-to-date IP phone with integrated gigabit switch, which provides PC and IP phone with connection via single physical line. The device offers advanced functionality, PoE support, high quality and modern design.

Business Solution

VP-17P is designed for companies with high requirements to reliability, usability and quality of voice data transmission.

High Quality of Sound

All the main audio codecs used in VoIP networks (G.711 (a-law, μ-law), G.726-24, G.726-32, G.729) are supported by VP-17P. Echo cancellation, Silence Detector, 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-17P is switched automatically to a redundant SIP server with main server state control.

AutoProvision

AutoProvision (AP) subsystem is used for automatic configuration, update and state monitoring of subscriber's equipment. The subsystem allows transparent replacing of a phone with a new one (any vendor). After replacing a phone, the configuration will be adapted automatically.

Elph Desktop

Elph Desktop application is designed for phone services extension. The application allows answering a call, arranging calls via click-to-call mechanism, viewing call history by searching and filtering according to subscriber's name, phone number, etc.

Usability

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

Features and capabilities

Phone features
– 2 SIP accounts configured independently
– Support for BLF
– Programmable keys
– Flexible dialplan
– Displaying of caller name and number (CallerID)
– Mute
– Redial
– Different ringtones for accounts
– Opportunity to upload ringtones
– Hotline
– Support for speakerphone
– Support for headset

Additional services
– Call Hold
– Call Transfer
– Call Waiting
– Call Forward on Busy (CFB)
– Call Forward on No Response (CFNR)
– Call Forward Unconditional (CFU)
– Do Not Disturb mode (DND)
– Caller Line Identification Restriction (CLIR)
– Local 3-Way Conference
– Support for remote conference according to RFC4579
– Hotline/Warmline
– Answering an intercom call
– Automatic Call Answer
– Call Pickup

Display, indicators and keys
– 3.2 inch (81 mm) backlit monochrome display with 128×64 resolution
– Functional keys with LED indication: message, headset, microphone mute, speakerphone
– Functional keys without LED indication: conference, call hold, call transfer, redial, volume adjustment
– 6 programmable keys

VoIP protocol
– SIP

Audio codec support
– Codecs: G.711 a-law, μ-law, G.726-24, G.726-32, G.729
– DTMF: In-band, RFC2833, SIP INFO
– Voice Activity Detection (VAD)
– Acoustic Echo Suppression (AES)
– Comfort Noise Generation (CNG)

Phonebook
– Local phonebook for 1000 phone numbers
– XML and LDAP remote phonebooks
– LDAP Remote Phonebook
– Searching in phonebooks
– Call history: dialed, received, missed, and forwarded calls

Integration with IP PBX
– Geographical redundancy of SIP server (support for up to 3 redundant servers)
– Operation without SIP server
– Busy Lamp Field (BLF)

Diagnostics
– Monitoring of device status via web interface
– Displaying of debugging information in Syslog

Monitoring and management
– Management in web interface
– SSH
– Telnet
– Autoprovision (DHCP)
– Factory reset, reboot

Network protocols and security
– Data network connection (Static, DHCP, No IP)
– Time and date synchronization via NTP
– QoS 802.1p, DSCP traffic marking
– Support for NAT-traversal: Public IP
– DHCP VLAN
– Support for SIP over TLS
– Support for LLDP MED
– Support for SRTP

