– **Power over Ethernet (PoE) for VP-15P**

– **2 SIP accounts**

– **Headset connection**

– **Large LCD**

VP-15(P) are up-to-date IP phones with an integrated gigabit switch, which provides a PC and IP phone with a connection via a single physical line. The VP-15P phone supports Power over Ethernet technology.

Business Solution

VP-15, VP-15P are 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, G.723.1, G.726, G.729) are supported by the VP-15, VP-15P phones. 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-15 and VP-15P are switched automatically to a redundant SIP server with main server state control.

AutoProvision

AutoProvision (AP) subsystem is used for centralized 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

The Elph Desktop application is designed for phone services extension. The application allows answering a call, arranging calls via a 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 management web interface with multilingual support provide usability in corporate telephone networks.

ELTEX Integrated Networking Solutions

Features and capabilities

Phone features
– 2 SIP accounts configured independently
– Redundancy of SIP server (up to 4 SIP redundant servers)
– Flexible dialplan
– Operation without SIP server
– Caller name and number displaying (Caller ID)
– Mute
– Redial
– Different ringtones for accounts
– Possibility to upload ringtones to the phone
– Call history
– Local phonebook for 200 phone numbers
– LDAP Remote Phonebook
– Speakerphone mode
– Short text messages receiving (SIP MESSAGE)
– Voice mail counters viewing
– Message Waiting Indicator (MWI)
– Remote phonebook
– Displaying of watched subscriber line status (BLF)

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 (DND)
– Caller Line Identification Restriction (CLIR)
– VAS code transfer from the phone
– Synchronizing VAS with a remote server
– Local 3-Way conference
– Support for remote conference according to RFC4579
– Hotline/Warmline
– Answering an intercom call
– Automatic Call Answer
– Call Pickup
– Remote Call Control
– Remote Ring service for issuing a custom Ringtone to a phone from Softswitch (over RTP stream)

Display, indicators and keys
– 3.2 inch (81 mm) backlit monochrome display, 128×64 resolution
– Interface language selection (English or Russian)
– 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
– G.711 a-law, μ-law
– G.723.1
– G.726
– G.729

Voice standards
– Voice Activity Detection (VAD)
– Acoustic Echo Suppression (AES)
– DTMF signal reception and generation
– Microphone adjustable gain control (AGC)

Diagnostics
– Device status monitoring via web interface
– Display of debugging information in Syslog, Telnet

Monitoring and management
– Management web interface (Russian and English versions)
– SSH
– Telnet
– AutoProvision automatic configuration (Static, DHCP)

Network protocols and security
– Data network connection: Static, DHCP
– Time and date synchronization via NTP
– QoS 802.1p, DSCP traffic marking
– Support for 802.1X
– Support for NAT traversal: STUN mode, Public IP
– Support for LLDP, LLDP MED
– Support for SIP over TLS
– Support for SRTP
– LDAP over TLS

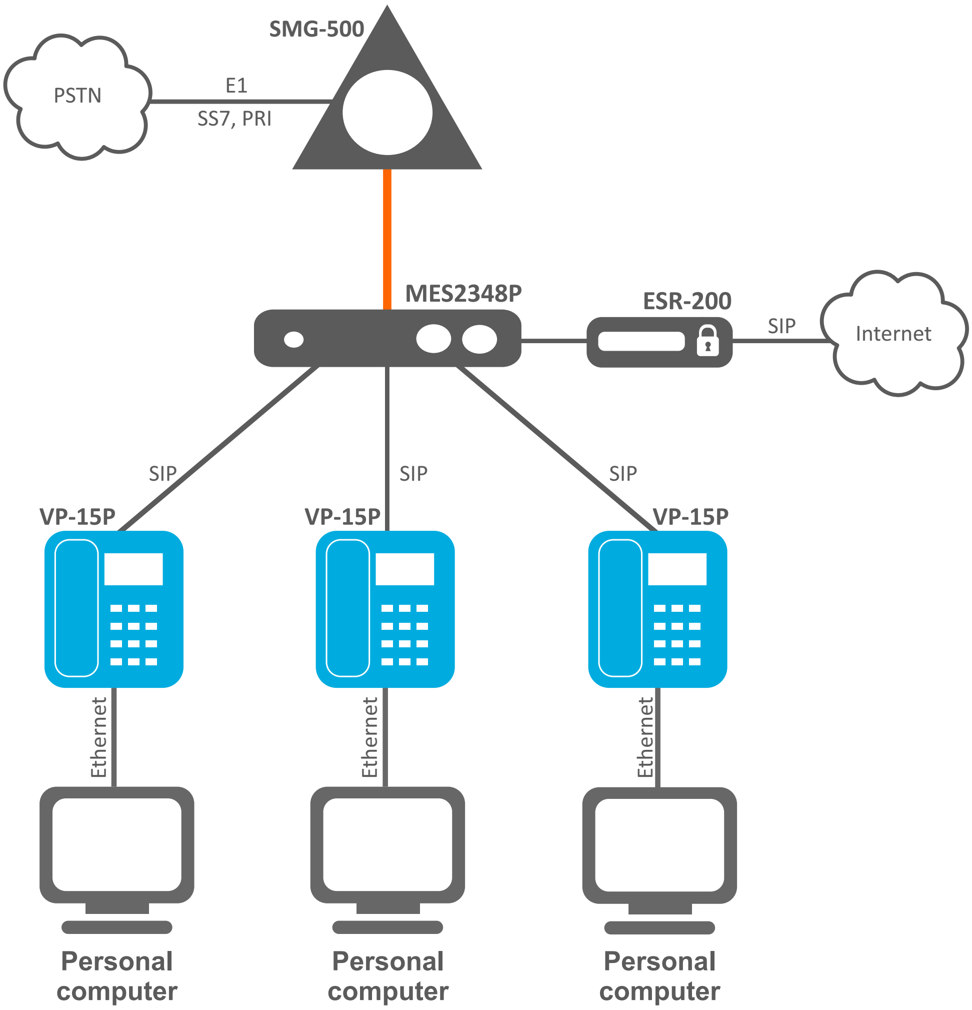
Specifications
– RFC 3261 SIP 2.0
– RFC 3262 SIP PRACK
– RFC 4566 Session Description Protocol (SDP)
– RFC 3263 Locating SIP servers for DNS lookup SRV and A records
– RFC 3264 SDP Offer/Answer Model
– RFC 3311 SIP Update
– RFC 3515 SIP REFER
– RFC 3891 SIP Replaces Header
– RFC 3892 SIP Referred-By 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)
– RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initial Protocol (SIP)
– RFC 4235 An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
– Broadsoft: SIP Access Side Extensions Interface
– DTMF "RFC2833+SIP INFO"
– Support for P-Remote-Ring header (Remote Ring)

Technical features
– SDRAM 128 MB
– SPI Flash 16 MB
– Linux OS
– Software redundancy

Physical features
– 2 × (RJ-45) 10/100 Mbps Ethernet ports
– 1 × RJ-9 (4P4C) port for handset connection
– 1 × RJ-9 (4P4C) port for hands-free connection
– Support for PoE 802.3af technology (for VP-15P) (power class 2)
– Maximum power consumption: 4 W (maximum input current consumption 0.8 A)
– Power adapter: 5 V DC, 2 A (optional)
– Operating temperature: from 0 to +40 °C
– Relative humidity at +25 °C: up to 80 %
– Desktop implementation
– Dimensions (W × H × D): 205 × 86 × 210 mm
– Weight: no more than 0.83 kg

ELTEX
Integrated Networking Solutions

Use case



Ordering information

Name	Description
VP-15	IP phone VP-15: 2 SIP accounts, 2 × 10/100BASE-T (RJ-45) ports, LCD
VP-15P	IP phone VP-15P: 2 SIP accounts, 2 × 10/100BASE-T (RJ-45) ports, LCD, PoE
Related products	
ECSS-AP-10	ECSS-AP-10 option for 10 subscriber devices auto-configuration
ECSS-AP-50	ECSS-AP510 option for 50 subscriber devices auto-configuration
ECSS-AP-100	ECSS-AP-100 option for 100 subscriber devices auto-configuration

Contact us

+7 (383) 274 10 01
+7 (383) 274 48 48

eltex@eltex-co.ru

www.eltex-co.com

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.