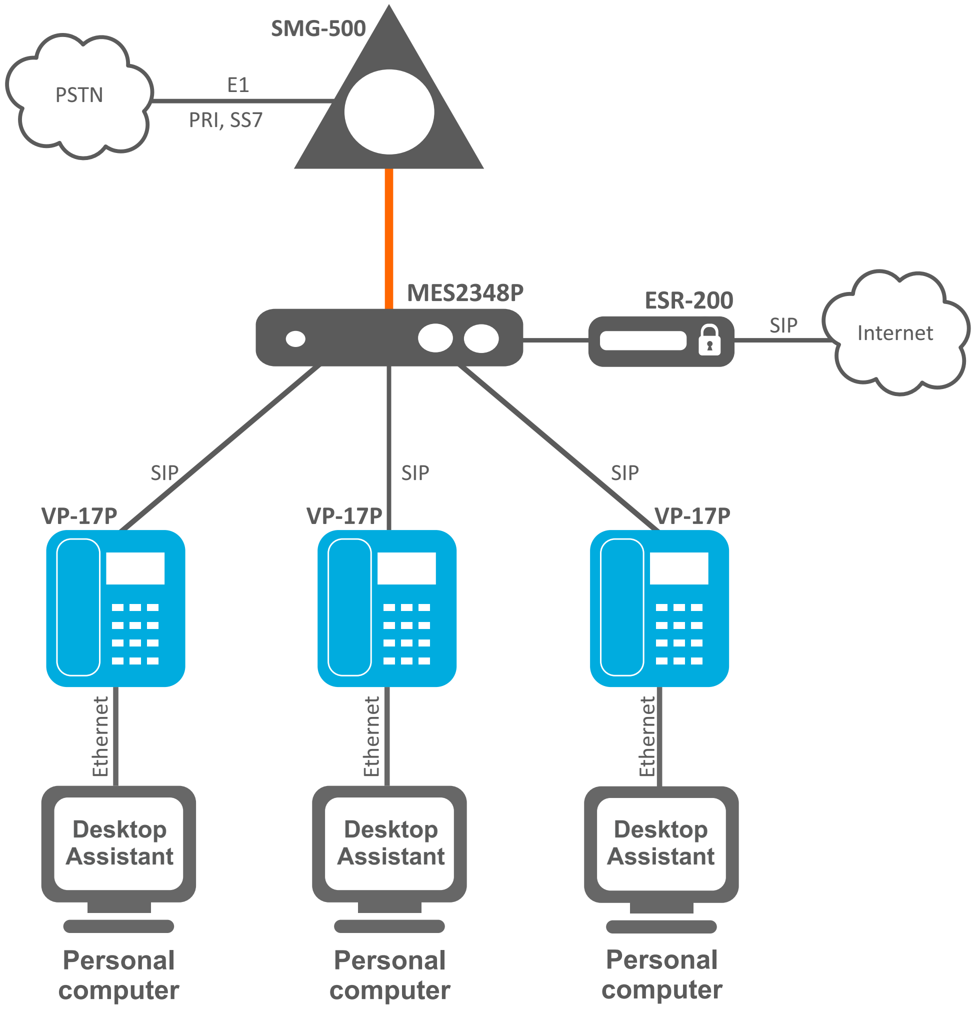
Specifications
– RFC 3261 SIP 2.0
– RFC 3262 SIP PRACK
– RFC 4566 Session Description Protocol (SDP)
– RFC 3263 Locating SIP servers for DNS lookup SRV and A records
– RFC 3264 SDP Offer/Answer Model
– RFC 3311 SIP Update
– RFC 3515 SIP REFER
– RFC 3891 SIP Replaces Header
– RFC 3892 SIP Referred-By 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 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
– RFC 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)

Technical features
– SDRAM 512 MB
– SPI Flash 256 MB
– OC Linux 4.4
– Software redundancy

Physical features
– 2 × RJ-45 10/100/1000 Mbps Ethernet ports
– 1 × RJ-9 (4P4C) for handset connection
– 1 × RJ-9 (4P4C) for headset connection
– Support for PoE 802.3af
– Power consumption: no more than 4 W (maximum current consumption 0.8 A)
– Power adapter: 5 V DC, 2 A (optional)
– Operating temperature: from +5 °C to +40 °C
– Relative humidity at 25 °C: up to 80%
– Desktop implementation
– Dimensions: 205 × 210 × 86 mm
– Weight: 0.83 kg

Integrated Networking Solutions

Use case



Ordering information

Name	Description
VP-17P	VP-17P IP phone: 2 SIP accounts, 2 × 10/100/1000BASE-T (RJ-45) ports, LCD, PoE
Related products	
ECSS-AP-10	Option ECSS-AP-10 for 10 devices auto-configuration
ECSS-AP-50	Option ECSS-AP-50 for 50 devices auto-configuration
ECSS-AP-100	Option ECSS-AP-100 for 100 devices auto-configuration

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About ELTEX

ELTEX Enterprise is a leading Russian developer and manufacturer of communication equipment with 30 years of history. Complete solutions and their seamless integrability into the Customer's infrastructure are the priority growth areas of the company.